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TECHNICAL REVIEW

No. 1 — 1973

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Calibration of Hydrophones

by

Peter A. Levin, M. Sc.

ABSTRACT
As no single hydrophone calibration method is suitable over the full frequency range of interest 0,1 Hz to 200 kHz, separate methods are outlined for each of the frequency ranges 0,1 Hz – 1 Hz, 50 Hz – 4 kHz and 4 kHz – 200 kHz. The measuring set-ups required for each method are described, and the details to be taken into consideration for the design of a small indoor water tank are pointed out. The frequency response results are presented as well as the measurements of useful parameters such as the equivalent noise pressure and the impedance of the Hydrophone Type 8100.

SOMMAIRE
Il n'existe aucune méthode d'étalonnage des hydrophones qu'on puisse utiliser sur toute la gamme des fréquences utiles (0,1 Hz – 200 kHz). L'article décrit les différentes méthodes utilisées sur chacune des gammes 0,1 Hz – 1 Hz, 50 Hz – 4 kHz et 4 kHz – 200 kHz. Les systèmes de mesure nécessaires pour mettre en oeuvre chaque méthode sont décrits et on met en lumière les détails à prendre en compte pour réaliser un petit réservoir d'eau d'intérieur. Les résultats relatifs à la réponse en fréquence sont donnés ainsi que les mesures de paramètres intéressants tels que la pression équivalente de bruit et l'impédance de l'Hydrophone Type 8100.

ZUSAMMENFASSUNG
Da keine Methode der Hydrophonkalibrierung für den ganzen interessierenden Frequenzbereich 0,1 Hz – 200 kHz anwendbar ist, werden unterschiedliche Methoden für die Frequenzbereiche 0,1 Hz – 1 Hz, 50 Hz – 4 kHz und 4 kHz – 200 kHz angegeben. Die für die einzelnen Methoden benötigten Meßanordnungen werden beschrieben; außerdem werden Details, die beim Entwurf eines kleinen Wassertanks zu berücksichtigen sind, aufgezeigt. Die Ergebnisse der Frequenzgangmessungen sowie weitere sinnvolle Meßgrößen, wie der equivalente Schalldruck des Eigenrauschens und die Impedanz des Hydrophons 8100, werden mitgeteilt.

Introduction
The most significant parameter specifying a hydrophone is its free field sensitivity S_H expressed as a function of frequency. Mathematically it is defined as the ratio.

$$S_H = \frac{e}{p}$$

where e is the electrical output voltage and
p is the sonic pressure acting on the hydrophone

the units being Volts per N/m^2 or Volts per μpascal. The calibration of the sensitivity is carried out either by using the comparison method or reciprocity method both of which require a free field environment. Natural sites such as lakes, ponds and reservoirs which provide free field environment have all been used for calibration purposes. However, care has to be taken to ensure that the ambient noise is low, which is not always possible on account of ship traffic, rain or wave motion. Also currents, temperature gradients, marine life, bubbles and pollutants make it difficult to achieve a free field environment. These problems on the other hand, can be overcome in an anechoic water tank with absorbent walls and bottom surface. However, if continuous waves are used the dimensions required of the water tank increase as the frequency decreases, making the anechoic water tank an impractical economic proposition. To overcome this limitation, the pulse technique can be utilized to facilitate the calibration of the hydrophone in a relatively small water tank of simple construction.

Water Tank Design

When designing the water tank, due considerations should be given to the following parameters:

Pulse duration (τ ms)
Distance between transducers
The repetition rate of the pulses

Pulse duration

Fig.1 shows a water tank of dimensions h x L x b with the two hydrophones at a distance d apart from each other. One bydrophone acts as a transmitter while the other is the receiver. From the figure it can be seen that the pulse length should be short enough so that measurements are not disturbed by reflected pulses (echoes) arriving at the receiver before the termination of the direct signal. This sets the following limitations to the pulse duration.

$$\tau \leqslant \frac{2d}{c} \quad \text{(reflection between transducers)} \quad (1)$$

$$\tau \leqslant \frac{L-d}{c} \quad \text{(reflection from wall)} \quad (2)$$

$$\tau \leqslant \frac{\sqrt{h2 + d2} - d}{c} \quad \text{(reflection from upper or bottom surface} \quad (3)$$

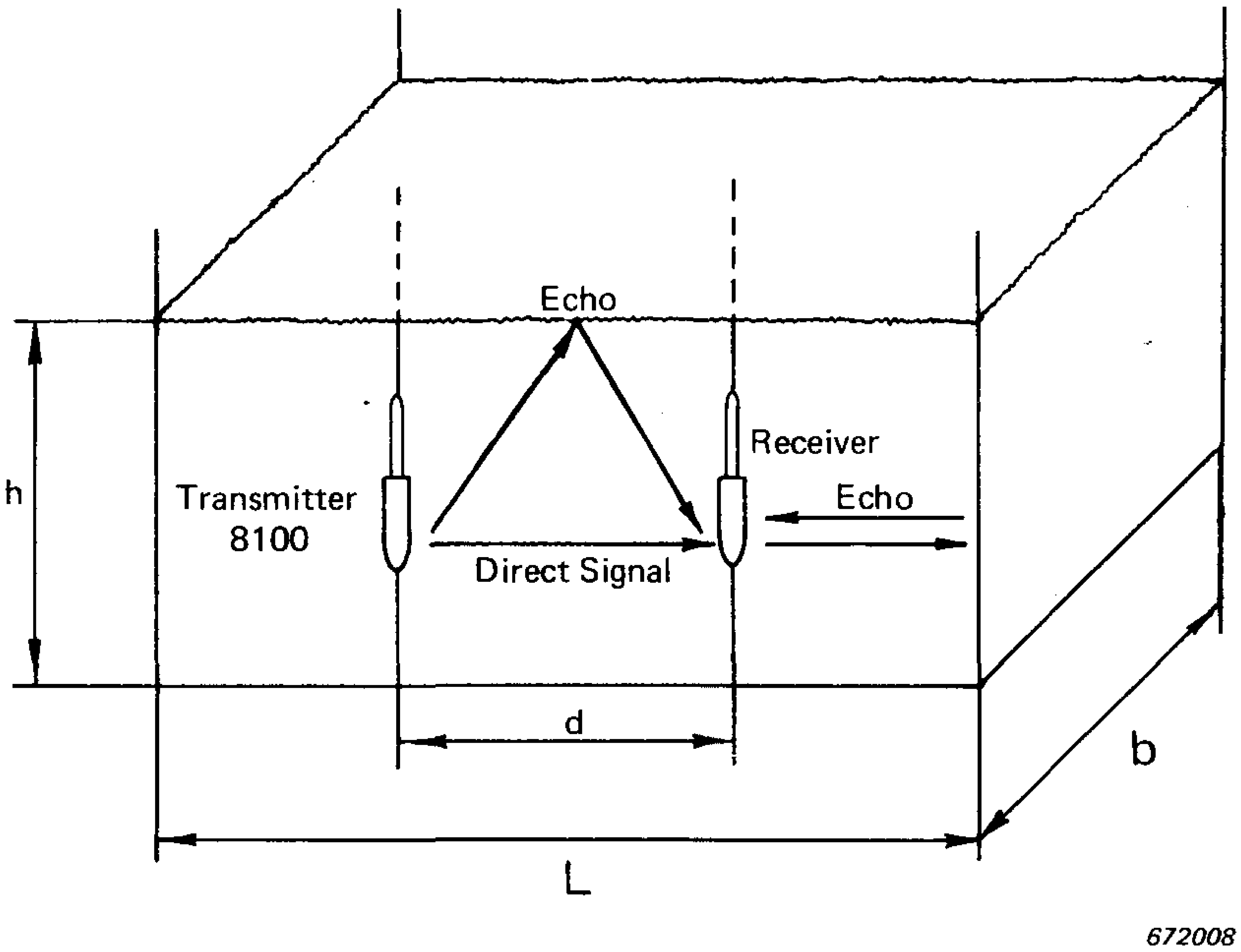

Fig. 1. Sketch of water tank dimensions

c is the velocity of sound in water (1430 m/s nominally)

The pulse length would also be governed by the lower limiting frequency at which the calibration is to be carried out. If the low frequency limit of interest is 3 kHz, the pulse length

$$\tau \geqslant 0{,}33 \text{ ms} \tag{4}$$

if there is to be at least one cycle of the signal. However, to obtain a steady state voltage amplitude across the receiver the minimum pulse length in practice should be at least twice as large i.e.

$$\tau \geqslant 0{,}7 \text{ ms} \tag{5}$$

on account of the transient response of the two transducers.

Distance between transducers

To ensure that the receiver is placed in the far field the distance between the transducers should be

$$d \geqslant \frac{a^2}{\lambda} \tag{6}$$

where a is the largest dimension of the transmitter (projector) and λ is the shortest wavelength used. Taking the value of a to be 10 cm for the Brüel &

Kjær Hydrophone Type 8100 and the upper limiting frequency to be 100 kHz.

$$d \geqslant 60 \text{ cm}$$

The mathematical conditions given by equations (1) – (6) are drawn in Fig.2 for the tank described in the following section and illustrate the restrictions placed on the allowable "working area" shown with hatched lines. The conditions (1) – (3) are governed by the dimensions of the water tank while conditions (4) – (6) are determined by the lower and upper frequency limit and the dimensions of the transducer. Thus the choice of the lower limiting frequency lays restriction on the minimal size of the water tank required.

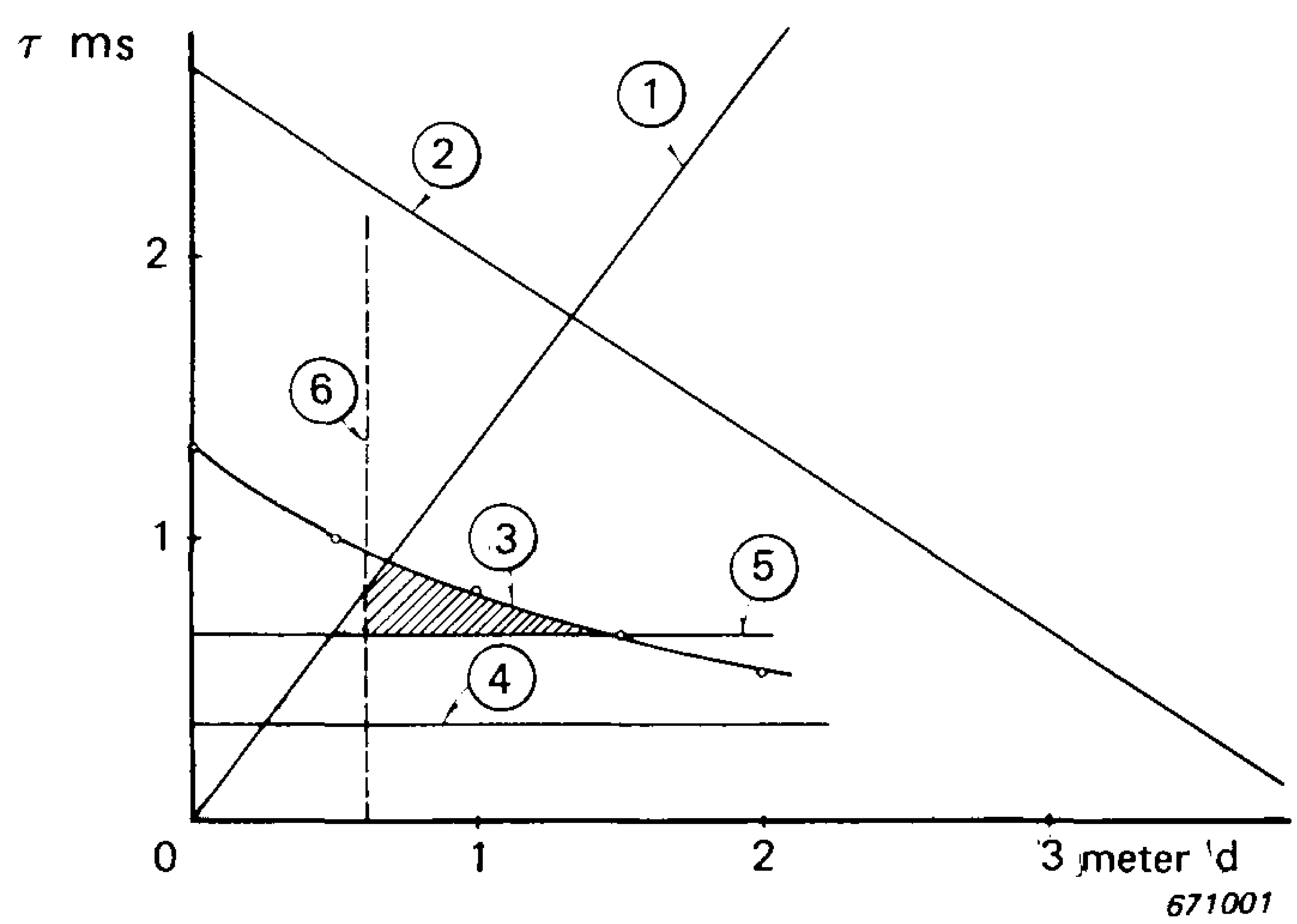

Fig.2. Limitations to pulse duration and transducer spacing for a given water tank

Repetition Rate of Pulses

The choice of the repetition rate of the pulses is governed mainly by the reverberation time (which depends on frequency, the size and shape of the water tank and the absorption coefficient of the tanks inner surfaces). With the aid of an oscilloscope, the repetition rate can be conveniently adjusted to allow sufficient time for reverberant decay.

The above considerations were taken into account when designing the water tank shown in Fig.3 together with its associated mechanisms. The water tank dimensions were decided upon to be 4 x 3 x 2 meters specifying the lower limiting frequency to be 3 kHz. The transducers were suspended from carriages which could be moved along rails on the platform, facilitating adjustment of the distance between transducers and their depth in water.

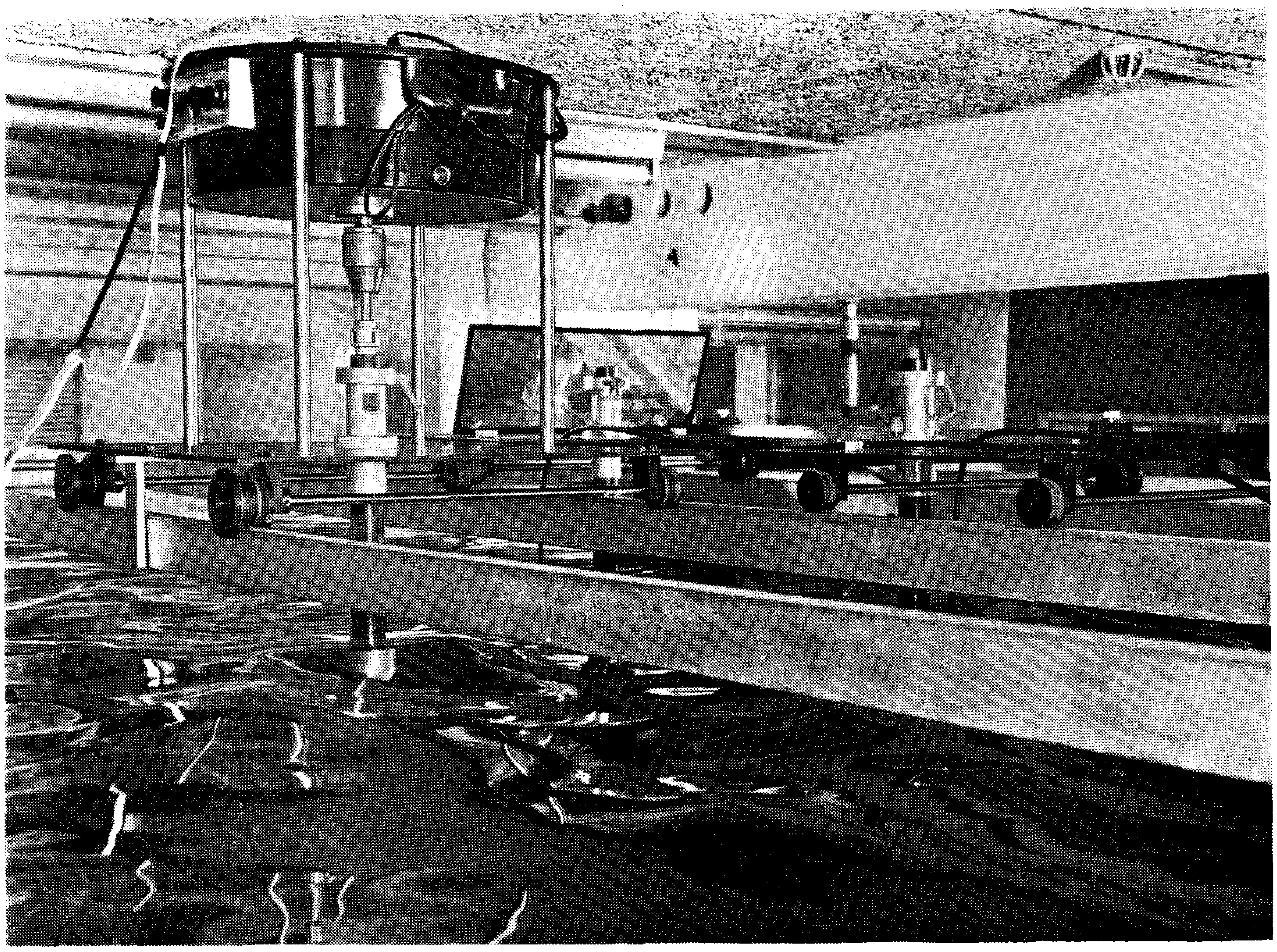

Fig.3. Photograph of the water tank at the Brüel & Kjær development laboratories

For recording directivity patterns the transducers can not only be rotated about a vertical axis, but can also be oriented in the hanger to obtain patterns in different planes.

Calibration Methods

Since no single hydrophone calibration method is suitable over the wide frequency range 0,1 Hz – 100 kHz, three different methods have been used for the following frequency sub-ranges.

3 kHz – 100 kHz
50 Hz – 4 kHz
0,1 Hz – 1 Hz

Calibration in the Frequency Range 3 kHz – 100 kHz

For the above frequency range the "Calibrated Projector Method" is used which requires a standard hydrophone as the transmitter and the unknown hydrophone as the receiver whose sensitivity is to be determined. The measurement set-up utilized is shown in Fig.4. Since the water tank is a

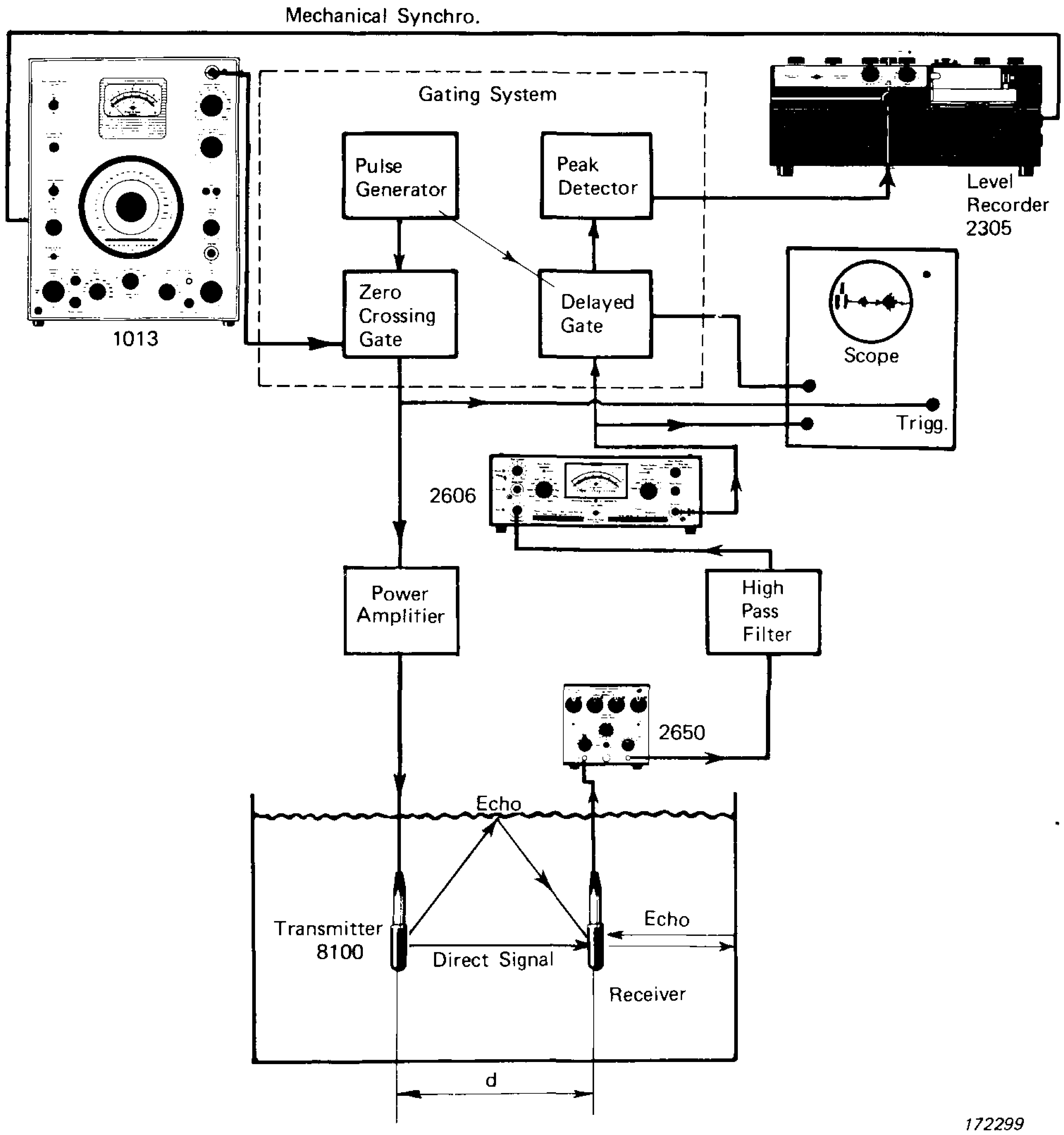

Fig.4. The measuring arrangement for pulse calibration

bounded enclosure reflected pulses (echoes) cannot be avoided. However, the use of the gating system, makes it possible to measure the open circuit voltage developed across the hydrophone on account of the direct pulse only, while blocking the reflected pulses. It can be seen from the measurement set-up that the gating system has two sections, the transmitting and the receiving.

