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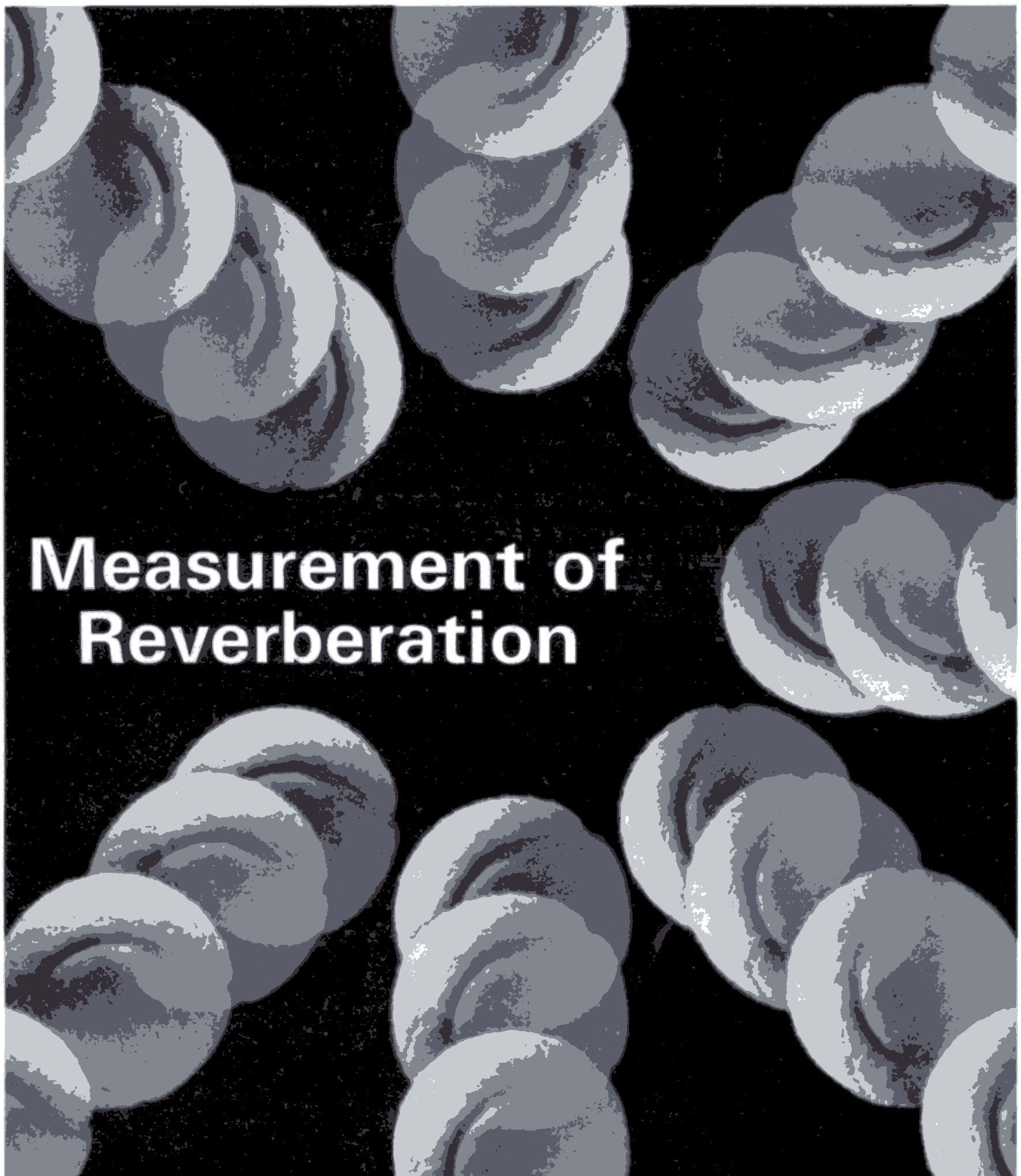
Measurement of Reverberation



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On the Measurement of Reverberation

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ABSTRACT

After a brief "historical" introduction some implications of reverberation theory and the wave nature of sound are discussed. Various methods of exciting the room acoustically are mentioned, and the development of reverberation measurement technique and evaluation over the past decades are reviewed in general terms. It is pointed out that to determine the acoustical quality of rooms it seems necessary to study certain details of the reverberation process, and that a simple statement of an average reverberation time is a rather insufficient measure. On the other hand, the detailed fluctuations in the reverberation process which are caused by beats between closely spaced room resonances should preferably be more or less averaged out so that more typical average trends can be observed.

Several methods may be used for this purpose, all of which are of a more or less integrating type. One of the methods, described by M. R. Schroeder, has been relatively thoroughly investigated at Brüel & Kjær. It has been found that use of this method may result in exceptionally smooth reverberation decay curves, and in comparison with other known methods, a considerably more accurate determination of the initial slope of the curve is possible. This is a great advantage, as this part of the decay curve is supposed to be of special importance when considering the acoustic quality of a room. However, in cases where only an average reverberation time determination is required, i.e. in cases of sound absorption measurements, the method seems to bear no significant advantage over already existing measurement schemes.

SOMMAIRE

Après une brève introduction donnant l'historique de la théorie de la réverbération et de la nature des ondes sonores, on en discute quelques implications.

Diverses méthodes d'excitation acoustique de la salle sont citées et l'on passe en revue, en termes généraux, les progrès de la technique de la mesure et de l'évaluation de la réverbération durant les dernières décades. On remarque que pour déterminer la qualité acoustique des salles, il paraît nécessaire d'étudier certains détails du processus de réverbération moyen constitue une mesure plutôt insuffisante. D'autre part les menues fluctuations dans le processus de réverbération, qui sont causées par des battements entre résonances de la salle très peu espacées, devraient de préférence être plus ou moins adoucies de telle sorte qu'on puisse observer des tendances moyennes plus typiques.

Plusieurs méthodes peuvent être utilisées à cet effet, chacune faisant plus ou moins à appel à l'intégration. Une des méthodes, déscrite par M. R. Schroeder, a fait l'objet d'investigations relativement poussées chez Brüel & Kjær. On a trouvé que cette méthode peut donner des courbes exceptionnellement régulières et, en comparaison avec d'autres méthodes, une détermination considérablement plus précise de la déclivité initiale de la pente est possible. Cela prsente un grand avantage car cette partie de la courbe d'extinction est considérée comme de vant présenter une importance spéciale lorsque l'on étudie la qualité acoustique d'une pièce. Cependant dans les cas où l'on ne désire la détermination que d'un temps de réverbération moyen, c'est-à-dire dans les cas de mesure d'absorption sonore, la méthode semble ne devoir présenter aucun avantage notable par rapport aux processus de mesure déjà utilisés.

ZUSAMMENFASSUNG

Nach einer kurzen "historischen" Einführung werden einige Folgerungen behandelt, die sich aus der Nachhalltheorie und der Wellennatur des Schalls ergeben. Verschiedene

Methoden der akustischen Anregung einem Raumes werden erwähnt, und in einem allgemeinen Überblick wird die Entwicklung der Nachhall-Meßtechnik und -Rechenverfahren während der letzten Jahrzehnte dargestellt. Es wird betont, daß zur Bestimmung der akustischen Qualität eines Raumes offenbar gewisse Einzelheiten des Nachhallvorganges studiert werden müssen und daß es nicht genügt, einfach eine mittlere Nachhallzeit anzugeben. Andererseits sollten die einzelnen Fluktuationen im Abklingvorgang, die auf Schwebungen zwischen eng benachbarten Eigenresonanzen des Raumes beruhe, nach Möglichkeit mehr oder weniger ausgeglichen werden, so daß der typische mittlere Verlauf beobachtet werden kann.

Für diesen Zweck gibs es verschiedene Methoden, die mehr oder weniger auf eine Integration hinzielen. Eine dieser Methoden, die M. R. Schroeder beschrieben hat, wurde bei Brüel & Kjær relativ sorgfältig untersucht. Man fand, daß diese Methode äußerst glatte Nachhallkurven ergibt und daß im Vergleich zu anderen bekannten Methoden eine bedeutend genauere Bestimmung der Anfangsneigung der Kurve möglich ist. Das ist sehr vorteilhaft, weil dieser Teil der Kurve anscheinend für die Beurteilung der akustischen Qualität eines Raumes besonders wichtig ist. In den Fällen jeoch, wo nur die Bestimmung einer mittleren Nachhallzeit erforderlich ist, d.h. bei Schallabsorptionsmessungen scheint die Methode keinen besonderen Vorteil gegenüber den bereits existierenden zu besitzen.

Introduction.

In the year 1922 Wallace Clement Sabine's "Collected Papers on Acoustics" were published. With this event the real foundation of modern room acoustics was laid. In his "Collected Papers" Sabine had, for the first time, introduced scientific principles and methods to the study of architectural acoustical problems and had answered many of the fundamental questions connected with sound absorption and transmission. As Sabine had clearly seen, a distinct relation exists between the reverberation of the sound in a room and the sound absorption of the room. Much of Sabine's work was, because of this, dedicated to the determination of the sound absorbing properties of various materials commonly used by architects in the design of lecture and music halls.

Considering the measuring apparatus available for the measurement of reverberation (and thus sound absorption) at the time Sabine made his famous investigations, his results are really remarkable. The sound source used in many of his experiments consisted of an organ pipe (middle c, 512 Hz) fed from a double tank, water-sealed air supply and controlled by an electropneumatic valve, similar to that used at that time in large church organs. As microphone, "simply" the ear of the experimenter was used and the time measuring device consisted of an electrically controlled chronograph. The reverberation time was then determined as the time taken from shutting off the sound from the organ pipe until it could be no longer heard in the room. Normally this corresponded to a change in sound level of approximately 60 dB, a level difference which has since been used for the definition of the reverberation time.

To allow for a reasonably accurate and reproducible "definition" of the lowest sound to be percieved by the listener, the external noise penetrating into the room where measurements were made had to be very low and consistent, and

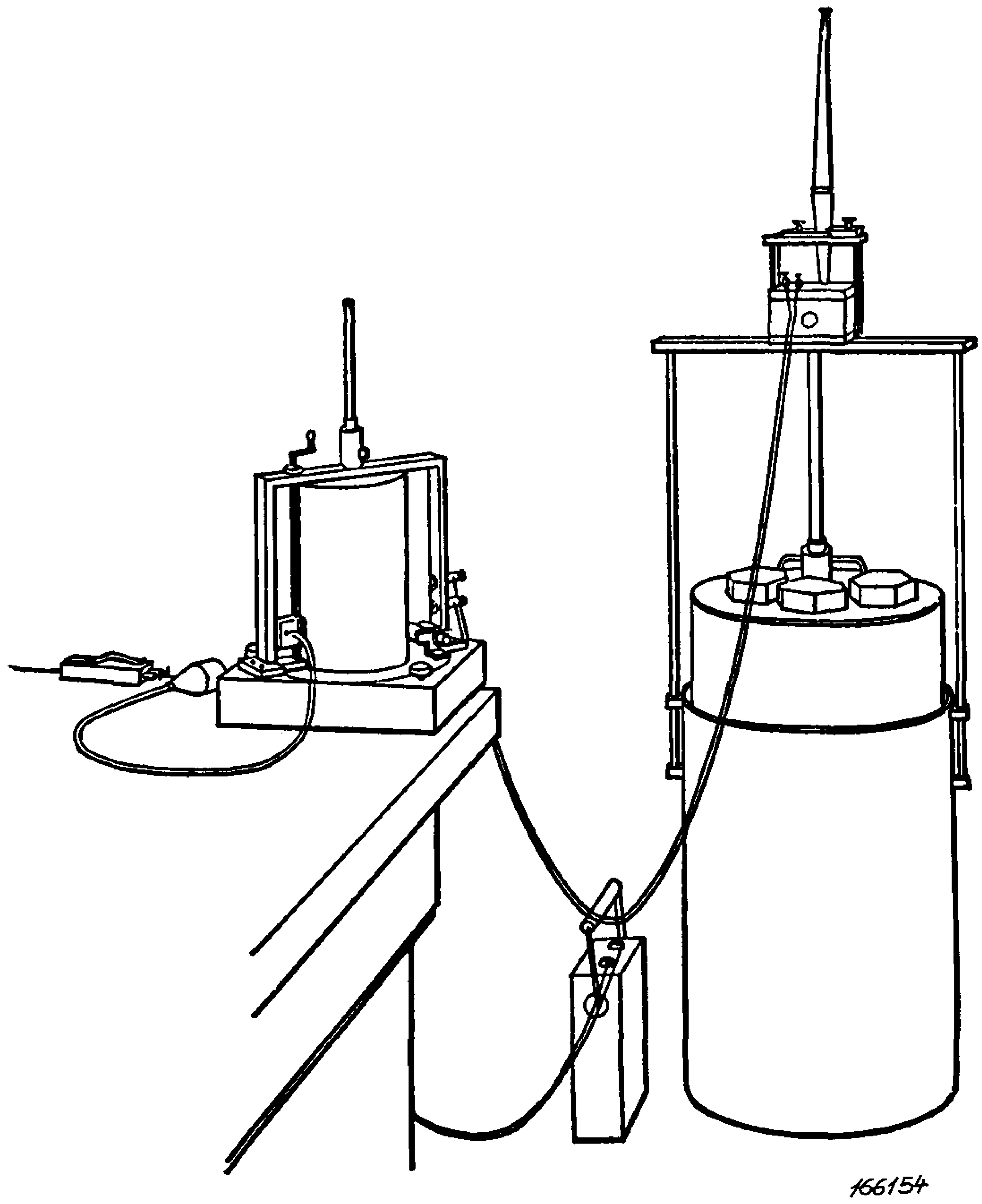

Fig. 1. Chronograph, battery and air reservoir, the latter surmounted by the electropneumatic valve and organ pipe. (W. C. Sabine).

most of the measurements were therefore carried out at night. This was, of course, rather inconvenient but the reproducibility of the measurements was surprisingly good. The average deviation of ten measurements from the mean was initially some 0.03 seconds and the maximum deviation 0.05 seconds. After "several nights practice", however, the "personal" factor could be reduced to 0.02 seconds.

Even though more "objective" methods of measuring sound pressure levels and level changes had been suggested it was not until the advent of electronic amplifiers and the invention of reliable measuring microphones that such methods were seriously introduced and utilized in the measurement of reverberation. Also, by that time complete electronic control of the sound source was possible, and loudspeakers driven from electronic, variable frequency sources replaced the organ pipe as sound producing instruments.

In this article a brief review of some of the more important methods of reverberation measurements which have been developed and utilized since the time of Sabine will be made, including a relatively new method, first published by M. R. Schroeder of Bell Laboratories in the Journal of the Acoustical Society of America, March 1965.

Some Basic Considerations.

