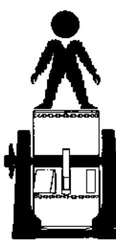
Acoustics Bulletin

October 1986 Volume 11 Number 4

Special Feature

MUSICAL ACOUSTICS

INSTITUTE OF ACOUSTICS



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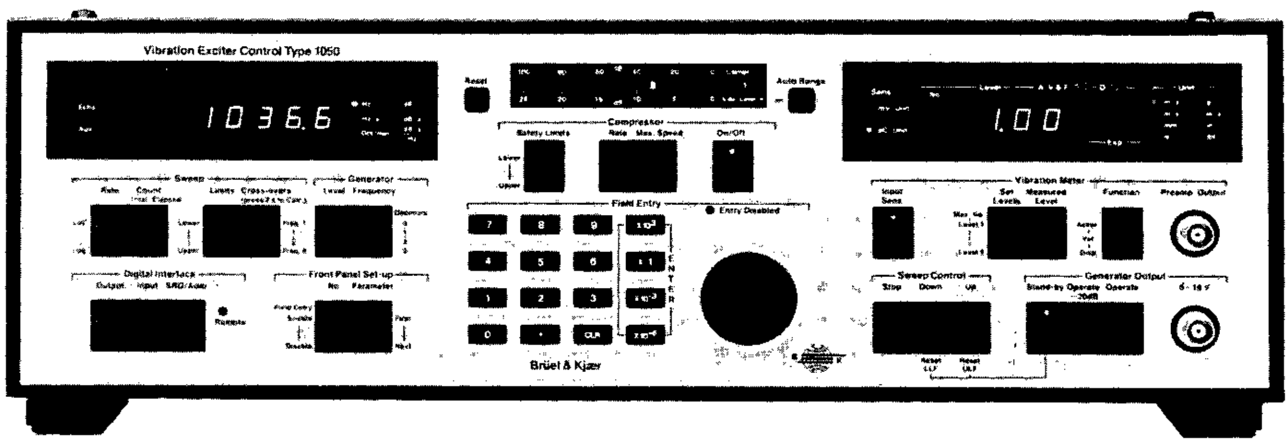

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The Institute of Acoustics was formed in 1974 by the amalgamation of the Acoustics Group of the Institute of Physics and the British Acoustical Society and is now the largest organisation in the United Kingdom concerned with acoustics. The present membership is in excess of one thousand and since the beginning of 1977 it is a fully professional Institute.

The Institute has representation in practically all the major research, educational, planning and industrial establishments covering all aspects of acoustics including aerodynamic noise, environmental acoustics, architectural acoustics, audiology, building acoustics, hearing, electroacoustics, infrasonics, ultrasonics, noise, physical acoustics, speech, transportation noise, underwater acoustics and vibration.

Membership of The Institute of Acoustics

Membership of the Institute is generally open to all individuals concerned with the study or application of acoustics. There are two main categories of membership, Corporate and Non-corporate. Corporate Membership (Honorary Fellow, Fellow, Member) confers the right to attend and vote at all Institute General Meetings and to stand for election to Council: it also confers recognition of high professional standing. A brief outline of the various membership grades is given below.

Honorary Fellow (HonFIOA)

Honorary Fellowship of the Institute is conferred by Council on distinguished persons intimately connected with acoustics whom it specially desires to honour.

Fellow (FIOA)

Candidates for election to Fellow shall normally have attained the age of 35 years, have had at least seven years of responsible work in acoustics or its application, and have made a significant contribution to the science or profession of acoustics.

Member (MIOA)

Candidates for election to Member shall normally have attained the age of 25 years, must either (a) have obtained a degree or diploma acceptable to Council and have had experience of at least three years of responsible work in acoustics, or (b) possess an equivalent knowledge of acoustics and cognate subjects, have had experience for not less than seven years of responsible work in acoustics or its application, and must have been a Non-corporate member of the Institute in the class of Associate for not less than three years.

Associate

Candidates for election to the class of Associate shall have attained the age of 18 years and (a) be a graduate in acoustics or a discipline approved by Council, or (b) be a technician in a branch of acoustics approved by Council, or (c) be engaged or interested in acoustics or a related discipline.

Student

Candidates for election to the class of Student shall have attained the age of 16 years and at the time of application be a bona-fide student in acoustics or in a related subject to which acoustics forms an integral part. Normally a student shall cease to be a Student at the end of the year in which he attains the age of 25 years or after five years in the class of Student, whichever is the earlier.

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Dear Fellow Member,

On returning from a rather long (enforced) absence, I found an ever-so-gentle reminder from our Editor that a 'letter' from me was expected like yesterday — at least to prove that I was still around! Anyone who has had to concoct regular progress reports will appreciate my immediate reaction.

And yet, on looking through the July '86 issue of the Bulletin one can identify a number of points, some pleasing and some causing concern. I would like to highlight the issues raised by two of our Vice-Presidents, because both (in their own different ways) emphasize the significance of participation by membership in the affairs of the Institute.

Mike Ankers (p 11) is unable to hide his glee at the 'dozens of offers (from members) to arrange meetings and/or to present specific papers'.

Geoff Kerry (p 33), on the other hand, is sending out a 'cry for help' for volunteers to participate in forming a full committee for what is the largest specialist group of the IOA, the Industrial Noise Group. As we all know, 'industrial noise' covers many significant aspects of acoustics and (with membership of well over 500) one hopes that this is only a temporary situation.

I think we ought to remind ourselves from time-to-time that the IOA is the second biggest learned society in Acoustics, after the Acoustical Society of America. We can only continue to function satisfactorily, organizing interesting professional meetings and providing a forum for the dissemination of knowledge in our own fields by working at it.

This letter reads like a sermon — apologies. I hope we all have an interesting year of activities.

Yours sincerely,

Orhan Berktay

Appreciation

Christopher Hans Hodges

With the death of Chris Hodges at the untimely age of 44, the UK acoustics community has lost a major figure, at a time when his full influence was just becoming apparent.

Chris came late to acoustics, as he had followed the doctorate which rounded off an outstanding student career at Cambridge by more than a decade of research in aspects of solid state physics in the USA, Canada and France. Returning to the rather different Cambridge of the new wave of science-based consultancy companies, and as a modest, unassuming man older than most of his more extrovert new colleagues, he nevertheless rapidly earned their great respect and affection on both a technical and a personal level.

Characteristically undervaluing his own contribution, he claimed that his initial work in acoustics simply drew analogies from the mathematics of imperfect crystals to that of vibrations of almost periodic structures. Soon, however, the traffic became two-way, and his elegant modelling of quantum diffusion using the ideas of acoustics alerted his erstwhile colleagues to the merits of inter-disciplinary contact from which we were already deriving such benefits. He continued to foster this cross-fertilization, most notably by a symposium at Cambridge drawing together experts in long range propagation in the oceans and atmosphere, and in solid state, as well as in the acoustics of complex fluid-loaded structures — a topic which was by now absorbing most of his energies.

The measure of Chris's success in this field lies not in the count of papers and limited-issue internal reports he and his co-workers produced, nor directly in the length or scope of the joint review article with Jim Woodhouse (whose publication in Reports of Progress in Physics was imminent at the time of his death). Rather it lies in the fact that the newcomer, surveying that paper, its vitality, depth and breadth, would have no consciousness of the blank wall of empiricism and ignorance that constituted our total knowledge a decade ago. Though Chris still insisted that his steps were small compared with those of the giants of the past, even he conceded that they were enough, and of sufficient purpose, to have travelled a very significant distance.

Though devoted to his work, Chris was never oblivious of its wider context, or of the recreational side of acoustics. He loved music, and especially choral works, though sadly the increasing breathlessness of his illness made him only a listener towards the end. He felt deeply the need to work out for himself the ethical implication of his studies — unable to accept the absolute rejection of any military or defence connection advocated by some people he respected — but clear in his own mind of the dangers of anodyne reassurances. He listened, then made his own decisions, with the same overwhelming concern for integrity he brought to all his life.

His friends and colleagues hope to mark that life by establishing an endowment fund in his memory at the Cavendish Laboratory. Anyone wishing to contribute is asked to contact Professor J E Ffowcs Williams at the University Engineering Department, Cambridge.□

I Roebuck

New Elections

At its meeting on 29 May 1986 Council approved the following elections to corporate and non-corporate membership.

Fellow

R I Damper
R C Hill

Member

L A Beirne	G A V Greenhough	N H Kitchin	G Russell
P Economou	E Hands	R Limb	P G Simpkin
C S Elliott	D M Howard	A S Munro	J T Wade
S D Fitzgerald	A P Hussey	C Phillips	M R Warwo
C M Frier	F Irving	B L Ritchie	A R Woolf
G P Goodwin	R Irwin		

Associate

S Ashton	E Davies	N J Hoyle	L Poultney
P T Bassett	J K Davies	H A Hull	M E Powell
J L Batchelor	M C C Donaldson	G P Huppler	J M Prince
D H Bisson	P Ford	S A Jackson	G A J Probert
D R Blyth	R M Ford	M Jones	J R Pyke
C M Bootle	A E Francis	C Joslin	S Rock
N J Bourne	R A Gibson	I C Keagle	A Rupkus
P Bourne	F C Goodall	J Kettlewell	M S Siblock
G C Bouttell	B J Griffiths	P Knox	S P Smith
R P Brailsford	C J Grimwood	D A Lyon	T J Smith
K A Broughton	A A F Gumbley	P McDermott	C R Stagg
M D Brown	D A Gunn	J McGowan	M G Stephenson
D L Burdon	J H Hamilton	A Moneylaws	M A Stokes
R D Burnett	A Hargreaves	D R Moore	N D Taylor
S N Chandler-Wilde	I C Harris	R Muirhead	D G B Thomas
I D Chapman	N R Hartley	D R Natolie	D I Torrance
J Chapman	K M Harvey	G Norman	E Towers
P Clayton	M D Hills	A C Parker	A G Tull
N J Collins	S Hodgson	B Platts	A F Turton
N D Cooper	S J Holmes	J C Polden	E J Walker
P R Cumberlidge	S G Houldcroft	M J Porter	R J Woodcock
E A Cutler			

Student

D W Anderson	M R Lester	D J Munden	R M Windle
M J Henery	R J McKinnell	H F Routh	R D Wright
S T Lee	A T Moorhouse		

Noise Council Conference

The recently launched Noise Council is to organize its first conference in London on 29 January 1987. The meeting will be on the theme of **Noise at Work — what kind of law do we want?** and will concentrate on the interpretation of the EEC requirements, with representatives from all sides of industry giving their views on the implications of new legislation to a similar representative group of delegates. Topics will include: Review of the nature of the problem; EEC and legal background; 'As far as is reasonably practicable'; Employers' viewpoint; Employees' viewpoint.

Lord Elliott will introduce the meeting, the organization of which has been undertaken by the Meetings Secretariat of IOSH.□

Workshop on Sound Propagation in Forested Areas and Shelterbelts

The Eco-acoustics research group in the Laboratory of Experimental Plant Ecology of the Catholic University of Nijmegen, Netherlands, organized an international workshop from March 3-6, 1986. Sixteen Dutch and foreign scientists presented papers and another twelve attendees were involved in the subsequent discussions. The workshop was structured around five sessions.

Introductory Session

Professor Linskens (Botany Dept, Nijmegen) recalled, in his opening address, how the classical articles of Eyring (1948) and Embleton (1963) focused the attention of the Nijmegen botanists on the theme of sound propagation in the environment. Dr Embleton (Nat Res Counc, Ottawa, Canada) presented a general overview of current understanding of theories of outdoor sound propagation. He discussed the influence of ground surface, meteorological conditions such as turbulence and refraction conditions producing 'creeping waves', and landscape on the sound field. Dr Martens (Botany Dept, Nijmegen) reviewed the acoustical research of the Eco-acoustics group in forests and other environments. He described measurements of ground surface impedance, of the influence of the vegetation structure and species composition, and of the vibration characteristics of plant organs. He also drew attention to research on natural sound sources and receivers in vegetation, like songbirds. Dr Johannesma (Biophysics Dept, Nijmegen) discussed the development of a method for complete signal analysis and representation based on the Coherent Spectro-temporal Intensity Density of the acoustical or electrical signal. This function has two arguments (frequency and time) and assumes complex values (amplitude and phase); it can be represented as a two-dimensional chromatic image, the photochrome. This photochrome supplies a base for understanding localization and identification of signals (eg sound) by an animal. For several of the workshop participants and attendees who were interested in this method an *ad hoc* visit and a demonstration in the Biophysics laboratory were arranged.

Sound Propagation and Ground Effect

Dr Attenborough (Open University, Milton Keynes, UK) reviewed the different ground models. A four-parameter model of a rigid porous medium based on flow resistivity, porosity, pore shape and tortuosity, and a ground model of a soft soil layer backed by a rigid layer may be used to describe the acoustical characteristics of forest floors. The effect at surface roughness was modelled as a change in the effective surface impedance. Dr Lee (Michigan Techn University, Houghton, USA) explained the impulse measuring technique developed to investigate acoustical properties of snow. He showed that snow often resembles a layered soil floor. Mrs Hess (Open University, Milton Keynes, UK) showed the use of acoustic measurements in studying soil characteristics. The method was shown to be useful in deducing the air-porosity of soils which is an important agricultural property for plant growth. Mr Wempen (Department of Physics, University of Oldenburg, FRG) demonstrated the results of a newly developed method for measuring soil surface impedance and admittance. The results of actual outdoor measurements and theoretical predictions based upon a thin layer ground model were shown to be in close agreement.

Sound Propagation in Forested Areas

Dr Fricke (Sydney University, Australia) presented a great variety of measured data concerning the influence of meteorological conditions on long range sound propagation in different situations and in different vegetations. The enormous spread in the data causes difficulties in interpretation. On the other hand, gross correlations could be established between sound attenuation rates and relative humidity. In her paper, Mrs Price (Open University, Milton Keynes, UK) explored the benefits and shortcomings of models including both ground effects and scattering to explain measured sound attenuation data in different forests. Dr Jacobs (Agricultural University, Wageningen, The Netherlands) presented a fundamental experimental study on air flow and turbulence around a (sound abating) screen. The results of this study may be useful to interpret sound propagation in some actual field situations. It is likely that a combined meteorological and acoustical experiment will offer a better understanding of sound propagation in relation to flow and turbulence. Presenting the results of a literature survey, Mr Ringheim (Voss, Norway) showed that there are still deficiencies in our knowledge of traffic noise attenuation by forest shelterbelts, since most models and measurements are restricted to sources and receivers close to the ground. As a result of the subsequent discussion, literature references on high-elevation measurements were exchanged.

Demonstrations at the Faculty of Sciences

The workshop participants and attendees visited the Botanical Garden of the Faculty of Sciences. There, the construction and the acoustic features of different types of willow-woven sound abating screens were explained and discussed. During a visit to the acoustics laboratory, Mrs Geveling (Botany Department, Nijmegen) demonstrated the laser-Doppler-vibrometer-scanning equipment that is used to study sound induced vibrations in plant leaves. Drs Huisman (Botany Department, Nijmegen) showed the current state of development of a ray tracing program that accounts for meteorological and ground effects. Possible ways of improving the model by simulating turbulences were discussed.

In the evening a discussion was held on several topics. The usefulness of the different experimental methods to measure acoustical properties of soil surfaces were discussed. Furthermore several ideas were developed of how to tackle the lack of accurate experimental results on sound transmission in the complex forest environment in relation to species composition and meteorological conditions. With regard to animal communication, we discussed the problem of sound detection by different animal species and how to carry out precise measurements.

Animal Communication in Relation to the Acoustical Environment

In the last session of the workshop Dr Brown (University of Missouri, Columbia USA) presented experimental results concerning the acoustics of three East African primate habitats. He showed the relationship between these results and intraspecific communication behaviour at short and long distances. He also demonstrated the morphological adaptations and perceptive adaptations of the studied species. Drs Reijnen (Inst Nature Res, Leersum, Netherlands) presented a study of woodland breeding bird populations in forest habitats at different distances from motorways. A relationship between breeding population density and distance to the road could be shown. He discussed the possible impact of traffic noise, since the breeding bird populations were negatively correlated with traffic noise levels.

In the last paper Mr Foppen (Botany Department, Nijmegen) reviewed the literature on long distance bird communication in relation to the acoustical environment of different habitats, forests, woodlands and open fields. He presented some hypotheses concerning the relation between the acoustical environment and bird song characteristics, which should be validated or negated by future experimental research. In the final discussion it turned out that a number of Dr Brown's experimental results supported the hypotheses stated by Foppen. A main conclusion of the final discussion was that much more experimental research is needed to develop a more profound understanding of sound propagation on the one hand and animal communication in relation to sound propagation on the other hand.

Demonstration of Willow-woven Sound Screens

The workshop included a coachtrip to the neighbourhood of Utrecht and Sliedrecht where it was possible to see the actual use of willow-woven screens in sound abatement and their adaptation to the landscape as an eco-hygienic element. In a plant-nursery near Sliedrecht some other plant-steel plate combinations were shown.

The workshop was held at Groesbeek in the eastern hilly part of The Netherlands, which despite the cool weather, turned out to be a most hospitable and pleasant venue. Financial subsidies were provided by the Dutch Foundation for Pure Scientific Research (ZWO), The Hague, the Environmental Science Branch of the Department of the US Army Research Development and Standards Group, London, and Mostert & de Winter Groenvoorzieningen, Sliedrecht.

Full proceedings of the workshop including the discussions are in preparation. Orders should be sent to Dr M J M Martens, Department of Experimental Plant Ecology, Group of Eco-acoustics, Faculty of Sciences, University, Nl-6525 ED Nijmegen, The Netherlands (price: Dfl 35).□

M J M Martens
K Attenborough

IOA Awards 1987

Professor M R Schroeder of the University of Göttingen is to be the Rayleigh Medallist for 1987 and will present his lecture at the IOA's Spring Conference in Portsmouth. The 1987 R W B Stephens Lecture will be given by Prof Dr Hab A S Sliwinski, of the University of Gdansk, also at Acoustics 87 in Portsmouth.

Honorary Fellowship of The Institute of Acoustics will be presented to James Moir at the meeting on Reproduced Sound in November 86 at Windermere.□

Noise Legislation — Its Effectiveness for Noise Control

The new Council's first publication, a 16-page booklet, reviews the successes and failures of nearly two decades of noise control legislation and draws attention to current legislation problems. It is available from Mrs Cathy Mackenzie, Institute of Acoustics, 25 Chambers Street, Edinburgh EH1 1HU, price £3.25 inclusive of postage and packing.□

Musical Acoustics

What is the use of Musical Acoustics? This question may well be asked at a time when the value of research is measured in kilopounds per SERC grant. There has been a long history of research into the operation of musical instruments and the perception of musical sounds. However, as Professor Taylor points out in his article, the tendency has often been for the scientist to reduce a musical problem to an absurdly abstract level. On the other hand, the musician — and especially the instrument maker — has often been suspicious of the scientist's activity; occasionally a wall of pseudo-scientific theorizing has been erected to protect the musician's domain. In any case, the practising musician is probably right to echo the words of Descartes: 'Am I to refuse my dinner because I do not understand digestion?'

Undoubtedly many musical problems have seemed deceptively simple — elementary text-books give plausible explanations of the action of organ pipes and vibrating strings. How many, on the other hand, can attempt a realistic treatment of the physics of the excitation mechanism at the mouth of an organ pipe or the subtleties of the bowing mechanism in the violin? In most cases a detailed treatment (with attendant mathematical sophistication) is necessary to arrive at anything near a reasonable physical explanation of what the practising musician experiences daily.

The ear-brain system of the hearer can process musical sounds in ways that reveal the slightest nuances of pitch, loudness and tone-quality. If musical acoustics is to have a practical application, for example in modifications to an instrument or the design of a concert hall, it must take account of the fine detail which the human listener is able to detect and which makes or mars his enjoyment. Fortunately, in some ways, the state of knowledge is becoming equal to this task although there is always the anti-scientific doubter who will claim that nothing significant will ever be achieved. (Is it only *his* ears, un-deafened by science, that hear the purest sounds unsullied by the tamperings of scientists?)

This group of articles has no pretensions to being a complete state-of-the-art presentation of current ideas in musical acoustics. It consists of a series of sketches of the interests of a few workers in the subject in Britain (or maybe those whom I could persuade to contribute!) Inevitably our kind of work is done by enthusiasts in ones and twos — many of whom are professionally involved in other branches of physical science. However, it is gratifying to note that in recent years there has been a renewed interest in musical problems. The physics of traditional instruments contains many obscurities as does the perception of musical sounds. The creating of new sounds through the resources of electronics is an expanding area of activity. In these areas there is much to be achieved in the pursuit of basic knowledge. In addition there is a growing realization that scientific investigation can point the way to desirable changes in such matters as musical instrument design. Some enlightened players and instrument makers find it worthwhile to ask our advice as a way of clearing a way through the jungle of anecdotes and old wives' tales surrounding their activities. We can reasonably claim that there *is* a use for musical acoustics — and we may make a few honest kilopence into the bargain!

Edgar Brown

Musical Acoustics — A Historical Perspective

CHARLES TAYLOR

Former Professor of Physics, University College Cardiff

It is popularly believed that the Science of Musical Acoustics began with Pythagoras who, it is alleged by some authors, invented the monochord and had clear notions about harmony and discord in terms of the perfection of the mathematical ratios of the frequencies involved. Unfortunately he, or perhaps we should say 'they', left no written record and we have to rely on the claims by the 'Pythagoreans', a long line of disciples, who succeeded in so muddying the water that it is not even clear how many individuals might have made up the collective personality of Pythagoras. But during classical Greek and Roman times the work was confined largely to philosophical speculation with little basis in experiment and, while it is possible to find some quite plausible statements about the nature of sound, there exist also many quite unacceptable notions. Before moving on to more recent times it is perhaps worth mentioning that even 2250 years ago when mathematicians were arguing about the detailed theory of the Pythagorean system of harmony and discord, a pupil of Aristotle called Aristoxenes, who was a musician, was claiming that only the ear could judge whether an interval was consonant or dissonant. It is a great pity that some physicists of the first half of the present century did not pay more attention to that precept! It might have saved many pointless arguments.

AS IN so many other branches of science, it was Galileo who first questioned the philosophical ideas about music that came down from the classical period and he did his famous experiments on pendula and resonance, inspired by watching the swing of heavy lamps and candelabra in Church. Galileo and his contemporary Mersenne both performed experiments on the behaviour of vibrating strings and published accounts in the Seventeenth Century and could therefore be deemed to have a better claim than Pythagoras to be the instigators of research in musical acoustics. But the real flowering began in the Eighteenth Century and, as D C Miller writes in his *Anecdotal History of the Science of Sound* there occurred in 1722 an important event that is rarely mentioned in scientific treatises. . . the publication of the first volume of the '48' by J S Bach. This established the equal tempered scale and Bach himself said: 'Music is the greatest of all Sciences'. Tartini drew attention to the phenomenon we would now call combination tones, Chladni devised his well known and beautiful experiment using sand figures to demonstrate the modes of vibration of a plate and Euler, D'Alembert, Bernoulli and Lagrange all published works on the theoretical aspects of vibrations in strings and pipes.

In parallel with these scientific developments were exciting developments in musical instruments. The organ began to assume a form that made Bach's Preludes and Fugues playable, the marvellous transition from viol to violin pioneered by the Cremona school came to full flower and attempts to standardize and rationalize the woodwinds began in this period, to be brought to perfection by Boehm in the early Nineteenth Century. But throughout this period scientists and musicians worked in more-or-less watertight compartments.

In the Nineteenth Century itself the science of acoustics, and musical acoustics in particular, occupied a great deal of attention. But still, the musical instrument makers paid little attention to the scientific developments and most of the remarkable improvements in their instruments came through superb craftsmanship and patience. One of the few scientists to be involved in the design of an instrument that has survived outside the laboratory is Wheatstone, who patented the Concertina in 1829.

Among the researchers of the Nineteenth Century that are of great significance for musical acoustics today we should perhaps pick just three or four. Thomas Young, whose famous experiments on interference of light waves were also applied to sound, pointed out that if a string is struck at the nodal point for a particular mode, then the corresponding harmonic will be absent from the tone produced. The siren was developed into a useful instrument for pitch-frequency measurements and attempts were made to arrive at a standard pitch; the telephone, the phonograph and the oscillograph all appeared in forms that we should now regard as rudimentary — but what changes they foreshadowed in our science!

By the end of the Nineteenth Century the science of Musical Acoustics was very well established and the four great heroes of the day were Rayleigh, who dominated the theoretical aspects of sound, Tyndall, the lecture-demonstrator *par excellence*, Helmholtz who established the basis for modern theories of the hearing process, and Koenig, perhaps the greatest scientific instrument maker of the pre-electronic age.

During the first half of the present century my personal choice of names that stand out are Miller, who made exhaustive studies of wave shapes and was the father of wave analysis, Sabine, who was the father of auditorium acoustics, Webster, who laid the foundations of horn theory, Alexander Wood, who as well as working on building acoustics, was one of the greatest teachers of acoustics (I was lucky enough to attend his undergraduate lectures). There is also a case to be made for the French physicist Bouasse, though because of his quarrels with his contemporaries, his farsighted investigations of the behaviour of wind instruments were not given recognition until the 1950s.

It is really quite astonishing to look back to the textbooks of the 1950s and to see how many myths were still being passed on. Few pointed out that no real mechanical musical instrument has truly harmonic overtones, though the theoretical basis of this in abstract studies of vibrations was known. It was still widely believed that it is the relative proportions of harmonics that determine the quality of a musical sound. The physics of musical instruments was still being discussed in terms of the steady state wave forms of single long notes and the contribution of transients and overall envelope shapes to the recognition of differences in tone between instruments was rarely mentioned. There was still a great deal of suspicion between instrument makers and physicists and few makers were prepared to accept that there could be a scientific contribution to their art. The extraordinarily complex behaviour of the ear-brain system in sound recognition was only just beginning to be recognized.

It is sometimes claimed that it was the arrival of digital computing and 'chip' technology, which of course has revolutionized the analysis of musical sounds, that was responsible for the explosive increase in understanding of musical phenomena during the last thirty years or so. While not wanting to detract from its contribution, I would claim that the real breakthrough came when physicists stopped testing instruments in anechoic chambers with artificial playing mechanisms producing single notes, and started to try to apply their science to instruments in the hands of real players and in natural surroundings. It is frequently said that when one begins an investigation it is necessary to simplify the problem by making all kinds of assumptions; but it is equally important to examine the solution in the light of those assumptions and to take into account all the complexities of the real situation. For some reason many of the workers in the 20s and 30s failed this second condition.

It was also necessary to build bridges of confidence between makers and physicists. Among many workers who have contributed to this process are Carleen Hutchins, whose skill as a maker of violins has enabled her to develop new scientific testing procedures that are really helpful to lutiers and Art Benade, whose skill as a player and a modifier of clarinets, as well as his great knowledge of theoretical physics has brought enormous practical benefits to clarinet players. Both of these workers have also published books that provide excellent reference material to the present state of the art.

So what is the present position? Although research on musical instruments is still carried out mainly by enthusiastic individuals who often have major interests in other fields of physics, it is being recognized more and more by grant-giving bodies, not least because there are often fall-out benefits in other branches of science and engineering; this is particularly true of some of the computer modelling techniques that are being used to study instrument behaviour. The membership of the Catgut Society, which consists of makers and players of stringed instruments as well as scientists in the musical acoustics field, now exceeds 1000 world wide and there is a steady stream of papers published in its twice-yearly journal. Meetings organized to study relatively specialized aspects of instrument making are beginning to draw a mixed population of players, makers and scientists, as for example a recent meeting at a College in the South West of England on varnishes for stringed instruments.

