BBC

ENGINEERING DIVISION

MONOGRAPH

BRITISH BROADCASTING CORPORATION

No. 8

AN AUTOMATIC INTEGRATOR FOR
DETERMINING THE MEAN SPHERICAL
RESPONSE OF LOUDSPEAKERS AND
MICROPHONES

by A. GEE, M.A.
D. E. L. SHORTER, B.Sc.(Eng.), A.M.I.E.E.
(*Research Department, BBC Engineering Division*)

No. 8

AN AUTOMATIC INTEGRATOR FOR DETERMINING THE MEAN SPHERICAL RESPONSE OF LOUDSPEAKERS AND MICROPHONES

by

A. Gee, M.A.

D. E. L. Shorter, B.Sc.(Eng.), A.M.I.E.E.

(RESEARCH DEPARTMENT, BBC ENGINEERING DIVISION)

AUGUST 1956

BRITISH BROADCASTING CORPORATION

FOREWORD

THIS is one of a series of Engineering Monographs published by the British Broadcasting Corporation. About six will be produced every year, each dealing with a technical subject within the field of television and sound broadcasting. Each Monograph will describe work that has been done by the Engineering Division of the BBC and will include, where appropriate, a survey of earlier work on the same subject. From time to time the series will include selected reprints of articles by BBC authors that have appeared in technical journals. Papers dealing with general engineering developments in broadcasting may also be included occasionally.

This series should be of interest and value to engineers engaged in the fields of broadcasting and of telecommunications generally.

Individual copies will cost 5s. post free, while the annual subscription is £1 post free. Orders can be placed with newsagents and booksellers, or BBC PUBLICATIONS, 35 MARYLEBONE HIGH STREET, LONDON, W.1

CONTENTS

PREVIOUS ISSUES IN THIS SERIES

AN AUTOMATIC INTEGRATOR FOR DETERMINING THE MEAN SPHERICAL RESPONSE OF LOUDSPEAKERS AND MICROPHONES

SUMMARY

To assess the performance of a loudspeaker or microphone in reverberant surroundings, it is desirable to know the mean spherical response. Unfortunately, the labour of computing this quantity from experimental data is prohibitive in all but the simplest cases.

Apparatus has been designed for use in conjunction with existing polar plotting equipment to compute automatically the mean spherical response; the necessary integrations can be carried out simultaneously with the measurement of the polar characteristics.

The operation of the circuit is illustrated by reference to the case of a loudspeaker rotated, under free-field conditions, before a stationary microphone. The audio-frequency output voltage from the microphone is amplified, rectified, and converted by a vibrating interrupter to a fixed frequency of 50 c/s; the resulting signal is applied, through separate amplifiers, to the voltage and current coils of a kilowatt-hour meter which thus registers a quantity proportional to the square of the sound pressure and hence to the rate of energy flow per unit area of wave front. By rotating the loudspeaker so that every part of the radiated sound field is explored, the total acoustic power output, expressed in terms of the mean spherical response, is determined.

Commercial kilowatt-hour meters are not intended to register accurately over a wide range of voltage; in the meter employed for the present purpose, this difficulty has been overcome by the use of flux-sampling feedback in the voltage coil amplifier and of an independent driving system for friction compensation.

1. Introduction

The sound heard from a loudspeaker in a live room arrives at the listener's ear not only by the direct path but also by indirect routes involving one or more reflections from the walls, floor and ceiling. In general, the ear is conscious of both the direct and reflected sound, and it has long been recognized[1] that in any objective assessment of the over-all effect, both must be taken into account. Various workers[2,3,4] have used as a measure of the total reflected sound, the mean spherical response of the loudspeaker, a quantity analogous to mean spherical candle power in photometry and obtained by averaging the sound intensity radiated in all directions. This quantity when presented as a function of frequency is known as the 'mean spherical frequency response' and can be used, in conjunction with the usual frequency response characteristics relating to the direct sound, to assess the effective characteristics of loudspeaker plus room. It may be of interest to note that a current British Standard[5] on loudspeaker testing includes a section dealing with the measurement of mean spherical frequency response.

Similar considerations apply to the mean spherical response of a microphone, which may be described as the response to sound arriving from all directions with random phase. This quantity is a measure of the sensitivity to reverberant sound and its variation with frequency has a bearing on the quality of reproduction.

One method of arriving at the mean spherical frequency response of a loudspeaker or microphone is by calculation from polar characteristics measured at a series of single frequencies or narrow bands of frequency, using Rousseau's or Kelly's graphical construction, or other methods, involving plotting polar diagrams on a distorted scale. This procedure is however tedious even when the instrument under test has axial symmetry; where this is not the case, the labour of computation becomes prohibitive.

Alternatively, a direct measurement of mean spherical response may be made by placing a loudspeaker and microphone, one of which is the subject of the test, in a highly reverberant room[1,3,4], such as is used for measuring acoustic absorption, the two instruments being so disposed that direct transmission of sound from one to the other is negligible. The principle of the method is analogous to that of the integrating photometer and likewise assumes complete diffusion of the radiated energy. In acoustic measurements, however, the wave-lengths involved are relatively long so that adequate diffusion is not easily achieved; various precautions are therefore necessary to avoid anomalous results at low frequencies, particularly in the case of loudspeakers, the efficiency of which may be modified by the acoustic load imposed on them by the room. In one application[4] of the method, for example, the required conditions were approximated by the use of cylindrical reflectors together with a large vane which was slowly rotated to vary the interference pattern in the room.
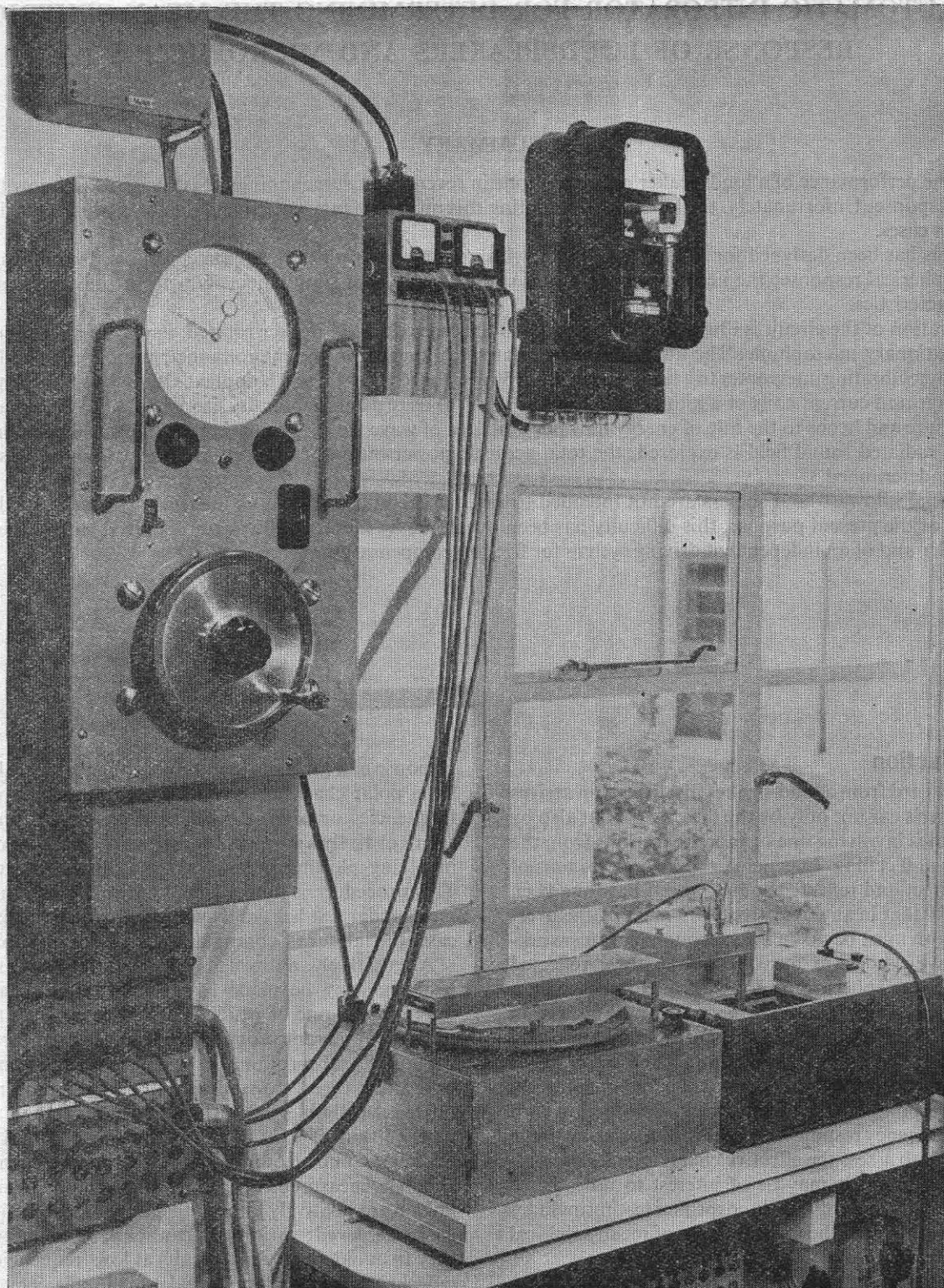
Fig. 1 — Polar Plotting and Integrating Equipment. General View