Certain aspects of the reverberation process, and of room acoustic problems in general, can be understood only by taking into account the wave nature of sound. This has long been recognized (Sabine himself was very much aware of it) but the full importance of it was probably first seriously considered by Knudsen (1932) who showed experimentally that reverberant sound has characteristic frequencies which are properties of the room and not necessarily only that of the source which initiates the reverberation. These characteristic frequencies can be found by solving the three-dimensional wave equation governing the sound propogation in the room and are commonly termed room resonances. They set up a complicated scheme of standing wave patterns, which change completely as the frequency of the sound changes. Now, as an abrupt change in sound level (the source is switched off) can be represented by a complex frequency spectrum the reverberation process contains signals with more than one frequency even if the steady state sound source consisted of a pure tone only. If these signal frequencies coincide with some of the room resonances, beats will occur and the reverberation decay curve will show a number of fluctuations. The less damped the room resonances are, and the "purer" the tone of the sound source, the greater will the fluctuations in the reverberation process be. It has therefore become common practice to use a "band of frequencies" as sound source, centered around that particular frequency for which it is desired to study the reverberation process. In this way the number of room resonances taking part in the decay will be so great that the fluctuations in the decay curve will, more or less, average out. The "band of frequencies" may consist of a band of random noise, a warbled tone, a number of closely spaced pure tones (often called a multitone) or simply a toneburst or pistol shot.*)

To ensure the excitation of as great a number of room resonances as possible within the frequency band produced by the sound source, the position of the source in the room must also be considered. In small rooms a suitable position would be in one of the room corners. In large rooms, however, the location of the source is of no great importance, and in rooms which are used for special purposes the source is normally placed at the relevant position (in a theatre on the stage, etc.).

Also the microphone used for picking up the sound is, in smaller rooms, placed in a corner or a number of different microphone positions are used and their outputs averaged.

The output signal from the microphone is then fed to an electronic amplifier, a rectifier circuit and some sort of recording device indicating the variation in sound level with time. It is particularly in the method of indication, that a

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*) A very sharp pulse contains infinitely many frequencies over a wide frequency band. A more narrow band for study is then obtained by filtering the output from the microphone before the reverberation process is recorded.

wide variety of equipment has been suggested during the past decades and in the following a brief description of some of these is given.

Methods of Reverberation Time Evaluation.

One of the first objective recording arrangements utilizing electronic equipment consisted of a microphone, an amplifier and a recording oscillograph. This arrangement was described by E. Meyer in 1927 but the measured results were difficult to interpret, partly because the reverberation decay was recorded to a linear scale, and partly because a "pure" tone was used as sound source which resulted in a very irregular decay curve. In 1928 Meyer and Just introduced the warble tone in their experiments and considerably smoother decay curves were obtained. Still, however, the recording was made to a linear scale. A further improvement with respect to smoothness of the recorded reverberation decay curve was introduced in 1930 by W. Kuntze who "averaged" the outputs of two different microphones.*) Other methods of sound field "diffusion" were also introduced, like rotating, sound reflecting panels (and even rotating sound sources, even though their importance with respect to reverberation measurements might be considered doubtful).

As the determination of reverberation times from the kind of reverberation decay recordings described above required a considerable amount of tedious work, a number of experimenters designed various "automatic" measurement equipment. Around 1930 Meyer, Strutt, Wente and Bedell et. al. described a series of such designs. Common to all of them was that as soon as the sound source was shut off an electric timing device was started. This device was again stopped when the output level from the microphone had fallen a predetermined amount. Similar, but often more refined automatic measuring devices, have been described in technical journals from time to time ever since. One of these might deserve a little more detailed description, because it allowed not only an automatic reverberation time measurement to be made, but also an automatic averaging of ten measurements was achieved. Furthermore, by making the lower switching level adjustable, the complete, averaged reverberation decay curve could be plotted point by point in a "simple" manner. This apparatus is described by Hunt in 1936 and is based on a "continuous" on-and-off switching of the sound source. The continuous on-and-off switching of the sound source, and thus the production of a "sawtooth" kind of signal, see Fig. 2, was not new, as it was described as early as 1931 by Hollmann and Schultes as "raumakustische Kippschwingung" (room acoustical sawtooth wave). However, the averaging procedure used by Hunt, as well as the adjustable lower switching level offered new possibilities for accuracy in reverberation measurements.

*) This method is still very much used, and the averaging of outputs from four to six microphones is not uncommon.

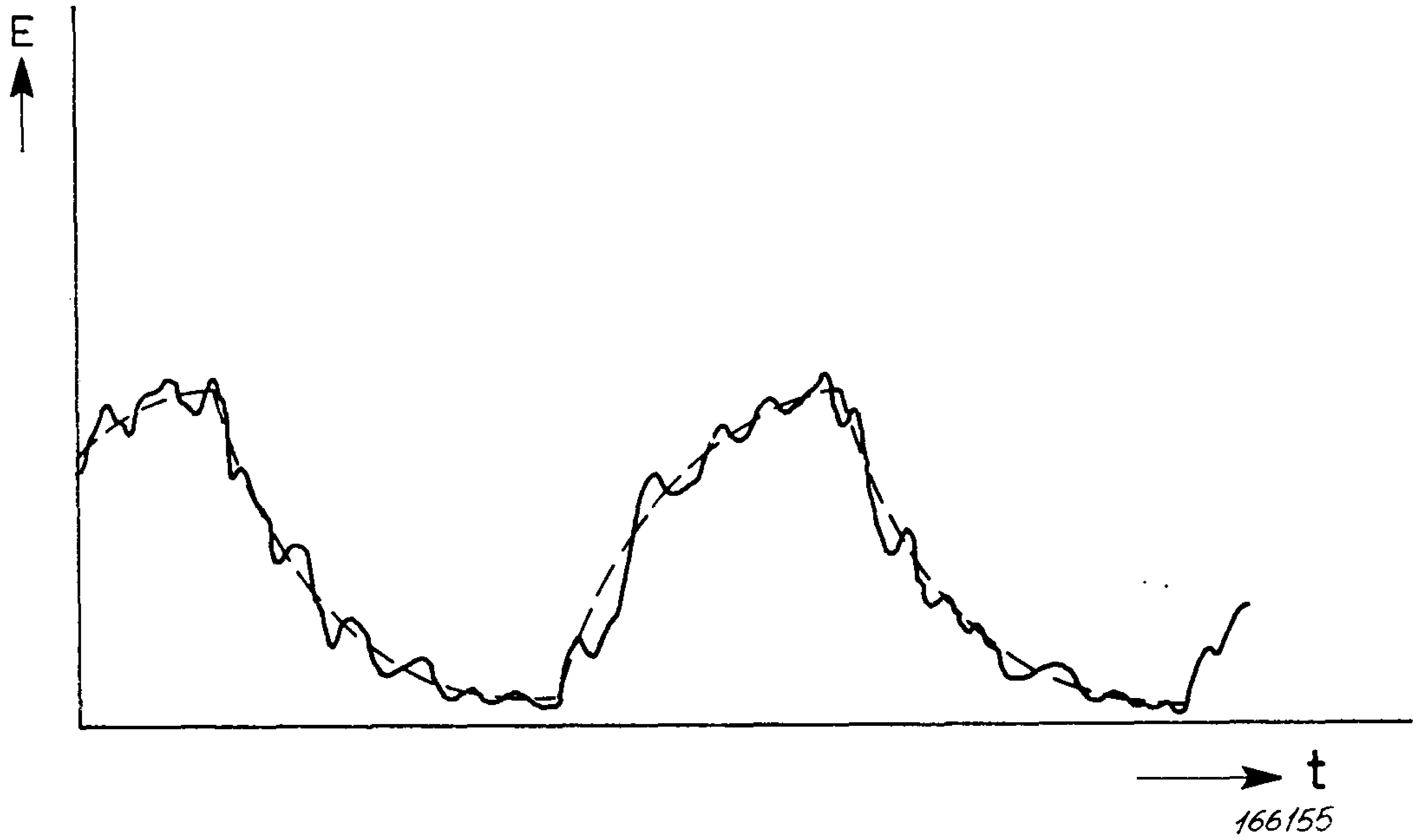

Fig. 2. Typical room acoustical sawtooth waves (Raumakustische Kipp-schwingung).

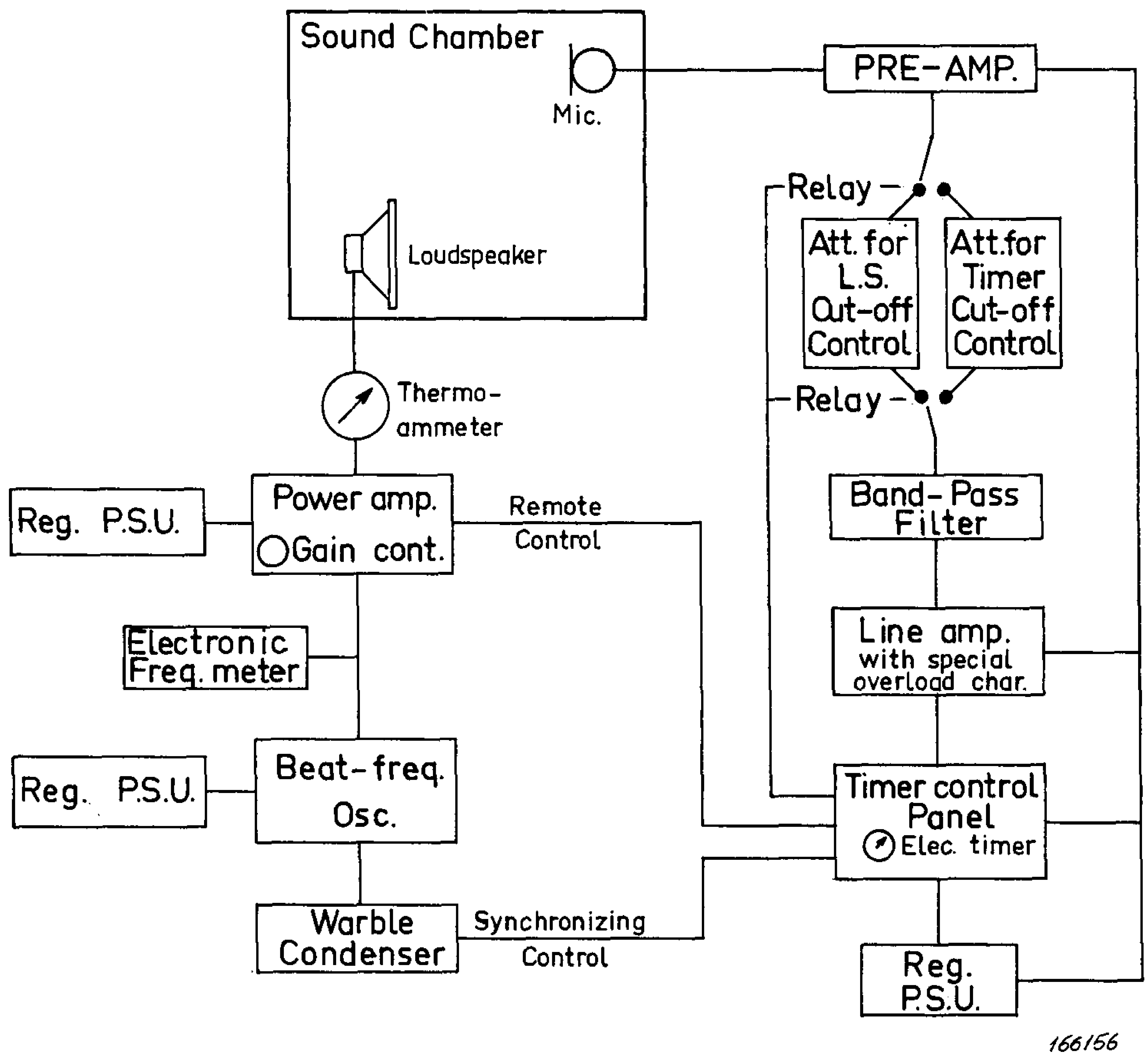

Fig. 3. Measuring arrangement for "continuous" on–off switching of the sound and automatic averaging over few measurements. Here also the lower switching level (off) can be adjusted in steps of 5 dB. (F. V. Hunt).

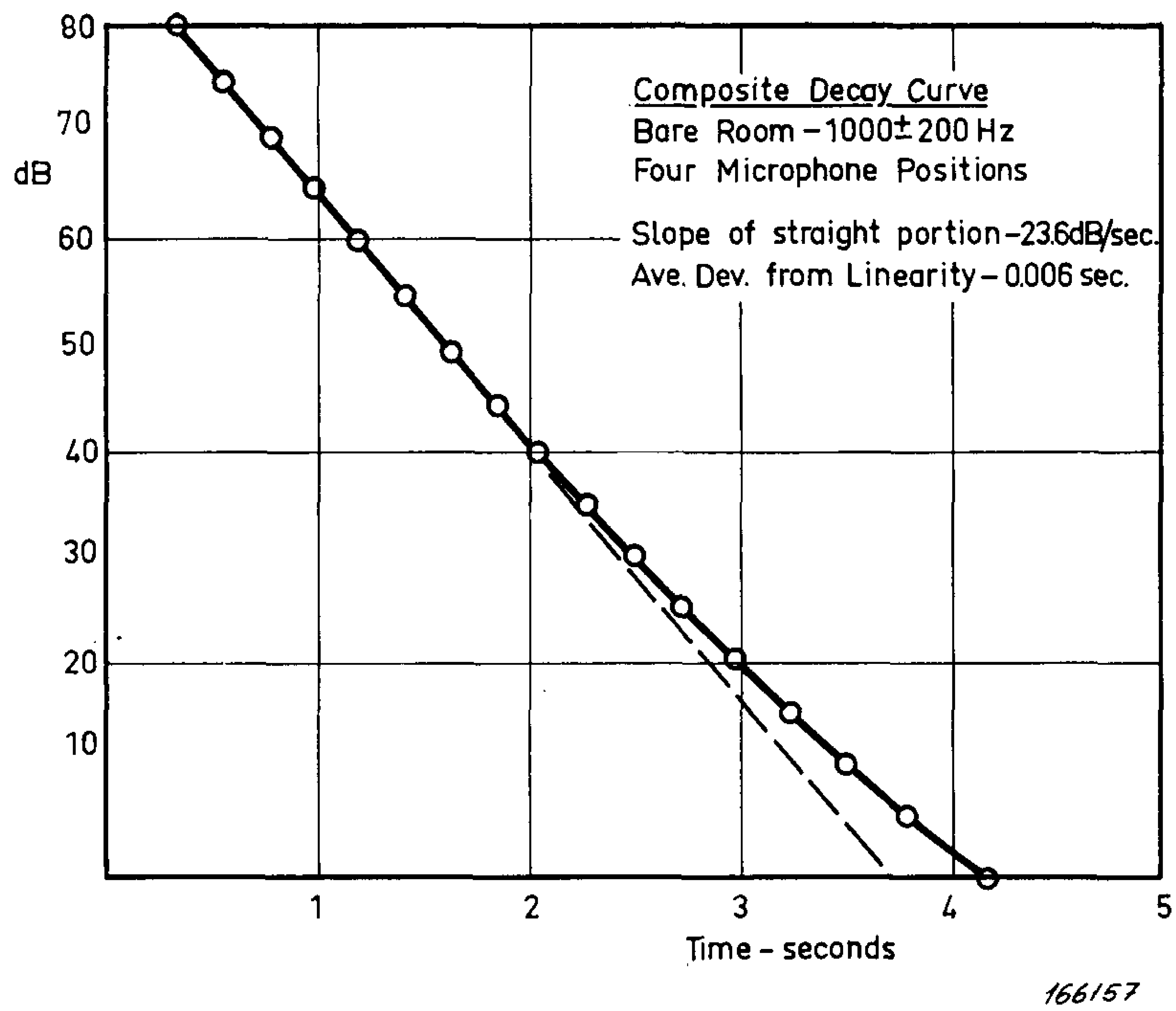

Fig. 4. Averaged reverberation decay curve obtained from measurements made by means of the arrangement shown in Fig. 3. (F. V. Hunt).