The signal from the Beat Frequency Oscillator Type 1013 is fed into the transmitting side of the gating system which consists of a square pulse generator having the possibility of adjusting the pulse length and the repetition rate of the pulses. With the aid of the zero crossing gate, the output from the gating system would appear as tone bursts starting and stopping at zero voltage cross-over time. The tone bursts are fed into the transmitting hydrophone via a low distortion power amplifier. The signal

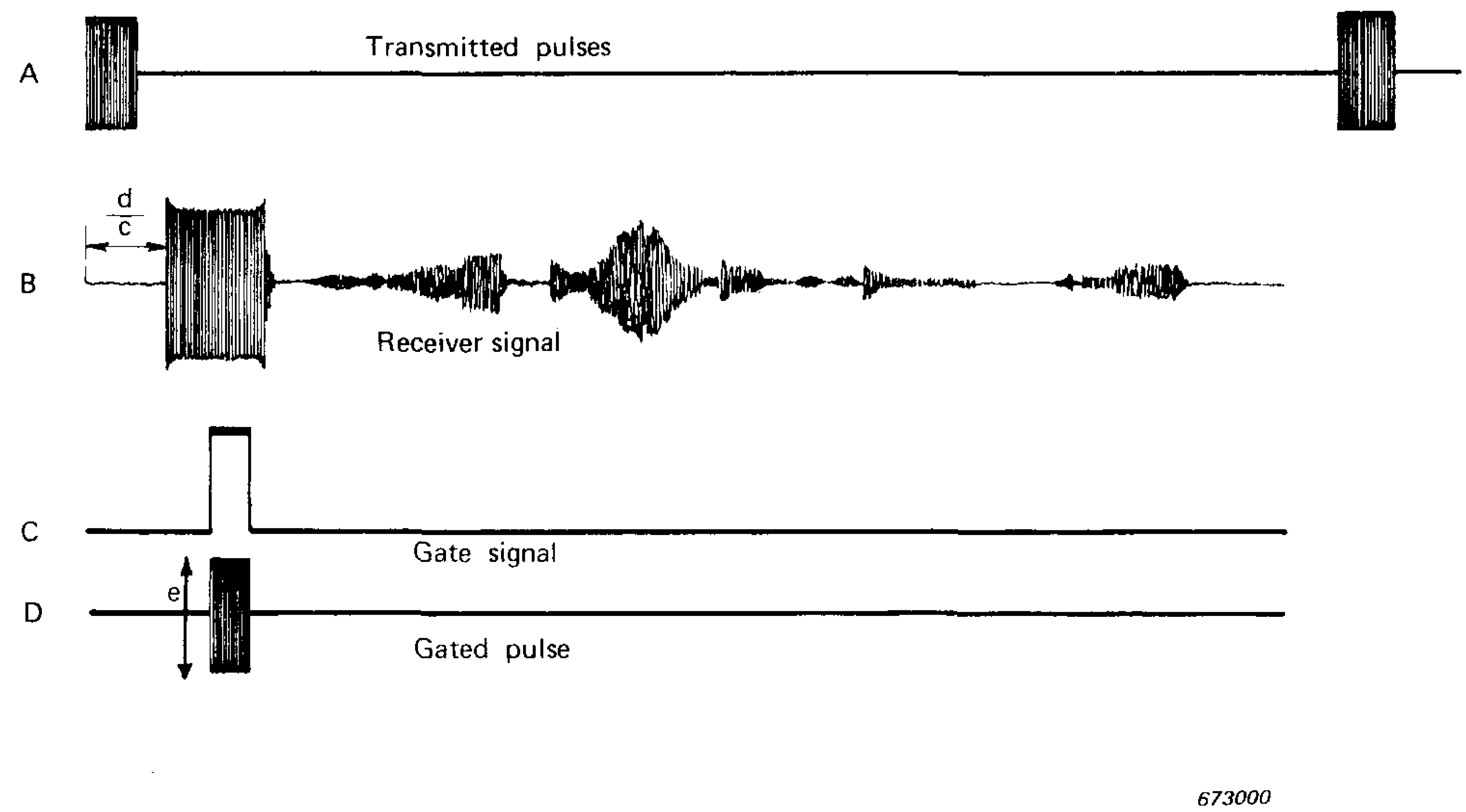

Fig.5. Signal waveforms measured at 70 kHz in the measuring arrangement of Fig.4

from the receiver hydrophone is displayed on one channel of a dual beam oscilloscope after being conditioned and amplified in the Conditioning Amplifier Type 2650 (or 2626) and Measuring Amplifier Type 2606 respectively. (A high pass filter with cut-off frequency 2 kHz may be inserted before the Measuring Amplifier if components of line frequency and its harmonics are found to exist on account of grounding problems).

To avoid the reflecting pulses disturbing the measurements, the output from the Measuring Amplifier is fed into the receiving side of the gating system, called the measuring gate. The measuring gate is synchronized with the pulse generator with a delay approximately equal to the signal transmission time. A final adjustment of delay and gate width allows the measuring gate to deliver the steady state part of the received signal to the peak detector which feeds the Level Recorder Type 2305. Fig.5 shows the signals appearing at different stages in the measuring set-up.

The transmitter hydrophone is previously calibrated by the reciprocity method for which the voltage response is known. If the transmitting hydrophone generates S_v Pascals/volt at a distance of 1 meter and the voltage applied across the transmitter is E volts, the sound pressure squared generated at the unknown hydrophone at a distance d would be

$$p^2 = \frac{E^2 S_v^2}{d^2}$$ according to the inverse square law

If the voltage output from the receiver is e volts the sensitivity S_H (volts per Pascal) of the receiver is given by

$$S_H{}^2 = \frac{e^2\ d^2}{E^2\ S_v{}^2}$$

Taking logarithms of both sides of the equation we obtain

$$20\ \log_{10} S_H = -20\ \log_{10} E - 20\ \log_{10}\ S_v + 20\ \log_{10} d + 20\ \log_{10} e$$

Thus by applying a constant voltage E to the transmitter and recording e, on the level recorder as the frequency is scanned, the value of S_H in dB can be calculated over the frequency range of interest.

By suspending the receiver from the Turntable Type 3921 polar directivity patterns can be plotted directly, since the turntable can be synchronized with the level recorder.

Fig.6 shows a typical receiving sensitivity curve for the Hydrophone Type 8100 obtained by reciprocity calibration method. The frequency response as can be seen is practically flat (within ±2 dB) up to 50 kHz. The radial resonance mode is observed to be at 60 kHz, while the longitudinal one appears at about 150 kHz.

Since the hydrophone is a stiffness controlled piezoelectric transducer, a constant voltage applied to it would result in a constant displacement response of the hydrophone. Also the acoustic sound pressure generated by the transmitter is proportional to the acceleration. Therefore, as the frequency is increased (keeping the input voltage constant) the sound pressure level would increase at a rate of 12 dB per octave since for constant

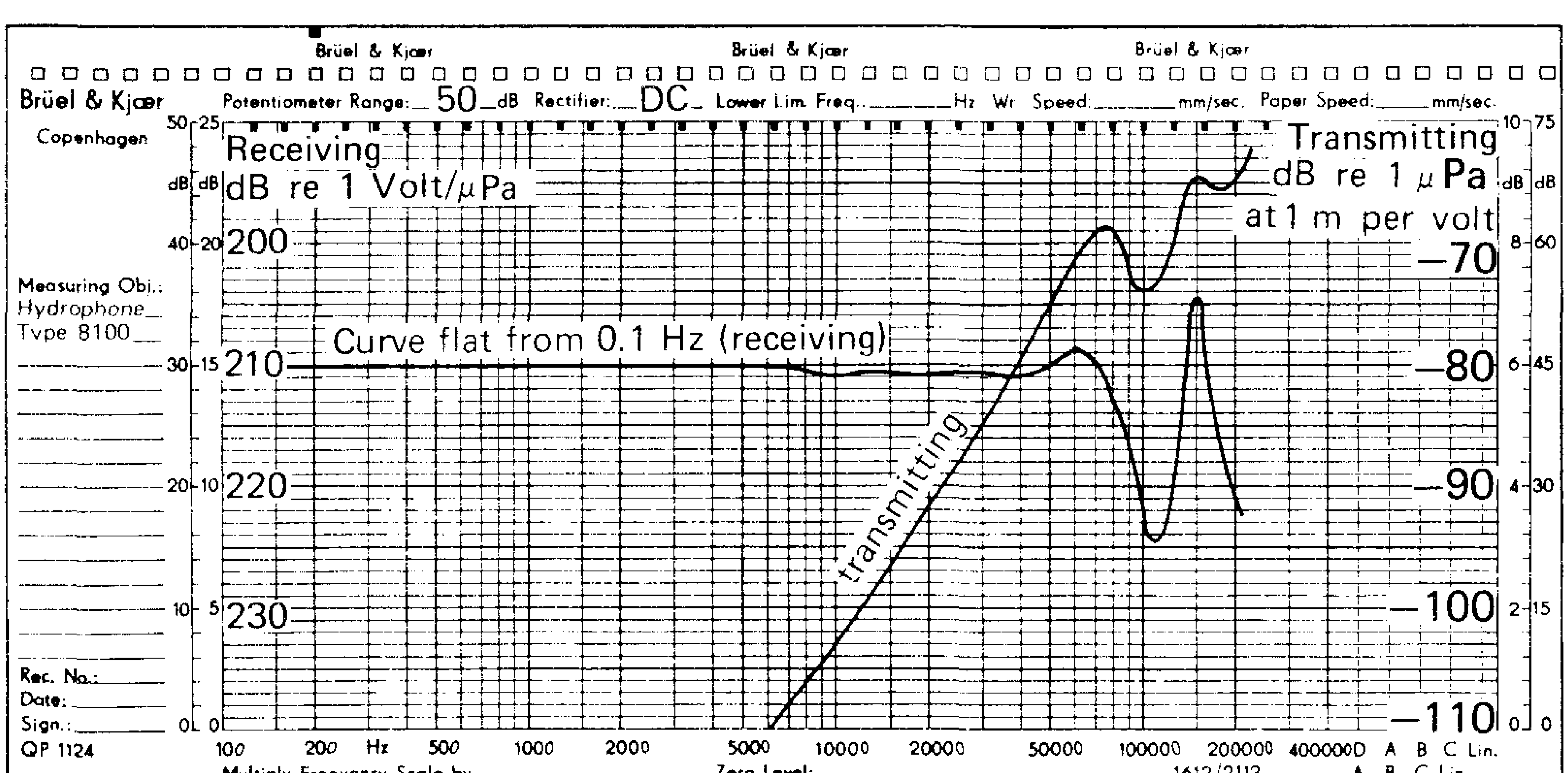

Fig.6. Typical receiving and transmitting response of Hydrophone Type 8100 in water

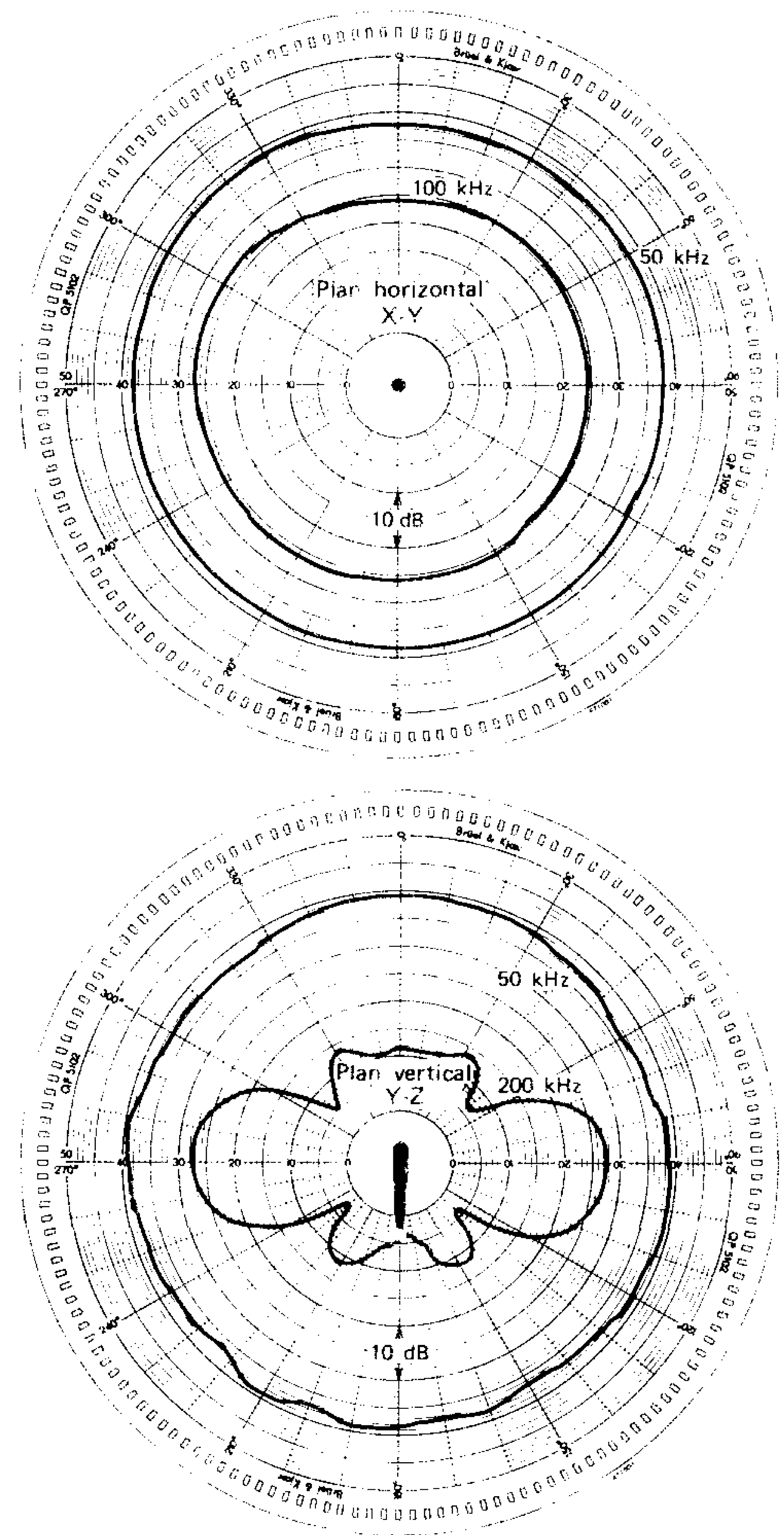

Fig.7. Typical polar directivity patterns in water

displacement, acceleration increases at the same rate. The transmitting voltage response shown also in Fig.6 is a curve of slope 12 dB/octave in the frequency range where the dimensions of the transmitter are small in comparison with a wavelength in water (i.e. below the first resonance). (A constant current applied to the transmitter would give a constant velocity response of the hydrophone resulting in a curve of slope 6 dB per octave).

The directivity patterns in the XY and XZ planes are shown in Fig.7 and are within ±2 dB up to 100 kHz and 50 kHz respectively. It should be pointed out, however, that the support fixture may affect the directional characteristics, especially if it has any large reflecting surfaces.

Calibration at Sonic Frequencies

If the acoustical impedance of the hydrophone is high enough so that its radiation impedance can be neglected and if measurements are performed in

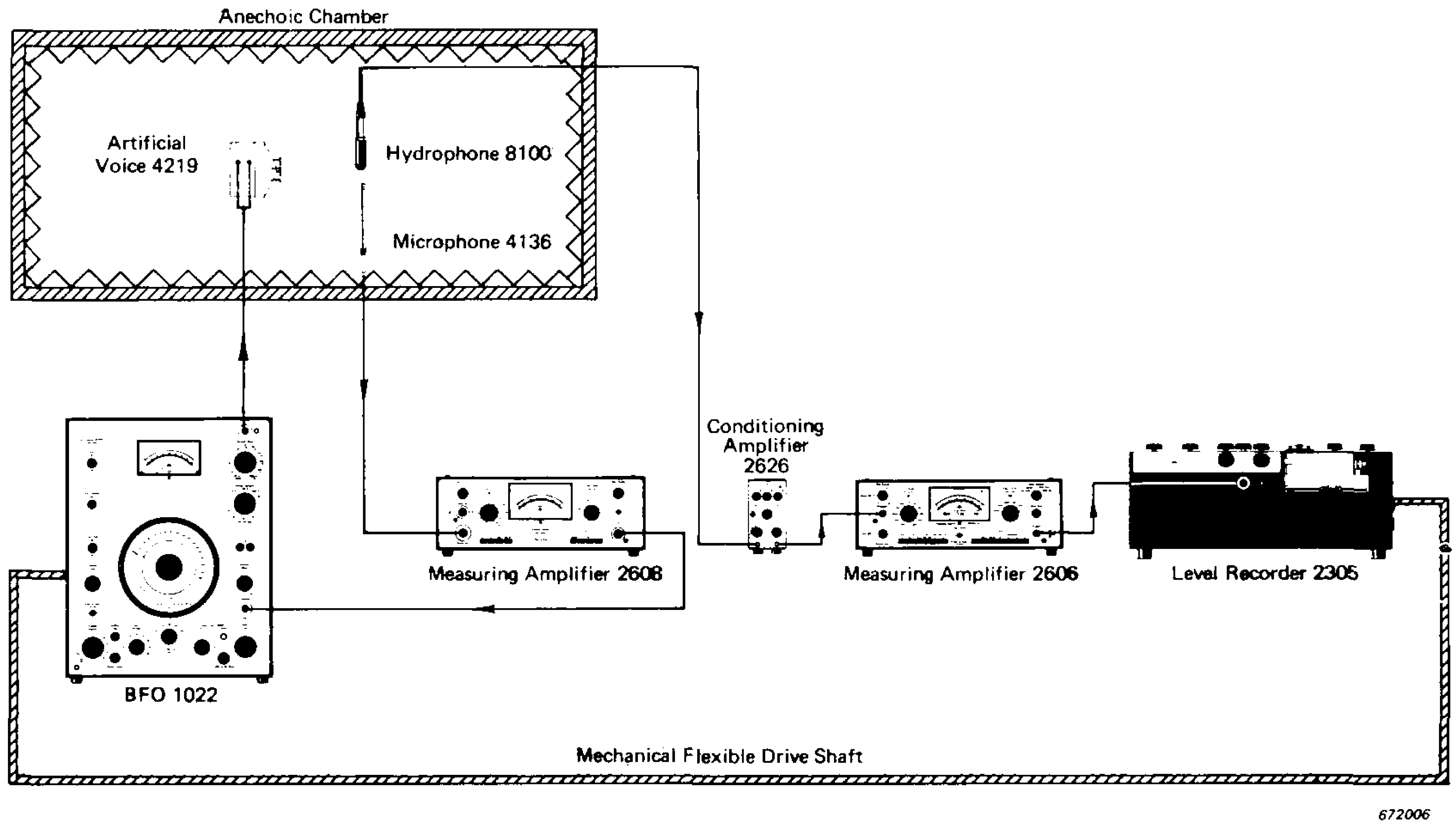

Fig.8. Measuring arrangement for calibration in anechoic chamber

the frequency range where diffraction phenomena can also be neglected, then the receiving sensitivity of the hydrophone would be the same in air as it is in water. With these assumptions hydrophones can be calibrated in an anechoic chamber in the frequency range 50 Hz – 4 kHz.

