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Contents

Conventional & On-line Methods of Sound Power Measurements

by

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ABSTRACT
The article contains description of sound power measurement techniques in free-field, diffuse-field and semi-reverberant field environments. Experimental results are included to illustrate accuracies achievable by the three methods using the same wideband sound source. Finally, an automatic method of sound power measurement is included for which the hardware and software are described. Advantages of this system over conventional instrumentation are also discussed.

SOMMAIRE
Cet article contient la description de méthodes de mesure de puissance acoustique en champ libre, champ diffus et dans un milieu semi-réverbérant. Des résultats expérimentaux illustrent la précision obtenue avec les trois méthodes utilisant la même source sonore à large bande. Enfin, l'appareillage et le processus mis en oeuvre dans une méthode automatique de mesure de puissance sonore ainsi que les avantages que présentent ce système sur les ensembles conventionnels, sont étudiés.

ZUSAMMENFASSUNG
Der Aufsatz enthält eine Beschreibung der Schalleistungs-Meßtechnik im freien Schallfeld, im diffusen Feld und in halb-halligen Umgebungen. Die erzielbare Genauigkeit dieser drei Methoden unter Benutzung derselben Breitband-Schallquelle wird durch experimentelle Meßergebnisse illustriert. Weiterhin wird eine automatische Meßmethode für die Schalleistung angegeben, und die benötigte Hardware und Software werden beschrieben. Schließlich werden die Vorteile dieses Systems gegenüber konventionellen Meßausrüstungen diskutiert.

Introduction
With the continuous increase of noise in everyday life contributed by higher speeds in transportation, prime movers and all forms of machinery, the public has inevitably become more and more noise-conscious. As a result "noiselessness" as a quality in itself, has become a prerequisite in the production and sale of modern day utility items such as washing machines, vacuum cleaners, refrigerators, air conditioning systems etc. Management in industry has therefore not only recognized the importance of making quieter pro-

ducts but also laid emphasis on providing better employee working conditions for the improvement of overall efficiency of human beings. Noise control as can be seen plays an important role from an economic as well as from an annoyance and health point of view.

In order that noise may be controlled effectively, it is primarily necessary to be able to measure the noise emitted by machines objectively according to internationally accepted procedures. (ISO Recommendation R 495). The results can then be assessed in the light of predetermined criteria for acceptance.

By determining the radiated sound power and its directivity, the four aims for measurement of physical characteristics of noise of machines (as outlined in the ISO Recommendation) can be achieved. They are:

a) Verification that the noise of a given machine conforms to a certain standard.

b) To make a comparison between the noise emitted by machines built to the same specification.

c) To make a comparison between the noise emitted by different machines.

d) The determination of noise received at a distance.

The noise produced by a machine (usually a byproduct of its primary function) is dependent upon factors such as its sound radiation characteristics and the environment in which it is placed. On the other hand the sound power emitted by the machine, which is little affected by the acoustic environment, may depend upon the installation. In a measurement report, it is therefore normally necessary to specify the operational characteristics of the machine, and to give a description of its installation.

Since equipment for measurement of acoustic intensity is generally not available, sound power inevitably has to be determined indirectly via sound pressure measurements, in the frequency range of interest. It is therefore of prime importance, that the environment in which the measurements are made, are such that the sound pressure squared is directly proportional to the sound intensity or sound energy density. This can be achieved respectively in anechoic chambers assimilating out-door free-field* conditions and in reverberant rooms providing diffuse-field** conditions.

(In all cases the indicating meter of the measuring apparatus should read the true rms sound pressure at the microphone location). Unfortunately such ideal conditions are not always available to the practicing engineer. As more often than not, machinery installed in factories cannot be moved around, measurements have to be carried out in what is termed a Semi-reverberant field. In such spaces the acoustic field is a resultant of free-field and reverberant field conditions. Sometimes, however, measurements are difficult to perform in such environments on account of reflecting surfaces in the neighbourhood of the sound source, or high background noise levels. These circumstances may necessitate sound pressure level measurements to be made in the near-field and when appropriate they allow determination of sound pressure level at a reference radius. While the main text of this article is concerned with sound power measurements, near-field measurements are described in Appendix B. Sound power measurements, as will be shown, require considerable amount of time using conventional instrumentation. An automatic method was therefore developed to speed up the procedure, making use of Real-Time Analysis and on-line digital computer processing.

Free-field measurements

The determination of sound power in a free-field is based on the assumption that a sound source will radiate free progressive spherical waves. The acoustic power can then be found conceptually by adding the products of the areas times the acoustic intensities for the areas on a hypothetical sphere around the source.

Since in a free-field the sound pressure squared, a certain distance away from the source is directly proportional to the acoustic intensity, the sound power can be determined practically by taking the spatial average of the sound pressure squared measured on equal areas on the sphere. However, the radius of the hypothetical sphere must be large enough so that far-field conditions are ensured i.e. the particle velocity is essentially in the direction of propagation of the sound wave.

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*Outdoor free field is considered to be one in which no sound reflecting obstacles are present apart from the ground.

**A diffuse field is a sound field of uniform energy density for which the directions of propagation of waves are random from point to point.

This condition can easily be checked by making use of the inverse square law which merely states that the sound pressure is inversely proportional to the distance, i.e., by doubling the distance from the sound source the sound pressure level reduces by 6 dB. (Though this distance should actually be measured from the acoustic centre of the sound source, it is often sufficiently accurate to measure it from the source surface). Under these conditions the sound power is related to the sound pressure level by the equation

$$P = \frac{4\pi r^2}{Q \rho_o C} p_m{}^2 \tag{1}$$

where P is power in watts
r is the radius of the sphere in meters
$\rho_o C$ is the characteristic impedance of air in mks rayls
p_m is the mean sound pressure in N/m^2
Q is the directivity factor defined later

For a point source Q takes the values 1, 2, 4, and 8 when the source is placed in mid-air, on a hard floor, on an edge between two adjacent hard surfaces and in a corner of three hard surfaces respectively.

In order to achieve spherical radiation in practice, difficulties are usually encountered. Firstly the machine would need to be suspended and secondly an anechoic chamber — a costly construction is required.

However, the problem can be side-stepped by carrying out "out-door" measurements under hemispherical radiation conditions. The sound source must then be placed on a hard flat surface in open air and the sound pressure measurements carried out, this time around a hemisphere, ensuring that far-field conditions exist. The radius required will be slightly larger than for spherical radiation, as near-field conditions extend generally to a greater distance on account of the reflecting surface. As a rough guide far-field conditions exist at distances greater than one wavelength away from the source or at two to three times the largest linear dimension of the machine — whichever is the greatest.

For hemispherical radiation of sound in a free-field, i.e. Q = 2, the sound power level is given by:

$$10 \log_{10} \left(\frac{P}{P_o}\right) = 20 \log_{10} \left(\frac{p_m}{p_o}\right) + 10 \log_{10} \left(\frac{2\pi r^2}{S_o}\right) \tag{2}$$

where P_o is the reference sound power 10^{-12} Watts.

p_o is the reference sound pressure of 2×10^{-5} N/m^2
S_o is a reference surface area of 1 m^2

The acoustic impedance of air $\rho_o C$ is implied in the above expression and its value is assumed to be 400 mks (rayls) for normal room temperature and pressure. Appendix (C) may be referred to for corrections for other room temperatures and pressures.

The quantity $p_m{}^2$ is the mean value in space of the squares of the rms sound pressures recorded at the measurement stations on the hypothetical hemisphere around the source.

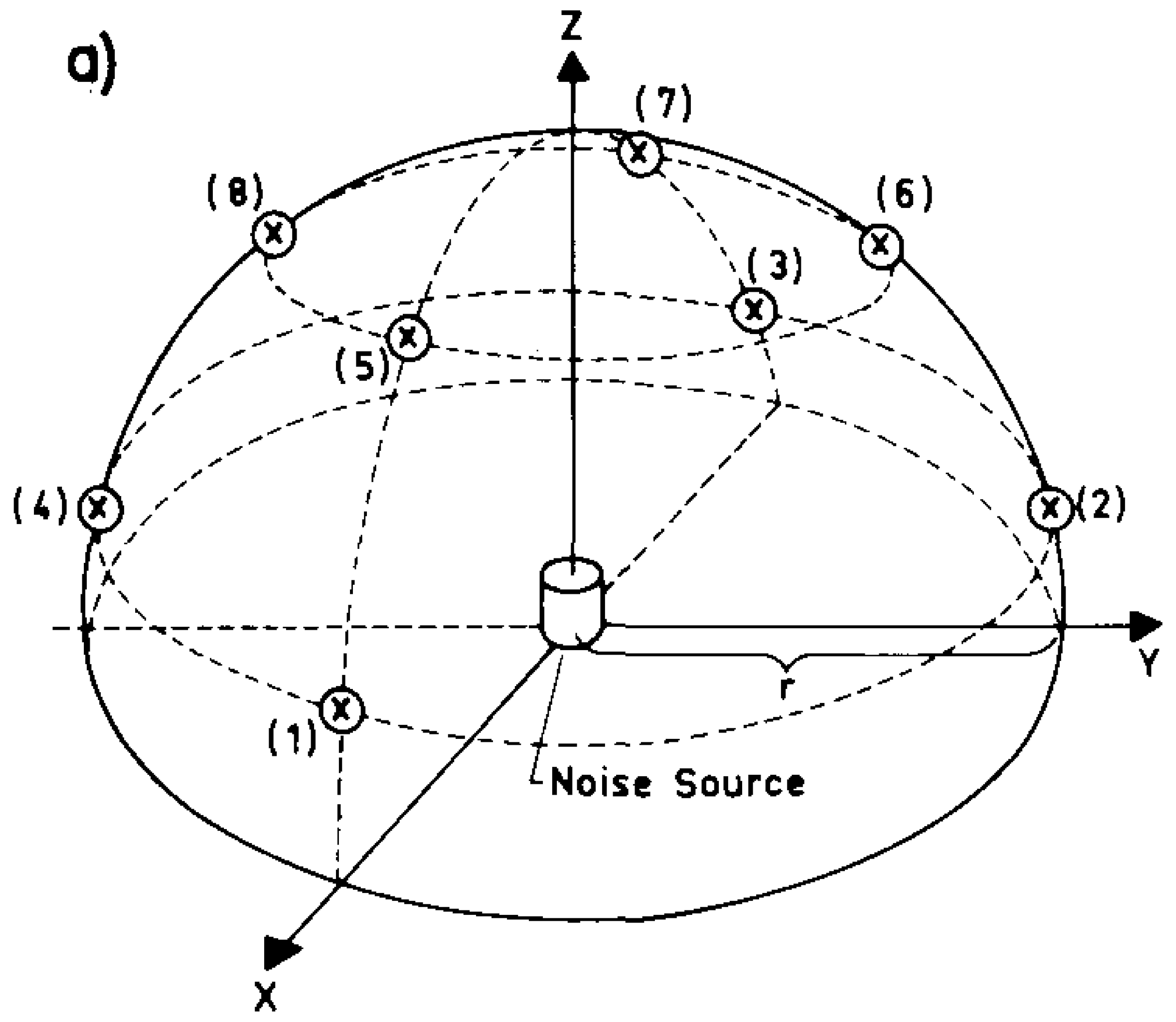
When the measuring points are chosen so that they are associated with equal areas on the hypothetical hemisphere p_m can be evaluated simply from

$$20 \log_{10} \left(\frac{p_m}{p_o}\right) = 10 \log_{10} \frac{1}{n} \sum_{i=1}^{i=n} \left(\frac{p_i}{p_o}\right)^2$$

where $$\sum_{i=1}^{i=n} \left(\frac{p_i}{p_o}\right)^2 = \left(\frac{p_1}{p_o}\right)^2 + \left(\frac{p_2}{p_o}\right)^2 + \left(\frac{p_3}{p_o}\right)^2 + \ldots\ldots\ldots\ldots \left(\frac{p_n}{p_o}\right)^2$$

Suitable schemes for the choice of six and eight measurement stations associated with equal areas are shown in Fig.1. However, more points can be chosen if greater accuracy is required. See Appendix (C). As an alternative a simple scanning microphone moving in an arc of a circle can also be used. The sound pressure level should be measured in 1/3 or 1/1 octave bands over the frequency range of interest. The background noise at the same measuring points should be determined using the same filters, when the machine is not operating. Ideally the background noise ought to be 10 dB lower than the readings at each point. However, corrections given in Appendix (C) should be applied when differences are less than 10 dB. For differences less than 3 dB the validity of results become dubious. It sometimes occurs in practice that a reflecting surface is associated with a noisy piece of equipment, e.g. an air-conditioner placed on the floor against a wall. The wall then has to be considered as an integral part of the noise producing equipment and sound power must be measured with the air-conditioner mounted in its proper place. A suitable solution to the problem would be to place the machine on an edge of a hard floor in an anechoic chamber with one of its walls covered with a hard surface. The sound power would then be radiating in a quadrant of a sphere and could be calculated from equation (2) by replacing $2\pi r^2$ by πr^2, the rest of the procedure being unchanged.

To exemplify the procedure experiments were carried out at Brüel & Kjær on an industrial type vacuum cleaner (of dimensions $0.62 \times 0.65 \times 0.65\ m^3$). The cleaner was placed on a hard floor in an a)

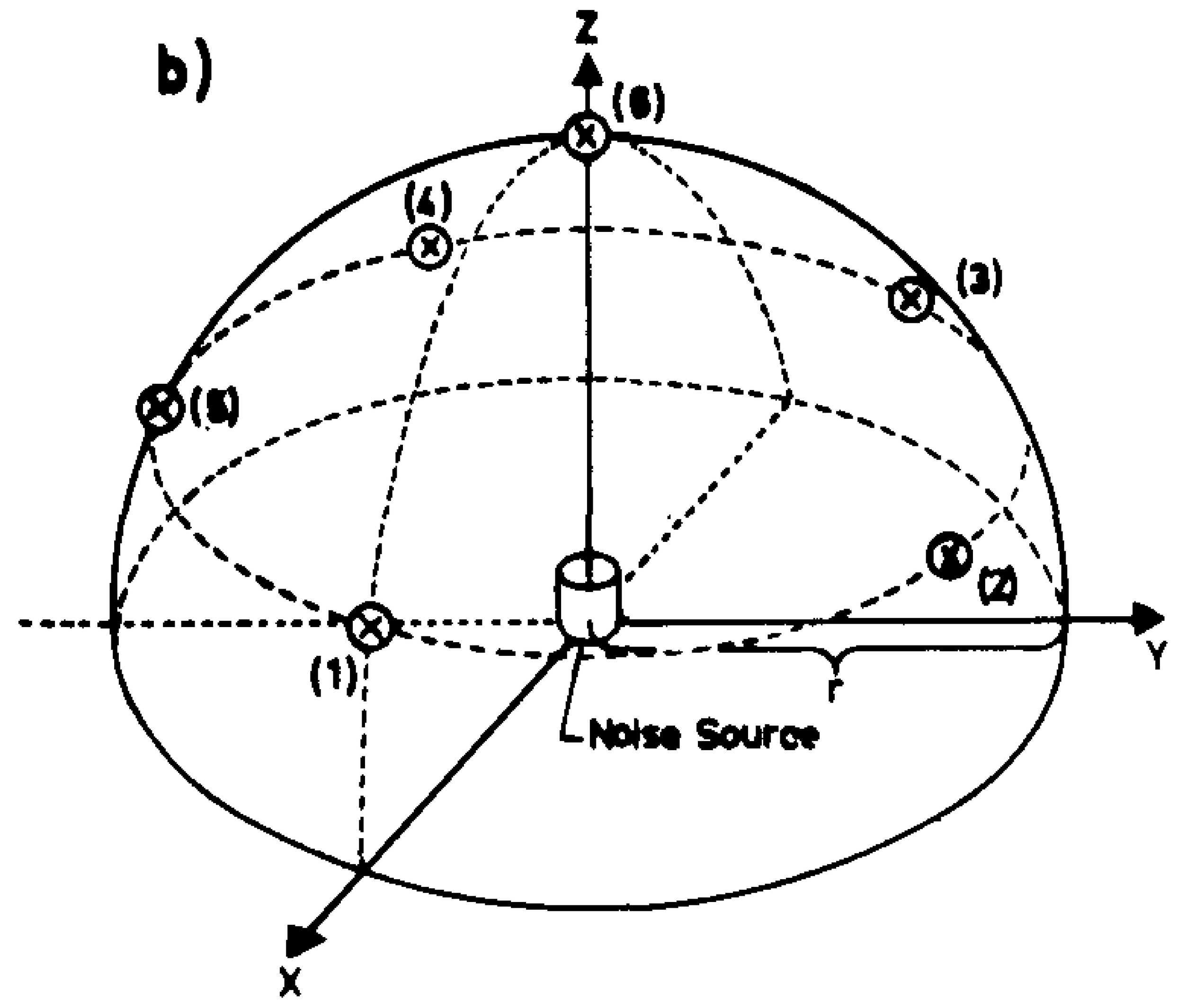


Micr. Pos.	X	Y	Z
(1)	0.97r	0	0.25r
(2)	0	0.97r	0.25r
(3)	-0.97r	0	0.25r
(4)	0	-0.97r	0.25r
(5)	0.63r	0	0.78r
(6)	0	0.63r	0.78r
(7)	-0.63r	0	0.78r
(8)	0	-0.63r	0.78r

r= Radius of Hemisphere

⊗ = Microphone Positions

b)



Micr. Pos.	X	Y	Z
(1)	0.89r	0	0.45r
(2)	0.28r	0.85r	0.45r
(3)	-0.72r	0.53r	0.45r
(4)	-0.72r	-0.53r	0.45r
(5)	0.28	-0.85r	0.45r
(6)	0	0	r

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Fig. 1. Distribution of 8 and 6 measuring points over a hypothetical hemisphere surrounding the source

anechoic chamber (of volume 295 m^3) to obtain hemispherical radiation conditions. Measurements were made at six microphone positions located on a hypothetical hemisphere as sketched in Fig.1 of radius 2 meters.