Fig. 3 shows a block diagram of the measuring equipment and in Fig. 4 an example of a reverberation decay curve obtained by means of this arrangement is shown. (Each point is here the average of 400 decays!). The operation of the equipment is briefly as follows:

When a predetermined sound level is attained in the room the source is turned off and an electric timer is started. As soon as the sound level has fallen to the (adjustable) lower threshold the timer is stopped and the source is turned on again. The recurrent cycle is stopped after 10 repetitions and the average time for a single decay can be read directly from the timer. Undesired level fluctuations in the reverberation process, which would upset a correct functioning of the device, are minimized by the use of an electric low-pass filter at the output of the rectifier and by using a band of noise or warble tones as sound source.

As the reverberation process is assumed to follow, basically, an exponential decay function the use of "exponential" measuring devices lay near at hand. Such devices have also been suggested but have never become really popular. One such method consisted in the charging of a capacitor whereby the potential on the capacitor under certain predetermined conditions, is a direct measure of the reverberation time. Another method consisted of comparing the discharge of a known RC circuit with the reverberation decay, a method which also led to the development of a reverberation measurement "bridge", Fig. 5. In the "bridge" circuit, which in principle is a comparison method, the reverberation process is "compensated' by a capacitor discharge

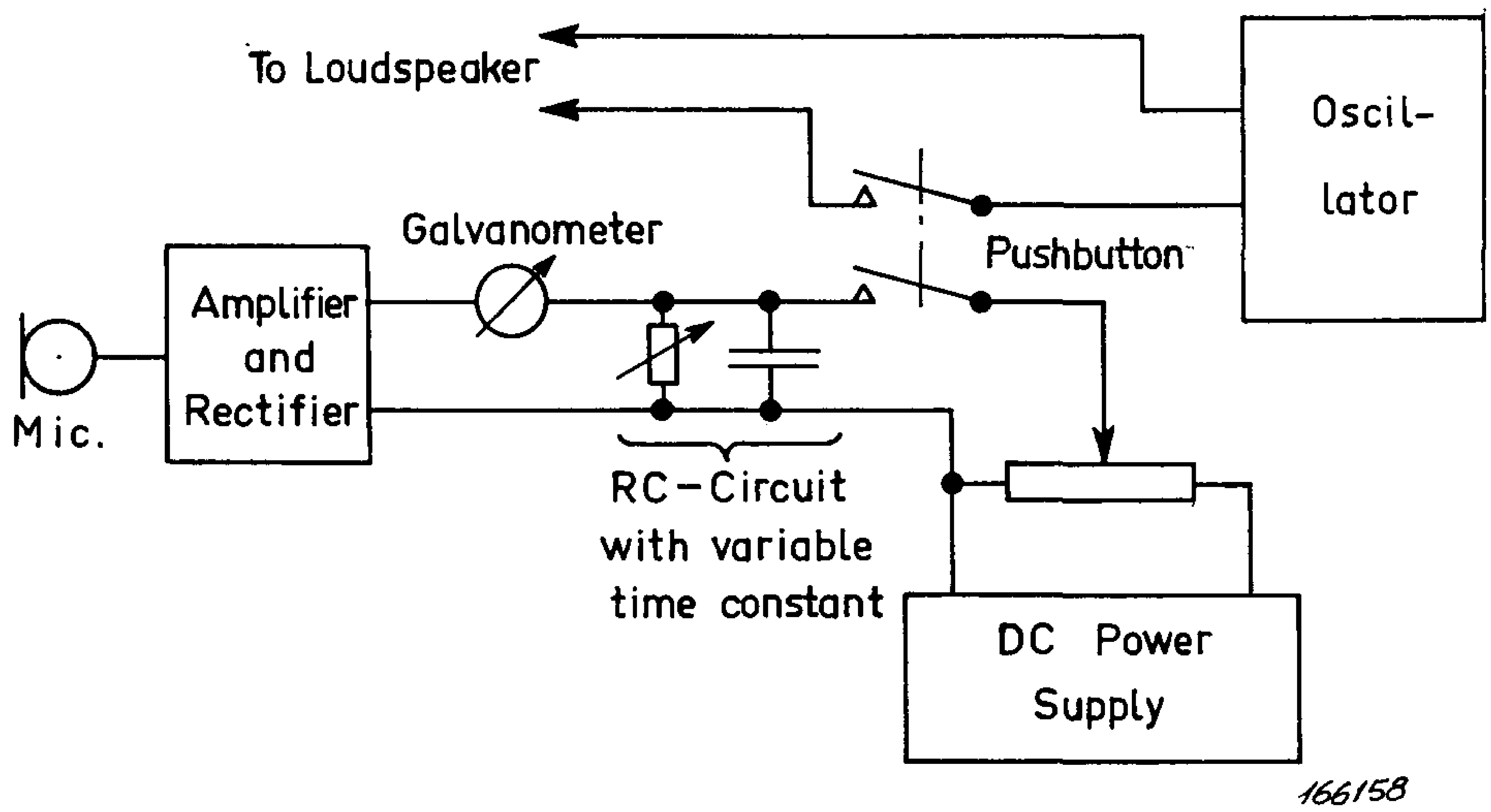

Fig. 5. Reverberation measurement bridge. (Olson and Kreuzer).

process, and the time constant of the RC circuit is adjusted until the galvanometer, G, in Fig. 5 deflects equally to the positive and negative side during the reverberation decay.

Most of the measuring arrangements described above were mainly designed to determine *the reverberation time* and they disregarded the details of the decay curve itself. It was soon found, however, that some knowledge of certain details of the curve was necessary in order to give a satisfactory description of the acoustics of the room in which reverberation measurements were made.

A great step in the direction of obtaining these details was made by the development of a logarithmic level recording device. (When recorded to a logarithmic scale an exponential decay curve becomes a straight line).

Such devices had been developed already in the early nineteen thirties by Ballantine, Meyer and Keidel, and von Braunmühl and Weber. While Ballantine used a logarithmic amplifier and linear recorder both Meyer and Keidel and von Braunmühl and Weber based their instruments on servo principles employing logarithmic input potentiometers. A variation of the "fluid" potentiometer used by Meyer and Keidel was also suggested by Nielsen in 1937.

The use of a servo system with logarithmic input potentiometer has two very great advantages in comparison with the use of logarithmic amplifiers and linear recorders: Firstly, the requirements to linear dynamic range of the rectifier are less stringent, and secondly the dynamic recording range of the recorder can be changed simply by changing the input potentiometer. For these reasons also more recently developed recorders employ the servo principle. The actual pen drive mechanism, however, is today normally different from that suggested by von Braunmühl and Weber, and commercially utilized in the socalled Neumann-recorder. In the Neumann-recorder the pen drive was effected by means of a small electric motor while newer recorders

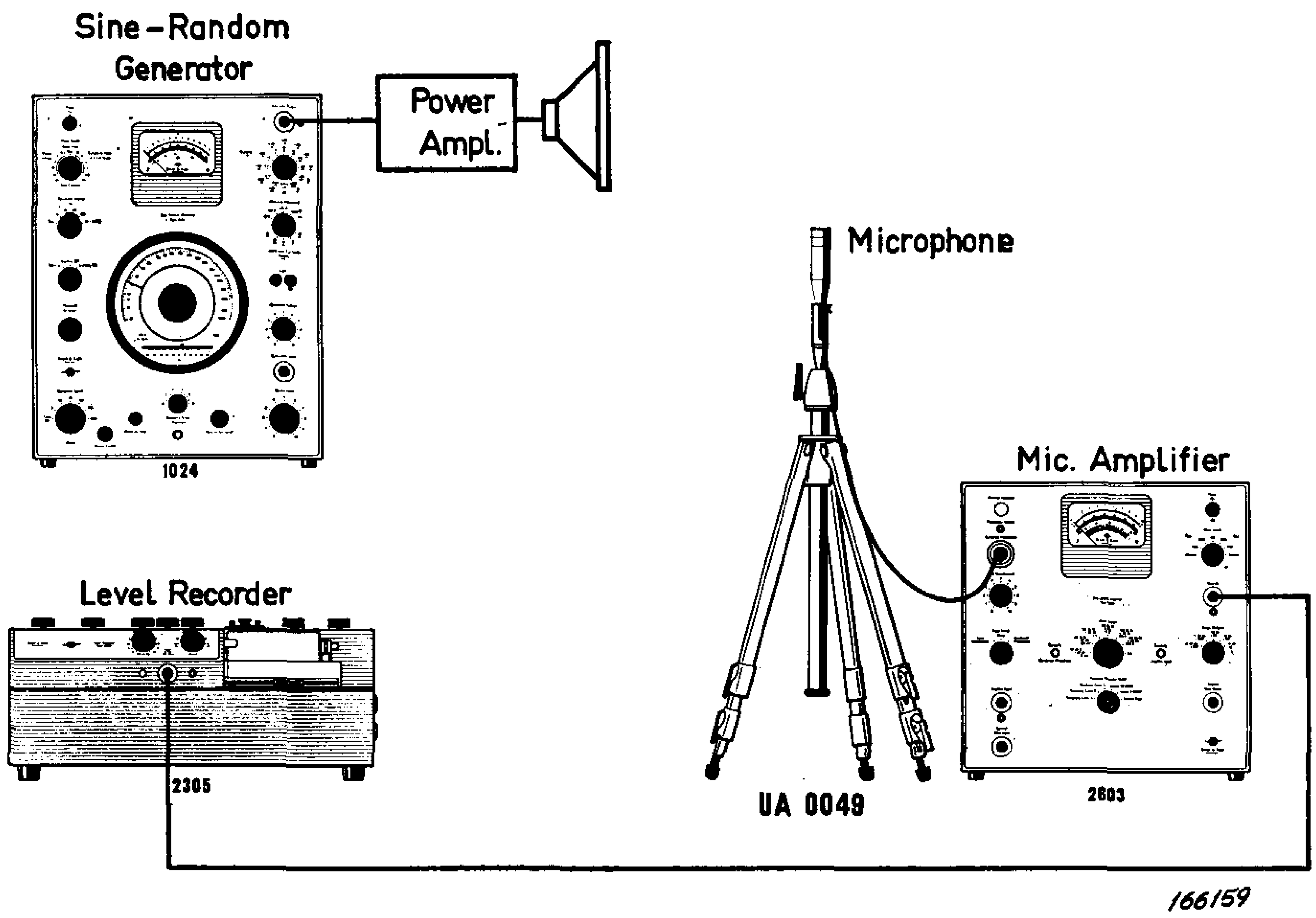

Fig. 6. Measuring arrangement utilizing a modern high speed level recorder for the recording of reverberation decay curves.

utilize direct electro-dynamic drives whereby much greater writing speeds can be obtained.

A typical measuring arrangement utilizing a modern, high-speed, logarithmic level recorder (Brüel & Kjær Type 2305) is shown in Fig. 6, and in Fig. 7 an automatically recorded decay curve is given. The curve reveals in this case one detail which would have been lost if only a reverberation *time* measurement had been carried out: it shows a typical "bend" in the curve at a certain decay level. This indicates the existence of a so-called "flutter"-echo which in most cases is a very undesirable room-acoustic effect and which occurs when a too great part of the total sound reflection takes place between two parallel walls.

However, even though certain details of the decay curve like the one just described are desirable to know, the detailed *fluctuations* should be "averaged" as much as possible to be able to obtain clear trends in the curves. By

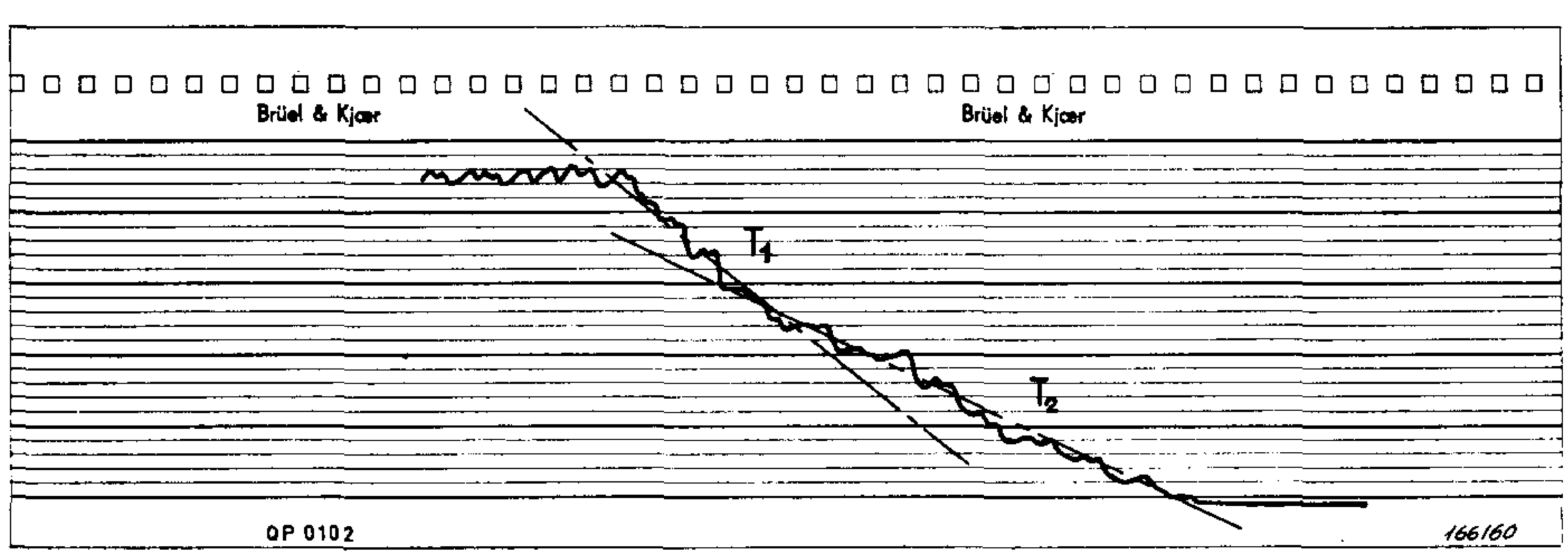

Fig. 7. Example of a recorded decay curve revealing the existence of a "flutter"-echo.

choosing a not too high writing speed of the recording device, still keeping the writing speed high enough to follow the main decay rates, these fluctuations may be minimized. On the other hand to be sure that important trends are not "averaged out" the writing speed should not be made too low whereby a certain part of the fluctuations will normally remain.