The measurement set-up used is shown in Fig.8 in which the artificial Voice Type 4219 is used as a constant sound pressure source. A regulating microphone Type 4136 is placed in front of it to maintain a constant sound

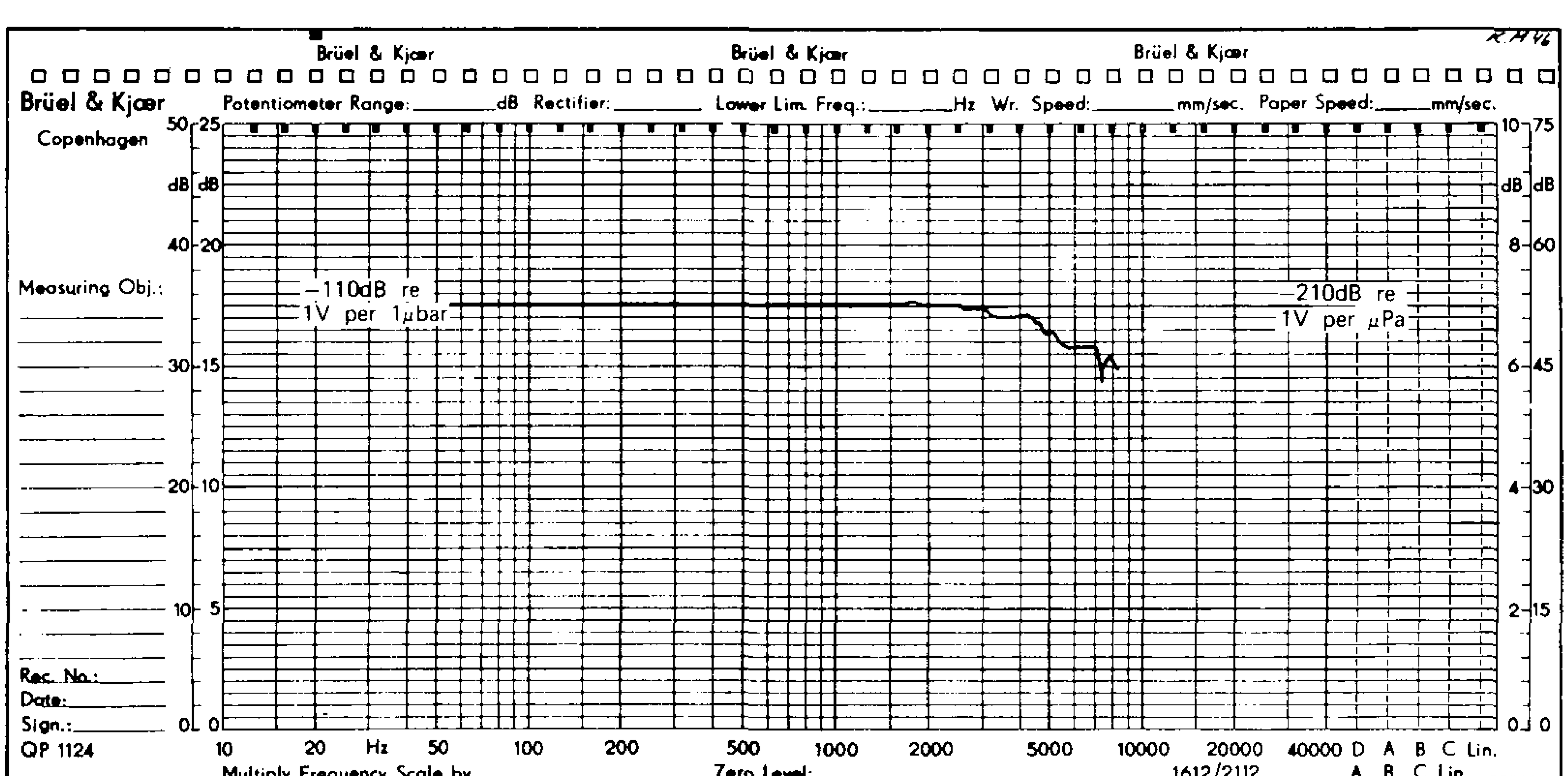

Fig.9. Typical receiving response of Hydrophone in air

pressure level acting on the hydrophone. The output from the hydrophone Type 8100 is fed to a Measuring Amplifier Type 2606 via a Conditioning Amplifier Type 2626. As the Beat Frequency Oscillator Type 1022 is synchronized with the Level Recorder Type 2305 the frequency response of the hydrophone can be recorded automatically on a frequency calibrated chart. Typical free field voltage sensitivity response of the hydrophone Type 8100 in air is shown in Fig.9 and is found to be within ± 1 dB up to 4 kHz. The directivity patterns in all planes are the same within ± 1.5 dB in this frequency range.

Calibration in the Infrasonic Frequency Range

Calibration of the hydrophone in the frequency range 0.1 Hz to 1 Hz is carried out in the water tank described earlier. By varying the depth of the hydrophone in water periodically as indicated in Fig.10 an oscillating pressure

$$\Delta p = \Delta h \, \rho g$$

is acted upon the hydrophone. ρ is the water mass density, g the acceleration due to gravity and Δh the double amplitude of vertical motion. The sensitivity of the hydrophone is measured utilizing the Low Frequency Charge Amplifier Type 2628 shown in Fig.10 and evaluated from the relationship

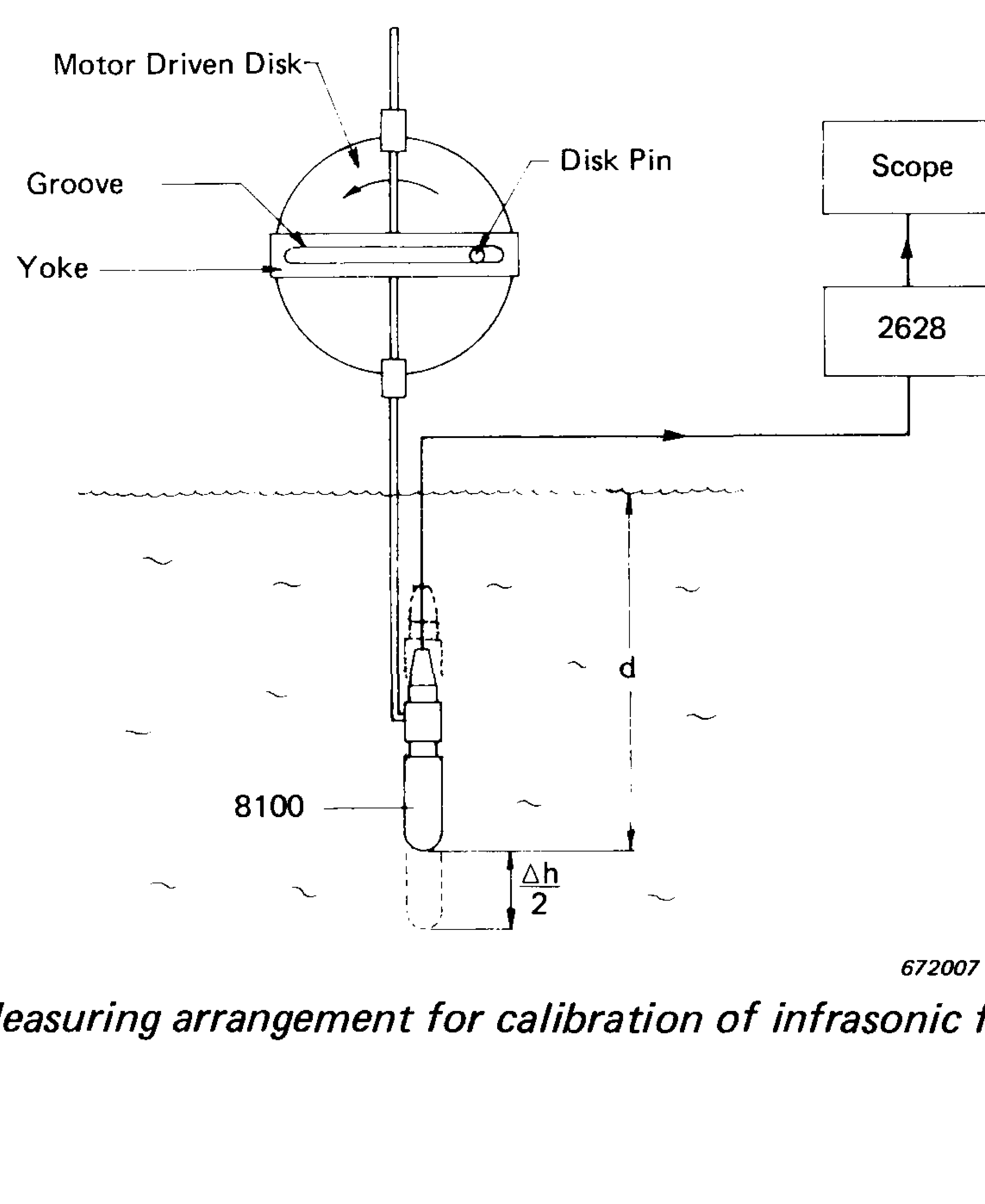

Fig.10. Measuring arrangement for calibration of infrasonic frequencies

$$S_H = \frac{\Delta e}{\Delta h \, \rho g}$$

where Δe is the peak-peak voltage measured on the oscilloscope. At infrasonic frequencies quasi-static pressure fluctuations are indistinguishable from sound pressure and absolute pressure calibration of the hydrophone is obtained. The inertial effects of the hydrophone are minimized on account of its vertical orientation.

There is no lower limiting frequency for this calibration method, but the upper frequency is limited by turbulence and laminar flow effects normally between 1.5 – 4 Hz.

Impedance Measurement

Often the impedance of a hydrophone is a useful parameter, to know when it is used as a transmitter. The electrical impedance measured at the terminals of the hydrophone can not only be used for calculation of the hydrophone efficiency and driving current from its transmitting voltage response, but also for impedance matching between the transducer and the transmitting or receiving equipment.

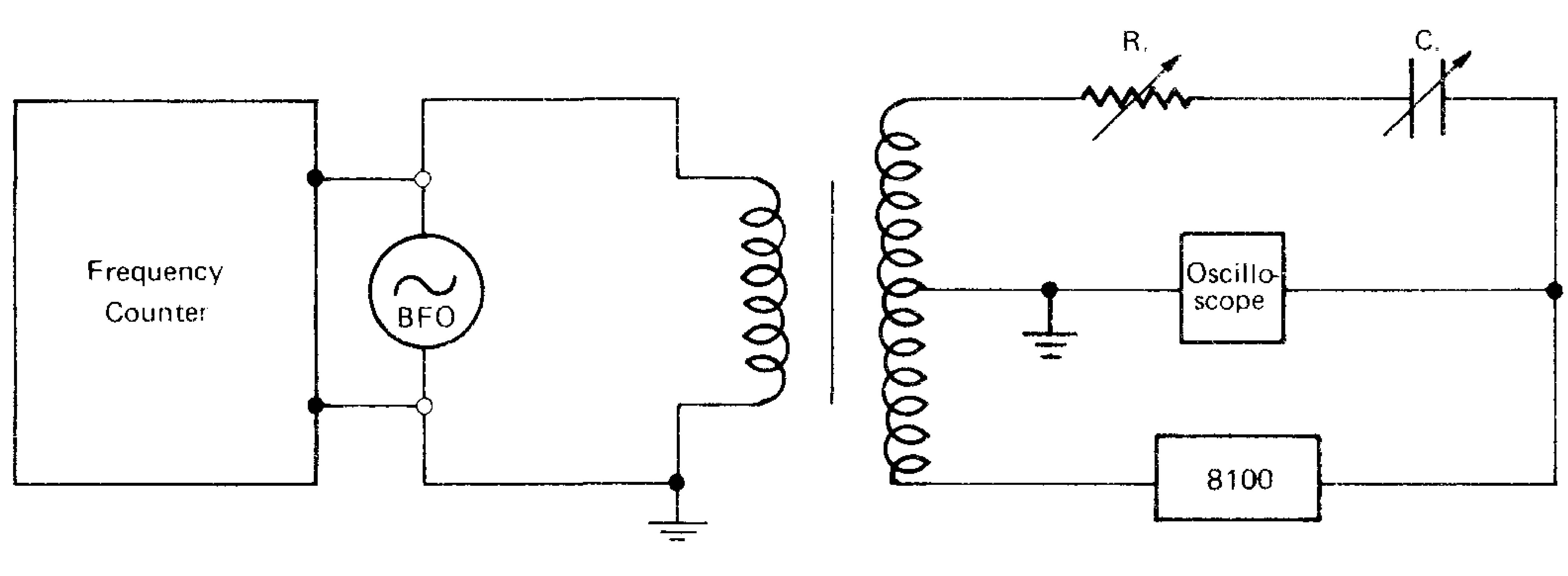

Fig.11. Bridge arrangement for impedance measurements

When carrying out impedance measurements the hydrophone should be loaded as it is loaded in actual practice. Although impedance is measured electrically it is a function of mechanical mass, stiffness as well as the acoustical characteristics of the hydrophone.

The measurement set-up used, based on the bridge principle, is shown in Fig.11. The Beat Frequency Oscillator Type 1013 supplies a low distortion signal to a differential transformer with ferrocube core, the secondary winding of which is grounded at its middle point. Thus two voltages of same amplitude but of opposite phase are applied across the two impedances: one to either a standard hydrophone or to a variable resistance and capacitance and the second to the unknown hydrophone.

The frequency of the Beat Frequency Oscillator is measured by a frequency counter, while the balance between the impedances is read out on the oscilloscope. The results obtained for the Hydrophone Type 8100 submerged in water are shown in Fig.12.

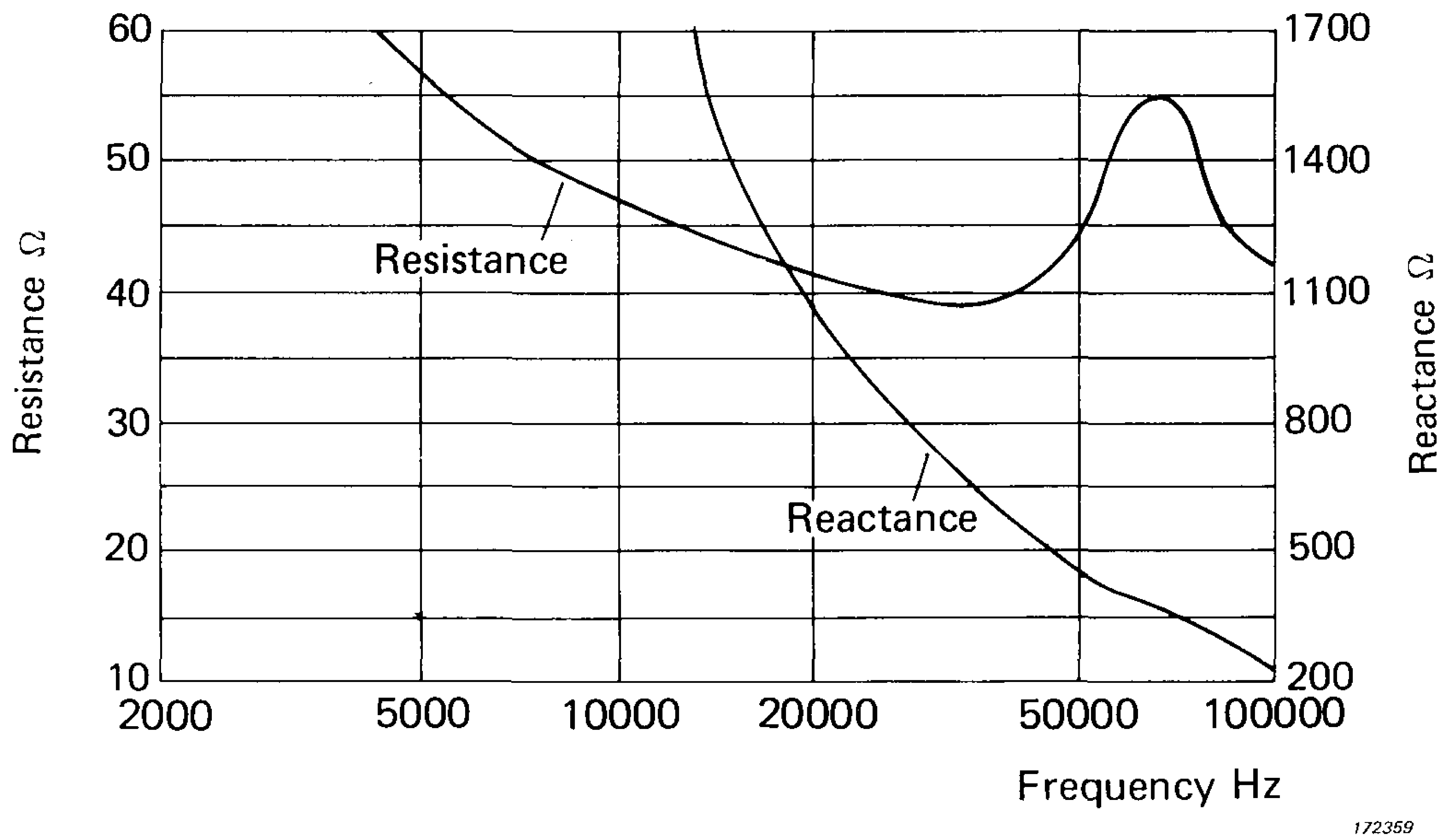

Fig.12. Typical resistance and reactance curves for a Hydrophone Type 8100

Equivalent Noise Pressure

The concept of Equivalent Noise Pressure is often used as a quality factor for the hydrophone. The equivalent noise pressure is defined as the RMS pressure that would produce a hydrophone open circuit voltage equal to the inherent electrical noise of the hydrophone

$$p = \frac{e}{S_H}$$

where S_H is the free field voltage sensitivity of the hydrophone and e is the electrical noise voltage measured. Noise levels are expressed as spectral levels in dB re 1 μPa in a 1 Hz bandwidth (or re 1 $\mu Pa/\sqrt{Hz}$).

Since inherent noise is also contributed by the preamplifier used, Fig.13 shows typical equivalent noise pressure curves of the hydrophone when used with different Brüel & Kjær preamplifiers.

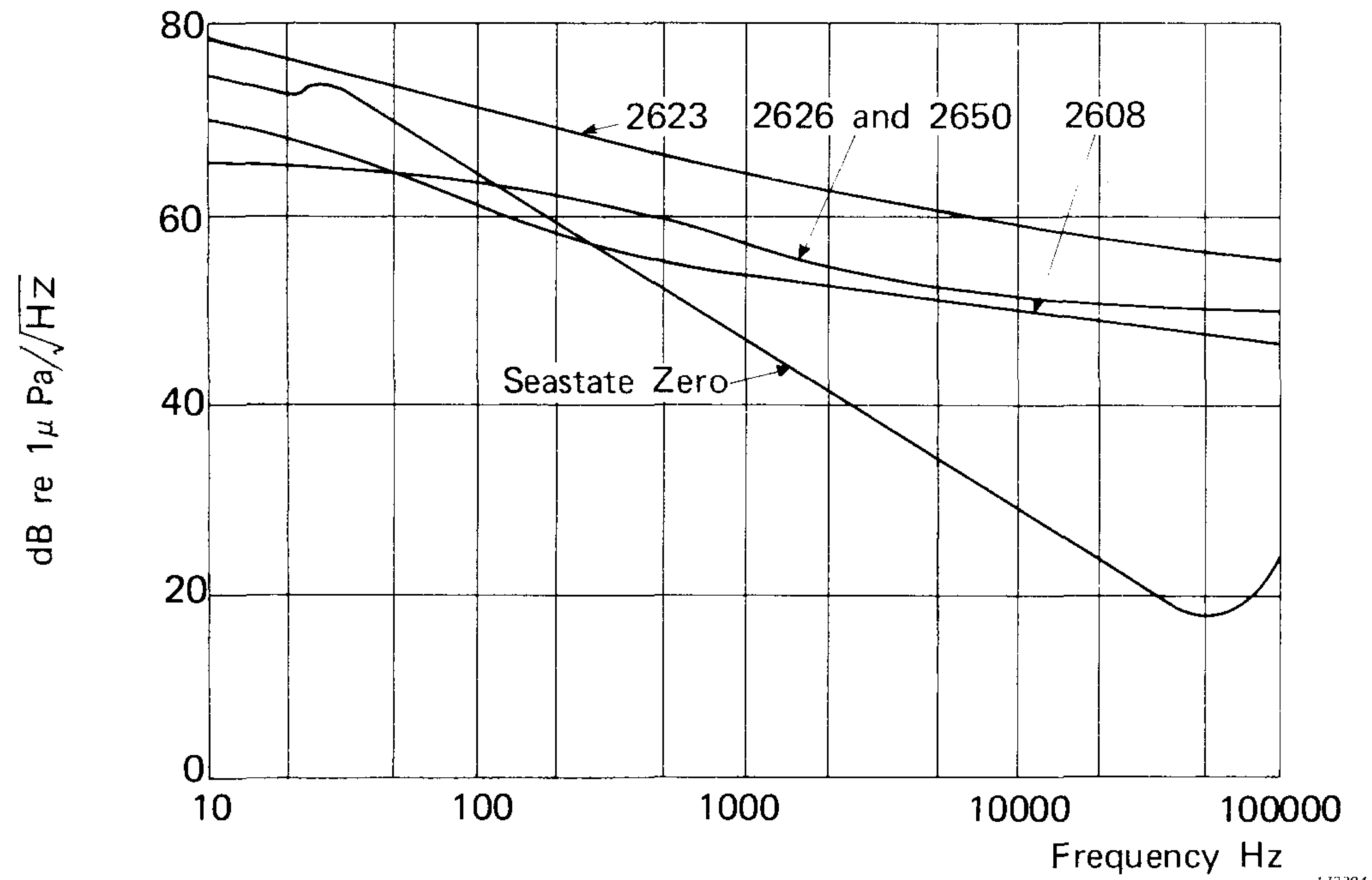

Fig.13. Equivalent noise pressure of the Hydrophone connected to different preamplifiers

Conclusion

The hydrophone calibration methods outlined here, as can be seen, are relatively simple to carry out. However, if more accurate results are desired, the reciprocity method of calibration could be carried out using the set-up of Fig.4 and utilizing three hydrophones placed equidistant from each other in a triangular form. A slightly easier comparison method could also be carried out in which the voltages of the standard and the unknown hydrophone placed equidistant from a transmitter, are compared.