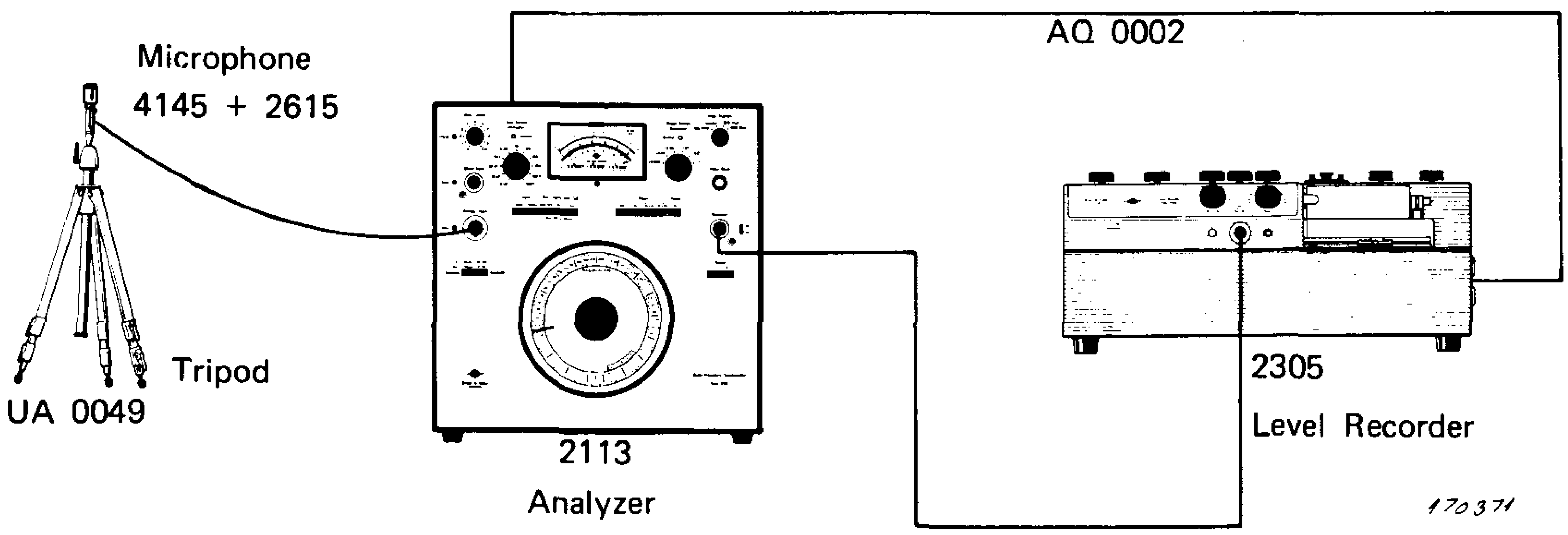

Fig.2. Measuring arrangement

The measuring arrangement used is shown in Fig.2. The system was calibrated at 1000 Hz with the Calibrator Type 4230. The sound pressure was analyzed in 1/1 octave bands by the Frequency Analyzer Type 2113 and was recorded automatically on the Level Recorder Type 2305. A typical frequency spectrum is shown in Fig.3. Sound power levels were evaluated and are shown in Table 1 below. No corrections for background noise levels were necessary as they were found to be more than 10 dB lower than the values stated in the table.

In order to obtain the mean sound pressure level, the averaging should be carried out on energy basis as shown already. However, the error obtained in using simple arithmetic averaging of decibels is less than 0.7 dB for a spread of 5 dB in the measured values and less than 2:5 dB for a spread of 10 dB.

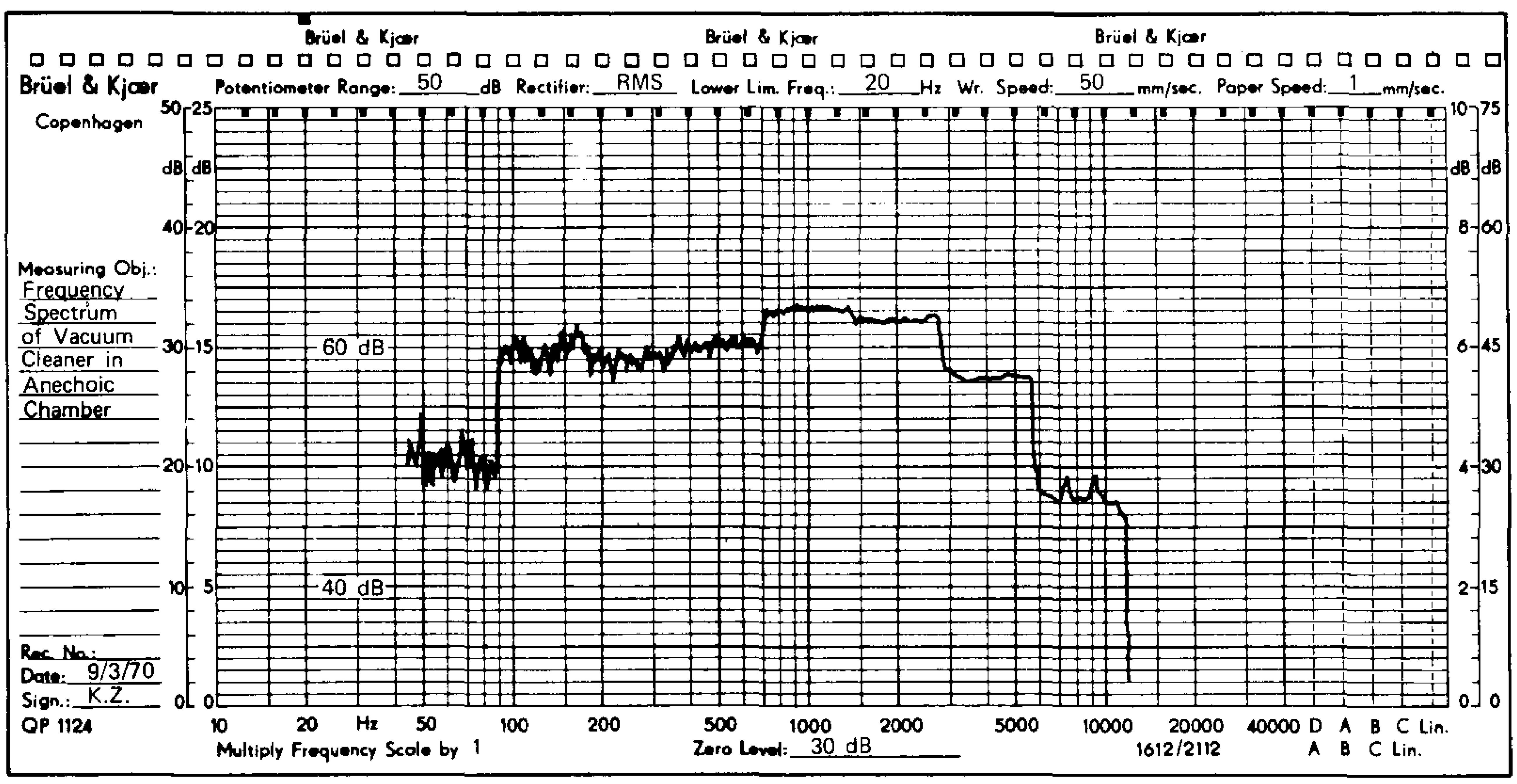

Fig.3. Frequency spectrum of vacuum cleaner in an Anechoic Chamber

Centre Freq. of Octave Band Hz f_o	Sound pressure level in dB re. $p_o = 2 \times 10^{-5}$ N/m^2						Mean $20 \log_{10} \frac{p_m}{p_o}$	Sound power level in dB. re. $P_o = 10^{-12}$ W
	Microphone positions							
	1	2	3	4	5	6		
63	52	52	55	54	50	56	53.2	67.3
125	58.5	60	62.5	61	59	59	60	74.1
250	59	60	61	61	59	60.5	60	74.1
500	59	60	60	60	60	62.5	60.2	74.3
1000	61	63	64	63	63	62	62.7	76.8
2000	62	62.5	67	66	62.5	61	64	78.1
4000	57.5	57.5	58.5	59	57.5	59	58.2	72.3
8000	51	48.5	50	49.5	47	45	49	63.1

Table 1

REF (2) & (5). Therefore to simplify calculations the following rule of thumb may be applied. Since the average of the decibel readings is always lower than the decibel of the arithmetic mean, add 1 dB to the average of the decibel readings when the spread is around 10 dB and no correction when the spread is 5 dB or less. (The error in the mean value will usually be less than ± 1 dB).

Diffuse Field Measurements

As mentioned in the introduction the second type of well defined environment in which the sound power can be measured directly, is the diffuse field. This method of sound power measurements is well suited for machines which do not have a pronounced directivity. In an ideal diffuse field the sound energy density is uniform everywhere in space. These conditions can be obtained approximately in practice in a large reverberant enclosure. Diffuseness of the field governs the accuracy of this method and is not always easy to achieve to a high degree especially at low frequencies and when the source is radiating pure tones.

Proper acoustic conditions for this method require a suitably shaped room of adequate dimensions whose surfaces can be considered hard over the frequency range of interest. By ensuring that the boundaries of the room are oblique, the sound waves will have great many reflected wave trains crossing each other, resulting in an almost uniform sound pressure level in the room. The recommended ratio for height : width : length of the room is 2 : 3 : 5 : or $1 : \sqrt[3]{2} : \sqrt[3]{4}$.

The volume of the room, however, depends on the physical dimensions of the machine and the wavelength of the lowest frequency band being considered, so that it can contain at least 20 modes of vibration at the lowest frequency.

In accordance with this requirement the volume for 1/1 octave analysis should be $V \geqslant 3\lambda_0^3$ and for 1/3 octave analysis should be $V \geqslant 9\lambda_0^3$ where λ_0 is the wavelength of the centre frequency of the lowest frequency band.

As regards the position of the machine, it is suggested that it should not be placed in the centre of the room, as an appreciable number of the resonant modes in the chamber may not be excited. When a hard floor, wall, or an edge is associated with a sound source, it must be placed in the corresponding position in the reverberant chamber during measurements. To further increase the diffuseness of the room, hard reflecting objects or rotating vanes may be placed in the room.

The deviation of the sound field from the ideal diffuse state near the sound source and the boundaries, makes the selection of microphone positions rather critical. For 1/1 octave frequency analysis, the microphone should be placed at least $\lambda_o/4$ distance away from all the room surfaces. To make sure that the microphone is not placed in the near field of the sound source, it should be placed at least one major linear dimension away from the source but not less than a distance (2/3) $V^{1/3}$ which is very nearly equal to the mean-free path. (V is the volume of the room).

In a diffuse field, the steady state sound energy in the room is equal to the difference between the sound energy transmitted by the source and that absorbed by the room boundaries. Since the sound energy is directly proportional to the sound pressure squared, the relation between the sound power emitted and the reverberant sound pressure level can be shown to be

$$P = \frac{R}{4\rho_o C} p_m^2 \qquad (3)$$

where R is the room constant (a function of frequency) defined as

$$R = \frac{S\alpha}{1-\alpha} \qquad (4)$$

S is the area of the room boundaries and
α is the average absorption coefficient in the chamber.

Under the assumption that the walls are hard, i.e.

$\alpha < 0.06$, the room constant is simply $R = S\alpha = A$ where A is a measure of the total absorption in the room.

The quantity A can be determined indirectly with the aid of the Sabine formula

$$T = 0.161 \frac{V}{A} \qquad (5)$$

by measuring the reverberation time of the room, T. (T is in sec and V in m^3). Substitution of equation (5) in (3) for R = A gives

$$10 \log_{10} \left(\frac{P}{P_o}\right) = 20 \log_{10} \left(\frac{p_m}{p_o}\right) - 10 \log_{10} \left(\frac{T}{T_o}\right) + 10 \log_{10} \left(\frac{V}{V_o}\right) - 14 \text{ dB} \qquad (6)$$

where T_o is the reference reverberation time 1 sec and V_o is the reference volume 1 m^3.

By determination of the mean sound pressure p_m in the permissible region of microphone positions, the sound power can be calculated from the above expression once the reverberation time has been measured for each frequency band. Theoretically a single measurement over the frequency range would suffice, if the diffuse field was ideal. It is recommended, however, that a spatial average be obtained from different measurement positions.

From the very nature of the sound field used in this method, it can be seen that the directivity index cannot be determined on account of the directional characteristics being drowned in the environment.

To again exemplify the procedure practically, the same vacuum cleaner and the measuring equipment were used as for the free-field environment. The vacuum cleaner was placed in a Reverberation Chamber (hard room) and the sound pressure levels were measured at five different locations in the room. The shape of the "hard" room was rather irregular and none of its walls were parallel to each other, Fig.4. It can also be seen that the dimensions of the room were rather small (volume 70 m^3). For this volume the lowest frequency at which proper measurement conditions exist is 120 Hz given by the expression $V \geqslant 3\,\lambda_0^3$. However, measurements were carried out also in the octave band of centre frequency 63 Hz. The reverberation times of the room in each octave band were measured with a high speed Level Recorder Type 2305 and are shown in column 2 of Table 2.

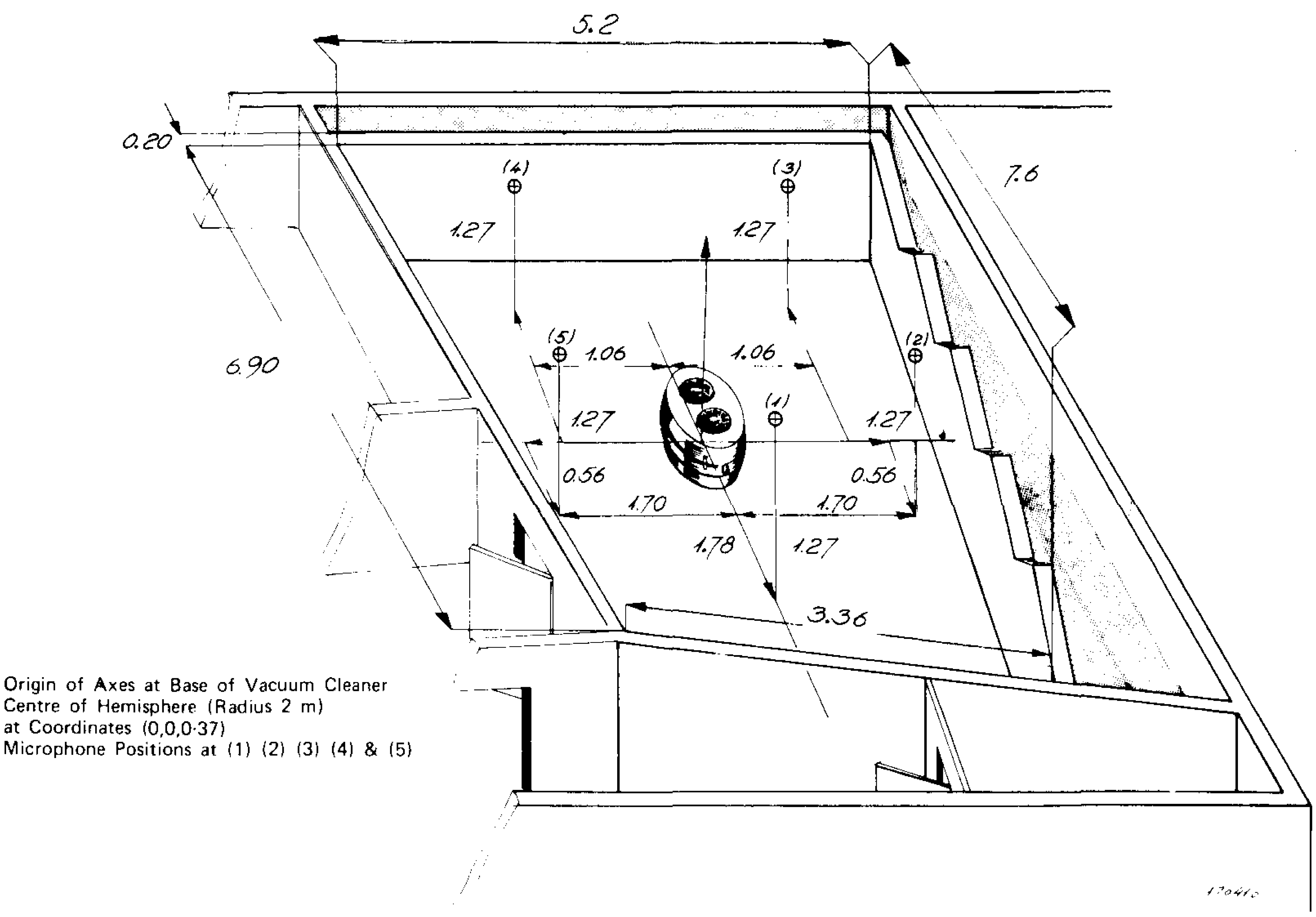

Fig.4. Sketch of the reverberation room used for measurements

Centre Freq. of Octave Band Hz f_o	Reverberation Time sec. T	Sound Pressure Level in dB re. $p_o = 2 \times 10^{-5}$ N/m^2 Microphone positions					Mean $20 \log_{10} \frac{p_m}{p_o}$	Sound power level in dB re. $P_o = 10^{-12}$ W
		1	2	3	4	5		
63	5.5	72	67	74	72	66	71.1	68.2
125	7	78	76	76	75	74	76	72.1
250	4.5	76	76	75	76	75	75.5	73.5
500	4.5	75.5	76	75	76	75	75.5	73.5
1000	3.5	77.5	77.5	77.5	77.5	77	77.5	76.6
2000	2.8	79	77.5	78.5	78.5	79	78.5	78.5
4000	2.6	72.5	72	72	73	72.5	72.5	72.8
8000	2.5	63.5	63	63.5	64.5	64	63.7	64.3

Table 2

Semi-reverberant field measurements

The ideal environmental conditions of free-field and diffuse field obtained in Anechoic and Reverberant Chambers respectively, are not always available in practice on account of economic reasons. Machines installed in factories, often cannot be moved from their location, making out-door tests impractical. Measurements under such circumstances need to be carried out under actual operating conditions in a laboratory, office or a factory where the environment has neither free-field, nor diffuse field conditions. In such semi-reverberant environments, the sound field decays more slowly and less smoothly near the source than for conditions of spherical divergence. Neither is it always possible to obtain a uniform sound field away from the source. Even in such cases the emitted sound power may be estimated from sound pressure level measurements. However, the results are often less precise than those obtained by use of the methods described earlier in this article.