An interesting new method of obtaining reverberation data which, theoretically, should minimize the fluctuations and represent the true average response of an enclosure to interrupted random noise has, as mentioned in the introduction to this article been described lately by M. R. Schroeder.

To try and evaluate the practical use of this method a number of experiments have been carried out at Brüel & Kjær and the measuring arrangements used and the results obtained will be described and discussed below.

Schroeder's Integrated Impulse Method.

It can be shown analytically that the ensemble average*) of the square of the reverberation decay in an enclosure equals the time integral of the squared impulse response**) of the enclosure (see Appendix A).

In the analytical derivation, however, it is essential that the signal used to excite the enclosure for "normal" reverberation measurements consists of random noise. As random noise may be looked upon as a signal consisting of an infinite number of impulses, the relationship stated, which at first might look a little surprising, may in fact be quite understandable, and the method bears two great advantages:

1) The squared reverberation decay curve as averaged over an infinite number of measurements can be obtained from a single measurement of the integral of the squared impulse response of the enclosure (room).

2) The curve obtained in this way minimizes unwanted irregularities and emphasizes only the main trends in the decay, which may otherwise have been hidden by the irregularities.

This is not only a convenience in conjunction with the measurement of reverberation time but it also opens up possibilities for a closer study of the reverberation decay which again may lead to new and efficient room-acoustical criteria for the design of music halls and theatres.

After a study of the theory behind the new method the question remains how, in practice, the squared impulse response of the room can be measured, especially in restricted frequency bands. Schroeder here suggests a very elegant solution and points out that as long as the duration of the exciting signal is very short in comparison with the impulse response of the enclosure,

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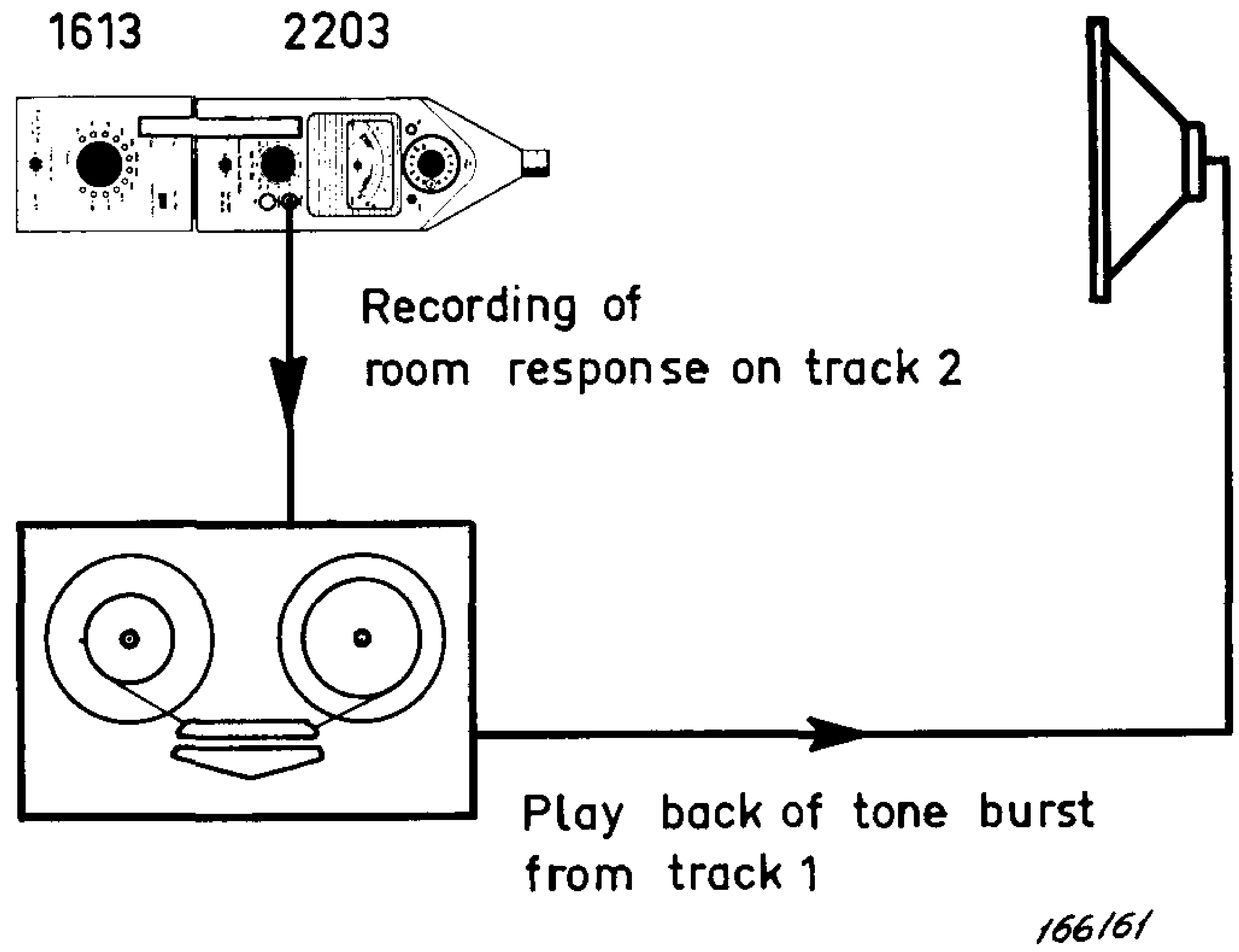
*) Ensemble average = Average over "infinitely" many experiments.

**) Enclosure impulse response = Response of the enclosure to a unit impulse excitation.



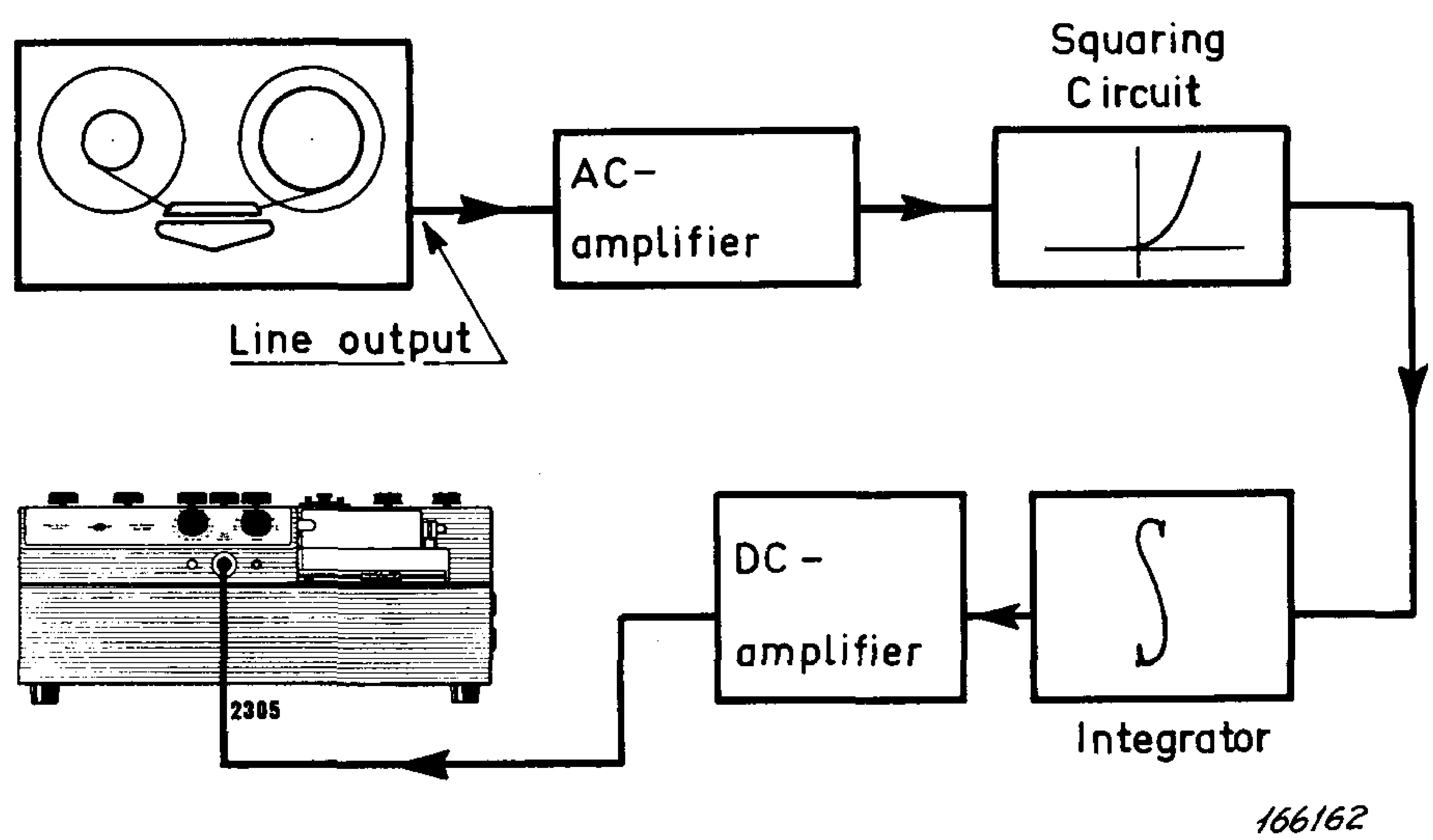


Fig. 8. Sketch of the measuring arrangement used for measurements according to the "integrated impulse method".
a) Arrangement used for measurement of room response.
b) Set-up used to determine the integrated square impulse response.

it may be considered an impulse type excitation. This is very important and makes it possible in general to restrict the frequency bands of the pulses used to 1/1 or 1/3 octave whereby also the frequency dependence of the reverberation decay can be studied.

The practical method of obtaining the squared and integrated impulse response as suggested by Schroeder is as follows (see also Fig. 8):

"Tone bursts whose spectra cover the desired frequency bands are radiated into the enclosure from a loudspeaker. The response of the enclosure to each tone burst is picked up by a microphone whose output is recorded on magnetic tape. The tape recording is played back in reversed-time direction. The output signal from the tape recorder is squared and integrated by means of an RC-integrating network. The voltage on the capacitor is then the desired decay curve (on a reversed-time scale)."

In the experiments carried out at Brüel & Kjær two different methods of impulse excitation were employed. One method consisted in firing a pistol and recording the room response directly onto magnetic tape. The filtering then took place during play back of the tape for evaluation. In the second case 1/3 octave and 1/1 octave filtered pulses (tone bursts) were radiated into the room via a loudspeaker. Also in this case the room response was recorded on magnetic tape. The length of the pulses were chosen to be a fraction of the period of the band center frequency $T = \left(\frac{1}{10 \times f_o}\right)$.

To allow for a practical evaluation of the "integrated impulse method" measurements utilizing impulse excitation were compared with measurements utilizing bands of random noise, all measurements being carried out under exactly the same experimental conditions.

Measurements were made in three different types of rooms: a medium sized

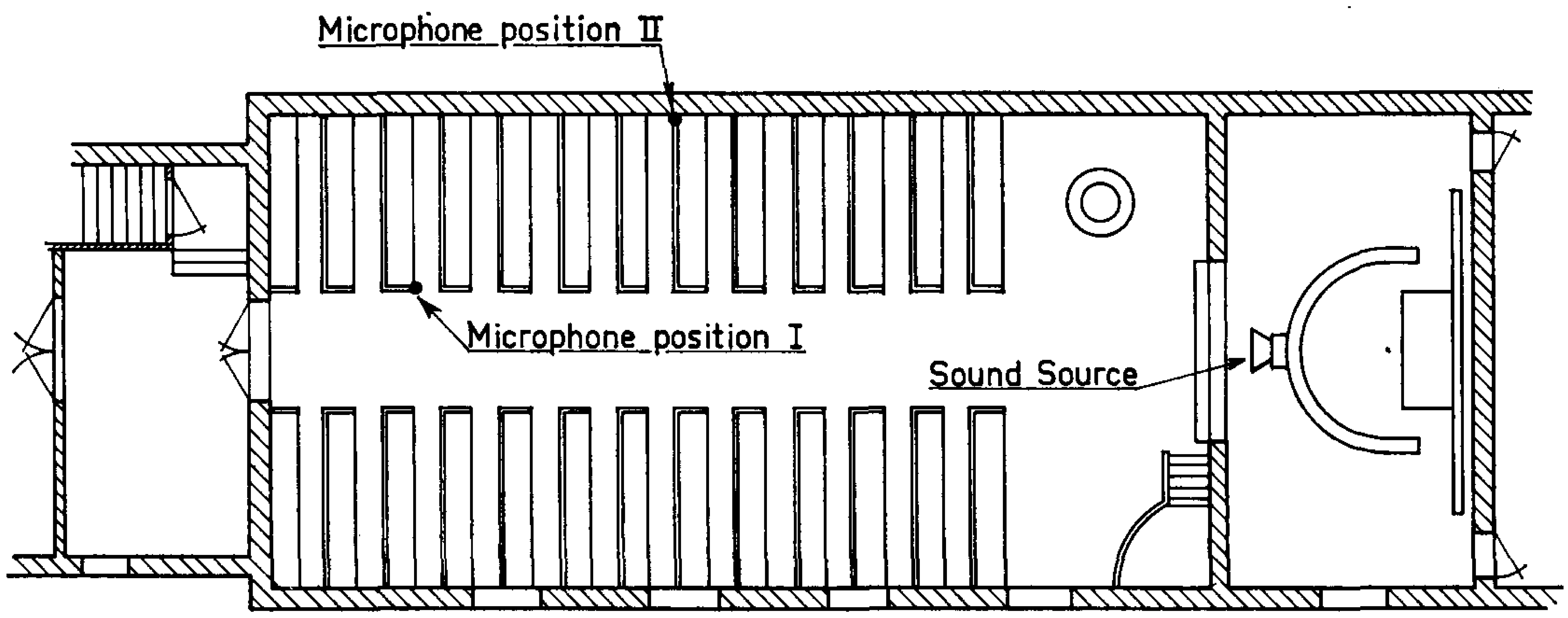


Fig. 9. Sketch of the church in which the measurements reported here took place. Source and microphone positions are also shown.