However, apart from the instrumental complications which may be involved in the use of reverberation rooms, computations based on free-air measurements have the advantage for research purposes that the total solid angle surrounding the loudspeaker or microphone may, if desired, be divided into sections and a separate integration carried out for each. By this means, the response integrated over the front hemisphere alone may be compared with the total for front and back together, a procedure which has been used[a] to obtain a figure of merit for microphones of the so-called 'unidirectional' type. Again, in the case of loudspeakers it might be possible, by weighting the results obtained in different zones, to give more prominence to the front response, and so arrive at an empirical figure, intermediate between the axial and mean spherical response, which would give a useful approximation to the overall effect obtained in an average listening room.

All these possibilities are, however, conditional on the use of some device to reduce the labour of computation to manageable proportions. The present monograph describes equipment developed to meet this requirement.

angular position of the instrument concerned; the Magslip system is used to rotate both a turntable carrying a sheet of polar graph paper and a pointer moving over a dial, giving a direct indication of angle. The polar response of the loudspeaker or microphone concerned is plotted by means of a pen moving along a radius of the turntable and operated by a servo-type level recorder through a suitable mechanical coupling. The rotating shaft is operated by a hand control through the medium of a Selsyn drive. Microphones under test are rotated in a sound field set up by a fixed loudspeaker; loudspeakers which are light enough to be supported on the shaft are rotated in front of a stationary measuring microphone while in other cases the loudspeaker is fixed and the microphone, mounted on the end of a boom attached to the rotating shaft, is moved in a circle about it.

Fig. 1 shows part of the equipment described above. The hand control for the rotating shaft and the dial indicating its angular position are contained in the unit on the left, which for convenience is referred to as the boom control. The plotting turntable and level recorder can be seen in the lower part of the figure.
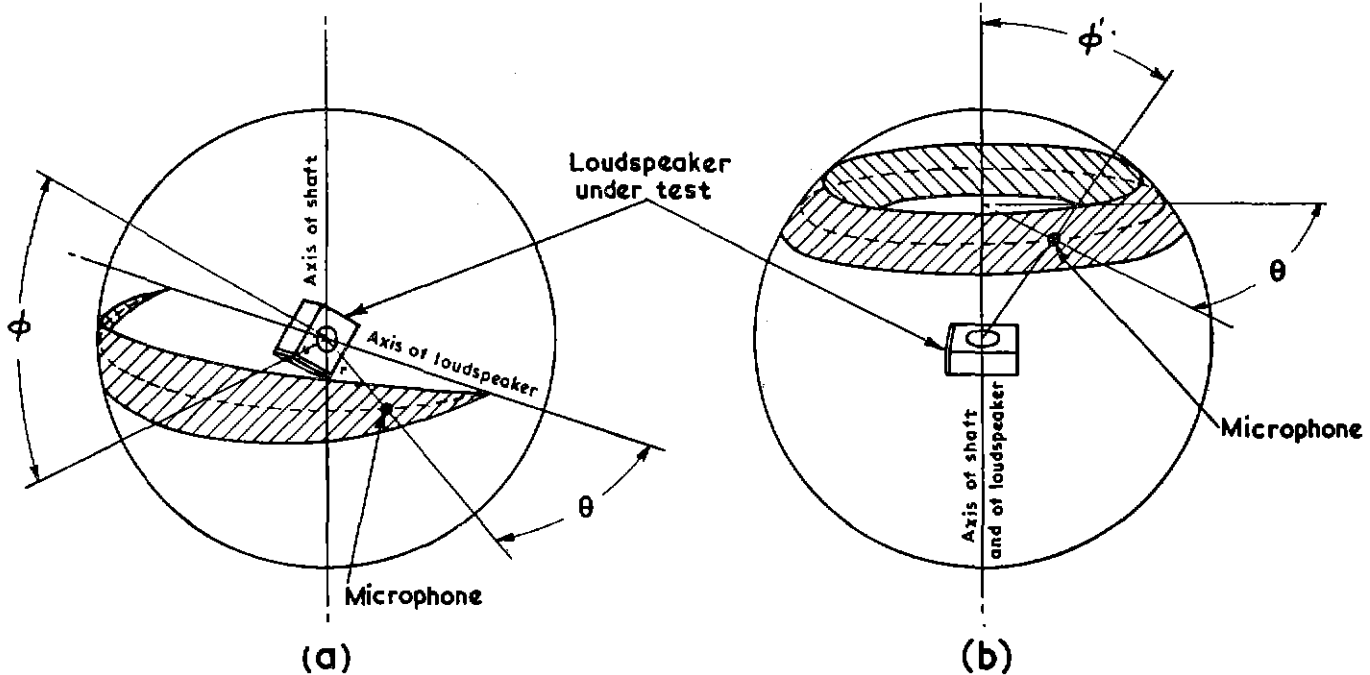

Fig. 2 — *Alternative Methods of Dividing Sphere of Integration*

2. General Principles

The apparatus to be described is designed to work in conjunction with the existing technical equipment of a lagged room used for measuring the polar response of loudspeakers and microphones. This equipment consists of a rotatable vertical shaft, on the end of which a microphone or a small loudspeaker can be mounted, together with a Magslip circuit for reproducing at a remote point the