Although the method is of limited accuracy, it is often the only method available and is therefore of significant practical importance. Limited information can be obtained from directivity measurements, and the results should therefore be used only for rough orientation purposes.

No specific requirements are made as regards the relative magnitude of the direct and reverberant components of the sound field in the test room. However, the room should be adequately large in order that the microphone may be placed in the reverberant part of the sound field without being too close to the room boundary surfaces. Long narrow rooms should be avoided. The machine should be mounted in the same manner as it is normally used and will typically be associated with a hard floor or a hard wall. Also it should be placed at least $\lambda_0/4$ distance away from all other boundary surfaces. This method is again not recommended for sound sources emitting spectra of narrow bandwidth. Only sound sources emitting broad-band noise are considered.

For steady-state conditions in a semi-reverberant room, it is obvious that the sound power absorbed is equal to the sound power emitted by the source. It can be shown from here, that the sound pressure p_m at a distance r from the source is made up of two components, sound that has travelled without reflection and sound that is reflected at least once from the boundaries. The relation between sound power and pressure can be shown to be

$$10 \log_{10} \left(\frac{P}{P_o}\right) = 20 \log_{10} \left(\frac{p_m}{p_o}\right) - 10 \log_{10} \left(\frac{Q}{4 \pi r^2} + \frac{4}{f(a)}\right) \qquad (7)$$

where Q is the directivity factor and f(a) is a valid measure of the absorption in the room.

The validity of this expression rests on the assumptions, that the sound energy densities in the free and reverberant fields are additive, REF. (2) and secondly that the sound pressure p_m is measured in a region excluding the near-field. There is, however, some disagreement as regards the most appropriate analytical expression to be used for f(a). Beranek, REF. (2), suggests f(a) = Room Constant $R = \frac{S\alpha}{1-\alpha}$ where S is the area of the bounding surfaces and α is the average absorption coefficient of the room. Others prefer to use f(a) = Sa where a is the average Sabine coefficient defined as $a = \log_e r$, where r is the average reflection coefficient $(1 - \alpha)$.

The mean sound pressure p_m must be determined as described under "Free-Field Measurements" and the room constant R can be determined from

$$R = \frac{V}{\frac{T}{0.16} - \frac{V}{S}} \qquad (8)$$

by measurement of the reverberation time T. V is the volume of the room.

The quantity $(10 \log_{10} (P/P_o) - 20 \log_{10} (p/p_o))$ gives a measure of the "environmental effect". From eq. (7) it can be seen that its value would be $-10 \log_{10} (Q/(4 \pi r^2))$ if the room was anechoic and would be a function of radius r. If, however, the microphone was placed in the far-field where the sound pressure did not change with distance the environmental effect would be $-10 \log_{10} (4/(f(a)))$ and would be a function of the absorption in the room. If in the "in-between" case of the semi-reverberant field where the direct and reflected sound contribute equally to the measured average pressure, an error of only 3 dB is introduced by neglecting either one of the components. From the above reasoning it is advisable to place the microphone in the far-field for non-directional sources, and near the source for highly directive ones.

The accuracy of this method of sound power measurement can be significantly increased if a reference source of known sound power level, and of similar frequency characteristics as the sound source is available. It has been shown in REF. (6) that identical spectrum shapes for the two sources are not, however, strictly required. Neither is the directivity of the two sources critical. What is actually accomplished by this method, is that a comparison between the sound power of the two sources is established, by which process many sources of error are eliminated.

Sound pressure level measurements are taken at a number of positions preferably over a hypothetical hemisphere with the sound source at its centre. The sound source is then substituted by the reference source and the measurements repeated at the same microphone positions. By equating the "environmental effect" for the two sources the sound power can be calculated from

$$10 \log_{10} \left(\frac{P}{p_o}\right) = 10 \log_{10} \left(\frac{P_r}{p_o}\right) + 20 \log_{10} \left(\frac{p_m}{p_o}\right) - 20 \log_{10} \left(\frac{p_{mr}}{p_o}\right) \qquad (9)$$

where P_r is the sound power of the reference source and p_m and p_{mr} are the mean sound pressures of the machine and the reference source respectively, at the surface of the hypothetical hemisphere.

If, however, a suitable sound source is not available, a prototype of the machine to be tested may be taken to an acoustical laboratory where its sound power can be established under ideal conditions. The prototype can then be used as a reference source.

If the machine cannot be moved, it might be possible to find an acoustically equivalent location for the reference source. In such cases, however, it may be more advantageous to choose measurement positions in the reverberant field in the room, than on a hypothetical hemisphere surrounding the source.

To include a practical example for measurements in such environments a semi-reverberant room of dimensions 6.75 x 5.75 x 4.5 m^3, and the same vacuum cleaner were chosen. The test room was cleared as much as possible, but certain objects like table, chairs, and a book-shelf were left in the room to simulate conditions of contingency met in actual practice. Measurements were taken on a hemisphere of radius 2 meters with the vacuum cleaner at its centre. As reference source a fan of similar dimensions and frequency characteristics, supplied by "F.S. Andersen" was used. The fan was now substituted for the vacuum cleaner and the measurements repeated for the same microphone positions. The sound power evaluated from equation (9) are shown in table 3.

Centre Frequency of Octave Band Hz f_o	Sound Power Level of Ref. Source dB re. $P_o = 10^{-12}$ W	Vacuum Cleaner Mean $20 \log_{10} \frac{p_m}{p_o}$	Ref. Source Mean $20 \log_{10} \frac{p_{mr}}{p_o}$	Sound Power Level of Vacuum Cleaner dB. re. $P_o = 10^{-12}$ W
63	51.3	54.8	51.6	54.5
125	60.6	62.7	56.7	66.6
250	69.0	65.4	62.8	71.6
500	59.5	65.9	51.0	74.4
1000	57.1	67.6	48.1	76.6
2000	58.0	69.9	48.8	79.1
4000	56.3	64.1	47.7	72.7
8000	51.5	55.6	42.1	65.0

Table 3

Use of Real-Time Analysis & On-Line Digital Computer Processing

It is obvious that normal methods of sound power measurements would require a considerable amount of time for scanning common analog filters and later carrying out the calculation. To speed up the procedure and simultaneously improve on accuracy an automatic method was developed. By utilizing a Real-Time Frequency Analyzer, the scanning time is saved on account of parallel filtering and the use of a computer facilitates almost instantaneous evaluation of the data. The computer transfers results to the peripheral devices and also controls the operation of the whole measuring system.

Other parts of the system include interface devices and analog input instruments to the Real-Time Analyzer. To overcome the limitation of one input channel, a computer-controlled eight channel multiplexer was innovated by means of which eight measuring transducers could be connected successively to the input of a Real-Time Analyzer Type 3347 Fig.5. Each of the eight

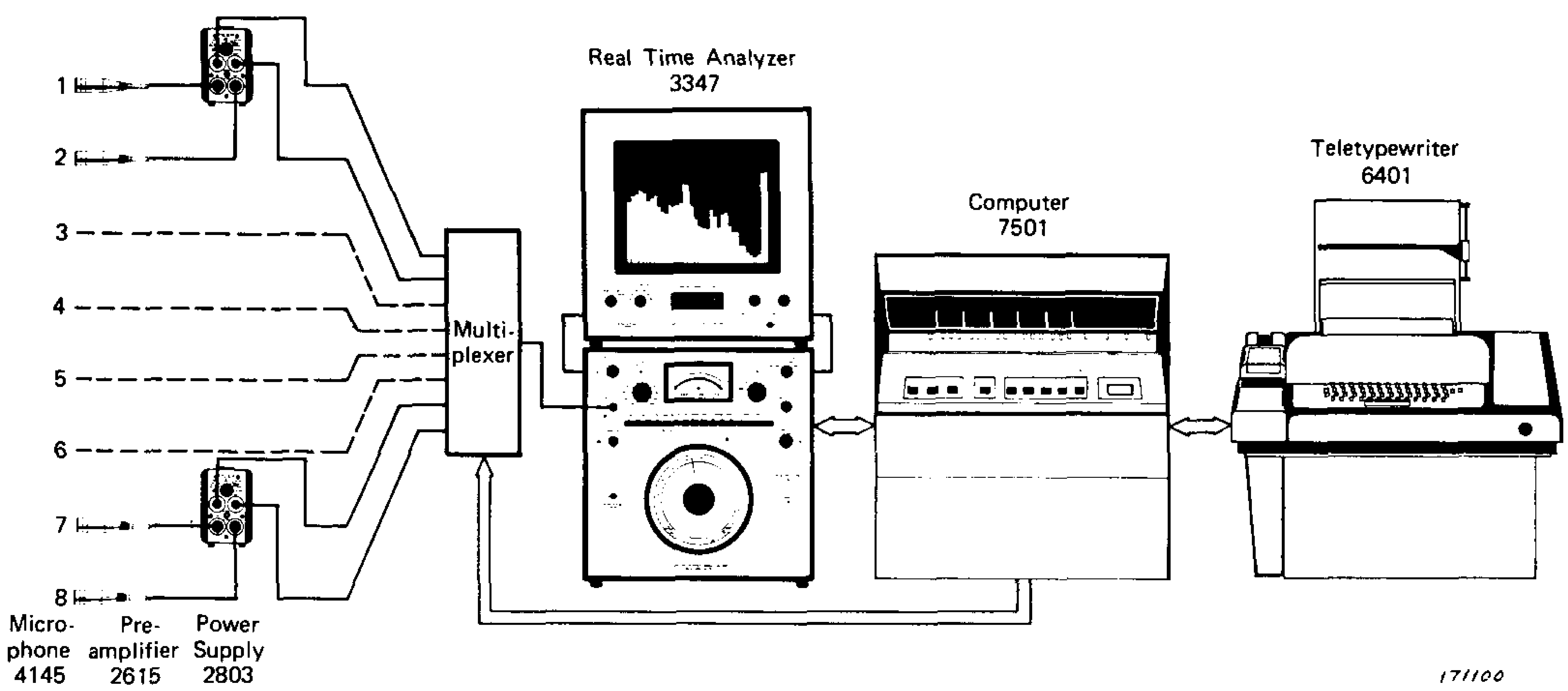

Fig.5. Measuring arrangement for real time analysis and computer processing

channels of the multiplexer contain a microphone, a suitable preamplifier and a power supply for the microphone. The signal in the channel selected by the computer is transmitted without level or waveform changes to the Real-Time Analyzer where it is analyzed, and the output transmitted in digital form to the computer connected on-line. After the data is processed in the computer the results are transferred to a teletypewriter. The automatic operation of the system, as well as the scanning rate and order of the microphones by the multiplexer are controlled by the computer program.

Fig.6 illustrates a block diagram of the multiplexer.* It contains fast electronic switches actuated by corresponding number of drivers. The drivers are controlled by logical impulses from the decoder, the blocking logic of which ensures that only one channel at a time is selected. The commands from the computer to the multiplexer are fed via the interface unit which provides all functions necessary for interconnection of an external device to the computer.

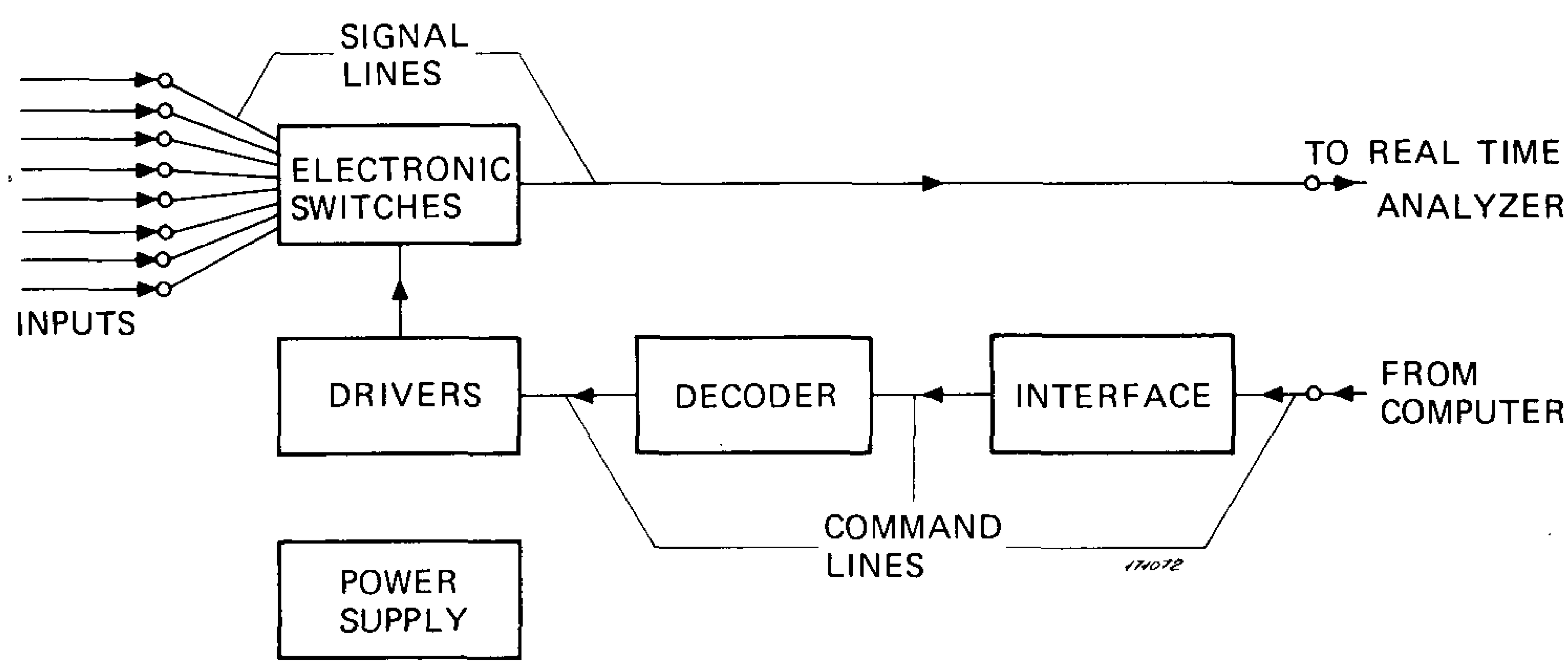

Fig.6. Block diagram of multiplexer

A Varian computer of memory 4 K was used for which the program was written in assembler language. Fig.7 illustrates the flow chart of the system software. The data to be read-in the program are the time interval between each spectrum, and the distance in decimeters between the sound source and the microphone if free-field environment is used. If, however, measurements are carried out in a diffuse field, the reverberation times in each 1/3 octave band and the volume of the room must be read in, instead.

*For details of the multiplexer see article "An Experimental Channel Selector System" by V. Kop page 28.

