LOUDSPEAKER TESTING USING DIGITAL TECHNIQUES

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A survey of research work into the evaluation of loudspeakers using digital methods to correlate objective and subjective assessments, carried out by KEF Electronics Limited in conjunction with the Department of Applied Acoustics of the Post Graduate School at the University of Bradford.

Published March 1975

KEF Electronics Limited, Tovil, Maidstone, Kent, England.

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Foreword by Raymond E. Cooke B.Sc.(Eng.), F.R.S.A., F.B.K.S.T.S.
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The quest for a perfect loudspeaker has been going on for more than half a century. So far none has been found, though not for lack of effort and enthusiasm, but rather because as speakers become more refined it becomes relatively more difficult to judge progress. The research work described in this booklet was undertaken as this Company's contribution to the knowledge of the subject. We hope that it will illuminate some hitherto obscure aspects of speaker behaviour and consequently quicken the rate of development.

"He who knoweth not mathematics cannot know any other science and what is more cannot discover his own ignorance or find its proper remedies."

Roger Bacon (1214 - 1292)

Historical Background

The need for adequate objective methods of evaluating loudspeaker performance scarcely requires justification. For more than half a century development has proceeded by a mixture of science, art and intuition, but no-one would regard the situation as satisfactory. As loudspeakers improve it becomes relatively more difficult to distinguish one from another. Scientific methods are needed to clearly expose the effects of small changes in construction. Accurate measurements are also a necessary part of any quality control system and the substantial elimination of the human element is a prerequisite for any test method which is to be applied to very large numbers of manufactured products.

Earlier efforts at an entirely scientific approach were constrained by available instrumentation, and measurements were therefore limited mainly to amplitude-frequency response characteristics. Despite the almost total dependence on frequency response curves as a means of visually portraying loudspeaker performance, particularly in commercial literature, it has been well known for many years that such measurements did not correlate well with subjective assessments of reproduced sound quality.

Speech and music are transient in nature, notwithstanding the fact that some sounds exhibit repetitive waveform patterns which audibly resemble steady state signals. It is therefore more logical to use transient test signals when evaluating transducers which are intended for the reproduction of musical sounds. This much was clearly understood by leading research workers in the field ever since the introduction of the moving coil loudspeaker in 1925.[1] Early investigations were carried out by McLachlan,[2] using a unit step signal, by Helmbold[3] using interrupted tone and Shorter[4] who also used interrupted tone to investigate the later stages of the transient. Shorter succeeded in refining the method to a stage where decay spectra could be presented in three dimensional displays representing amplitude, frequency and time. Still later, Corrington,[5] working in the United States, showed the connection between various modes of cone resonance, tone-burst decay patterns and corresponding fluctuations in the amplitude-frequency response characteristic. Work on transient behaviour has continued sporadically over the years and investigations by Hentsch,[6] Schaumberger[7] and others have contributed to the knowledge of transient phenomena. Problems of instrumentation nevertheless dominated the situation until quite recently. Earlier methods suffered from poor signal-to-noise ratio and were tedious and time consuming. The results were also difficult to interpret and none of the techniques described came into general use, largely because no evident correlation emerged between what was observed and what was heard.

Recent Advances

If a loudspeaker is assumed to be a linear device, the steady state frequency response, in terms of both amplitude and phase, as well as the transient response can all be derived from its response to a short duration impulse. In fact an accurate record of a loudspeaker's impulse response will enable its response to any other signal, whether transient or steady state, to be predicted. The foregoing is a most important fundamental proposition on which the whole of what follows entirely depends.

A knowledge of a loudspeaker's amplitude response alone would be sufficient to enable its general performance to be predicted, if it were a minimum phase-shift device. In general however, loudspeakers are not so well behaved and therefore phase information is also necessary. Investigations into phase distortion in loudspeakers were carried out by Wiener[8] as early as 1940, but he was unable to eliminate the effects of the linear phase-shift arising from the distance between the measuring microphone and the loudspeaker under test. Subsequently Ewaskio and Mawardi[9] measured group delay and succeeded in eliminating linear phase-shift. Later still Stroh[10] used a delay line for the same purpose.

The problems of phase-shift measurement were subsequently solved in elegant fashion by Heyser[11] using analogue methods. He has reported his work in an important series of articles which are compulsory reading for all students of loudspeaker evaluation.

Research by KEF Electronics Limited

A linear system may either be completely described in the frequency domain by its frequency response, both amplitude and phase, or in the time domain by its impulse response. This latter approach was selected by KEF Electronics in arranging a programme of research into transient behaviour which commenced in 1971 in consultation with R.V. Leedham of the University of Bradford. The methods evolved overcame all previous disadvantages of poor signal-to-noise ratio, slowness and tediousness due to the use of modern instrumentation techniques. Signal averaging is used to obtain a wide dynamic range and subsequent processing of the impulse response using a digital computer is employed to provide other displays from the initial data.

An interim progress report was made in a paper presented to the Audio Engineering Society in 1973 by L.R. Fincham and R.V. Leedham.[12] Since then there have been further refinements in measuring technique, signal processing and the presentation of data. The method is now in full use for development work and will shortly be extended to production testing and quality control. An up-to-date report was given in papers by J.M. Berman[13] and L.R. Fincham[14] to the A.E.S. 50th Convention in 1975.

Description of the Method

In essence, the method consists of applying to the loudspeaker an electrical impulse of short duration and recording the corresponding acoustical output as detected by a microphone located at a suitable distance away. The impulse is repeated in sequence with a regular interval which is sufficiently long to ensure a return to quiescence after each excitation. The measuring environment need not be anechoic, but should be large enough for the response from the transducer to die away before reflections arrive from nearby surfaces. Most industrial anechoic chambers are not entirely non-reflecting. Residual reflections and structural resonances which would pass unnoticed on the usual analogue measurements can seriously jeopardise the validity of impulse responses. It has been found preferable to carry out these tests either in the open air on top of a high tower on in very large rooms.

The resulting impulse responses are digitised, stored and averaged by a computer. The response is successively reinforced relative to ambient and random noise up to a limit imposed by the analogue to digital converter. By this means an effective signal-to-noise ratio of 60dB is readily achievable and even higher ratios are possible using appropriate converter equipment.

Presentation of Data

The stored impulse may be viewed directly, and visual inspection has been found useful in making rapid assessments without the need for further computation in every case. The corresponding amplitude and phase response may be computed using the Fourier Transform and permanent visual recordings can be made with the aid of an X-Y plotter. Further processing yields a three dimensional display of the transient behaviour which can also be drawn on an X-Y plotter by means of a sophisticated computer programme which causes the tracing pen to lift at intersections between overlapping lines. The recorded digitised impulses are stored either on magnetic disc or on magnetic tape for subsequent replay during further experiments without the necessity of recourse to the original loudspeaker and its acoustical environment.

Using present equipment it takes approximately ten minutes to record a satisfactory impulse response using 500 signal averages. Plotting the graphical record takes only a few seconds longer. Likewise, computing and plotting amplitude and phase response from the recorded impulse takes a few seconds. Producing a graphical record of the three dimensional display currently takes up to one hour, mainly because of the slowness in retrieving information stored on the magnetic tape. Substituting a magnetic disc recorder will considerably shorten the processing time.

Typical Results

A large number of drive units and systems has already been investigated and a tremendous amount of data recorded. The following examples have been chosen to illustrate some interesting applications and some of the visual consequences resulting from various constructional techniques.

Fig. 1 shows a typical impulse response for a single drive unit in a small closed box, a) before signal averaging and b) after 500 averaging operations. The reduction of random noise can be seen quite clearly, especially in the tail of the response. Expanding the tail section reveals considerable recorded detail. Fig. 2 shows the impulse response of 1b expanded by 8 and 64 times respectively. The visual resolution obtained at the higher magnification is greater than 1000:1 or 60dB.

Direct visual inspection of the impulse response can be used to observe certain differences in performance. For example

Fig. 3 shows the effect of changing the voice coil in a 110mm diameter cone type unit. The ringing at 7kHz, due to the mass of the coil resonating with the cone neck compliance, is clearly seen in the tail of the impulse response (3b). By comparison, the normal production type (3a) exhibits quite a different pattern. Changes in cabinet construction are also shown up in a similar way. Fig. 4 demonstrates the resulting change in impulse response from the same drive unit housed a) in a substantially constructed box made from 12mm thick particle board laminated with 12mm thick layers of panel damping material and b) in a box constructed from 6mm thick hardboard. The effects of panel vibration can be seen by comparing the later stages of the impulse response.

Subsequent processing of the impulse response enables the complete frequency response (amplitude and phase) to be computed. The phase response is useful in optimising the relative planes of the drive units so that linear phase-shift due to time delay may be eliminated prior to designing the crossover network.

There is no doubt that the more powerful and dramatic form of presentation is the three dimensional cumulative spectra showing the response of the loudspeaker in terms of amplitude, frequency and time. This often reveals defects which are not otherwise apparent in either the impulse or the frequency responses. A clear example is shown in the decay spectra Fig. 5, where a reflection from the back of the box appears as a ridge running parallel with the frequency axis at approximately $t = 1ms$. The impulse and frequency responses are shown juxtaposed for comparison.

The defective unit illustrated in Fig. 3b is seen again in Fig. 6 where the low damped resonance at 7kHz is clearly evident in the decay spectra. By comparison a corresponding display for a well designed dome type mid-range unit is shown in Fig. 7. The smooth and regular nature of the decay characteristic is obvious.

Very many types of loudspeaker have been investigated using this method. Examples include electrostatics, ribbons, isodynamic types and horn-loaded devices. Widely differing patterns of decay spectra are produced which exhibit characteristic features not readily observed from steady state frequency response curves and other simple data.