Fig. 2 shows how the simple process of taking a polar curve can be extended to obtain the mean spherical response. To avoid needless repetition, the description of the operation throughout this monograph will refer only to the case of a loudspeaker which is rotated in front of a stationary microphone; similar considerations apply when the mean spherical response of the microphone is required, the positions of the two transducers being then inter-

changed. In Fig. 2(a) the loudspeaker is shown mounted on the end of the rotating shaft and tilted about its axis by some angle ϕ to the horizontal. If the shaft is rotated through 360° while the output of the microphone is recorded and the process is repeated with various values of ϕ, the sound radiated by the loudspeaker in all directions may be sampled. If the microphone output signal were proportional to the rate of acoustic energy flow per unit area, the total acoustic power radiated could be obtained by integrating this signal over any closed surface containing the loudspeaker. In practice, however, only the sound pressure or the particle velocity can be directly sampled. While it would be possible to use a pressure microphone and a velocity microphone placed close together, applying the amplified output signal to some kind of audio-frequency wattmeter circuit, such a complication is not generally considered to be justified. Instead, it is usual to employ a pressure microphone alone and to integrate over the surface of a sphere, treating the loudspeaker as a source of negligible size at the centre. The rate of energy flow per unit area of the surface of integration can then be assumed to be proportional to the square of the sound pressure registered by the microphone. The errors which may result from this assumption are discussed in the appendix.

For the purpose of the integration, the spherical surface may be divided into a number of parts, such as that shown shaded in Fig. 2(a), for each of which the integral to be evaluated will be proportional to:

$$\int_0^\pi e^2\sin\theta d\theta$$

where e is the r.m.s. output voltage of the microphone and θ the angle of rotation of the vertical shaft. This method of division is preferable where the loudspeaker has axial symmetry, so that a single integration of the form given

above will suffice, or where the degree of asymmetry is so slight that only a few values of ϕ need be taken.

Alternatively, the sphere of integration may be divided up into zones as in Fig. 2(b); for each zone, an integral proportional to:

$$\sin\phi' \int_0^{2\pi} e^2 d\theta$$

will then have to be evaluated. This method has the advantage that some form of weighting could, if desired, be applied to the individual zones depending on their distance from the loudspeaker axis.

From the foregoing it will be seen that the function of the equipment is to evaluate integrals of the form:

$$\int e^2 d\theta \text{ or } \int e^2\sin\theta d\theta$$

where e is a function of θ. However, $d\theta$ can be replaced by $\frac{d\theta}{dt}.dt$; if the loudspeaker is rotated at constant angular velocity, i.e. if $\frac{d\theta}{dt}$ is constant, the integration with respect to angle can be replaced by an integration with respect to time. It is then possible to make use of an integrating wattmeter, applying to one set of coils a signal proportional to e and to the other a signal proportional to e or $e\sin\theta$, the sine law being obtained by an appropriate form of potentiometer operated by the rotating shaft. The only readily available power-integrating meters are those designed for use at 50 c/s and to employ one of these it is necessary to provide a 50 c/s source modulated in accordance with the amplitude of the signal e. Moreover, certain modifications to such meters are required to enable them to deal with wide variations in the input to the voltage coils, a condition
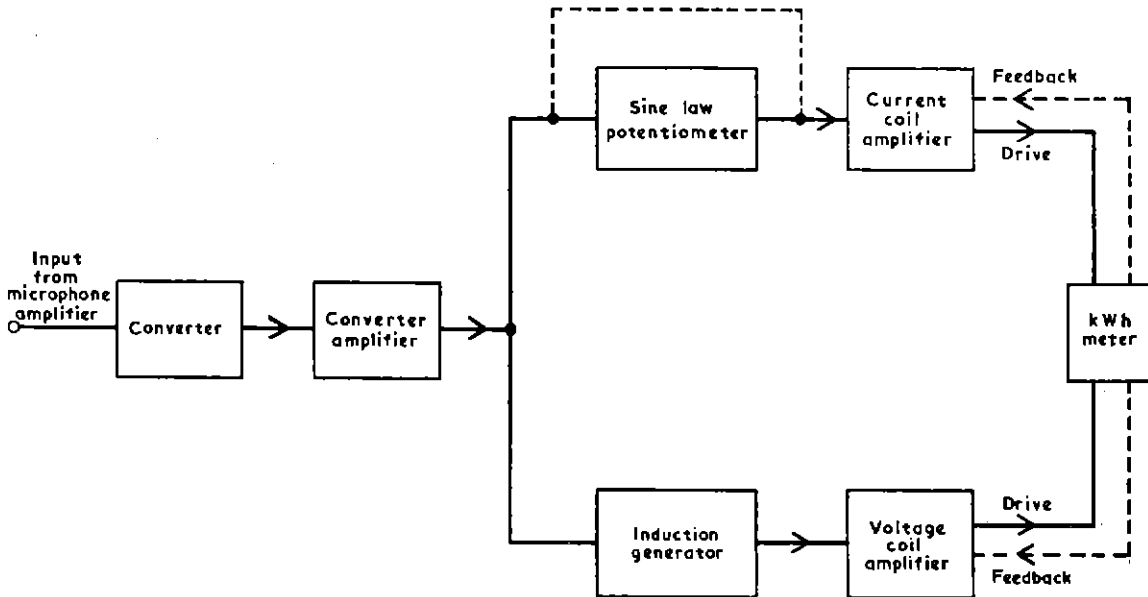

Fig. 3 — Simplified Block Schematic Diagram

8

for which they are not intended. However, the slight instrumental complications thus entailed are still considered worth while for it would be hard to find a cheaper and more readily available piece of equipment for integrating the product of two electrical quantities than the common commercial kilowatt-hour meter.

So far, constant angular velocity of the loudspeaker relative to the microphone has been postulated. It is often inconvenient in practice to preserve this condition, and it was for this reason that the existing equipment had been designed for hand operation rather than with a motor drive. For the purpose of the integration however the desired result can be obtained by introducing a factor proportional to $\dfrac{d\theta}{dt}$; this end is achieved by passing the 50 c/s signal intended for one set of meter coils through an induction generator—a device similar to a Magslip—which is driven from the vertical shaft and delivers a voltage proportional both to the input and to the angular velocity.

Finally, means have to be provided for producing a 50 c/s signal proportional to the output voltage from the microphone. This function is performed with sufficient accuracy for the present purpose by a linear rectifier, the smoothed output from which is interrupted mechanically at 50 c/s and filtered to remove harmonics. The use of the rectified mean instead of the r.m.s. value of the fluctuating signal is permissible here since the form-factor of the incoming signal, which may consist of single frequencies or bands of noise, is constant throughout the measurement.

Fig. 3 shows a block schematic of the complete integrating circuit working on the principles outlined above. The unit marked 'converter' contains the linear rectifier which converts to d.c. the audio-frequency output from the microphone amplifier, together with the interrupter and smoothing circuits which convert this d.c. to a 50 c/s signal. It should be noted that the sine-law potentiometer is only included in the circuit when integrating by the method of Fig. 2(a). Where the method of Fig. 2(b) is employed, the integrals for the individual zones have to be weighted in some way by the appropriate values of sin ϕ' before being added together.

The design of the individual units shown in Fig. 3 will be described in the following sections.