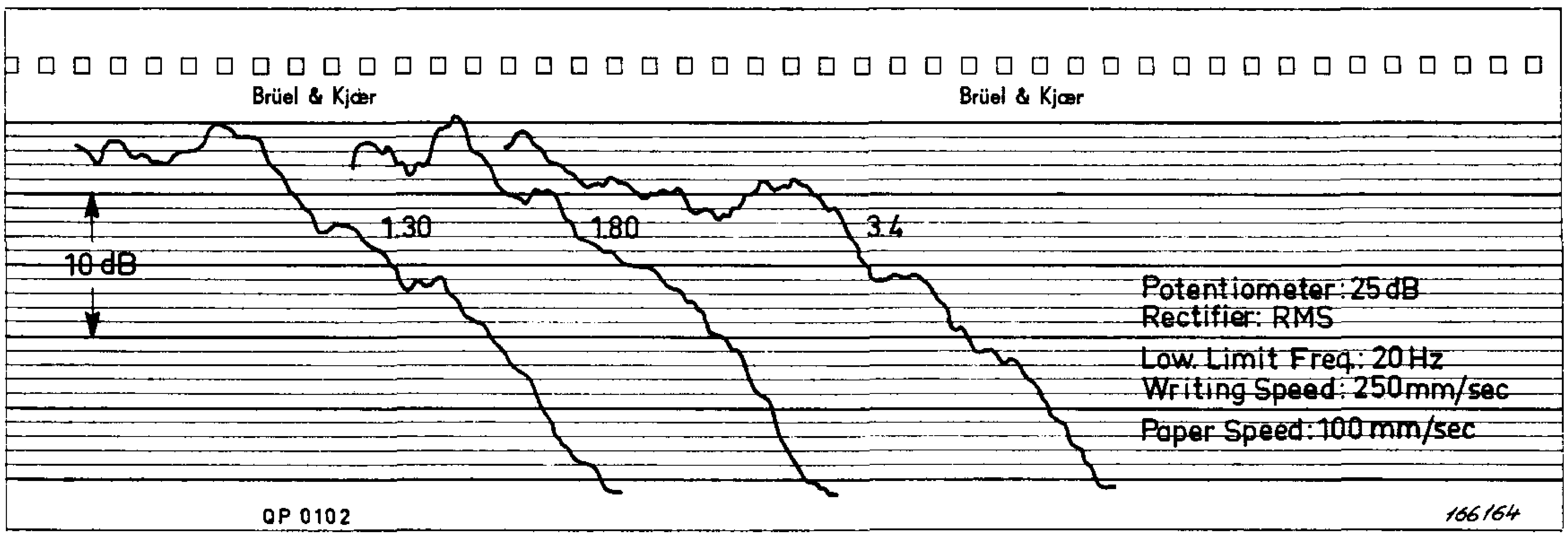


Fig. 10. Three typical reverberation curves obtained from measurements according to the interrupted noise method. (Measurement position II, 1/3 octave noise bandwidth, 1ooo Hz center frequency).

church with relatively long reverberation, a movie theatre with a slightly shorter reverberation, and an ordinary room with very short reverberation. In each case several microphone positions were investigated. Only the main measurements made in the church will be discussed in more detail in the following ,as the conclusions drawn from these measurements are representative for all of the investigations.

Fig. 9 shows a sketch of the church and two of the microphone positions used. In both microphone positions "sets" of 10 identical measurements were made to obtain a picture of the relative spread in results. Measurements were carried

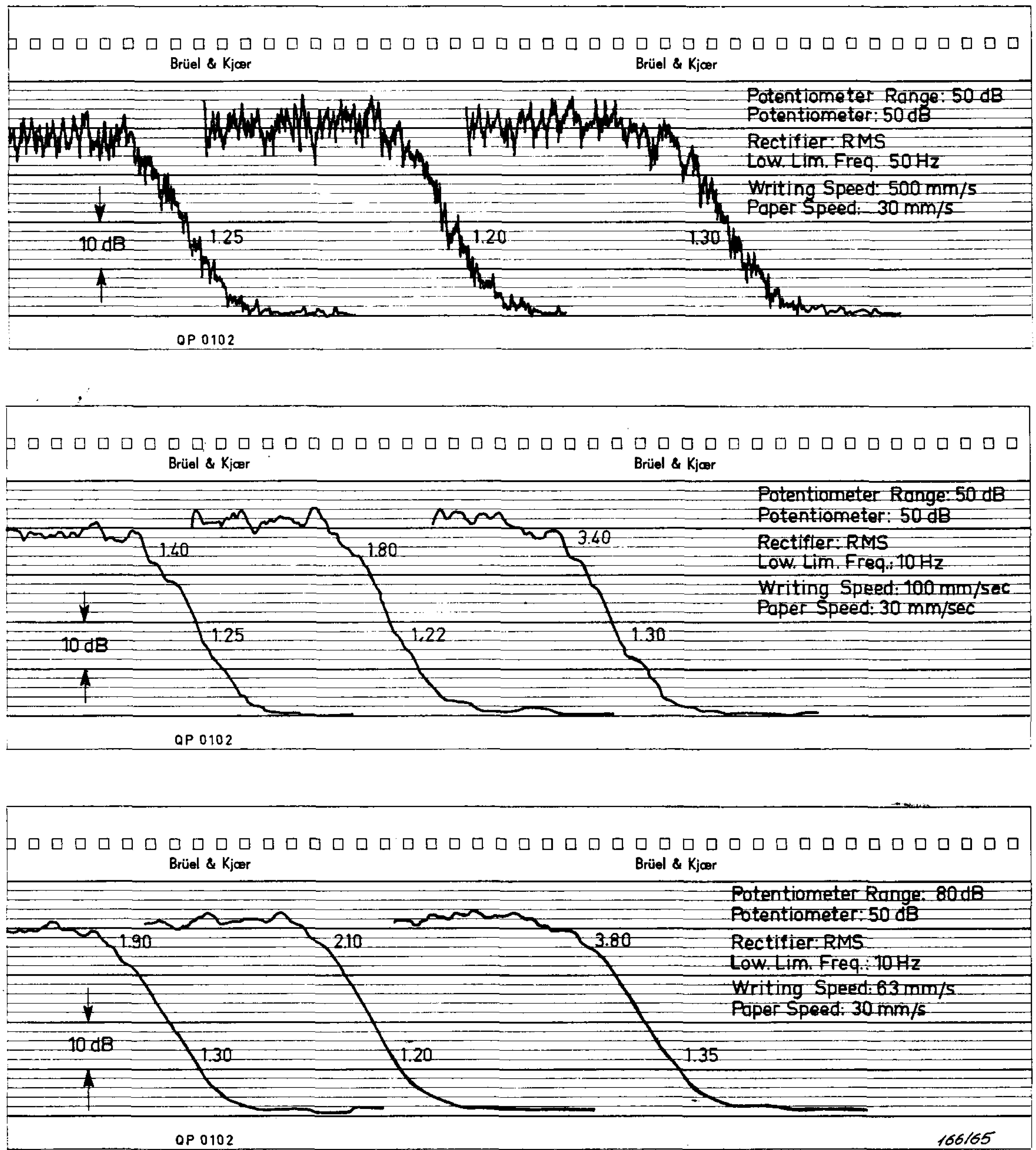


Fig. 11. Recording of the same decay processes as shown in Fig. 10 using different writing speeds on the level recorder.

out for band center frequencies of 500, 1000, and 2000 Hz and the results carefully analyzed as discussed below.

Fig. 10 shows three typical reverberation curves obtained for microphone position II according to the interrupted noise method. The noise bandwidth was 1/3 octave and the band center frequency was 1ooo Hz. To obtain a picture of the averaging effect that might be obtained simply by changing the writing speed in the level recorder the same decay processes are reproduced in Fig. 11 but with different writing speeds. The averaging effect is clearly noticed.

A further observation which can be made is that the slope of the middle position of the curve is very easy to determine and that the spread in values obtained for this slope is expected to be very small, which is also confirmed by looking at the table in Fig. 14.

On the other hand, the spread in value of the initial slope seems to be rather large, see also the table Fig. 14. By initial slope is here meant the slope of the uppermost 10 dB of the decay curve. As this part of the curve is supposed to be of great importance in judging the acoustical quality of rooms the spread is very undesirable.

Fig. 12 shows three curves obtained by means of the "integrated impulse method" under exactly the same conditions as those shown in Fig. 10 (pulse bandwidth 1/3 octave, center frequency 1ooo Hz, microphone position I).

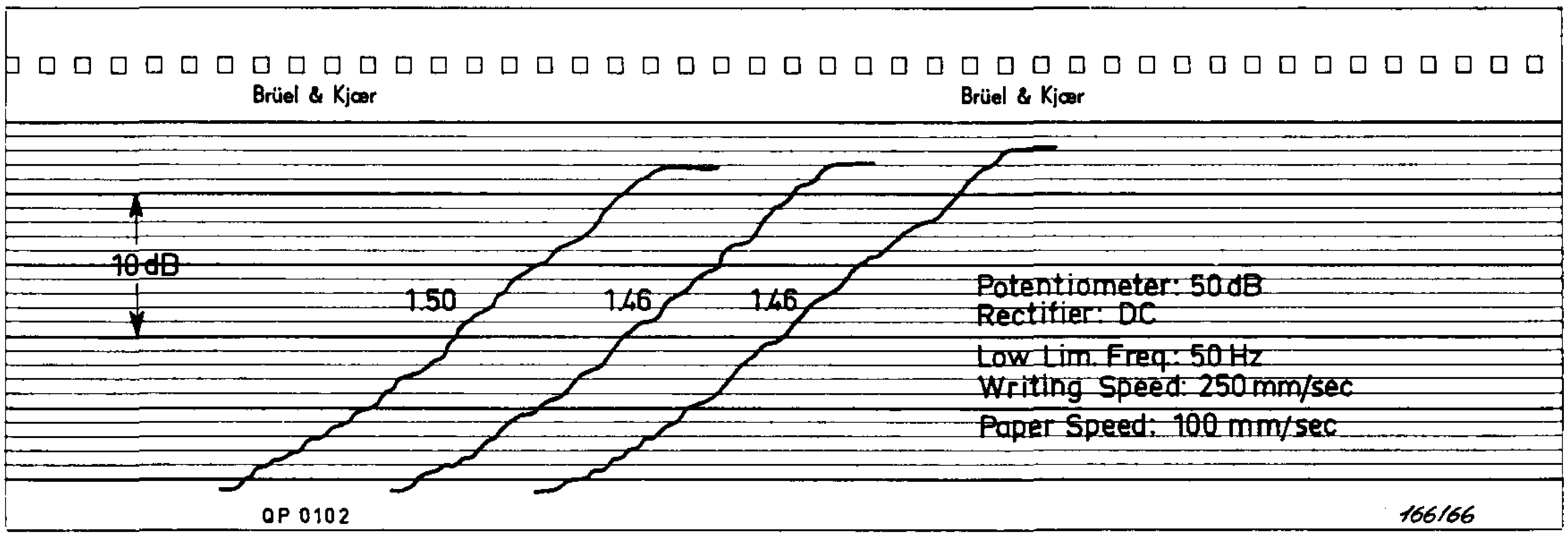


Fig. 12. Curves obtained by the "integrated impulse method" using "toneburst" excitation (1/3 octave 1ooo Hz filter excited by a 100 μsecond impulse).

It is clearly seen that the spread in initial slope is now reduced considerably (theoretically the spread should be zero!) While the curves recorded in Fig. 12 were obtained from measurements using 1/3 octave tone bursts to excite the room Fig. 13 shows a similar set of curves obtained by means of pistol shot excitation. From a look at the curves given in Figs. 12 and 13 the differences seem negligible which is further confirmed by the figures given in the table Fig. 14. This table gives a more comprehensive picture of the measurements using 1/3 octave bandwidths at 1ooo Hz and microphone position I.

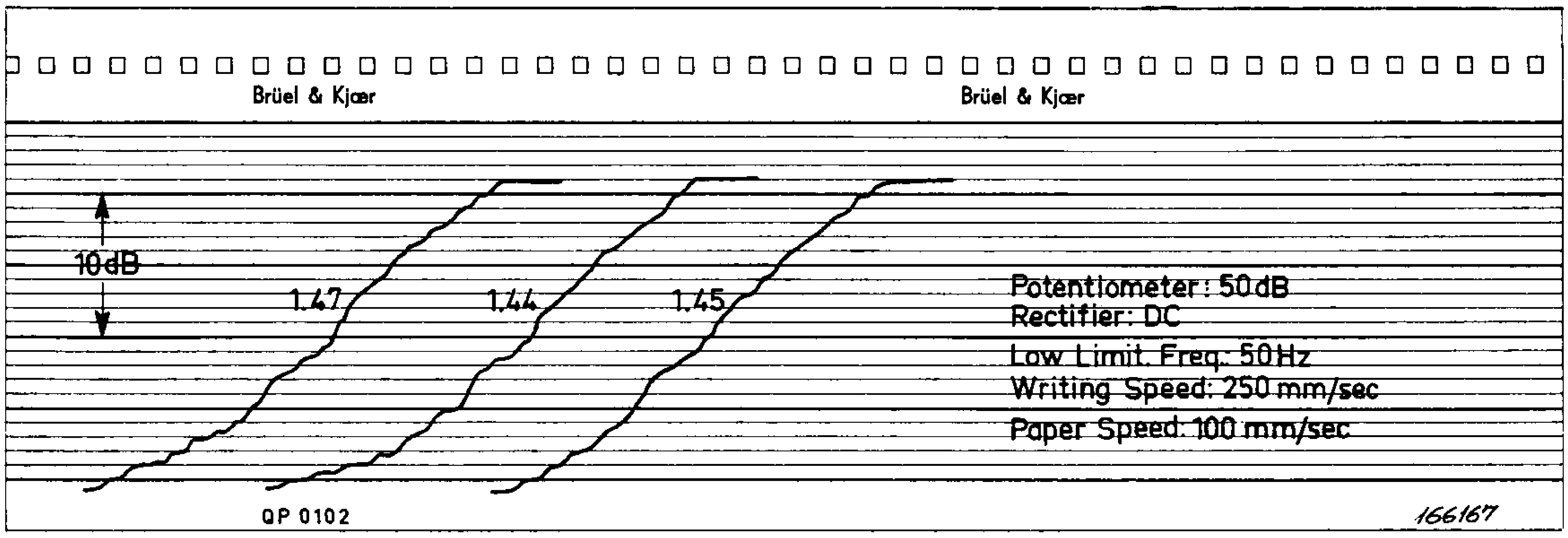


Fig. 13. "Integrated impulse" curves obtained with pistol shot excitation of the room.

1ooo Hz, 1/3 octave; Microphone position I

Sound Source	Reverberation time			
	0–10 dB		Full decay	
	T/sec	σ/sec	T/sec	T/sec
Interrupted noise	1.30	o.09	1.25	o.02
Pistol shot	1.45	o.02		
Filtered square-pulse	1.46	o.01		

Fig. 14. Table of the results of series of 10 measurements at microphone position I, 1/3 octave bandwidths, 1ooo Hz.

Still further data obtained both at microphone position I and II and both for 1/3 octave and 1/1 octave frequency bands are given in the tables Figs. 15, 16 and 17. These data show that for a determination of the slope of the middle portion of the reverberation curve no immediate advantage is gained by using the "integrated impulse method", as the spread in determined value is so small that it is simply due to the always existing evaluation error. However, in determining the initial slope of the reverberation curve the "integrated impulse method" seems to be far superior to the interrupted noise method. This is, of course, also to be expected as the initial decay conditions furnished by interrupted noise necessarily *must* vary from experiment to experiment while the impulse response for a certain source-microphone configuration is *one* single deterministic function.