I confidently predict that Musical Acoustics, having regained some of the respect of the musical community that was lost through a kind of short-sighted arrogance that once bedevilled it, will go from strength to strength and, as leisure activity becomes an increasingly important factor in Society, will more than justify the modest funds expended on it.

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Don't Shoot the Piano Tuner

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Oscar Wilde was well aware of the problems faced by the pianist; it is a pity that little is usually said in defence of the piano tuner. For a start, it is impossible to define a 'well-tuned' piano without specifying the requirements of the music, the pianist and the acoustic environment and even (since perception of the pitch of low-fundamental-frequency complex tones varies from person to person) each listener. The statement in a recent New Scientist article that 'pianos sound better when they are slightly out of tune' begs the question, is not 'sounding best' a definition of being 'in tune'? But despite the phycisist's idea that pitch corresponds quite closely to frequency and the musician's view that musical scales consist of defined pitch relationships, being 'in tune' is rather more complicated to define — and more difficult to achieve — than a specified sequence of frequencies.

PYTHAGORAS, we are told, when not preoccupied by right-angled triangles, made the happy discovery that subjectively consonant musical intervals are produced by whole-number ratios in the lengths of otherwise identical plucked strings; that, as we know, means whole-number ratios in frequency. Thus, consonant intervals are defined by the frequency ratios:

2:1	octave
3:2	perfect fifth
4:3	perfect fourth
5:4	pure major third
6:5	pure minor third

Helmholtz explained this observation in terms of consonance produced by matching harmonics of the fundamentals, while fundamental frequencies in less tidy relationships produce beating or roughness between mismatched harmonics and are perceived as dissonance. Clearly, of course, there is more to it than that, or the average human might have less tendency to sing flat! However, the aim of a musical scale is to offer the musician a set of notes, frequencies, which give consonant intervals from which melody and harmony may be built.

The problem is that the sizes of the consonant intervals are not consistent with one another. The harmony of the spheres does not work, and Pythagoras' dismay can be imagined. Seven octaves (frequency ratio $2^7 = 128$), despite what may be deduced from a piano keyboard, is not the same as twelve perfect fifths ($3/2^{12} = 129.7$); the difference is known as the Pythagorean Comma, but can be little consolation to Pythagoras. The fifth and the fourth together make a correct octave ($3/2 \times 4/3 = 2/1$) and the major and minor thirds together make a correct fifth ($5/4 \times 6/5 = 3/2$), but inconsistencies exist between two octaves and a third ($2 \times 2 \times 5/4 = 5 = 80/16$) and four fifths ($3/2^4 = 81/16$) — the Comma of Didymus; between three major thirds ($5/4^3 = 125/64$) and an octave (128/64) — the Lesser Diesis; and between four minor thirds ($6/5^4 = 1296/625$) and an octave (1250/625) — the Greater Diesis. The extensive history of the various tunings of the standard diatonic major scale is concerned with ways of optimizing the tuning of inconsistent intervals.

Conventionally, the piano is supposedly tuned in 'equal temperament'; the ultimate compromise which abandons the attempt to make any interval but the octave exactly correct, and divides the octave into twelve equal semitones. Thus the semitone interval is given by a frequency ratio $\sqrt[12]{2} = 1.0594631$ and the fifth, for example, is $1.0594631^7 = 1.498$ instead of 1.5. No interval is exactly correct, but (unlike other forms of tuning) every key is tuned the same. This may be fine for the piano, although its practicability demonstrates the shortcomings of Helmholtz's theory; but stringed instruments such as violins tune their strings to perfect fourths and fifths, singers are said to (try to) tune all their intervals pure — and the tuning of wind instruments (because of the compromises involved between making an air column vibrate at given frequencies and making that vibration stable with near harmonic components) rarely conforms to any calculated scale — so conflict is inevitable.

The harmonics are used to tune an interval. The lower and upper notes of a perfect fifth (in frequency ratio 3/2) have third and second harmonics which coincide in frequency and so do not beat together (Helmholtz's criterion for consonance); when the desired interval is governed by a different frequency ratio, such as the equal tempered fifth of 1.498, the harmonics do not match exactly but will beat at a calculable rate. Therefore, if the correct beat rate is set between the harmonics of two notes, there is a fifty-fifty chance that the notes form the correct interval; it is necessary to observe whether the rate increases or decreases as the pitches are changed, to ensure that one has the correct interval of the two possibilities.

So, in principle, the task of the tuner is to set calculated beat rates between the various notes of a scale. Simple, but of course not easy; the skill of listening to the correct frequencies is laboriously acquired by piano tuners but defeats many musicians, and the accuracy with which beat rates can be judged is rather less than that needed to ensure that intervals are consistent with one another — so tuning systems incorporate cross-checks to minimize cumulative error.

Arriving at the basic scheme, the problems are just beginning. The principles of beat rate calculation, as well as being difficult to implement accurately, are themselves based on inaccuracies. Piano strings, especially in the bass, are far from ideally flexible and uniform and are not adequately described by the one-dimensional wave equation; upper notes have the additional complication of being provided by two or three strings. The stiffness of thick strings (heavily overwound to provide low frequencies without excessive length) introduces significant fourth-order bending wave behaviour, with modes of vibration which are not exact harmonics but higher in frequency. This inharmonicity means that an octave which is 'correct' on the criterion that the upper note matches the frequency of the second mode of vibration of the lower involves more than doubling frequency; thus, piano octaves are 'stretched'. Negative inharmonicity can also occur, if a string has a kink or other discontinuity. So the piano is not really tuned to equal temperament, but has a rather large basic octave; beat rate calculations are aiming at the wrong frequencies. Nor are beat rate calculations correct, since they assume that the partials present are harmonic. Piano tuning is an inaccurate implementation of incorrect calculations. Nor is it possible to correct the calculations significantly; inharmonicity varies from string to string, never mind from piano to piano. Some relief is provided by Earle Kent[1], who finds that in practice the problems of defining either the scale or the means or tuning it are overcome by introducing a different criterion for 'good' tuning, that for any interval its beat rate should progress smoothly across the compass of the keyboard.

Further problems are introduced with multiple stringing. These can be referred to, misleadingly, as unisons. It is (of course) hard to tune these as exact unisons (and how exact is exact? — though there is evidence that the coupling of modes produces unison); it is impossible to maintain unisons for any length of time. The variation in stress on the piano frame as a string is tuned has an effect on the other strings (so tuning must not disturb the status quo too much — overall pitch changes must be done gradually) and soon after tuning, atmospheric fluctuations in temperature and humidity affect the 'unisons'. The one often-cited piece of research (Kirk,[2]) on the tuning of unisons reports that listeners (all with some musical training) prefer tunings with 1 to 2 cents spread among the strings of a unison group (a cent is 1/100th of a semitone) to exact unisons (or to large mistunings!). Whether pianos are tuned like this because they sound 'good', or the preference exists because that is the way pianos inevitably are, is a classic candidate for a chicken-and-egg debate. Most tuners staunchly maintain that they should — even that they do — tune unisons exactly; Kirk states that 'artist tuners' (apparently mythical beasts, the mention of whom provokes derision in most tuners) tune to the preferred deviations from unison. Some explanation for this preference might lie in the characteristic 'double decay' pattern of piano sound; an initial 'prompt sound' with rapid decay marks each note, but sustain or 'aftersound' is provided by the subsequent slower decay. Weinreich[3] demonstrates that this is provided by the existence of two polarizations of vibration in the string, which couple differently to the widely different horizontal and vertical impedances of the bridge and soundboard; in multiple stringing, the coupling of string modes of vibration which are close in frequency contributes to this effect.

The tuning of a piano has rather less to do with achieving a specified set of frequencies than the theory of scales (or the advocates of electronic tuners) might imply. The factors which determine perception of pitch receive considerable investigation; the perception of 'correct' tuning, the interaction of pitches, is still more complex. The piano tuner cannot please everyone, but is surely worthy of respect in what is achieved.

In the end, is it all in the service of music? A large London choir rehearses with a piano which happens to be cared for by one of its tenors, who has a day job with Steinway. Its 'equal tempered' major thirds should be around 400 cents — a little more, allowing for inharmonicity; pure major thirds (ratio 5/4) are 386 cents. The conductor is rarely satisfied with the choir's tuning. 'Those thirds. . .not like the piano. . .I want them brighter' (musicians' term for sharper). So the piano is not right; but shouldn't pure thirds be flatter? We are far from understanding it all yet; but the tuner does his best!

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A Divertimento for Woodwind

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Where do you put the Tone-Holes?

We all know that the tone-holes in a woodwind instrument change the tube length so that it may play the notes of a chromatic scale. Are there 'correct' positions for the holes? The recorder family demonstrates that the holes can be placed reasonably conveniently for the hands on the different sized instruments while still maintaining the correct tuning. There is, in fact, a trade-off between hole location and both hole diameter and depth; practical instrument makers have known this as an article of faith without any need for a physical explanation. With Boehm's re-design of the flute in the early nineteenth century, the idea of the 'correct' positioning of the tone-holes gained some ground.

A LARGE open hole, of a diameter comparable with that of the tube on which it is placed, effectively *does* cut the tube short at that place and the placing of the hole may almost conform to the text-book fraction of the resonant wavelength. Of the instruments with tone-holes, only the (modern) flute and the saxophone pay any sort of lip-service to this scheme. On all the other instruments, the length corresponding to the internal standing wave extends well beyond the first open tone-hole; this may be of modest size and possibly be drilled through quite a thickness of wood. An extreme case of this occurs in the bassoon where drilling long oblique holes allows the fingers to cover some of them directly. The large size of the bassoon also ensures that the holes are far from their acoustically 'correct' positions in the above sense. (The present-day bassoon is a mixture of 'ancient and modern' in this respect.)

Richardson (1929) first treated the tube with side holes as an acoustical transmission line. At the location of an open side hole, it and the continuing portion of the tube act as a parallel combination of acoustic impedances terminating the nearer portion of the line. Continuing this process back to the mouthpiece for all holes that are open reveals the input acoustic impedance there; the resonance frequencies occur when this is purely real and large (for a reed) or small (for a flute embouchure). These calculations are straightforward but tedious and Nederveen (1969) has described a simplification of the method which can be used for practical design work.

Is there *no* special placing for the tone-holes? The answer to this lies in the tone quality desired. Hague first suggested that a series of open holes at the further end of a tube could, in conjunction with the intervening tube pieces, act as a high-pass filter. Benade (1960) has pursued this idea with great fruitfulness and has worked out many consequences of the 'tone-holes lattice cut-off frequency'. Components of the internally-generated sound spectrum lying above this frequency propagate through the portion of the tube containing the open holes and are radiated by them. Components below the cutoff are evanescent in this region; they are effectively limited to the portion of the tube nearer the mouthpiece where they combine to produce a stable oscillation at the playing frequency. The tone quality of the note heard is thus influenced by the location of the cutoff frequency. Since the filter characteristics are governed by the shunt inductance (tone-hole diameter and depth) and series capacitance (tube diameter and hole spacing), the holes must be chosen to give the desired cutoff frequency *as well as* the correct tuning of the scale. The failure to fulfil this double requirement led to the demise of many attempts to 'rationalise' woodwinds in the nineteenth century. Only the flute succeeded because, for it, musical taste was ready to accept the change in tone quality that went with the louder sound emerging from Boehm's large tone-holes — not to mention the rationalized technique for the player.

The Action of a Reed

The reed, either single as in the clarinet and saxophone, or double as in the oboe and bassoon, has been considered as a pressure-operated valve. A succinct account of its small-amplitude motion has been given by Backus (1965) with respect to the clarinet reed. The reed must admit air when the pressure at the reed end of the instrument is at a maximum. Consequently the reed must be a stiffness-controlled vibrator and hence closed and open. A positive-going pressure step sent out from the reed returns after a double transit of the tube as a negative-going one and closes the reed. The resulting pressure reduction returns after a second double transit as a pressure increase and reopens the reed. The fundamental wavelength of the standing wave thus corresponds to four times the tube length and the pressure signal contains a preponderance of odd harmonics — all in conformity with the simple theory of the clarinet.

In our laboratory we decided to have a look at the pressure signal inside a conical instrument (first of all a bassoon), together with the reed motion. We observed that the pressure signal from the (missing) vertex. We found that the conical tube coupled to the reed cavity formed a kind of parallel tuned circuit whose oscillations were triggered by the reed closure (Figure 2). Only half a cycle of oscillation was able to take place before the rising pressure reopened the reed, damping any further oscillation. The 'inductance' part of the circuit depends on the cone angle, the 'capacitance' on the reed volume. For a given cone input diameter, the period of oscillation (the pulse length) should fall as the cone angle increases. This was observed by comparing the bassoon (cone angle 0.014 rad) with a heckelphone (similar to an oboe, cone angle 0.025 rad) and with various 'home-made' conical tubes all blown with the same reed.

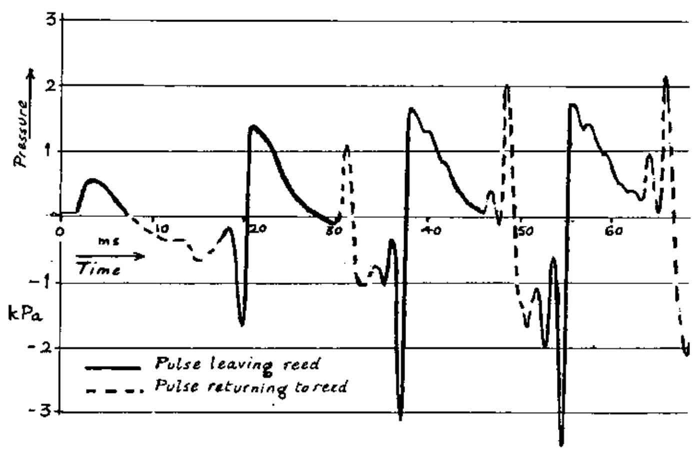

Figure 1 Pressure waveform at start of note Bb_1 (58.3 Hz) Point of observation: 660 mm from reed, tube length: 2511 mm

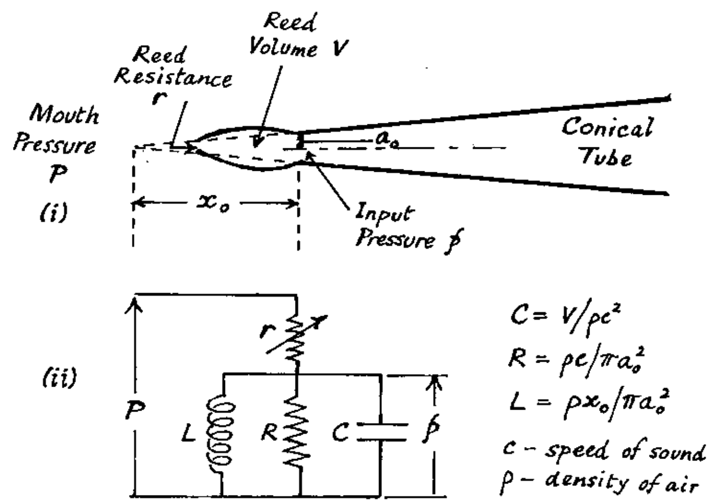

Figure 2 (i) Reed coupled to conical tube (ii) Equivalent circuit

operate below its resonant frequency. Other subtleties are explainable by small-amplitude theory; for instance, reed damping introduces a delay between internal pressure and reed opening. It is as though the reed is attached to a slightly longer tube than it actually is and the played note is lower than the passive tube resonance. Benade (1970) has shown that a nonlinear relationship between internal pressure and air flow through the reed is essential in explaining the stabilization of playing level and the varying harmonic generation at different levels. Also, air input at the fundamental frequency can be influenced by mouthpiece pressure at harmonic frequencies. Hence a good instrument will be 'well aligned' — its passive resonances will closely approximate to a harmonic series. McIntyre and his colleagues (1983) have examined the vibration of a clarinet reed in the time domain. They note that, even at moderate playing levels, the reed motion is far from sinusoidal, being closer to a square-wave alternation between contained sharp pulses (Figure 1) whose width was not significantly affected by the tube length appropriate to the note being played although, of course, the pulse repetition rate was so affected. Observing the pressure signal at various points allowed us to unscramble the pulse going out from the reed from that returning towards it. We found that the emitted pulses were negative-going and an inspection of the reed motion showed that, for a low note, the reed stood open for a large part of the cycle. It then closed (producing the negative pressure pulse) and reopened after a short time. The emitted pulse returned from the open end of the tube as a positive-going pulse, but its trailing edge went sufficiently negative to trigger the reed closure once more. Hence the periodic time was a double transit of the tube plus the pulse length. What caused the reed to reopen and after how long? Clearly some mechanism in the vicinity of the reed is responsible. This may be understood if it is noted that the wave in a cone is a spherical wave diverging Many effects of tuning of the conical woodwinds together with their overblowing properties and tonal spectra have been explained in this way. The exact details of pulse length control do, of course, involve the shape of the reflected 'trigger' pulse and hence the nature of pulse reflection at the outer end of the tube containing the lattice of open holes. Work is continuing on this subject.

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Perceptual and Signal-processing Approaches to Separating Speech from Noise

Meeting of the Speech Group held at the University of Nottingham on Friday 25th July 1986

On the Intelligibility of Speech in Noise

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A facility has been established at the Royal Signals and Radar Establishment for conducting speech intelligibility tests using the Diagnostic Rhyme Test (DRT). A permanent listening panel has been recruited who participate in tests for three mornings a week. This arrangement has been running for 18 months and a considerable amount of data has been gathered on the perception of speech in noise. This paper will start by presenting a working definition of speech-to-noise ratio, and continue by giving intelligibility results on the use of LPC-10 in simulated Tornado cockpit noise. The effects of noise at both source and destination will be discussed.

The Performance of Temporal Structure Investigation Based Pitch Detection Algorithms under Conditions of Noise

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The performance of two time-domain pitch detection algorithms, each employing temporal structure investigation of the speech waveform in order to obtain estimates of the fundamental frequency of the speech, is examined with particular reference to their behaviour when presented with noise contaminated speech. The pitch detection algorithms comprise the previously published parallel processing method, and an improved pitch detection algorithm based on multiple feature examination of speech waveform peaks. Each pitch detection algorithm is evaluated using both real and synthetic speech at varying levels of signal-to-noise ratio (SNR). Types of noise include white, gaussian, and more realistic background noise such as air conditioning, telephones, computer keyboards, and other voices. The results of the evaluation show the multiple-feature based algorithm to be more robust under such conditions. Results also indicate reasonable bounds of SNR under which time-domain pitch detection algorithms may be expected to perform for the different types of noise.

Two-Talker Separation

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The ideal hearing aid would selectively amplify a target voice and attenuate competing voices and other sources of interfering noise. In exploring one instance of this problem, we are evaluating two algorithms for separating two voices speaking concurrently with periodic excitation at similar overall intensities. Both approaches exploit the regularity in the harmonic structure of voiced speech. The first involves attenuating the harmonics corresponding to the competing voice, via a cepstrum-like representation. The second method was derived from the procedure for harmonic selection described by Parsons (JASA, 1976, 60, 911-918). The fundamental frequency of each voice is estimated from the amplitudes and frequencies of all harmonic peaks resolvable in the composite spectrum. From this fragmentary information, the spectrum of each voice is reconstructed. The two procedures were evaluated by synthesizing five steady-state vowels with fixed fundamental frequencies and mixing them in pairs containing a low (120 Hz) and a higher-pitched vowel. Each procedure improved the intelligibility of both the higher- and the lower-pitched vowels, both for listeners with normal hearing and for listeners with moderate hearing impairments of cochlear origin, particularly when the difference in fundamentals exceeded about a quarter of a semitone. Comensurate with its greater sophistication and complexity, harmonic selection gave larger improvements in intelligibility than the cepstral method and lead to a clearer impression of there being only one talker present in the enhanced speech.

Noise-tolerance Characterization of an improved Formant Tracking Algorithm based on Pole Enhancement

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The ability to measure the centre frequencies of areas of resonance (formants) in the short-time power spectrum of speech is of paramount importance in the recognition of voiced speech sounds in a feature-extraction-based continuous speech recognition system. Additionally, the provision of a tracking algorithm, by which the loci of formants with respect to time can be estimated, yields formant transition information which helps identify phonetic features which are of short duration. Noise robustness in formant estimation is an essential attribute for recognition systems which are used in the office environment and in military applications. The novel technique presented in this paper provides a noise-robust method of extracting formant centre-frequency information from the short-time speech spectrum and consequently improves the signal-to-noise performance of the associated tracking algorithm. Formant estimation is based on modelling the vocal tract frequency response using linear prediction coding (LPC) techniques. However, the estimation of formant centre frequency in any analysis frame is greatly improved by employing off-axis spectral estimation coupled with a progressive increase in vocal tract model order, which together provide vocal tract pole enhancement. Finally, the use of a formant-weighting filter function applied within each frame aids in conferring high noise immunity to the estimation process. The pole enhancement technique is shown to offer an improvement of 10-15 dB in signal-to-noise immunity as a formant frequency estimator over conventional LPC-based spectral estimation. In its application to formant tracking, it is shown that the technique also offers improved separation of formants which tend to merge. The combination of both properties leads to a 15 dB noise immunity improvement in formant tracking consistency when compared against other LPC-based formant tracking algorithms. An additional advantage of the technique is its relative insensitivity to choice of vocal tract model order, which produces an inherently speaker-independent formant estimation algorithm.

On the Use of LMS Adaptive Filtering for the Treatment of Speech Recorded in Reverberant Environments

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Reverberant energy can seriously undermine the intelligibility of speech recorded in rooms exhibiting reverberation times in excess of 100 msec. Results are presented here, which demonstrate the efficacy of using very long length adaptive filters, up to 18,000 stages with a recording bandwidth of 5 kHz (equivalent to 1.8 sec), in reducing the damaging reverberant energy and decorrelating the effects of a room's impulse response. The theoretical and practical limitations of using this approach in 'dereverberating' speech signals are then discussed. Large amounts of reverberant speech material that constitute the Speech Intelligibility in Noise (SPIN) test have been processed with adaptive filters of varying lengths, and the results of the subsequent formal listening tests are given. Whilst no dramatic improvement in intelligibility has been observed, filter lengths up to 1024 stages do afford some meaningful enhancement. Extending the filter lengths beyond 1024 stages seems to have little advantage, with the speech recorded in a room having a 2 sec reverberation time. If the exact position of the sound source (ie a speaker's mouth) and microphone remain unaltered during a conversation, an alternative adaptive filtering scheme is proposed which is shown to render a good inverse filter of the room's impulse response. The attendant improvement in the intelligibility of the speech material processed in this manner is significant.

The Role of 'Perceptual Centres' in Separating Speech from Noise

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A Computational Model of Auditory Selective Attention

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This paper will describe a multiple channel, three-stage computational model of acoustic image interpretation. The model is based on physiological descriptions of peripheral auditory processing and on concepts derived from psychophysical studies of auditory perception. It is intended to have properties enabling sound sources to be isolated from their background and to be isolated from each other. A computational version of the model will be a contribution to effective automatic speech analysis capable of operating in noisy environments. The premise of the research is that stimulus identification cannot be achieved until the focal stimulus has been separated from other simultaneous but irrelevant effects. Its short term aim is to provide a working model of the process of fusion of auditory subcomponents into a coherent image. The model will be so constructed as to provide good quantitative predictions of human behaviour when dealing with sequences of monaural and binaural tone complexes; predictions whose origins can readily be understood in terms of the interaction of peripheral effects preceding the psychophysical decision stage.

On the Possible Integration of 'False Harmonics' into Vowel Percepts

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Vowels are often heard against a background of other periodic sounds. Exclusion of extraneous sounds, which is necessary for accurate estimation of F1, might be achieved by a 'harmonic sieve'; only components which are harmonics of F0 are accepted as 'belonging' to the vowel. However, extraneous components which are integral multiples of F0/2 may not be excluded, as the resulting series consists only of components harmonically related to F0/2. These components may act as 'false harmonics'. If a 'false F1', comprising two 'false harmonics', is added to token /3/ on the lower skirt of F1, then as its level increases the percept changes from /3/ to /i/, suggesting the inclusion of the 'false harmonics' in the F1 estimation. However, any integration must be partial, as high levels of the 'false F1' are required to reach the phoneme boundary, and low tones are often heard accompanying the perceived vowel. Furthermore, no significant fall in pitch occurs for stimuli labelled as /i/, suggesting either that the action of the 'harmonic sieve' differs for estimation of pitch and phonemic quality, or that the /3/ to /i/ conversion does not reflect an integration of the 'false F1'. At the phoneme boundary, the harmonic closest to F1 appears clearly resolvable in the excitation pattern. Hence the conversion does not reflect masking of the F1 peak which might cause the 'false F1' to become the perceived F1 by default. However, the conversion may result from a perceptual inference about those components masked beneath the 'false F1' rather than from integration. Further investigation is needed. Delaying vowel onset relative to the 'false F1' increases the level of the 'false F1' required to reach the phoneme boundary. This might be explained by adaptation or by perceptual segregation on the basis of onset of asynchrony. The increase asymptotes around 200-300 ms delays. The asymptotic level of the 'false F1' is sufficient to mask the original F1 peak and thus to become the F1 by default.

Effects of Acoustic Context of Perceived Vowel Quality

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The perception of vowel quality is influenced by the local acoustic environment through mechanisms of perceptual grouping (Darwin and Sutherland, QJEP, 1984) and adaptation (Summerfield et al, Percep and Psychophys, 1984); is it also influenced by properties of the more remote environment? If extra energy is added to a vowel at a harmonic near F1, its phonetic quality changes. The change in phonetic quality can be reversed if the added energy is perceptually grouped out from the vowel. In our experiments, vowels with added energy near to F1 are embedded in sequences of tones. If the sequence of tones has the same frequency as the harmonic, then the added tone makes little contribution to vowel quality. This effect increases as the number of tones preceding the vowel is increased, and is greater if the tone sequence continues after the vowel. A similar effect is observed if the vowel is embedded in a sequence of tones which descends in frequency (by whole tones, or by half-harmonic steps), coinciding with the vowel at the harmonic frequency. However, if the tone sequence *ascends* in frequency, then the added tone *does* alter the perceived vowel quality, indicating that the tone sequence has not captured the embedded tone. In a further experiment a single tone at various frequencies preceded the vowel by various time intervals. The experiment showed that the difference between ascending and descending series was due to the cumulative individual effects of the two tones that immediately preceded the vowel, rather than to the configuration of the series. We could find no effect attributable to the ascending and descending contour of the tone sequences.

The Perception of Inharmonic Complex Tones

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When a single low partial in a harmonic complex tone is shifted progressively in frequency from its harmonic value, the pitch of the whole complex (residue pitch) shifts with it, in a proportional manner, until the shift exceeds a certain value. Beyond that, the pitch shift is less than proportional and eventually declines. The mechanism determining the pitch of complex tones apparently rejects components which are sufficiently mistuned. A component which is sufficiently mistuned also tends to be heard as a separate component, standing out from the complex as a whole. The present paper considers whether these two perceptual effects reflect the same underlying process. Thresholds for hearing a mistuned harmonic as standing out from a complex tone increase as the duration of the stimulus decreases. The degree to which a partial can be mistuned before it makes a reduced contribution to residue pitch also increases with decreasing duration, but not to the same extent. For long-duration tones and for moderate degrees of mistuning (3-4%) the mistuned harmonic is clearly heard as a separate tone, but at the same time it can make a strong contribution to the pitch of the complex as a whole, indicating a form of 'duplex perception'. Finally, it is demonstrated that, when subjects compare the pitches of complex tones with common harmonics, they do not simply compare the pitches of individual components; rather, judgements are made on the basis of residue pitch.