In the methods described, the voltage sensitivity of the hydrophone has been measured. The charge sensitivity S_q can, however, be evaluated from

$$S_q = S_v \times (\text{Capacitance of Hydrophone} + \text{Cable})$$

where S_v is the voltage sensitivity.

It should be noted that the voltage sensitivity depends on cable capacity and hence the length of the cable, while the charge sensitivity remains constant. Prior to carrying out any measurements with the hydrophone, the neoprene rubber boot should be carefully cleaned with a mild solvent or detergent and a suitable wetting agent then applied, such as soap suds to ensure that air bubbles do not adhere to the surface of the hydrophone. Also the hydrophone should be allowed at least one minute per ^{0}C temperature change to stabilize in the medium before measurements are carried out. On account of practical limitations mentioned earlier, calibration of the hydrophone in the frequency range 1 Hz – 50 Hz is usually carried out in a specially designed pressurized calibration chamber.

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The Measurement of Reverberation Characteristics*)

by

J.R. Hemingway

ABSTRACT

The reverberation time is recognized as being a major parameter for assessment of the acoustic quality of lecture halls, auditoria and studios. The main methods of measuring reverberation curves are discussed and compared. Practical instrumentation set-ups for the recording of reverberation curves are described, together with the analysis and assessment of the curves.

SOMMAIRE

Il est admis que le temps de réverbération est le paramètre principal lorsqu'il s'agit d'évaluer les qualités acoustiques des salles de conférence, des auditoriums et des studios. Les principales méthodes de mesure des courbes de réverbération sont discutées et comparées. Les montages pratiques permettant l'enregistrement des courbes sont décrits, ainsi que l'analyse et le dépouillement des courbes.

ZUSAMMENFASSUNG

Als einer der wichtigsten Parameter für die Beurteilung der akustischen Qualität von Studios, Hör- und Konzertsälen wird die Nachhallzeit angesehen. Im Aufsatz werden die wichtigsten Nachhallzeit-Meßverfahren durchgegangen und miteinander verglichen. Ferner werden einige praktische Geräteanordnungen für die Aufzeichnung von Nachhallkurven beschrieben sowie deren Analyse und Auswertung dargelegt.

Introduction

If a sound in a studio, auditorium, concert hall or theatre is abruptly switched off, the sound intensity in the room will not fall immediately to zero. The sound will only gradually be absorbed with each succesive reflection at the internal surfaces of the room, and at high frequencies by the air in the room.

*) This paper was first published in "Fernseh und Kino-Technik", Berlin, 26 (1972) No. 12.

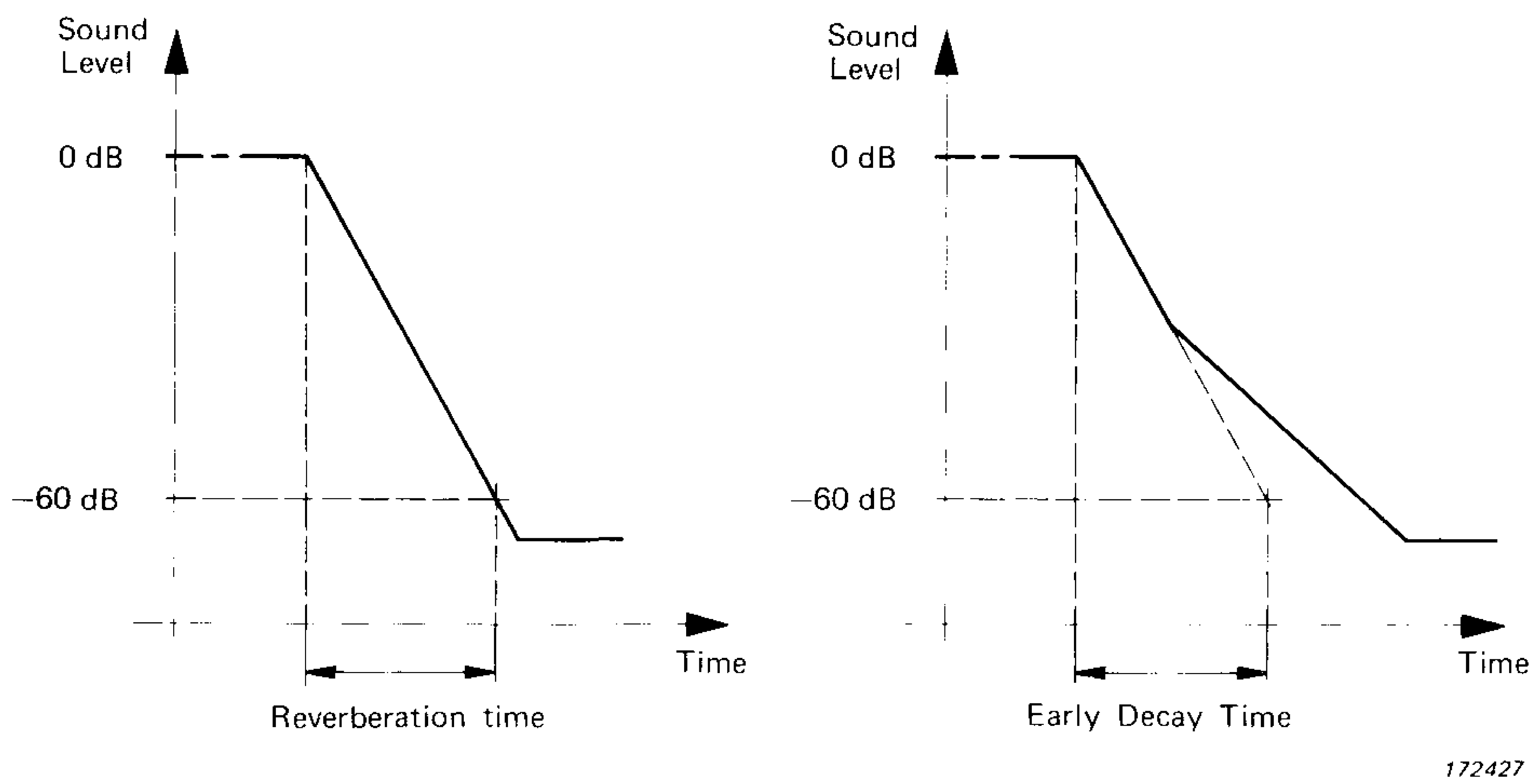


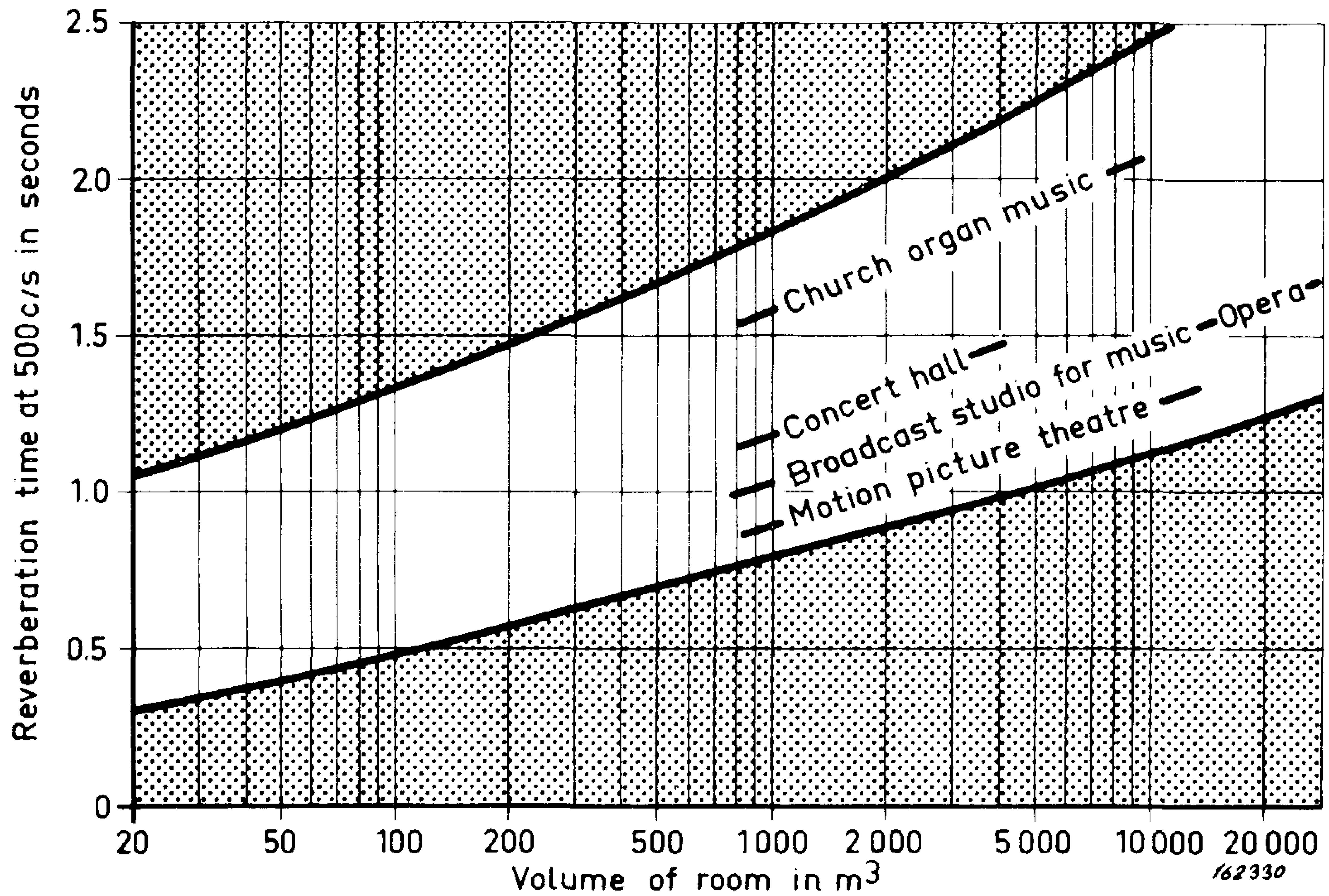
Fig. 1. The Definition of Reverberation Time and Early Decay Time

The way in which sound decays away in a room is termed the reverberation characteristic of that room. It has long been recognized that the reverberant characteristic is an important parameter in the acoustic assessment of rooms designed for the purpose of sound emission and reception. Much of the early work in the field of reverberation studies was performed by Sabine (1). Sabine defined the reverberation time (RT) of a room as being the time in seconds which the sound level in the room requires to decay to 60 dB below its initial level. In a well diffused room, free from standing waves, the decay will be exponential. If such a decay curve were recorded logarithmically, it would appear as a straight line, see Fig.1 (a).

Fig.1 (b) shows a typical non-exponential decay curve which can occur in practice. The latter part of the curve has a smaller slope than the initial portion. Such a decay curve can result when a significant portion of the decay takes place between parallel walls and a standing wave pattern is established. For such a curve it is convenient to define the Early Decay Time (EDT) as the reverberation time extrapolated from the initial portion of the curve only, the later portion being neglected.

Having defined the reverberation time, we would expect to be able to specify the optimum reverberation time for a certain room. Unfortunately this is not so, for the simple reason that the subjective impression of the acoustic quality of a room varies considerably from person to person. It is, however, possible to take an average of the recommendations of various investigators, and this is shown for music in Fig.2.

Fig.2 (a) shows the recommended variation of the reverberation time with room size for various types of room. In general the reverberation time



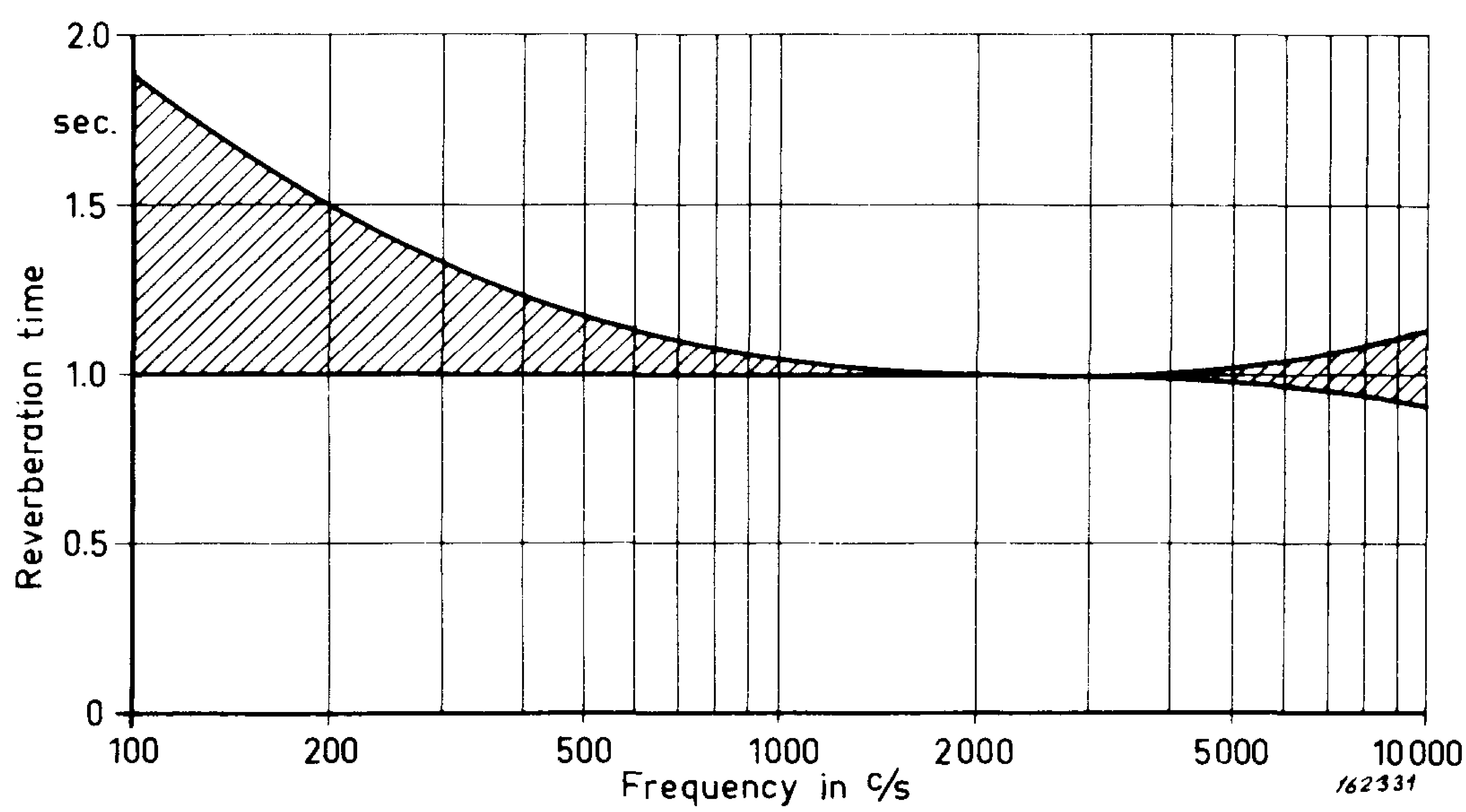


Fig.2. The Variation of the Optimum Reverberation Time for Music with Room Size, Room Type and Frequency

should increase with room size. Cinemas should have low reverberation times as the correct reverberation characteristics have already been "built-in" to the sound track, studios and concert halls should have slightly increased reverberation times, and churches should have long reverberation times for the best performance of church music written for these conditions.

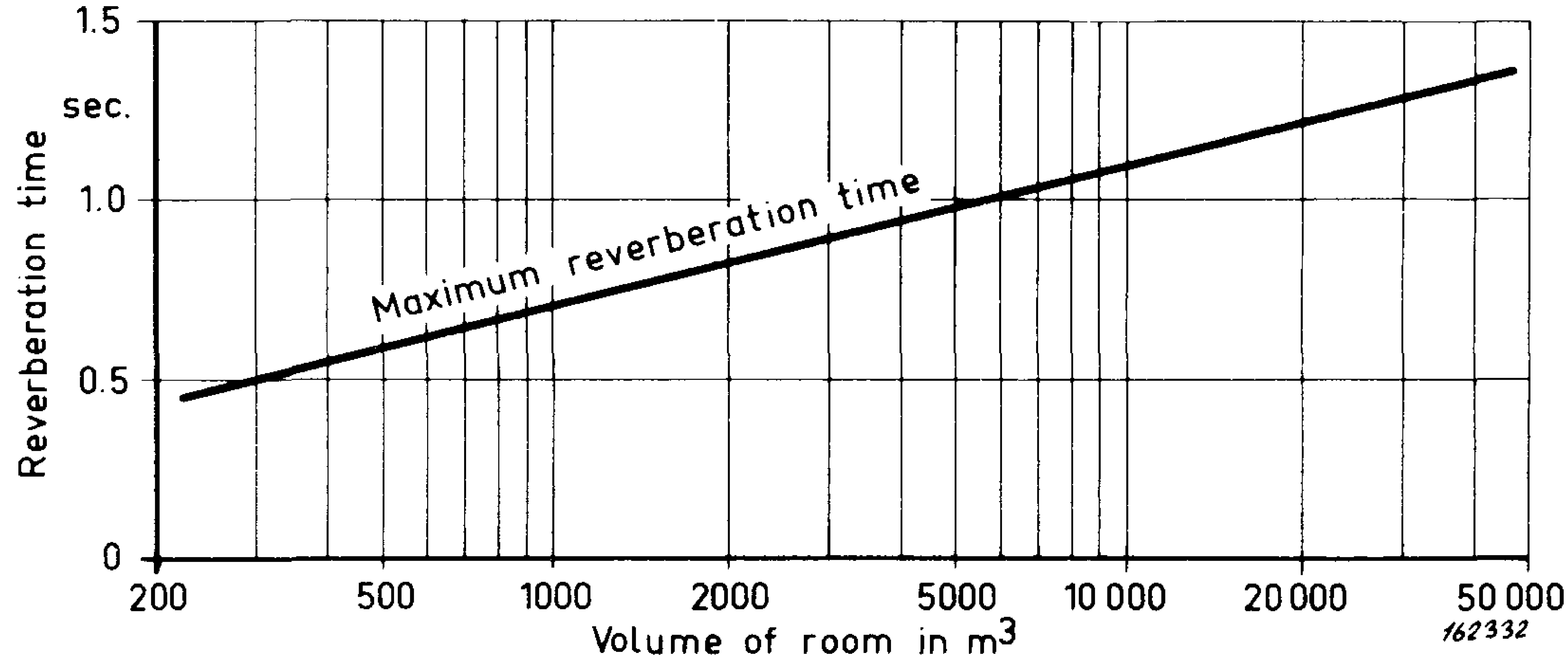


Fig.3. Variation of the Maximum Reverberation Time for Speech with Room Size (from "Sound Insulation and Room Acoustics" by P.V. Brüel)

Fig.2 (b) shows the range of data for the variation of reverberation time with frequency. In general the reverberation time should be constant across the frequency range, but with slightly increased values at low frequency.

For lecture halls, auditoria and theatres the criteria is that the reverberation time should not be long enough to interfere with the intelligibility of the speech. Thus a maximum reverberation time can be specified, the variation of which with room volume is given in Fig.3.

Types of room excitations signal

Sinusoidal Excitation

Room excitation by a sinusoidal or pure tone signal would seem to be the obvious first choice, but despite the advantage of simplicity, it has two great disadvantages. The first is comprehensive testing over the audio-frequency range (20 Hz – 20 kHz) is impossible using discrete frequency excitation. The second disadvantage is that even though the room excitation is at one frequency, the room response will contain components at slightly different frequencies, which gives beating between components. Thus decay curves obtained using pure sinusoidal excitation tend to be very irregular in nature. The time variation of a sinusoidal excitation at a frequency of 100 Hz is shown in Fig.4 for comparison with later examples.