3 Design Requirements

3.1 Linearity

Ideally, the system should be linear, i.e. the speed of revolution of the kilowatt-hour meter disk should be proportional to the product of the signals applied to the voltage coil and current coil amplifiers. The error resulting from any non-linearity of the system will depend upon the range over which these signals vary; it will be greatest when a large proportion of the total registered by the integrating meter is recorded at low speeds at which the friction compensation is only partially effective. Some idea of the order of error to be expected in extreme cases of this kind may be gained by considering a hypothetical axially sym-

metrical loudspeaker having a polar diagram of the form shown in Fig. 4.

Let the response be taken as unity over the angle 2Ψ and α at all other angles. Imagine, for example, that the integrating system is linear from the maximum meter speed to a level 20 dB below it, that for levels between 20 dB and 30 dB below the maximum, the integrating meter runs at half the proper speed, corresponding to an error of 3 dB, and for still lower levels fails completely to register. The error arising in these circumstances is tabulated below. It is assumed that the sphere of integration is divided as shown in Fig. 2(a), that the loudspeaker is rotated with constant angular velocity, and that the maximum permissible meter speed is attained but not exceeded.
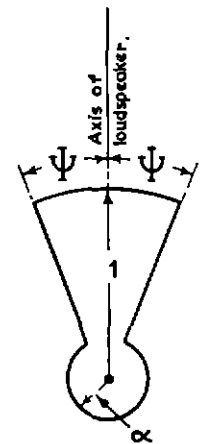

Fig. 4

Ψ	$20 \log_{10}\alpha$(dB)	Error (dB)
5°	−30	0·39
5°	−20	<0·01
10°	−30	0·26
10°	−20	<0·01
15°	−30	0·12
15°	−20	<0·01

It will be seen that quite serious departures from linearity can be tolerated at levels more than 20 dB below the maximum, provided that the registration is accurate at higher levels. In practice, the overall amplitude characteristic of the system is linear within ± 1 dB for at least the upper 20 dB of its range and it is estimated that the error in the integral will seldom exceed $\frac{1}{2}$ dB.

3.2 Speed of Operation

The time required to carry out a given series of measurements is governed by two principal factors.

(a) Inertia of rotating system

In view of the high moment of inertia of the rotating shaft when loaded with a loudspeaker or with a microphone mounted on a long boom, the hand drive has been designed with a 36 : 1 reduction gear and the minimum time required in practice for one revolution of the shaft is about twenty seconds.

(b) Reading accuracy of integrating meter

To achieve a significant result the rotor of the kilowatt-hour meter must make a certain number of revolutions to allow a sufficiently high reading accuracy. The smallest division on a standard 250V meter of 10A rating is not usually less than 10 Wh, and if the reading accuracy is required to be 2 per cent, this means that a total of 0·5 kilowatt-hour must be registered. If 100 per cent overload of the current coils* is allowed, with full voltage on the voltage coils, this would take six minutes and since con-

---

* To conform to B.S.I. Specification No. 37 : 1937, kilowatt-hour meters must be capable of carrying continuously, and registering accurately, 100 per cent overload.

Indicators

Voltage coil.

Feedback coils.

Current coils

Fig. 5 — Modified Kilowatt-hour Meter

tinuous full speed operation is not achieved in practice, a considerably longer time would be required to obtain the necessary precision. By modifying the meter in the manner described in the next section, the reading accuracy has been so far increased that it no longer limits the speed of measurement.

4. Design Details
4.1 *Integrating Meter*
4.1.1. *General*
The integrating meter shown in Fig. 5 is a commercial kilowatt-hour meter to which certain modifications have been made to adapt it for the purposes of this equipment.

To achieve the required reading accuracy, the original gear train and dial system were rebuilt. A pointer was fitted to the spindle of the gear driven directly from the worm on the rotor; this pointer moves over a dial large enough for individual rotor revolutions to be resolved. A second pointer was arranged to indicate 500 revolutions of the rotor before repeating and finally, a dial reading hundredths of a revolution was fitted to the rotor shaft. These dials and pointers may be seen in Fig. 5.

In normal practice such meters do not have to deal with large variations in voltage and in their design advantage is taken of this fact in two ways: (*a*) the torque required for friction compensation is derived from the flux pro-

duced by the voltage coil, a quantity which can with sufficient accuracy for the purpose be taken as constant. (b) economies are effected in the construction of the magnetic circuit by allowing a certain degree of non-linearity.

For the present application, wide voltage variations are to be expected and special measures are necessary to secure the required degree of overall linearity.

To obtain a constant torque for friction compensation it was decided to use a 400 V three-phase four-wire meter, which contains three separate driving elements; the first of these is used to provide a small but constant torque to offset the effect of friction, while the second and third are available for driving the rotor. The voltage coil of the first element is connected to the mains through a variable resistor which provides an adjustment for the friction compensation; the corresponding current coils are not used.

The driving torque acting on the rotor disk is proportional to the product of the magnitudes of the fluxes produced at the disk by the driving magnets and the sine of the phase angle between them. The inevitable non-linearity of the magnetic circuit therefore causes non-linearity of the driving torque with respect to driving voltage. This difficulty was overcome by fitting small flux-sampling coils to the pole tips of the driving magnets, as close to the rotor disk as possible; these coils provide voltages very nearly proportional to the fluxes which produce the driving torque, and by using the voltages as feedback signals for the appropriate amplifiers, the driving torque can be made proportional to the corresponding input voltages. The feedback coils can be seen in Fig. 5.

4.1.2 Working Range

The working range of the integrating meter, i.e. the ratio of the maximum to the minimum electrical input at which accurate registration takes place, will now be considered.

The maximum speed of the rotor can be increased in two ways; by driving the coils beyond their normal working rating, and by reducing the flux of the braking magnets. The extent to which the former expedient can be used is limited on the one hand by the maximum safe dissipation of the coils, and on the other by self-braking action.* The latter effect becomes apparent when the flux producing the driving torque is comparable with the flux of the braking magnet; the total braking flux is then increased, and linearity suffers. It was found that some increase in input was possible without encountering either of these limitations, and full advantage was taken of this fact to extend the working range.

It might at first appear that the range of the meter could be further increased by weakening the flux of the braking magnet, thus allowing a higher rotor speed for a given input to the coils. In fact, the opposite is the case, for if the control exercised by the magnet were reduced, the effects of friction at low speeds would be more apparent while the effects of self-braking at high speeds would be unchanged.