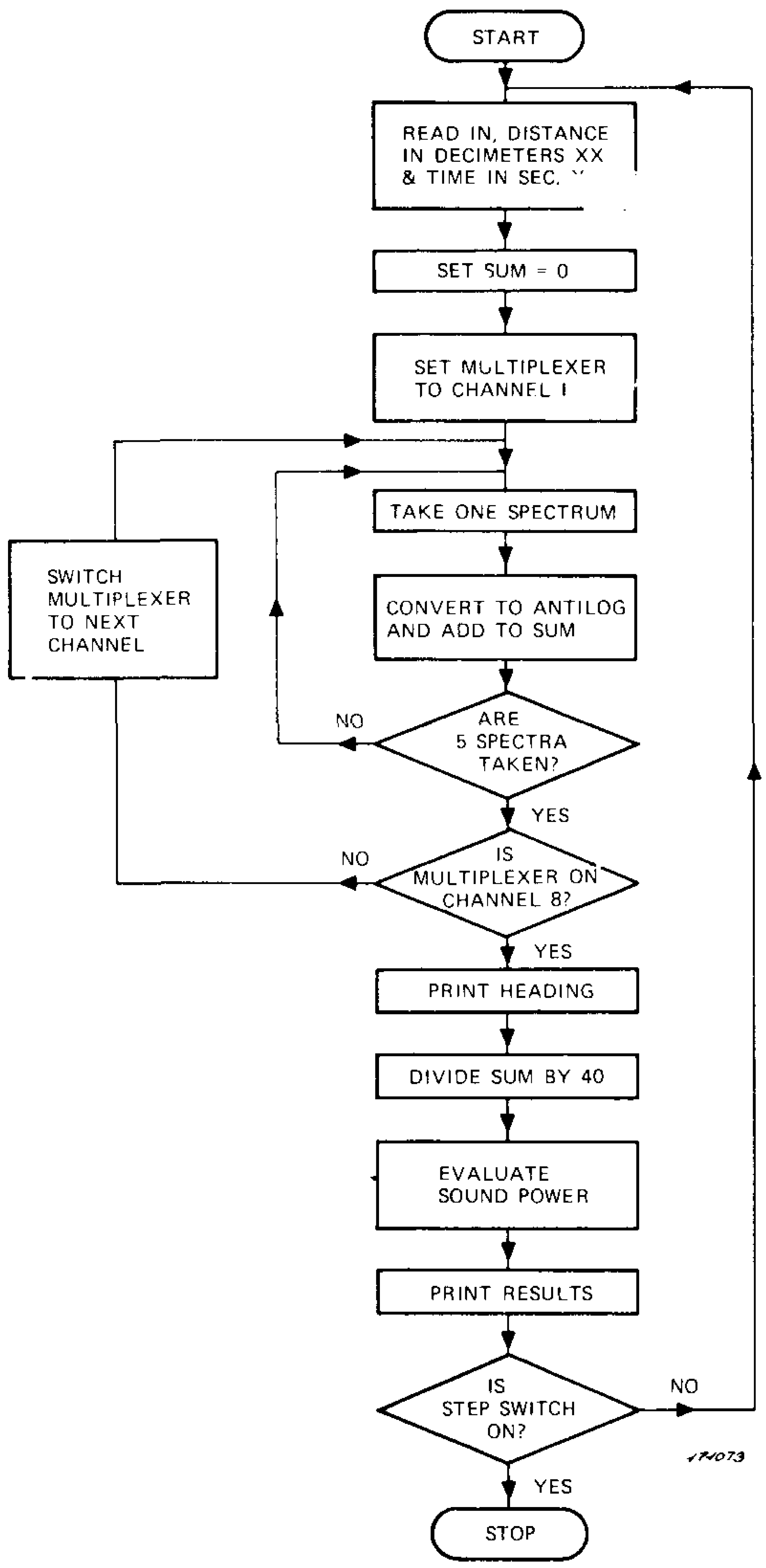

Fig.7. Flow chart of computer software

In each frequency channel the sound pressure level is set to zero (i.e. "Sum" = 0). The multiplexer is set to microphone position 1, and a spectrum is taken. The dB values are changed to ratios and added to the "Sum". The procedure is repeated until five spectra are taken after which the multiplexer switches to the next microphone position following a command from the computer. A total of forty spectra are taken from the eight microphones, the ratios of which are added in the "Sum" for each frequency channel. The "Sum" is then divided by forty and sound power evaluation is carried out. The results are plotted on the teletypewriter as a 1/3 octave spectrum shown in Fig.8. The numbers under "Frequency Channel Number" correspond to ($10 \log_{10} f_o$) where f_o is the centre frequency of the 1/3 octave band. Each "x" in every channel represents 1 dB above the reference level typed at the beginning of the spectrum. If the "Step" switch

```
DISTANCE IN DECIMETRES = 20
READ-IN INTERVALS (SEC) =  1

12345678

SOUND POWER LEVELS IN ONE-THIRD OCTAVE BANDS

FREQUENCY   SOUND
CHANNEL     POWER
NUMBER      LEVEL                                         SOUND POWER
                                                          MEASURED BY
  REF.      40.0 DB                                       CONVENTIONAL
   19       xxxxxxxxxxxxxxxxx  56.5 DB                    INSTRUMENTS
   20       xxxxxxxxxxxxxxxxxxxxxx  61.9 DB                   DB
   21       xxxxxxxxxxxxxxxxxxxxxxxxxxx  66.7 DB              70
   22       xxxxxxxxxxxxxxxxxxxxxxxxxxxxx  68.7 DB            69
   23       xxxxxxxxxxxxxxxxxxxxxxxxxxxx  68.4 DB             69
   24       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  69.8 DB           70
   25       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  69.8 DB           70
   26       xxxxxxxxxxxxxxxxxxxxxxxxxxxxx  68.8 DB            70
   27       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  70.6 DB          71
   28       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  71.2 DB          71
   29       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  70.3 DB           70
   30       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  69.9 DB           70
   31       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  73.1 DB        73
   32       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  74.7 DB      75
   33       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  72.6 DB        72
   34       xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx  73.4 DB        74
   35       xxxxxxxxxxxxxxxxxxxxxxxxxxxx  68.3 DB             68
   36       xxxxxxxxxxxxxxxxxxxxxxxx  64.4 DB                 65
   37       xxxxxxxxxxxxxxxxxxxxxx  61.8 DB                   63
   38       xxxxxxxxxxxxxxxxxxx  59.3 DB                      61
   39       xxxxxxxxxxxxxxxxxxx  58.6 DB                      59
   40       xxxxxxxxxxxxxxxxxx  57.9 DB                       58
```

Fig.8. Results as they appear on teletypewriter

on the computer is "on" the computer is set in the "stand by" condition and the operation of the system is instantly stopped.

To check the operation of the system the same vacuum cleaner was used and was placed on a hard floor in an anechoic chamber. Eight microphones were placed at a radius of 2 meters from the source, and sound power evaluated. Sound power was also measured in 1/3 octave bands by sequential filtering, and the results obtained were found to be in fair agreement with the automatic method.

On an average the difference was 0.5 dB and a maximum difference of 3.3 dB at 125 Hz was obtained.

Conclusion

In order to compare the accuracy achieved by the three different methods of sound power measurements, Table 4 was compiled. Assuming the sound power levels obtained in anechoic chamber are true values, the errors in dB obtained in hard room measurements and semi-reverberant field measurements are given in columns five and six respectively. Fig.9 shows a plot of the sound power levels obtained by the three methods against frequency.

Centre Frequency of Octave Band Hz	Sound Power Level of Vacuum Cleaner			Error dB Sound Power in Anechoic Room — Hard Room	Error dB Sound Power in Anechoic Room — Semi-Rever. Room
	In Anechoic Room	In Hard Room	In Semi-Reverberant Room		
63	67.3	68.2	54.5	−0.9	12.8
125	74.1	72.1	66.6	2.0	7.5
250	74.1	73.5	71.6	0.6	2.5
500	74.3	73.5	74.4	0.8	−0.1
1000	76.8	76.6	76.6	0.2	0.2
2000	78.1	78.5	79.1	−0.4	−1.0
4000	72.3	72.8	72.7	−0.5	−0.4
8000	63.1	64.3	65.0	−1.2	−1.9

Table 4

The measurements carried out in the anechoic chamber and in the hard room agree to within 2 dB in all frequency bands. The error in the semi-reverberant field measurements has a maximum value of 2.5 dB except for the two lowest frequency bands. The error could be attributed possibly to insufficient mode density. In fact the first octave band has intentionally been included in the results to throw light on the short-comings of sound power measurements at low frequencies. The unsteady value of the sound pressure level itself can be seen from the frequency spectrum Fig.3.

It should be noted that the values of sound power level should be quoted to the nearest half decibel. The decimal place, however, has been included here to illustrate the magnitude of difference in results obtained for different sound power measurement methods. From the overall results, it can be concluded that the sound power of sources emitting broad-band noise, can be determined in semi-reverberant spaces with accuracies adequate for practical engineering purposes, when ideal environmental conditions are not available.

The computer-controlled system is characterised by several advantages over conventional measuring instrumentation, the primary one being saving in time. Measurement and evaluation of the data take roughly a minute and another minute is required for print-out of the results. Thus by providing facilities for quick interchange of specimens, the system is ideal for fast serial work. Although the maximum scanning rate of the multiplexer can be of the order of 1000 channels per second, the scanning rate in practice is governed by the time constants of the Real-Time Analyzer. To increase accuracy any number of spectra can be taken, on account of the nature of the computer program, as long as overflow in the "Sum" is avoided.

The ease with which the computer software can be changed makes the

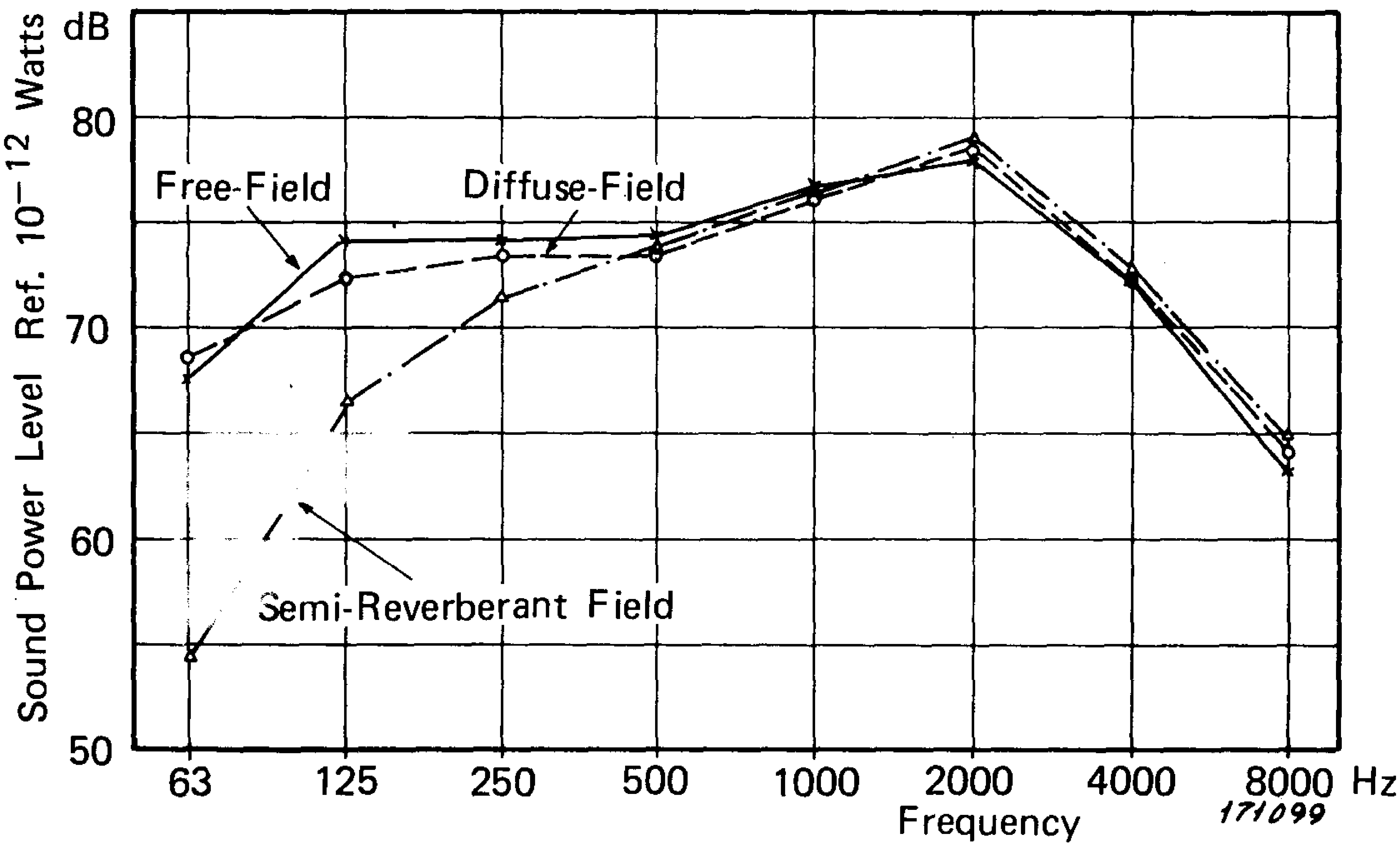

Fig.9. Sound power levels measured in free-field, diffuse-field and semi-reverberant field

system flexible for carrying out different types of tests. With modifications to the computer program only, the directivity could be measured at intervals of 45° around a circle, with 8 measuring channels. Although ISO Recommendation suggests 30° intervals (requiring 12 measuring channels) 8 measuring channels would suffice, either if the source was not highly directive, or radiated sound power symmetrically. If, however, more channels are required, additional multiplexing units may be connected in series.

Computer program has also been written for measurements of sound power in a hard room of known volume and reverberation times.

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6. WELLS, R.J. & WIENER, F.M.: On the determination of the Acoustic Power of a source in Semi-Reverberant spaces. Noise Control Vol. 7. No. 1, p. 21.

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APPENDIX A

Directivity Factor and Directivity Index

The directional characteristics of a sound source are described in numerical measures by quantities termed, directivity factor and directivity index. The directivity factor Q (f) is the ratio of the intensity on a designated axis of a sound source, at a specified distance r and frequency band, to the intensity that would be produced at the same position by a non-directional point source radiating the same power.

From the definition it can be seen that for the axis of maximum radiation the directivity factor is always greater than unity and for a non-directional source the factor is unity for all directions.

The directivity index is defined as

$$\text{D. I. (f)} = 10 \log_{10} Q\text{ (f)}$$ where (f) denotes function of frequency.

In terms of sound pressure level the directivity index is given by

$$\text{D. I. (f)} = 20 \log_{10} \frac{p_i}{p_o} - 20 \log_{10} \frac{p_m}{p_o} \qquad (10)$$

where p_i is the sound pressure in the i^{th} direction at a distance r and p_m is the mean sound pressure on a hypothetical sphere of radius r with the sound source placed at its centre.

For hemispherical radiation conditions with the sound source placed on a hard reflecting surface the directivity index is given by

$$\text{D. I. (f)} = 20 \log_{10} \frac{p_i}{p_o} - 20 \log_{10} \frac{p_m}{p_o} + 3\text{ dB} \qquad (11)$$

Correction for background noise levels should be carried out if necessary.

Values of directivity index at intervals of 30^o would generally suffice.

APPENDIX B

Near-Field Measurements
When none of the described three methods of sound power measurement is practical, either on account of reflecting surfaces or high background noise levels, the sound pressure level may be measured at a number of points suitably distributed around the source in the near-field, to minimize the effect of background noise.

This is done as follows:
A "prescribed surface" conforming approximately to the outside casing of the machine should be marked out around the source, and should be as simple as possible so that its area can be easily calculated. The number of microphone positions and their disposition will depend on the size of the machine and the irregularity of the acoustic field. A typical disposition of measuring points on a "prescribed surface" is shown in Fig.B.1. The area S of the "prescribed surface" and its average distance from the source should be noted. The values of the sound pressure levels measured around the source must be averaged to determine p_m as described under section "free-field measurements". If the source is not highly directive and the variation not greater than 5 dB, an arithmetic mean would suffice. The area of the "prescribed surface" is used in the determination of the radius of an equivalent hemisphere in the equation

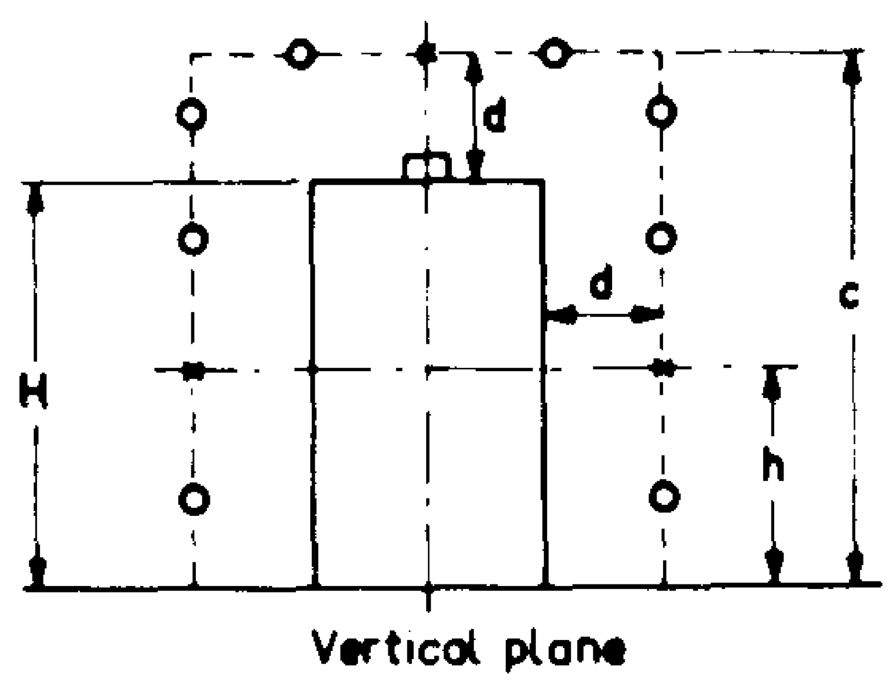
$$r = \sqrt{\frac{S}{2\pi}} \qquad (12)$$

The sound pressure level p_d at a "reference distance" d can be evaluated from

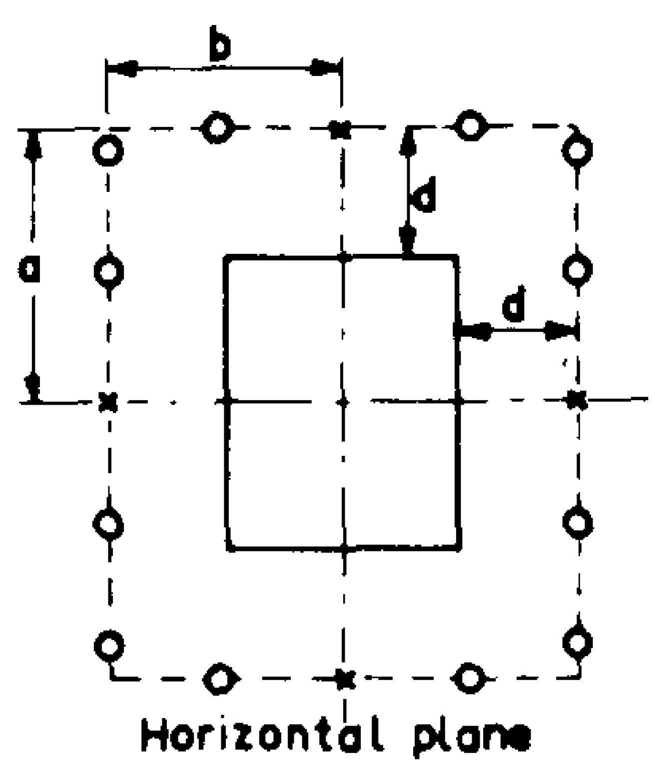
$$20 \log_{10} \left(\frac{p_d}{p_o}\right) = 20 \log_{10} \left(\frac{p_m}{p_o}\right) - 20 \log_{10} \left(\frac{d}{r}\right) \qquad (13)$$

The recommended values for "reference distance" d are 1, 3 or 10 meters. The approximate value of the sound power can be evaluated under certain circumstances. From eq. (2) it can be seen that the numerical value of the sound power for hemispherical radiation conditions at reference distances of 1, 3 and 10 meters would be 8, 18 and 28 dB greater than the sound pressure level at the reference distance.