Warble Tone Excitation

The basic disadvantage of pure sinusoidal excitation can be overcome by using a "warble-tone" as shown in Fig.5. In this type of excitation the

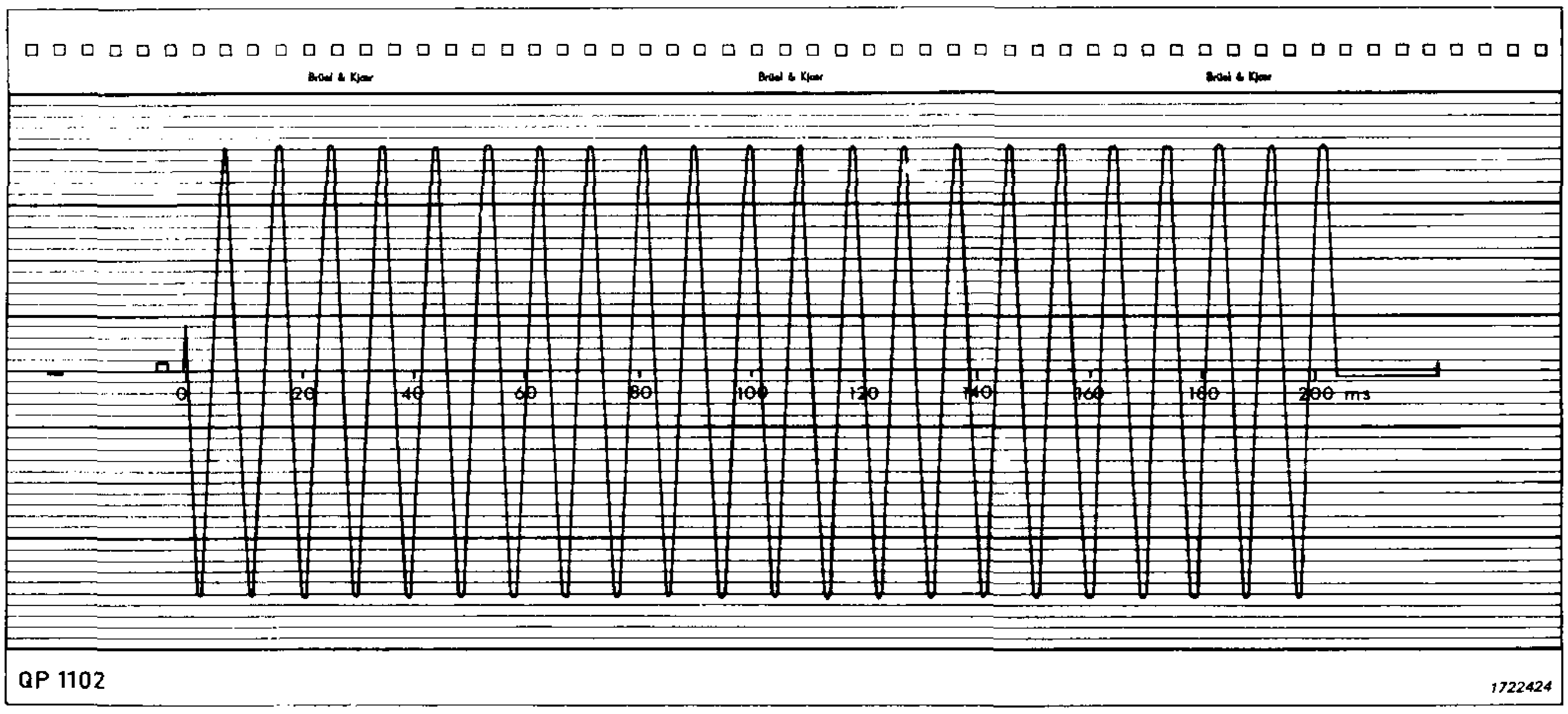


Fig.4. Sinusoidal Excitation Signal

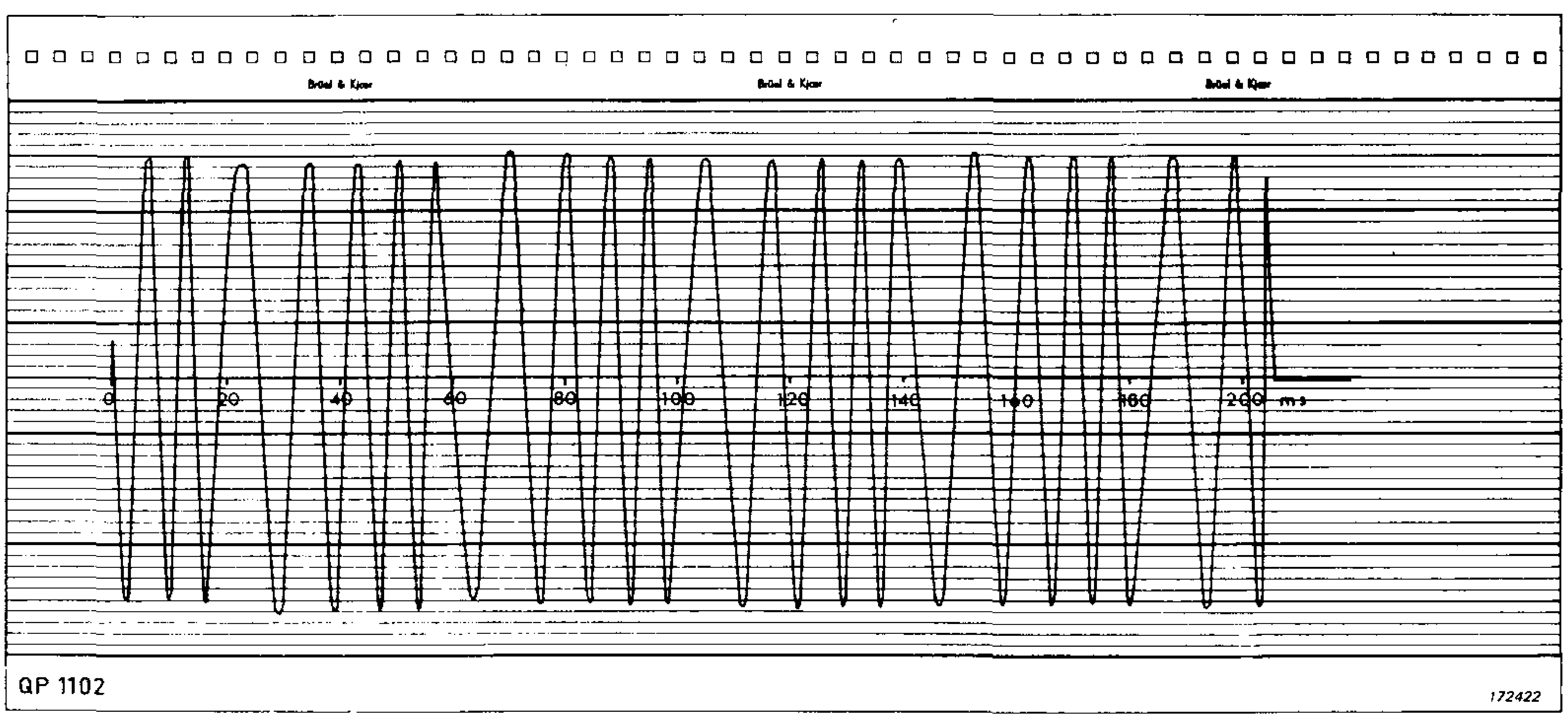


Fig.5. Warble Tone Excitation Signal

frequency varies either side of the centre frequency, thus giving a range of frequency excitation. In Fig.5 the centre frequency is 100 Hz, with a frequency range of 63 Hz. Decay curves obtained using a warble-tone are smoother than for pure sinusoidal excitation, but as the frequency at the point of sound interrupted is random poor repeatability between succesive decay curves results.

Random Noise Excitation

Room excitation by random noise again has the advantage that the excitation signal covers a band of frequencies continuously either side of the centre frequency. The usual excitation signal is white noise (random noise with a flat frequency characteristic) filtered in third octave or octave bands, these being band pass filters whose bandwidth is a constant percentage of

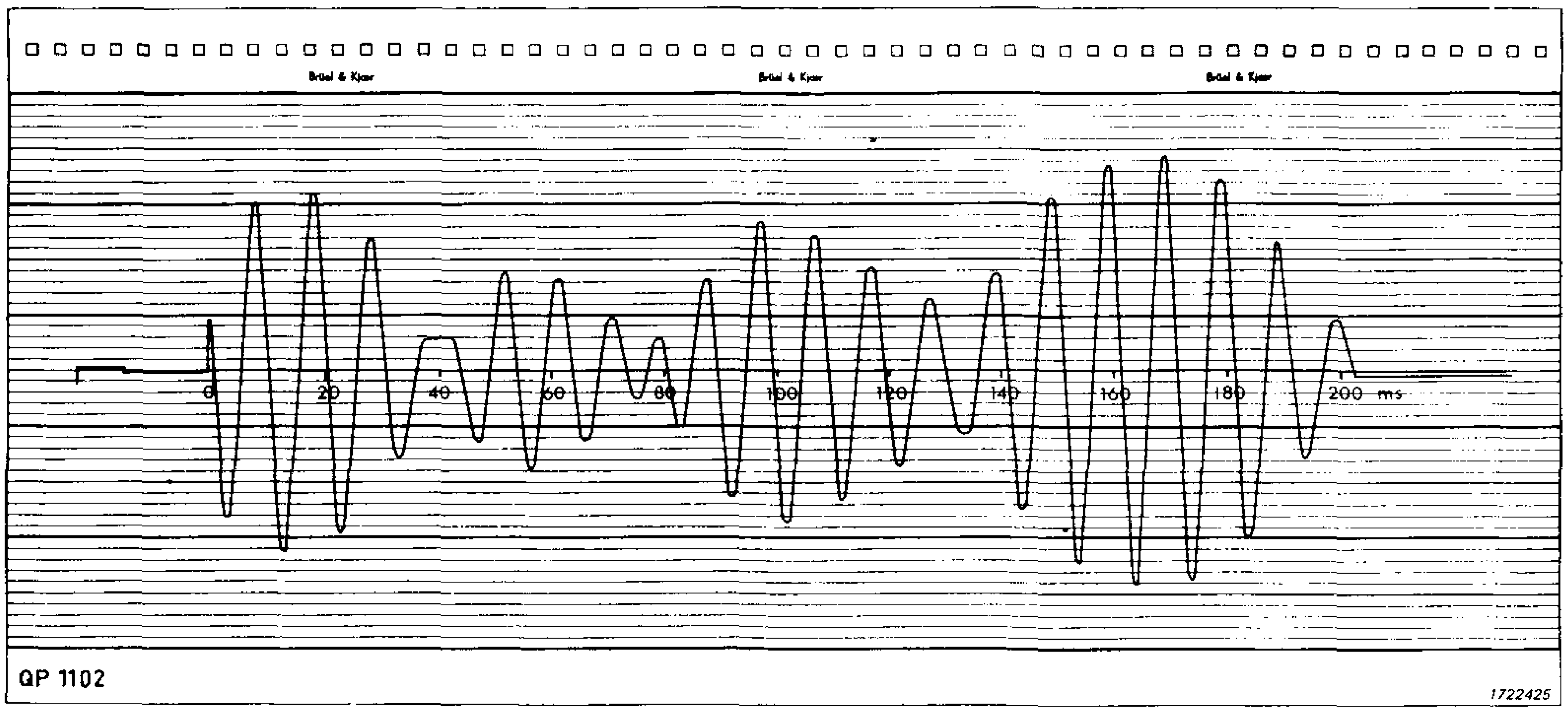


Fig.6. Filtered White Noise Excitation Signal

the centre frequency. The audio range from 20 Hz to 20 kHz is divided into 30 third octave or 10 octave bands. The time variation of a white noise signal third octave filtered at 100 Hz is shown in Fig.6.

The random nature of the signal can clearly be seen. Again the signal at the point of switch off is random, resulting in poor repeatability between succesive decay curves.

Impulsive Excitation
A common method of impulsive room excitation has been by pistol shot. In this method the entire room response is measured, possibly tape recorded to save shots, and filtered in third octave or octave bands for analysis. Although decay curves using the pistol shot technique tend to have better

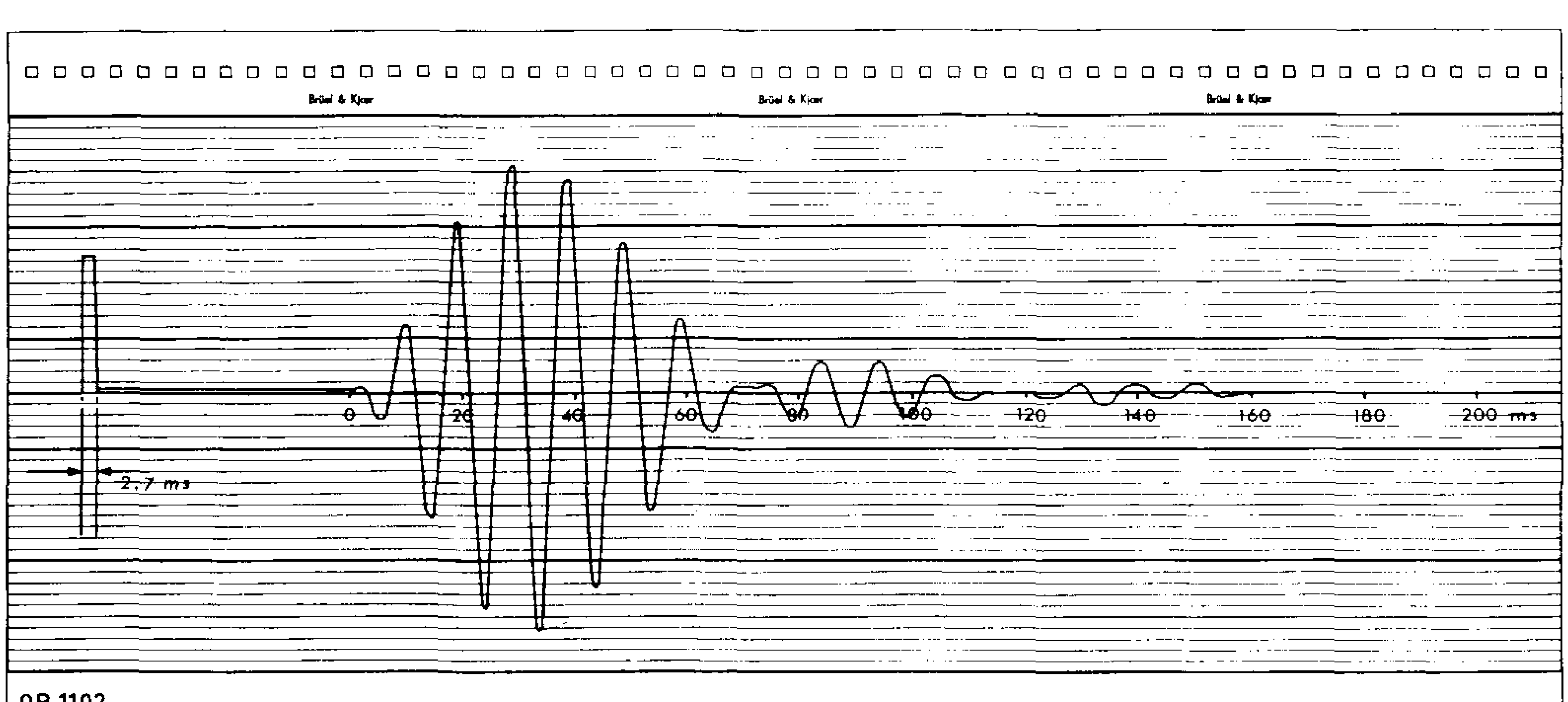


Fig.7. Filtered Rectangular Pulse Excitation Signal

repeatability than those of previous methods, a disadvantage is that very little control can be exerted over the excitation, in particular ensuring adequate excitation over the audio frequency range. A type of impulsive excitation which overcomes these difficulties has been recently developed by Schroeder (2). In the Schroeder technique, a third octave or octave filter is excited by a rectangular pulse and the resulting filter ringing is used as the room excitation. An example of the excitation is shown in Fig.7.

A typical rectangular pulse is shown, together with the response to this pulse of a third octave filter with a centre frequency of 100 Hz. Decay curves using such excitation are highly repeatable.

Practical Reverberation Measurement Set-Ups

The Interrupted Noise Method

The measurement set-up for reverberation investigations using the interrupted noise method is shown in Fig.8.

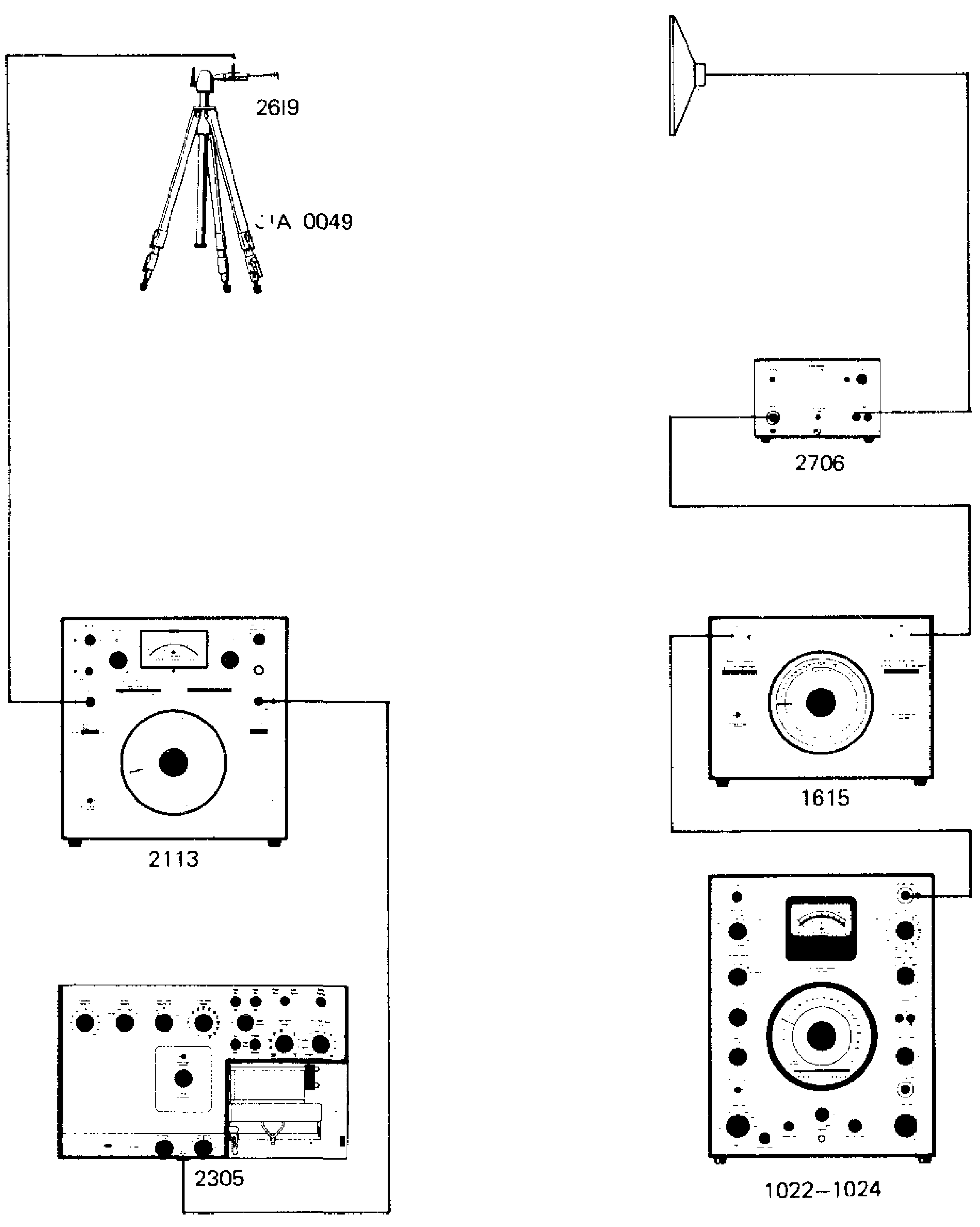


Fig.8. Measurement Set-up for Reverberation Investigations using the Interrupted Noise Method

Warble tone or random noise excitation are the normal signals used for this method. For warble-tone excitation, a B & K Beat Frequency Oscillator Type 1022 is used, and for random noise excitation a B & K Sine-Random Generator Type 1024 followed by a B & K Third Octave Filter Set Type 1615. The excitation signal is passed to a B & K Power Amplifier Type 2706 and to a loudspeaker inside the room under investigation. In the figure the room response is received by a single microphone, but in practice the results from several microphones placed around the room may be averaged to give a more representative result. A B & K Half Inch Microphone Type 4133 or 4134 is used, mounted on a Microphone Preamplifier Type 2619. The signal from the microphone is passed to a B & K Frequency Spectrometer Type 2113 for amplification and also third octave filtering to improve the signal to noise ratio. The output from the Frequency Spectrometer is passed to a B & K Level Recorder for logarithmic recording of the decay curve. Sudden interruption of the sound in the room is achieved by using the "generator stop" button on the Oscillator or Random Generator. Typical repeated decay curves obtained using the above apparatus with random excitation are shown in Fig.9.