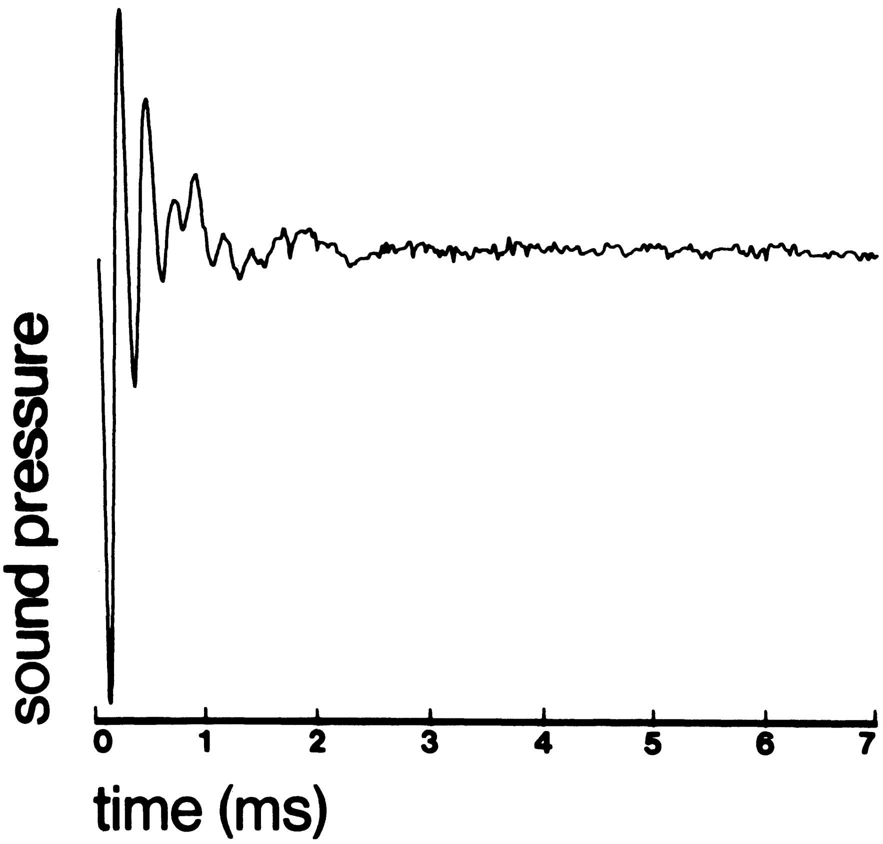


Fig. 1a.

Impulse response of 110mm moving-coil unit in 7 litre closed box

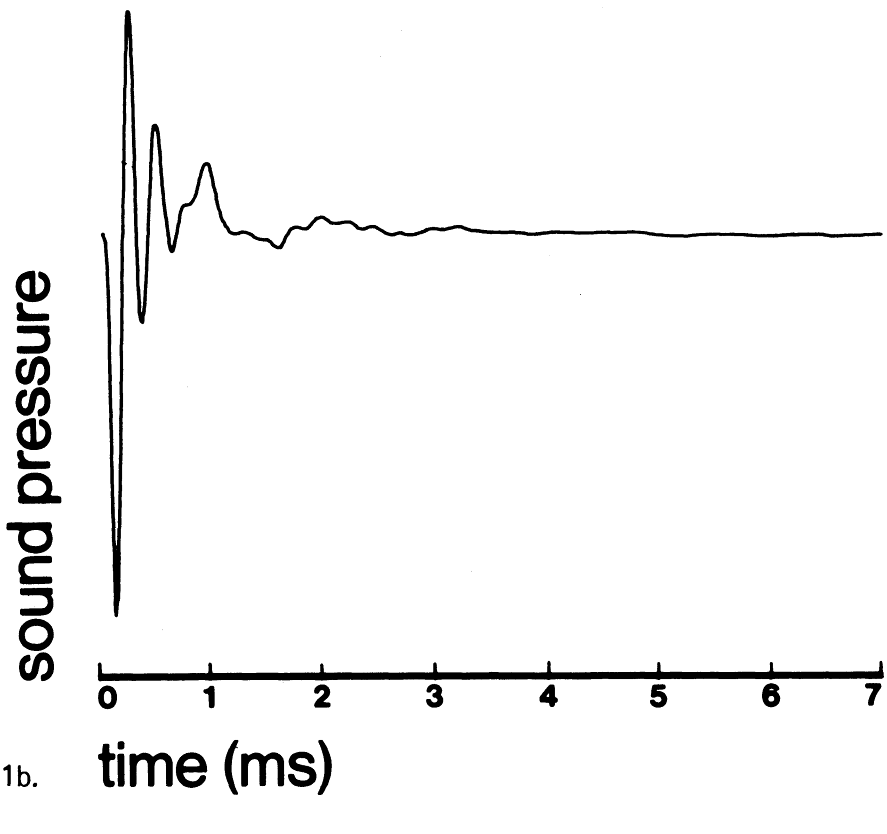


Fig. 1b.

Impulse response averaged 500 times

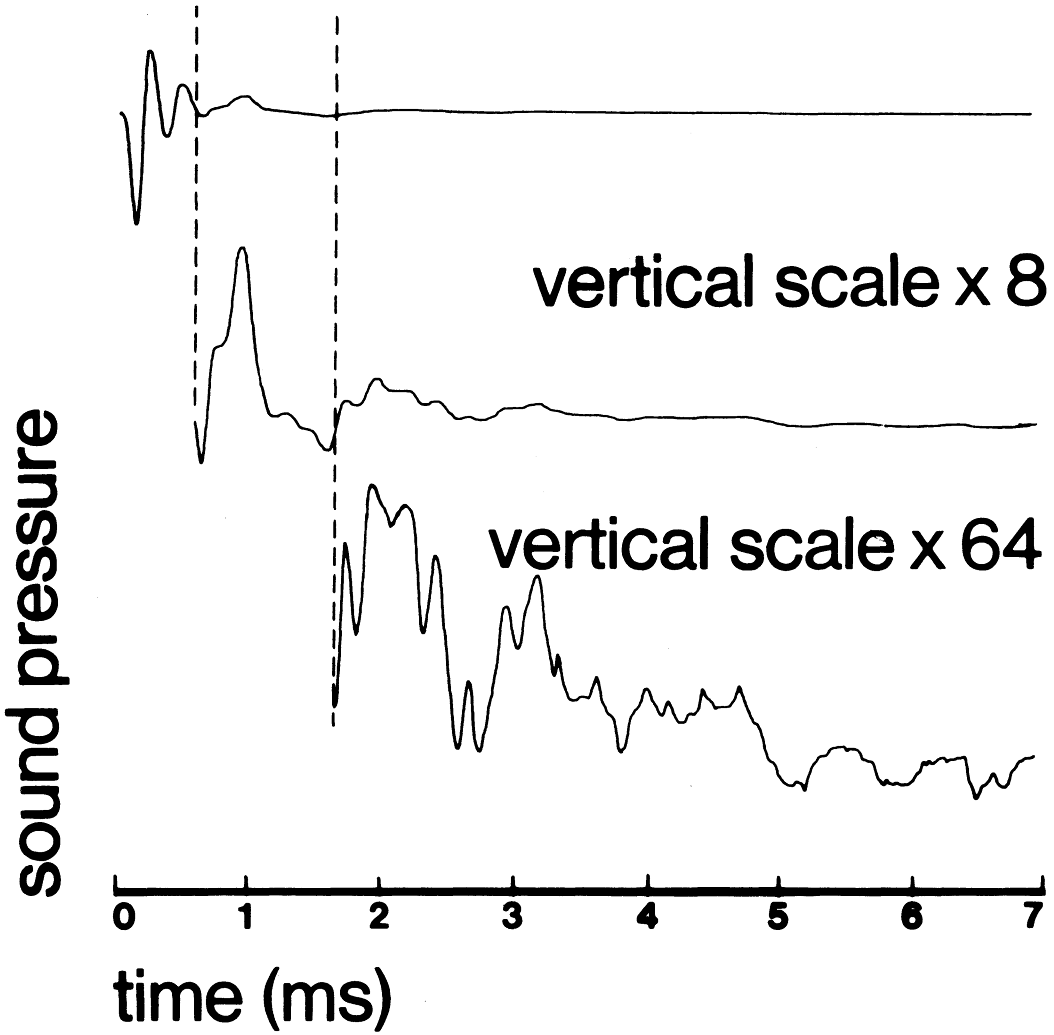


Fig. 2.

Impulse response averaged 500 times with tail magnified to give visual resolution of 500:1 (50dB)

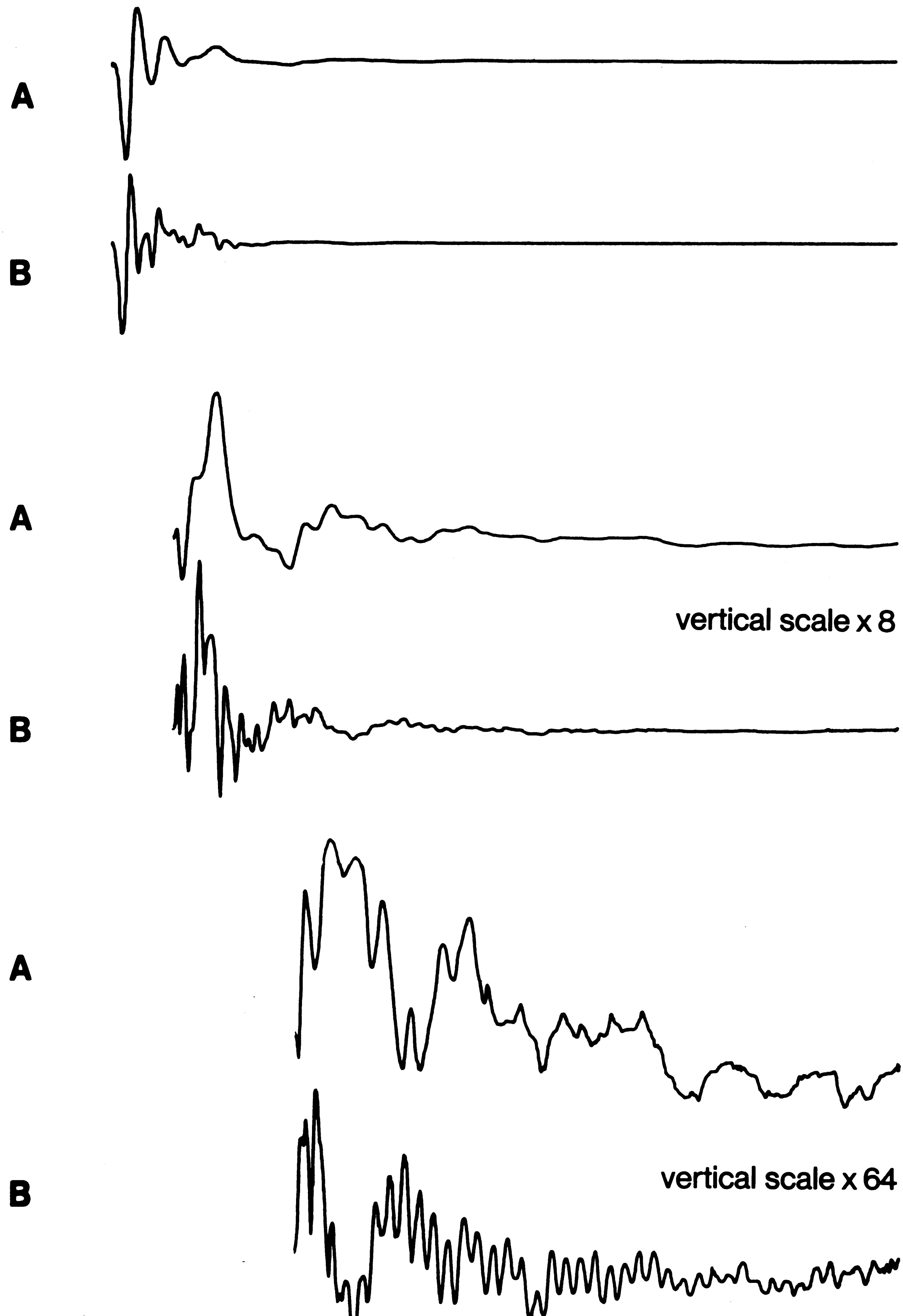


Fig. 3

110mm moving coil units in 7 litre closed box
A control unit
B as control unit but fitted with lightweight voice coil