This method is chiefly used for comparison of performance of different machines. With this in mind, the reference distance should be the same for all machines and preferably slightly greater than the radius of the equivalent



l (Metres)	d (Metres)
$> 0{\cdot}25$	1
$< 0{\cdot}25$	$4l \leq d \leq 1$ $d > 0{\cdot}25$



l = Maximum linear dimension of machine
$h = \frac{H}{2}$ but not less than 0·25 Metre
x = Key measuring points
o = Other measuring points marked off at intervals of 1m from key points

Fig. B.1. Disposition of measuring points on a "prescribed surface"

hemisphere for the largest machine considered. Use could also be made of "A" weighting network instead of 1/3 and 1/1 octave frequency bands.

If it is desired to determine the effect of the environment, a direct comparison may be made by placing the machine in a space with no reflecting objects and repeating the measurements. Care should, however, be taken to ensure constant conditions of machine noise. If this is impractical, the sound pressure may be measured over two surfaces, whose area have a ratio of 2 or 4. The sound pressure level difference between the two surfaces should then be –3 and –6 dB respectively. It is recommended, however, that the influence of the environment be kept to a minimum, since it's accurate determination is not possible.

APPENDIX C

Corrections For Room Temperature & Pressure

Correction values to be added to the sound pressure level for appropriate temperature and barometric pressure. Atmospheric pressure is in mm of mercury and temperature in degrees Kelvin. Zero correction is for 20°C and 760 mm of mercury. Fig.C.1.

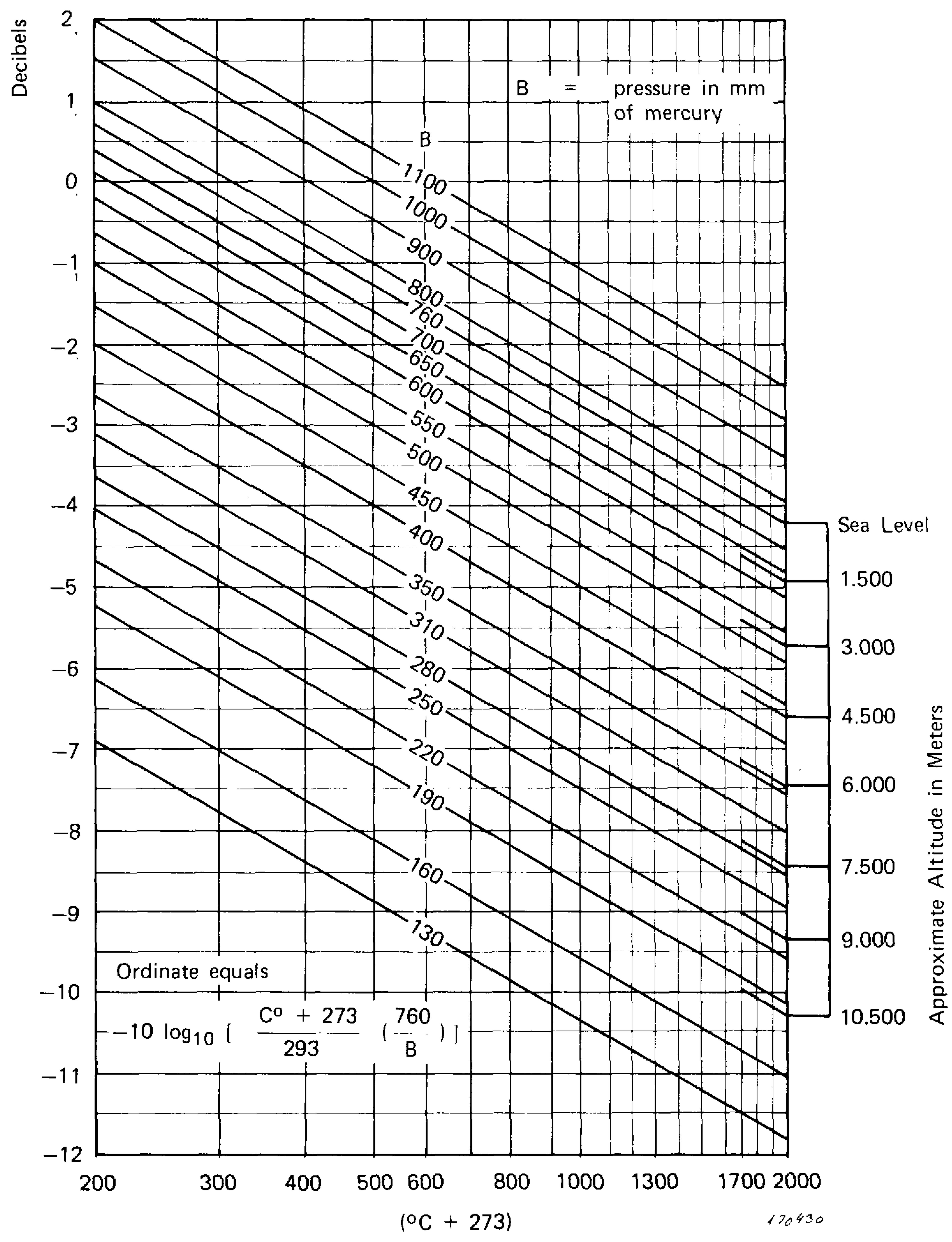
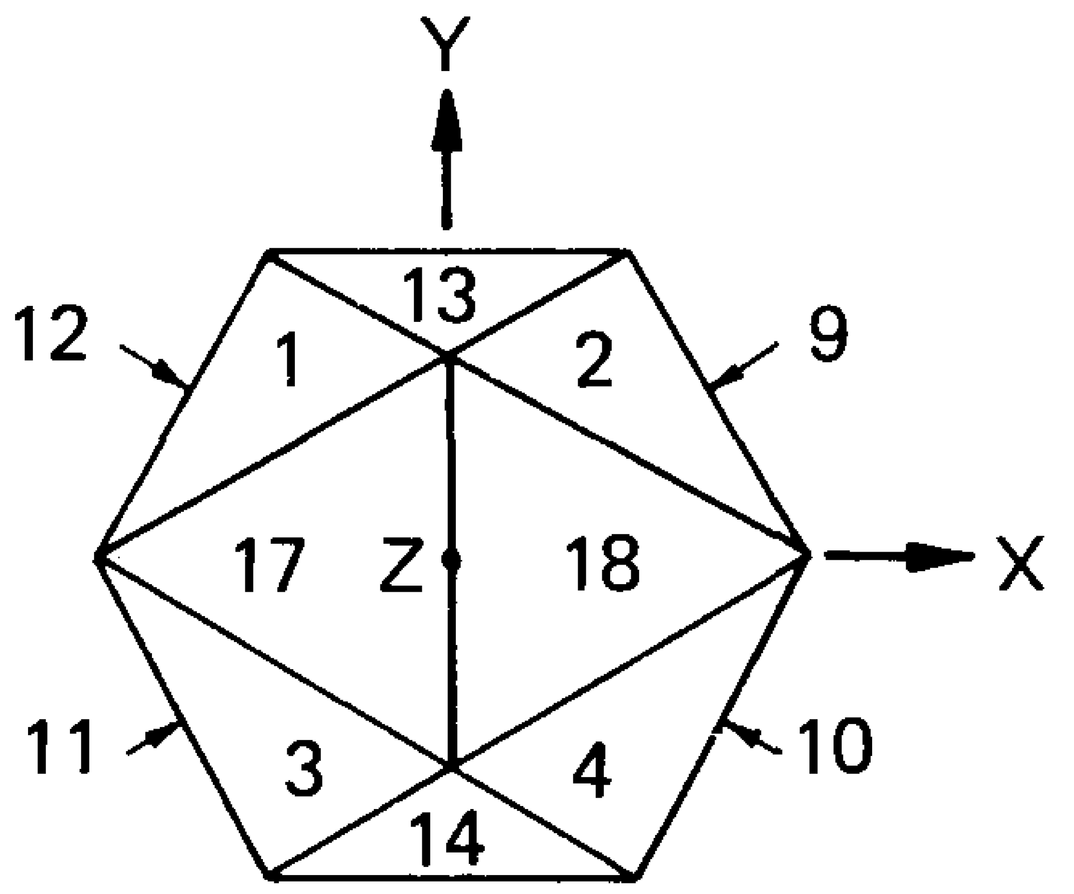


Fig.C.1. Correction chart for ambient temperature and pressure

Disposition Of 20 Measuring Positions

Fig.C.2 shows division of the surface of a sphere in 20 equal areas of identical shapes. The co-ordinates of the mid-points of the sectors are given in the table.



View from top of sphere

Appendix B

Coordinates of mid-points of sectors			
Sector numbers	Coordinate		
	X	Y	Z
1 - 8	$\pm 1/\sqrt{3}$	$\pm 1/\sqrt{3}$	$\pm 1/\sqrt{3}$
9 - 12	$\pm$ 0.934	$\pm$ 0.357	0
13 - 16	0	$\pm$ 0.934	$\pm$ 0.357
17 - 20	$\pm$ 0.357	0	$\pm$ 0.934

171098

Fig.C.2. Disposition of 20 measuring points for equal areas on a sphere

Corrections For Background Noise

Since the presence of background noise at the place of measurement affect the results, correction values must be subtracted from the sound pressure level being measured. In Fig.C.3 the correction values ΔL_N are plotted against the difference between the sound pressure level measured L_{S+N},

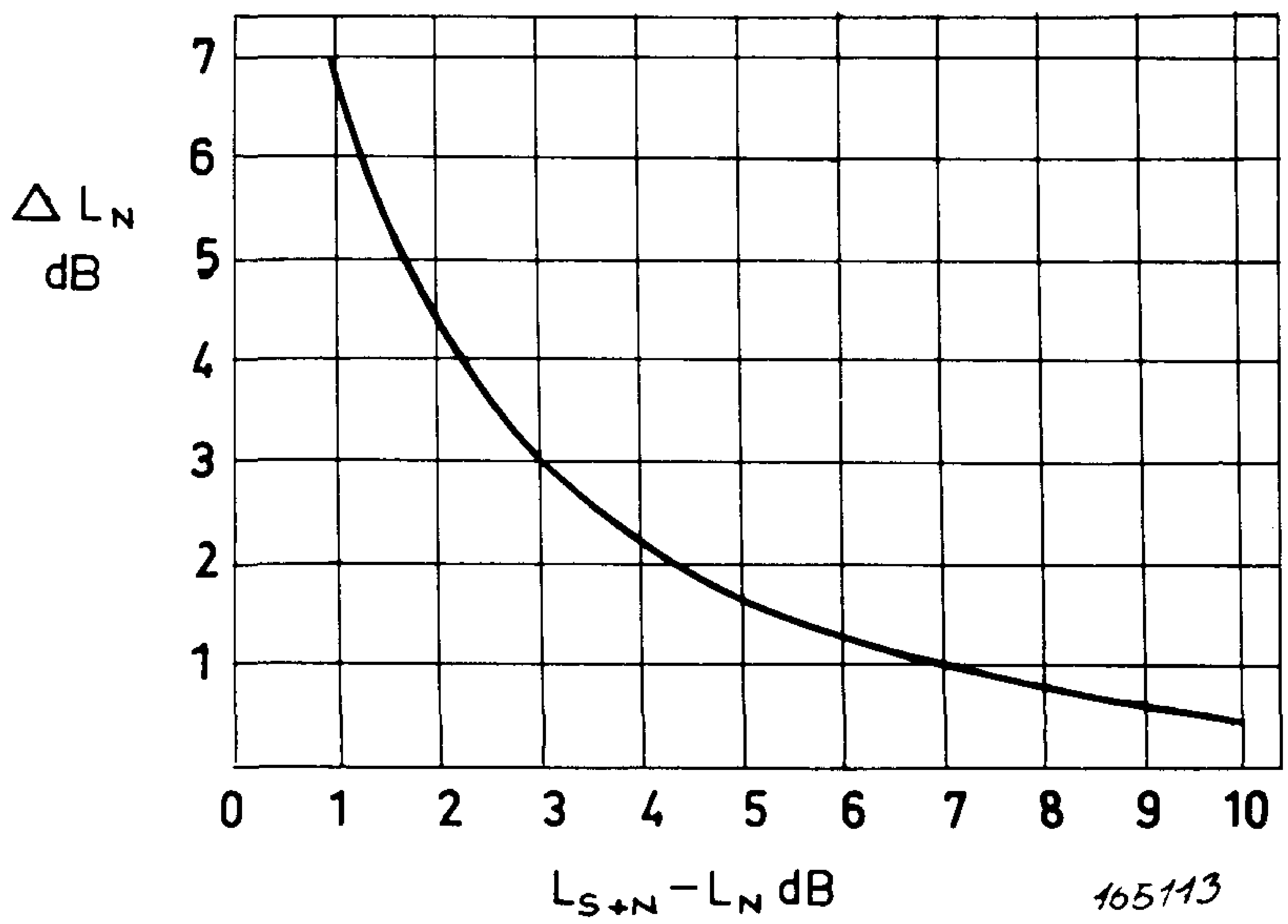


Fig.C.3. Correction chart for background noise levels and the background noise level L_N when the sound source is shut off. As can be seen from the graph, no correction (<0.5 dB) is needed when the difference is 10 dB or more, while 3 dB must be subtracted for a difference of 3 dB. For difference between 10 dB and 3 dB an approximate correction value can be derived from the chart. For differences less than 3 dB, it is advisable generally to move the sound source to a quieter place.

An Experimental Channel Selector System

by

V. Kop, M. Sc.

ABSTRACT

Modern instrumentation systems often require the measurement and/or monitoring of signals from a number of measurement positions.

The article describes briefly an experimental channel selector system which allows such multipoint measurements and/or monitoring to be carried out either manually or automatically. When switched to operate automatically the system can be controlled from an external generator (clock), or the control can be carried out from a digital computer programmed accordingly.

The channel selector system has been used successfully for on-line determination of sound power in conjunction with a minicomputer and a real-time analyzer, as well as in vibration testing for multipoint control in the feed-back loop.

A variety of other applications for the system are suggested.

SOMMAIRE

Les ensembles expérimentaux modernes nécessitent souvent la mesure et/ou la régulation de signaux provenant de nombreux points de mesure.

Cet article décrit brièvement un ensemble sélecteur multi canaux qui permet de telles mesures en plusieurs points soit manuellement soit automatiquement. Lorsqu'il est destiné à fonctionner automatiquement, l'ensemble peut être commandé à partir d'un générateur extérieur (horloge) ou bien d'un calculateur numéral convenablement programmé.

Le sélecteur a été employé avec succès associé à un petit calculateur et à un analyseur en temps réel pour la mesure de puissance sonore aussi bien que lors d'essais en vibrations pour une régulation en plusieurs points dans une boucle d'asservissement. D'autres applications de ce système sont suggérées.

ZUSAMMENFASSUNG

Moderne Meßeinrichtungen erfordern häufig die Messung und/oder Überwachung von Meßdaten an einer Vielzahl von Meßpunkten.

Hier wird ein experimentelles Kanalwählersystem beschrieben, das derartige Mehrpunktmessungen und/oder -überwachungen automatisch oder manuell auszuführen gestattet. Beim automatischen Betrieb kann das System sowohl von einem externen Taktgenerator als auch von einem entsprechend programmierten Computer gesteuert werden.

Das Kanalwählersystem wurde bereits für die automatische Messung der Schalleistung in Verbindung mit einem Minicomputer und einem Echtzeitanalysator erfolgreich eingesetzt, ebenso bei der automatischen Schwingprüfung zum Zweck der Mehrpunkt-Regelung. Eine Vielfalt weiterer Anwendungen für das System wird vorgeschlagen.

Introduction

Experimental acoustic investigations, control and monitoring at a large number of positions, require devices capable of measuring, processing and evaluating data automatically. With the increasing demand for quick analysis, the use of real-time instruments are inevitable. Flexibility of the measuring system can be greatly increased by the use of digital equipment with a small computer as a central element.

To overcome the limitation of one input channel to such systems, some experiments have been carried out with an electronic channel selector which could either be operated manually, by an external generator or be computer-controlled. By means of the channel selector eight measuring transducers could be connected successively to the input of a real-time analyzer. If, however, more than eight channels are required, additional selector units can be connected in series.

The channels are selected in a progressive sequence when operated manually or by an external signal generator. When used in conjunction with a computer, the dwelling time in each channel and the order of the channels may be chosen arbitrarily by the software.

Description

Fig.1 shows the use of the channel selector system when utilised in conjunction with other instruments.

Each of the signal channels contains a transducer (microphone, accelerometer or other pick-up, depending on the measurement problem) together with a suitable preamplifier and a power supply.

The signal in the selected channel is transmitted by the channel selector system without any level or waveform changes to the input of a real-time frequency analyzer. Blocking logic of the channel selector system ensures that only one channel at a time is selected.