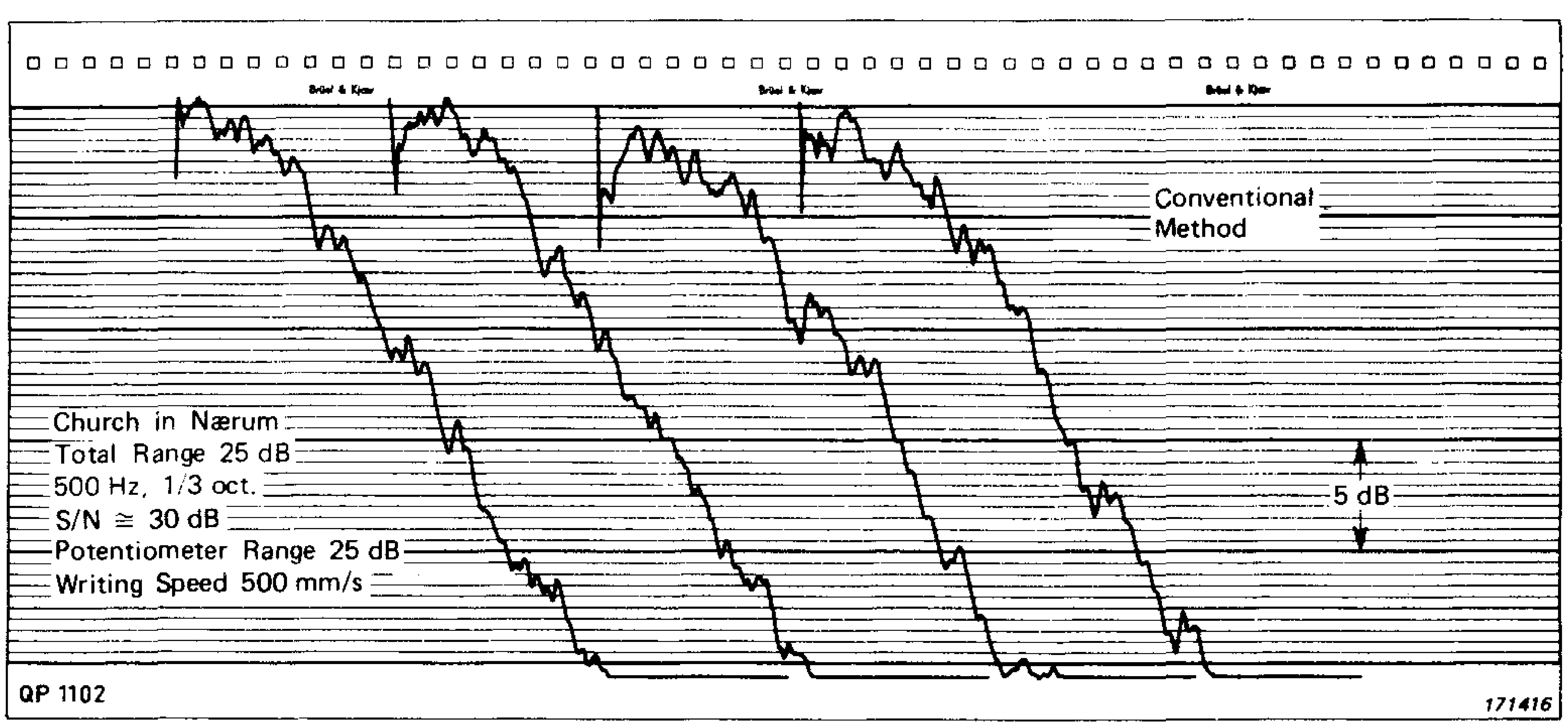


Fig.9. Typical Decay Curves Obtained Using the Interrupted Noise Method

The characteristic statistical uncertainty of decay curves using this method is clearly visible. When using this method it is necessary to take many curves at each frequency, and average the results.

The Schroeder-Kuttruff Method

The Schroeder-Kuttruff method followed from the integrated impulse theory of Schroeder (2) which was later modified by Kuttruff (3) who suggested a practical measurement set-up. Basically the result of the Schroeder theory is that the average of infinitely many squared reverberation decay curves using random noise excitation may be obtained by squaring and integrating the room response to a single impulsive excitation signal. The impulsive excitation is the filtered rectangular pulse previously discussed in section 2.4.

A typical measurement set-up using the B & K Reverberation Processor Type 4422, an instrument specifically designed to make use of the Schroeder-Kuttruff method, is shown in Fig.10.

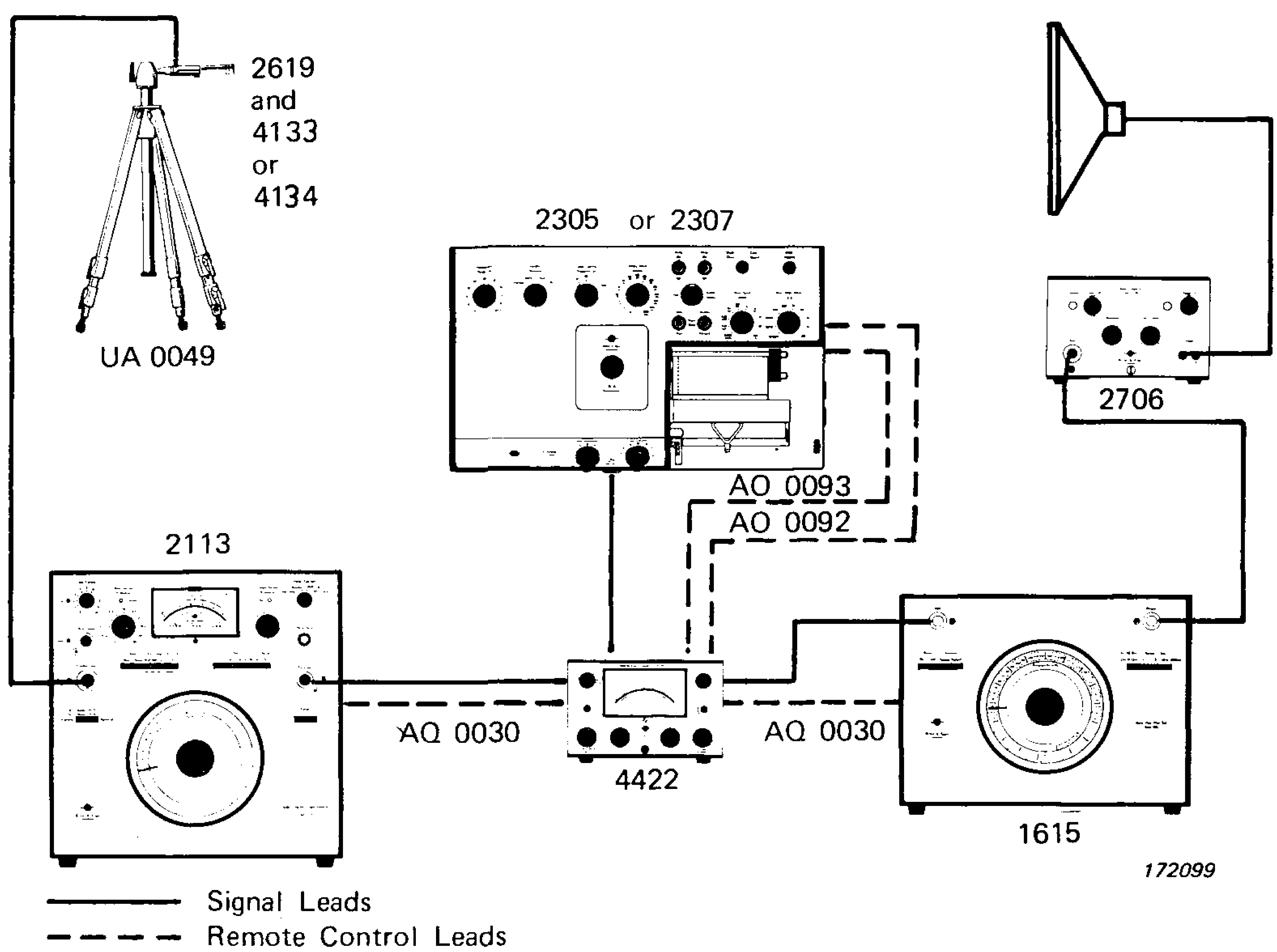


Fig.10. Measurement Set-up for Reverberation Investigations Using the Schroeder-Kuttruff Method

The Reverberation Processor produces a single rectangular pulse which is passed to a Third Octave Filter Set. The resulting impulsive signal is passed to a Power Amplifier and from there to a loudspeaker in the room under investigation. The room response is received by a Half Inch Microphone mounted on a Microphone Preamplifier and passed to a Frequency Spectrometer for amplification and third octave filtering. The signal is then returned to the Reverberation Processor for squaring and integrating. The resulting signal is passed to the Level Recorder for logarithmic recording of the decay curve. Typical decay curves obtained by this method are shown in Fig.11.

Due to the absence of statistical uncertainty in the excitation signal and the smoothing of the integration process, the curves are highly repeatable and of a regular nature. The curves are also representative of the true reverberation characteristic of the room under investigation and can be examined in detail.

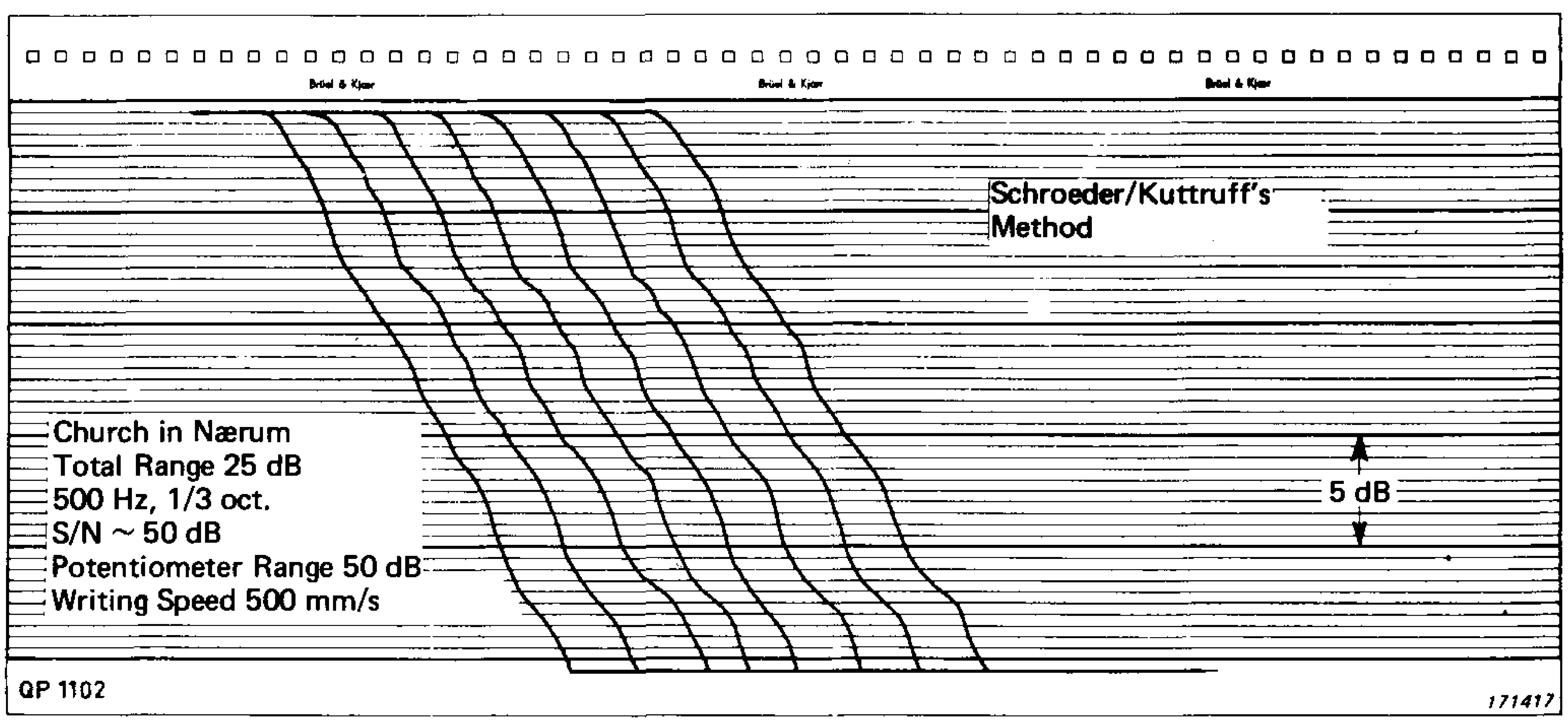


Fig.11. Typical Decay Curves Obtained Using the Schroeder-Kuttruff method

The Analysis of Reverberation Decay Curves

It is a simple matter to obtain the reverberation time from decay curves which are approximately exponential, and give a straight line when recorded logarithmically. A typical reverberation time analysis using the B & K Reverberation Protractor Type SC 2361 is shown in Fig.12.

The protractor reading has been factored to allow for the paper width (x 2) squaring (x 2) and a paper speed different to that specified (x 0.1). For non-exponential decays, for example those showing the bent characteristic of Fig.1, no single straight line can be drawn through the curve, and the description of such curves by a single reverberation time is difficult. To overcome this problem it is necessary to measure the subjective reverberation times of such curves as assessed by the human ear. An investigation of this type was performed by Atal et al. (4) who found that it was the Early Decay Time (EDT) assessed from the initial portion of the curve only which agreed best with the subjective assessment of the reverberation time. Good agreement in two concert halls for music was obtained by measuring the

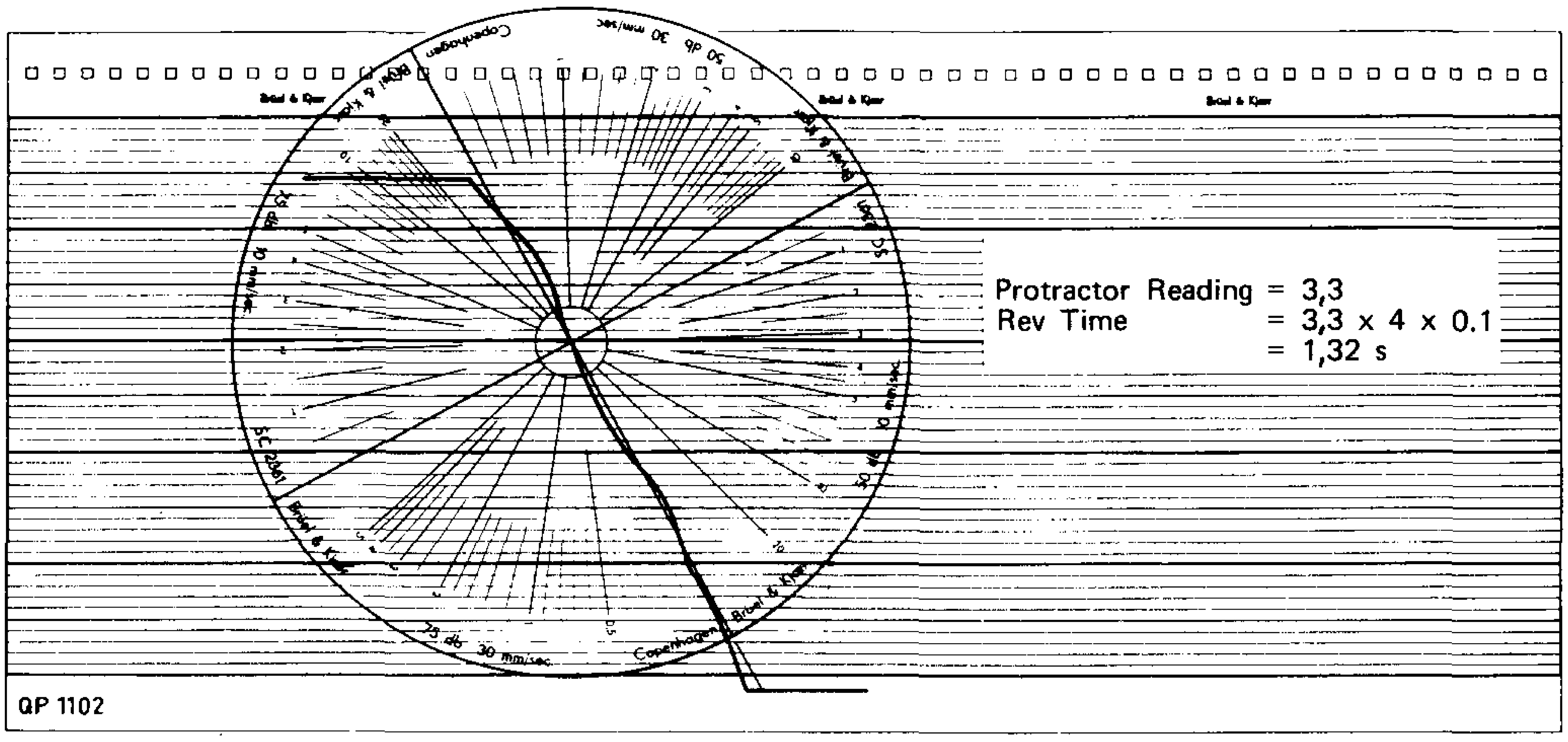


Fig.12. Analysis of a Decay Curve Using the Reverberation Protractor

EDT over the first 15 dB of the decay. Such a measurement would involve a large error for curves obtained by the interrupted noise method due to the irregularity of the curves. Curves using the Schroeder-Kuttruff method do not suffer from such irregularity, and accurate measurement from the first 15 dB of the curve is easily possible.

The B & K Reverberation Processor has the built-in facility of evaluating the EDT automatically without a recorded reverberation decay curve. The instrument obtains the EDT from the time for the sound to decay from the −1 dB to −10 dB (or −15 dB) points. A meter readout of EDT is provided.

Conclusion

The interrupted noise method of reverberation investigation, despite its inherent disadvantage of statistical uncertainty, has found widespread use in the field of acoustics. The recent technique of Schroeder and Kuttruff removes this statistical uncertainty from reverberation measurement, and allows the examination of reverberation decay curves in detail. It is to be hoped that the existence of a more sophisticated technique will result in a more exact definition of optimum reverberation characteristics, and enable better design and assessment of the acoustic quality of studios, auditoria, concert halls and theatres.

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Adaptation of Frequency Analyzer Type 2107 to Automated 1/12-Octave Spectrum Analysis in Musical Acoustics

by

Miroslav Filip)*

ABSTRACT
A measuring system is described which performs stepped semitone (1/12-octave) frequency analysis of tape recorded signals automatically. The system utilizes a Frequency Analyzer Type 2107 in which the standard rotary switch for external filter switching has been interchanged with a special semitone switch.

Further the measuring system includes a Level Recorder and control circuits which control tape recorder modes "start"/"stop" and filter tuning in conjunction with the semitone switch. The system is part of a measuring system to be developed for automatic analysis of predetermined segments of a non-stationary signal and other applications in musical acoustics.

SOMMAIRE
L'article décrit un système de mesure effectuant automatiquement l'analyse en fréquence en demi-ton (1/12-octave) sur des signaux enregistrés sur bande magnétique. Ce système utilise un Analyseur de Fréquence Type 2107 où le commutateur circulaire normal, utilisé pour faire commuter un filtre externe a été remplacé par un commutateur spécial.

Le système comprend en outre un Enregistreur de Niveau et des circuits qui commandent la mise en marche et l'arrêt de l'enregistreur magnétique et l'accord du filtre en conjonction avec le commutateur. L'ensemble fait partie d'un système de mesure qui permettra l'analyse automatique de segments prédéterminés d'un signal non stationnaire et d'autres applications en acoustique musicale.

ZUSAMMENFASSUNG
Es wird ein Meßsystem beschrieben, mit dem sich Frequenzanalysen von Signalen, die auf Magnetband gespeichert sind, automatisch in Halbtonschritten (1/12-Oktave) durchführen lassen. Das System basiert auf einem Frequenzanalysator Typ 2107, in dem die normale Nockenscheibe für die synchrone Umschaltung eines Terzfilters durch einen speziellen Halbtonschalter ersetzt wurde.

Ferner enthält das Meßsystem einen Pegelschreiber sowie Steuerschaltkreise für "Start/Stop" des Bandgerätes und für die Filterabstimmung in Verbindung mit dem Halbtonschalter. Das System ist Teil eines zu entwickelnden Meßsystems für die automatische Analyse von bestimmten Ausschnitten aus einem nichtstationären Signal und andere Anwendungen in der musikalischen Akustik.

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Introduction

In musicological applications of spectrum (SPL density) analysis, the bandwidths established and internationally standardized in acoustic noise analysis, i.e. the one-octave and third-octave bands, are often too wide. Many important sound structure details could be averaged out in the third-octave spectrum levels and sometimes essential information might be lost. In general, if significant tone components represented by narrow spectral peaks are expected, as is usually the case with musical signals, more narrow bands are to be employed to achieve desired frequency resolution, although one is well aware that this can be done only on account of the degraded time resolution, due to the uncertainty principle.

One of the best suited instruments for this purpose is the B & K Frequency Analyzer Type 2107 with the constant percentage bandwidth (at the –3 dB points) variable in steps from 20% down to approx. 6%, the latter being very close to the tempered semitone, i.e. one twelfth of an octave. Even if it must be borne in mind that a 6% bandwidth at a centre frequency, say, 100 Hz is equivalent to nearly 170 msec of transient time of the filter, the superiority of the constant *relative* bandwidth analysis over the constant absolute bandwidth analysis is evident especially in musical acoustics.