Noise and Vibration in the Aircraft and Spacecraft Industries

One-day meeting jointly organized by the Industrial Noise Group and the East Midlands Branch on 30 October 1986 at Birmingham International Airport

Multi-Channel Airport Noise Monitoring

A Myles, J Ollerhead and R Lambert
Loughborough University and CEL

First-hand experience has been gained in the design, manufacture and installation of airport noise monitoring systems at several major international airports with the most recent being Birmingham International Airport.

The various technical requirements of design in achieving the required system performance are discussed with particular reference to climatic protection of the microphones and associated electronics, data transmission integrity, and display and reporting procedures.

Potential problems during the installation phase are highlighted in the light of experience gained.

Aircraft Noise: Measurement, Impact and Control

J Simson
London Scientific Services

In Great Britain, individuals who are troubled by the noise of aircraft flying over their homes are precluded from taking any legal action against the operators. As far back as 1963, Sir Alan Wilson recognised the special responsibility that this places on the aviation industry and environmental planners to keep noise nuisance under control. Much attention has been paid to the major airports, but general aviation and heliports should not be ignored as a potential source of major environmental conflict. Noise control is usually most effective and less costly if measures can be taken to prevent a nuisance occurring by establishing and developing within the 'Environmental Capacity' before communities are antagonized by uncontrolled growth or ill planned operations.

Noise monitoring within the community can provide information on noise trends which is not available from published NNI contours. Such information can help both to assess the likely environmental impact and to provide information which can assist in the day to day control of noise. However, a clear distinction must be drawn between the noise generated by individual aircraft and the noise generated by the airport operation as a whole. With the universal application of noise certification requirements aircraft noise emission levels meet exacting standards and are therefore generally well controlled at source but there is no guarantee that noise certificated aircraft will automatically prove acceptable during day to day operation. Monitoring of noise by local authorities in the Heathrow area for example shows that the NNI contours do not tell the whole story.

Birmingham International Airport Development — the Local Authority's Noise Control Considerations

L Poultney
Metropolitan Borough of Solihull

The paper includes an introduction on the Department's investigation for and issue of the Prior Consent under the Control of Pollution Act 1974 for the development. In addition, it outlines the history of complaints relating to noise from Birmingham International Airport before and after the construction of the Earth Bund.

Details are then given in relation to noise surveys undertaken by Officers of this Department on and around the development. The results of the surveys illustrate the effectiveness of the Earth Bund for controlling noise from aircraft taxiing operations at Birmingham International Airport, noise levels being at least 15 dB(A) lower at the boundary of the nearest residential property than they would be without the presence of the Bund.

The Assessment and Control of Aircraft Engine Ground Running Noise

J B Large and I H Flindell
ISVR, University of Southampton

Aircraft engine ground running noise can constitute a significant noise problem at airports with aircraft and aeroengine manufacturing, maintenance, and research and development facilities. This paper reviews the authors' current philosophy on the measurement, prediction, assessment, and control of aircraft engine ground running noise as based on recent experience at Filton Airport, Bristol, and the three major London Airports.

Predictions using a simple 11 dB per doubling of distance attenuation rate give good agreement with actual measurements at long range subject to allowance for meteorological variation, specific topography and aircraft orientation. Community noise level criteria can be set both in terms of a limit for unacceptable noise exposure and a limit for desirable noise environments. Notwithstanding these critiera some nearby long term residents appear to tolerate otherwise unacceptable noise exposure levels whereas other more distant residents may complain at much lower noise exposure levels. Not all noise control measures are equally effective.

Noise Disturbance at Night near Heathrow and Gatwick Airports

P Brooker and J B Critchley
Directorate of Research, Civil Aviation Authority

In 1981 the Government stated that it would undertake a review of the restrictions on Heathrow/Gatwick night flights when most of the noisier movements had been phased out. This paper describes research carried out in 1984 as part of that review.

The 1984 research repeated the methodology of the 1979 study for five of the noisiest areas originally surveyed near Heathrow and Gatwick airports. The social surveys, with concurrent noise exposure measurement, were designed to establish the extent and causes of sleep disturbance around both airports.

The 1984 study confirmed the main results of the 1979 study, in particular that sleep disturbance occurs frequently, even in areas where little or no aircraft noise is heard at night. It showed that people tend to attribute their disturbance to aircraft noise to a greater extent for increased aircraft noise exposure, although the total reported disturbance, ie from all causes, does not increase at the same rate. A statistically significant increase in total disturbance is generally detectable only in the noisiest areas.

The 1984 study demonstrates that the Government's night restrictions policy has resulted in reduced noise exposure and broadly reduced night disturbance.

Military Airfields: Prediction of Noise Contours and their Application to the MOD's Compensation Scheme

R J Weston and P A Wilson
Royal Air Force Institute of Community and Occupational Medicine UK

Airnoise, a mathematical model for computing military aircraft noise contours, has been developed by the NPL, to the specification of the Royal Air Force Institute of Community and Occupational Medicine. This paper outlines the capabilities of Airnoise and the noise compensation scheme to which it is applied.

Application of the model is then described using examples of compensation schemes which have been put into practice.

This part of the paper concludes with some of the practical limitations of the model and the direction in which the Institute of Community and Occupational Medicine would like the model to be developed.

A Review of the UK MOD's compensation arrangements near military airfields follows. Differences in noise levels and compensation arrangements at military and civil airfields together with the public pressure for a higher environmental standard of living, led the (UK) Ministry of Defence to review its noise compensation policy for those living near military airfields in the UK. This paper relates the matters examined in the course of the review which took place from 1983-85.

Aspects covered include the payment of noise insulation grants and compensation for reduction in property value due to increased noise; the purchase of dwellings subject to very high noise levels; the adequacy of acoustic secondary double-glazing; the validity of noise measurement methodology and techniques; weighting for night-flying; public presentation of the policy.

Main changes recommended by the review, which were announced in November 1985, were that the qualifying noise level for sound insulation grants should be reduced; special attention should be taken of maximum noise levels where a significant amount of night flying takes place; and a new post of Environmental Noise Officer should be appointed to co-ordinate all efforts within MOD to reduce the effects of noise on the community. The review also confirmed that the Leq method of noise measurement is the best index for the purpose.

The paper summarizes the work undertaken in the review and estimates the likely effect of implementing the changes in policy.

Acoustic Resonance in Turbomachinery, a Source of Blade Vibration

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M C Welsh
CSIRO, Melbourne, Australia

An experimental investigation of flow-induced acoustic resonances in a single-stage, axial compressor is described, which was conducted at the Department of Mechanical Engineering, University of Swansea. The results show a range of circumferentially propagating acoustic modes can be generated by flow induced vortex shedding from blade rows and that these can force a rotor blade response both at and far removed from the blade natural frequency.

A further experimental investigation of tandem plates in a wind tunnel, as a model of turbomachinery blade rows, is also presented which was conducted at the Division of Energy Technology Laboratory, CSIRO Melbourne, Australia. Acoustic resonances generated in the tunnel due to resonant vortex shedding was able to be suppressed by varying the axial spacing of the tandem plates.

The paper describes a proposed mechanism for the transfer of energy from a flow to an acoustic field and suggests ways in which noise generating systems could be controlled in the future.

The Vibro-acoustic Response Prediction of a large Communications Satellite using a Computer based SEA Technique

P J Williams, J P C Wong and R J Cummins
British Aerospace

For a number of years the Structural Acoustics Group at Filton have been involved in the prediction of acoustically induced vibration levels in spacecraft structures and systems. The work has been sponsored by two major spacecraft operators, in an effort to improve the test specification accuracy and hence equipment package reliability, for spacecraft subjected to the broad band random environments typical of launch system payload bays.

Recently British Aerospace have been contracted by INTELSAT to supply methods and techniques, compatible with current engineering practice, capable of predicting launch induced vibration levels on the types of structure and the structural configurations to be encountered on Intelsat's next and future generations of communications satellite.

Using Statistical Energy Analysis (SEA) techniques, similar to those originally developed under contract to the European Space Agency and in particular the SEA computer code GENSTEP2, predictions of good engineering accuracy have been obtained for analyses carried out on the INTELSAT VI Satellite soon to begin commercial operation.

Autumn Conference 1986 — Speech and Hearing

28-29 November 1986 at the Hydro Hotel, Windermere

Is the Third Formant Necessary for Vowel Normalization?

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The centre frequency of the third formant is fairly constant across different vowels produced by the same speaker. That frequency varies inversely with the length of a speaker's vocal tract. These facts have suggested that listeners normalize vowels by estimating the length of a speaker's vocal tract from his third formant centre frequency. To test this hypothesis, subjects were asked to identify two- and three-formant versions of synthetic isolated RP vowels. For a male, a female, and a child's voice, eight vowels were made in each version. Two experiments with the items presented in random orders showed no advantage in identification for the three-formant vowels. This outcome contradicts the hypothesis under test. The results of a third experiment using a blocked, cross-over design also will be reported.

Analyses of the stimuli and the data were done at the Speech Laboratory of the IBM UK Science Centre.

A study of frequency transition detection using synthetic vowel sounds

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The main purpose of this study was to obtain formant frequency transition detection thresholds. These have been estimated for the first and second formant for the vowels /a/, /3/, /i/ and /u/. Control experiments were performed both on isolated formant and on pure tone transitions at the same initial frequencies. These were carried out in order to decide whether there is any significant difference between frequency transition detection thresholds for speech and comparable non-speech stimuli. The vowels used were chosen to represent different conditions of F_1 and F_2 spacing in order to test also for any effects on threshold produced by the proximity of neighbouring formants.

Diminution of High Frequency Energy as a Cue to the Voicelessness of Following Consonants

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UCL, SPAR Group

Spectrographic analysis of speech reveals fluctuations in high frequency energy (often quite massive) which are not generally considered to be invariant properties of the segments with which they are associated. 'Reference analysis', using an electrolaryngograph signal in addition to standard spectrographic and simple waveform techniques, indicates that some of these fluctuations may be associated with rule-governed variations in larynx activity. An increase in the relative open phase duration of the glottal cycle before final and intervocalic voiceless consonants, for example, can be seen to correlate with loss of high frequency acoustic energy in this environment.

The present work describes perceptual experiments designed to test the role of this energy loss as a cue to the voicedness of a following consonant. Utterances of (ɑ: 'sɑ:) and (ɑ: 'zɑ:) were synthesized by copy; tokens representing intermediate degrees of voicing were then synthesized. Preliminary perceptual testing indicated the position of the categorical boundary on this continuum. The next categorical assessment was based on the use of a continuum of tokens derived from this boundary pattern by the systematic variation of the preconsonantal larynx source spectrum. Results for individual listeners are discussed.

The Perceptual Integrity of Initial Consonant Clusters

Anne Cutler and Sally Butterfield
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Detection time for word-initial phonemes is faster when the phoneme is followed by a vowel (as in *band*) than when the phoneme is part of a cluster (as in *brand* or *bland*). We investigated this effect as a function of the *model* which subjects were given for the target phoneme. Subjects listened for word-initial target phonemes in continuous utterances. Subjects who were told throughout the experiment to listen for '/b/ as in bowl' were faster with words like *band* than with words like *brand* or *bland*, as previously found. However subjects told to listen for '/b/ as in blue' were faster with words like *bland* than with words like *band* or *brand*, while subjects told to listen for '/b/ as in brave' were faster with words like *brand* than with words like *band* or *bland*. We further demonstrated that no such effects are found with consonant-vowel onsets; varying the models '/b/ as in Bill' and '/b/ as in Ben' had no effect on time to detect /b/ in words like *builders* or *benches*. The 'matching model' effect is specific to consonant clusters, and suggests that word-initial clusters are perceived as integral units.

Forward Masking and the Perception of Stop Consonants: Psychoacoustical and Electrophysiological Experiments

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Disputes exist concerning some speech perception phenomena: do they require speech-specific phonetic processes, or do they have auditory explanations? One phenomenon believed to be evidence for specific processes related to speech production (Motor theory of speech perception) concerns stop consonants: removing the silent interval corresponding to the vocal tract closure preceding release of a stop leads to 'suppression' of the acoustically present stop consonant, eg /spin/ is heard as /sin/. We compare the results of neurophysiological experiments on animals with those from psychophysical tests on humans, since animal experiments exclude the effects of possible speech-specific perceptual processes.

Naturally spoken German words /stahl/ and /spott/ were prepared with different length silent intervals between the fricative /sh/ and the corresponding /t/ or /p/, and various fricative sound levels. It is found that:

1. The spectral properties of the signals and the psychoacoustical results do not contradict the assumption of forward masking.
2. Psychophysical results with different fricative levels support this assumption.
3. The neurophysiological results correspond well to those measured in forward masking experiments with simple signals.
4. Psychoacoustical and neurophysiological findings with the speech signals show qualitative analogy.

These results are evidence that auditory masking plays a role in the suppression of stop consonants and that these phenomena can be explained without assuming special phonetic processes in speech perception.

Discrimination of Speech Intonation Contour. Evidence for Tonetic Categories?

A Faulkner
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The idea that there are perceptual categories for intonation contour has particular attractions in speech recognition and speech synthesis. If this notion is correct, then tonetic category boundaries should, according to the logic of categorical perception, be observable as discrimination peaks.

A series of ABX experiments will be reported which examine discriminability along stimulus continua formed by 1- and 2-segment logarithmic f_0 slopes imposed on synthetic speech. The discrimination peaks that have been found are consistent with boundaries between the tonetic categories of 'rise', 'shallow fall', and 'fall'.

Modelling the Perception of Simultaneous Vowels

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Listeners can report more accurately than chance both members of a pair of simultaneous vowel sounds that possess the same amplitude contour and fundamental frequency (Fo). Generally, they do so without perceiving two separate sound sources. Our aim was to model their performance. Two sets of the vowels /i a u)3/ were generated: (1) by cascade formant synthesis and (2) by additive harmonic synthesis with just 3 pairs of harmonics, straddling the frequencies of the lowest three formants. Fo was 100 Hz with the first pitch pulse occurring at the same time in relation to the amplitude envelope of each vowel. Six listeners with normal hearing identified both vowels in all possible pairings using the same five vowels as responses. The (arcsin-transformed) proportions of the response options given to each 'double' vowel may be considered to reflect its *perceived* similarity to each single vowel. These proportions were compared with the predictions of several models of the *auditory* similarity of the single and double vowels. Correlations between perceived and auditory similarity were quite high ($r = 0.82 - 0.90$) when spectra were expressed as (i) the negative part of the second differential of the auditory excitation pattern, and (ii) the frequencies of the formant peaks in the gross envelope of the auditory excitation pattern. Representations that assign equal weight to spectral peaks and valleys gave lower correlations. Together with the finding of similar performance with the two synthesis types, these results confirm that spectral peaks and shoulders are important in the perception of vowels. Additionally, they suggest that listeners may solve some aspects of the problem of identifying concurrent sounds in part by making multiple spectral comparisons rather than by perceptually 'subtracting' one sound from another.

Evidence for right-to-left Flow of Information in Human Speech Perception

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Many current models of word recognition in running speech implicitly or explicitly assume that words are recognized in sequential order, on the basis of the acoustic information corresponding to the word itself together with the prior context, and that a word's identity is known at least by the acoustic offset of that word.

Evidence is reviewed which argues that there is also a reliable role for context which follows the word in question. An experiment is described which provides further evidence for the existence of such 'right context effects'. The experiment uses a word-level 'gating' technique to determine the extent of the word-by-word intelligibility of a large corpus of spontaneous speech. Normative data is presented which characterizes the circumstances in spontaneous speech which are most likely to give rise to right-context effects.

The implications of these data are discussed with reference to models of human word recognition and to attempts at speech recognition by machine.

Towards Improved Modelling of the Link between Aerodynamic Processes and Acoustic Sources in Speech

G Clark and Celia Scully
SPAR Group, University of Leeds

In our existing model of speech production processes, aerodynamic and acoustic processes are treated separately. In the aerodynamic block, low frequency components of air pressure and volume velocity of airflow at several points in the respiratory tract are derived. In natural speech, aerodynamic conditions at the glottis combine with the position and mechanical state of the vocal folds to generate a quasi-periodic volume velocity waveform, called the voice source. This is the acoustic component of volume velocity of airflow through the glottis. These processes are represented in a highly simplified way in the voice source block of the model. In our previous modelling, consistency between these two descriptions of transglottal airflow has not been attempted.

This study considers simplified representations of the physical processes involved in this aerodynamic to acoustic mapping. Methods of implementing them in the modelling will be described.

Improved representation of the processes responsible for the voice source should assist the modelling of noise sources also. The relationships between the different acoustic sources will be discussed.

Simulation of Two Women Speakers of English with a Model of Speech Production Processes

Celia Scully and G Clark
SPAR Group, University of Leeds

It is possible to produce, by trial and error with auditory monitoring, acceptable synthetic versions of men's speech and singing, using our model of the processes of articulation, aerodynamics and acoustics. The problems of modelling similarly the speech and singing of women are even greater, because there are very few quantitative descriptions of women's articulation and little acoustic modelling of their vocal tracts has been done. Parameter values for voice source generation need to be better understood also. Different speakers may exhibit different patterns of transglottal losses during the voicing cycle for example.

This study describes attempts at modelling two women speakers of English. Controlled phonetic contexts and data obtained in our laboratory as well as published descriptions are used for initial estimates of parameter values in the model. Modelling techniques for improving the match between the natural and the synthetic speech will be described.

STAR-PAK: A Signal Processing Package for Acoustic-Phonetic Analysis of Speech

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Automatic speech recognition based on acoustic-phonetic feature extraction demands extensive application of speech signal processing. The aim of such signal processing is to provide a maximally rich description of the speech signal in several transform domains, where each transform domain offers a characterization of signal information in a complementary form. In this way, a highly correlative broad knowledge base of the signal characteristics can be maintained.

Elements from various transform domains in the knowledge base can then be used in a rule-based phonetic analysis system capable of producing BroadClass to FineClass phonetic interpretation of the speech waveform. Additionally, an ability to alter signal processing parameters within any given kernel transform, according to the performance of the rule-based system, or other processing elements at the lexical/syntactic/semantic levels of the system, offers improved overall recognition performance.

This paper describes the philosophy and processing architecture of the Signal Transform Analysis for Recognition software package (STAR-PAK). The architecture provides both an array of kernel transforms and combinations of kernel outputs which form the first stage in the acoustic front-end processor of a feature-extraction-based continuous speech recognition system. In addition to providing a representation of the signal in various transform domains, the architecture also allows for the possibility of both overall system performance evaluation and intelligent speech signal processing driven by a system controller.

The Application of Analytic Signal Analysis in Speech Processing

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This paper explores the application of complex time domain signal analysis in speech processing. The basis of the analysis is the treatment of the speech pressure waveform as the real part of a complex or analytic signal. This treatment allows the unique specification of signal amplitude envelope and instantaneous frequency. Of these two signal time domain attributes, the former is perhaps the more familiar since it is related to signal energy. The interpretation of signal instantaneous frequency in speech analysis, and indeed in the analysis of acoustic data generally, has not yet been fully explored. Real speech data is used here to assess the utility of this much neglected signal time domain attribute and exemplify possible applications.

The starting point of the assessment is a comparison of the instantaneous frequency of voiced and voiceless speech signals. Results of the comparison illustrate the potential of this signal time domain attribute for voicing determination. It is also shown that although there is no one-to-one correspondence between time domain and Fourier domain frequencies, suitable manipulation of signal instantaneous frequency does yield information commonly assumed to be obtainable only through short-term Fourier analysis of the signal. These findings demonstrate the power of appropriate time domain processing techniques, and suggest that their use may result in valuable insights into how information is encoded in the speech pressure waveform itself.

The Wigner Distribution as a Speech Signal Processing Tool

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The representation of speech signals by conventional spectrum estimation techniques (such as filter bank analysis and the discrete Fourier transform of time windowed signals) suffers from the well known problem of compromise between temporal and frequency resolution. Specifically the information-theoretic inequality tells us that the minimum attainable bandwidth Δf from a signal of duration Δt is determined from $\Delta t \Delta f > 1/2$. Therefore if we wish to examine transient behaviour (for example on the scale of a pitch period) the time window must be narrow, the effective signal duration short and hence the bandwidth large leading to a smearing of spectral features such as formants and harmonics of the fundamental frequency. Similarly if fine spectrum resolution is required then an accurate representation can only be obtained if the signal is stationary. Since speech is a signal with time varying spectrum content, this means that conventional spectrum analysis does not exploit the full information present in the speech waveform.

However there are alternative signal processing techniques, based on the Fourier transform of correlation functions, which retain the optimum information content of a signal. This paper will discuss one such technique known as the Wigner representation. After a brief historical background the conventional and Wigner representations will be illustrated for several test functions and segments of synthetic and human speech to demonstrate the advantages of the Wigner formalism. Finally the problems of this approach and its future relevance to speech recognition will be discussed.

SEGLAB: An Interactive Environment for Phonetic Segmentation and Labelling of Speech

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SEGLAB is a component of the speech transcription system under development at the Centre for Speech Technology Research, University of Edinburgh. In that system processing follows different stages: digital signal processing, segmentation and labelling, lexical lookup, and the formation of syntactic strings. SEGLAB constitutes the second of these stages.

The basic technique is feature extraction (as opposed to template matching). In the first version of SEGLAB, described here, a number of acoustic parameters from signal processing were analysed into phonetic features, and then sequenced into MidClasses. The MidClasses are a classification of the 44 phonemes of English into acoustically similar groups.

SEGLAB is rule-based, with different kinds of rules to do the feature extraction and the sequencing into MidClasses. It provides both an environment in which the rules are developed, and an environment in which they are routinely run. The forms of the rules themselves are described in another paper. This paper describes in general what sorts of facilities are involved, and how they are used.

An End-point Detection Algorithm Incorporating Acoustic-phonetic Knowledge

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In some speech system applications, such as word-spotting, target word identity is known in advance of any processing. Such *a priori* knowledge can be exploited in the design of end-point detection algorithms. An improvement in accuracy and a decrease in word-rejection caused by extraneous noise, can be achieved if acoustic features, specific to a word or class of words, are used in the boundary detection algorithm. This paper investigates how *a priori* knowledge of acoustic features at word boundaries can be incorporated in an end-point detector. A comparison is made with an established end-point algorithm which has no stored speech knowledge about the input word.

Cepstral and Spectral Approaches to separating Concurrent Voices

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The ideal hearing aid would selectively amplify a target voice and attenuate competing voices and other sources of interfering noise. In exploring one instance of this problem, we are evaluating two algorithms for separating two voices speaking concurrently with periodic excitation at similar overall intensities. Both approaches exploit the regularity in the harmonic structure of voiced speech. The first involves attenuating the harmonics corresponding to those regularities of spectral structure that correspond to the competing voice, via a cepstrum-like representation. Five steady-state vowels were synthesized on fixed fundamentals of 115 Hz and 190 Hz and mixed in pairs containing a low- and a high-pitched vowel. When the competing vowel was suppressed, the intelligibility of the lower-pitched vowel increased by 25%, and of the higher-pitched vowel by 11%. Informal tests suggested that analogous improvements occur in the intelligibility of naturally-produced all-voiced sentences. The second method was derived from the procedure for harmonic selection described by Parsons (JASA, 1976, 60, 911-918). The fundamental frequency of each voice is estimated from the amplitudes and frequencies of all harmonic peaks resolvable in the composite spectrum. From this fragmentary information, the spectrum of each voice is reconstructed. Preliminary experiments on paired vowels and natural discourse suggest that the greater sophistication and complexity of this second method can lead to greater enhancement than that from the cepstral approach.

Spectral Continuity as a Relaxation Constraint on Grouping in Speech Spectrograms

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We present a relaxation scheme for isolating perceptually salient aspects of a spectrogram. The aim of the project is to design algorithms capable of isolating and grouping together features (consisting of energy peaks in frequency or in time) which are visually prominent in a Bark-scale spectrogram. Peaks in a Bark-scale spectrogram may correspond to harmonics (below about 1 kHz for a male voice) or formants. The algorithm first identifies each spectral peak as a separate datum. Continuity constraints then group adjacent peaks together, if they are similar in intensity and orientation, to give a line sketch of the original spectrogram in which each harmonic or formant is represented by a single continuous horizontal line. Abrupt onsets of wide-band energy such as stop bursts are represented by vertical lines. We will compare the performance of relaxation algorithms using different levels of constraint, including those that are specific to speech.

Speech Spectral Segmentation for Spectral Estimation and Formant Modelling

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Linear Predictive Analysis is one of the most extensively used speech processing techniques and has been employed as a time domain method for the estimation of the basic speech parameters. It is in the area of spectral estimation that Linear Prediction (LP) can also be used as an effective means of formant analysis or speech spectrum smoothing for speaker/speech recognition.

The application of these ideas to selected portions of the signal frequency response, rather than uniformly over the entire spectral range leads to the formulation of the 'Selective Linear Prediction' technique. However, the choice of frequencies which divide the individual regions and the predictor order to adopt in each portion of the overall spectrum is essentially intuitive and is based on merely an approximate notion of the spectral energy distribution of the utterance under consideration.

This work describes a method of splitting the signal spectrum into several segments using the approach of Dynamic Programming in order to produce the 'best' fit between real and model spectra on the basis of a suitable error criterion. This technique is then modified for the segmentation of a frequency response into speech formant regions and implemented for several vowel utterances.

Estimates of formant position are then calculated from the Selective LP model parameters associated with these segments. The results of this procedure are finally incorporated into cascade formant synthesizer configurations and are compared with those available using standard LP modelling.

This work is carried out in collaboration with the IBM UK Science Centre, Winchester, UK and we are grateful for their help.

Formant Tracking

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A distinguished succession of speech engineers from Walter Lawrence to John Holmes have extolled the virtues of parallel formant synthesizers for the production of realistic speech but it has proved very difficult to extract formant tracts automatically from real speech. Dr J C Manley and I have attempted to solve this problem by dividing the 12-pole lpc log spectrum into 14 one-third octave channels, and using AI techniques for picking out the three or four most prominent ridges in the resulting sequence of discrete spectra. In favourable circumstances three such tracks, together with voicing parameters, provide adequate information for the production of acceptable speech.

An Improved Formant Tracking Algorithm based on Pole Enhancement in Autoregressive Signal Processing

G Duncan and M A Jack
Centre for Speech Technology Research, University of Edinburgh

Extraction of acoustic-phonetic features from a speech waveform relies on accurate tracking — with respect to time and frequency — of the short-time spectral resonances (formants) present in the acoustic speech waveform, so as to provide a detailed acoustic interpretation of voiced speech. Formant tracking algorithms generally rely on deconvolution of voiced-excitation and vocal tract filtering functions so that effects due to the excitation spectrum can be rejected. In addition, such algorithms generally use some degree of knowledge concerning vocal-tract dynamics in order to track formants from one instant to the next.