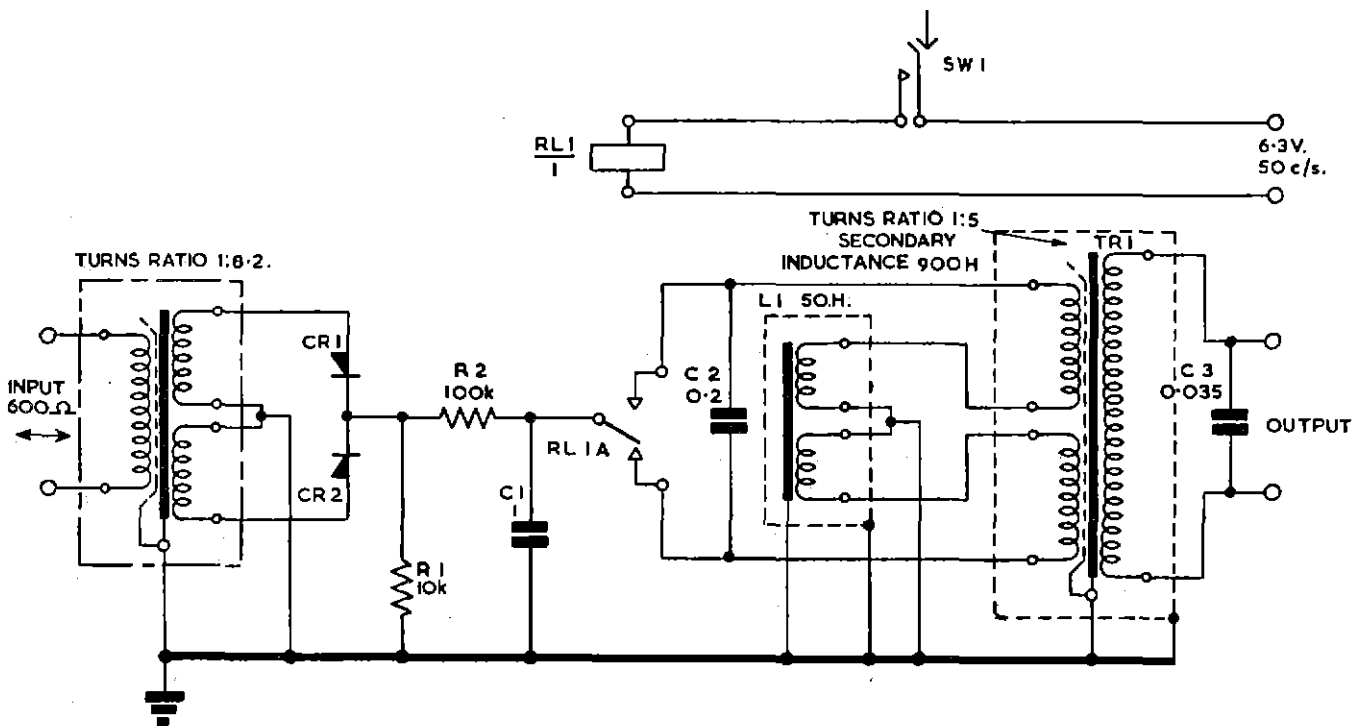* S. James. 'Electricity Meters and Instrument Transformers'. Chapman and Hall Ltd, 1952, p. 112 et seq.

Fig. 6 — Converter Unit. Simplified Circuit

11

The working range could be extended by increasing both the braking flux and the driving power, but for the limitation already imposed by the safe heat dissipation of the coils.

In their original form the current coils required more than 20 A for full energization. To obviate the necessity for a current transformer mounted close to the meter, or for cumbersome heavy wiring from the amplifier, the coils were rewound to an impedance of about 300 ohms, the current required being then only 130 mA.

The maximum working speed of the meter disk is about one revolution per second. The dial readings are of course in arbitrary units and the scale factor required to give the mean spherical response is obtained by rotating the shaft supporting the loudspeaker through 360° while applying a known and constant input to the converter unit, thus simulating the effect of an omnidirectional radiator.

4.2   Converter

Fig. 6 shows the circuit of the converter unit. The input signal is rectified by the germanium diodes CR1 and CR2 and smoothed by the filter R2, C1.

The frequency response of the input circuit is within $\pm\frac{1}{2}$ dB between 30 c/s and 20 kc/s.

The conversion of the d.c. signal to a.c. is effected by the polarized relay RL1 which is driven from a tapping on the mains transformer of the apparatus and which applies the voltage developed across C1 alternately to each half of a transformer TR1. To avoid unnecessary wear on the contacts, the relay coils are fed through a separate switch SW1 so that they need only be energized for the duration of each test.

The series inductor L1 together with the shunt inductance of transformer TR1 and the capacitors C2 and C3 constitute a tuned circuit designed to discriminate against harmonics of the 50 c/s signal while minimizing the variation in fundamental output with change of mains frequency. The frequency response of this circuit, shown in Fig. 7, was measured by applying a constant direct voltage across R1 and energizing the relay coils from an oscillator of variable frequency.

The maximum input to the converter is about 8 V and this gives a 50 c/s output of about 5 V. The input impedance of the converter is approximately 600 ohms.

4.3   Induction Generator

The induction generator is geared directly to the driving handle on the boom control, and makes 360 revolutions for one revolution of the loudspeaker under test. The output from the generator during a measurement varies between about 0·5 V and 10 V. The only significant deviation from strict proportionality between output, input and speed takes the form of a small voltage appearing at the output when the rotor is stationary. The phase of this spurious signal varies with the angular position of the shaft and cannot therefore be easily compensated. The amplitude is however less than 50 mV for maximum input;

the resulting error is negligible unless the rotation of the loudspeaker is interrupted for a long period in the middle of a test, and even so can be avoided by temporarily switching off the converter relay.
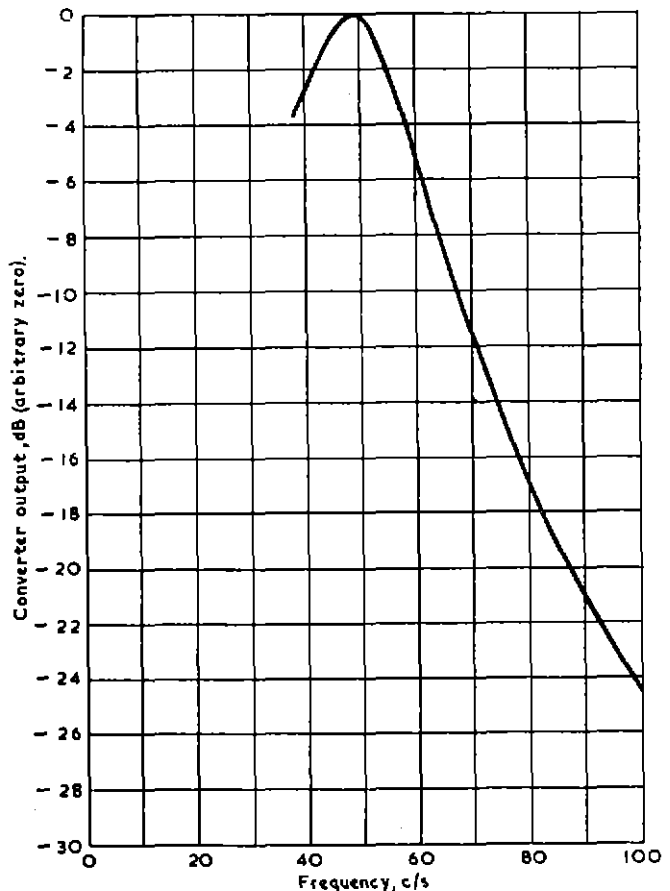

Fig. 7 — Converter Unit. Variation of output with Mains Frequency

4.4   Sine-Law Potentiometer

The function of the sine-law potentiometer is to weight the signal transmitted to the current coils of the integrating meter by a factor proportional to $\sin\theta$ while θ varies between 0 and π (see Fig. 2(a).) Much time can be saved if the two diametrically opposite sections of the sphere can be dealt with in succession by making a complete revolution of the shaft supporting the loudspeaker, i.e. by performing

the integration $\int_0^\pi e^2\sin\theta d\theta$ for $\phi=\phi_1$, and then for

$$\phi=\phi_1 +\pi.$$

To avoid having the two integrals of opposite sign, the potentiometer is so constructed that the sine-law weighting remains positive for values of θ between π and 2π.