Further, the Analyzer Type 2107 is continuously tunable over the whole audiofrequency range by an external drive and may also be tuned remotely from the Level Recorder Type 2305 via the Flexible Shaft UB 0041. Naturally, the choice of this instrument excludes the possibility of the real-time spectrum analysis, but in the applications desired the increased time consumption of the repetitive mode of analysis has not been considered a serious drawback.

The following requirements have then been specified:

a) The tuning of the analyzer should take place in semitone steps in such a way that during each playback of the signal in repetitive analysis the resonance frequency of the selective filter is kept *constant,* then the analyzer is tuned one semitone higher, the tuning stopped, a new playback (of the same signal) started, etc.

b) The tuning of the analyzer should be remotely controlled and the whole measurement process *automated.*

Provided that the Analyzer Type 2107 and the Recorder Type 2305 are at disposal in the laboratory, only a slight adaptation of the analyzer and a construction of a relatively simple control unit is necessary to form an instrumental basis for various analytic procedures utilizing the 1/12-octave spectrum analysis. It is believed that owing to moderate cost of the additional equipment this may be of interest to musical acousticians, and perhaps also to others who encounter in their practice a need for an automated high-resolution frequency analysis.

Analyzer Adaptation*)

The Analyzer Type 2107 is equipped (from the serial no. 132953 up) with a rotary switch in the form of a circular printed board plate placed on the main frequency tuning axis, and two contacts, designed for synchronous application with the third-octave filter as described in the Instruction booklet. Therefore, the original switch, Fig.1, connects the two contacts once every 1/3-octave. It is only necessary to replace the original printed board by another one which, if turned some small constant angle in the tuning direction, connects the same contacts and disconnects them after the

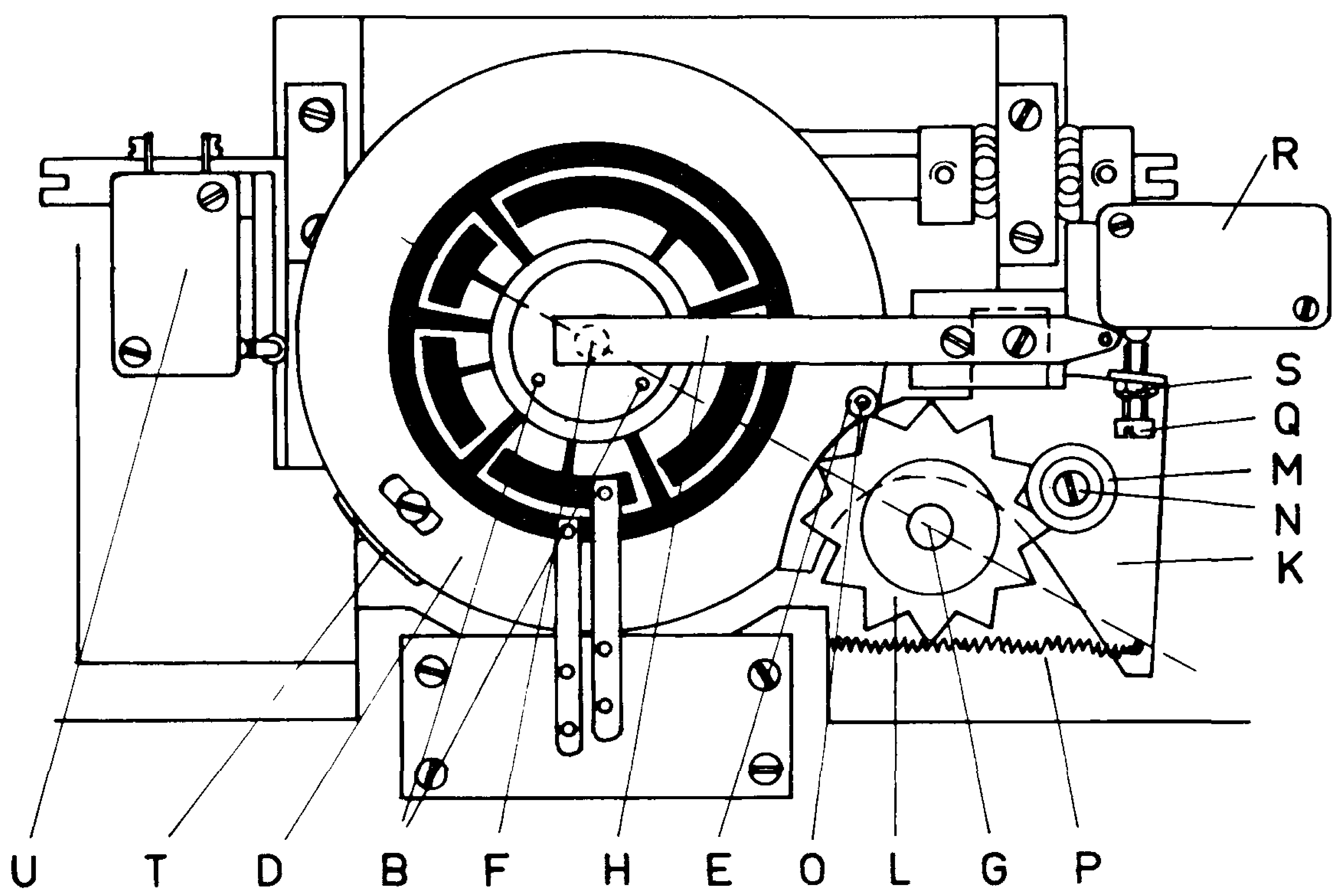

Fig.1. Original third-octave switch mounted in the Frequency Analyzer as shown in the service manual (black means conducting)

*) A similar adaption would be possible for the new Frequency Analyzer, Type 2120 in which the available filter bandwidths are 1%, 3%, 10% and 1/3-octave. (Editors note).

analyzer has been tuned exactly one semitone higher with respect to the starting position. This new rotary switch, Fig.2, controls the switching circuit described in the following section. Naturally, the semitone switch should also be rhodium plated as is the original one.

The optimum width of the *non*-conducting segments of the board, Fig.2, depends somewhat on the shape of the switch contacts. In the analyzer tested (serial no. 300433), the optimum angular values ("widths") of the non-conducting segments is about 7 degrees. The angular positions of the "leading" edges of these segments should correspond to the desired semitone increments on the tuning scale, because it is at these edges where the tuning is stopped and the tape transport started.

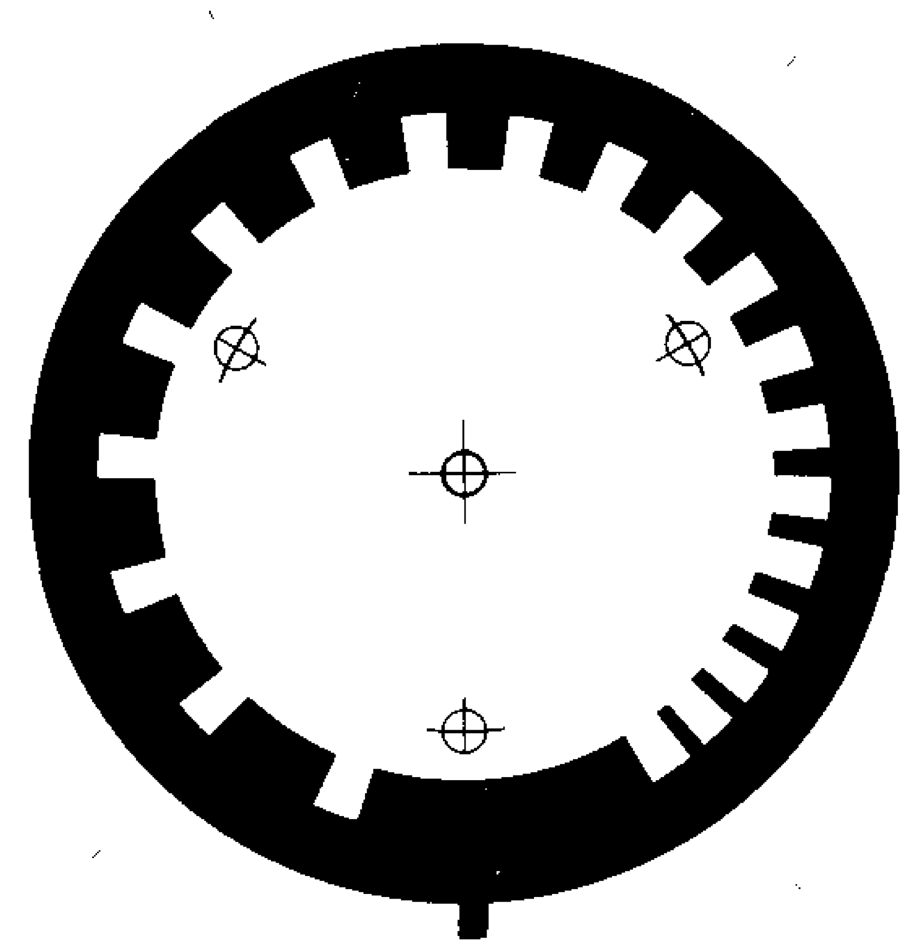

Fig.2. New twelfth-octave switch (black means conducting)

For the frequencies of the individual semitone steps it seems reasonable to choose the corresponding (i.e., derived from "equidistant" mantissas) antilogarithms, nearest to the internationally standardized preferred numbers (frequencies), see Table 1, although they do not fully agree with the equal-temperament frequency values related to the 440Hz standard. Also, the twenty tempered semitones are not exactly equal to one half of a log frequency decade (corresponding to one complete revolution of the frequency tuning axis), but the error is negligible as the musical interval equal to exactly one half of a log frequency decade is $39.86 \log_{10} \sqrt{10} = 19.93$ semitones as compared to the 20.00 semitones. This small discrepancy cannot be avoided because the angular positions of the frequency scale points defined by the rotary switch must remain the same in all the half-decade ranges of the instrument, but the magnitude of the error, when related to one range, amounts to only 0.35%.

F_a	200	212	224	237	251	266	282	299	316	335
F_b	631	668	708	750	794	841	891	944	1000	1059
F_a	355	376	398	422	447	473	501	531	562	596
F_b	122	1189	1259	1334	1413	1496	1585	1679	1778	1884

Table 1. *F_a are preferred numbers ("frequencies") nearest to corresponding semitone steps at the outer side of the frequency tuning scale. F_b are those at the inner side of the scale*

Control Unit

The signal to be analyzed is first recorded on one track of an endless tape loop on a remotely controlled tape deck. On the second track there is a timing command in the form of a short tone burst with about 3.5 to 4 kHz "carrier" frequency and about 0.2 to 0.3 sec duration. Since the shape of the timing command-envelope could never be perfectly rectangular, some processing is necessary, Fig.3.

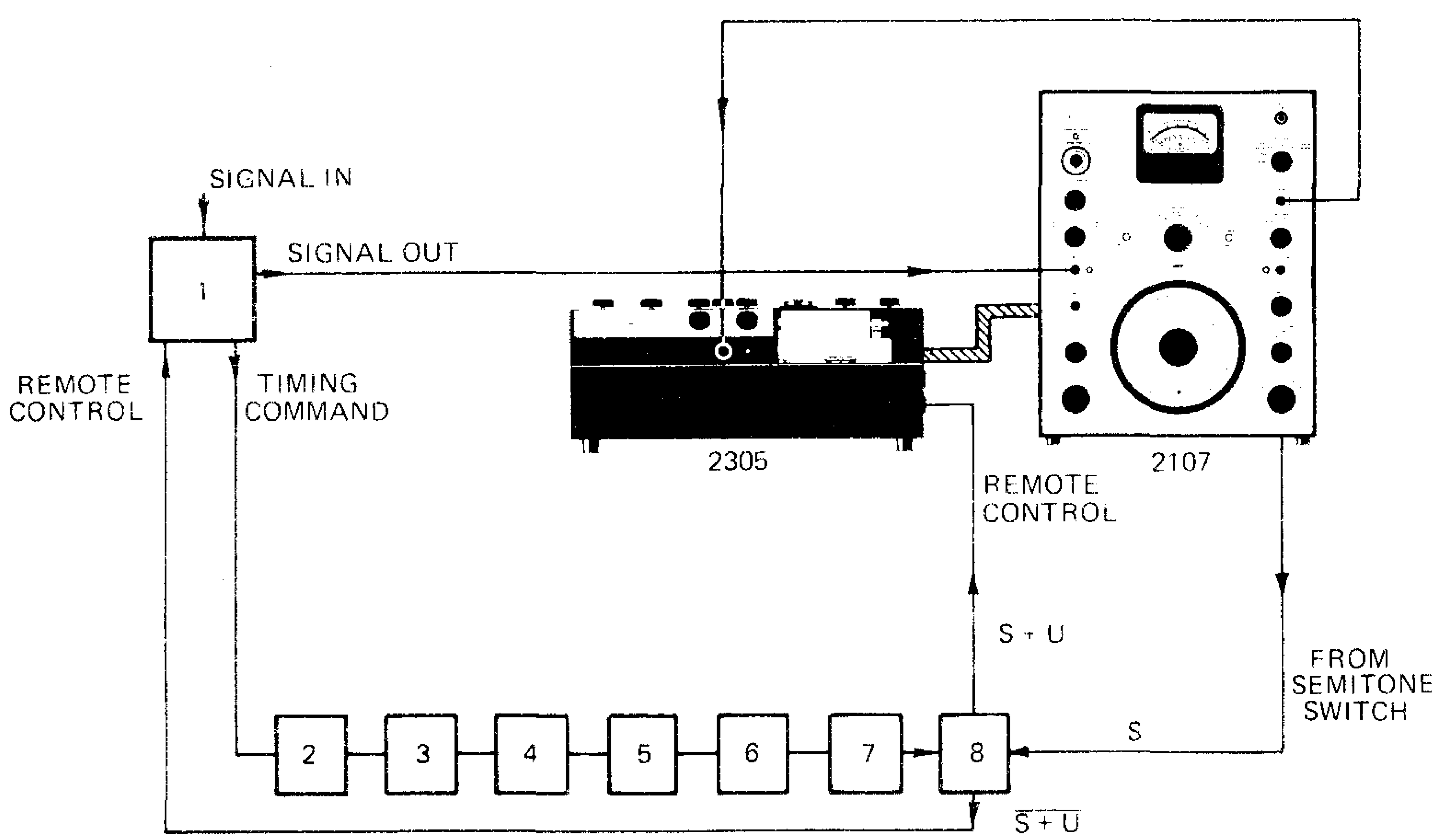

Fig.3. *1 . . . endless tape loop recorder*
2 . . . emitter follower buffer
3 . . . third-order high-pass Chebysheff active filter
4 . . . class C amplifier
5 . . . charge accumulating detector
6 . . . bistable circuit (Schmitt trigger)
7 . . . monostable circuit
8 . . . switching circuit (described in the main text)

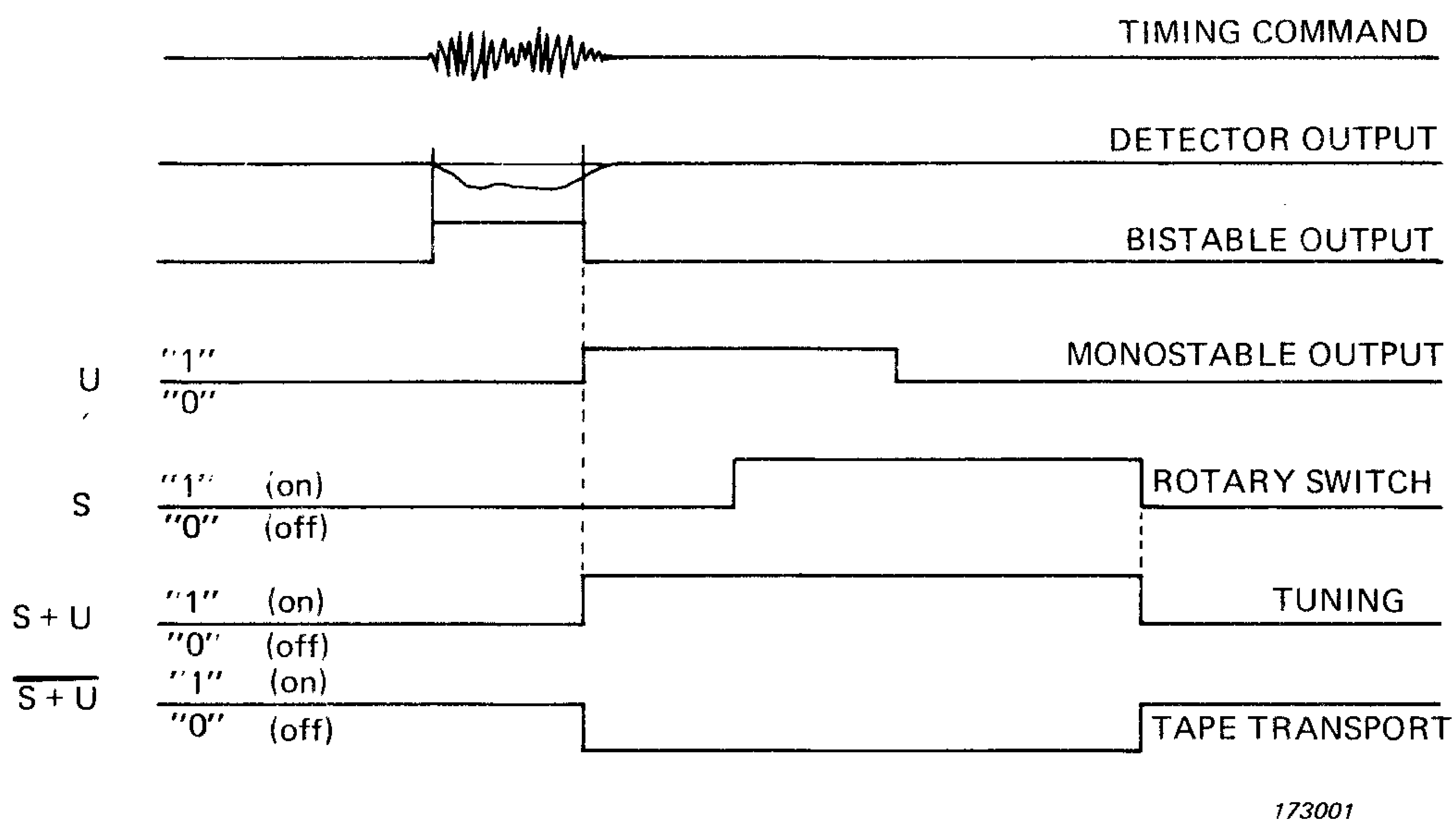

Fig.4. Typical waveforms, referred to Fig.3 and explained in the text

To reject low frequency components and transients, a high-pass filter is inserted. To prevent undesired multiple switching (more than once per each loop cycle), which could be caused by random impulses, a charge-accumulating type of detector is used. Thus a sufficient number of periods is necessary to trigger the bistable circuit, and individual random spikes remain inactive. The resulting waveform is then used to trigger the switching circuit in the way shown in Figs.3 and 4.

When activated, the switching circuit stops immediately the tape transport and, being now controlled for some time by the monostable circuit, initiates the tuning of the analyzer via the remote control of the magnetic clutch in the recorder. After a small time interval the 1/12-octave rotary switch in the analyzer takes over this function and the monostable may then return to its quiescent state.

Due to the described function of the rotary switch the tuning is continued until the analyzer has been tuned one semitone higher. Thereby the switching circuit is reset, the magnetic clutch disengaged, the tuning stopped, and the tape transport started. After one cycle of the loop the timing command initiates again the whole procedure as has just been outlined.

If we suitably interpret the functions U and S in Figs.3 and 4 as boolean, then the switching circuit could simply be seen as an OR-gate, and the S + U signal in logical "1" state starts tuning and stops tape transport, while in logical "0" state the reverse takes place.