This paper describes a formant tracking technique based on excitation-source deconvolution using autoregressive (AR) signal processing. The technique offers improved accuracy and signal-to-noise performance over conventional formant tracking systems based on AR spectral analysis. In addition, it is shown that the ability to take a longer-term contextual overview of the characteristics of the speech waveform offers enhanced performance.

Vector Excited Wiener Filtering: An improved Stochastic Model for Speech Production

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In this paper we will introduce a speech production model based on Wiener filtering theory, applicable to speech coding at low bit rates. We view the excitation signal as a zero mean Gaussian stochastic process, which is vector quantized. The coefficients of an adaptive synthesis filter and an excitation vector are computed to minimize the expected squared error between the synthetic and natural versions of the waveform. The optimum synthesis filter is identified as a Wiener filter. Since the excitation as well as the synthesis filter are computed to obtain best possible synthesis for a block of speech, this configuration is optimal in terms of distortion.

The FAAF — A Single General Purpose Test with Multiple Special-purpose Applications

J R Foster and M P Haggard
MRC Institute of Hearing Research, Nottingham

There have been three main phases in the development of the Four-Alternative Auditory Feature test (FAAF):

1 Choice of vocabulary and response format according to ergonomic design principles, and the choice of electro-acoustic conditions of presentation, to show up small differences between people or conditions in audiology and communications engineering. (Mark I).
2 The optimization of the Mark II test towards revealing small differences between conditions, in the light of item-analysis on extensive bodies of applications data (the present topic).
3 Further development in progress using a subset of the 80 FAAF items towards a computer-controlled clinical package to run adaptively, providing various indices including speech-reception thresholds in quite and in noise.

In phase II we have concentrated on the methodology of special-purpose subscores for revealing small differences, eg in S/N ratio, drawing upon several bodies of human-factor and audiological data. We have shown that differences between conditions or subject groups may be as powerfully demonstrated by using a specific subset of items as by using all 80 — thereby as much as halving the length of clinical or ergonomic evaluations, eg a 5- rather than a 10-minute test. As an obvious example, difficult conditions differing only slightly in difficulty are best distinguished by the easiest items and vice versa; but the principle is more far-reaching. Sizeable differences between the effects of two different frequency-responses (FR) (Flat and +9 dB/oct) have been shown ($p<0.001$), even though the overall scores are identical. By the same principle any overall advantage for one FR can be attributed specifically to its effects in the low, mid or high-frequency region. Particular ranges of signal/noise ratio likewise have optimal distinguishing subsets. Such subsets are constrained by the intensity ranges or the dominant frequency ranges of the various known acoustic cues to phonemic distinctions, but have been empirically optimized for the test speaker and for the particular acoustic conditions used. To reflect the use of information from each of three *a priori* frequency bands (low, mid, high), only 10 or fewer classes of error are counted, although these are each based on pooling feature error responses to potentially about 40 of the items.

Further examples are given of tested hypotheses that support similar claims for subscores developed for sensitivity to S/N ratio in various regions of S/N ratio, and degrees of hearing impairment. The benefits of the foregoing analyses of applications data over the last six years lie in the updated scoring programs for licensed users and an imminent definitive publication illustrating the derivation and use of subscores.

STI Measurements using a Dual Channel Analyser

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The Speech Transmission Index is based on the Modulation Transfer Function. The MTF can either be measured directly in the modulation domain, as suggested by T Houtgast and H J M Steeneken, or calculated from the impulse response. The method described in this paper measures the noise-free impulse response and calculates the MTF according to the definition by M R Schroeder. The noise-free MTF is then corrected for the influence of noise by a measurement of the background noise.

The theory and instrumentation will be outlined and illustrated with a measurement example.

On the Intelligibility of Synthetic Speech

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Increasing interest is being shown for the use of synthetic speech in military communications tasks. As a consequence suitable assessment methodologies need to be developed in order to quantify the effectiveness of synthetic speech products from different manufacturers.

This effectiveness will depend on a variety of factors, one of which is intelligibility. This paper will briefly describe the Diagnostic Rhyme Test (DRT) and present the results of some tests carried out on a range of commercial and experimental speech synthesizers. These devices have been tested in the presence of noise and with bandpass limiting. Whilst the DRT was developed for human-human studies it is hoped that the results from the synthetic speech experiments will form a useful bench-mark for comparing systems.

Synthetic Speech Tests and Usable Hearing

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Many profoundly deaf persons do not use a hearing aid. Such persons have commonly had a trial (perhaps many years ago) and decided an aid was of no benefit. But experience over three years with a screening procedure for cochlear implants has highlighted the difficulty in finding candidates who actually completely fail to benefit from amplification.

This paper presents the screening procedures used, with particular reference to speech tests. The use of natural speech tests is generally inappropriate with profoundly deaf persons, whereas synthetic speech provides detailed control and a wide range of degrees of difficulty. The use of synthesis in testing as well as in speech therapy is reviewed.

We present results on the speech processing abilities of profoundly deaf persons before and after use of an electrical implant or conventional amplification. Finally, we discuss reasons for success or failure with various types of prosthesis, with particular reference to the significance of therapy.

Measurement of the Benefit of Hearing Aid and Surgery in Patients with Asymmetric Hearing Losses, using an Adaptive Free-field FAAF Test

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The free-field FAAF test has two major faults when used on individuals with asymmetric hearing losses. Firstly the speech is presented from in front of the individual and in these circumstances the better hearing ear determines the score, which is therefore abnormally high. Secondly a fixed signal to noise ratio is expected to encompas a wide range of hearing losses.

To overcome these problems an adaptive version of FAAF has been devised which alters the signal to noise ratio keeping the percentage correct score at a pre-determined level of say 70%. A further refinement is to perform the test with the speech presented first in the straight ahead position and then at an angle of 45 degrees towards the worse hearing ear. This gives scores in the positions of both least and maximal disability. By subtracting these scores a disability score is obtained. A comparison of disability scores before and after a hearing aid fitting or surgery gives a benefit score. Benefit scores between different treatment modalities can be statistically analysed.

Results from individuals with asymmetric conductive hearing losses will be presented to illustrate the effectiveness of this test in assessing disability and benefit.

Making a Hearing Aid Tester

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A D Wallis
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Hearing aids should be objectively tested far more often than is the usual practice. One reason tests are not made is the lack of test equipment. The RNID has spent more than ten years developing various devices, all intended to provide relatively inexpensive test equipment for schools, clinics and dispensers.

We will review current technology, and describe four RNID systems: two stand-alone devices and two based on microcomputers. Problems of signal generation and measurement will be reviewed, with reference to test enclosure design, transducers and use of signal processing.

Finally, we will discuss:

1) problems encountered in getting from a prototype to a commercially available device;
2) testing to recognized standards vs cost, complexity, and user acceptance;
3) manufacture, distribution and user requirements;
4) the advantages and disadvantages of using microcomputers;

5) where we went wrong in the past, and our current plans.

Acoustical and Perceptual Correlates of Stutterers' Speech after Treatment

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Many stutterers who have gone through one of the fluency shaping therapies appear to produce speech containing only a small number of disfluencies. Yet, their speech is often not completely natural, so that it is easily distinguished from the speech of non-stuttering control subjects. In our contribution we will describe a number of acoustical and perceptual features of post-treatment speech that are responsible for its deviant character. We have made an attempt to automatize the acoustic analysis, in order to make it a useful tool for a complete assessment of the results of a stuttering therapy and for the analysis of the changes in clients' speech that appear to occur in the 6 to 12 months after the end of the formal therapy.

Vowel Production using Visual Feedback

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This paper presents a device which is capable of displaying tongue position for vowel articulation. It is a common feature amongst those who have experienced hearing loss in adult life that vowel production tends to deteriorate; it can also be a problem for the pre-lingually deaf. This device is intended to be used in a teaching program to provide visual feedback on vowel production.

A mid-sagittal plot is displayed on a suitable screen showing the essential parts of the articulatory apparatus. The shape of the tongue is determined from basic characteristic features extracted from the speech waveform. The device is based upon Ladefoged *et al's* PARAFAC analysis[1] which gives an empirical relationship relating formant frequencies to tongue shape. The formant frequencies are extracted from the speech waveform by simple peak picking of the spectral signal derived from a 16-channel filter bank.

Two tongue shapes are generated, one as a reference, the second the users attempt to mimic it. By the process of trial and error, the users are thus able to correct their vowel production.

The paper presents the technical aspects of the device along with results obtained on field trials.

Reference

1. Ladefoged, P, Henshman, R, Goldstein, L and Rice, L (1978), *Generating Vocal Tract Shapes from Formant Frequencies*, J Acoust Soc Am, 64(4), 1027-1035.

A Psycho-physical Assessment relating to Speechreading

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Since instruction in speechreading was formalized, tutors have stressed the importance of certain visual attributes, for example visual acuity and the speed of visual perception. Within the last 30 years, the reliance of speechreading ability on discrete psychological and physiological factors has been more fully investigated. Correlation has been reported with many visual attributes including pattern recognition and latency of visual evoked potentials. It is accepted that speechreading is a psycho-physical task in which skills are utilized. It would, therefore, seem logical to assume that many such discrete attributes interact to determine an individual's aptitude as a speechreader. Assessment of speechreading aptitude must involve this interaction. A proven test would enable hearing therapists to alter their emphasis of therapy between speechreading and amplification techniques.

A basic model of speechreading has been used to design a test which aims to assess the overall ability of an individual to perform speechreading tasks. The test involves rudimentary psycho-physical factors, such as speed of perception and pattern recognitions, and is contained in a micro-computer based test, which is quick and easy to perform.

Results are available from 40 severely hearing impaired subjects whose average length of time of wearing hearing aids is 26 years. This group should have optimum speechreading skills.

A highly significant correlation (0.57) was found between the test score and speech-reading as assessed with a video-based sentence test. This effect holds after the removal of subjects' age as a co-variate. Correlation between the speechreading score and other visual function measures was insignificant even with effects of partial correlates removed.

Visual Presentation of Voicing and other Cues as an Aid to Lipreading

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The possibility of using a visual display of acoustic features as an aid to lipreading is being investigated. A device has been developed which illuminates

(a) a single light in response to a voiced sound,
(b) one light in response to voiced sounds and another light in response to other sounds, or
(c) two lights as above and a third light in response to strong sibilants.

Experiments have been performed to measure the effectiveness of this device in aiding the identification of consonants in an /aCa/ context. It has been found that the recognition score for lipreading alone was about 33%, but for lipreading using the device the score increased to about 54%.

A New Method for Quantifying the Contribution of Vision to Speech Perception in Noise

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The intelligibility of sentences presented in noise improves when the listener can view the talker's face. Our aims were: (a) to quantify the benefit, and (b) to relate it to individual differences among subjects in lipreading ability and among sentences in lipreading difficulty. Auditory and audio-visual speech-reception thresholds (SRTs) were measured in 20 listeners with normal hearing. Stimuli were 60 sentences selected to range in the difficulty with which they could be lipread (with vision alone) from easy to hard. Using the ascending method of limits with a step size of 2 dB, the SRT was defined as the lowest signal-to-noise ratio (S/N) at which all three 'key words' in each sentence could be identified correctly. In the auditory-alone condition, runs started at -18 dB S/N. In the audio-visual condition, sentences were presented first at $-\omega$S/N to measure subjects' lipreading ability, and runs then started at -30 dB S/N. Measured as the difference in dB between auditory-alone and audio-visual SRTs, 'visual benefit' averaged 11 dB, ranging from 6 to 15 dB among subjects, and from 3 to 22 dB among sentences. As predicted, the benefit from vision in audio-visual conditions, measured as a difference in S/N, is a measure of lipreading ability, being highly correlated with performance with vision alone ($n = 20$, $r = 0.86$, $p<0.01$). Likewise, freed by the SRT method from any ceiling effect, those sentences which were easiest to lipread gave a higher measure of benefit from vision in audio-visual conditions with noise than did sentences that were hard to lipread ($n = 60$, $r = 0.92$, $p<0.01$). The results establish the basis of an efficient test of speech-reception disability in which the lipreadability of the test sentences is controlled.

Speech Research at RSRE

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Following the official transfer of the Joint Speech Research Unit (JSRU) to the Royal Signals and Radar Establishment (RSRE) on 1 November 1985, the speech research programmes at RSRE and JSRU have been combined and established as the RSRE SPEECH RESEARCH UNIT. The new Unit will continue to provide the focus for the UK's long-term strategic speech technology research by means of a strong internal research programme and through collaboration with industry and the universities.

This paper will present an overview of the structure of the RSRE Speech Research Unit and will outline the main objectives of the integrated speech research programme. The overall research methodology will be discussed and, in particular, it will be shown how the problems of speech recognition and synthesis are being tackled using a structural/stochastic modelling approach. It will be argued that this approach is aimed ultimately at the development of a rigorous theory of speech pattern processing.

This paper will provide the background for the other, more technical, papers from the RSRE Speech Research Unit, which will be presented at this conference.

Comparative Isolated Word Recognition Experiments

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This paper reports comparative recognition results for two alternative isolated word template-matching recognition strategies, which were assessed in the same low cost microprocessor-based hardware.

The first strategy used templates constructed from features consisting of the first two major spectral energy peaks and the total speech energy, extracted from time averaged speech frames. Linear time normalization was used in an attempt to overcome speech temporal variability.

The second strategy used only the first two major spectral energy peaks from each frame but used a symmetric dynamic time warping algorithm.

The results show that for recognition

systems based on minimal distinctive features, non linear dynamic time warping makes a far more significant contribution to recognition accuracy than an additional descriptive feature such as utterance energy. So significant is the increase in recognition performance that it may be concluded that even low cost systems should be designed with dynamic time warping, or similar techniques, before attempting to increase the number of descriptive features employed in the speech pattern.

Dynamic Speaker Adaptation in Speaker-Independent Word Recognition

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Within a speech recognition system conducting a dialogue, a large amount of information is available relating to the current speaker's characteristics. Ignoring this information is equivalent to assuming that each utterance in the dialogue is spoken by a different speaker. The possible advantages of a system which adapts to the speaker are a reduction in the recognition search space and an improvement in recognition accuracy due to reduced inter-speaker pronounciation confusability.

In this paper we consider a method of dynamic speaker adaptation suitable for use in pattern matching where it is assumed that each word is represented by multiple reference templates obtained via cluster analysis. These templates are considered to represent various speaker subpopulations and the method attempts to adapt dynamically to the speaker by identifying those subpopulations which are relevant. Results are presented which indicate the performance of the method on a speaker independent digit recognition task.

An Isolated Word Recognition System with Progressive Adaptation of Templates

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An isolated word recognition system recently implemented in software on a minicomputer will be described. This incorporates linear or acoustically-based time segmentation of input and reference word patterns, and several stages of word comparison with different numbers of segments per word allowing elimination of unlikely recognition candidates at an early stage. It also provides the option of progressively adapting the reference patterns to the characteristics of the recognized input by a weighted averaging procedure, so as to update existing speaker-specific reference patterns or to adapt initial speaker-independent ones to a particular speaker. The adaptation can be made conditional on a measure of the degree of confidence in the recognition, or on feedback from the user as to the correctness of the recognition decision. In the latter case, adaptation away from misrecognized input is a possibility. Results, to date, of experiments with various initial training and adaptation conditions will be presented, and an indication of proposed future work will be given.

Experiments in Isolated Digit Recognition using Hidden Markov Models

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Over the past year the RSRE Speech Research Unit has been conducting an extensive series of experiments on speaker dependent isolated digit recognition using hidden Markov models (HMMs). The purpose of these experiments is to provide a reliable set of comparative results which show the implications for recognition accuracy of particular choices of HMM parameters and algorithms. Issues which have been addressed include the effects of size of training set, number of states in the HMM, topology of the underlying Markov model, alternative methods for estimating initial statistics and alternative parameter estimation and recognition algorithms.

This paper will begin with a description of the database which forms the basis of the experiments, including the criteria used for choosing speakers. Some of the more interesting results of the study will then be presented, including a comparison of recognition accuracy obtained using hidden Markov modelling and traditional 'template matching' techniques based on Dynamic Time-Warping, a study of four different algorithms for initial HMM parameter estimation, and a comparative study of the Viterbi and Baum-Welch algorithms for parameter estimation and recognition.

Improved Duration Modelling in Hidden Markov Models using Series-parallel Configurations of States

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Hidden Markov Models (HMMs) form the basis of many current speech recognition systems. In this approach a pattern arising from a particular utterance is treated as a sequence of quasi-stationary regions, each of which is modelled as the output of a stationary stochastic process. These processes are identified with the states of an underlying Markov chain, which models the temporal structure of the utterance. The key to the success of the HMM approach is the availability of rigorous mathematical techniques for model parameter estimation and recognition.

However several properties of HMMs are inappropriate for modelling speech signals. For example state duration in a HMM is modelled by a geometric probability density function (pdf), so that the probability p(D) of remaining in a particular state for duration D is proportional to $a^D(1-a)$ $(0<a<1)$. Hence as D increases, p(D) decreases. This is undesirable in a model of durational structure for speech signals.

The approach to improved duration modelling proposed here is motivated by the fact that the pdf describing the combined duration statistics of two states in series in a Markov model is the convolution of the state duration pdfs of the individual states. Therefore by identifying the quasi-stationary regions of an utterance with series-parallel configurations of states, rather than individual states, more appropriate state duration pdfs can be obtained. The advantage of this approach is that the underlying model retains its first-order Markovian properties, hence the standard parameter estimation and recognition techniques are applicable.

After a review of the background theory, this paper will describe the duration pdfs which are obtainable from particular configurations of states. The implications of the method for parameter estimation will be discussed and the state duration pdfs which arise when the method is applied to speech patterns will be presented. Finally, results of speech recognition experiments using these techniques will be presented.

The Multi-layer Perceptron as a Tool for Speech Pattern Processing Research

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A pressing problem in speech pattern processing is the construction of stochastic models which have appropriate internal (hidden) structures. Normally, the structure of a speech pattern model is defined using *a priori* constraints based on some understanding of the nature of the relevant patterns; for example in hidden Markov modelling the number of states in the model may be defined by the expected number of roughly stationary spectrum regions in a word. The stochastic parameters in the models are then estimated using *a posteriori* information extracted from a set of training examples of actual patterns; in hidden Markov modelling the state transition probabilities and the state output probabilities are derived in this way.

However, recent work in this laboratory and elsewhere has been concerned with model building strategies which are capable of learning structural and stochastic information simultaneously.

This paper will introduce a particular adaptive network approach based on the 'multi-layer perceptron' (error back-propagation network). Examples will be presented of the technique being used to determine the 'hidden' internal structure of a variety of speech-related problems. The conclusion will be drawn that the multi-layer perceptron is a very useful tool for speech pattern processing research.

Experiments with a Learning Network for a Simple Phonetic Recognition Task

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The error back-propagation algorithm of Rummelhart, Hinton and Williams is a solution to the long-standing problem of learning in multi-layer perceptron-type networks. Most examples of its use to date have been on illuminating but carefully-contrived small artificial problems. We present initial results on the use of BP to perform part of a well-established coarse phonetic class labelling task, given a few acoustic measurement 'traces' as input. The initial goal is to replicate the performance of a computer program which was intended to encapsulate a phonetician's experience in 'reading' such traces (Roach and Roach 1983).

Some Quantitative Comparisons between Speech Fundamental Frequency Estimation Algorithms

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Various comparison techniques are being investigated to make possible a quantitative comparison of fundamental frequency estimation algorithms against a reference that makes use of a laryngograph. These measures are carried out on a pulsatile output from the devices, where each pulse corresponds to an acoustic excitation due to a vocal fold closure, and they include:

1) Cross correlation.
This gives the time shift for best-fit between the output from the device under test and the reference.

2) Receiver operating characteristic.
This is a plot of the probability of detecting a vocal fold closure as indicated by the reference, versus the probability of a false alarm. It is intended that this measure should give an indication as to how well the device under test performs with respect to the reference, in a way that is independent of the actual threshold criterion used in the test device.

3) Jitter distribution.
This is a probability plot of the differences in the times of occurrence of output pulses in the reference and the corresponding time-aligned pulses from the device under test. It is hoped that this measure will give an indication of how precisely and consistently devices are able to place pulses with the respect to the reference.

Results will be discussed which illustrate the use of these measures in the assessment and optimization of fundamental frequency estimation devices.

A Pitch Detection Algorithm Optimized for Pitch Microperturbation Analysis

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This paper examines the performance of several versions of the time domain parallel processing method of pitch period estimation of speech, highlighting the limitations of each. An improved algorithm, based on the temporal investigation of the speech waveform, is described and evaluated. It is shown to be superior in both long term pitch period estimation accuracy, and cycle-to-cycle accuracy. The accuracy in cycle-to-cycle estimation of this algorithm permits its use in the detection of pitch microperturbation. A new method of evaluating the performance of pitch detection algorithms, based on measuring their stability with respect to variation of the time origin of the input speech, is also described.

TMS320: Knowledge Based Pitch Detection in Real Time

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Software processing of speech in real time has become convenient in recent years, thanks to the introduction of specialized signal processing chips. The TMS320 is especially useful, since its built-in facilities make relatively easy and efficient programming possible.

This paper describes the project in two parts: the first part extracts several features in the time domain, the second evaluates their probability distribution in various voicing states, and makes a decision on the most likely voicing and pitch frequency condition.

At present, the first part of the implementation is up and running. It provides continuous voicing information and pitch frequency in real time. This part uses a simplified evaluation module. The second phase of the project containing the full evaluation module is being implemented at the present time.

The completed project will ultimately provide a front end processor which can be readily used in speech research, in the recognition of intonation patterns, low rate coding and compression.

The facilities of the TMS320 make it also possible to provide a portable, low power consumption and inexpensive source of pitch information in aids for the handicapped: cochlear implants, tactile hearing aids and visual feedback in speech therapy for the deaf.

MidClass Phonetic Analysis for a Continuous Speech Recognition System

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This paper describes a feature-based acoustic phonetic rule component of a continuous speech recognition system which classifies speech sounds into a number of phonological MidClass categories, such as voiceless stop, voiceless fricative, nasal, etc. Where phoneme recognition fails, recognition at this MidClass level will provide information which can be exploited by a lexical access component that includes knowledge of the phonotactic properties of the words in the system lexicon, since the average equivalence class size of words 'spelled' in MidClass terms is relatively small.

The current system takes as input a vector of digital signal processing parameters computed for each time frame and derives the MidClass segmentation by applying various types of rules. First, a number of threshold rules are run to locate segments of the input data that meet empirically determined criteria such as the relative energy levels in various frequency bands, the locations of dips in these energies, estimated formant frequencies, etc. The feature segments created by these threshold rules are the input to a set of sequence rules which allow durational and contextual criteria to be applied.

The rules and the general principles underlying the methods employed in deriving them are discussed, and an evaluation of the system on a multiple speaker data base is presented.

A Discussion of some Criteria for Designing an experimental Text-to-Speech Synthesizer

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Text-to-Speech systems have traditionally been built as a collection of processes, each doing some part of the transformation from orthography to sound, and working serially. While this organization may be effective for a completely self-contained system, it is not suitable for Speech Researchers wishing to understand and control actions at various stages in order to experiment with, and adapt, the types of transformations that are being done.

The criteria for designing such an experimental organization are discussed, with reference to: the usability of the system from the point of view of Speech Researchers and Speech System designers; the ease with which it is possible to interact with the system to modify data and processes; and the ease with which the results of different actions can be compared, both visually and aurally.

The work is described with reference to the High-Quality Speech Synthesis project being done at the IBM UK Scientific Centre.

Selection of a Formant Synthesizer Model for Text-to-Speech Synthesis

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In a Text-to-Speech synthesis system the quality of the final speech output depends on a wide range of factors. In this paper we will review the choices available when selecting a synthesizer strategy and architecture and will discuss the consequences such choices have on the range of parameters required to drive that synthesizer. We will review the possible methods for deriving a formant parameter library relating a narrow phonetic description of the desired utterance to the corresponding synthesizer control parameters and show how an adoption of the appropriate synthesizer model can significantly facilitate the construction of the formant parameter library. The effect these choices have on the quality of the final speech will also be discussed.

The Preprocessor Module of the CSTR Text-to-Speech System

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The task of a Text-to-Speech system is to generate an appropriate pronunciation for any sentence in unrestricted, well-formed text. The role played by a Preprocessor module in such a system is to convert anomalous strings of alphanumeric and other characters into strings of 'neutral', lower-case ASCII characters that the Linguistic Processor(s) of the system can accept for further processing. Types of anomalous strings include numbers, proper names with initial capitals, acronyms, dates, abbreviations, mathematical symbols, chemical formulae, etc. This paper discusses the structure of a Preprocessor designed not only to facilitate the generation of the pronunciation of conventional text-sentences, but also to generate the appropriate pronunciation for the task of reading aloud in a proof-reading function.

Alignment of Phonemes with their Corresponding Orthography

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A technique has been developed for aligning the phonemes in a phonemic transcription of a word with the graphemes in its orthographic representation. For example creationism /kri:eɪʃnɪzəm/ can be aligned as

	c	r	e	a	t	io	n	i	s	m	
/	k	r	i:	eɪ	ʃ	ə	n	ɪ	z	əm	/

Tables of phonemic-to-othographic correspondences are given together with the frequency of occurrence of each correspondence in the texts constituting the *Lancaster-Oslo/Bergen Corpus* (LOB, 1978). The technique has been used as part of the inferential process in the building of rules for the automated phonetic transcription of English text, as an aid in the production of a computerized dictionary of English pronunciation, and to extract from this pronouncing dictionary all possible pronunciations of English prefixes. Uses are also seen in automatic speech recognition as part of the process of decoding the phonetic information extracted from the analysis of the acoustic signal. The algorithm has been checked using a phonemically tagged version of the LOB Corpus. The accent of

English used in this work is Received Pronunciation (RP), see Gimson (1980).

Cosegmentation in the IBM Text-to-Speech System

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Segment concatenation as part of a synthesis-by-rule process must take account of variations in synthesis parameters. Such variations include modifications to target values as a result of immediate segmental context, as well as the way in which parameters are interpolated between the modified target values for adjacent segments. Rules are currently being developed based on direct observations of a single RP speaker. Those rules which are already in use will be described and demonstrated.

Morphophonology in the CSTR Text-to-Speech System

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In a Text-to-Speech system which has as its first steps a decomposition of input items into morphemes, it is essential to formulate algorithms for pronunciation of sequences across morpheme boundaries. For example, the sequence 'ungracious' will be separated into 'un' + 'grac' + 'i' + 'ous'. When pronounced, it is necessary to change the 'n' in 'un' to the velar 'ng' and to ensure that the sequence 'c' + 'i' is pronounced 'sh'. Given rule ordering, it is relatively easy to construct rules for generating the right pronunciation in such cases. There are, however, several classes of alternations in stem pronunciations which are not signalled adequately by spelling but which must be dealt with, for example leap/leaped, please/pleasure, divine/divinity. This paper will outline how this and other similar problems are handled in the CSTR system.

Syntax and Morphology in Text-to-Speech

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Computer parsing techniques are normally top-down, or bottom-up, or based on charts (chart parsing). The third method combines the best features of the other two. None of these methods is satisfactory for a Text-to-Speech system which involves morphological decomposition and requires information about word-class and about phrase structure. The system to be described performs word class assignment, involving morphological information, information about closed-classes and dictionary look-up. Correct stress-assignment requires information about word-class. The procedures for grouping words into phrases proceed from word class assignment.