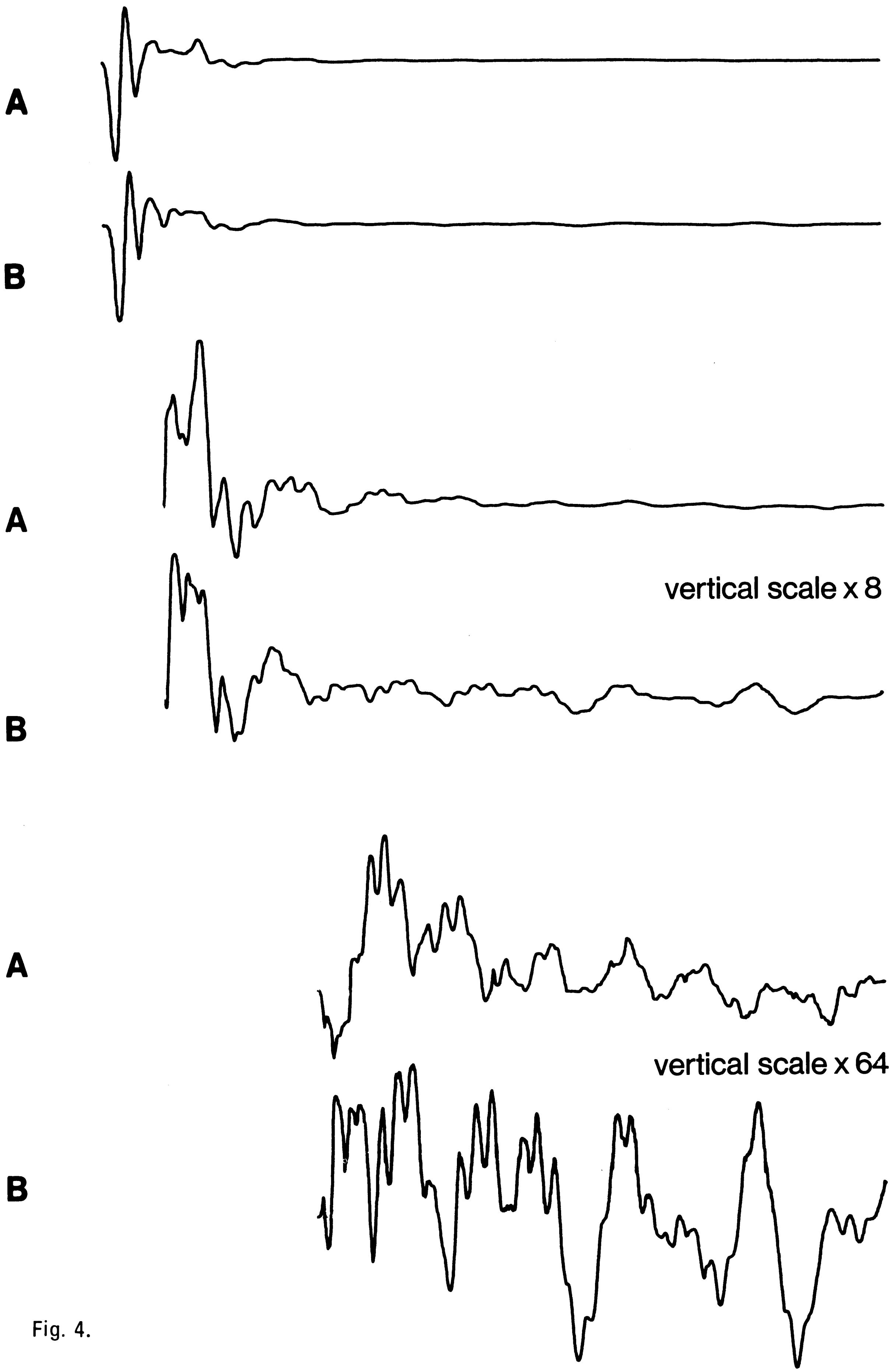
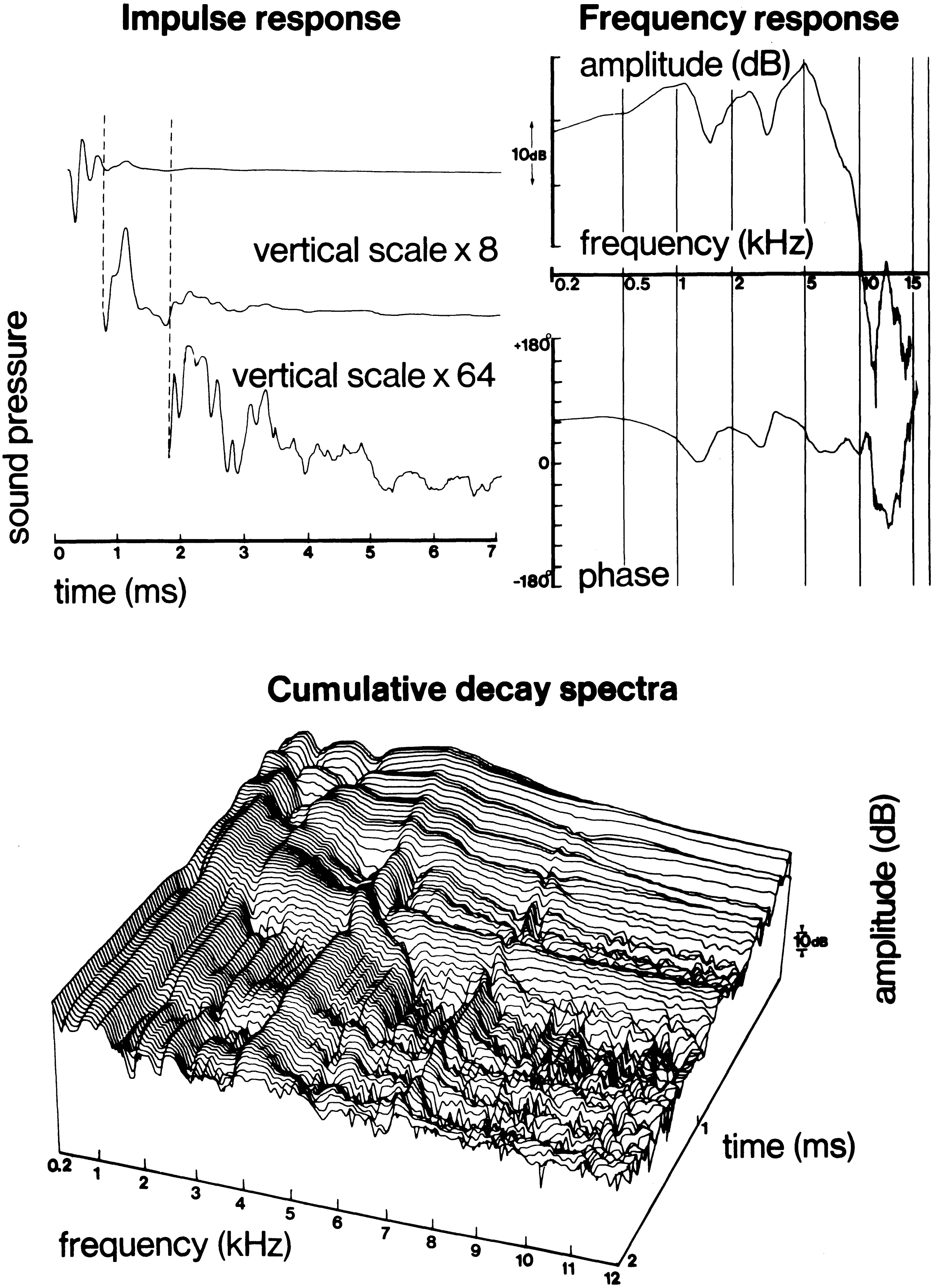


Fig. 4.

110mm moving coil units in 7 litre closed box

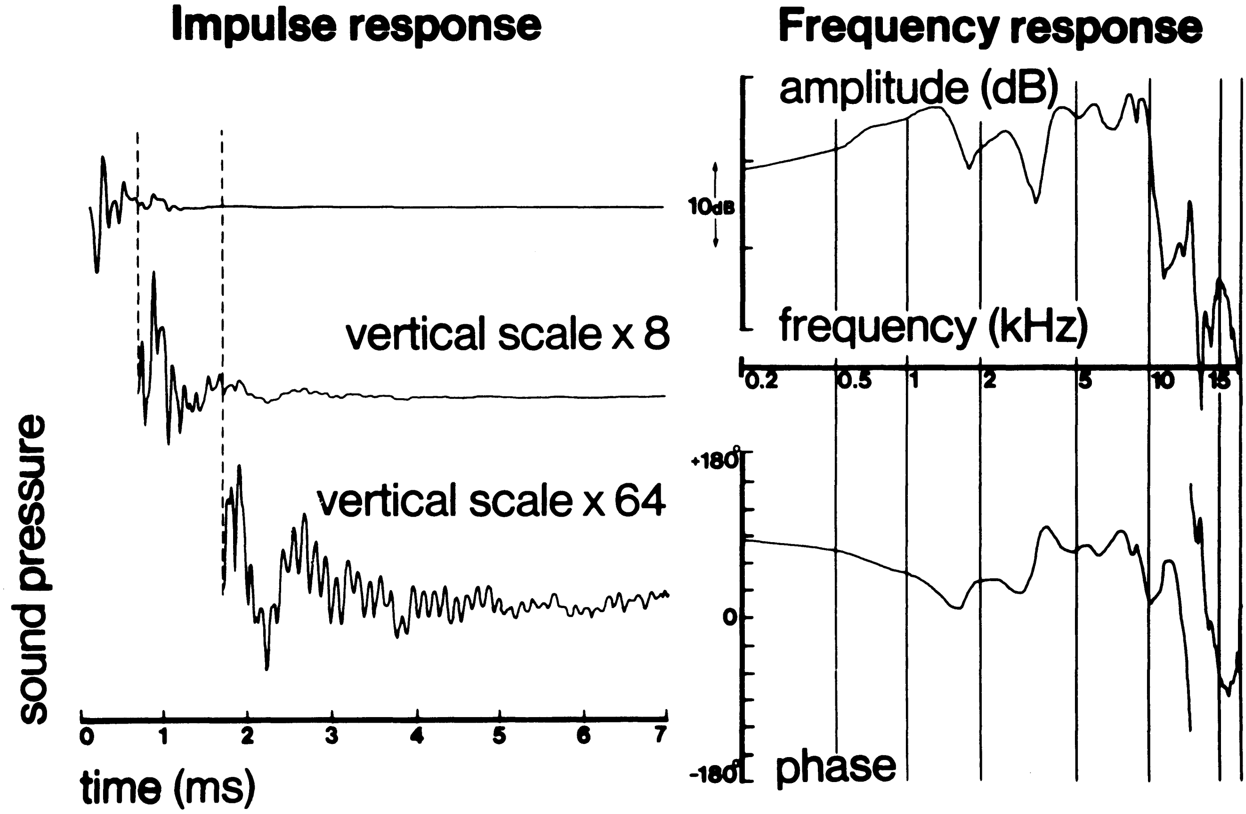
A control unit (box 12mm chipboard with 12mm panel damping material)