Before finishing this discussion on the "integrated impulse method" the influence of background noise and signal to noise ratio upon the measured results should be briefly touched upon. Schroeder also mentions this and states that "if the decay curve is to be converted to a logarithmic amplitude scale, 500 Hz

Sound Source	Filter Octave	Micro-phone position	Reverberation time			
			0–10 dB		Full decay	
			T/sec	σ/sec	T/sec	T/sec
Interrupted noise	1/3	I	1.21	o.06	1.25	o.011
	1/1	I	1.42	o.10	1.32	o.009
	1/3	II	1.74	o.09	1.25	o.015
	1/1	II	1.28	o.02	1.31	o.007
Integrated impulse method Pistol shot	1/3	I	1.98	o.009		
	1/1	I	1.75	o.020		
	–	–	–	–		
	1/1	II	1.79	o.009		
Filtered square-pulse	1/3	I	1.50	o.032		
	1/1	I	1.56	o.017		
	1/3	II	1.68	o.038		
	1/1	II	1.54	o.014		

Fig. 15. Table of results obtained from measurements at 500 Hz.

1ooo Hz

Sound Source	Filter Octave	Micro-phone position	Reverberation time			
			0–10 dB		Full decay	
			T/sec	σ/sec	T/sec	T/sec
Interrupted noise	1/3	I	1.30	o.09	1.25	o.018
	1/1	I	1.43	o.06	1.32	o.012
	1/3	II	1.66	o.19	1.30	o.024
	1/1	II	1.26	o.06	1.31	o.008
Integrated impulse method Pistol shot	1/3	I	1.45	o.020		
	1/1	I	1.34	o.019		
	1/3	II	1.23	o.018		
	1/1	II	1.41	o.010		
Filtered square-pulse	1/3	I	1.46	o.012		
	1/1	I	1.47	o.027		
	1/3	II	1.47	o.017		
	1/1	II	1.40	o.004		

Fig. 16. Table of results obtained from measurements at 1ooo Hz.

2000 Hz

Sound Source	Filter Octave	Micro-phone position	Reverberation time			
			0–10 dB		Full decay	
			T/sec	σ/sec	T/sec	T/sec
Interrupted noise	1/3	I	1.03	0.08	1.13	0.012
	1/1	I	1.15	0.04	1.15	0.008
	1/3	II	0.96	0.05	1.15	0.017
	1/1	II	0.97	0.07	1.13	0.012
Integrated impulse method	1/3	I	1.32	0.011		
	1/1	I	1.31	0.016		
	1/3	II	1.31	0.022		
Pistol shot	–	–	–	–		
Filtered square-pulse	1/3	I	1.28	0.018		
	1/1	I	1.38	0.017		
	1/3	II	1.19	0.008		
	1/1	II	1.26	0.018		

Fig. 17. Table of results obtained from measurements at 2000 Hz.

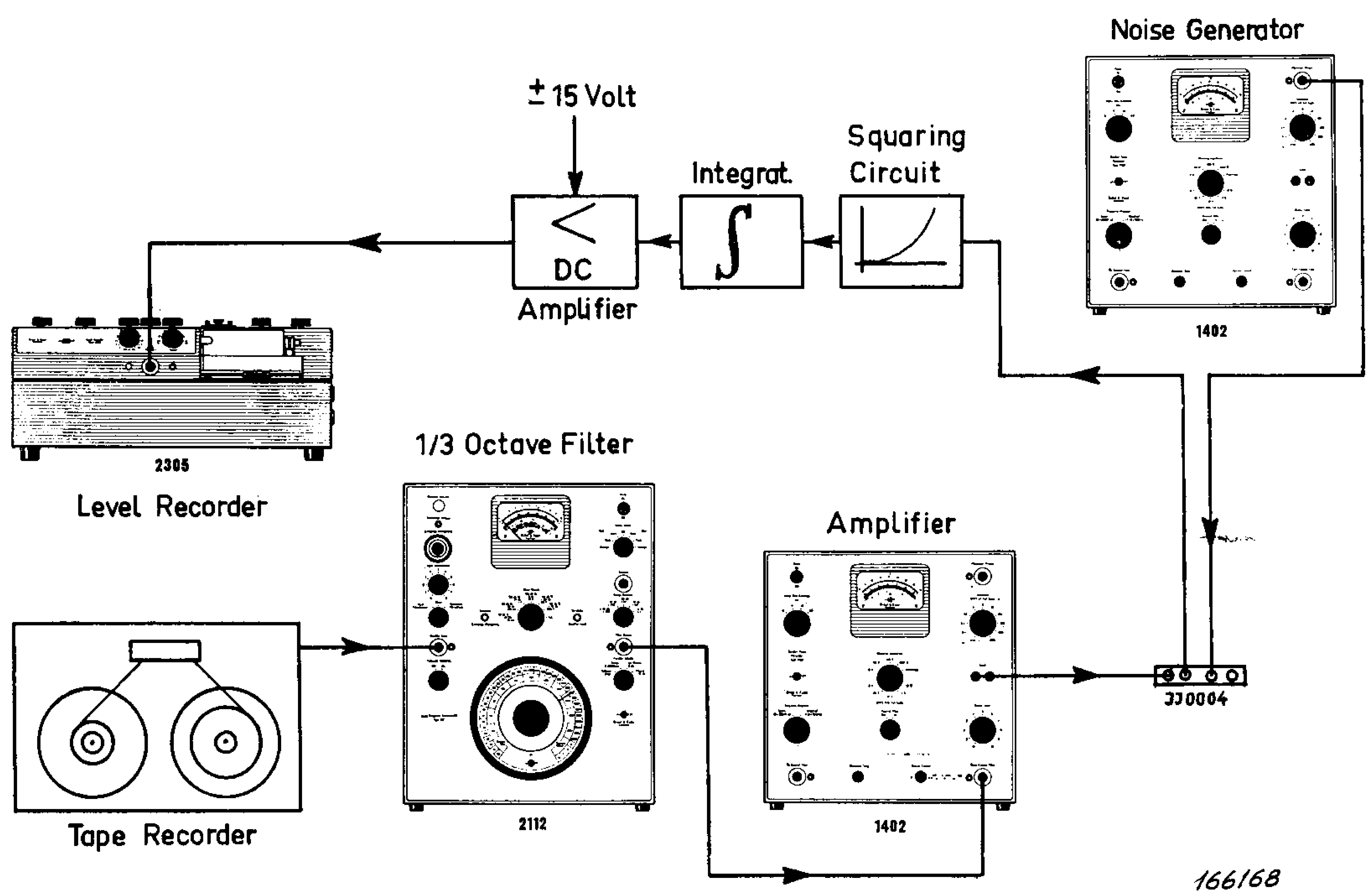


Fig. 18. Measuring arrangement used to estimate the influence of background noise.

care must be taken that the integration start after the reverberation signal exceeds the noise background". To show the effect of background noise some simple experiments have been carried out. They consisted in employing the "integrated impulse method" to measurements on a bell, and adding background noise artificially. The impulse response of the bell was recorded on magnetic tape.

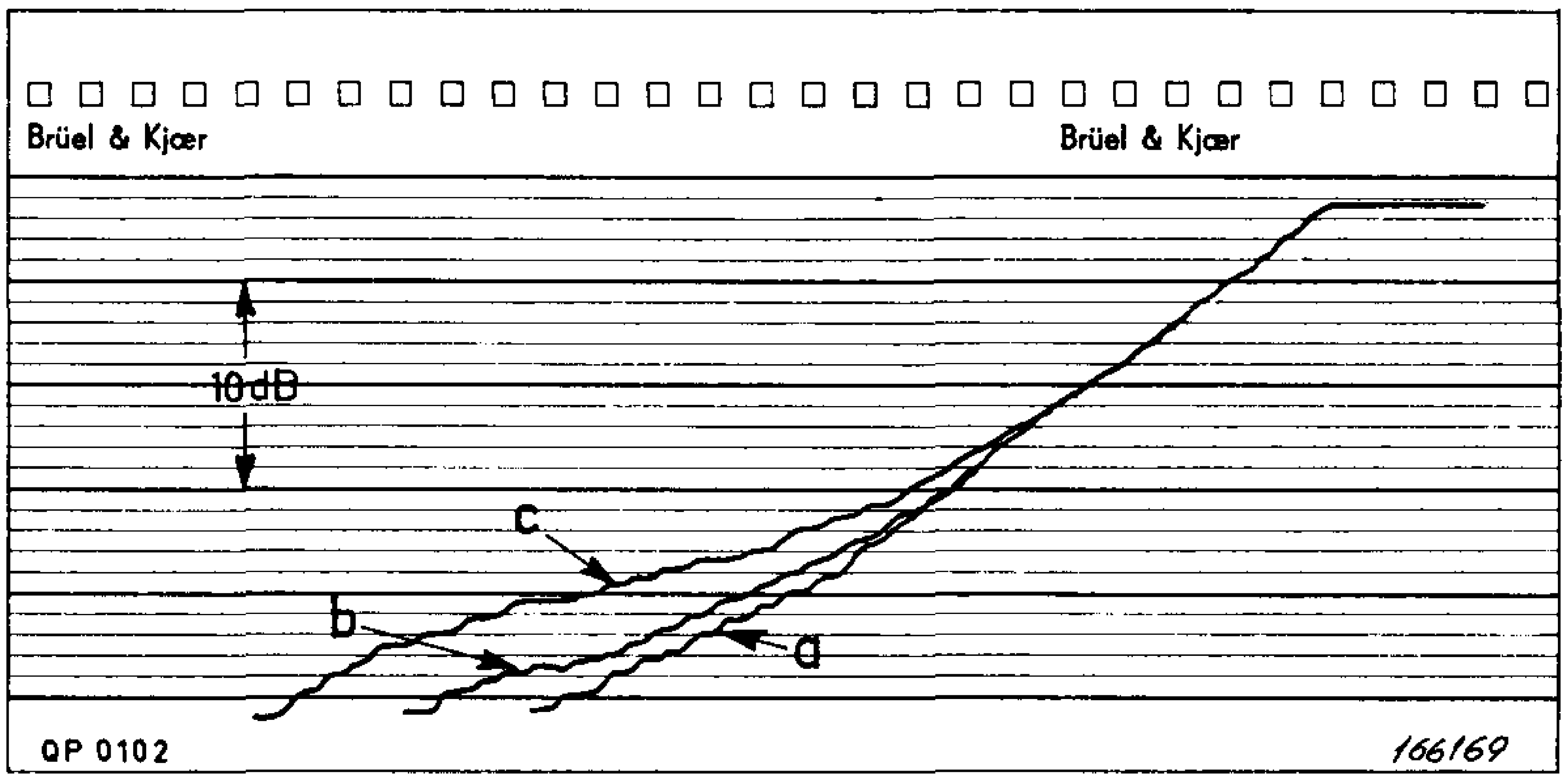


Fig. 19. Curves obtained with the arrangement shown in Fig. 18.

By playing the tape back and adding controlled random noise to the signal at the input of the squaring circuit, Fig. 18, various maximum signal-to-noise ratios could be produced. Fig. 19 shows the results of such measurements. The curve marked a) was recorded without the addition of "extra" noise, the maximum signal-to-noise ratio being approximately 35 dB. Curve b) was obtained by adding a certain amount of "extra" noise, thereby reducing the signal-to-noise ratio to about 26 dB. The integration was in this case also started a little earlier than in the case of curve a).

Finally curve c) shows a recording with a maximum signal-to-noise ratio of roughly 18 dB. The curves clearly indicate the influence of the background noise.

It should be noted, however, that in practical impulse response measurements it is only the noise which is present on the tape that actually disturbs the measurements as the internal noise produced by the measuring amplifiers can be balanced out in the DC amplifier prior to recording (see also Appendix B).

Acknowledgement.

The authors would like to express their sincere thanks to Dr. V. L. Jordan for his suggestion to use the decaying sound of a bell as test signal for the squaring and integrating circuits, as well as for valuable comments to the "historical" section of the manuscript.

Appendix A

Theory of the Integrated Impulse Method.

The basis of Schroeder's "Integrated Impulse Method" is, as stated in the text, that the ensemble average of the square of the reverberation noise decay in an enclosure equals the time integral of the enclosure squared impulse response. To arrive at this result Schroeder first considers a room excited by "stationary white noise" which is suddenly shut off. Except for the "whiteness" of the noise these are common experimental conditions for reverberation measurements. The implications of the "whiteness" will be discussed below. *If* the noise is stationary and white this can be matematically stated by the equation

$$< n(t_1) \times n(t_2) > = N \times \delta(t_2 - t_1) \tag{1}$$

where t_1 and t_2 are two arbitrarily chosen instants of time and $\delta(t_2 - t_1)$ is the Dirac δ-function. $< n(t_1) \times n(t_2) >$ is the autocovariance function of the noise In words equation (1) states that the noise is *uncorrelated* (which is the same as "white") and that the average $< n(t_1) \times n(t_2) >$ only has a value different from zero when $t_2 = t_1$.

Schroeder next considers the response of any linear network to an arbitrary time function, $n(\tau)$:

$$s(t) = \int_{-\infty}^{t} n(\tau) \times r(t-\tau)\, d\tau \tag{2}$$

Here $r(t-\tau)$ is the impulse response of the network at the time t to a *unit impulse* occurring at time τ.

By considering the room a linear acoustic network and squaring equation (2) one obtains the following double integral:

$$s^2(t) = \int_{-\infty}^{\tau=t} d\tau \int_{-\infty}^{\Theta=t} d\Theta \times n(\tau) \times n(\Theta) \times r(t-\tau) \times r(t-\Theta) \tag{3}$$

The upper limits of integration should, furthermore, be taken to be 0, if this is chosen as the instant of time when the noise is shut off (no "forcing" input to the system occurs after the source is shut off!).

Averaging the above expression over the ensemble of noise signals (averaging over "infinitely" many reverberation measurements) and utilizing equation (1):

$$< n(\tau) \times n(\Theta) > = N \times \delta(\Theta - \tau)$$

one obtains:

$$< s^2(t) > = \iint_{-\infty}^{0} N \times \delta(\Theta - \tau) \times r(t-\tau) \times r(t-\Theta)\, d\Theta d\tau \tag{4}$$

As $\delta(\Theta - \tau)$ is zero except when $\Theta = \tau$ and as the integral over the delta function equals unity equation (4) finally becomes:

$$< s^2(t) > = N \times \int_{-\infty}^{0} r^2(t-\tau)\, d\tau \text{ *)} \tag{5}$$

It may be worth while to discuss the expression (5) a little further:
From $\tau = 0$ the function $< s^2(t) >$ represents the ensemble average of the *squared* reverberation process. To obtain this function an "infinite" number of measurements would be necessary and the reverberation time determined according to normal procedures would be *half the actual reverberation time* due to the squaring.

On the other hand, the time integral $\int_{-\infty}^{0} r^2(t-\tau)\, d\tau$ represents, basically, a single measurement of the squared impulse response of the linear network (room) integrated over an infinite time.

The general practice in reverberation measurements has previously been to determine $< s(t) >$ (not $< s^2(t) >$!) approximately from a certain number (not "infinitely" many) measurements.

It seems therefore a very fascinating idea to be able to obtain an "infinite" ensemble average from a single measurement, quite apart from the fact that the main (average) trends in the process would then show up very clearly!