The Implementation of Lexical Stress Rules in the CSTR Text-to-Speech System

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In a morpheme-based Text-to-Speech system, lexical stress patterns cannot, in the majority of cases, be marked in the dictionary, since stress placement is influenced by the presence of affixes. Instead, primary and secondary stress must be assigned by rule. The Text-to-Speech system currently under development at CSTR is morpheme-based, and uses Fudge's (1984) rules as the basis for its stress assignment module. For each word, the input to the module is a list specifying the orthographic and phonemic form of each morpheme, with the stress-determining properties of any affixes marked, and the output is a phonemic string in which primary and secondary stress are indicated by the use of diacritics. This paper describes the implementation of the module.

An 'Accent Unit' Model of Intonation for Text-to-Speech Synthesis

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An intonational model for speech synthesis by rule (SbR) is being developed, based on the 'accent unit'. Implementation of this model in the JSRU SbR system has enhanced the naturalness of the output.

Accented syllables are rhythmically stressed and pitch prominent. The final accent in a tone-group carries the nuclear tone. An 'accent unit' consists of an accented syllable together with any following unaccented material.

Auditory and quantitative analysis of natural speech, delivered in a 'declarative' discourse style, indicates that the degree of relative accentual Fx obtrusion and inter-accentual excursion used by RP speakers is less constrained than a determinate mathematical model would predict.

In the present model the peak Fx on accented syllables is varied according to a probabilistic algorithm. This generates patterns of Fx between accented syllables appropriate to context, and at the same time, attempts to replicate the phonetic variability encountered in natural speech.

The Synthesis of British English Intonation using Tonetic Stress Marks

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A recent popular method of intonation synthesis-by-rule (Pierrehumbert 1980, Liberman & Pierrehumbert 1984) makes use of linguistic units (pitch accents) derived from the 'American' school of intonation analysis. An intonation synthesis-by-rule method is described that makes use of the framework of the 'British' school of intonation analysis. Thus the initial prosodic representation is framed in terms of tonetic stress marks. This system is being used for the prosodic transcription of a corpus of spoken English, and is a modification of the systems used by O'Connor and Arnold (1961) and Crystal (1969).

The data comprises long and fluently-spoken natural utterances, intonationally transcribed. Rules convert the intonational units to 'target values' on a ten-level scale, as in Pierrehumbert (1980). A declining topline and level baseline are then superimposed, yielding at least one frequency value per syllable. The resultant F0 contour is compared with the original. The close match obtained suggests that this theoretical method is a valid starting-point for intonation synthesis from annotated text. The effects of removing declination are also discussed.

Definitely Not Pattern Matching: A Method in Automatic Speech Recognition

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A technique for use in automatic speech recognition (ASR) is reported which does not employ traditional pattern matching techniques. The theoretical basis derives from Popper's theory of the growth of scientific knowledge, and claims to solve the problem of applying highly abstract theoretical knowledge to the problem of speech recognition. This problem lies with the construction of bridging relations between theoretical and observation languages. Existing ASR systems attempt to express abstract knowledge in observation language terms. Observation languages are incapable of describing abstract theories. The new technique is a combination of recognition heuristic and research methodology. A detailed description of the technique is presented in the form of a commentary on its application to the detection of the voicing pulse. Starting with a trivial universal theory of the voicing pulse, constraining conditions are derived which exclude all hypotheses which fail to conform to the theory. The tests are implemented as a Prolog program and the results shown. No comparisons are made by the program between prototypical patterns. A new theory is then constructed to take account of the deficiencies in the old one and the process repeated. It is demonstrated that with the growth of knowledge an increasingly greater recognition accuracy is achieved.

The Problem of Capturing Linguistic and Phonetic Knowledge

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It is generally agreed that speech synthesis and automatic speech recognition systems need to incorporate linguistic and phonetic knowledge. Ideally that knowledge needs to be just what a native speaker knows about language in general and about his own language in particular. Since most people cannot externalize their knowledge, however, ss and asr designers look to linguists and phoneticians to provide the necessary information.

This paper examines whether what a linguist knows is in fact different from what the ordinary speaker knows, and whether the linguistics and phonetics models are entirely suited to providing the necessary information. One or two knowledge formats will be discussed, with attention being focused on concepts such as 'explicitness', 'description', 'simulation', 'certainty' and 'probability' with a view to proposing a suitable format for the simulation in ss and asr of the native speaker 'expert' in speech production and speech perception.

Exploiting Physiological Codes in ASR

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The physiological code for transmitting auditory information from the ear to the brain is in the form of parallel trains of spikes responding to different frequency passbands. At low frequencies, the spike trains contain phase information and at all frequencies they carry information about changes in amplitude as well as reflecting the ongoing amplitude level. To exploit the potential of this coding system in automatic speech recognition devices we need algorithms which mimic the transduction of mechanical to neural activity in mammalian nervous systems. This paper reviews the merits of a number of existing computational procedures which model this process.

Underlying Phonetic Explanations: Representing Causal Knowledge in a Phonetic Knowledge Base

A J H Simons
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It is argued in this paper that phonetic knowledge can be put to much more effective use in automatic speech recognition than has been achieved to date. Reported phonetic expert systems typically use production-rules; hypotheses about broad classes of sounds are progressively refined; the system reasons with tokens such as acoustic cues and linguistic features. Following an established specification for the design process of expert systems, the scope of the problem is redefined to show the types of phonetic knowledge which a true expert will have to model. The conceptualization of phonetic knowledge is reappraised with a view to providing appropriate symbolic tokens, defined in markedly different domains, for reasoning in causal explanations rather than surface classifications. The kinds of formalization required are discussed, showing the degree of detail necessary and the interaction between symbolic tokens within and across the different knowledge planes.

Error Correcting Parser for Automatic Speech Recognition

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This paper reviews some of the implications of psycholinguistic studies on human speech perception and production. These indicate the necessity for an Automatic Speech Recognition (ASR) system to cope with incomplete and ungrammatical spoken commands. The studies reveal the need to incorporate semantic and pragmatic knowledge in the recognition process.

The paper also reports the development of an error correcting parser designed for an airborne ASR system. This parser attempts to resolve some of the identified problems by utilizing lexical collocational restrictions within a constrained task domain.

A Probabilistic Control Strategy for Parsing Multiple Strings on a Lattice

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When many alternative strings (in the form of a word lattice) are produced by a connected word recognition system we are faced with the problem of controlling the search space of all possible strings when attempting to parse the 'correct utterance' (1).

A further problem (having obtained a parse tree of alternative strings) is the necessity to fit a measure of reliability to the various constituents of the parse tree.

In this paper, a solution is proposed whereby probability distributions constructed from the recognition scores obtained during training are used to make probabilistic judgements of word recognition likelihood and string recognition likelihood. The probabilities are used both to queue the alternative words between nodes on the lattice and to provide the reliability measure for words and strings. A bottom-up chart parser (2) is used to perform the lattice parsing, and results are presented which indicate the performance of the method on a connected word recognition task.

1. Young, S J, *Generating Multiple Solutions from Connected Word DP Recognition,* Proc Institute of Acoustics, 1985.

2. O'Shea, T and Eisenstadt, M. *Artificial Intelligence Tools, Techniques and Applications,* Harper and Row 1984.

Word-structure Reduction Rules in Automatic, Continuous Speech Recognition

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In this paper a rule interpreter is described for deriving variant pronunciations ('Reduced Forms') from 'Citation Forms' in a 4000-word lexicon. The lexicon is part of a feature based, automatic speech recognition system and the rule interpreter has been encoded in LISP on a Xerox 1108 ('Dandelion').

The Citation Form is the phonemic representation of a maximally careful and formal production of a given lexical item in isolation; the Citation Form entry also includes an initial and final word boundary, syllable boundaries assigned on the basis of the 'maximum onset principle' (Kahn, 1976), and primary and secondary stress. Currently a small number of complex rules (embodying a large number of more simple rules) have been written which predict some of the modifications that Citation Forms can undergo in fast, informal speech production. As in Chomsky and Halle (1968), the rules consist of a *pattern* which is to be matched and an *action* which is to be taken if the pattern is matched. Pattern and action can consist of any phonemes, word and syllable boundaries and stress marks. In addition, phonemes have been defined in terms of *feature matrices* which are composed of bundles of *distinctive features* to which the rules may refer. The rule interpreter also allows optionality, conjunctivity and feature copying to be included in the rules. Using this system, it has been possible to expand the Citation Form lexicon by a factor of about 2. Implications for automatic speech recognition and speech synthesis are discussed.

An Interpreter for Phonological Rules in Automatic Speech Recognition

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A given word may have many different phonemic realizations all of which a phonemically based speech recognition system must be able to recognize as that word. Our approach to this problem is to generate *reduced* forms of a word by phonological rules from *citation* forms and then to match against the reduced and citation forms at recognition time. An interactive software tool for developing and running a set of phonological rules for this purpose is described. The user can edit rules, run rules, examine the effects on the lexicon of running the rules, and undo these effects so that he may edit and try again. Records are kept such that for a given reduced form the user can trace the path of rules to the citation form from which it was generated, as well as examine the tree of reductions generated from a given citation.

Lexical Stress, Lexical Discriminability and Variable Phonetic Information in Speech Recognition

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Huttenlocher (1984) claims that the phonetic information contained within the stressed syllable of a word is most informative relative to other portions of that word in terms of distinguishing it from rival candidates in the lexicon. Huttenlocher demonstrates this finding within a model of lexical access in which only partial phonetic information is used; specifically, 6 'broad' classes of phonemic unit (Huttenlocher & Zue, 1984). This paper describes an attempt to test Huttenlocher's findings within a model of lexical access in which variable phonetic information is used: 44 'FineClass' phonemic units, and a number of broader 'MidClass' units each of which is a superordinate grouping of several of the FineClass phonemes (Laver *et al*, 1986). A number of simulations are described in which different assumptions are made concerning where, within a word, FineClass descriptions are output by a notional acoustic front-end. Two conditions are compared, in which FineClass descriptions are output at random, and in which they are output only within stressed syllables. The results, which question Huttenlocher's claims, are discussed in relation to the recognition of continuous speech by machine.

Variable Phonological Categories and Lexical Discriminability in Speech Recognition

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This paper examines the relationship between three levels of phonological classification and lexical discriminability. The three levels under discussion are BroadClass (eg stop), MidClass (eg voiced stop, voiceless stop), and FineClass or phonemic/allophonic (eg / b,d,g,p,t,k /). Re-writing FineClass into MidClass and MidClass into BroadClass predicates a loss of information, and the power to discriminate between lexical items decreases. Lexical discriminability is examined using the notion of equivalence class, defined by Zue (1985) as 'a collection of words having the same representation'. The size of the equivalence classes at each level has implications for economy of lexicon-searching, and indicates where best to focus analytic effort. The discriminative power of a given MidClass category is calculated by re-writing one BroadClass into its MidClass members within a BroadClass lexicon. Similarly, the discriminative power of the different FineClass categories is calculated by re-writing one MidClass into its FineClass members within a MidClass lexicon. This may be of value in devising analysis strategies for incoming acoustic material.

Filtering of Phonetic Hypotheses in a Speech Recognition System using Grammatical Tag Transitions

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Many speech recognition systems have a top-end filter representing grammatical knowledge in some form. The usefulness of such a filter is generally accepted; styles in the construction of grammatical filters vary from the mathematically sophisticated and linguistically simple (eg IBM's trigram system) to the linguistically sophisticated but computationally unwieldy (eg the use of an ATN in HWIM). An attempt to steer a middle path between these extremes uses transitional probabilities obtained from a statistical survey of a corpus in terms of grammatical categories, rather than words, and distinguishes several grammatical environments in which various sets of probabilities apply. The result is a probabilistic grammar with a single layer of phrase structure, and finite state transitions between grammatical categories within phrases. Some preliminary results obtained using this system are presented.

Acoustic Variability in Velar Stop Consonants

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Velar stops are often classified as [compact], the acoustic realization of which is a single spectral prominence in the mid-frequency region (about 1-3.5 kHz). Many velar stops do not fit this description however; their detailed realization depends on phonetic context and individual speaker characteristics such as vocal tract length. This paper reports a preliminary investigation into the acoustic correlates of the distinctive feature [+compact]. Successive short-time spectra of the bursts and following releases of prevocalic velar stops spoken in various phonetic contexts by different talkers have been examined. Analyses include standard DFT and LPC spectra and the output of auditory excitation pattern models (eg Bladon and Lindblom (1981) *JASA* 69, 1414-1422, and Moore and Glasberg (1983) *JASA* 74, 750-753). The same analyses are being applied to synthetic stop bursts forming acoustic continua from /g/ to /b/ and from /g/ to /d/, for which identification functions have been obtained in listening tests. Results will comprise identification of factors that affect the degree of 'classical' compactness of velars in standard spectra, and an evaluation of the contribution of auditory transform models to the specification of the acoustic properties associated with the feature [compact].

Acoustical and Physiological Descriptions of Voice Onset

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The way in which the vocal folds start vibrating at a voiceless-voiced transition seems to be highly characteristic for a given speaker. This impressionistic knowledge has, however, never been substantiated by means of comprehensive and detailed measurements. We have recorded the acoustic speech signal and the laryngogram in a wide variety of voiceless-voiced transitions, in VCV and CVCV words, where the word stress can be on the first or second syllable. The onset of voicing will be described in both the time and frequency domain; descriptions are based either on the acoustic speech wave or on the laryngogram. The alternative descriptions will be compared in order to try and obtain an explanation in terms of the physiology of voice production.

Time-Structure Measures of Vocal Fold Vibration

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Two new measures, based on laryngographically derived vocal fold period estimation, are presented. These are the probability-density functions of:

a) periods of larynx silence (Sx), and
b) periods of voiced excitation (Vx).

Results of some preliminary investigations of these measures are discussed. These investigations had as their aims:

i) verification of the distributions,
ii) estimation of optimal input sample size for distribution stability, and
iii) establishing parameters of these distributions for normal speakers.

Analysis and Perception of the Open Phase to Period Ratio of Glottal Pulses

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Although it is widely accepted that in the source — filter model of speech production the source contributes to the naturalness of the speech output, very little is known about what changes in the source waveform occur in natural speech and how these changes affect the quality of the perceived speech.

The analysis to be described is concerned with assessing how the open phase to period ratio of the voiced excitation pulse varies with changes in pitch and vocal effort. A pitch synchronous closed phase inverse filtering technique has been developed to extract the glottal excitation waveform from natural speech. Experiments have also been performed to determine how great a change in open to period ratio is required before there is a detectable difference in the perceived speech output.

Variation in Glottal Open/Closed Phase Ratio for Normal and Pathological Female Speakers of English

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UCL and *NPL, SPAR and SPAC Groups

The realization of contrastive segmental targets is associated not only with supraglottal changes but also with structured variation in larynx activity; prosodic contrasts are also associated with qualitative variations in larynx activity in addition to basic fundamental frequency adjustments. One parameter of this laryngeal variation is the relative duration of the open and closed phases of the glottal cycle. The present work is founded on the analysis of this feature of speech and larynx activity in the speech of women for both normal and pathological production. Two types of speech material have been examined:

(1) contrasts in VCV sequences involving the alveolar consonants of English in different vocalic environments;
(2) extended stretches of speech.

In addition to standard spectrographic and simple acoustic waveform examination techniques, an electrolaryngograph signal has been used to provide the basis for 'reference analysis', giving a more detailed account of speech events than is possible on a purely acoustic basis. We present the results of analysis by specially-written programs which

(i) extract from the electrolaryngograph signal timings of the open and closed phase of the glottal cycle;
(ii) calculate and display distributions of occurrence of open/closed phase ratio values at different fundamental frequencies across each subject's range.

Open/closed phase ratio variation has marked effects on acoustic output, and the extent to which improved knowledge of such variation may help to define pathology and improve synthesis by rule and recognition algorithms is discussed.

A Peripheral Auditory Model for Speech Processing

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Following the arguments of Javel *et al* (JASA, September 1983) we have implemented a computational model of the peripheral auditory system which explicitly incorporates the frequency-, intensity- and phase-specificity of excitation and suppression phenomena.

The input waveform is first put through a bank of band-pass filters which in turn provide the input to the second stage. Each channel in this second stage receives input from a number of different filters which may be excitatory or suppressive depending on the polarity of the wave and the frequency relation between the filter and the channel. Positive-going waveforms provide variable excitation, negative-going provide variable suppression, both with appropriate threshold and saturation effects.

The output of the model will provide temporal information that is potentially useful in sound separation (Weintraub, PhD thesis 1985), and which is not found in conventional spectrograms based on the energy from a bank of linear filters.

Towards a Semi-symbolic Representation of Speech based on Auditory Modelling

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Current speech analysis techniques may not be sufficiently robust to serve as the initial processing stage of large vocabulary, continuous speech recognition systems. Auditory models, on the other hand, are more likely to provide a representation in which perceptually important features are preserved and enhanced, although work is still required to reduce the neural code to its salient form. The work reported in this paper is an extension of earlier research which resulted in a computational model of the peripheral auditory system. Contained in this new formulation are components reflecting several important auditory phenomena — basilar membrane filtering, two-tone suppression, phase-locking and short-term adaptation of auditory-nerve fibre responses. The current investigation is motivated by the desire to produce an early semi-symbolic representation of speech — the 'speech sketch'. This paper reviews the auditory model and presents some preliminary views of the first level auditory sketch.

Auditory Modelling for Automatic Speech Recognition

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It is known that the performance of many speech recognizers is limited by the ability of the initial signal analysis to completely and accurately represent the 'important' characteristics of that signal. The human auditory system, however, provides sufficient information for very reliable recognition, and this paper describes a computational model of such a system, written in FORTRAN. This implementation is based on the model developed by Richard F. Lyon at the Fairchild Laboratory for Artificial Intelligence Research, simulating the auditory process from the middle ear through to the first level of neural firing. Results will be presented, showing the high temporal and spectral resolution offered by this model when driven by both synthetic and natural stimuli. This increased resolution should allow recognition algorithms to identify speech sounds more accurately and reject extraneous noise more effectively.

Identification of Similar-sounding Speakers by Means of an Auditory Long-term Spectrum

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Similar-sounding speakers (controlled for accent, age, stature, occupation and domicile; seven of them male and seven female) were recorded twice, three years apart. Long-term spectral analyses were carried out on both the 1983 and 1986 samples of their speech, using not the acoustic spectrum (nor some rather arbitrary parts of it), but an 'auditory spectrum' transformed to reflect certain time-independent properties of peripheral auditory analysis. Measures of congruence of auditory-spectral shape, and of auditory-spectral tilt, were obtained. Results indicated some support for the ability of these measures to achieve better discrimination among contemporary samples of different speakers than between time-separated samples of the same speaker. However, contaminatory effects such as changes in recording conditions, or the differential use (on one occasion but not the other) of breathiness, could sometimes have overriding consequences.

Towards an Auditory Primal Sketch

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The application of psychological ideas to the problem of automatic speech recognition is often advocated but rarely practised. This paper outlines the use of ideas developed by Marr (1982) in vision to segment the auditory input stream. Lessons learned from the identification of colours under different illuminations are also applied to the labelling of the derived segments. The algorithms are embedded in a single word recognizer implemented on a BBC micro which will be demonstrated. The input to the system is a zero cross count and the energy in three frequency passbands which are sampled at only 100 Hz. The principles could be applied, however, to a wide range of potential inputs.

Phonetic Tokens for Symbolic Reasoning: an Appraisal of Distinctive Feature Theories

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Phonetic expert systems for automatic speech recognition need to be able to reason symbolically with tokens corresponding to natural classes in the acoustic, articulatory, auditory or linguistic domain. Distinctive Features have traditionally fulfilled such a function in linguistics. This paper describes the roles that symbolic reasoning tokens would be expected to play in an expert system and examines the pedigree of various descriptive and feature systems in this light. Traditional phonetic descriptions and three well-known Distinctive Feature systems are discussed. They are found wanting in various respects: sometimes they make the wrong classifications of speech-sounds, they cannot be guaranteed to provide a unified evaluation-metric for expert system performance and, most significantly, they fail to provide a fine-grained causal explanation of underlying phonetic behaviour. Examples of symbolic reasoning tokens appropriate to different domains are proposed, together with examples of how knowledge sources might interact with them in the process of automatic speech recognition.

Fluctuation Phenomena in Underwater Acoustics

International Conference organized by the Underwater Acoustics Group

9-10 December 1986, Weymouth

Invited Paper: Successes and Failures of Multiple Scatter Theory

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Data from accurate ocean acoustic experiments has led to the rapid development of multiple scatter theory over the last five years, and has allowed this theory to be tested. An important achievement of recent theory has been to explain the time spectra of acoustic intensity fluctuations caused by internal waves.

Recently much effort has been devoted to investigating the accuracy of approximations used in dealing with point sources. This has been prompted by the desire to explain residual disagreement between theory and experiment. The use of curvilinear co-ordinate systems has helped to resolve this question and has also allowed the extension of the theory to the case of heavily curved ray paths.

At the same time the numerical simulation of wave propagation in a randomly scattering medium has advanced to the stage where theoretical predictions for point sources can be tested. Simulations can be performed even in curvilinear co-ordinate systems, yielding valuable insight into the spatial behaviour of intensity fluctuations in the real ocean. Thus both theory and simulations of the intensity fluctuations due to a monochromatic point source can successfully explain available experimental data.

A notable exception, however, has been the failure of theory to deal adequately with the cross-correlation of intensity fluctuations at different wave frequencies. Investigations have shown that while the basic multiple convolution solutions are still accurate in the two frequency case some fundamentally new method will be needed to evaluate these expressions.

Scattering of Waves by Refractive Layers with Power-law Spectra

E Jakeman and J H Jefferson
RSRE, Malvern

We investigate the statistical properties of waves which have been scattered by a layer of random medium whose gradients of refractive index are self-affine. This problem is of interest not only as a realistic model for certain scattering systems, but also because of its analytical and computational simplicity. The case in which the scatterer is composed of independent increments is particularly simple and we have shown that it can be solved exactly over the whole near to far field region without further statistical assumptions. More generally, the effect of spatial integration of ray density fluctuations at the detector has been investigated. We have shown that in the limit where the refractive index becomes smoothly varying, spatial integration over geometrical singularities is similar to diffraction smoothing, giving the same structural form for the scintillation index as a function of propagation distance. This provides further insight into the properties required for any model for the statistics of the scattered wave amplitude to be valid over the whole range of propagation distances.

Scattering of Acoustic Waves by Turbulence

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The scattering coefficient of the mean acoustic field has been calculated for random turbulent media, using a Green's function approach. The correlation function of the medium inhomogeneities due to turbulence has been described by the von Karman function. Using the Bourret approximation (corresponding to fine grained random media), the dependence of the scattering on the parameter ν in the von Karman function has been calculated. The scattering increases monotonically with both ν and ka where a is the correlation radius.

Computation of Ray Statistics beyond a Multi-Scale Diffuser

J H Jefferson, E Jakeman and J E P Beale
RSRE, Malvern

When high-frequency waves are refracted through a narrow region of turbulence or are reflected from a rough surface, the statistics of the emanating ray-directions are determined uniquely by the statistical properties of the surface. In this paper we analyse in detail a multi-scale model with finite inner scale and show how the statistics and correlation properties of the rays evolve as we move away from the surface, a purely geometrical effect. Our model is solved exactly and we derive expressions for the factorial moments and correlation function of the rays as functions of distance from the surface. These results are compared with numerical simulations and are presented graphically together with ray-diagrams which show very simply how the statistics of the rays scale with distance.

The Description and Simulation of Correlated non-Gaussian Noise Processes

R J A Tough
RSRE, Malvern

The description of noise processes in terms of Fokker Planck (FP) equations and their equivalent stochastic differential equations will be discussed, with special attention being paid to the role of multiplicative noise in determining the static statistical properties of the process. After discussing several simple illustrative examples the description of the correlated K distributed noise process (which has found extensive application in the study of propagation through random media) will be considered in more detail. The relationship between the FP model and the birth-death-migration model of Jakeman will be discussed. The analysis will be extended to include the effect of mixing the noise process with a coherent signal ('homo-dyning') and provides us with a useful model for propagation through an extended random medium. Complementary simulation methods, based on the numerical integration of stochastic differential equations, will also be described and their results used to illustrate the discussion.

A Numerical Approach for Rough Surface Scattering

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In this paper a method is described for dealing with acoustic scattering from a time-independent pressure-release surface. As in previous work, the problem can be formulated as a boundary integral equation on the surface. In principle, this formulation takes account of arbitrarily large surface roughness and multiple scatter phenomena at the surface. It is hampered, however, by the high resolution necessary to adequately resolve the wave interaction with the surface.

The problem is re-formulated as an integral equation in frequency space. This can be discretized and solved for a given arbitrary surface. Previous work with sinusoidal surfaces indicates that this approach may be a more natural one. The problem is then solved for many realizations of a surface obeying Gaussian statistics. Averaging such results allows the evaluation of statistical averages of the acoustic pressure field. Numerical results will be given for various surfaces and a comparison made with previously obtained solutions for a sinusoidal surface.

Invited Paper: Fluctuations of Signals and Noise in the Sea and their Effect on Sonar Target Detection

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Fluctuations of underwater sound may be said to be either deterministic or stochastic. The former are generally of long period and can be predicted quantitatively from our fund of knowledge about sound in the sea; the latter are of short period and are predictable only statistically. Based upon the idea that the sound received from a steady source consists of a mixture of a steady and a random component, the Rician, or generalized Rayleigh, distribution is obtained for the fraction of the sample of the level of a received signal above or below the average. Such curves have been verified by a variety of field data. Also, both theory and observation show that ambient sea noise at the output of a detector has a chi-square distribution with degrees of freedom equal to the bandwidth-time product. As far as detection is concerned, it is easy to see that long-range targets are detected better during signal-to-noise surges than they would be in the absence of fluctuations; on the other hand, short-range targets tend to be lost during moments of fading. Quantitatively the effect of fluctuations on detection can be expressed by 'transition curves' of detection probability against signal excess that depend on the fluctuation law involved. Fleet exercise data yield transition curves indicating that in the real world the signal-to-noise ratio is log-normally distributed, with a standard deviation of 6-8 dB, as a result, apparently, of the Central Limit Theorem applied to the sonar equations.

The Effects of Fluctuations on Coherent Synthetic Apertures

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Producing large aperture sonars for high resolution seafloor imaging has always been made difficult by the random fluctuations inherent in the ocean. Nowhere is this effect more critical than in the operation of a truely useful synthetic aperture sonar.

Synthetic apertures based on short duration pulsed transmissions have so far not produced a viable SAS and the major reasons for this lack is the effect of ocean turbulence on the phase coherence across the aperture.

The effects of turbulence may be minimized by using some form of continuous transmission continuous reception sonar coded in some way to retain information about the range of the targets and a sonar which is inherently insensitive to path length variations.

A twin receiver CTFM sonar (Continuous Transmission Frequency Modulated) is now being evaluated as a side looking synthetic aperture sonar and recent results are presented.