The analyzed signal is then transmitted to the analog and/or digital data receivers (level recorder, tape puncher, printer, magnetic tape recorder, computer "on-line" etc.). When the channel selector unit is used in conjunction

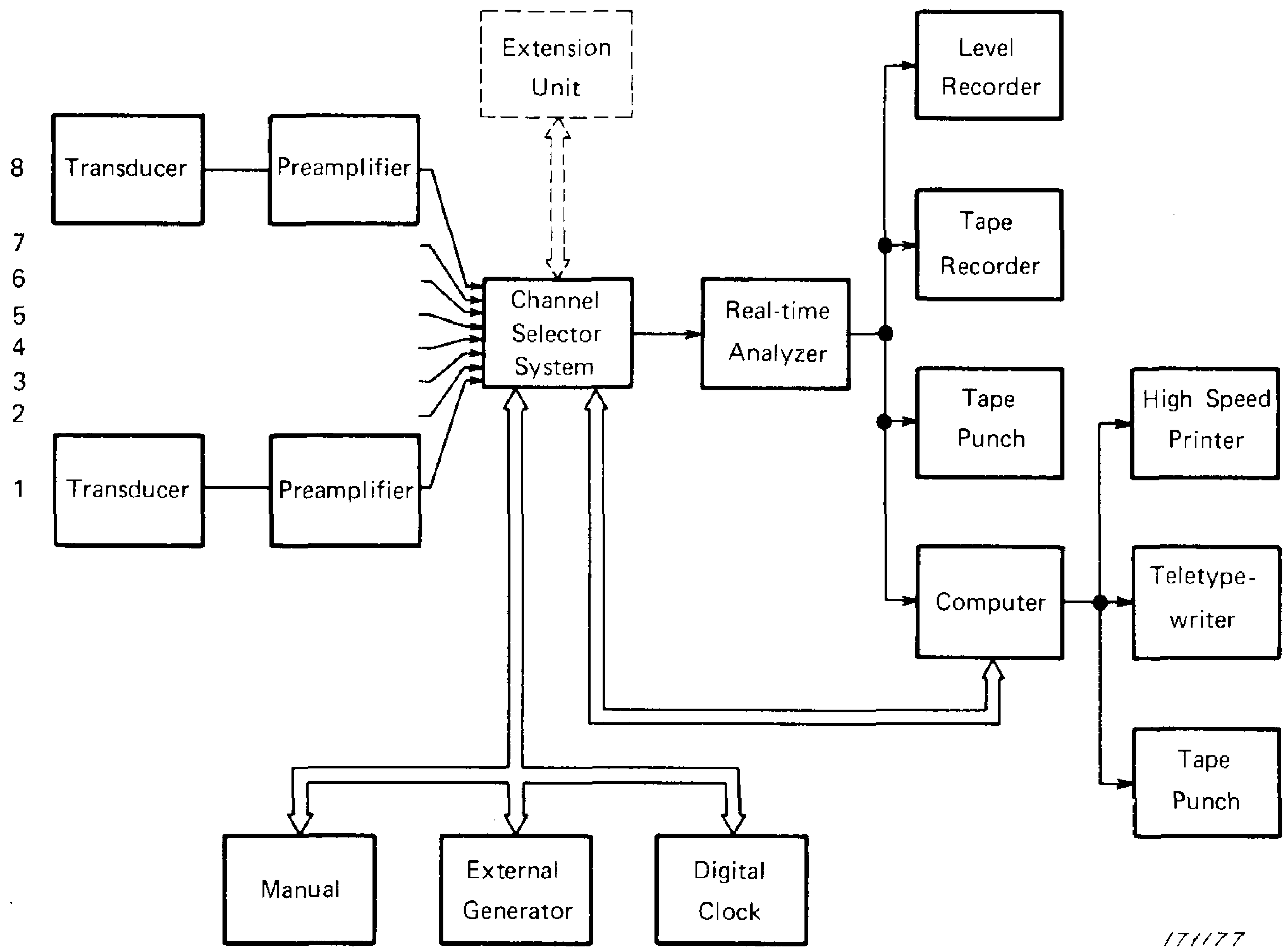


Fig. 1. *General measuring and processing arrangement with the channel selector system*

with a computer, the automatic operation of the whole system may be controlled by the computer programme.

The principle of operation of the automatic and computer-controlled channel selector system is illustrated by Fig.2.

Fast FET switches activated by a corresponding number of drivers are used as switching elements. Special circuits were used to avoid problems arising from switching transients.

The drivers are operated by means of logical impulses depending on the mode selected. The scanning of channels in a progressive sequence is carried out by the digital logic circuitry, consisting mainly of a decade counter and ar "one-out-of-eight" decoder. A triggering circuit is provided for scanning with the aid of an external signal generator.

The number of channels to be scanned in one cycle can be selected by means of a switch. The scanning cycle repeats itself automatically, starting from channel No.1. A second switch is included for manual reset of the system at any time.

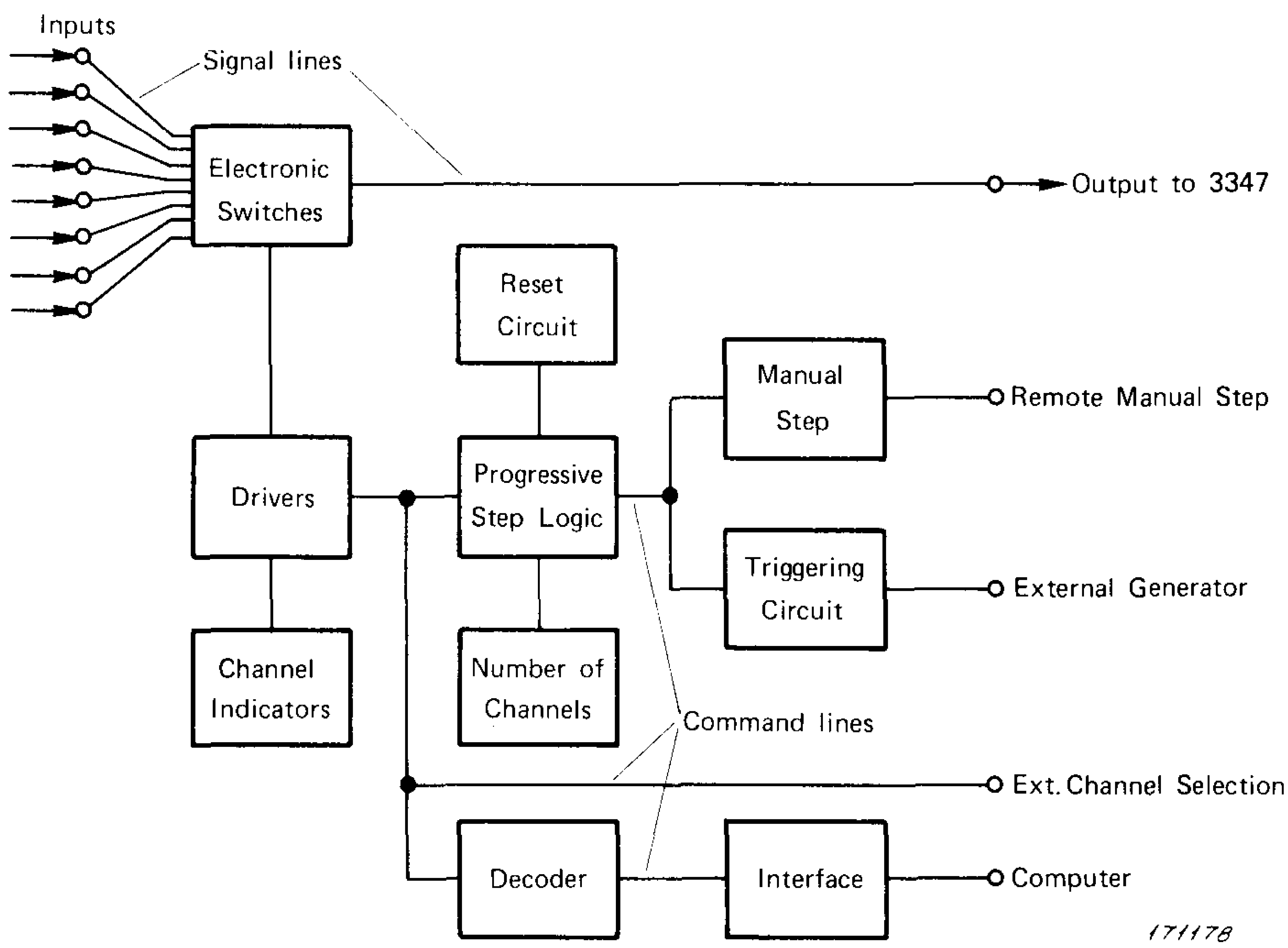


Fig.2. Block-diagram of the channel selector system

Separate logic circuitry is provided which allows the channel selector system to be used in conjunction with a computer. It consists mainly of a decoder (which decodes binary coded inputs to 8 mutually exclusive outputs) and an interface unit, providing all functions necessary for interconnection of the selector system to the computer (e.g. it processes signals indicating that the operation of an external device is in progress and that a device address and a control code are placed in the corresponding computer bus).

Logic circuit and sockets are built-in for connecting additional channel selector units in series and for remote manual scanning of channels.

Examples of Application

Measurement Of Noise Emitted By Machines

Upto 8 microphones are located in the test chamber where the machine under investigation is situated. If the test environment is fully reverberant, the microphones may be placed randomly. For free-field or semi-reverberant conditions the microphones should be placed symmetrically around the machine. The measuring system analyses and evaluates the noise emitted by the machine in each microphone position.

As an example, the measurement of the average sound spectra of the noise emitted by an industrial type vacuum cleaner were made. The measuring arrangement with the vacuum cleaner placed in an anechoic room appears in Fig.3.

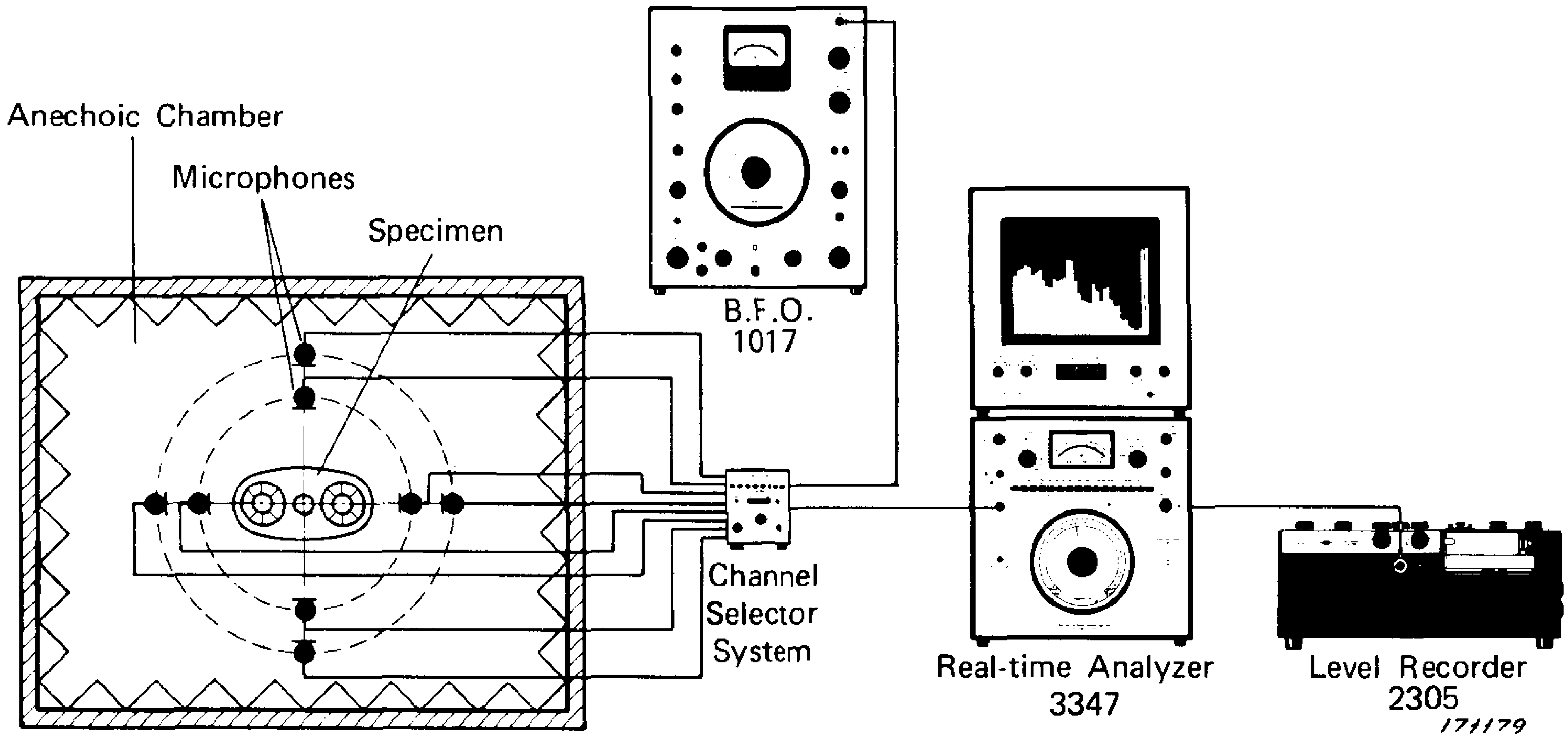
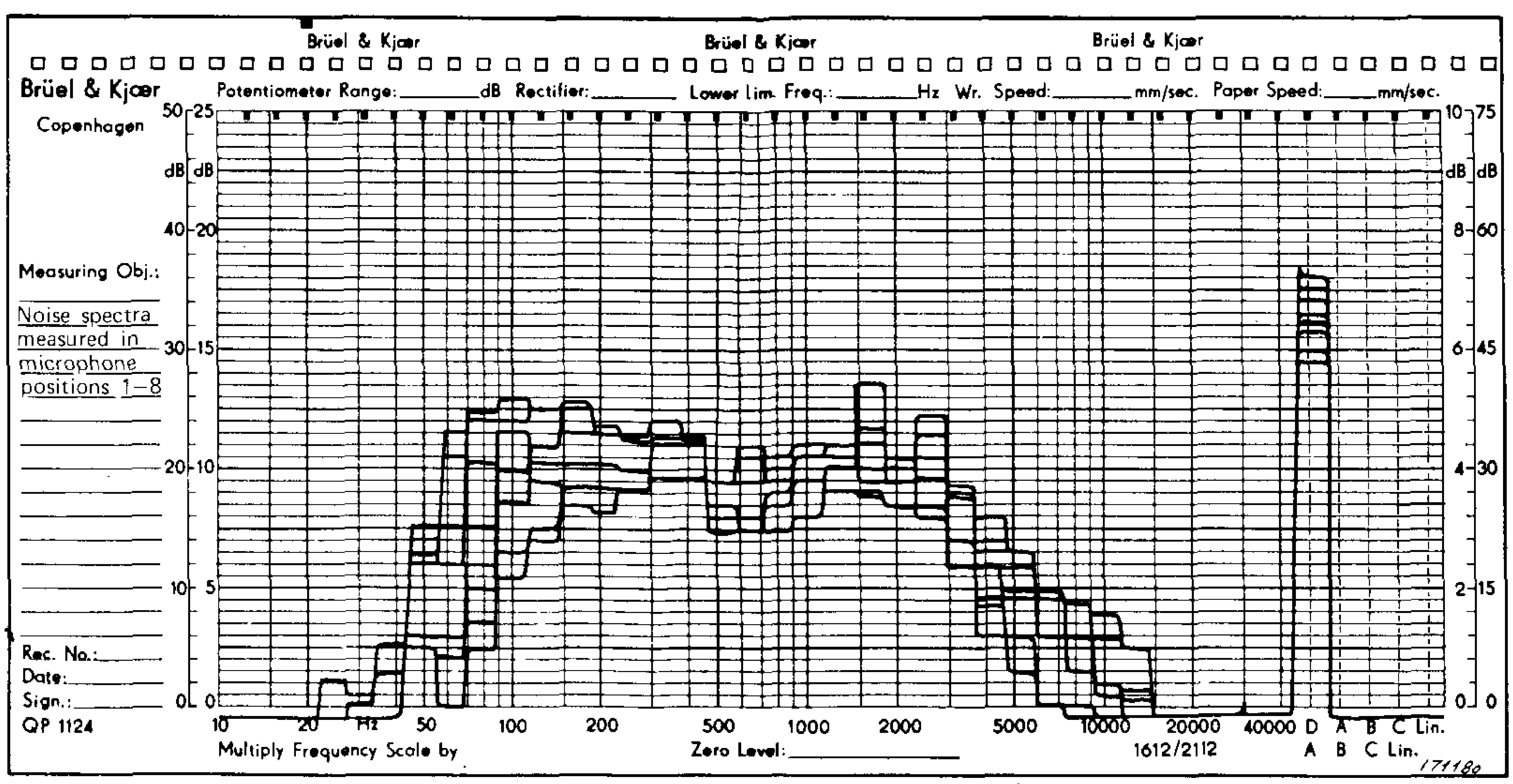


Fig.3. Channel selector system in the arrangement for the measurement of the average sound spectra

The microphone signals are transmitted successively to the Real-Time Analyzer Type 3347, the scanning rate being determined by the frequency of an external signal generator. The Analyzer Type 3347 provides an instantaneous frequency analysis in 30 parallel, third octave bands. By using a suitable time constant mode of the analyzer and a suitable signal channel advance frequency, the system provides a close approximation to the spatially averaged sound spectra in real time.

The spectra measured in each particular microphone position are shown in Fig.4a, while Fig.4b shows the directly measured average spectrum.