The wiper of the potentiometer is attached directly to the vertical shaft which supports the loudspeaker under test. It consists of a horizontal insulating plate, to the face
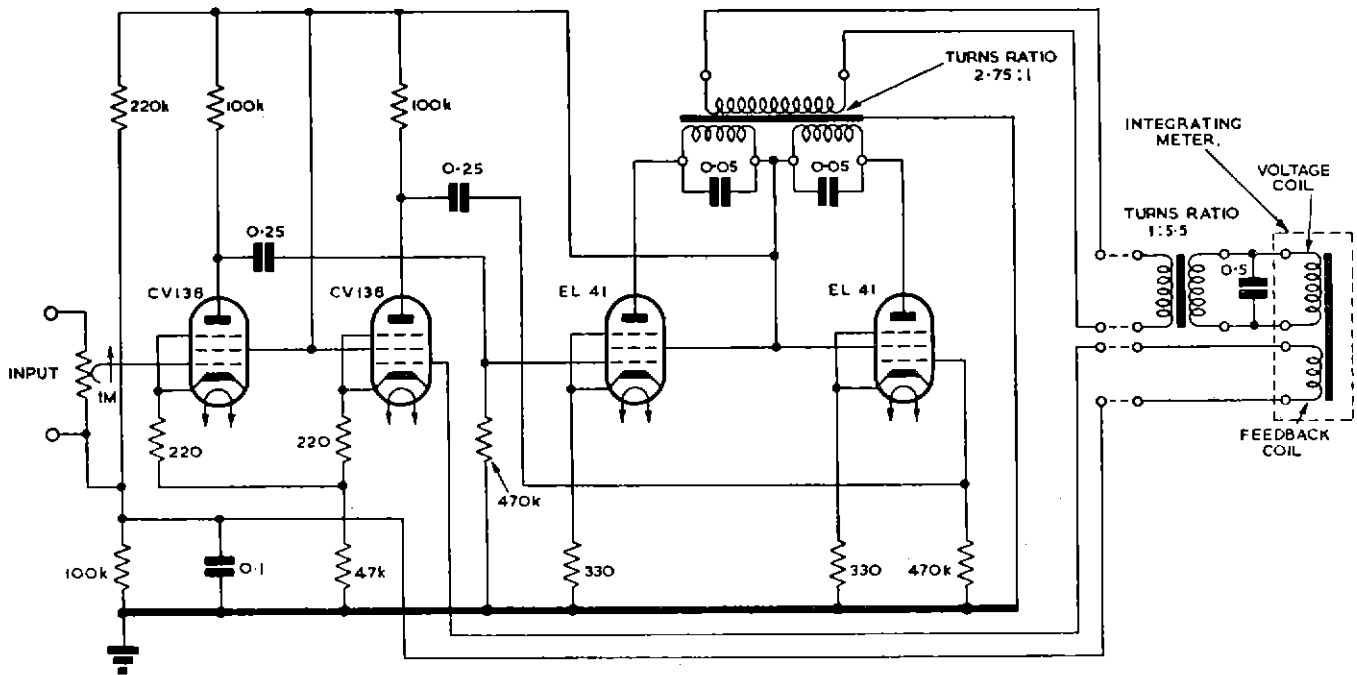
Fig. 8 — Voltage Coil Amplifier and Meter Connections. Simplified Circuit

of which is attached a curved wire making contact with a fixed, linear resistance card (extracted from a wirewound potentiometer) mounted along a radius from the shaft. The incoming voltage is applied across the whole of the resistance and the contour of the contact wire is such that the fraction of this voltage tapped off is equal to |sin θ|.

The maximum input to the sine-law potentiometer is 50 V. The maximum output depends upon the extent to which the response of the loudspeaker-microphone combination falls as the value of sin θ rises. Before making a measurement, the gain of the amplifier feeding the sine-law potentiometer is so adjusted that the output from the potentiometer never exceeds 5 V.

4.5 Turntable Unit

The turntable unit shown in Fig. 1 is not an essential part of the integrating equipment, although it may be desirable to integrate and record polar diagrams simultaneously. The unit can however be used to enable the integrator to operate from data supplied by existing polar curves. To this end, the input terminals of the level recorder are supplied with a 50 c/s signal controlled by a hand-operated potentiometer, thus allowing the pen on the turntable to be set to any point along its line of travel; the same signal voltage is applied to the input of the integrating equipment. By rotating the boom control and following with a dry pen the line of an existing polar diagram placed on the turntable, the variations in response with angle can be reproduced and the integration carried out as if the loudspeaker concerned were actually on test. In the present instance, it was not convenient to dispose the equipment so that an unaided experimenter could rotate the boom while following the polar diagram with the hand-

controlled pen; on the few occasions that this operation is carried out, the boom control would be turned by an assistant.

4.6 Amplifiers
4.6.1 General

Fig. 8 shows in simplified form the circuit of the voltage coil amplifier and its connections to the integrating meter. The remaining two amplifiers, which supply respectively the current coils of the meter and the induction generator, all employ the same basic circuit.

Each amplifier contains four valves: a pair of CV 138s in a long-tailed-pair circuit, acting as phase splitter, and a pair of EL 41s in the output stage. Negative feedback is used to improve linearity and to stabilize gain; the feedback signal is applied to the grid of one of the phase-splitting valves, while the input signal is applied to the grid of the other.

The overload point of the amplifiers feeding the integrating meter is adjusted in each case to protect the voltage and current coils against accidental over-running. To ensure that the overload levels, once set, are accurately maintained, a voltage-stabilized H.T. supply is provided.

4.6.2 Converter amplifier

This amplifier raises the level of the signal from the converter from approximately 5 V to a maximum of 50 V to feed the induction generator and sine-law potentiometer. The induction generator, which presents a partly reactive impedance, draws a maximum current of 0·17 A, corresponding to a power of about 6 W; the current drawn by the sine-law potentiometer is negligible by comparison.

13

The amplifier circuit needs no special comment. Negative voltage feedback is derived from the secondary winding of the output transformer, the loop gain being approximately 20 dB.

4.6.3 *Voltage coil amplifier*

The maximum input to the voltage coils is about 400 V and 60 mA, i.e. 24 VA; the impedance of these coils is largely reactive and the maximum dissipation is only about 4 W. To economize in amplifier power, the circuit was therefore tuned.

It was at first intended to tune the coils with a series capacitor mounted in the meter case and so to avoid applying 400 V to the cords, jackfields, and tie-lines connecting the various parts of the equipment. The large phase shifts associated with the series tuned circuit, however, made it difficult to achieve stability in the feedback system. Eventually the coils were parallel tuned and fed by an external step-up transformer as shown in Fig. 8.

As already stated, the magnetic circuits associated with the voltage coils are non-linear and when a sinusoidal voltage is applied, the waveform of the gap flux contains harmonics. To produce a pure flux waveform, a voltage having an appropriately pre-distorted waveform would have to be applied to the coils. This condition would in ordinary circumstances be brought about automatically by the use of feedback but cannot be achieved in the present instance because the coils, being tuned, must necessarily receive a substantially sinusoidal input. However, the presence of harmonics in the voltage coil flux does not influence the torque on the rotor of the meter as long as there are no corresponding components in the flux produced by the current coils.

The gain in the feedback loop is approximately 20 dB and the maximum input to the amplifier about 5 V.