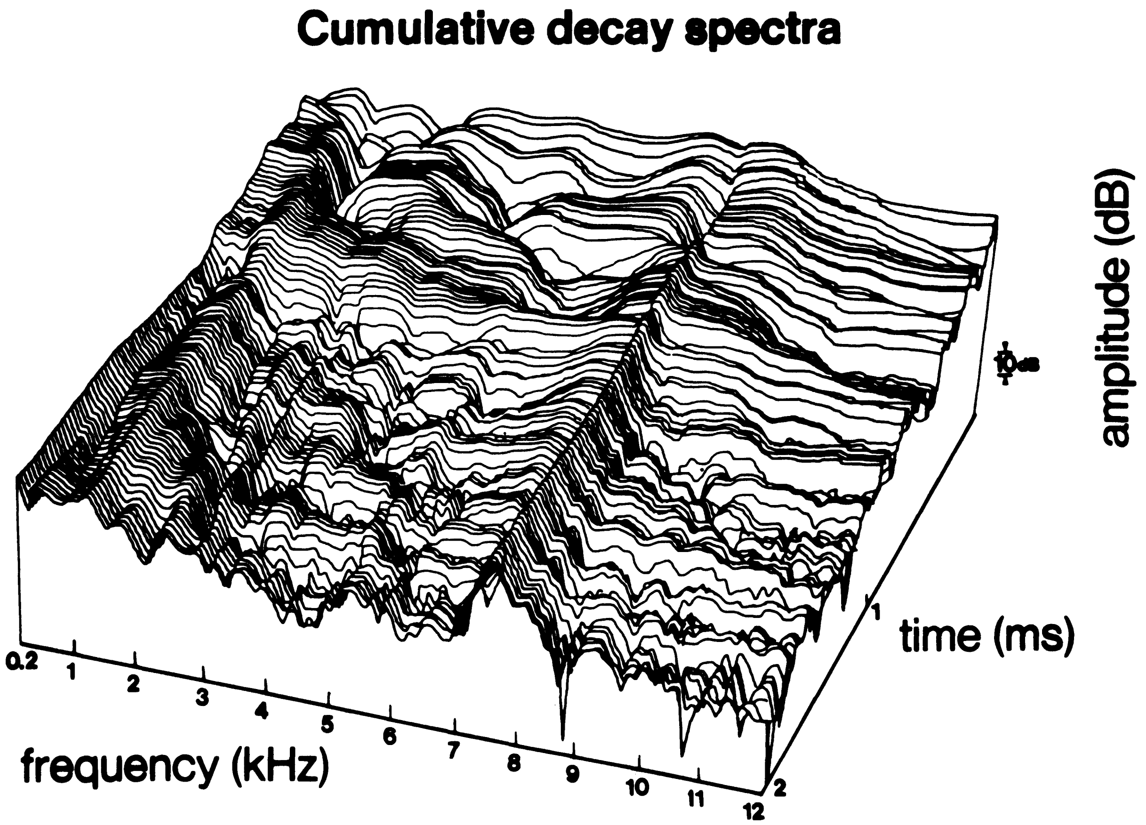
B as control unit but box 6mm hardboard without panel damping



110mm moving coil bass/mid-range unit in 7 litre closed box.

Fig. 5.





110mm moving coil bass/mid-range unit with lightweight coil in 7 litre closed box.

Fig. 6.

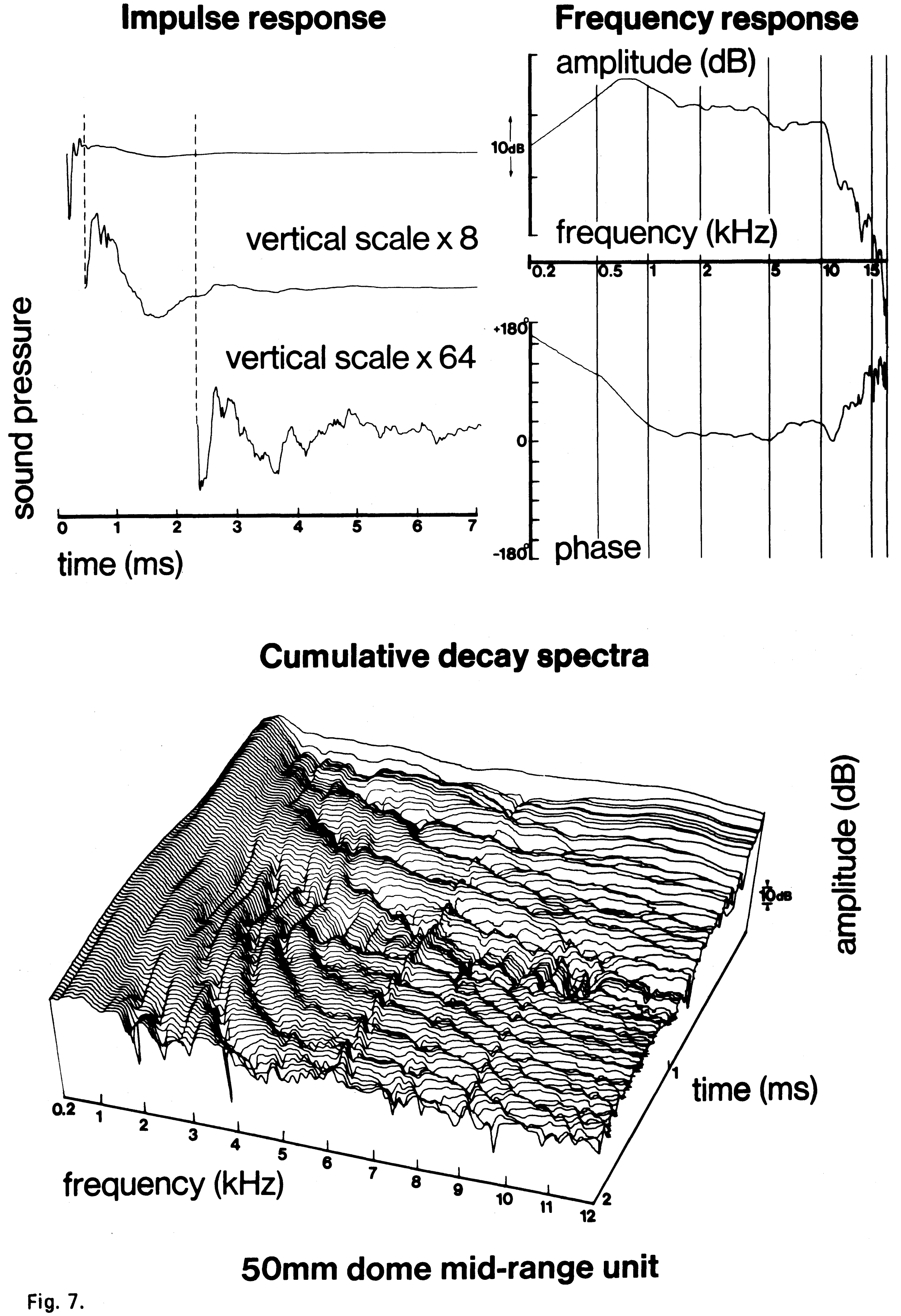

Impulse response
Frequency response
amplitude (dB)
10dB
vertical scale x 8
frequency (kHz)
0.2
0.5
1
2
5
10
15
sound pressure
+180°
vertical scale x 64
0
-180°
phase
0
1
2
3
4
5
6
7
time (ms)
Cumulative decay spectra
amplitude (dB)
10dB
1
time (ms)
0.2
1
2
3
4
5
6
7
8
9
10
11
12
2
frequency (kHz)
50mm dome mid-range unit

Fig. 7.

Applications

(a) Cabinet Design

The use of the technique is not confined to considerations of drive unit behaviour. It is perfectly well adapted for investigations into enclosure design. Useful results have already been obtained from tests on various materials of construction and it is intended to progress measurements towards a full scale evaluation of enclosure performance. In general, cabinets exhibit relatively low damped resonances in comparison with modern high quality drive units and evidence of faulty behaviour is therefore seen in the later stages of the response tail.

(b) Crossover Network Design

The design of dividing networks has hitherto been pursued by measurement of frequency response, terminal voltages and listening tests. These methods have never been satisfactory, particularly because many designs have involved compromises between circuit configurations which give satisfactory performance when judged subjectively and others which produce smoother axial frequency response curves. It has to be remembered that in the present state of the art, considerable importance continues to be attached to amplitude-frequency response curves, particularly in commercial spheres. There is however growing evidence to suggest that an entirely new approach to crossover network design is both necessary and possible. This will be directed towards the elimination of audible phase-shift and the compensation for time delays caused by spatial separation of drive units in multi-speaker systems. Rapid inspection of impulse responses and time delay data is vital in such work.

(c) Quality Control

Because the impulse response of a loudspeaker is, so to speak, its own individual and characteristic signature, since no two are identical, the storage of impulses on magnetic tape provides a convenient means of achieving on-line quality control of production quantities. Such records facilitate highly detailed studies of trends and faults for statistical analysis and greatly assist in improving consistency of performance by careful control of those parameters which significantly affect sound quality.

Currently available loudspeakers may be over-engineered in certain respects in an effort to give a better measured performance which has no subjective significance whilst neglecting other previously unmeasurable aspects which have far greater effect upon the reproduced sound quality.

(d) Performance Criteria

The use of digital techniques for processing impulse responses gives greater insight into the physical behaviour of loudspeakers. Having made it possible to measure a variety of defects, it is necessary to know which faults affect subjective judgement of quality. A technique has therefore been evolved to include the digital processing of music so that the subjective behaviour of a loudspeaker may be assessed. An interesting feature of the method is that such assessments may be carried out on simulated or imaginary loudspeakers as well as on physically existing transducers.