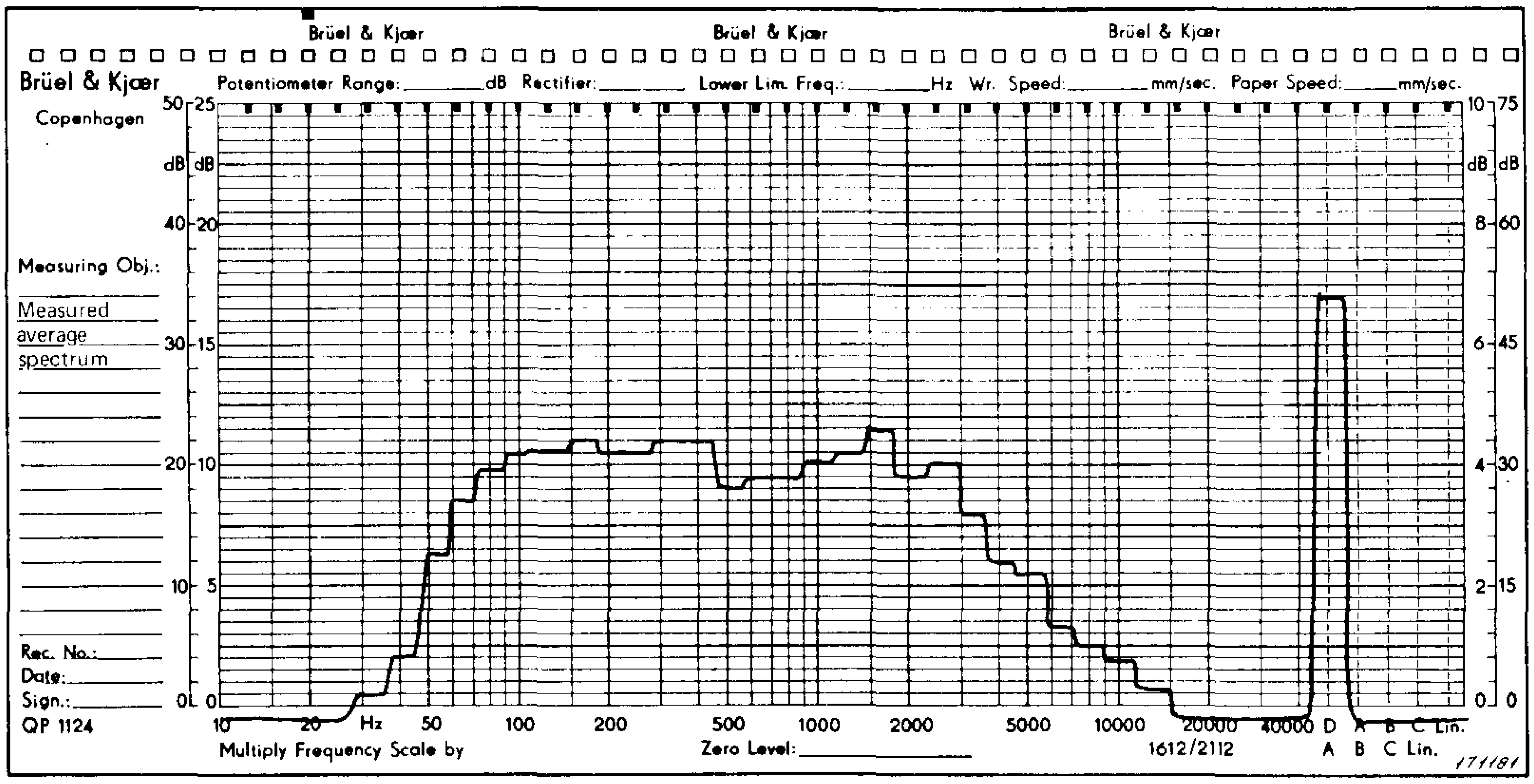


Fig.4. The signal spectra measured in all microphone positions (a) and the average spectrum (b)

Very good agreement between the measured and the calculated spatially averaged spectra was obtained. The deviations in the frequency range of interest (i.e. 100 Hz to 10 kHz) were less than 0.5 dB.

Automatic Sound Power Evaluation

The arrangement with the channel selector system connected to the Real-Time Analyzer Type 3347 and the Computer Type 7501 (see Fig.5) can be advantageously used for the automatic evaluation of the sound power radiated by a machine.

Having a suitable computer program, the arrangement provides automatic measurement and frequency analysis of the noise in each microphone

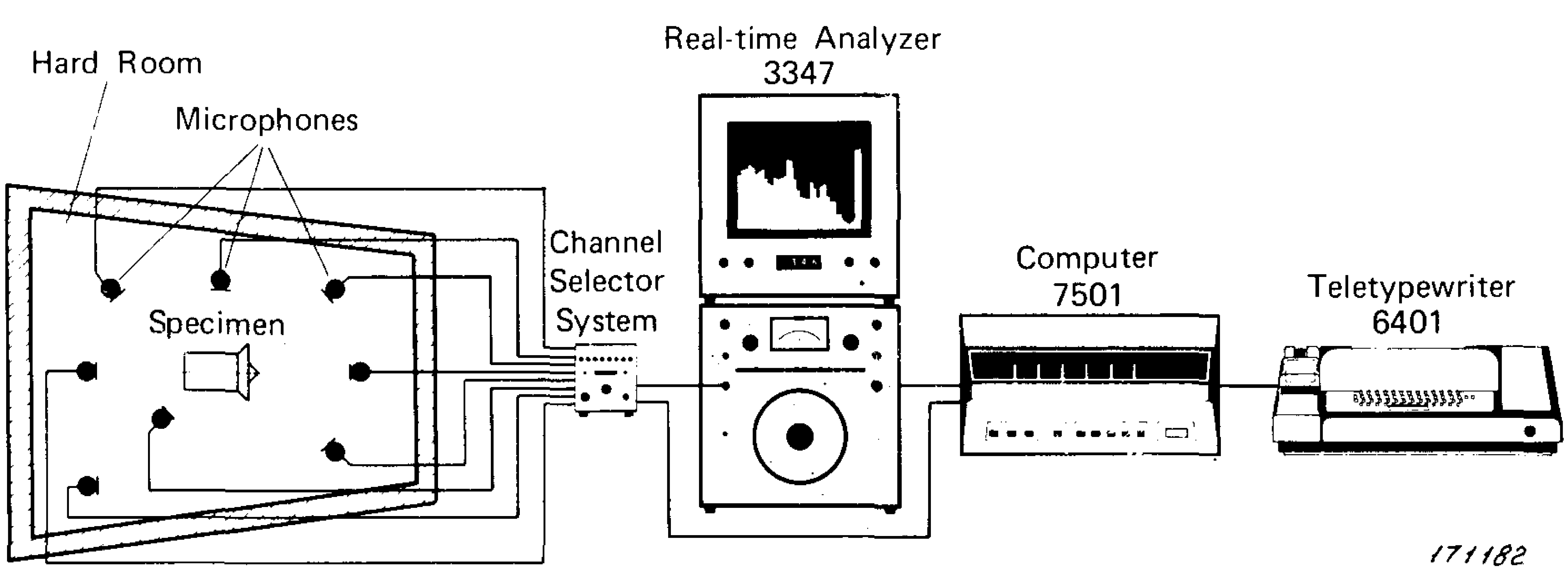


Fig.5. Use of the channel selector system for the automatic evaluation of sound power

position. Processing the data, computing the sound power in third octave bands and recording the results on peripheral devices (e.g. a teletypewriter, a puncher etc.) take one to two minutes of automatic operation of the system. The computer software can include all constants and correction factors necessary to evaluate the sound power under different measuring conditions, e.g. from measurement in an anechoic chamber, a hard room or in a semi-reverberant room.

The above described arrangement was used for the automatic evaluation of sound power of an industrial vacuum cleaner and a rather large fan type ventilator. The results obtained by measurements on the vacuum cleaner placed on a hard floor in an anechoic chamber are shown in the previous article. For the second experiment a fan of diameter 0,55 m and length 1.5 m was placed on the floor of a hard room. The known acoustic room characteristics were inserted in the computer program and forty frequency spectra from the eight randomly situated microphones were used in the computation. The automatically evaluated sound power levels in third octave bands (125 Hz to 10 kHz) are shown in Fig.6.

CENTRE FREQUENCY HZ	125	160	200	250	315	400	500	630	800	1000
REVERBERATION TIME SEC	7.0	6.5	5.0	4.5	4.0	4.0	4.5	3.5	3.5	3.5
	1250	1600	2000	2500	3150	4000	5000	6300	8000	10,000
	3.2	3.2	2.8	2.8	2.8	2.6	2.2	2.4	2.5	2.5

READ-IN INTERVALS (SEC) = 1

12345678

SOUND POWER LEVELS IN ONE-THIRD OCTAVE BANDS

FREQUENCY CHANNEL NUMBER	SOUND POWER LEVEL	SOUND POWER MEASURED BY CONVENTIONAL INSTRUMENTS
REF.:	50.0 DB	DB
21	xxxxxxxxxxxxxxxxxx 68.2 DB	73.5
22	xxxxxxxxxxxxxxxxxxxxxxxx 73.6 DB	77
23	xxxxxxxxxxxxxxxxxxxxxxxxxxxxx 79.0 DB	81
24	xxxxxxxxxxxxxxxxxxxxxxxxx 75.4 DB	77
25	xxxxxxxxxxxxxxxxxxxxxxxxxxx 76.8 DB	78
26	xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx 81.2 DB	82
27	xxxxxxxxxxxxxxxxxxxxxxxxxx 75.8 DB	77
28	xxxxxxxxxxxxxxxxxxxxxxxxxxxx 77.8 DB	78.5
29	xxxxxxxxxxxxxxxxxxxxxxxxxx 76.0 DB	77
30	xxxxxxxxxxxxxxxxxxxxxxxxx 74.9 DB	76
31	xxxxxxxxxxxxxxxxxxxxxxxxx 75.1 DB	76.5
32	xxxxxxxxxxxxxxxxxxxxxxxx 73.8 DB	74.5
33	xxxxxxxxxxxxxxxxxxxxxxx 73.4 DB	74
34	xxxxxxxxxxxxxxxxxxxxxx 71.8 DB	73
35	xxxxxxxxxxxxxxxxxxxx 69.8 DB	71.5
36	xxxxxxxxxxxxxxxxxxxx 69.7 DB	71
37	xxxxxxxxxxxxxxxxxxx 68.7 DB	70
38	xxxxxxxxxxxxxxxx 65.5 DB	66.5
39	xxxxxxxxx 58.7 DB	60.5
40	x 50.5 DB	52.5

Fig.6. Automatically evaluated sound power levels of a fan

Vibration Monitoring

For monitoring and control of vibrations on complicated structures (e.g. heavy machinery, constructions, buildings, transport vehicles etc.) it is necessary to measure vibrations at several positions, and specially where they are likely to be critical. The channel selector system permits up to 8 (or more) vibration pick-ups (accelerometers) to be connected, and, once mounted, measurements can be taken without interrupting the operation of the machine. The arrangement shown in Fig.7 provides quick visual control of the vibration spectra at different points. With the aid of a plain plexiglass screen, (mounted in front of the CRT display screen of the Analyzer Type 3347) marked with preselected tolerance limits, the visual display may be used for real time checks on the operational conditions of the machine.

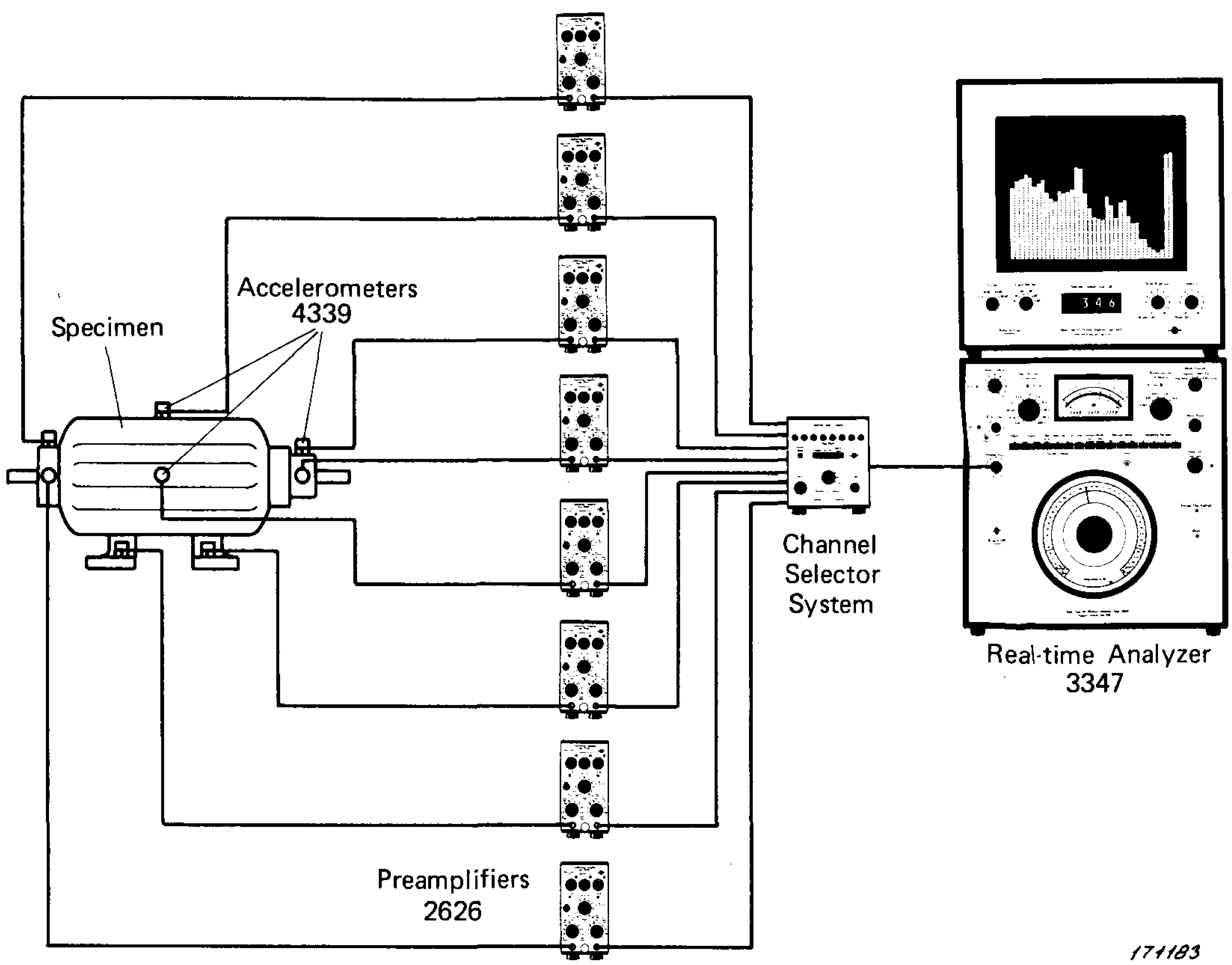


Fig.7. Channel selector system as a part of a vibration measurement and control set-up

Production Quality Control

Noise and vibration tests play an important role in production line quality control. The arrangements employing the channel selector system(s) can advantageously be used:

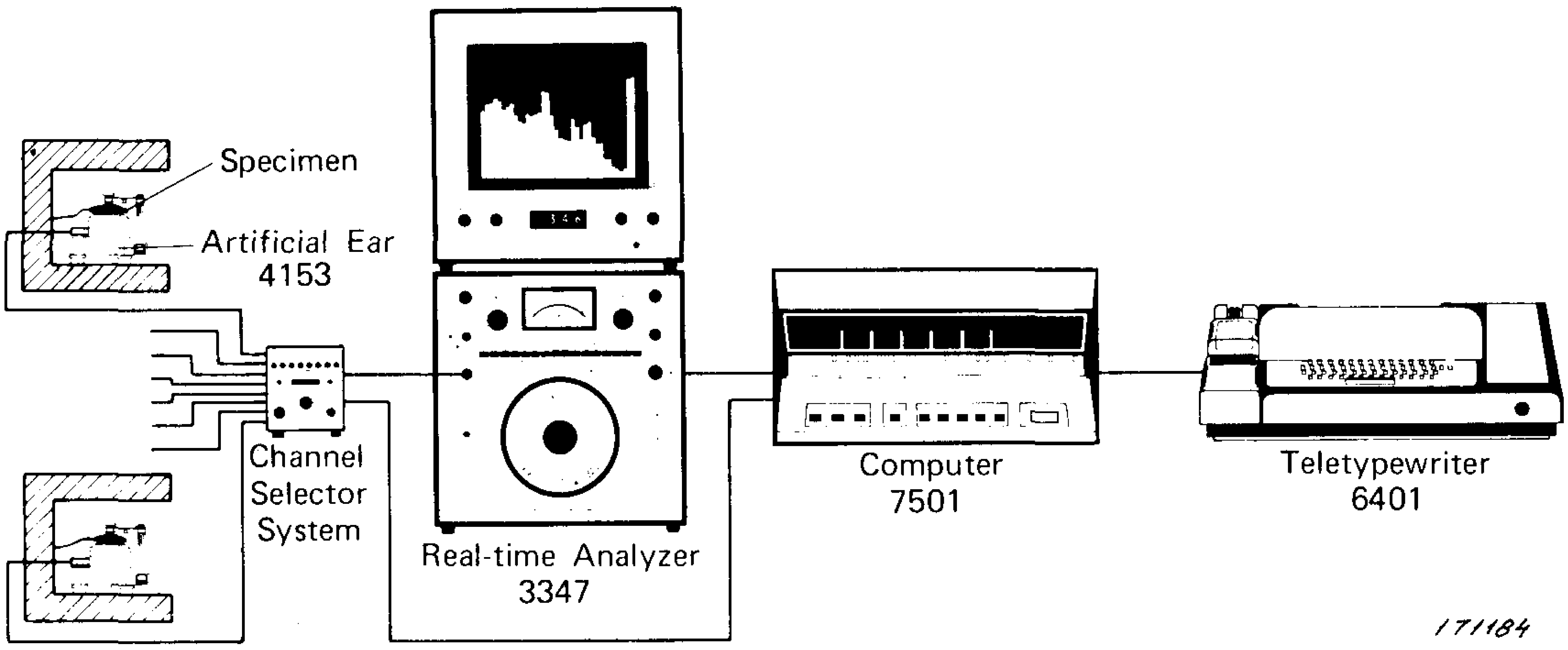


Fig.8. Channel selector system in the arrangement for production quality control

1) to check that the radiated noise and/or vibration levels do not exceed acceptable limits
2) to classify the products according to noise and/or vibration levels
3) to detect which part of the tested specimen might be faulty.