Sea Surface Wave-Height Effects on Shallow Water Matched-Field Processing

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Sea surface wave-height fluctuations cause a mismatch in environmental conditions and therefore affect the detection and localization performance of a matched-field processor. A sensitivity study of this mismatch phenomenon was performed in an idealized, range-independent, Pekeris channel at 150 Hz with a nominal water depth, D, of 100 m covering 150 m of sediment with an effective critical angle of 22 deg. A 50 m deep source was placed at a range of 2.65 km from a 21-hydrophone equally spaced vertical line array covering half the water column (from D/4 to 3D/4). Variations in the sea-surface height of ±3.5 m were considered as an extreme, but realistic, case and the output signal-to-noise ratio (SNR) and predicted range and depth of the source were determined from a series of range-depth maximum likelihood ambiguity surfaces. Surface height variation caused systematic errors in range estimations, such that when the depth was increased due to a wave-crest, the target range appeared shorter than the actual range. The opposite held for a wave-trough. In the case studied a maximum range error of ±7.5% was observed. The corresponding calculations of target depth were consistently biased towards greater depth, with an average error of 2%. Calculations indicated that small variations in surface height (~1%) can cause losses of 15 dB in detection performance. The ambiguity surfaces corresponding to the various wave heights were averaged and the resultant degradation in SNR was 6-9 dB. The average range and depth errors on the composite surface were approximately 5% for 0dB input SNR, and 0% for 15 dB input SNR.

Invited Paper: Remote Sensing of Astrophysical Plasmas using Scintillation Methods

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Radio waves from astronomical sources propagate through a variety of inhomogeneous media in travelling to the Earth's surface. These include the intergalactic gas, the interstellar medium, the heliosphere and solar wind, and the ionosphere. Observations of suitable sources which irradiate the media with sufficient coherence reveal scintillation phenomena caused by all zones except the first. Fortunately the timescales of the fluctuations are usually widely separated so that effects due to the different media are not confused. Detailed observation and analysis of the spatial, temporal and radio frequency structure of the scintillation have provided information both on the media and on the sources which is often unattainable by other methods.

Such information includes the wavenumber spectrum of density inhomogeneities in the interstellar plasma, the spatial velocities of pulsars, the distance between emitting zones in a neutron star magnetosphere, the speed of the solar wind at high heliolatitudes, the mapping and tracking of interplanetary disturbances responsible for geomagnetic storms, the angular sizes of quasars etc.

This paper will present a broad review of the achievements and principles of scintillation methods as an observational tool in astrophysics.

Underwater Laser Systems

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The absorption of electromagnetic radiation by natural seawater is extremely high across the complete spectrum except in three narrow regions. These regions are, firstly the cosmic ray region and secondly, at the opposite end of the spectrum, the extremely low frequency region where wavelengths are typically several kilometres. There is also a dip in the attenuation in the blue/green part of the visible region of the spectrum between 450 nm and 550 nm, and it is in this region that the Laser Systems Department is specifically interested.

This paper is intended to be an overview of the underwater uses and techniques of lasers which operate in the blue/green region. The various laser sources available such as frequency doubled Neodymium-Yag lasers, Raman-shifted Xenon-Chloride lasers and Mercury Bromide lasers, will be described and compared. Also, the systems in which these sources can be incorporated will be discussed. Such systems include underwater range-gated LIDAR, laser enhanced imaging, and sub-sea communications.

In addition to the above, the paper will discuss some of the problems of the propagation of laser light in water where scattering and absorption are important. The results of laboratory work on underwater LIDAR will also be presented.

Numerical Calculations of the Propagation of Laser Light through Water

D B Radcliffe and R L Cooke
British Aerospace, Sowerby Research Centre, Bristol

In any systems analysis for underwater laser systems it is imperative to accurately model the propagation of the laser pulse from the source through the scattering medium to the receiver. This is so because of the exponential attenuation law where small uncertainties in the attenuation coefficient lead to large errors in the predicted laser power requirements for a given range. Analytic techniques can be used to predict beam spread and attenuation only in cases where multiple scattering does not occur. Numerical Monte Carlo techniques are therefore more powerful.

In this paper we will outline the usefulness of Monte Carlo codes with particular reference to a specific code developed at NOAA, where the approach is to model the transport of photons through the water as a series of individual scattering and absorption events. The mean free path and the scattering angle are chosen randomly but weighted to the various inherent properties of the water such as the volume scattering function and the absorption coefficient. The code therefore predicts the spatial and temporal spreading of an initially narrow laser pulse.

We also discuss the propagation of laser pulses through water containing air bubbles. The scattering properties of a single bubble are well understood allowing algorithms to be constructed to model, by Monte Carlo methods, pulse propagation through a medium containing a distribution of bubbles of varying size.

Invited Paper: Acoustic Propagation and Ocean Internal Waves and Finestructure

T E Ewart
Applied Physics Laboratory and the School of Oceanography, University of Washington, Seattle, USA

We have progressed in our understanding of wave propagation in random media to the point where theoretical predictions of the fluctuations in sound waves propagating through a medium with a known autocorrelation function are quite accurate. The analytical predictions of the range-depth-time phase and intensity fluctuations agree very well with results from numerical solutions of the parabolic wave equation for propagation in media with the same correlation function. Measurements of the complex amplitude of sound waves that have propagated through ocean regions where the statistics of the media are well measured provide tests of the theories. In this review the models of the acoustic index of refraction correlations that result from oceanic tides, internal waves, and finestructure processes will be discussed. Comparison of measured and predicted acoustic fluctuations will be presented to indicate the level of our understanding. Our knowledge of the scattering processes allows us to consider the problem in an inverse sense. That is, we use the equations of scattering to obtain a stochastic inverse from acoustic data. Examples will be given from the Mid-Ocean Acoustic Transmission Experiment, MATE, of both the forward problem and the inverse problem. Models of the oceanic internal wave, finestructure and tidal processes obtained from the measured acoustic fluctuations, are compared with those measured. A discussion of the future expectations from stochastic inverse analysis will be given. Work supported by the Office of Naval Research.

Phase Fluctuations due to Acoustic Propagation through a Turbulent Tidal Flow

D Farmer
Institute of Ocean Sciences, Sidney, Canada
J Verrall
Department of Physics, University of Victoria, Canada
S Clifford
National Oceanic and Atmospheric Administration, Colorado, USA

Sound travelling through a time-dependent inhomogeneous medium such as the ocean is subject to various distortions which can lead to fluctuations in phase and in the angle-of-arrival measured at some distant receiving plane. We consider the effects of two possible sources of such distortion: first, the random fluctuations caused by fully developed turbulence and second, the more slowly evolving variations caused by changes in the mean component of current normal to the acoustic path. Random fluctuations of short period may be interpreted in terms of the time dependent random components of the refractive index field. In weakly stratified, tidally stirred water, the smaller scale fluctuations may lie within the inertial subrange, in which case the spectrum of angle-of-arrival fluctuations can be related in a simple way to a refractive index structure parameter describing the sound speed perturbations. A quite different effect occurs as a result of a mean flow perpendicular to the acoustic path. In this case the acoustic waves are advected by the flow, resulting in a change in the mean angle-of-arrival, the sin of which is equal to the Mach number of the flow.

An experiment using a fixed path across a tidal channel provides an opportunity for studying these effects. The spectrum of fluctuations in the angle-of-arrival is consistent with an inertial subrange model for the refractive index field. Distortion of the acoustic path by the mean flow produces a small, but measurable effect, quite consistent with independent current measurements. The results discussed here provide both an explanation of acoustic effects in terms of environmental parameters and conversely indicate possible approaches by which such measurements may be inverted so as to probe the intervening medium.

Observations of Temperature, Salinity and Sound Speed Microstructure on the UK Continental Shelf

P F Dobbins
British Aerospace, Underwater Research and Engineering Unit, Weymouth

Microstructure measurements were carried out through the summer of 1986 at various locations on the continental shelf around the British Isles. High spatial resolution vertical profiles were taken using an instrumentation package carrying temperature, salinity, sound speed, depth, two-axis current and other sensors. The current velocity fluctuation data were analysed to produce profiles of the rate of kinetic energy dissipation as a function of depth, and the horizontal structure was measured at those depths showing high dissipation rates using a specially built towed fish carrying temperature, salinity, sound speed and depth sensors. In this paper, examples are presented of the records obtained along with the computed spatial correlation functions and wavenumber spectra. The implications for sound propagation in shallow continental shelf waters are discussed.

Fluctuations in Acoustic Transmission Loss: Comparisons between Measured Data and Predictions of the Safari/Parabolic Equation Numerical Models

H B Ali and G J Tango
Naval Ocean Research and Development Activity, Ocean Acoustics and Technology Directorate, NSTL, Mississippi, USA

The results of acoustic transmission loss measurements in a shallow water area of the North Atlantic have revealed significant fluctuations in the acoustic intensity levels received by a 60 km-distant vertical array of hydrophones. The attempt to attribute these fluctuations to easily identifiable properties of the medium is frustrated by the complexity of the latter. An added complication arises from the location of the measurement site in a geographical area characterized by the convergence of two major water masses of differing environmental properties, viz, Arctic water and Atlantic water. Moreover, the resulting Polar oceanic front is not stationary, but, rather, oscillates with semi-diurnal tidal periodicity.

Propagation factors tentatively identified as contributors to the acoustic fluctuation include the steep gradients in environmental parameters arising from the different water masses, changes in water depth, currents, and possibly other effects. Numerical studies were performed in an attempt to separate the various effects and, in particular, to assess the relative contributions made by the tidally advected changes in water masses, *per se*, and the concomitant changes in other environmental parameters. In particular, using simplified models of the test environment, computations were made using SAFARI FFP and a wide-angle version of the Parabolic Equation model (IFDPE). Although neither model by itself is entirely adequate for the environment considered, both range-dependence and ocean-bottom shear being of interest, the combination does provide some insight into the suspected phenomena.

Experimental Work on Acoustic Fluctuations of Broad-band Sound in the Sea

J R Potter
Saclant ASW Research Centre, La Spezia, Italy
S Miller
DAMTP, University of Cambridge

In October 1985 an experiment was carried out in the Tyrrhenian Sea to investigate acoustic fluctuations over a propagation distance of 4 km. A broad-band source was employed, transmitting significant acoustic energy between 100 Hz and 2 KHz. This is the first such experiment to use broad-band sound, which allows cross-frequency correlations to be calculated. Some 250 transmissions were made, at various intervals between 15 mins and 1 hour. A linear, vertical, 32 hydrophone array with 2 m spacing was moored 250 m below the surface to record the intensity of the sound field due to direct arrivals. The primary aim was to record both spatial and temporal intensity fluctuations. Absolute arrival time fluctuations could not be measured, but relative arrival time differences down the array could be seen and have been shown to agree with results derived from multiple scattering theory. Simultaneous oceanographic observations of the temperature and salinity field were made by a towed oscillating body which enables a direct estimate of the sound speed inhomogeneity scales to be made. Multiple scattering theory is then applied with a known spatial correlation for the medium. Internal wave activity is found to be roughly comparable with other sources of sound speed inhomogeneity.

Amplitude Fluctuations of High Frequency Underwater Acoustic Signals in the Marginal Ice Zone

J W Posey and M A Wilson
Naval Ocean Research & Development Activity, NSTL, Mississippi, USA

Amplitude fluctuations of high frequency volume-propagated acoustic pulses were measured in the Fram Strait marginal ice zone. Sources at 8, 9, 10, 11 and 13 kHz were suspended independently at a depth of 60 metres. Twelve hydrophones were suspended at the same depth and a horizontal array of eleven receivers was suspended at 12 metres. Sources and receivers were up to six kilometres apart and moved with the ice. This gave data from a wide variety of ranges and azimuths but complicated the signal processing required. Range data was updated every 30 seconds with pulses being emitted four to eight times per minute. The dependence of fluctuations on frequency, range, and direction of propagation are investigated. Spatial and temporal properties of the fluctuations are explored. Representative autocorrelations, coefficients of variation, and other statistical information are presented. In a 20 minute sample set, large random variations in amplitude are found even at short ranges. Fluctuations of sound speed on this time scale were not measured directly, but the acoustic data imply a normalized variance, $\langle(c - c_0)^2\rangle/c_0^2$, on the order of 10^{-7}.

Spatial Coherence of a Source Received on a Seismic Towed Array after Long Range Propagation

A Plaisant and S Leroy
Thomson Sintra, Activités Sous-Marines, Cagnes-Sur-Mer, France

A propagation and spatial coherence experiment was carried out in deep water with a moving source transmitting long constant frequency pulses received on a 96-channel seismic towed array after propagation to distances up to 2000 kilometres.

Results are presented concerning spatial coherence, its effect on array signal gains with classical and optimum beamforming, and the behaviour of these gains depending on array length used and propagation distance.

Speech Input/Output: Techniques and Applications

A MEETING of especial interest to the Institute of Acoustics was held recently in London. It was entitled **Speech Input/Output; Techniques and Applications** and took place on March 24-26. Although the administration was under the auspices of the Institution of Electrical Engineers, it is probably fair to say that the main driving component came from the Speech Group of the Institute of Acoustics, since this group had the greatest representation on the organizing committee. In addition to the IEE and the IOA, four other sponsoring societies were involved: British Society of Audiology, College of Speech Therapists, the Institute of Mathematics and its Applications, the Institution of Electronic and Radio Engineers, and the Royal Association for Disability and Rehabilitation.

Altogether about 250 participants and exhibitors attended, coming from some 17 different countries. Australia and China were the most distant parts of the globe represented and, perhaps most obviously to workers in speech, France, Holland and Sweden were most numerous in the foreign contingent. Altogether 65 papers and poster presentations were made in 17 sessions.

The main aim of the meeting was to present and discuss new work in the field of man/machine communication using speech as the means of input, output and control. Ordinarily this type of meeting serves either one of two very separate groups of interest. Commercial, industrial and military aims are represented in one type of gathering whilst the needs of those working for the deaf, speech disabled and physically handicapped, for example, are served by another. On this occasion, both groups were united in a programme which was deliberately designed to bring them together. At first sight this grouping of interests may seem to give a potential advantage to the handicapped since future developments in Speech Sciences, signal processing, analytic methods and VLSI techniques seem most likely to come from the areas of greatest financial investment. As Mr Jack Ashley MP foreshadowed in his excellent opening address, however, many discussions and papers at the meeting showed that this was far from true. Indeed, some of the main presentations were concerned with commercially important developments but had their origins, at least in part, in the field of work for the handicapped. What, after all, are more handicapped than today's computer based 'speakers' and 'listeners'?

During the meeting investigations were discussed of, for instance, phonological structure, phonetic/acoustic transformations, speaker idiosyncratic features, and auditory constraints on speech receptive processing. These matters seem far removed from the immediate needs of speech technology but they gave some insights into the nature of our technological handicaps — as well as of the need for greater fundamental knowledge. The present use in speech technology of mathematically sophisticated diphone synthesis, and dynamic time warping based recognition applications, are in reality very removed from the human condition. However, to be able to define problems that are hardly addressed at present in current work, is also to define some of the directions in which future sources of improvement may be sought.

For the UK participants, the start of the Alvey programme of combined support for industry and university work in speech made the meeting timely — problems are best defined at the beginning — and it was encouraging to see the first fruits of this work and the shaping of further collaboration, both in the UK and with 'Europe'.□

A J Fourcin

New Materials and their Applications

The Institute of Physics is arranging a conference on this theme at the University of Warwick from 22 to 25 September 1987. The aim of the conference will be to provide a forum for scientists and technologists concerned with new ideas and areas of applications in the field of materials. Topics will include: Advanced materials; Joining of materials; Microstructure and its relationship to properties; Interfacing of materials; Materials forming techniques; Materials in a hostile environment; Materials characterization, including NDT; Electrical, electronic, magnetic, acoustic and optical properties; and Health and safety aspects of materials.

Contributed papers are invited on the above topics and associated areas. Offers of papers, with 2 copies of abstracts (300 words) on one side of A4 paper, should be sent by or before 16 March 1987 to Dr S G Burnay, AERE Harwell, Didcot, Oxon OX11 0RA. Further details of the conference and associated technical exhibition may be obtained from the Meetings Officer, The Institute of Physics, 47 Belgrave Square, London SW1X 8QX. The IOA is among the co-sponsors of the meeting.□

Diploma in Acoustics and Noise Control 1985

Titles of Diploma Projects

Colchester Institute

The formulation of a standard for new residential development affected by road traffic noise
Leisure centre plant room noise
A comparison of room noise levels using diffuse field and modified direct path theories
Sound insulation in conversions
Noise problems with a mixed industrial — residential area, and mobile refrigeration units
Lorry bans; sleep disturbance and noise assessment
The silencing of a small portable generating set
The legal and social consequences of issuing a public entertainments licence in relation to noise
Noise associated with a non-standard computer installation
Low frequency performance of air duct silencers

Cornwall College

A rotating platform
Presbycusis — or the diminution of hearing with age
Is there a relationship between hearing loss and the sexes?
Rotating microphone boom
Portable noise generator
To test a procedure and method for relating PNdB to peak dB(A)
Wind turbine generator

Leeds Polytechnic

Investigation of noise from a refrigeration compressor unit together with noise control techniques
Evaluation of the environmental impact of a paper corrugating machine
Investigation into noise caused by a milking machine on television sound recording
Planning and noise
Noise control survey at Maltby Parkgate Project

Liverpool Polytechnic

The use of excedence levels for the rapid assessment of facade insulation
Room acoustic characteristics as determined by calculation from the established room radius
Provision of an acoustic enclosure for the CEL 160 Graphic Recorder
Subjective and objective aspects of recording studio design
A short method for determining the airborne sound insulation of a partition

cont . . .

Perceptual and Signal-processing Approaches to Separating Speech from Noise

Meeting of the Speech Group held at the University of Nottingham on Friday 25th July 1986

This was a well-attended meeting. Sixty people heard 10 papers in a tightly packed afternoon session. Five papers were motivated by practical needs in signal processing and five by more fundamental concerns in speech perception and psychoacoustics, reflecting the two orientations in the title. Several factors underlie this upsurge of interest in the problems of retrieving speech signals from noise. They were well-represented in the papers presented. First, over the last 10 years several new markets for 'speech enhancement' have been identified, to be added to established interests in communications engineering. For example, forensic signal processing has emerged as a practical discipline. In addition, automatic speech recognizers are needed that can operate successfully in natural, thus noisy, environments, along with hearing aids that attenuate background noises while amplifying a target voice. These three interests have much in common. Each would benefit from algorithms for de-reverberating speech (Curtis), for separating the speech of concurrent talkers (Stubbs), and for tracking the fundamental frequency (Sutherland) and the formants (Duncan) of a voice in noise.

In parallel with the emergence of these practical interests, basic understanding of the spectro-temporal attributes that enable speech signals to be understood in quiet surroundings has reached the point where it is informative to ask detailed questions about how the attributes are registered when interfering sources of sound are present. In particular, how are sounds produced by one talker grouped together and segregated from background noises, so that elements of the background noise are not interpreted as elements of the talker's voice? Three papers addressed this question (Pattison, Roberts, Moore). A fourth (Smith) discussed the possibility that judgements of rhythm may show how coherence in speech streams is specified and registered in noise.

The two orientations, practical and fundamental, have come together at a time of huge increase in the availability of computers with sufficient processing power to make it necessary to devise efficient means for evaluating computationally intensive algorithms with large amounts of speech test materials (Pratt), and to allow models of auditory and perceptual processes to be tested computationally (Meddis). Given these motivations, the next ten years should see a further convergence of perceptual and signal processing approaches to separating speech from noise.

Titles and Abstracts of the papers presented at the Meeting are given on pages 13-14 of this Bulletin.□

Quentin Summerfield

Derbyshire College

Construction site noise — the effects of EEC Directives
Calculation of Road Traffic Noise: Leicestershire
Noise problems in workshops operated by British Coal
Opencast mining operations — noise considerations
Noise produced by axial main mine fans
Fingerprinting noise
Noise abatement zone procedures applied to a foundry
Heavy lorries — the environmental impact
Occupational noise exposure in the drop forging industry
Calculation of road traffic noise
Design considerations for new dwellings close to a motorway
Airport noise monitoring
Ventilating and air conditioning systems
Motorcycle noise
Exposure to noise from woodworking machinery
Diesel engine noise
Reduction of diesel engine noise by close-fitting shields
Baseline noise study for the proposed S Warks coalfield
A method for calculating noise breakout from factories

Newcastle Polytechnic

Acoustical insulation afforded by walls and floors in new housing estate
Acoustic properties of Trinity Hall
Sound insulation tests in converted town houses
Noise problems associated with refuse collection vehicles
Planning implications of railway noise
Appliances and domestic noise
Ambient noise levels around proposed open cast coal site
Conversion of nuclear physics laboratory into lecture room
Acoustic assessment of Middlesborough Town Hall
Environmental impact of noise from an oil rig fabrication yard
Acoustic absorption — standing wave and reverberation methods compared
Wayside noise from high speed trains

North Staffordshire Polytechnic

Reduction of the sound level of a vibrating sieve
'Is a car a point or a line source?'
Reduction of noise in a glass toughening factory
Vibration and noise characteristics of the outer casing of a 500 MW generator
Exposure to noise of persons employed in epoxy paint manufacture
Investigation of components of a record turntable
Practical assessment of sound insulation in lightweight construction timber-framed municipal flats
Examination of the acoustic performance of a motor cycle silencer
Noise control in a washing-off room

cont . . .

British Medical Ultrasound Society Annual Meeting

The 18th Annual Scientific Meeting of the British Medical Ultrasound Society will be held from 16 to 18 December 1986 at the University of Warwick, Coventry. The meeting is attended by over 500 delegates each year and includes a major Commercial Exhibition. Details may be obtained from: Mrs L Blench, General Secretary, BMUS, 36 Portland Place, London W1N 3DG, Telephone: 01-636 3714.□

Erratum

An error crept into the list of officers of IOA Branches and Groups published on page 32 of the July 1986 issue of Acoustics Bulletin. Please amend the details for the Secretary of Speech Group to read as follows:

Dr N M Brooke
Department of Computing
University of Lancaster
Engineering Building
Bailrigg
Lancaster LA1 4YR

BRANCH AND GROUP NEWS

Scottish Branch

In accordance with the decision taken at the last AGM, the Scottish Branch has had a quiet year but plans are well in hand for the 1986-87 programme. A visit was arranged in September to the recently re-designed studios of BBC Scotland in Glasgow.

The next AGM will be held at Heriot-Watt University in January 1987 and it is planned to precede the AGM with a technical meeting on environmental noise in the hope that this will attract a good attendance from all parts of Scotland.

North West Branch

Summer Social

With no tinsel, holly or mistletoe to be seen, the branch finally held its Christmas social event this July, when a total of 44 branch members and guests came on an evening cruise aboard the narrow boat *Castlefield*. We sailed along the Bridgewater canal from Worsley to Leigh as far away as possible from the noise of modern day life; this presented a view of industrial archaeology rather than one of scenic beauty. The canal at Worsley is coloured a deep orange due to the iron ore from the Duke of Bridgewater's mines, giving way to the more usual canal colour a few miles along the route. Near Leigh we passed several slightly decayed mills which remain from the industrial revolution, and now appear to lie empty. The *Castlefield* is the longest narrow boat on the canal with an overall length of 72 ft. Turning this large boat around in the canal basin at Leigh was quite a feat.

There was a bar on board and light refreshments were served during the evening. Hardly a word on acoustics was mentioned except for comments on the poor quality of the public address system, clearly a job for an acoustician here!

Evening Meeting

In November we have the pleasure of welcoming Dr Peter Thorne of the Institute of Oceanographic Sciences at Bidston, who will be talking about the acoustics of marine sediments. Peter gave the A B Wood Medal lecture at this year's Spring Conference and I for one am looking forward to hearing the latest developments in his field.

Chris Waites

Physical Acoustics Group

The Physical Acoustics Group of the IOA and IoP remains very active although still without adequate representation by IOA members (please contact Cathy Mackenzie or Geoff Kerry if interested). Meetings have been arranged on Ultrasonic Scattering from Bones (11 November 1986), Acoustic Microscopes (24 February 1987) and Microwave Acoustics and Acousto Optics (May 1987); and Ultrasonics '87 is to be held in London in July 1987.□

Material for the January issue of Acoustics Bulletin should reach Mrs F A Hill at 25 Elm Drive, St Albans, Herts AL4 0EJ, no later than Friday 14 November.

North East Surrey College of Technology

Investigation of the relative impact sound insulation of materials when used as the resilient layer in a platform floor
An investigation of noise and vibration from boiler plant
Survey of occupational noise exposure derived by stand-by diesel generators in the City of London
Investigation to establish the source, and suggest a remedy to noise arising in Council owned dwellings in a low rise block of flats
An investigation into the diffraction of sound at three different frequencies around a solid barrier
Comparison of impact and airborne sound insulation of the floors of three different types of dwellings
A case study: sound transmission between offices and music rooms in a school
An investigation into the sound insulation properties of a floor
Assessment of the sound insulation of floors between three pairs of flats using the Building Regulations 1976 and 1985
Swimming pool acoustics — a comparison
An investigation into the acoustic properties of 2 car wash/dryer systems and the effectiveness of differing sound insulation treatments
Investigation into sound insulation of floors between living rooms of 2 flats before and after some remedial measures had been carried out
A hush kit for the departmental vacuum pump
Noise problems associated with disco/live pop music from a function room in a public house
Investigation into the problem of achieving musical clarity as experienced by symphony orchestras using main vestry hall in Chelsea Old Town Hall
Assessment of early morning noise disturbance from heavy vehicle movements in a residential area
The Coach House, Farningham, Kent. An acoustic assessment, including its suitability for use for various types of entertainment
An investigation into problems of sound insulation between offices at the Civic Centre, Gravesend
Sound power measurement investigation
The impact of railway noise upon proposed housing development
Investigation of traffic noise and sound insulation properties of existing and improved facades at Flat 39a Malden Way, New Malden, Surrey
An investigation to see if there is likely to be a risk of hearing loss from the use of in-car entertainment systems
An investigation into the problem of noise from computer printers in offices
Investigation and assessment of Go-Kart noise, Camberley Go-Kart Track, Blackbushe Airport, Yateley, Hants
The development of Honeywood Industrial Park
An investigation into the noise levels produced by a Jet Provost during a ground engine run
An investigation into the effects of the Channel Tunnel Rail Link in the London Borough of Wandsworth
A comparison of ear defenders
Investigation of noise from a boiler room affecting an adjacent residential property
A noise study of an air handling unit, London Borough of Tower Hamlets
A report of an investigation into road traffic noise
Investigation into noise emissions produced by a petrol lawn mower and the effectiveness of its exhaust gas expansion box
Assessment of road traffic noise along a proposed new relief road serving the expanding Ferry Terminal at Ramsgate, Kent
The attenuation of road traffic noise. A study to compare the acoustic performance of earth bunds with close boarded fences
Nuisance associated with an MSE ultrasonic disrupter
To survey some of the noise emissions during the initial construction of the new underline bridge at Kingston Station
Investigation into noise exposure and the possible risk of hearing impairment to the operator of a Sanitary Landfill Compactor
An investigation into noise from a domestic swimming pool
A study of noise emission and control associated with a gluten and starch processing factory located near residential accommodation

Tottenham College

Noise annoyance from discotheques
Analysis of objective clicking tinnitus
The control of noise emission from a clay pigeon shooting school within Epping Forest District
A local authority investigation of an alleged noise nuisance
Washing machine noise
Noise in operating theatres
The assessment of railway noise affecting a proposed residential development
An investigation into the control of noise from automatic car wash fan dryers
Machine noise in the home
An investigation into the effects of varying environmental conditions on the propagation of noise from a motorway
An investigation into the removal of architectural features from a tower block in the London Borough of Newham
Traffic noise prediction
Determination of the variation of SRI of a test sample due to its position within a test aperture
Sound power levels in a working environment
Noise in woodworking and joinery shop
Approximation of early/late energy index
Measurement of the effects of meteorological and ground conditions on a sound source over 200 m

Members having a personal interest in finding out more about any of these projects should initially contact the course tutor for the Diploma in the appropriate college. Further contact is at the discretion of the student concerned; it is not normally possible to provide copies of project reports.☐

Chester M McKinney: ASA President Elect

Chester McKinney, recently awarded honorary fellowship of The Institute of Acoustics, is the new President Elect of the Acoustical Society of America. The citation giving details of Chester's career in acoustics at the time of the IOA award appears in the July 1986 issue of the Bulletin. We offer him our good wishes for a successful term of office.☐

Institute Prize awarded at the University of Surrey

At a short, informal ceremony on 26 June, Dr A E Brown in his capacity as External Examiner to the Physics with Modern Acoustics Course at the University of Surrey, awarded the Institute of Acoustics Prize for the best student in the Second Year to Nicole Porter. Last year Nicole, seen below, shared the Prize awarded for the best student in the First Year.