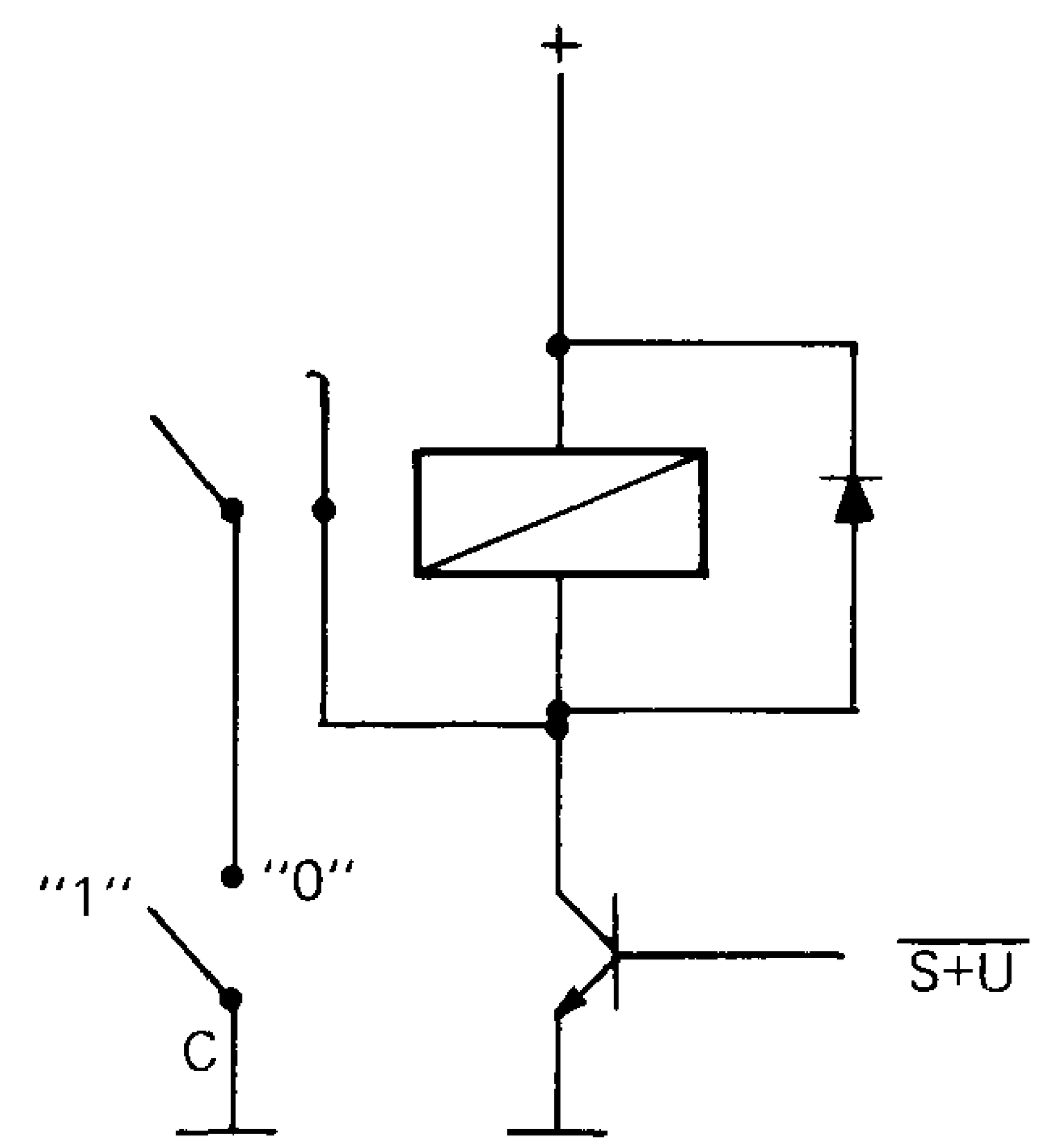

Fig.5. Tape recorder remote control relay with self-holding switch. Only the self-holding contact of the relay shown.
(For S + U in the logical "0" state the transistor receives a positive Voltage)

A self-holding stop arrangement, shown in Fig.5, is connected to the relay controlling the tape transport. If the switch C is set to "0" at *any* time, the tape will stop at the next arrival of the time command without being allowed to start a new cycle. Thus individual cycles of the tape loop can also be performed by simply setting the switch C to "1" and returning it to "0" immediately.[1)]

Final Remarks

With the aim to make the tuning of the analyzer and the paper transport in the recorder completely independent, the remote control of the magnetic clutch in the *analyzer* has also been experimented with various circuit configurations, but failed to satisfy the requirements because of small but unpredictable skips of the tuning axle depending on mutual positions of the two parts of the clutch at the moment of switching in. When the *recorder's* clutch is controlled, no such difficulties have been observed.

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1) This non-repetitive mode of loop cycling may be used with advantage also if the signal to be analyzed is itself tape recorded. Then the tape deck with the *original* record (set to playback) is remotely controlled by the same relay (that of Fig.5) as is the *loop* recorder (set to recording). Again, a simple switch can disengage the control of the tape with the original record during repetitive analysis from the loop.

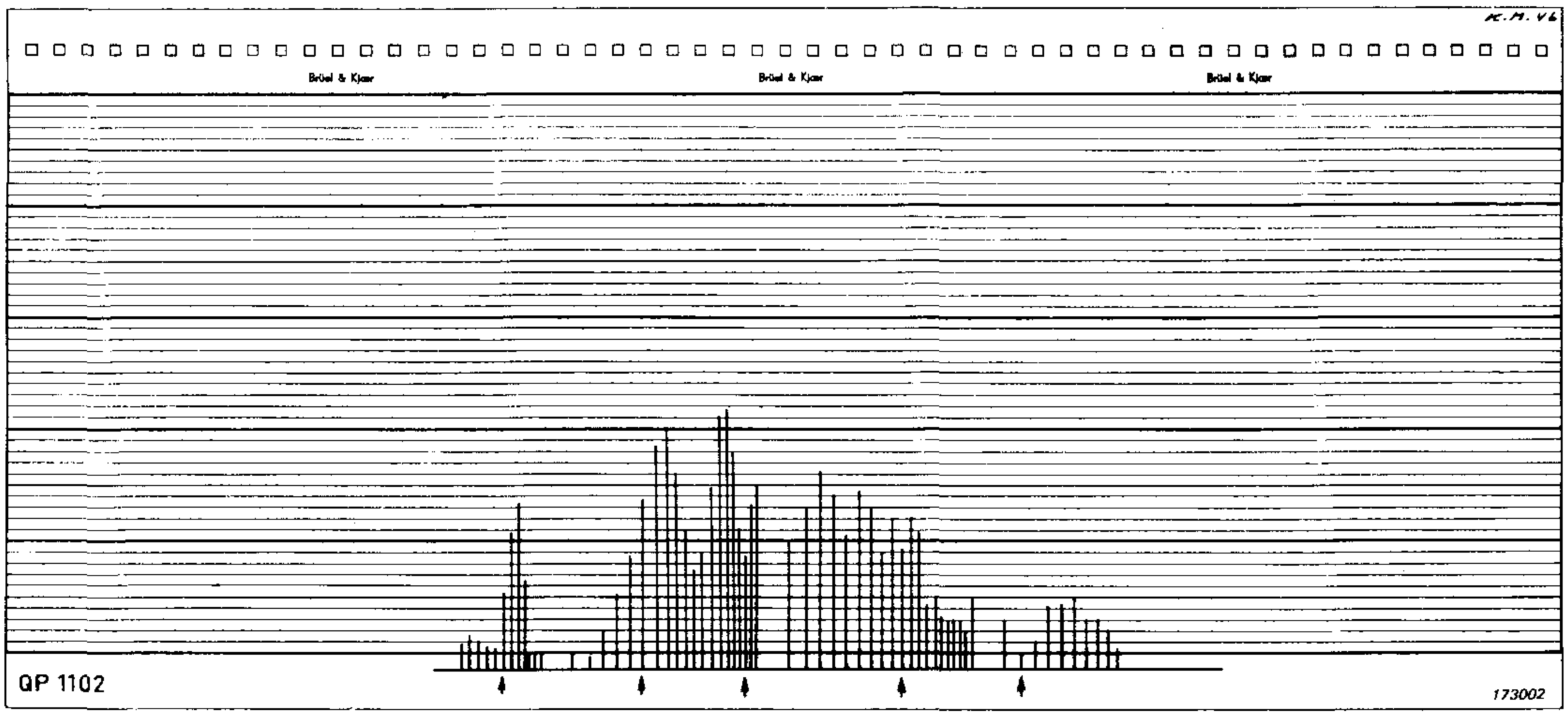

Fig.6. Example of semitone-sampled spectrum, frequency range 100 – 3350 Hz, maximum selectivity. Recorder settings: Potentiometer range: 50 dB, RMS, L.L.F.: 20 Hz, W.S.: 125 mm/sec (100 mm paper width), P.S.: 0.3 mm/sec (1 : 10 gear disengaged), Drive shaft speed: 12 r.p.m.Arrows indicate the musical notes c_0, c_1, c_2, c_3,

The Flexible Shaft is connected to the Drive Shaft II mechanical output of the recorder and in the described application its speed of 12 r.p.m. has been found convenient. Then the frequency tuning axis of the analyzer rotates at an angular velocity of 2.88 degrees per second (when driven), and a convenient duration of the quasistable state of the monostable circuit is about 2 sec.

In the basic mode of operation, apparent from this description a SPL density spectrum of a quasistationary signal is obtained in the form of *discrete spectrum samples* (recorded as vertical lines) separated by semitone intervals. Therefore, the spectrum can be recorded also on the uncalibrated paper, the Paper Speed of 0.3 mm/sec being then most suitable, see Fig.6.

The remote controlled semitone tuning, however, is intended also for automatic analysis of predetermined segments of a non-stationary signal, further for a quasi-three-dimensional analysis of time-varying semitone-quantized spectra, and for some other applications.

All these procedures, including that described here, are to be fully automated and programmable for a preset number of semitones. Therefore, the switching circuit is in fact slightly more sophisticated than it were necessary for the OR-function alone, because the described 1/12-octave automatic tuning arrangement, as already said, is intended to form just a central part of a multipurpose measuring set.[2] This, however, would exceed the scope of this brief communication and will be described later.

Acknowledgements

I wish to acknowledge my thanks to Mr. F.D. Jacobsen, B & K, for his interest in this topic and for his efficient assistance connected with the manufacturing of the semitone switch board, and to Mr. T. Szabo, SAV Bratislava, for his careful technical assistance.

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2) Among other functions, the switching circuit should also prevent from being analyzed the artifacts that may be caused during the tape (loop) starting time. This should be as short as possible.
(More elaborated tape systems have also been described in the literature, e.g. B & K Tech.Rev. 2/1970, pp. 21-25.)
Naturally, it would be much more attractive to have the signal stored in a digital memory instead of an analog tape loop memory. But even if we consider a frequency range of, say, 16 kHz, and a 46-dB signal-to-noise ratio, then the information flow represented by the signal amounts about 220000 bit/sec. With only 8.5-second samples (as are those used so far in the described equipment), this would require almost 2 Megabits of memory and the cost would exceed many times that of the basic equipment.

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).

Bekesy-Audiometry with Standard Equipment

by

B. Møhl *)

In laboratories investigating psycho-acoustic phenomena, the need often arises to measure the auditory sensitivity of human subjects. In many cases, the nature of the experiment does not justify the expenditure of specialised automated equipment and the audiogram is therefore taken manually, using available generators, sound sources etc. The purpose of this paper is to describe an automated method, which requires only general purpose equipment and yet gives a permanent record on calibrated paper.

The set-up used in our laboratory is shown in Fig.1. The output of a 1 volt RMS ac source (e.g. a filament transformer) is applied to the compressor input of an audiogenerator (Brüel & Kjær Type 2010) via a switch. The output of the generator drives a set of calibrated earphones (Beyer DT 480S) and is itself driven through a flexible shaft by a Level Recorder Type 2305 which is synchronized to it.

*) Institute of Zoology, Lab. A. University of Aarhus

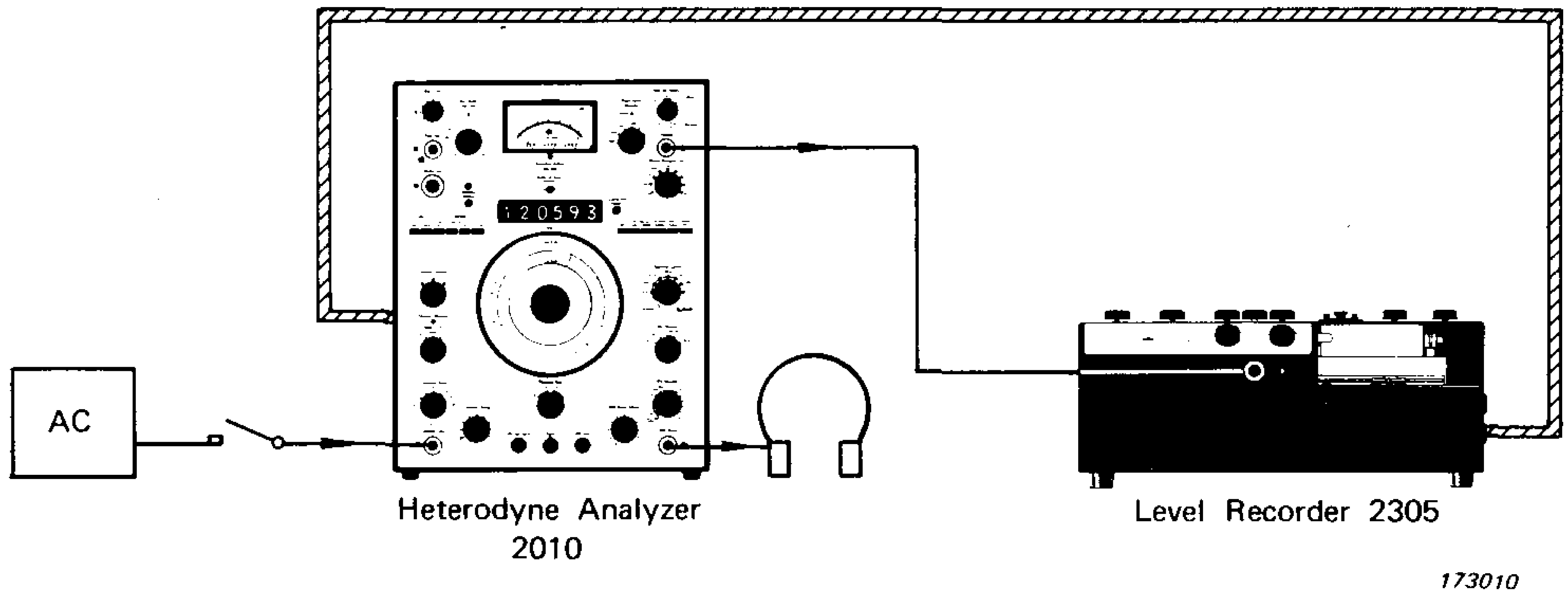

Fig. 1. Set-up for Bekesy audiometry

The principal feature of the set-up is the utilization of the compressor facility for adjusting the sound level (rather than employing a reversible, motor-driven attenuator). The compressor circuit is designed to adjust the output of the generator relative to a reference level, characteristically provided from a microphone in a closed servo-loop. In the set-up in Fig.1, the servo loop is opened, and the control voltage is supplied from the switch. Consequently, when the switch is open, the output will increase with a constant rate (selectable in steps from 3 to 1000 dB/sec) until the upper limit of the dynamic range of the compressor circuit is reached. With the switch closed, the output will decrease with a constant rate and eventually reach the noise level. The subject is required to press the switch, when a tone is heard and release it as soon as the tone disappears. Accordingly, the compressor circuit will adjust the input to the earphones to the threshold range (Fig.2).

An alternative way of describing the operation of the system is to consider that a new servo-loop is established, with the subject as an integral part. The reference level is then the threshold of the subject.

In order to establish the dynamic range of the recorder, the generator and the subjects, it is advisable first to open the switch, switch on the compressor circuit, set the generator output voltage at maximum output without clipping, turn the generator frequency to the lowest desired frequency (e.g. 100 Hz) and adjust the generator output step attenuator to the nearest step just above the threshold of the subject. This procedure ensures operation at the upper part of the compressor range. Next, the input potentiometer, and attenuator of the level recorder are adjusted so that the pen is approximately 10 dB above the scale used for the above mentioned setting of the generator. (For generators without an independent recorder output, it may be necessary to amplify the generator output before applying it to the recorder). The final step in the setting-up procedure requires a short track-

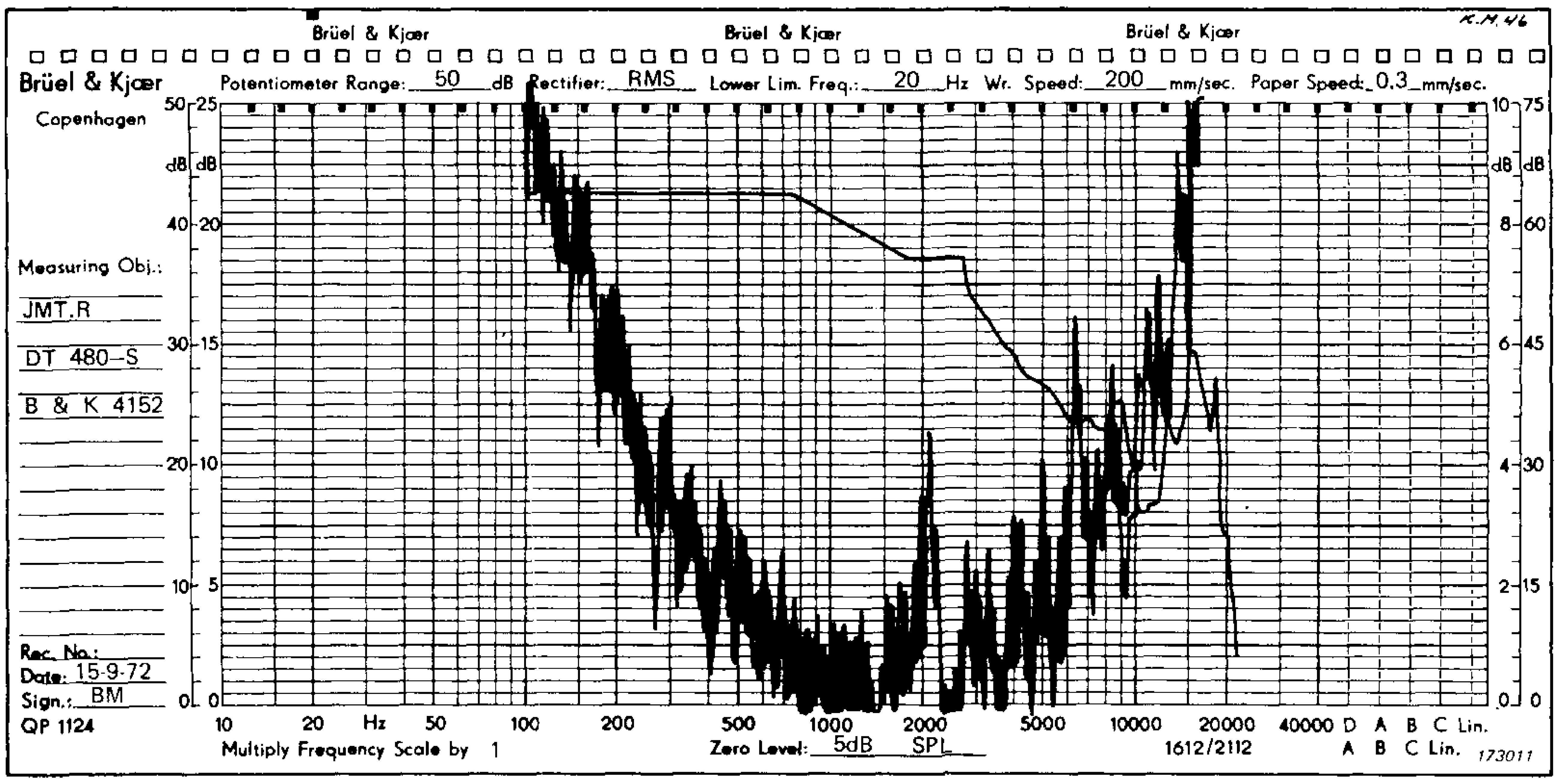

Fig.2. Audiogram of 25 year old male and calibration curve of the earphone, using a 6 cc coupler

ing of the threshold by the subject at his most sensitive frequency in order to ensure that the level is within the range of the recorder. If not, the experimenter has to decide which frequency range is of most importance and adjust the recorder accordingly. Alternatively, the attenuation of the recorder can be changed during the run, in which case, however, the set-up cannot be considered fully automatic.

Following the initial procedure, the generator is set at the lowest frequency of the audiogram, and the level recorder pen is positioned at the same frequency on the calibrated recording paper. The mechanical drive from the recorder is locked to the generator and the subject asked to track his threshold by pressing the switch on and off. When a stable response is obtained, the paper drive of the recorder is turned on, and the subject tracks his threshold as frequency increases (Fig.2).

To evaluate the obtained audiogram for absolute sensitivity, the frequency response of the earphone and the voltage applied must be known. The accuracy is improved if, after completion of the audiogram, the earphone is calibrated by an artificial ear on the same recording paper as was used for the audiogram. When justified by the nature of the problem, circuits can be inserted to normalize the response of the earphone as well as the combination "earphone-normal subject" as used in clinical audiometry. In other cases, where only changes in sensitivity are of importance, calibration procedures can be omitted altogether.

The limitations of the arrangement (apart from those set by the earphones) are the dynamic range of the compressor circuit (quoted by manufacturer to be 50 dB but measured to exceed 70 dB) and the dynamic range of the recorder. The adjustment procedure outlined ensures, that at low signal levels, noise will not mask the signal either for the listener or for the recorder. The decrease rate is dependent upon the magnitude of control voltage applied, as well as the compressor speed setting. The increase rate depends only on the latter. For symmetrical up-down regulation we have found 1 V RMS to be suitable.

Due to the versatility of general purpose instruments, the set-up is well suited to test the influence of settings for frequency increment rate, attenuation rate etc. For audiograms, it has been found that a compressor speed of 3 dB/sec, a writing speed of 200 mm/sec and scanning of the frequency range from 0.1 kHz to 10 kHz (log scale) in about 5 min. is suitable in most cases.

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