Because the physical interpretation of the integral $\int_{-\infty}^{0} r^2(t-\tau)\, d\tau$ might not be immediately obvious the following discussion might be helpful:
By definition $r(t-\tau)$ is the unit impulse response of the system under consideration. Now, if such an impulse occurred at $\tau = -\infty$ then the above integration merely states that the response (or rather squared response) has to be theoretically considered and integrated over an infinite period of time. In practice, of course, the response to a unit impulse is only measurable over a certain, very finite, period of time. The meaning of the integral is thus to consider the integration as long as the response of the system to a unit impulse can be determined in practice, and the limits of integration are chosen accordingly.
The "unit impulse" is in practice often obtained by means of a pistolshot, a tone burst (noise burst) or other short lasting sound phenomena. At this point, however, the previously mentioned implications of the "whiteness" of the noise should be considered.
In Schroeder's derivation, as outlined above, "white" (uncorrelated) noise was assumed for excitation of the room. Normally band-limited noise is used, to be able to determine the reverberation as a function of frequency. As soon as the noise is band-limited equation (1) does not hold in a strict mathematical

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*) This theoretical result has also been given by J. S. Bendat in 1958 in a slightly different form in his book "Principles and Applications of Random Noise Theory", Example 2—11, p. 76.

sense, because a certain time correlation is imposed upon the noise. If, however, the *effective correlation interval is small compared with any part of interest in the reverberation decay process* equation (1) is still valid in a practical sense. (The effective correlation interval of band-limited noise is of the order of the reciprocal of the bandwidth).
Schroeder "avoided" this difficulty by including the filter response in the room response, stating that $r(t-\tau)$ should be taken as the combined response of the filter and the room. The two responses then have to be separated at a later stage in the analysis, and the implications stated above are thus actually only postponed until this stage.
Regarding the length of the impulse used to determine the impulse response of the filter + room this should, to obtain the *true* impulse response, be short compared to the period of the filter center frequency.

Appendix B

Description of the Measuring Arrangement.

A block diagram of the set-up is shown in Fig. 8 of the text and in the following a description of the "blocks" which were specially designed for the purpose is given.

Squaring Circuit.

This circuit, which is shown in Fig. B.1, was designed for a maximum input voltage of 10 volts RMS.
The input signal s(t) is rectified and the squaring parabola, $Y = s^2(t)$, is approximated by means of three straight line segments as shown in Fig. B.2.

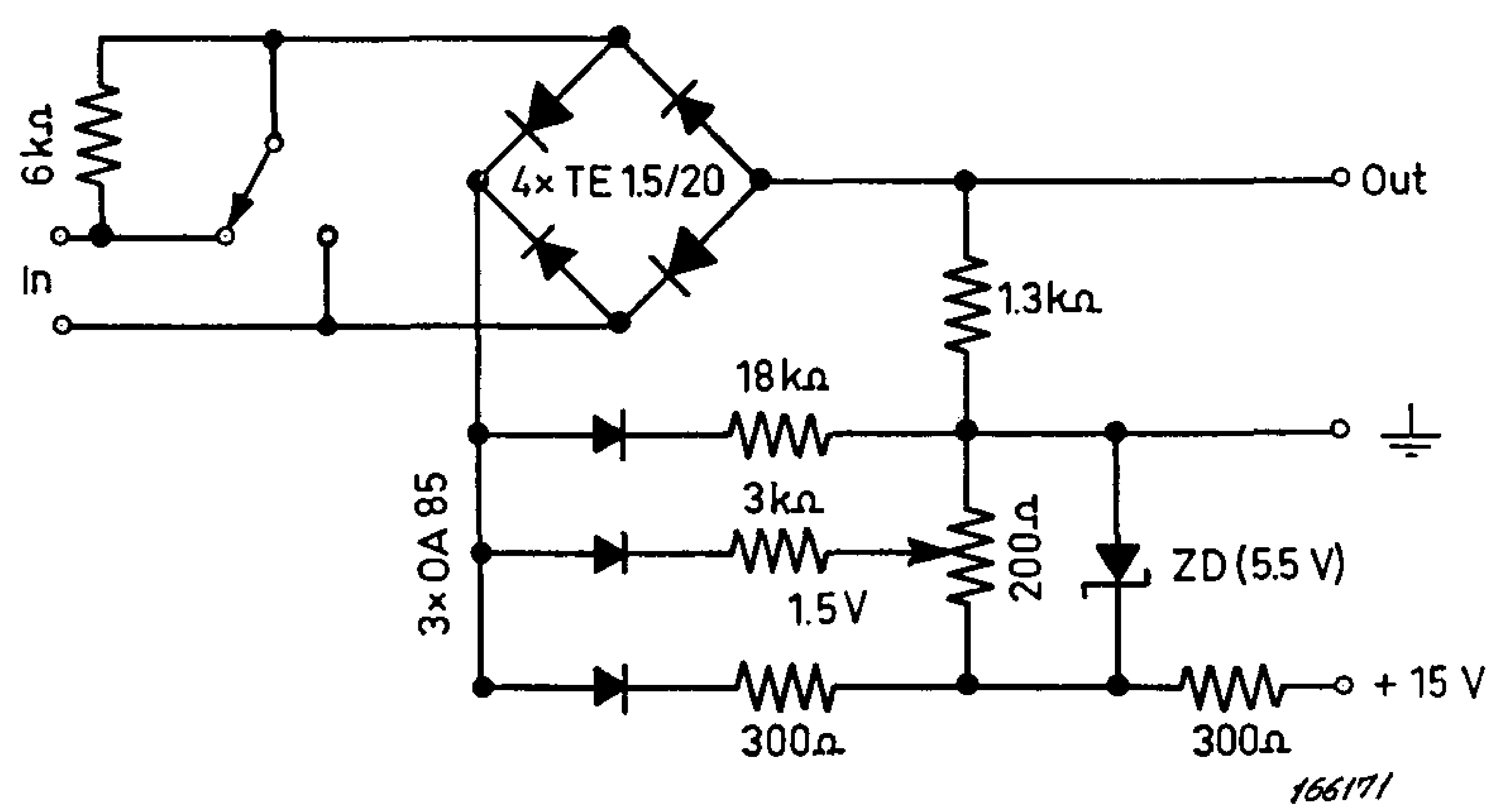

Fig. B.1. Schematic diagram of the experimental squaring circuit.

This is obtained by applying three different bias voltages to the diodes (OA 85), so that they will start conducting at different input levels.
The output from the circuit as a function of input is shown in Fig. B.3 to a log-log scale. It can be seen from the curve that a reasonably good accuracy is obtained, except at very low signal levels, if the input voltage is adjusted to a maximum level of 10 volts RMS in each experiment.

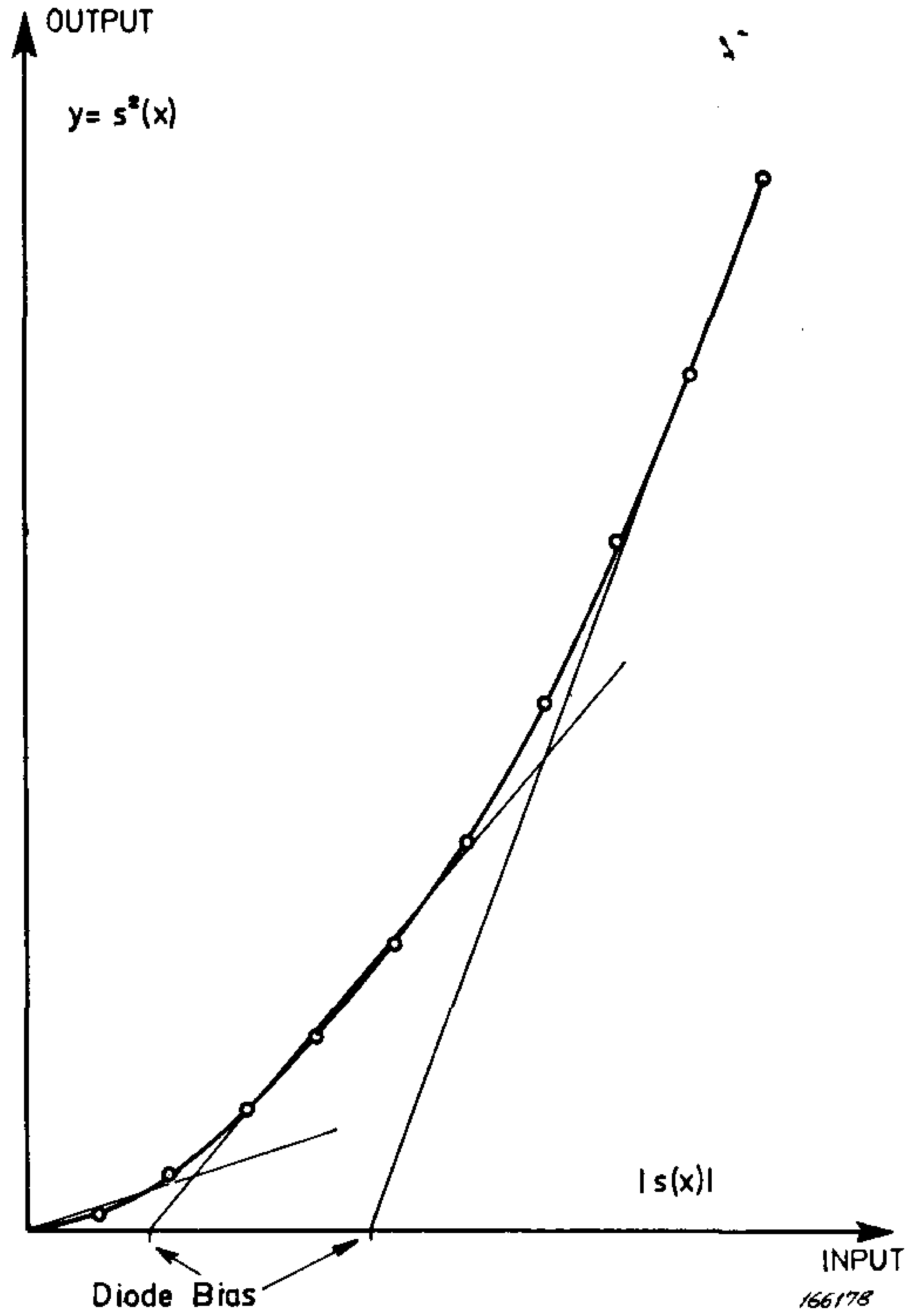

Fig. B.2. Sketch illustrating the functioning of the squaring circuit.

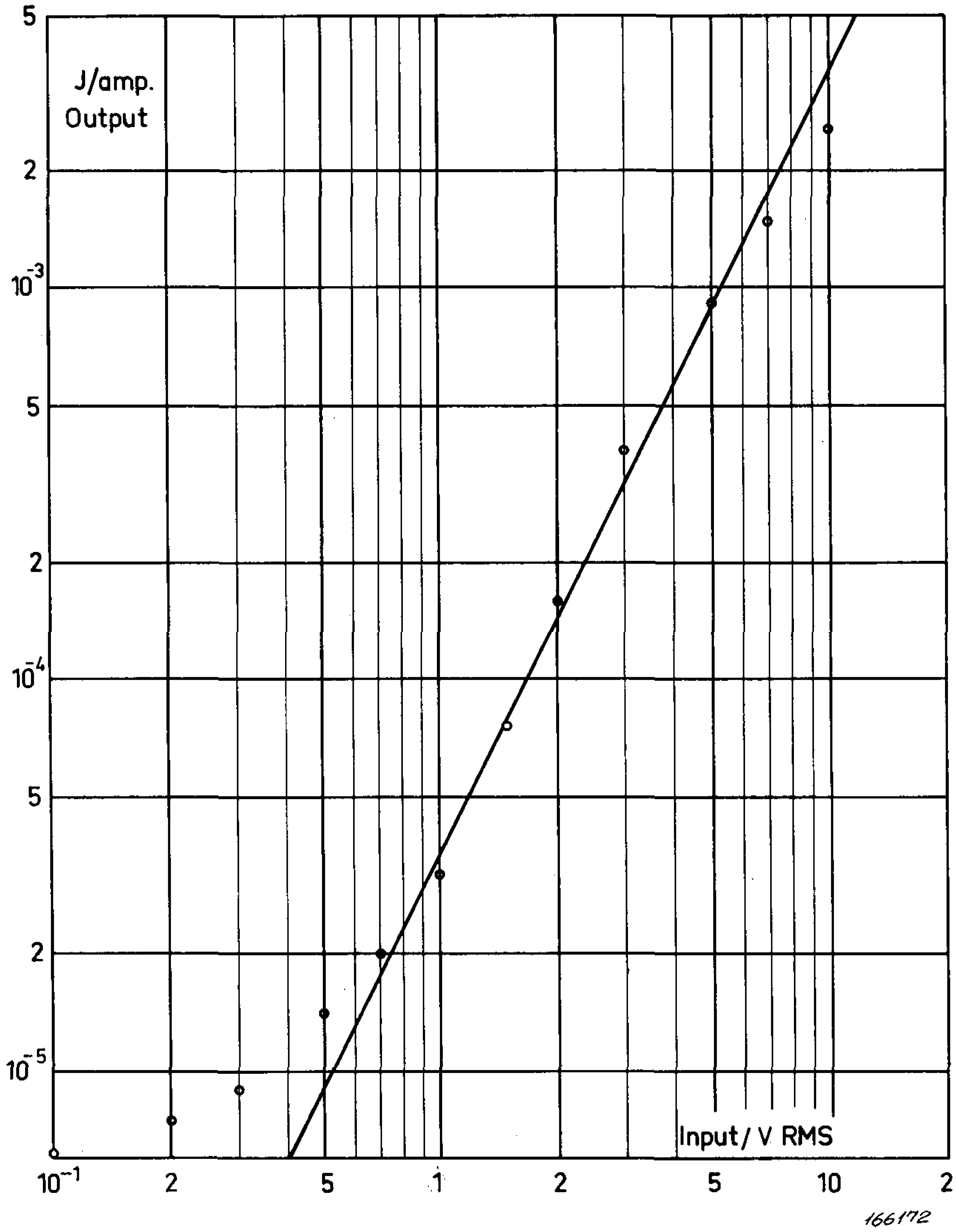

Fig. B.3. Input–Output relationship measured on the circuit, Fig. B.1.

Integrating Network.
The integrating network is a simple RC circuitry with R = 5.5 MΩ and C = 4 μF giving a theoretical time constant of T = RC = 22 seconds Fig. B.4. A check on the practical circuit revealed that T was not 22 sec. but 16.7 sec., see Fig. B.5, due to the loading effect of the input impedance of the DC-amplifier. From the measurements this input impedance can be calculated to be approximately 17 MΩ.