The prizes are awarded by the Department of Physics and the Institute to mark the final run of the Physics with Modern Acoustics Degree course which is being stopped as a result of the financial cutbacks suffered by the University of Surrey. Professor A G Crocker, the Head of Department, decided that the course could not be allowed to disappear without some mark of its passing, and so those students in the final cohort showing the best performance (not necessarily the ones who come top in the examinations) in each year are being awarded these special prizes sponsored by the Institute.☐

Special Feature continued from page 12

Mode Analysis of Musical Instruments:

B E RICHARDS

Department of Physics, University College Cardiff

J M BOWSHER

Department of Physics, University of Surrey

The Application of Holographic Interferometry and Finite Element Analysis

The tone quality, and in some cases the pitch, of musical instruments is intimately related to the modes of vibration of the body of the instrument. Mode analysis of the vibrating structure can lead to a quantitative understanding of the relationship between the instrument's construction and its sound quality. We describe here the application of holographic interferometry and finite element analysis to the study of stringed instruments, bells and trombones; these instruments are of research interest at University College, Cardiff and the University of Surrey.

HOLOGRAPHIC INTERFEROMETRY and finite element analysis find many applications for vibration and stress analysis in the fields of mechanical and civil engineering, but they are equally suited to the analysis of the complex structures found in musical instruments. An example of their use for the study of a freely-supported violin top plate is shown in Figure 1. Time-averaged holographic interferometry is an optical technique. It allows minute transverse vibrations to be visualized in the form of contour lines (interference fringes), overlaying the object's surface and mapping out its displacement. If the object is excited sinusoidally during the recording period of the hologram and care is taken to ensure that only one mode of vibration is excited, the contour pattern represents the eigenfunction of that mode. The method requires no special treatment of the object's surface and has a sensitivity of the order of 0.1 micrometre. By contrast, finite element analysis is a computational technique for modelling structures and obtaining approximations to their normal modes of vibration. These calculations require details only of the instrument's dimensions (shape) and material properties. The object is 'broken down' into smaller units (elements) for which the differential equations of motion can be solved. The computation is complex and in general commercial software packages are employed. The choice of package is important to ensure that it includes the correct elements, classes of materials and eigenvalue problems. We have successfully used ASAS, NASTRAN, PAFEC and LUSAS for our musical instrument studies.

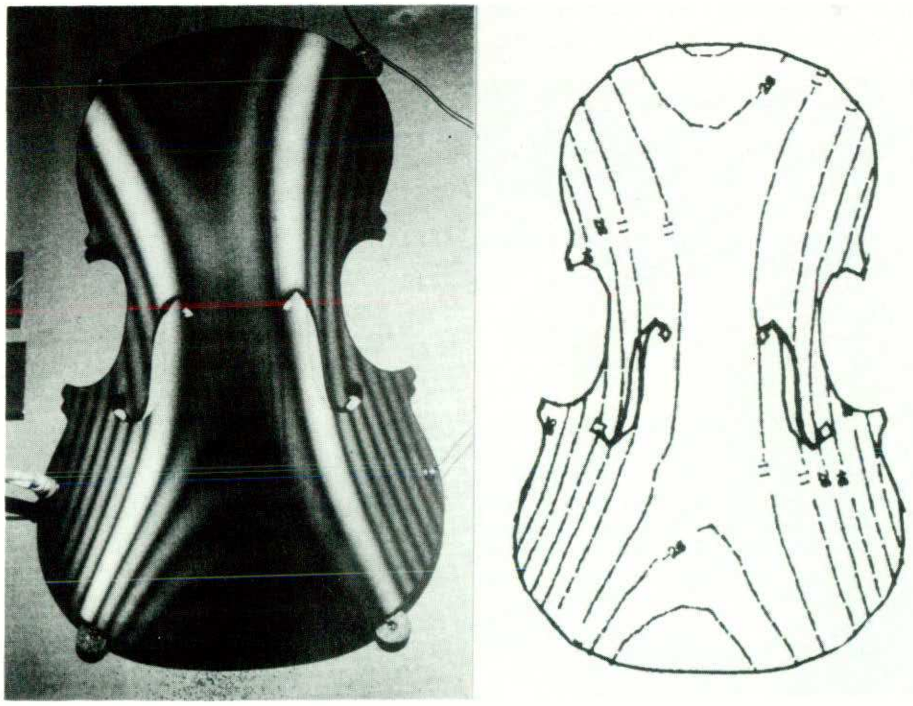

Figure 1 A comparison between experiment and theory. (Left) The second mode of vibration of a violin plate recorded using holographic interferometry at a frequency of 168 Hz. The two bright fringes are nodes. (Right) The same mode calculated by the finite element method at a frequency of 178 Hz. The hatched lines are contours of the displacement amplitude with the same spacing as the dark fringes in the accompanying interferogram.

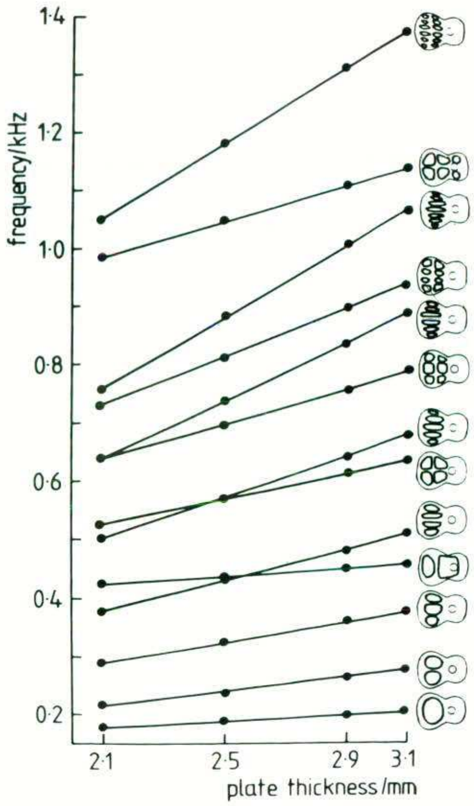

Figure 2 Finite element calculations of mode frequencies of a strutted guitar top plate with varying plate thickness. (From Richardson and Roberts 1985)

Stringed musical instruments form a most challenging problem for investigation. Coupling between the various wooden parts and to the air inside the instrument, and the orthotropic nature of the materials, make quantitative analysis extremely difficult. The function of the body is to increase the sound radiation from the strings, but in doing so it also acts as a formant filter, preferentially amplifying certain string harmonics. The frequencies, spatial distributions and Q-values of the body modes are thus of prime importance in determining the sound quality of the instrument. At University College, Cardiff we are studying the modes of guitars and violins in relation to the construction of the instruments. Holographic studies, using a four watt argon-ion laser, of real and experimental instruments provide data against which theoretical computations can be compared. We have begun to develop numerical models of instruments which will be 'playable'. The effect of changes in construction can then be directly related to the change in sound quality. As an example of this work, Figure 2 shows how the mode frequencies of a guitar top plate vary as the plate is thinned. Comparable practical studies would be impossible because of the difficulty in obtaining several pieces of wood with identical properties; the numerical model furnishes us with an infinite stock of any material.

A rather more simple application of these methods is the study of bells.

Figure 3 Bell tuning at Taylor's Bell Foundry, Loughborough

Their circular symmetry and isotropic metallic properties make analysis rather easier. The pitch of a bell is governed by the frequencies of several strongly-radiating modes, and therefore mode frequencies must be precisely tuned. Holographic studies have revealed some of the 'mysteries' associated with bell tuning. The traditional shape of bells has evolved to produce mode frequencies in approximately the correct ratios; fine tuning is then achieved by removing material from the bell wall in specific areas.

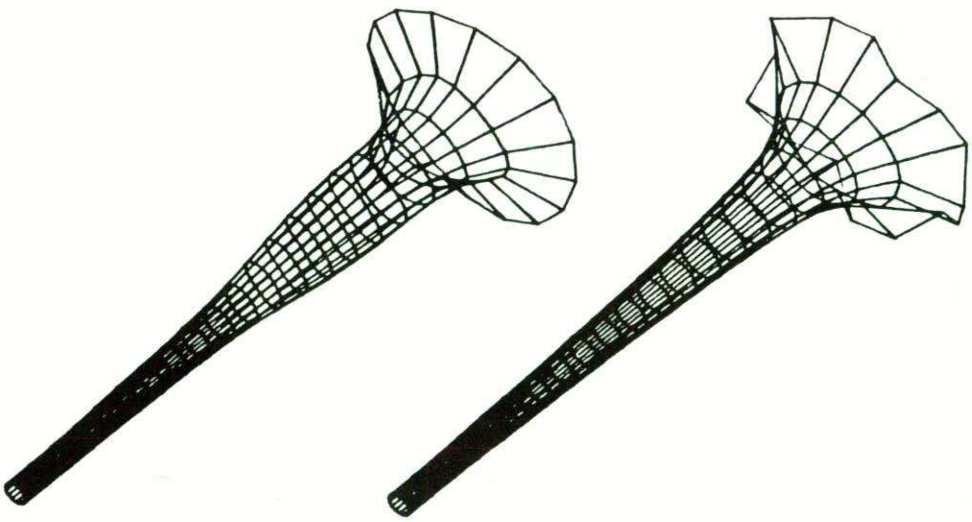

Figure 4 Trombone bell wall vibrations as predicted by a finite element calculation. The displacement amplitudes have been greatly exaggerated for clarity. The two modes shown are (left) at 478.6 Hz, a frequency where the coupling between the air-column and structure is comparatively strong, and (right) the next mode at 609.4 Hz, where the coupling is very weak. (From Watkinson and Bowsher 1982)

A striking example of the use of finite element analysis is the recent work of Andre Lehr of the Netherlands (Lehr 1986). He has used this technique to design a carillon bell with a major third rather than the traditional minor third partial. Through the use of computer modelling he was able to develop suitable bell profiles, wall thicknesses and tuning procedures without having to cast a single bell. This work represents one of the most progressive developments in the manufacture of bells for several centuries.

Finally, we turn our attention to the work at Surrey on a different type of bell: the bells of brass instruments. Sound production in brass instruments

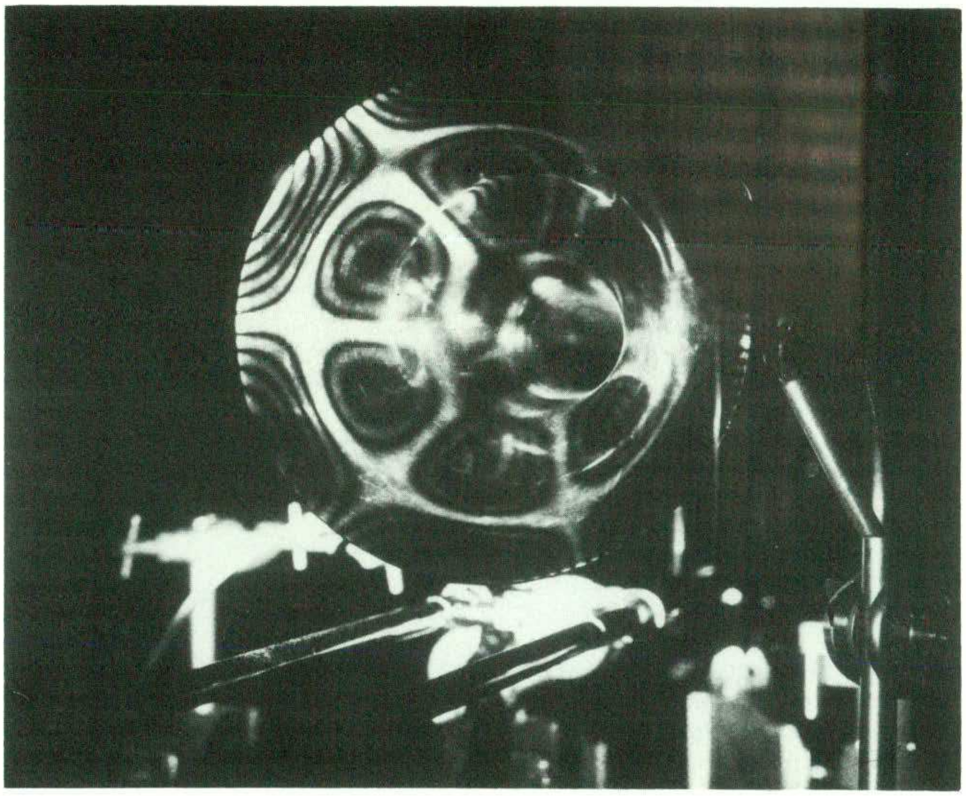

Figure 5 Trombone bell wall vibrations visualized by holographic interferometry (401 Hz)

is primarily the result of air-column vibrations. However, the bells of instruments such as trumpets or trombones have relatively thin and flexible walls, and modes of vibration occur at frequencies comparable with air-column vibrations, ie in the musical range. Depending on the symmetry of the structural vibrations, wall motion sometimes couples to air-column vibrations (Figure 4). Direct sound radiation from the bell motion to the audience is small in comparison with the sound radiated by the air-column, but it is sufficient to influence the player, particularly for frequencies where strong coupling occurs. Materials, bell shapes, rim construction and stay positions have all been studied in relation to sound quality. The indications are that wall thickness is far more important than wall material, at least for the range of brasses usually employed in manufacture.

References

A Lehr. *A carillon with major third bells,* Proceedings of the 12th ICA, Vol III, K6-4, 1986.

B E Richardson and G W Roberts. *The adjustment of mode frequencies in guitars: a study by means of holographic interferometry and finite element analysis,* SMAC '83 Vol II (Pub of the Royal Swedish Academy of Music No 46:2), 285-302, 1985.

P S Watkinson and J M Bowsher. *Vibration characteristics of brass instrument bells,* J Sound and Vib, 85, 1-17, 1982.

Electronic and Computer Music Synthesis

MICHAEL GREENHOUGH

Physics Department, University College, Cardiff

In the last few decades analogue and digital electronics have revolutionized our approach to sound and music generation. Both the power and the problems which this technology brings are perhaps best understood in relation to those of more traditional resources.

THE PHYSICAL controlling input to a musical instrument can be described in a 'space' bounded by dimensions such as breath pressure, finger positions, muscle tensions etc. The corresponding acoustical output space is bounded by pitch, dynamic level and the many dimensions corresponding to timbre. The instrument itself can therefore be characterized by a mapping which relates these two spaces. In the process of mastering an instrument, knowledge of this mapping is acquired largely empirically and subsequent performance is a highly intuitive business.

Technological enhancements of traditional musical instruments have resulted in an increase in the size of the acoustic output space and sometimes a corresponding increase in complexity of the physical input space, but the intuitive mapping has been preserved. The possibility of intuitive control is due largely to two characteristics of musical instruments. Firstly, such systems operate in real time, a change in the controlling input parameters producing an instantaneous change in the audio output. The resulting immediate feedback allows the player to arrive at a desired sound by experimentation, with the minimum of delay. Secondly, the instrument allows the player access to only a very limited region of the total sound space. Most instruments produce harmonic spectra with fundamental frequencies restricted in range to a few octaves and often quantized to the notes of the chromatic scale. Therefore the acoustic signals produced have a very high degree of information redundancy. Since our acoustic environment is rich in harmonic series this redundancy is of a particularly appropriate and intelligible kind. We can say, then, that the player's physical gestures are tightly mapped to a restricted sound space which is peculiarly rich in significant sounds.

With the advent of electronic oscillators, amplifiers and filters in the early part of this century the size of the potential sound space available increased very significantly as, necessarily, did that of the controlling input space. The proportion of the sounds which are of musical interest, though, is markedly smaller than for a traditional instrument. It has been common for users to restrict their freedom and increase this proportion by choosing configurations of synthesizer modules which model the traditional instrument. Thus the OSCILLATOR — ENVELOPE-SHAPER — FILTER style of patch reintroduces a reassuring form of redundancy whilst still offering a range of sounds unobtainable by non-electronic means. Most importantly, since analogue synthesizers operate in real time, in the search for a particular sound we can put together a likely patch and then home in on our target by adjusting knobs and listening. No detailed explicit knowledge of the sound's physical properties is needed.

Any sound can be represented uniquely as a pressure against time function, so when we introduce the computer, complete with digital-to-analogue converter, our output sound space expands to include all possible sounds. This seemingly attractive system has several penalties associated with it. Firstly, a very large amount of input data is essential to specify a particular output sound. Secondly, the overwhelming majority of possible sounds have no musical value or significance, being merely indistinguishable examples of 'off-white' noise. Thirdly, the relationship between the pressure waveform and the sound as perceived is most obscure. As a result early computer music systems, such as Music V[1], have tended to be modelled on analogue synthesizers which, in turn, reflect traditional instruments. Thus by the reintroduction of familiar redundancy the amount of input data required is greatly reduced, and the proportion of useful sounds is increased to a sensible level. Nowadays dedicated digital hardware is often used to replace software 'oscillator' and 'filter' routines, making real-time computer synthesis feasible. Even so, the degree of intuitive control may be limited.

A very notable recent method for coping with the vast potential of sound synthesis is the frequency-modulation technique of Chowning[2], which is used in the Yamaha DX series of digital synthesizers. This maps a very small number of input parameters on to a small but anomalously rich and useful region of the total sound space. The mapping, however, is not very intuitive.

Traditionally, then, we control details of the waveforms and spectra of sounds implicitly, using conventional instruments. Computer technology gives us greater generality through explicit control. Powerful systems with 'friendly interfaces' such as the 'Fairlight' Computer Music Instrument and the 'Synclavier' have been commercially available for a number of years, but at a high cost.

In striking contrast to our approach to lower-level, acoustic features, the longer-term (eg melodic and rhythmic) properties of music have always been specified explicitly by composers. In the Physics Department at University College, Cardiff, we are developing a system which uses the computer to achieve implicit control of these properties.

The system runs on an IBM-PC AT microcomputer which drives a Yamaha DX7 synthesizer via a MIDI interface. The timbral features of the music produced are thus predetermined by button pressing and so the computer is free to concentrate on which notes will be played when and for how long.

The musical structure of the output is controlled by a collection of probability distributions which can be displayed graphically on the VDU and adjusted during real-time performances. In this way the user can exert an indirect but powerful influence on the pitches and durations of notes in up to 16 voices at a time. After a likely starting configuration has been set up the user can home in on a particular effect by ear with no need for explicit musical knowledge.

The probability distributions which govern the system parameters are identical in structure and are simply interpreted in different ways to fulfil different musical purposes. Some represent the *a priori* weighting of pitch values and can be tailored to produce scales of a traditional, or quite arbitrary, form. A flat distribution will maximize the information content to produce chromatic chaos. Biasing the distribution to pitches of, say, a diatonic scale will introduce redundancy of the context-free variety and impose a crude tonal character on the series of notes output. A programmed series of such distributions can be used to give a harmonic progression. Whilst this progression is readily apparent, the output still makes little musical sense.

It is normal therefore to introduce some context-sensitive redundancy. The user chooses a number of features of the context as reference points. These may be the pitch or duration of previous notes in the voice being generated (or previous and presently sounding notes in any other voice). A probability distribution is then used to control the first and higher-order differences between each reference point and the event being determined. We thus control the horizontal, vertical and diagonal pitch and rhythmic relationships between the voices. Output can, for example, be given a natural tendency towards (or, indeed, away from) contrary motion, ostinato and canonical patterns etc.

The probability distributions representing parameter differences offer the user an alternative (and not unintuitive) domain of control. The musical significance of the distributions is clear from a consideration of simple calculus. Where pitch is concerned, 1st and 2nd differences represent, respectively, the gradients and curvatures of the melody, through which we can control the occurrence of extrema. The difference control structures can be extended to 3rd and higher orders and also to lower, negative orders, implying the summation of event values. This gives us control of the 'area under the melody' and hence of how long a voice may linger in different parts of its range.

The application of the same controlling structures to event durations can produce rhythmic behaviour which is stable and intelligible, downright bizarre, or anything in between. The context-free probabilities represent the tendencies of events to start at particular points in time, and so control the distribution of metric emphasis. The 1st-difference distribution therefore represents the durations of events. 2nd differences control the tendency of duration values to persist once they have occurred.

In addition to the fine adjustments which can be performed on a probability distribution a facility is available for controlling its overall shape by means of a single parameter, which could be derived from a potentiometer. Thus a two-dimensional joystick affords the user control of the gross tonal and rhythmic properties of the output. This single parameter also provides a powerful link between the melodic and rhythmic properties of the stream of events. For example, if desired, important pitches can be particularly favoured at points of strong metric emphasis.

The use of a difference method for event generation brings about a dramatic data reduction which facilitates the essential real-time operation but involves, inevitably, a corresponding loss of generality. The results so far encourage one to believe that some of this generality is expendable, that is, that the particular kind of redundancy in the output is consistent with certain general musical tendencies. By varying only the data fed to the system (ie probability values and context reference points) a wide variety of musical effects can be obtained, which are perhaps best described at present as 'plausible and promising'!

The introduction of high technology to sound and music synthesis has brought us limitless possibilities. An important task now is the design of interfacing software which limits our access to the more fruitful regions of timbral and music-structural space. These will not only provide powerful and intuitive tools for composition but also constitute, to some extent, models of how we perceive sound and music.

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1. M V Mathews. *The Technology of Computer Music*, The MIT Press, 1969.

2. J M Chowning. *The Synthesis of Complex Audio Spectra by Means of Frequency Modulation*, JAES 21, 526-534, 1973.

The Acoustical Quality of Violins

COLIN GOUGH

Department of Physics, University of Birmingham

Few problems in acoustics have generated as much interest as the century-old attempt to discover a scientific basis for the 'Stradivari secret' — the properties of sixteenth and seventeenth century Italian violins that essentially define what we mean by excellence in tone of a violin. From an artistic point of view the violin represents one of the supreme achievements of the 16th century Italian renaissance. Undoubtedly the beautiful carving, the translucence of the varnish and the sheer beauty of its outward form contribute to a violinist's appreciation of any old instrument. Indeed many people still believe it is the varnish that holds the secret; others believe it is something to do with the original treatment of the wood and its subsequent ageing; while others, including myself, believe it has more to do with the skills with which the great Cremonese masters selected and graded the thicknesses of the woods used throughout the course of the violin's construction.

WHATEVER the explanation, the acoustical properties must be completely determined by the dynamic and radiative properties of the violin. Such information must therefore define the quality of a violin, provided we know how to interpret it.

In crude acoustical terms the violin is simply a wooden shell that acts as a loudspeaker in its own bass-reflex cabinet. The shell is excited by bowing the string producing a force with a saw-tooth waveform on the supporting bridge, which is transmitted to the shell structure. The resulting sound is therefore very rich in harmonics.

Traditionally, much research on the physics of the violin has concentrated on what are known as the main body and Helmholtz resonances. The main body resonance is usually centered around 4-500 Hz and largely involves the front-plate vibrating like the cone of a loudspeaker. The hidden asymmetry of the violin (including the use of a soundpost and bass-bar) allows this mode to be an efficient source of monopole radiation. The Helmholtz resonator is formed by the enclosed air vibrating through the f-holes cut into the front plate. It is centered around 280 Hz and helps to boost the sound output below the main body resonance.

Whereas an ideal loudspeaker is designed to have as flat a response as possible, the output of a violin is full of deep peaks and troughs associated with the large number of structural resonances of the shell that can be excited. This is equivalent to the problem of a loudspeaker-cone breaking up into unwanted resonances. However, it is just this resultant 'colouration' of sound output that gives each violin its distinctive tone quality.

In Figure 1 we show some measurements which not only illustrate the complexity of vibrational response for four violins of widely different quality but which also illustrate the way that research in musical acoustics is important in the development of new ideas and techniques in acoustical research. These measurements were made using a novel technique of measurement and analysis introduced by Gabi Weinreich[1], in which the principle of reciprocity is invoked to obtain absolute measurements of the 'radiativity' of an acoustical object. In this method the violin is placed in a well-defined sound field and the velocity response at the point of string support on the bridge is recorded using a light-weight gramophone pick-up. The method not only provides meaningful absolute measurements but also involves only those vibrational modes that are acoustically significant.

The other innovative feature of such measurements lies in the method of analysis, in which the derivatives with respect to frequency of both the in-phase and quadrature response are evaluated before being squared and added to give the output. This has the effect of reducing the overlap between resonances, so that it is much easier to identify their positions and widths in a multi-mode structure. The 'Weinreich plot' is particularly easy to implement in any digital processing system and could be of value in any form of spectral analysis, including FFTs, for systems with closely spaced resonances.

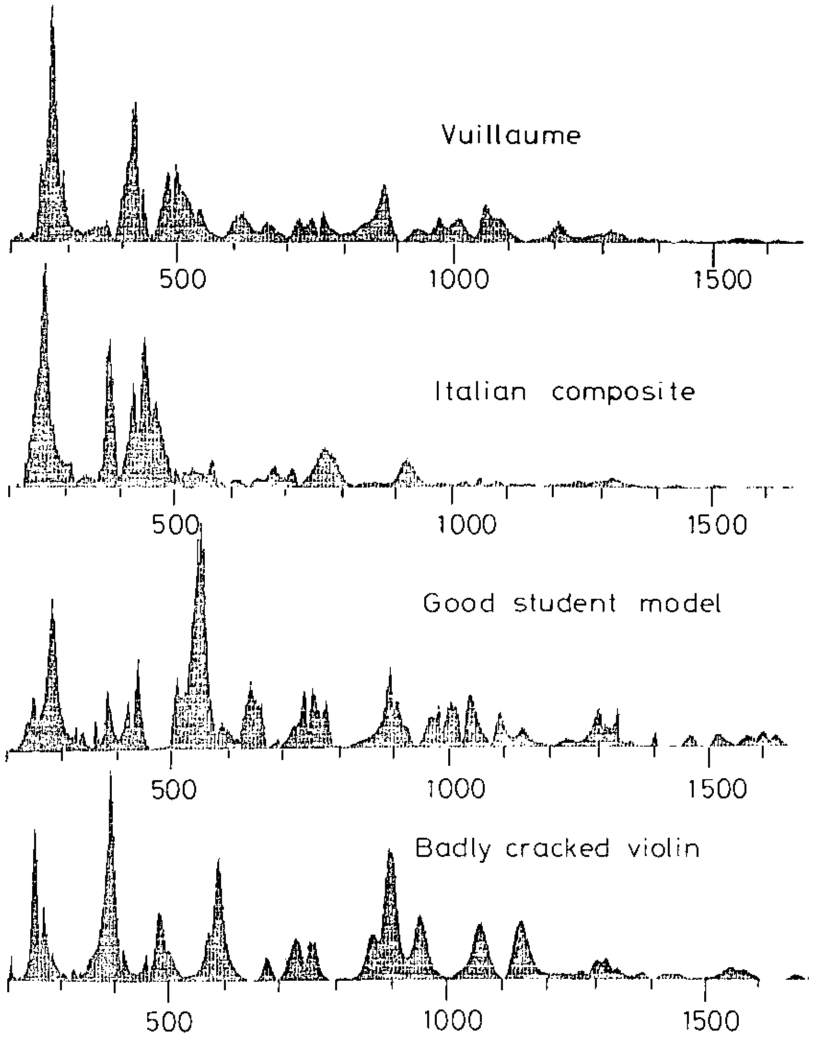


Figure 1 Acoustic resonance spectra

All four instruments exhibit a pronounced Helmholtz resonance around 280 Hz with a large number of structural resonances at higher frequencies. Such a plot provides a unique acoustic fingerprint for a particular instrument, which could be used for identification or verification purposes. Holographic measurements and, more recently, measurements using modal analysis have forcibly demonstrated that every mode involves the vibration of the instrument as a whole rather than an isolated component like the front plate.