4.6.4 *Current coil amplifier*

This amplifier must deliver up to 40 V and 130 mA to the rewound current coils. The associated magnetic circuit has a large gap so that non-linear effects are much smaller than in the case of the voltage coils. Moreover, with the large gap, the inductance of the circuit is low enough to make tuning unnecessary, so that the feedback is fully effective in reducing distortion of the flux waveform.

As in the case of the voltage coil amplifier, the loop gain in the feedback chain is about 20 dB and the maximum input, 5 V.

4.6.5 *Phase relationship*

To obtain the maximum torque on the rotor of the integrating meter, the fluxes produced by the two sets of driving coils, and hence the input signals to the amplifiers which feed them, must be brought into quadrature. Apart from the small phase shift introduced by the induction generator (less than 12°), the two inputs would normally be in phase, and to achieve the required quadrature relationship, simple R–C networks are used to advance the phase of one signal by about 45° and to retard the other by the same amount. By making the time-constants of the

two circuits equal, the effects of any small change in mains frequency are almost entirely self-compensating; the phase difference between the two signals remains 90° while the product of the amplitudes is constant within $\pm0\cdot6$ per cent for a mains frequency change of ±10 per cent.

4.7 *Monitoring facilities*

The various pieces of equipment shown in the block schematic diagram of Fig. 3 are connected together through the distribution unit which appears in Fig. 1 between the boom control and the integrating meter. Two a.c. voltmeters mounted on the front panel of this unit show the inputs to the voltage and current coils of the integrating meter. These meters must be watched by the operator, so that he can ensure that the maximum working levels are not exceeded. The input to the current coils is determined only by the circuit gain and by the polar characteristics of the loudspeaker. The input to the voltage coils, however, is proportional not only to the response of the loudspeaker but to the rate at which it is being rotated. Advantage can be taken of this effect by rotating the shaft supporting the loudspeaker faster when the response is seen to fall; wide variations in the input to the voltage coils can thus be avoided and errors due to non-linearity reduced.

5. Conclusion

The apparatus described considerably expedites the computation of the mean spherical response of a loudspeaker or microphone. The time taken for the integration process is little more than that required to record, on automatic polar plotting devices, the information on which such a computation could be based.

It is hoped that the data obtained with this equipment will facilitate the objective assessment of the performance of loudspeakers and microphones under practical conditions of use.

6. References

1. E. W. Kellogg. **Loudspeaker Sound Pressure Measurements,** J.A.S.A., Vol. II, No. 2, October 1930, pp. 157–200.

2. W. West and D. McMillan. **The Design of a Loudspeaker,** J.I.E.E., Wireless Section, Vol. 15, No. 44, June 1940, pp. 53–73.

3. Harz and Kösters. **Ein neuer Gesichtspunkt für die Entwicklung von Lautsprechern?** Technische Hausmitteilungen des Nordwestdeutschen Rundfunks, Vol. 3, No. 12, December 1951, pp. 205–8.

4. H. C. Hardy, H. I. Hall, and L. G. Ramer. **Direct Measurement of the Efficiency of Loudspeakers by use of a Reverberation Room,** Transactions of the IRE–PGA, 10, November–December 1952, pp. 14–24.

5. British Standard 2498 : 1954. **Recommendations for Ascertaining and Expressing the Performance of Loudspeakers by Objective Measurements.**

6. R. P. Glover. **A Review of Cardioid Type Unidirectional Microphones,** J.A.S.A., Vol. XI, No. 3, January 1940, pp. 296–302.

APPENDIX

Error in measurement of mean spherical response of a loudspeaker due to eccentricity of equivalent sound source

It has already been pointed out in Section 2 that the computation of the mean spherical response of a loudspeaker from pressure measurements made in free space rests on the assumption that the sound may be imagined to emanate from a point situated at the centre of the sphere of integration. At a finite distance from the loudspeaker however, the wave-front may not be spherical, so that no single point can be taken to represent the position of the source. Alternatively, even if the latter requirement is approximately met, the exact position to be assigned to the point may be in doubt; the following analysis serves to indicate the order of error which may arise in the latter case.

Let it be assumed for simplicity that the loudspeaker under test produces a spherical wave-front centred on a point S, which may be regarded as the position of the equivalent source, and that the pressure produced at a given distance from S is constant at angles up to Ψ from the loudspeaker axis but zero in all other directions. Consider now the effect of integrating over the surface of a sphere whose centre O is displaced from S, as shown in Fig. 9, by a distance σ along the axis of the loudspeaker.
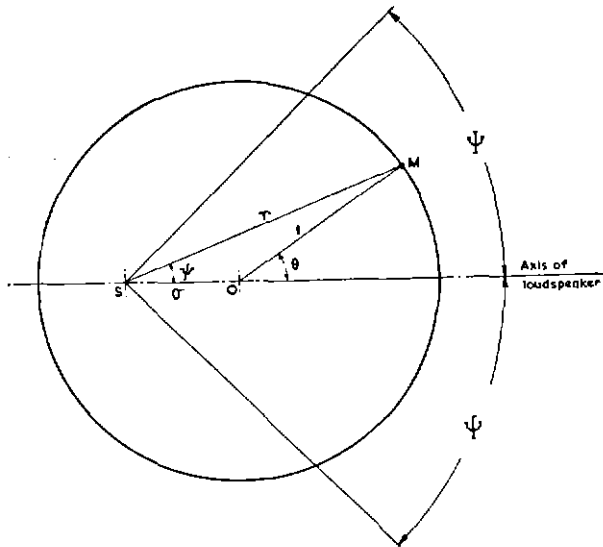

Fig. 9

Let the radius of the sphere of integration be taken as unity and let r be the distance of the microphone M from the source S. It will be seen that the sound reaching the microphone has left the source at an angle ψ to the loudspeaker axis but for the purpose of the integration is treated as if it had emanated from O at an angle θ to the axis.

Since the pressure registered by the microphone M is inversely proportional to the distance r and

$$r = \sqrt{1+\sigma^2+2\sigma\cos\theta}$$

it is clear that instead of the required integral $\int_0^{\Psi} e^2\sin\theta d\theta$,

where e is the microphone output voltage when $\sigma = o$, an erroneous result

$$\int_0^{\Theta} \frac{e^2\sin\theta}{1+\sigma^2+2\sigma\cos\theta}\cdot d\theta$$

will be obtained, where Θ is the value of θ when $\psi = \Psi$. Thus, the ratio

$$\frac{\text{Measured mean spherical response}}{\text{True mean spherical response}}$$

$$= \sqrt{\left(\int_0^{\Theta} \frac{e^2\sin\theta}{1+\sigma^2+2\sigma\cos\theta}\cdot d\theta \middle/ \int_0^{\Psi} e^2\sin\theta.d\theta\right)}$$

$$= \sqrt{\left(\frac{log_e\,(1+\sigma^2+2\sigma)/(1+\sigma^2+2\sigma\cos\Theta)}{2\sigma(1-\cos\Psi)}\right)}$$