Application of this technique will lead to the estimation of audible thresholds for phase distortion, time delay and amplitude-frequency distortion. The work may also be extended to include room-loudspeaker relationships and the influences of various characteristics on stereo image formation. In fact, by these means, the need for phase coherence throughout the entire recording-transmitting-reproducing chain may be assessed. Studies of these effects may well reveal inadequacies in many parts of the system which have not hitherto received much attention from the standpoint of phase distortion. In particular, tape recorders, multi-microphone systems and noise reduction circuits are all ripe for investigation. With the growth of interest in digital techniques for studio equipment, measured data of the type outlined above may soon be essential.

(e) Summary

The value of digital techniques lies in their great versatility and speed. The ability to obtain very detailed information easily and quickly is more likely to ensure progress in loudspeaker design than infrequent inspired invention. Important additional benefits result from the associated information storage facility, which is an essential factor in quality control. Correlation of objective measurements with subjectively assessed performance is also possible. Similar methods may be applied to the evaluation of other electro-mechanical transducers such as microphones, pickups and headphones as well as complete recording-reproducing systems.

Acknowledgments

KEF Electronics Limited wish to acknowledge the contributions made by Dr. J.M. Berman and Mr. R.V. Leedham at the University of Bradford by their dedication to the research programme during more than three years, Mr. Sri Welaratna for his assistance in preparing material for the A.E.S. papers and Mr. G. Corbett for help in preparing the three dimensional displays. Our thanks are also due to many engineers in all parts of the world who have followed the work with great interest and taken part in helpful discussions. In particular, we wish to thank Prof. J.R. Ashley, Mr. P.J. Baxandall, Mr. R. Heyser, Mr. E.R. Madsen and Mr. D.E.L. Shorter for their useful counsel. Finally, we acknowledge our indebtedness to Hewlett Packard Limited for the loan of equipment on several occasions and for their valuable encouragement of our efforts.

R. E. Cooke
L. R. Fincham

Tovil, March 1975.

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LOUDSPEAKER EVALUATION USING DIGITAL TECHNIQUES

by J. M. Berman, Ph.D., B.Sc.(Eng.)
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A paper presented at the A.E.S. 50th Convention, London, 4th March, 1975.

Introduction

Loudspeaker impulse responses with a wide dynamic range can be obtained with the aid of a digital computer. If it is assumed, in common with network theory, that the system is linear and time-invariant, such an impulse response completely defines the electro-acoustic transducer in its particular relation to the measuring microphone. From this one measurement alone we are then able to derive the corresponding response to any test signal and, in addition, to present the total system information in ways which may communicate more visual information about its behaviour.

This paper outlines the progress of a research programme which uses the impulse response as a starting point for loudspeaker investigations.

The Measurement Technique

An impulse response can be obtained by exciting a loudspeaker with a sufficiently narrow square pulse. The environment need not be anechoic but should be sufficiently large that the response of the transducer dies away before reflections arrive from nearby surfaces. Such a narrow pulse, however, cannot convey sufficient energy to the loudspeaker to give an adequate signal to noise ratio, and so this measurement has not previously been used to advantage. The difficulty is overcome, and this whole new approach made possible by the technique of signal averaging. The pulse is made to be repetitive with a period at least as long as the reverberation time of the room, and the resulting microphone responses are digitised, stored, and averaged by the computer. The impulse response is successively reinforced relative to any random noise and by this means the dynamic range can be improved up to the limit imposed by the analogue to digital convertor. Fig. 1 shows the experimental arrangement as used for the results in this paper.

This technique has been found to give results which are identical, within the limits of experimental accuracy, to traditional free field analogue measurements. Although this is taken in a non-anechoic environment, it is truly a free-field measurement whose low-frequency limit only depends on the size of the room; it is not subject, as are outdoor measurements, to the vagaries of the climate and does not suffer the variability of results from so-called 'anechoic' chambers.

The Impulse Response

One of the most remarkable results of this work has been the discovery of how much information can be obtained directly from a visual inspection of the impulse response prior to any computation.

Fig. 2(a) demonstrates the measured impulse response of a 5 inch unit in a 7 litre closed box. 2(b) and 2(c) show the tail expanded 8 and 64 times respectively, giving a visual dynamic range of over 50dB. Such waveforms are found to be repeatable, even in the tail of the response, for any given unit. Different samples of nominally identical units have impulse responses features which remain practically invariant, but any small alteration in materials or construction, however, produces a dramatic change in the response. It has been found that particular features can be related directly to surround, voice coil, cone, etc.

While the early part of the impulse response is related predominantly to the loudspeaker drive unit, the latter part is dominated by sound radiating from the enclosure walls. Each enclosure exhibits its own characteristic 'signature' waveform which is substantially independent of the unit. This same waveform can be identified in the box impulse response, obtained by firing a short pulse into the box. Since this method allows identification of the separate contributions of transducer and box, it provides a unique way of investigating both box colouration and the complex problem of unit/box interaction.

Frequency Response

The frequency response given in Fig. 3 computed from Fig. 2, gives the customary amplitude response accompanied by its phase response, (a), which completes the complex function and from which the associated group delay characteristic can be derived.

If the loudspeaker falls into the classification of a minimum-phase system, amplitude and phase are uniquely related; since one may be derived from the other by means of the Hilbert Transform, knowledge of either is sufficient completely to describe the system behaviour. To determine whether a loudspeaker is a minimum-phase system, the Hilbert Transform can be computed and compared with the measured phase response. The example given in Fig. 3(b) demonstrates that above 3kHz this loudspeaker exhibits some non-minimum phase behaviour. If equalisation of such a loudspeaker is attempted in order to flatten the frequency response in the non-minimum phase region, then regard must be paid to both amplitude and phase response of the equalising network. It is one of those recurrent ironies; we have to measure phase to find if it can be neglected.

Of the many types of transducer so far investigated, most have been found to exhibit minimum-phase behaviour over a substantial part of the frequency range. However, when combined by means of a crossover network, the resulting multi-way system is generally found to be non minimum phase.

Cumulative Spectra

Impulse response and frequency response are statements of system behaviour in the two mutually exclusive domains of time and frequency. The ear and brain, however, are not restricted to evaluating sound in one domain or the other, for we experience pitch, changing as a function of time. It seems sensible, this being so, to pursue a method of presenting the system information in a form which may correlate more closely with subjective response.

Three-dimensional cumulative spectra displays, which relate both time and frequency during build-up and decay, may be computed from the impulse response. Fig. 4 shows the first 2msecs of the decay spectra of Fig. 2. It can be shown mathematically that a cross-section parallel to the time axis gives the log. magnitude of the envelope of the toneburst response at that frequency. But this three-dimensional picture is capable of displaying far more subtle information than that which is obtained directly from the toneburst response as it is usually displayed. In particular, it can give us information which cannot be deduced from either the impulse response or the frequency response.

A considerable range of loudspeaker systems have now been investigated, and the form of the 3-dimensional relief is found to vary dramatically. An electrostatic speaker, for instance, gives a quite different result from that of a conventional moving coil, and different again from that of a dome unit.

Fig. 4 demonstrates some of the effects which can be observed with a moving coil unit in a closed box. There are clear examples of delayed resonances, ridges which lie across the entire frequency range, suggesting reflections (in this case from the rear of the enclosure) and complex frequency/time effects, some moving diagonally across the display. None of these are obvious from either the impulse, or frequency response.

Summary

Accurate and repeatable loudspeaker impulse responses with wide dynamic range may now be measured and stored using a digital computer without the need for an anechoic environment. The steady state and transient behaviour may be derived from such an impulse response and are at least as good as equivalent results from analogue measurements. In addition, digital processing techniques may be applied to the stored impulse response to reveal aspects of the system behaviour quite inaccessible with traditional measuring techniques.

At present, listening tests are still the only means of revealing many of the subtle differences between loudspeakers. There is now evidence that digital techniques may provide measured confirmation of these audible differences and so facilitate more objective approach to loudspeaker design.