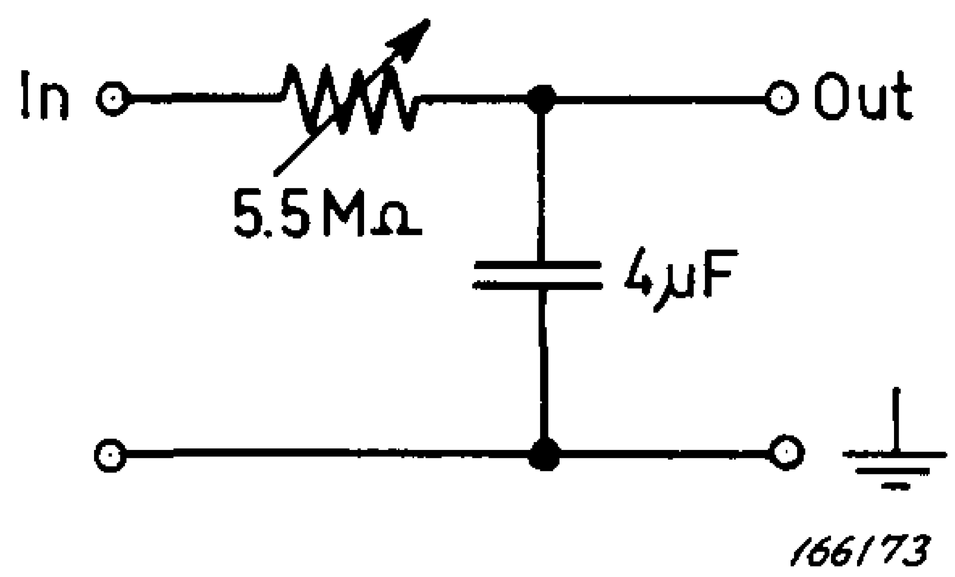

Fig. B.4. The integrating circuit.

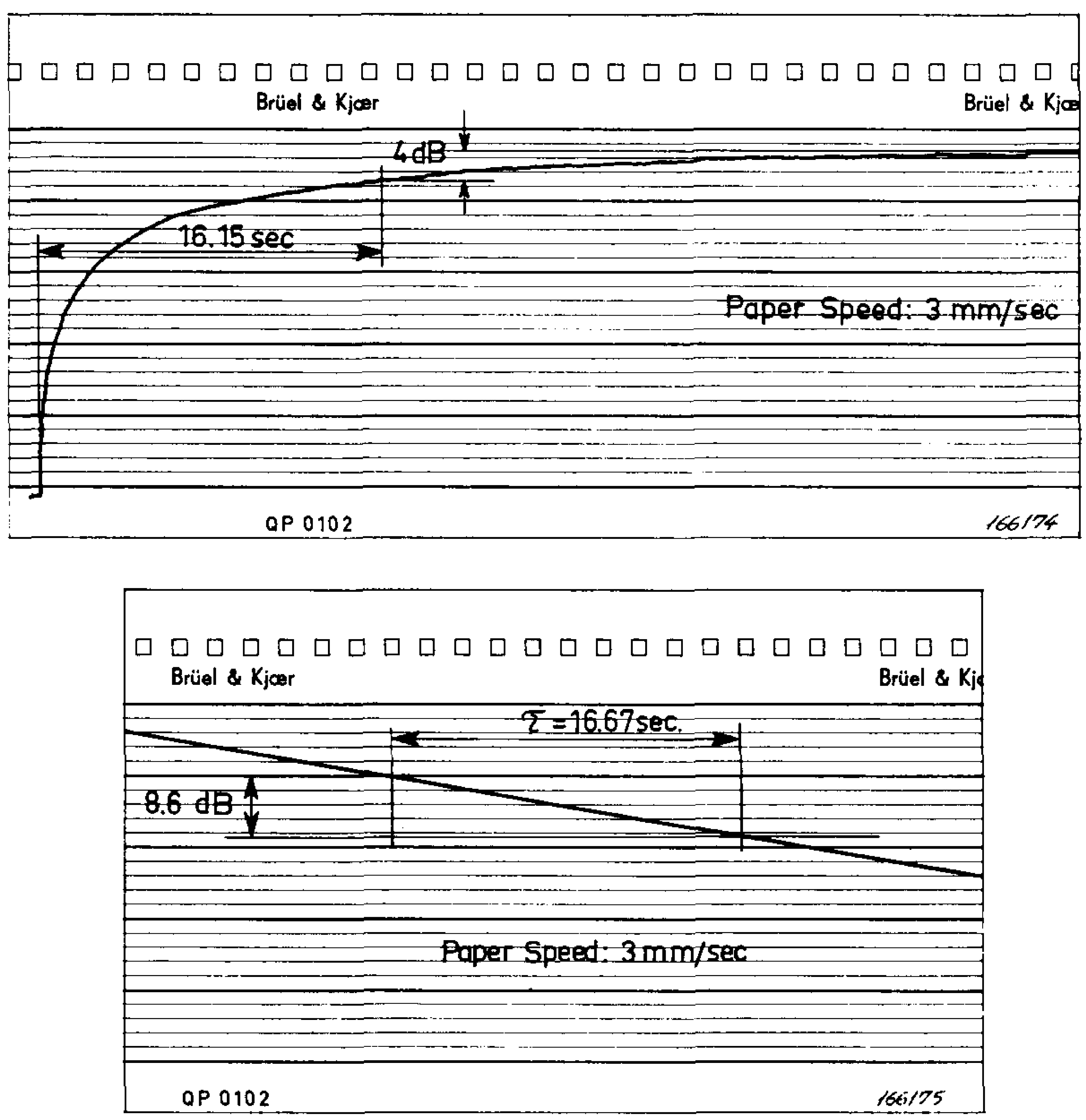

Fig. B.5. Typical charging and discharging curves for the integrating capacitor. a) Charging. b) Discharging.

DC-Amplifier.
In order to obtain a high input impedance and low temperature drift a balanced input stage with field effect transistors was used. The circuit is shown in Fig. B.6 and the specification were:

Gain:	35 dB
Frequency Range:	DC – 10 kHz (–1 dB)
Dynamic Range:	35 dB
Input Impedance:	Approximately 17 MΩ

Adjustment of the DC-amplifier was made by careful balancing of the input stage (with the squaring and integrating networks connected in circuit and zero input signal applied to the squaring device). Due to the biasing of the diodes in the squaring circuit, however, a certain residual voltage will exist

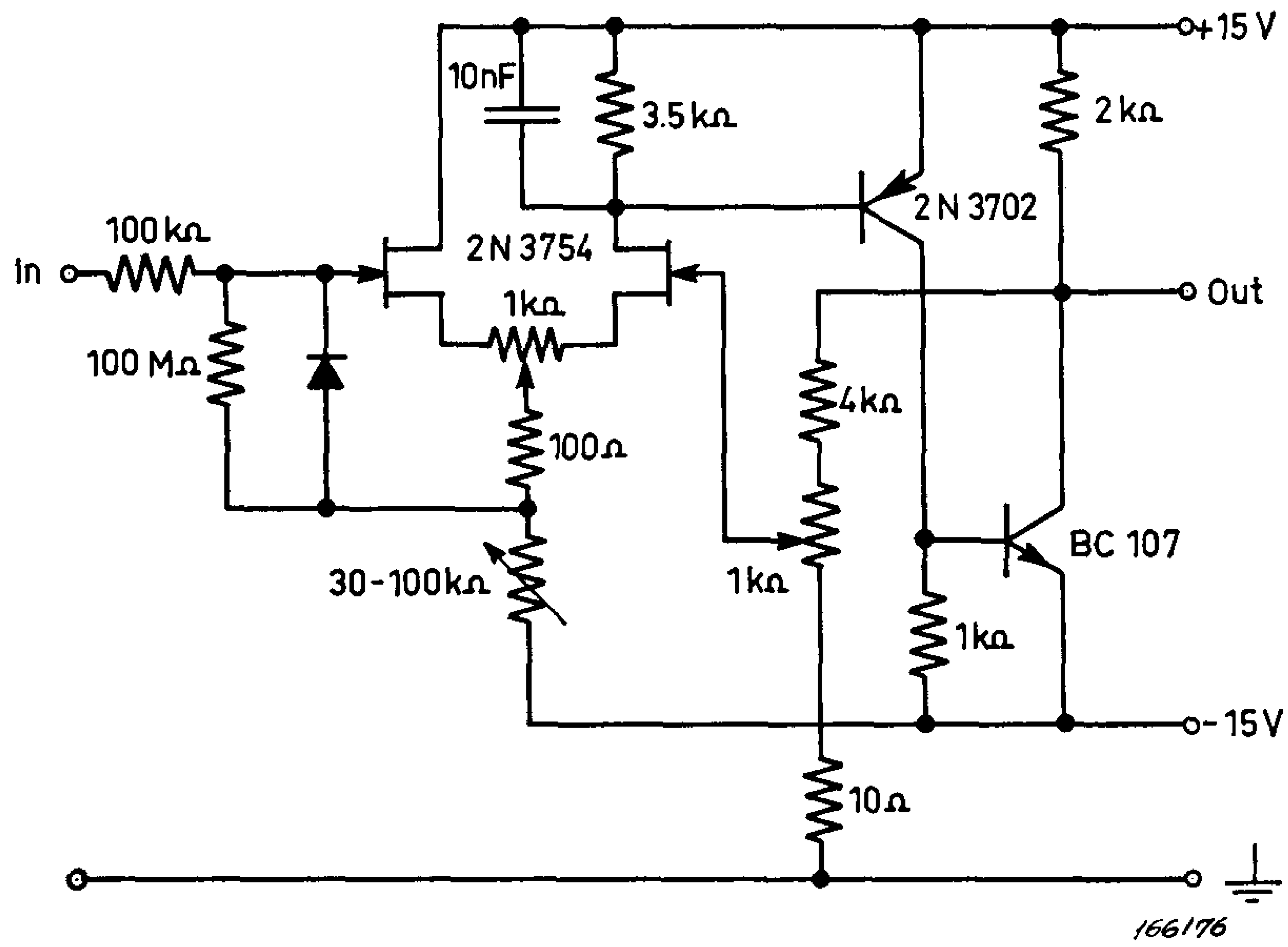

Fig. B.6. Circuit diagram of the DC-amplifier.

on the integrating capacitor and make a rapid resetting of the initial conditions after measurement impossible. Between each measurement it is therefore necessary to wait until the correct initial charge is re-established on the capacitor.

To avoid external (noise) signals producing an "extra" charge on the capacitor during this waiting time the input to the squaring circuit is short circuited by means of a switch, see Fig. B.1. At the same time a load resistor of 6 kΩ is connected across the output of the AC-amplifier (Type 1402) preceeding the squaring circuit.

Check on Correct Operation of the Arrangement.

To allow for a check on the complete squaring and integrating arrangement measurements were made on the reverberation of a bell. As the decay here takes place strictly exponentially, and the integral of the square of an exponential function is also an exponential function (with twice the value of the exponent) this signal should provide an excellent check on the equipment over the complete dynamic range. Furthermore, exponential decays result in straight lines when recorded on a logarithmic level recorder, thus making the

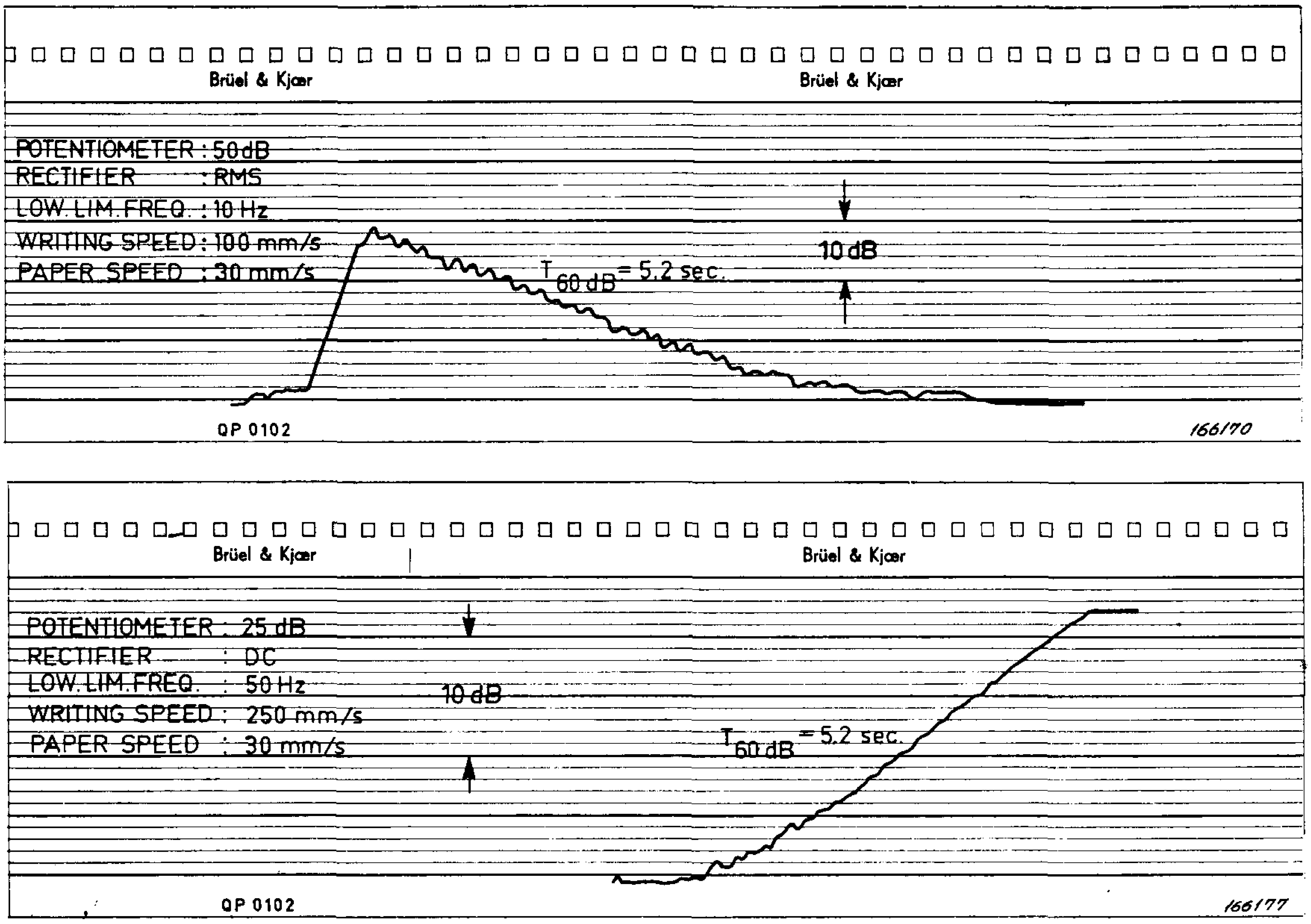

Fig. B.7. The sound decay of a bell recorded to a logarithmic scale.
a) Direct recording of the sound pressure level.
b) Recording obtained after squaring and integration of the sound pressure level (and with reversed time direction as used in the "integrated impulse method").

check straightforward. Fig. B.7 shows the result of the measurements. In Fig. B.7 a) the decay of the bell was recorded directly, while in Fig. B.7 b) the recording was made after the squaring and integration had taken place. It can be seen that the accuracy of the circuits is actually quite good.

Note: Due to the squaring the full dynamic range in Fig. B.7 b) corresponds to a dynamic range of the *input signal to the squaring circuit* of only 25 dB, even though a 50 dB range potentiometer was used on the Level Recorder.

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