It can readily be shown that

$$\cos\Theta = -\sigma\sin^2\Psi+\cos\Psi\sqrt{1-\sigma^2\sin^2\Psi}$$
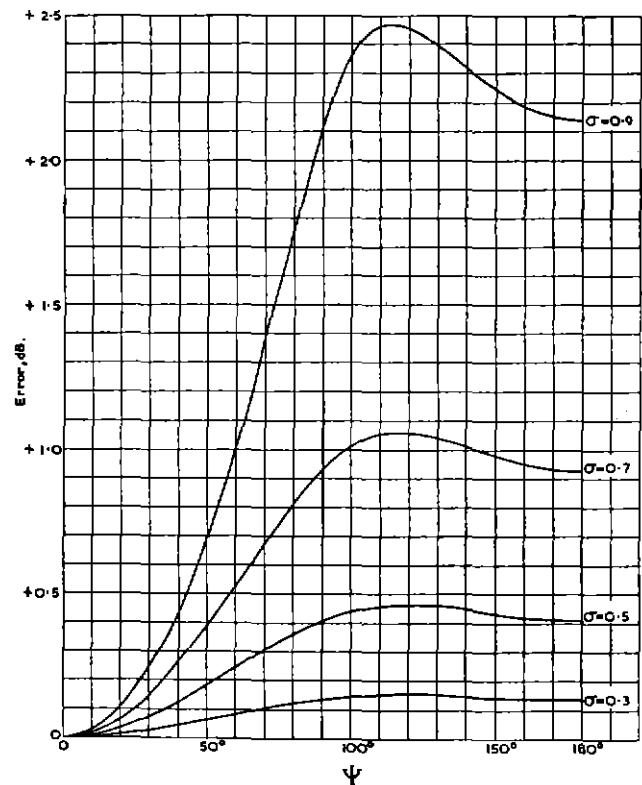
From the above, the curves in Fig. 10, giving the error

Fig. 10 — Calculated Error due to Displacement of Source from Centre of Sphere of Integration

15

in mean spherical response as a function of Ψ for various values of σ, have been calculated. It should be noted that as the radius of the sphere of integration has been taken as unity, σ is the ratio of the eccentricity of the source to the radius of the circle described by the microphone.

It will be seen from the curves that the errors, which are positive and substantially independent of the direction of

the displacement σ, are small when Ψ is small and are greatest when Ψ is about 120°.

Since in practice the response of a loudspeaker is usually measured at a distance greater than twice the depth of the cabinet, the value of σ is not likely to exceed 0·5 and the error due to eccentricity of the source should therefore be less than 0·5 dB.

A METHOD* OF RECORDING TELEVISION OR OTHER SIGNALS ON MAGNETIC TAPE

by

R. D. A. MAURICE, Ing.-Dr., Ing.E.S.E., A.M.I.E.E.
Research Department, BBC Engineering Division

Experience from sound recording proves that magnetic-dust tape of suitable characteristics can record a magnetic pattern with an effective uni-dimensional aperture of less than 0·0025 inches along the line of the recording pattern. In what follows an aperture dimension of 0·0025 inches will be taken by way of example without prejudice to any other suitable figure. A television picture made up of 405 horizontal lines with an aspect ratio of 4/3, width to height, could therefore be accommodated in about 1·4 square inches of magnetic tape. The reason why magnetic-tape recording of television signals has so far proved so difficult is because of the poor space utilization of the tape area and the loss of sensitivity which would result with

typical present-day magnetic tape owing to a reduction in aperture area of, for example, 50 to 1 due to using a square aperture instead of a rectangular one (say, 0·125 inches by 0·0025 inches) as at present used for sound recording. To record television signals along a single straight line on a piece of magnetic tape requires, from the above figure for effective aperture, a tape velocity of 13 000 inches/s. Such a velocity would introduce formidable problems if it were essential that the tape moved at this speed. It is, however, possible to 'lay' the television image on to the tape according to the same or a similar pattern as for normal television scanning, that is with a number of horizontal lines arranged one above the other in a stack of, say, 405 interlaced in two groups of 202½, or 405 in two sequential groups of 202½. By this means, and notwithstanding the requirement for a tape capable of retaining a maximum flux density, say, 50 times that used at present, the use of the whole area of the tape is achieved with a great saving in tape and more especially with a tape-speed reduction of 520 to 1. The tape may be run, therefore, at 25 inches/s like cinematograph film.

In order to achieve line scanning across the tape and therefore at right angles (or nearly) to the longitudinal axis and the direction of motion of the tape, it is necessary to arrange a lateral motion at line-scan frequency for the signal-modulated magnetic field.

A means of accomplishing this is described, and it relies upon the principle of conducting the magnetic field to the required portion of the tape through a relatively low-reluctance path. This low-reluctance path may consist of the two pole pieces 2′ of an electro-magnet 6, shown in Fig. 1a, the magnetic tape 1 upon which the magnetic image is to be recorded and two ferrite filled slots 2″ in two rotating copper disks 4, thus forming a magnetic Nipkow disk in which the state of magnetization of the path is controlled by the television signal whilst the position of the path is controlled by the scanning waveform to which the angular velocity of the disk is synchronized in modulus and phase. Locating pins 8 permit of accurate location of disks 4 one with respect to the other. In a specific example, drive motor 9 causes disks 4 to rotate at 6 000 rev./min. Disks 4 have a diameter of about 40 inches and each has 100 equi-spaced slots about 0·002 inches wide in the circumferential direction and filled with high-permeability material. Locating pins 8 are so disposed that the two sets of 100 slots 2″ are in superposed pairs. The gaps between pole pieces 2′ and slots 2″ should be as small as possible, say 0·001 inches. The gap between the two disks 4 should not exceed the tape thickness by more than, say, 0·003 inches, and should preferably be less. The width of the pole pieces 2′ would equal the diameter of the holes 2″, namely 0·002 inches. The thickness of the disks 4 close to the periphery where they rotate between the pole pieces 2′ should not be less than about $\frac{1}{16}$ inches each. The width of the tape 1 could be 1·3 inches and its velocity in the direction of a radius of the disk would be 25 inches/s.

It will be noted that longitudinal recording through the tape rather than lateral recording across its surface is here contemplated. It would appear to be more difficult to achieve lateral recording by this method although possible variants easily come to mind.
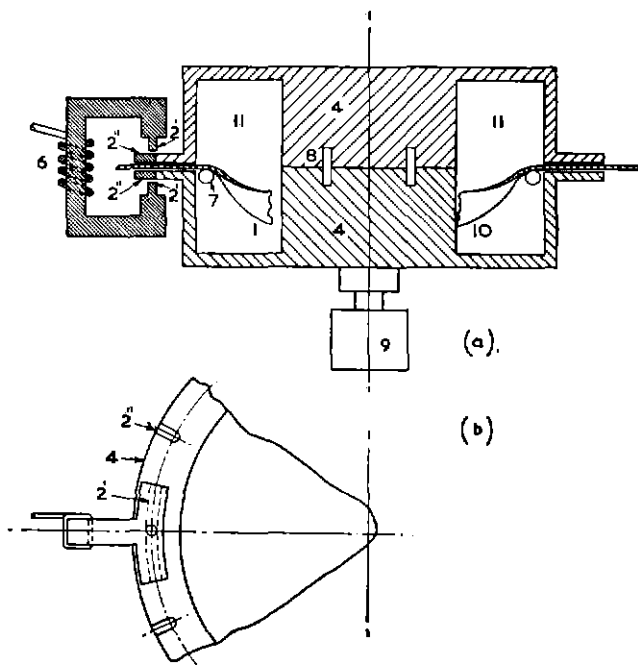

Fig. 1

* Patent applied for.