Layout for loudspeaker impulse measurements

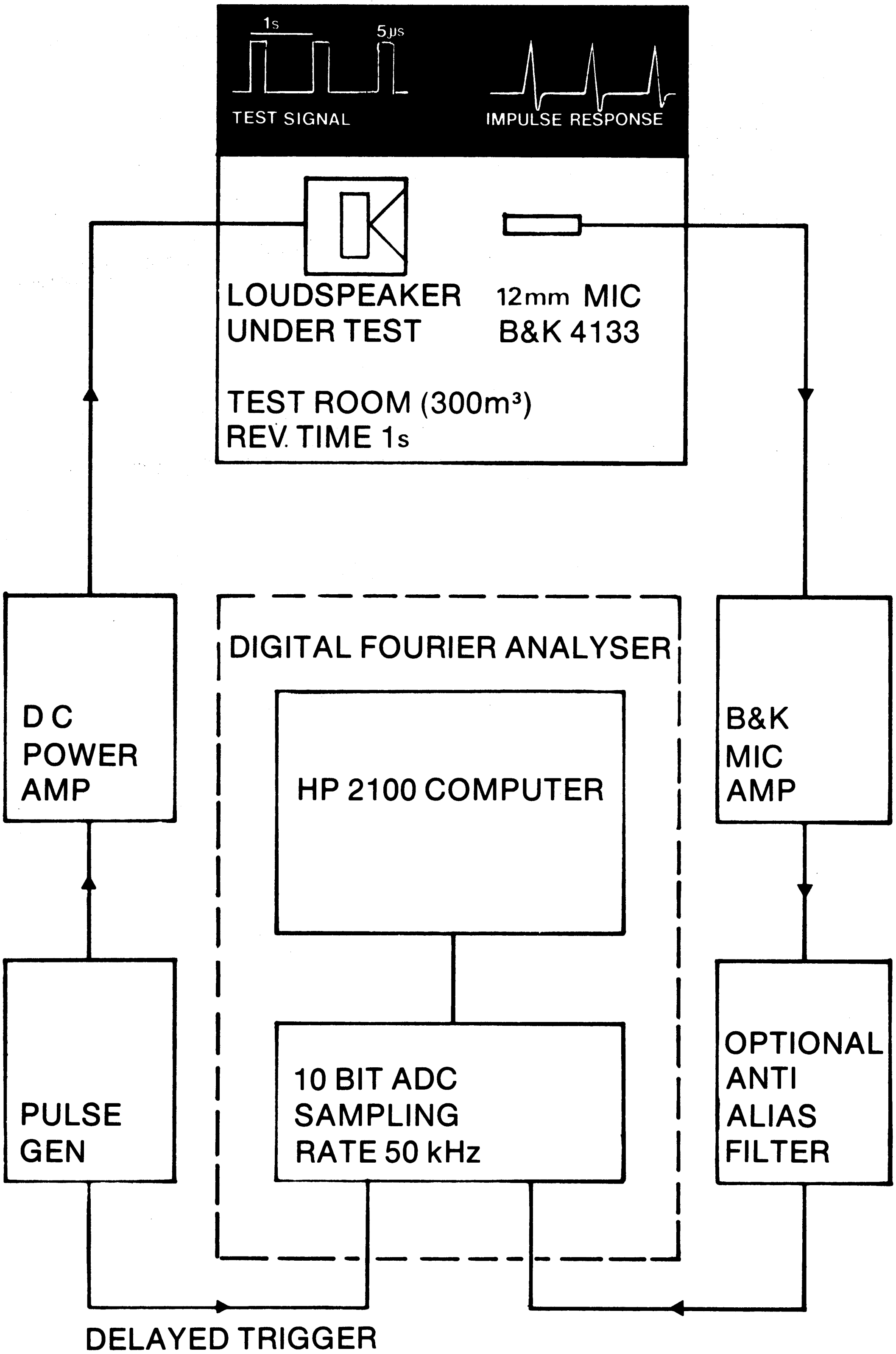


Fig I

Time

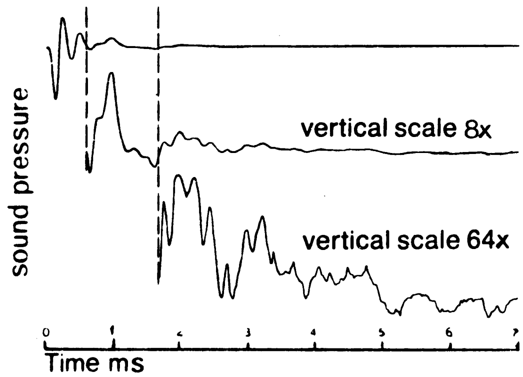


Fig 2 Measured impulse response of 5″ dynamic loudspeaker in 7 litre closed box

Frequency

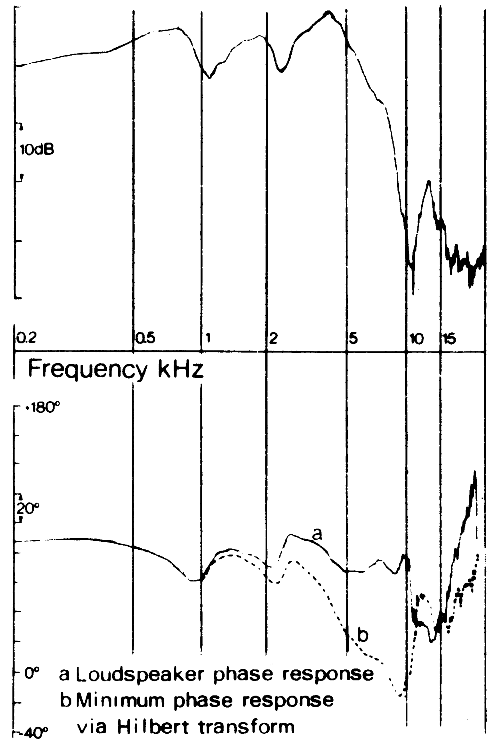


Fig 3 Frequency response computed from impulse response

Time/frequency

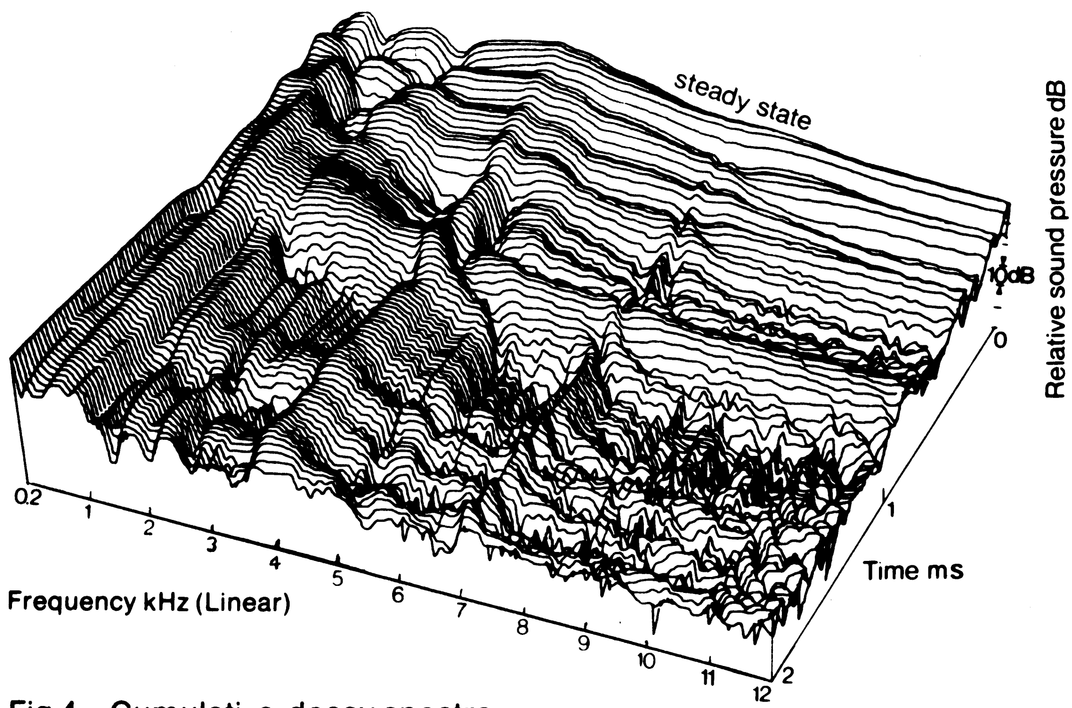


Fig 4 Cumulative decay spectra

LOUDSPEAKER SYSTEM SIMULATION USING DIGITAL TECHNIQUES

by L. R. Fincham, B.Sc.(Eng.)
Technical Director, KEF Electronics Limited

A paper presented at the A.E.S. 50th Convention, London, 4th March, 1975.

Introduction

Objective loudspeaker measurements are usually made in order to help the engineer to eliminate the causes of audible defects in a loudspeaker and enable the system design to be optimised. Unfortunately, certain defects, notably colouration, are not readily diagnosed using traditional analogue measuring methods although their presence is all too readily detected during listening tests. A new measuring method, using digital techniques, has been developed recently for obtaining the impulse response of a loudspeaker. Considerable additional insight into the objective performance of a loudspeaker is now obtained by either direct visual examination of the recorded impulse response or of other displays obtained from subsequent processing of the impulse response using a digital computer. All this additional information is of no value, however, unless its subjective significance can be established. It is a loudspeaker's sound quality that matters rather than its measured performance.

Listening Tests

The subjective performance of a loudspeaker is usually assessed by listening to its reproduction of a variety of programme material under carefully controlled conditions. The sound from the loudspeaker may then be compared either with a remembered impression of the original sound or directly with that of another loudspeaker (AB testing). Using this method, experienced listeners can often achieve a surprising degree of consistency in their rank ordering of a selection of loudspeakers and in their detection of aural defects. The factors affecting their choice, however, are seldom evident from a visual inspection of the measured characteristics of the loudspeaker. This lack of correlation may be partly because objective measurements are usually made with a single microphone under free field conditions, whereas listening assessments are made binaurally in a semi-reverberant room where there is inevitable interaction between the loudspeaker and the room due to reflections.

An alternative subjective assessment method may be used which eliminates many of the above variables and can prove useful in optimising the performance of a loudspeaker system since it is independent of the listening environment. The general arrangement for this method is shown in Fig. 1.(a). The loudspeaker system under test is placed in the environment used for its objective assessment and its output is relayed, via the measuring microphone and a reference loudspeaker, to a listener situated in a typical listening room. The input source material may be switched directly to the reference loudspeaker (B) or via the test loudspeaker/ microphone chain (A). Under these conditions there will be no audible difference between the two switch positions, for a subjectively "perfect" loudspeaker, providing the input levels for A and B are adjusted to be the same. For a less than "perfect" loudspeaker there will be a significant audible difference and modifications must be made to the loudspeaker system if this difference is to be reduced to an acceptable level.

It would be helpful if proposed improvements to the loudspeaker could be assessed subjectively without the need to make physical changes to the system. This is now practical using a modification to the listening test just described in which the loudspeaker and measuring microphone chain are replaced by a digitially realised system having the same measured impulse response h(t) (Fig. 1.(b)). The output from the test loudspeaker may be simulated digitally by processing the programme material using a computer, and the stored impulse response of the loudspeaker. The quality of the simulated loudspeaker and the original programme may then be compared directly using the reference loudspeaker. Any proposed changes to the loudspeaker are simulated through modifications to the stored impulse response and their subjective significance assessed directly.

<u>Digital Simulation of Linear Systems</u>

A linear time invariant system is completely defined in the time domain by its impulse response h(t) and in the frequency domain by its complex frequency response (amplitude and phase) or transfer function H(f) (See Fig. 2). The output of the system to any input signal may be calculated using the following relationships:

$y(t) = h(t)*x(t)$ Time domain convolution

Where $y(t) = \int_{-\infty}^{\infty} h(\tau)x(t-\tau)d\tau$ eq.(1)

and $Y(f) = H(f) \cdot X(f)$ Multiplication of spectra.

h(t), x(t) and y(t) are related to H(f), X(f) and Y(f) by the Fourier transform.

$$H(f) = \int_{-\infty}^{\infty} h(t)e^{-j\omega t}dt$$

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-j\omega t}dt \qquad \text{eq.(2)}$$

$$Y(f) = \int_{-\infty}^{\infty} y(t)e^{-j\omega t}dt$$

These integrals may be conveniently evaluated using a digital computer. The digital simulation of a system having a given impulse response may be realised by computing the output wave form y(t) for a given input x(t) by either:

(a) direct time domain convolution of x(t) with h(t) (See eq. 1)

(b) using the Fourier transform to obtain Y(f) (See eq. 2) and the inverse Fourier transform to evaluate y(t) from Y(f).

The indirect method (b) is faster if use is made of the fast Fourier transform (FFT).

<u>Practical Digital Simulation Method</u>

A diagrammatic representation of the procedure currently being used for programme processing is shown in Fig. 3. The programme material is sampled and digitised by means of a high quality 14 bit analogue to digital convertor (ADC) and stored on a digital magnetic tape unit. The sampling frequency is limited by the maximum data transfer rate to the digital magnetic tape unit to just over 16 kHz which allows a maximum upper frequency limit for the input signal of 6 kHz if a 9th order Darlington low pass filter is used to avoid aliasing.

Music programme with an upper frequency limit of 20 KHz is acquired by replaying 15 ips master tapes at ¼ speed on a high quality analogue recorder (NAGRA IV SJ). The programme material is processed by direct time domain convolution with the stored impulse response of a loudspeaker using the HP2100 computer. Convolution is preferred to the faster FFT approach for practical reasons and the need to retain the 14 bit overall accuracy necessary for high quality reproduction of music. Currently the processing time required for the convolution of a 4k data block with an impulse response of 256 points is 120 seconds. After processing the digitised music is played back via a 14 bit digital to analogue convertor (DAC), a 6 kHz low pass filter to reconstitute the analogue wave form, recorded at 3¾ ips on the NAGRA IV SJ and finally replayed at 15 ips.