As a special example the computer-controlled system for production classification of earphones is shown in Fig.8.

The experimental set-up was used for very fast automatic classification and sorting of quality earphones into 3 classes. The tested earphones were classified with respect to the overall distortion in the frequency range from 100 Hz to 5 kHz. The classes were determined by the distortion less than 1%, over 1% and less than 3% and above 3%. The output data in terms of classification against identification number of the tested object and test results were printed out on a teletypewriter. Because the complete testing and classification of one earphone takes only 4—4.5 secs, the test system seems to be ideally suited for fast serial work.

Vibration Testing with Multipoint Control

A very important part of an automatic sinusoidal and/or random vibration test system is the feedback loop containing the control accelerometer. By testing large non-rigid specimens it is, however, difficult to find a proper single control point.

In these systems more control pick-ups (accelerometers) should be selected in a time sequence by means of a time-division multiplexer. With the aid of averaging devices a true average control signal is built up. With the proper

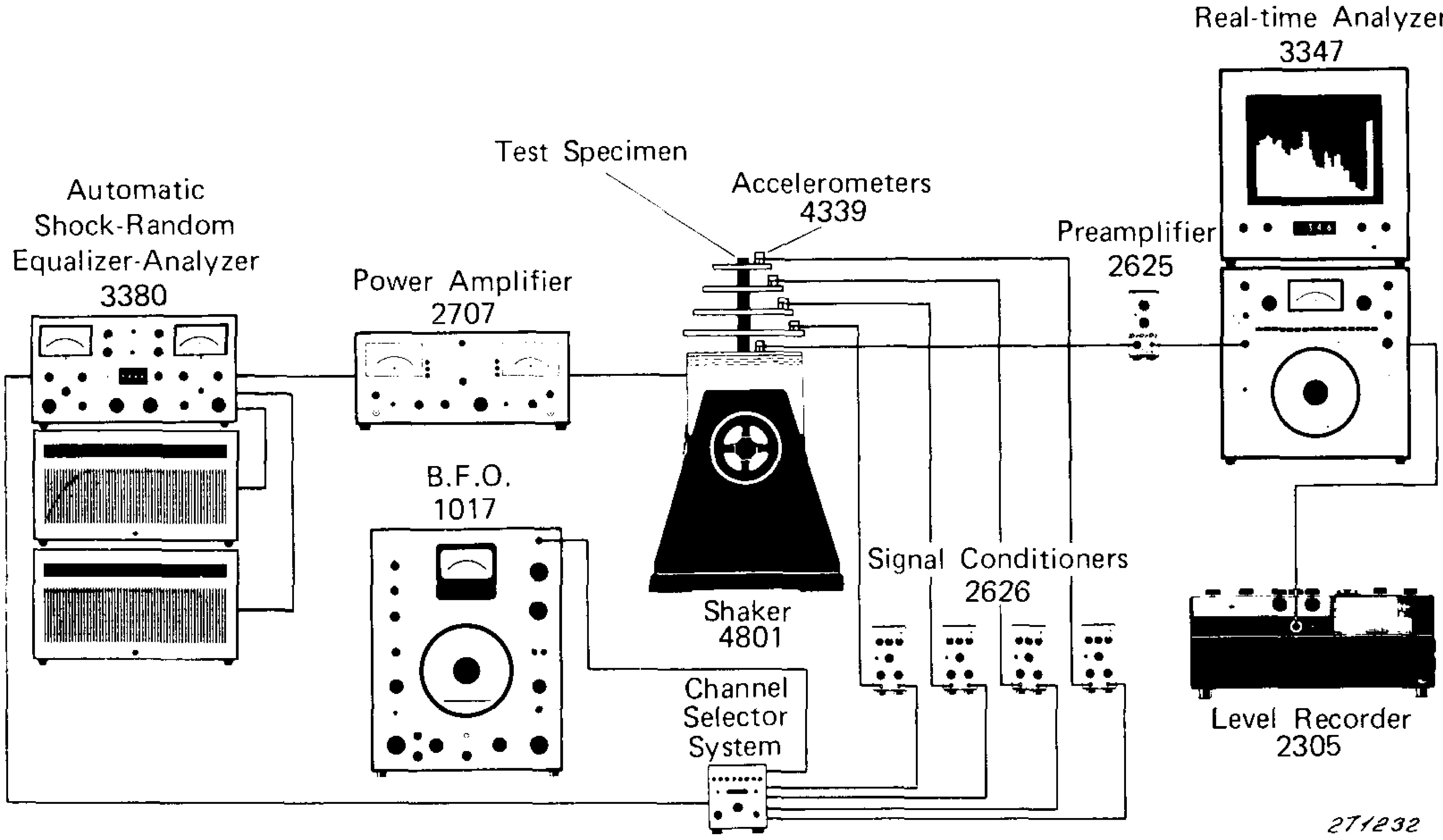


Fig.9. Set-up for average wide-band random vibration testing

selection of the control loop parameters (i.e. the advance frequency of the multiplexer, the time constants of the averagers and the gain of the AGC circuits) the actual controlling signal is proportional to the average of the individual control signals' magnitudes. It is furthermore independent of the relative phases of the individual control signals, and, when random signals are used, the resulting spectral density is very close to the average of the spectral densities of the individual signals.

An example of an arrangement, designed for evaluating a multipoint control system is shown in Fig.9.

In the control loop the channel selector system activated by an external generator is used. The smoothing and AGC circuits are included in each narrow-band channel of an Automatic Shock-Random Equalizer-Analyzer Type 3380, covering the frequency range from 20 Hz to 2 kHz (in 120 channels).

To check its correct operation, the system was excited by a wide band random signal (white noise) and 4 control accelerometers were connected to the channel selector. As indicating instruments the Real-Time Analyzer Type 3347 with the Level Recorder Type 2305 were used.

As an example, documenting very good agreement between the measured and the calculated average spectral density of the control signal, the curves

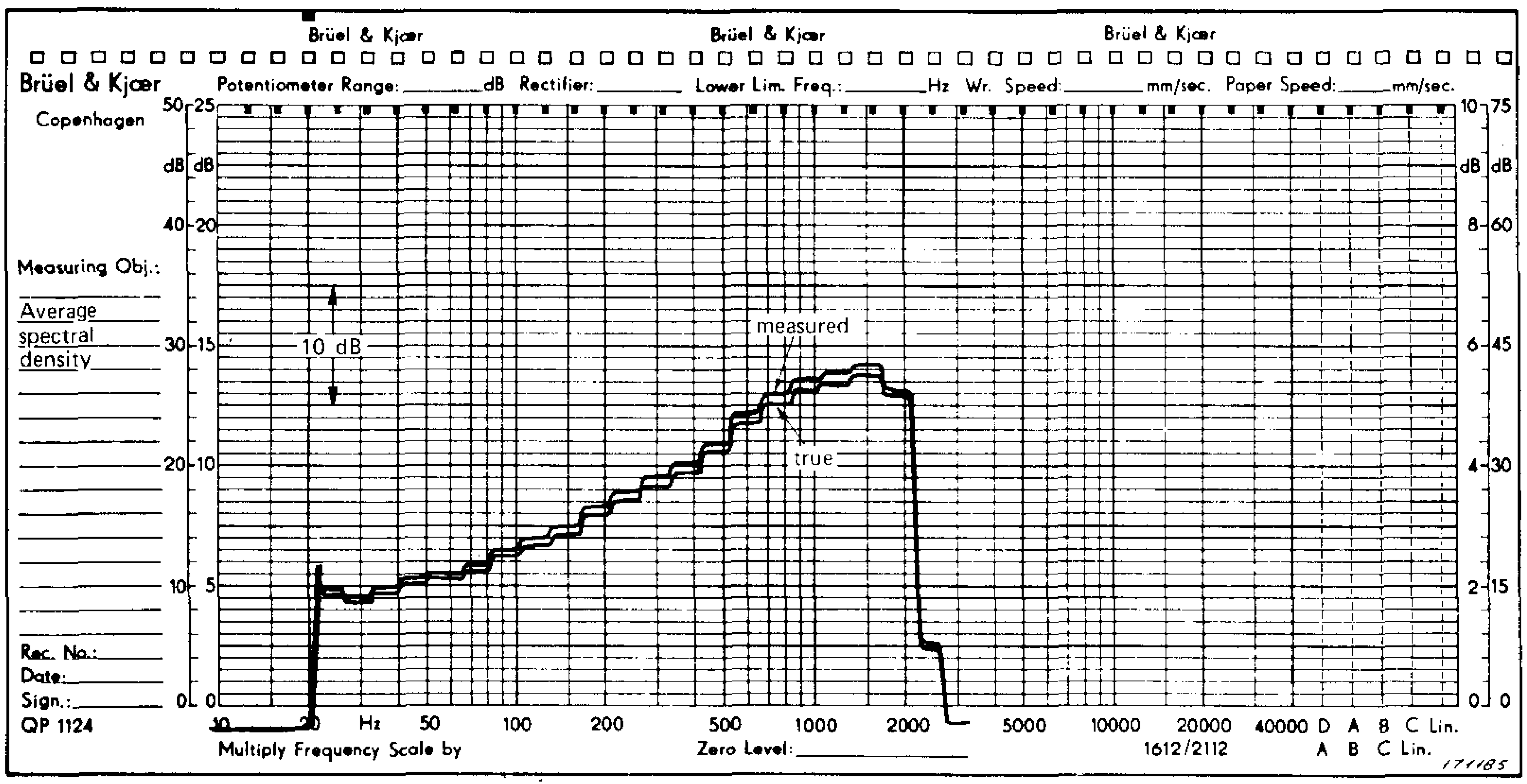


Fig.10. Measured and calculated average spectral densities of the control signal in the set-up in Fig.9

in Fig.10 are shown. In this particular experiment the level in one channel was 10 dB lower than in the remainding 3 channels.

Other Applications

The examples described by no means cover the possible applications of the channel selector system. The system could be used also for noise monitoring (in the vicinity of airports and other noise exposed areas), for automatic evaluation of sound transmission loss of a wall placed between two test chambers, for automatic vibration control systems, for sonic fatigue tests (where assessment of the average sound pressure requires more than one microphone) and for many other special control and monitoring purposes.

Strain, flow, temperature, moisture, mechanical dimensions, stress etc. are typical quantities, whose measurement and control can also be facilitated and speeded-up by the described channel selector system.

References

1. B. FREDERIKSEN: 1/3 Octave Spectrum Readout of Impulse Measurements, B & K Techn. Rev. No. 1–1969.
2. T. USHER, Jr.: Average Control for Sinusoidal and Random Vibration Testing, JASA, Vol. 41, No. 4, 1967.

N.B. The experimental channel selector described in this article is available from Brüel & Kjær as Type 5619 on special request.

Brief Communications

The intention of this section in the B & K Technical Reviews is to cover more practical aspects of the use of Brüel & Kjær instruments. It is meant to be an "open forum" for communication between the readers of the Review and our development and application laboratories. We therefore invite you to contribute to this communication whenever you have solved a measurement problem that you think may be of general interest to users of B & K equipment. The only restriction to contributions is that they should be as short as possible and preferably no longer than 3 typewritten pages (A4).

Comments on the Article: "On the Frequency Analysis of Mechanical Shocks and Single Impulses"

In the article "On the Frequency Analysis of Mechanical Shocks and Single Impulses" by Broch and Olesen (Technical Review 3/1970) the Fourier Spectrum of the rectangular pulse is given as

$$A(f) = AT \frac{\operatorname{Sin}(\pi f t)}{\pi f t}$$

This is, of course, correct for a pulse which extends in time from $-\frac{T}{2}$ to $+\frac{T}{2}$, and the spectrum of a pulse extending from 0 to T (as in Figs.2 and 8 of the article) is, apart from the phase distribution, identical with A (f) above. One might therefore reasonably use A (f) in a simple derivation of the filter response.

This, however, leaves some uncertainty as to how the filter response is related in time to the excitation.

Now, the Fourier transform of a pulse extending from 0 to T is given by

$$A(f) \times e^{-j\pi f t}$$

and the filter response is then

$$2 \int_{f_1}^{f_2} A(f)\, e^{-j\pi f T}\, e^{j\pi f(t-t_L)}\, e^{j2\pi f_o t_L}\, df$$

where

$$f_1 = f_o - \frac{\Delta f}{2} \text{ and } f_2 = f_o + \frac{\Delta f}{2}$$

In the process of integration A (f) should be taken outside the integral but not $e^{-j\pi fT}$.

The result is

$$2\,A\,(f_o) \times \Delta f \times \frac{\operatorname{Sin}\left[\pi \Delta f\,(t - t_L')\right]}{\pi f\,(t - t_L)} \operatorname{Cos} 2\pi f_o\,(t - \frac{T}{2})$$

where

$$t_L' = t_L + {}^{T}/2$$

and the transmission time t_L is seen to be the interval between the envelope's peak and the centre of the pulse.

Professor L.B. D'Alton, M.E.C. Eng., F.I.E.E.
Dept. of Electrical Engineering,
The University College, Dublin

Author's Answer:
We are very grateful to Professor D'Alton for pointing out to us the error that has slipped into the Figs.2 and 8 in the article on Frequency Analysis.

The input pulse time functions (top of figures) should, of course, have extended from $-\frac{T}{2}$ to $+\frac{T}{2}$, instead of as shown from 0 to T, as all the mathematics in the article have considered the input pulse to be symmetrical around the chosen amplitude-axis (t = 0).

Jens T. Broch *Hans P. Olesen*

Errata
Technical Review No.1, 1970, page 34, line 13, "ISO Recommendation 177" should read "IEC Publication No. 179".

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(Continued from cover page 2)

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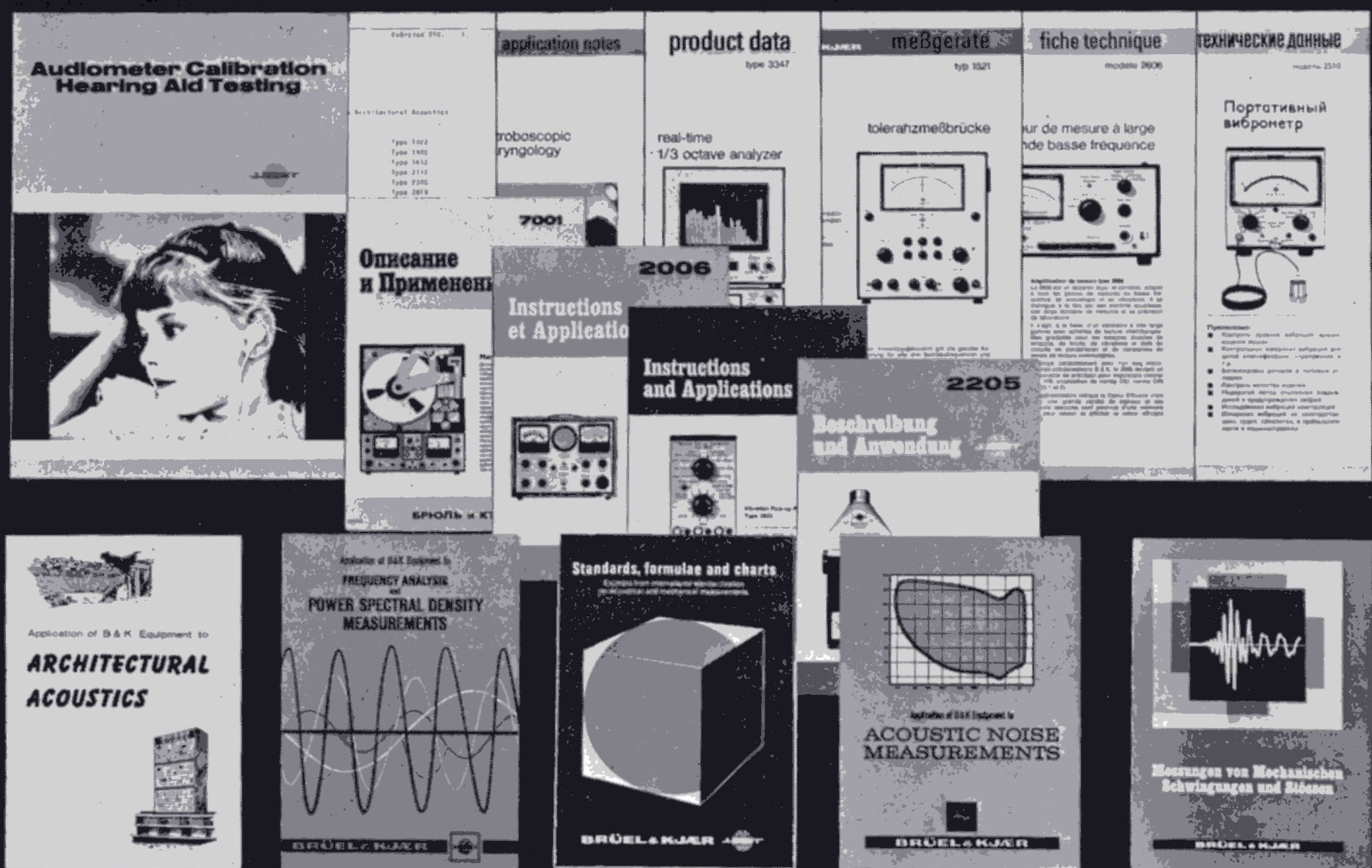

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