A word of caution about the interpretation of such measurements must be introduced. In such a plot an ideal loudspeaker would exhibit a single cone-resonance whereas the sound output remains constant until the acoustic wavelength became comparable with the cone size. It must always be remembered that resonant modes will contribute significantly to the sound output well above their resonant frequency.

Figure 2 illustrates the waveforms and associated Fourier spectra for notes of a whole tone scale bowed on the lowest string of the Vuillaume violin included in Figure 1. We note that the acoustic energy is distributed over a number of partials with waveforms and amplitudes of harmonics varying appreciably from note to note. In particular, the fundamental components of the lowest notes are extremely weak, as anticipated from the absence of any appreciable acoustic resonances at such low frequencies. These measurements also demonstrate the well-known psychoacoustic phenomenon of the 'missing fundamental', where the apparent pitch of a note is determined by the periodicity of a waveform, whether or not the waveform contains any appreciable Fourier components at the fundamental frequency.

Despite the large differences in spectral content, the tone of this violin, as judged by player or listener, appears not to change significantly from note to note. In passing, it is important to emphasize that the sound heard by the distant listener may be very different from that heard by the player, who also experiences the 'near-field' sound. The same scale played on another instrument would produce waveforms with spectral contents that would again vary from note to note but would differ appreciably from that of the first violin. Thus we have an apparent paradox — large changes in the spectral content of the notes sounded on a single instrument appear not to significantly affect the tone quality, whereas no two violins ever sound the same for reasons that we must attribute to the spectral content.

Such observations certainly imply that the quality of a violin's tone is unlikely to be identified with the position or quality factor of a particular structural resonance, such as the main body resonance, as has often been suggested. At best this can only affect the amplitude of a single partial, which we have already seen can vary significantly from note to note leaving the apparent tone unchanged.

The above measurements have concentrated on relatively low frequencies where the ear is relatively insensitive. It is well known from studies of speech that the spectral content of sounds in the 2-5 kHz range is important in characterizing sound quality and the same must also be true for the violin. At such frequencies the damping is such that the structural resonances blend into a continuum, so that it is only their density of states that becomes important. Furthermore, at such frequencies, the bridge acts as a tuned acoustic transformer that can enhance or degrade energy transfer over chosen frequency ranges. Changing a bridge or adjusting the transfer characteristics by selective carving can often dramatically affect the sound of a violin. Somewhat surprisingly the importance of the bridge on violin tone is often completely overlooked.

From what we have indicated above, it is clear that a great number of factors affect the tone of a particular violin but as yet there is still no very clear appreciation of which of these are most important. Any comparison of the physical properties of a violin with monetary value is a non-starter, since even the greatest makers have produced violins of inferior tone quality, which nevertheless retain their value as collector's items. One of the main problems is that we currently have no established language to define in acoustical terms what violinists mean when they refer to a violin having a bright, hollow, wooden, brilliant, nasal, shrill, warm, full, Italian or French sound.

What we need is a properly funded study involving physicists, psychoacousticians and distinguished professional violinists to work towards defining such a language, possibly in terms of a number of 'standard violins' or recorded sounds. Only by doing so is it likely that any real progress will be made in correlating quality and specific physical attributes of instruments.

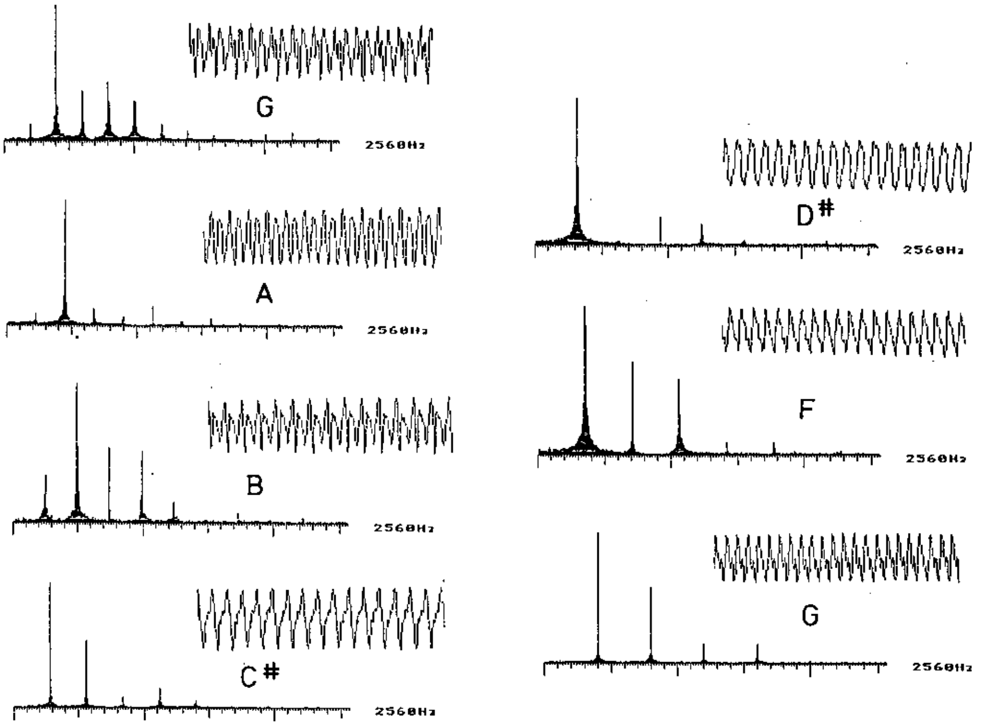


Figure 2 Wave forms and FFT spectra for bowed notes on Vuillaume g-string.

The potential rewards are high, particularly if some way could subsequently be found to help violin makers produce instruments that more consistently approach the quality of the great Italian masters. Thereby we might not only be helping many professional players, who cannot afford the vast sums now required to buy old Italian violins, but would also be helping millions of people world-wide who, at some stage in their life, learn to play the violin on instruments that currently give little encouragement to student, teacher or long-suffering parent alike.

I am grateful to Paul Withey and Michael Hume for the measurements of radiativity obtained in an undergraduate project.

Reference

1. E B Arnold and G Weinreich. *Acoustical Spectroscopy of Violins,* J Acoust Soc Amer **72,** 1739, 1982.

Vibration Control in Optics and Metrology

Sira are organizing an international conference on **Vibration Control in Optics and Metrology** in London on February 25 and 26 1987, with the aim of highlighting problems of vibration isolation in nanometre technology. The Conference will be preceded by two tutorial days on optomechanical design. Four technical sessions will cover: vibrational environments; specification, design and measurement of antivibration systems; and areas of application and case histories. An exhibition will accompany the Conference. For further details please apply to The Conference Office, Sira Ltd, South Hill, Chislehurst, Kent BR7 5EH.□

Royal Medal for Professor Ash

Her Majesty The Queen has approved the recommendation of the Council of the Royal Society for the award of a Royal Medal for 1986 to Professor E A Ash, in recognition of his outstanding researches on acoustic microscopy. Professor Ash, Rector of the Imperial College of Science and Technology, London, was among the first to recognize the possibilities of surface acoustic waves as the basis for delay lines in electronic circuits and more recently, with his group at University College London, has worked on other applications of surface acoustic waves and on acoustic and optical microscopy. Details of the award appear in the July 1986 issue of Royal Society News.□

Letter from the Vice-President Groups and Branches

I mentioned in my last letter the success our colleagues in Hong Kong had enjoyed with the Westpac II Conference. We have now received the branch annual report which showed that all had not been plain sailing over the preceding year. Three of the original branch committee who had successfully run a popular series of evening meetings on highly topical subjects had left Hong Kong and the new committee had been beset by illness and pressure of work. Having let the evening meeting schedule lapse the committee looked for an alternative way of encouraging the aims of the IOA.

With some help from the Australian Acoustical Society and the Acoustical Society of Japan they set about organizing Westpac (Western Pacific Region Acoustics Conference) with the resultant success already reported. In fact they produced a net profit of £7000 which they intend to spend on furthering the aims and objectives of the Institute in Hong Kong.

Now we have one or two branches in this country whose committees appear beset by all kinds of woes! How about learning from the colonies and hosting a major event?

At a recent Council meeting it was decided not to continue the Groups and Branches Representatives meeting on an annual basis but to call it together when there was a major item to discuss. I shall continue to report items of interest on Groups and Branches, as indeed I hope individual Groups and Branches will, on these pages.

Any problems that committees have can of course be brought to Council's or its standing committees' attention in the usual way by contacting either myself or Cathy.□

Geoff Kerry

In case you didn't hear...

The **Association of Consulting Engineers** has produced a short video which explains the functions of Consulting Engineers and demonstrates their involvement in a variety of contrasting projects around the world. An information sheet giving details of the video, which is standard VHS, are available from Mrs Cathy Mackenzie at IOA HQ in Edinburgh.

A consortium led by Leicester Polytechnic in partnership with Imperial College and Loughborough University are preparing a **Pollution Control Distance Learning** package, which will begin to be available from late 1986. The units cover the following broad subject areas: Air pollution control; Hazardous waste control; Noise and vibration; Water pollution control; and Environmental management. Those interested in being kept informed of the progress of the scheme should contact Peter Shepherd, Project Manager, Centre for Educational Technology and Development, Leicester Polytechnic, PO Box 143, Leicester LE1 9BH.

The **Proceedings of INTER-NOISE 86** are now available as a two-volume set containing 271 papers covering all areas of noise control engineering. The cost is $75.00 (surface postage included), and payment must be made in US funds on a US bank or on a bank having a correspondent bank in the US. The Proceedings may be ordered from Noise Control Foundation, PO Box 2469 Arlington Branch, Poughkeepsie, NY 12603, USA.□

Meeting Announcement and Call for Papers

Acoustic Microscopy and its Applications

The Physical Acoustics Group will hold a one-day meeting on Acoustic Microscopy and its Applications on 24 February 1987. The meeting will follow the same pattern as developed at the Group's previous highly successful meeting on this topic of providing a forum for the presentation of new developments in the technique, of applications and of short reports of new work in progress. Abstracts of offered contributions and requests for further information about the meeting should be sent to:

Dr D P Almond, School of Materials Science, University of Bath, Claverton Down, Bath BA2 7AY, Avon.□

Courses

Commins-bbm, acoustics noise and vibration consultants, are organizing a course on November 3-5 1986 on **Advanced Problems of Noise and Vibration Control.** The course, to be held in Paris, will be given by Dr Eric Ungar and Dr Istvan Ver, and will be in English.

Complex problems will be discussed such as: enclosures, structural dynamics, lined and unlined ducts, mufflers in acoustics and aerodynamics, viscoelastic materials, complex damping treatments, large fans, two-stage isolators, reciprocity, broad-band damping, aircraft and jet engine facilities, vibration design for high technology facilities. Anyone interested in participating should contact: Commins-bbm, BP 81, 91371 Verrières le Buisson, France, Telephone: 33-1-60. 13.32.50.□

Noise North of the Border

Protests are being voiced in Scotland at the news that Secretary of State Malcolm Rifkind is considering amendments to Scottish regulations for buildings standards, bringing them into line with those applying in England. It is believed that the effect of the changes will be to lower standards, and reduce Council controls on sound insulation. The Building Standards Advisory Council has sent the proposed regulations back for further consideration; meanwhile opposition to the changes continues to grow.□

NON-INSTITUTE MEETINGS

1986

22 October. *SEECO 86. Environmental Stress Screening.* Coventry. Contact: SEE, Owles Hall, Buntingford, Herts SG9 9PL.

28-29 October. *Secure Communication Systems.* London. Contact: Conference Services, IEE, Savoy Place, London WC2R 0BL. (Co-sponsored by the IOA).

26-28 November. *International Symposium on Local Communication Systems.* Toulouse. Contact: Prof J P Cabanel, Secretariat du Symposium IFIP — Laboratoire LSI, Université Paul Sabatier, 118 Route de Narbonne, 31062 Toulouse Cedex, France.

8-12 December. *1st Asian Pacific Region Conference on Deafness.* Hong Kong. Contact: Hong Kong Society of the Deaf, 901 Duke of Windsor Social Serv Bldg, 15 Hennessy Road, Hong Kong.

14-16 December. *Unhealthy Housing — a diagnosis.* Warwick. Contact: Rosemary McMahon, Health and Housing Conference Administrator, The Institution of Environmental Health Officers, Chadwick House, Rushworth Street, London SE1 0QT.

15-18 December. *Electronics for Ocean Technology.* Edinburgh. Contact: The Conference Secretariat, IERE, 99 Gower Street, London WC1 6AZ.

1987

25-26 February. *First International Conference on Vibration Control in Optics and Metrology.* London. Contact: Conference Office, Sira Ltd, South Hill, Chislehurst, Kent BR7 5EH.

24-26 March. *DAGE '87.* Aachen. Contact: Prof H Kuttruff, Institut für Technische Akustik, Templergraben 55, Aachen.

1-3 April. *Condition Monitoring.* Swansea. Contact: Mervyn H Jones, University College of Swansea, The Abbey, Singleton Park, Swansea SA2 8PP.

6-9 July. *Ultrasonics International 87.* London. Contact: Marija Vukovojac, Conference Organizer, Ultrasonics International 87, Butterworth Scientific Ltd, PO Box 63, Westbury House, Bury Street, Guildford, Surrey GU2 5BH.

8-10 July. *NOISE-CON 87.* Pennsylvania. Contact: Conference Secretariat, NOISE-CON 87, the Graduate Program in Acoustics, Applied Science Building, University Park, PA 16802, USA. (300-word Abstracts by 7 November 1986).

13-18 September. *4th European Conference on Non-Destructive Testing.* London. Contact: Conference Associates NDT, 27A Medway Street, Westminster, London SW1P 2BD.

15-17 September. *INTER-NOISE 87.* Beijing, China. Contact. INTER-NOISE 87 Secretariat, 5 Zongguancun Street, PO Box 2712, Beijing, China.

Information relating to meetings of possible interest to readers should be with the Editor at the address on page 1 no later than four months before the date of the meeting.□

Aircraft Noise Index Study

The Government has responsibility for Airport Policy and in this capacity is concerned that aircraft noise disturbance should be monitored and assessed as accurately as possible. A full study was therefore commissioned from the Directorate of Research of the Civil Aviation Authority which, together with an appropriate consultation process, will be taken into account in the development of future policy. The main objective of the research was either to substantiate the Noise and Number Index (NNI) or if necessary to devise some better index of aircraft noise exposure.

To ensure that the results of the study were nationally representative, twenty-three areas around five major civil airports — Heathrow, Gatwick, Luton, Manchester and Aberdeen — were selected. Fieldwork consisted of aircraft noise measurements and a programme of social surveys. About eighty randomly-chosen residents were interviewed in each of the chosen areas; in total over two thousand people were interviewed. The main conclusion of the study report is that there appears to be a case for replacing the NNI system of measurement by one based on the 24 hour equivalent continuous sound level (Leq).

Copies of the Report — *DR Report 8402: United Kingdom Aircraft Noise Index Study: Main Report* — are available by post, price £12.50, from Civil Aviation Authority, Greville House, 37 Gratton Road, Cheltenham, Glos CL50 2BN, Tel 0242 35151.□

ISVR Short Courses

The following short courses are to be held at ISVR during late 1986 and 1987.

1986

November 3-7	Applied Digital Signal Processing

1987

March 24-26	Laser Technology
March 30-April 3	Clinical Audiology
March 30-April 3	Noise & Vibration Control for Environmental Health Officers
April 6-10	Instrumentation and Measurement Techniques for Noise Control (D1)
April 7-9	Engine Noise and Vibration
May 18-22	Noise Control for Engineers in Processing Industries
May 27-29	Sound Intensity Measurement
June (date to be finalized)	Electronics for Mechanical Engineers
September 7-11	Advanced Noise and Vibration
September 14-18	Industrial Audiology
September 21-25	Technical Audiology
September 21-25	Instrumentation and Measurement Techniques for Vibration Control (D2)

For further details of any of these courses, contact the ISVR Short Course Organizer, The University, Southampton, SO9 5NH, Telephone: 0703 559122 Ext 2310 or 752.□

New Products

Submissions for inclusion in this section should be sent direct to J W Sargent, Building Research Establishment, Garston, Watford WD2 7JR.

Field FFT Analyzer Ono Sokki CF210

The CF210 Field FFT Analyzer is a development of the CF200 Analyzer. This latest battery powered model includes 144 k bytes of data storage and a GP-IB interface for control from an external computer. It is designed to be used with direct plug-in accelerometers and microphones to completely eliminate the need for signal preconditioning. The Dot-Matrix Liquid Crystal Display provides input signal time-axis waveforms and display of analysed data. It is also capable of three-dimensional display and provides direct digital reading of any point on the analysed data display.

The mass memory is extremely flexible and can store time or frequency data. Complete with built-in printer and battery pack the unit weighs only 8 kg.

High Speed Transient Event Recorder: Krenz TRC 4080

The TRC 4080 comprises a master controller, trigger setting, sampling clock and up to 4 parallel ADCs in the same rack mounting unit. Analogue signals can be sampled at a sampling rate of up to 50 kHz sampling rates and two time bases are available in order to make the best use of up to a maximum of 128 k samples of memory per channel.

Micro-computer control is available to suit all the standard interface protocols.

The system can be extended up to a maximum of 48 parallel channels of input each with an 8 bit resolution all controlled by one master unit. The TRC 4080 has the ability to output data to a variety of types of display both in analogue and digital modes. These include outputs to a two channel oscilloscope or a multichannel Y-t strip chart recorder. Digital data can be transferred for either storage or further downline processing. An intelligent interface is also available for HP compatible plotters.

Further details on Ono Sokki and Krenz instruments are available from Lucas CEL Instruments Ltd, 35 Bury Mead Road, Hitchin, SG5 1RT. Telephone: 0462 52731.

Industrial Noise Exposure Meter — Pulsar 85

The Model 85 series of Industrial Sound Level Meters show, directly on scale, both the noise level and its allowable exposure time. The meter scale of the 85 shows a measuring range of 80 to 120 dBA and gives the corresponding 'Code of Practice' exposure times from 8 hours (90 dBA) down to 7.5 minutes (108 dBA). The time zones are shown in green, orange and red to indicate safe, intermediate and dangerous areas. A '30 dB button' is provided on the front of the unit permitting measurements down to 50 dBA.

An electret condenser microphone is mounted on a telescopic boom to minimize case reflections. The 85E meets the requirements of BS 5969 Grade 3 while the Model 85E2 meets BS 5969 Grade 2.

Further details from Pulsar Instruments, 40-42 Westborough, Scarborough, North Yorks YO11 1UN. Telephone: 0723 371351.

New Sine and Sine/Noise Generators from B & K

Brüel & Kjær have produced two new instruments for use with Automatic Testing and Quality Control systems. They are the Sine Generator Type 1051 and the Sine/Noise Generator Type 1049. Both have linear and logarithmic sweep ranges with user set frequency limits from 0.2 Hz to 200 kHz and extended linear frequency sweeps from 0.001 Hz to 200 Hz with a resolution of 1 mHz. Single and repetitive sweep modes permit synchronous operation with graphic recording and oscilloscope monitoring equipment.

Output levels from 100 microvolts to 5 volts RMS are selectable with an attenuator accuracy of 0.03 dB. The amplitude is flat within 0.05 dB over the AF range with distortion better than −96 dB. Both generators have IEC/IEEE interface enabling them to be incorporated into automatic test systems.

Sets of control panel settings can be stored in memory for recall whenever required and in addition a 1024 point user-defined amplitude weighted frequency sweep can be stored for use when any form of 'constant output' test measurements need relating to frequency.

The Type 1049 Sine/Noise Generator has features indentical in every way to the 1051 except that it also provides narrow band random, white and pink noise outputs for electroacoustic measurements plus a logarithmic amplitude sweep and a compressor providing 126 dB amplitude regulation.

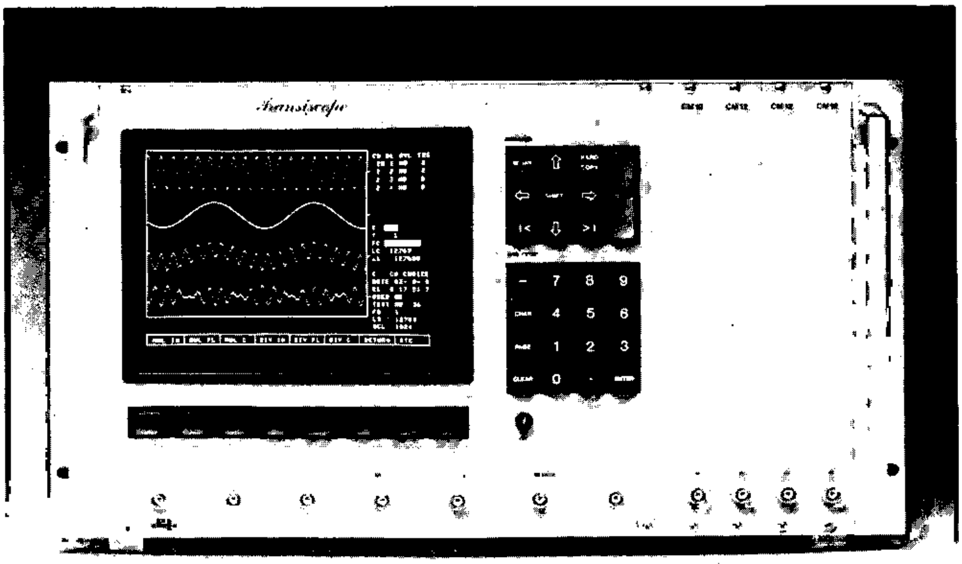
Further details from Brüel & Kjær (UK) Ltd, 92 Uxbridge Road, Harrow, Middlesex HA3 6BZ. Telephone: 01-954 2366.

Transient Recorder TS9004

The TS9004 is a four-channel instrument and is one of the Transiscope range of computing transient recorders now available in the UK from LMS-DIFA Ltd. The features of these recorders are:

12 Bit Precision Analogue to Digital Converter
64 K Words of Memory per channel
Programmable anti-alias Filters
Programmable sampling speed
Up to 10 timebases per recording, maximizing resolution where it is needed
Plot interface, for analog output on X-Y recorder
IEEE 488 and RS232 interface, for data I/O and complete remote control
DMA interface for high-speed 16-bit parallel data I/O
Centronics interface, cable and software for direct connection to Epsom FX 80 type printer

Scalasoft: Scalar software operations — Max, Min, RMS, Energy, Area, Risetime etc
Vectorsoft: Vector software operations — Add, Subtract, Multiply, Integrate etc
Spectrasoft: Fast Fourier Transform package — Amplitude Spectrum, Power Spectrum, Transfer functions etc



The TS9004 costs £10,000.

Other transiscopes are available up to 16 channels.

LMS-DIFA also offer the LMS-CADA Computer Aided Dynamic Testing and Analysis software library, the LMS-F-Series Fourier Systems F700/900 and the LMS-Consultancy services.

Further details from LMS-DIFA Ltd, Prospect House, 2 Prince Georges Road, London SW19 2PT. Telephone: 01-648 2595.

Mass Store for the AD-3522 FFT Analyser

A new enhanced version of the battery powered FFT Analyser AD-3522 is now available from Hakuto. Option 2 gives 40 non-volatile memory stores, 1/3 octave, 3 dimension waterfall and zoom functions. Option 2 fits directly into existing AD-3522 FFT analysers.

Further details from Hakuto International (UK) Ltd, 33-35 Eleanor Cross Road, Waltham Cross, Herts EN8 7LF. Telephone: 0992 769090. □

Institute of Acoustics Meetings

1986			
22 October	SB	Start up in Dynamics (A Middleton)	Southampton
23 October	NWB	Visit to Thwaites Brewery	Blackburn
30 October	ING/EMB	Noise and Vibration in the Aircraft and Spacecraft Industry	Birmingham
19 November	SB	Playing Musical Instruments: Acoustics and Performance (J Bowsher)	Southampton
November	NWB	Evening Lecture by P D Thorne	
6-9 November	M	Reproduced Sound. In collaboration with AES, ASCE, EMAS, APRS, and ABTT	Windermere
11 November	PAG	Ultrasonic Scattering from Bones	London
28-30 November	M	Autumn Conference: Speech and Hearing. In collaboration with the British Society of Audiology	Windermere
4 December	M	Sound Insulation of Buildings and Building Elements	BRE, Watford
4 December	SB	Current Trends in Loudspeaker Design (R Small)	Southampton
9-10 December	UAG	Fluctuation Phenomena in Underwater Acoustics	Weymouth
1987			
21 January	SB	**AGM** and Sound Insulation of Conversions (D Alphey)	Portsmouth
18 February	SB	Environmental Impact Assessment (G M Jackson and T F Murphy)	Southampton
20 February	M	Noise in Mechanical Services	London
24 February	PAG	Acoustic Microscopes, the physics and their application	
March	M	Subjective Perception of Sound Quality by those with Sensitive Hearing. Joint with BSA	
March	M	Open Cast and Quarry Noise	
4 March	SB	Investigation and Treatment of Hearing Disorders in Adults (A M Martin)	ISVR, Southampton
13-16 April	**M**	**Acoustics 87. IOA AGM**	**Portsmouth Polytechnic**
14-15 April	UAG	Sonar Transducers — Past, Present and Future	Birmingham
15 April	SB	Visit to Test Facilities of IAC (provisional)	Weybridge
May	M	Developments in Instrumentation and Computation in Acoustics	London
13 May	SB	Subjective Effects of Mixed Noises (C G Rice)	Southampton
1988			
29 August — 1 September	**M**	**7th FASE Symposium: Speech**	**Edinburgh**

Key:

M = Meetings Committee Programme
BAG = Building Acoustics Group
ING = Industrial Noise Group
MAG = Musical Acoustics Group
PAG = Physical Acoustics Group
SG = Speech Group
UAG = Underwater Acoustics Group
LEM = London Evening Meeting

EMB = East Midlands Branch
NEB = North East Branch
NWB = North West Branch
SB = Southern Branch
ScB = Scottish Branch
SWB = South West Branch
YHB = Yorkshire and Humberside Branch

Further details from:
Institute of Acoustics
25 Chambers Street
Edinburgh EH1 1HU
Tel: 031-225 2143

Christchurch Printing Company Ltd., 19 Woodside Road, Bournemouth BH5 2BA