The system described in this paper has only recently been developed, and the results obtained so far do no more than confirm the validity of the method. It is already clear, however, that digital simulation may be usefully applied to a number of presently unresolved loudspeaker problems. Among these are the subjective importance of amplitude, phase, and time-delay distortion and the loudspeaker room relationship.

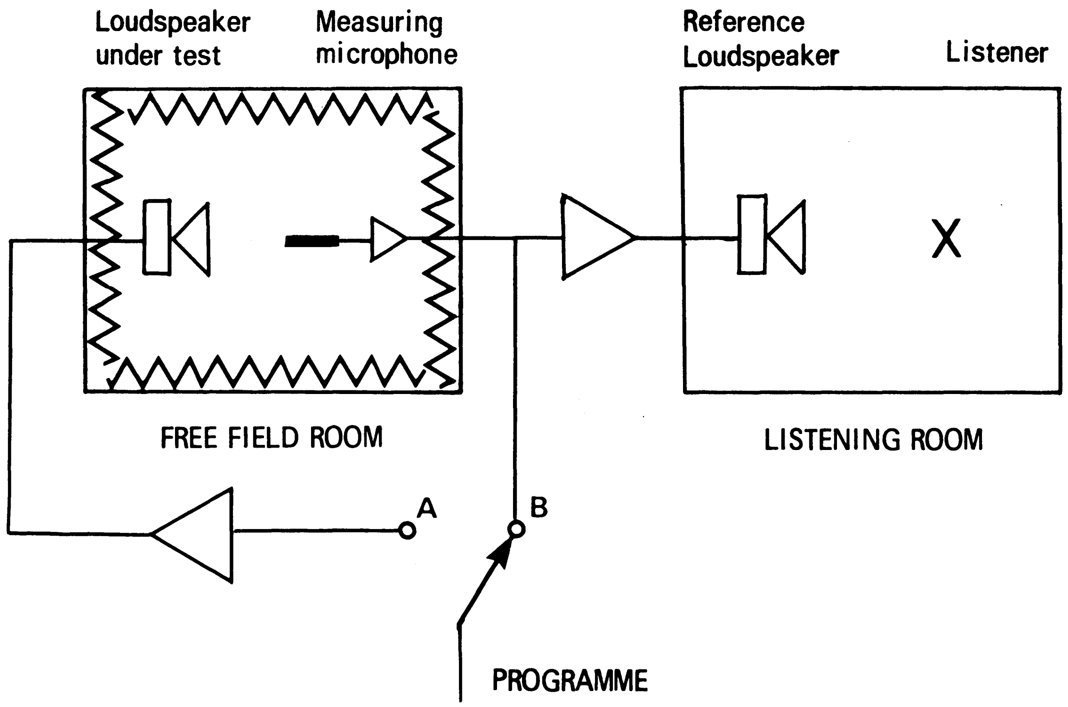

Loudspeaker
under test
Measuring
microphone
Reference
Loudspeaker
Listener
X
FREE FIELD ROOM
LISTENING ROOM
A
B
PROGRAMME

Fig. 1a.

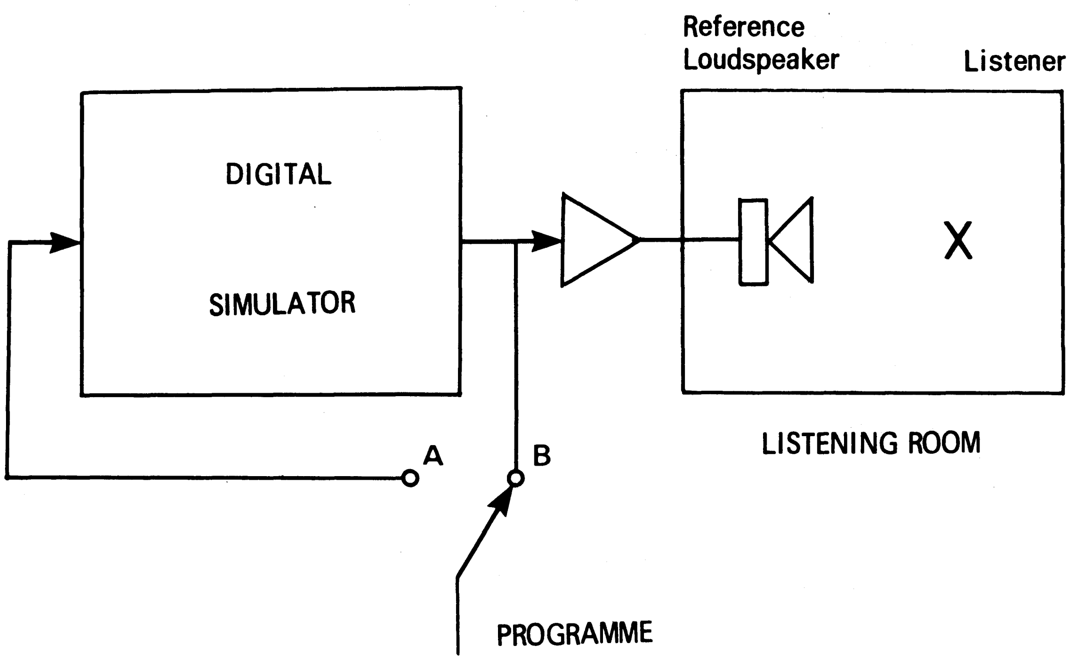
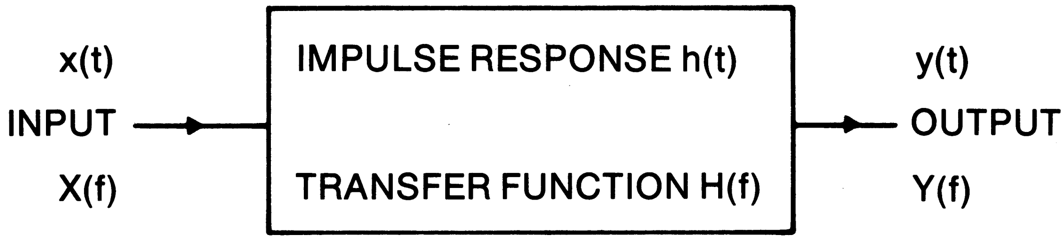

Reference
Loudspeaker
Listener
DIGITAL
SIMULATOR
X
LISTENING ROOM
A
B
PROGRAMME

Fig. 1b.

Loudspeaker system linear model

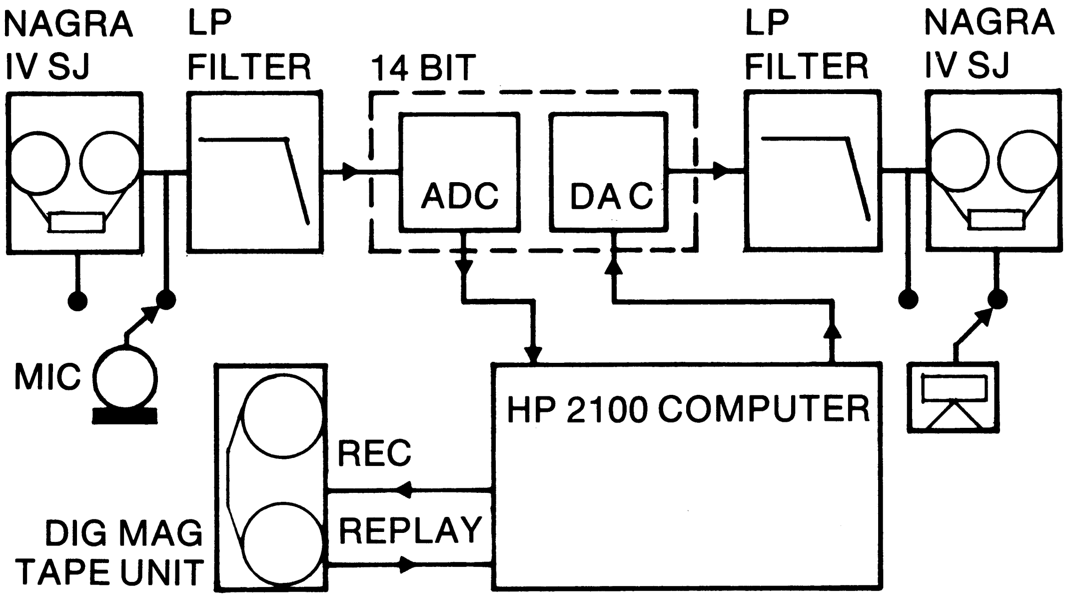


Y(f)=X(f). H(f) MULTIPLICATION OF SPECTRA

y(t)=x(t) ✱ h(t) CONVOLUTION OF TIME FUNCTIONS

Fig. 2.

Digital record/replay system



Time domain convolution process

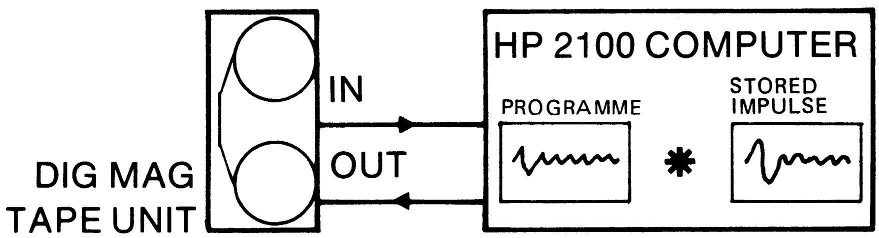
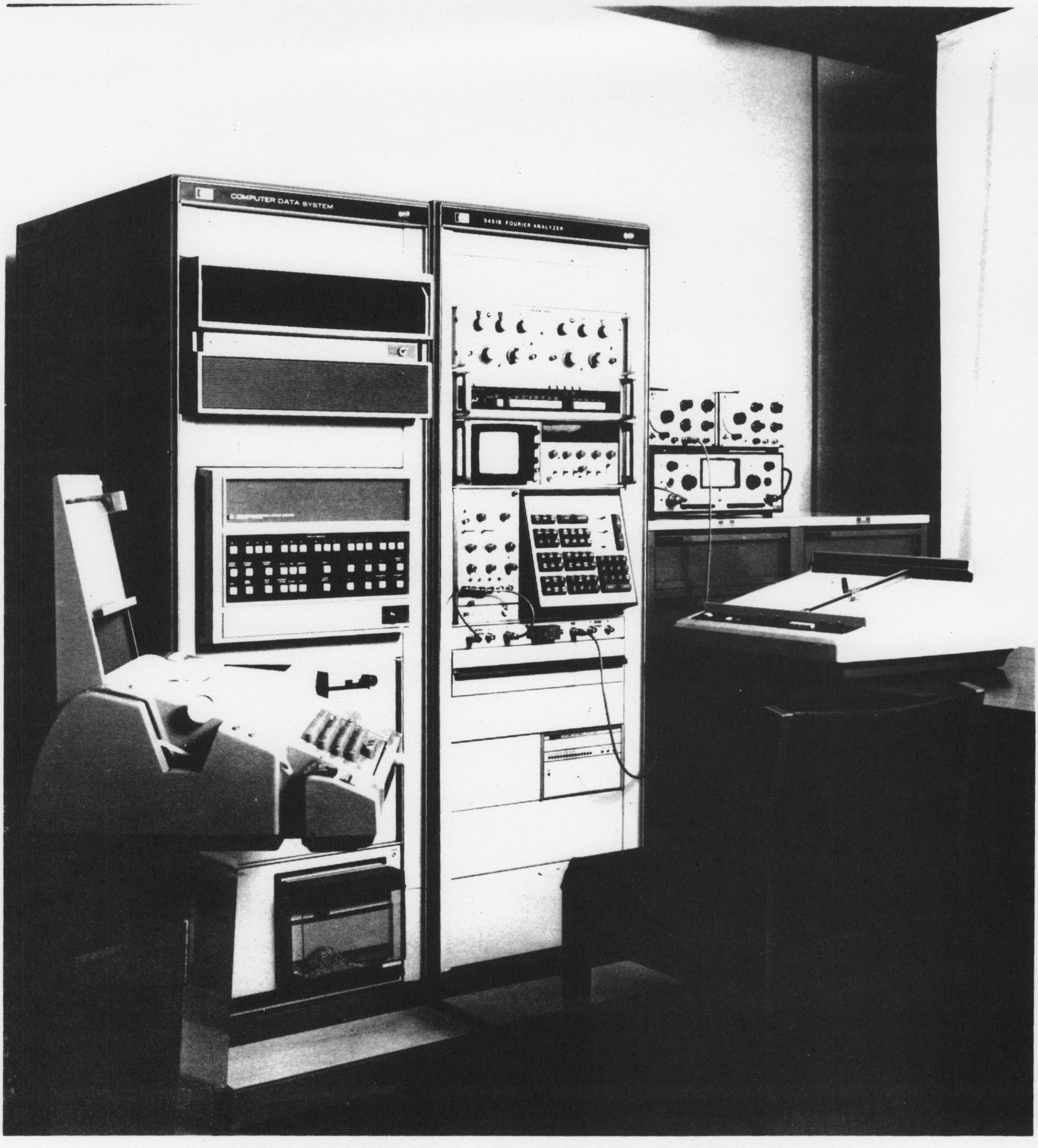
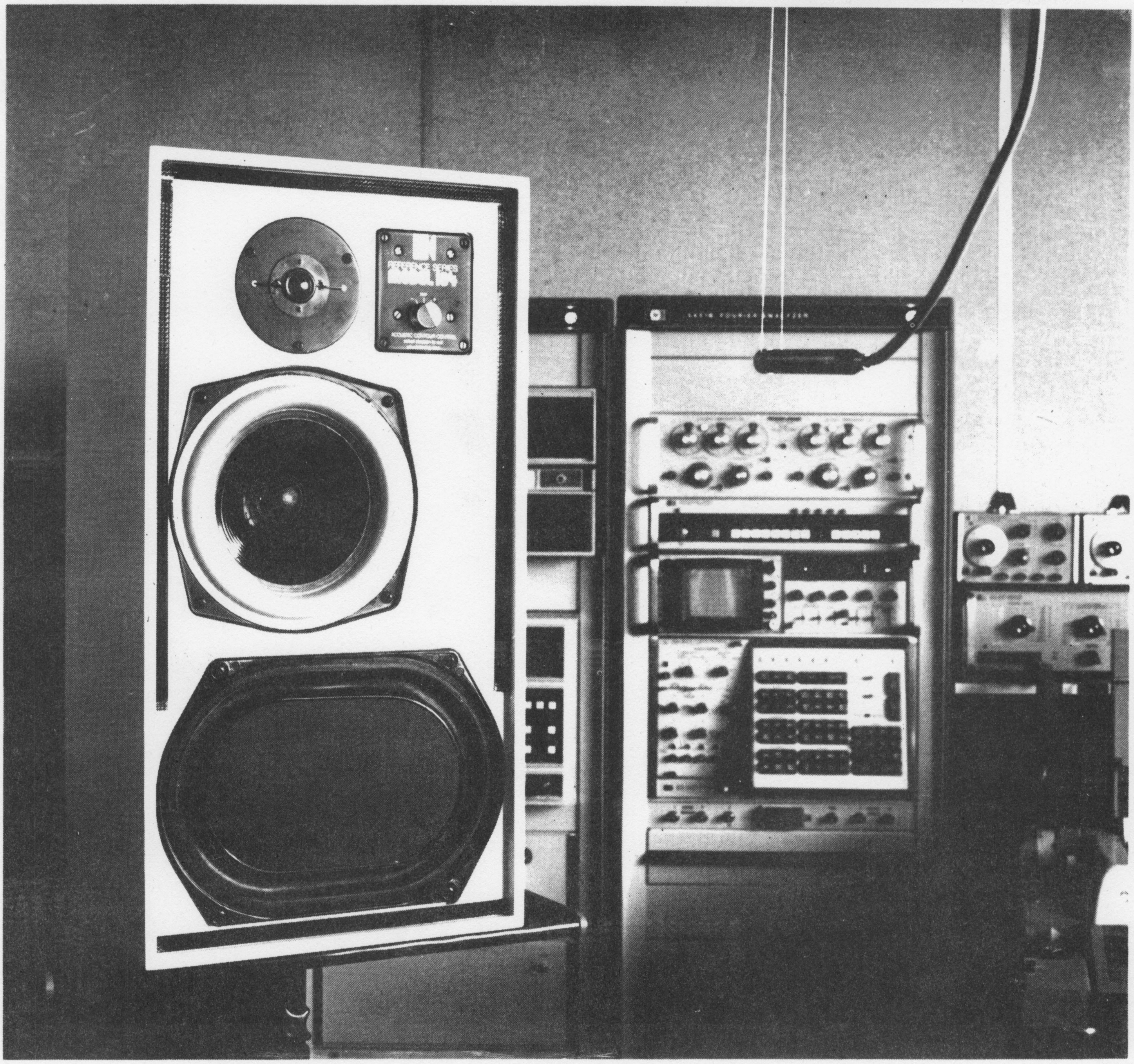


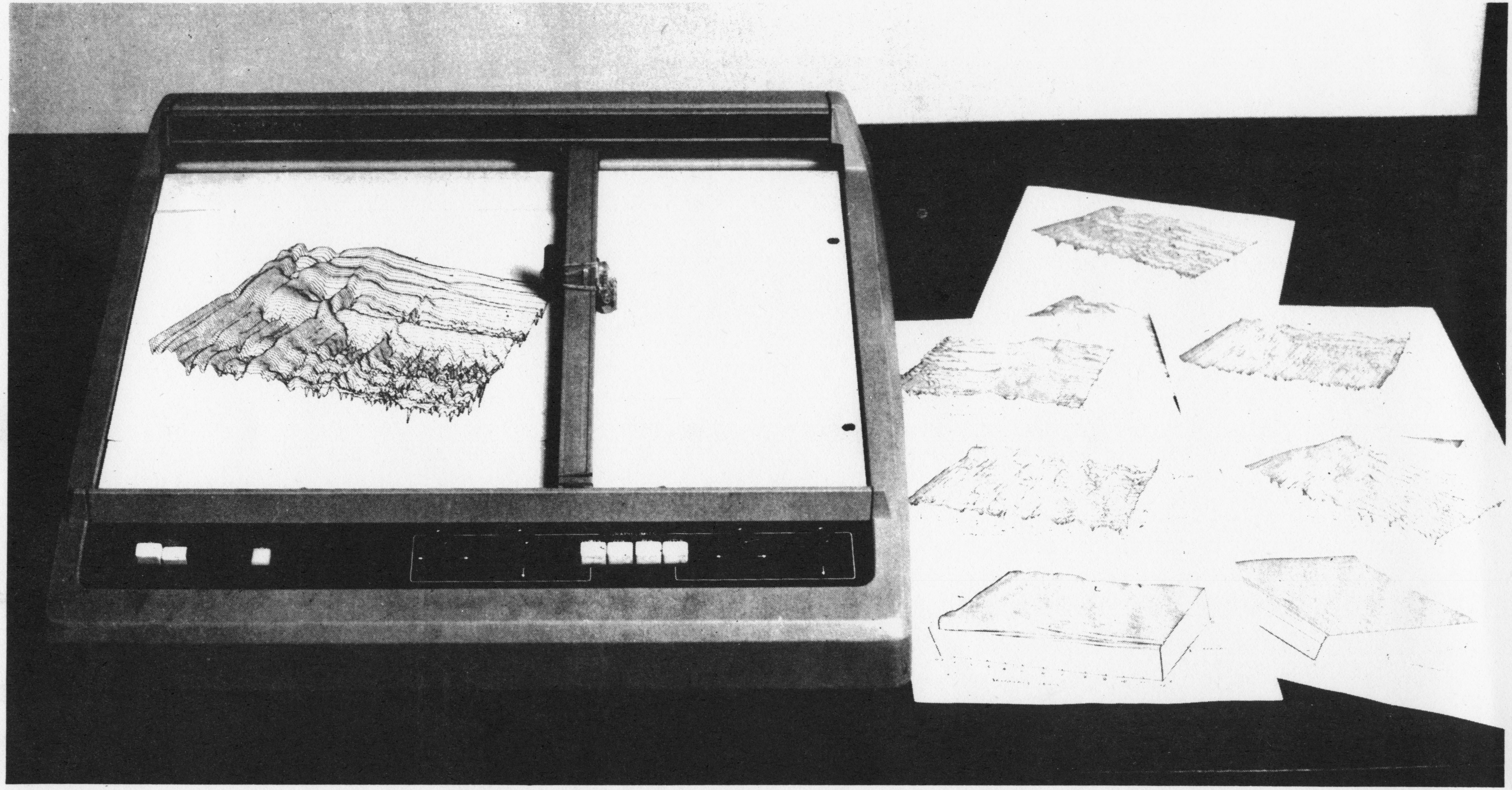
Fig. 3.



Computer equipment used by KEF for measuring impulse responses of loudspeakers and recording processed results.



Impulse response measurement on KEF Model 104 in a live room, showing capacitor microphone in foreground and Fourier analyser at the rear.



Three dimensional cumulative spectra for typical loudspeakers recorded by digital X-Y plotter.