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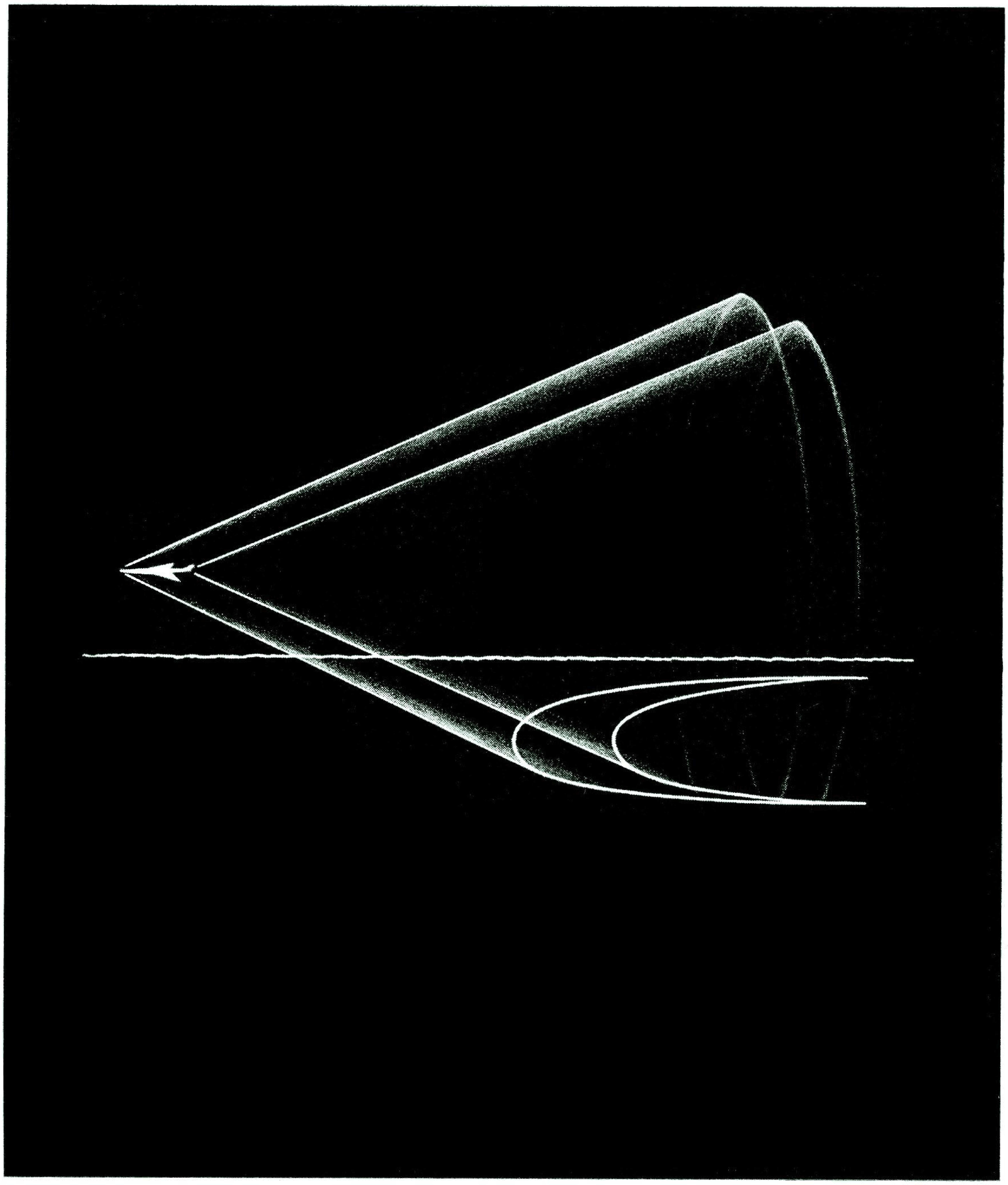

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TECHNICAL REVIEW

No. 1 — 1969

Contents

The Use of Digital Systems in Acoustical Measurements*)

by

Jan Søeberg, M.Sc.E.E.

ABSTRACT

The paper discusses some of the properties and problems involved in the use of digital systems for reception and evaluation of acoustical or vibrational measurements. Emphasis is laid on the fact that some of the acoustical equipment operating in the analog mode, will require an analog digital conversion.

SOMMAIRE

L'article discute quelques propriétés et problèmes liés à l'emploi de systèmes digitaux pour la réception et l'évaluation de mesures acoustiques ou de vibrations. L'accent est mis sur le fait que certains équipements acoustiques fonctionnant sur le mode analogique vont demander une conversion analogique-digitale.

ZUSAMMENFASSUNG

Einige Besonderheiten und Probleme werden erläutert, die bei der Anwendung von digitalen Systemen für die Erfassung und Auswertung akustischer oder schwingungstechnischer Meßdaten zu berücksichtigen sind. Breiten Raum nimmt die Tatsache ein, daß ein Teil der Meßgeräte diese Meßdaten in analoger Form liefert und somit für ihre digitale Weiterverarbeitung eine analog-digitale Umsetzung erforderlich wird.

Introduction

The digital technology and system knowledge have in the past decade expanded to a level far beyond any estimate made just a few years ago. Still new areas are being taken up to evaluation for possible use of digital systems. One of the reasons for this is the hardware technology, that is the development within the electronic components with integrated circuit blocks, and another reason is the development of the software part, that is the computer programming part which controls the process and organizes the system. Sometimes it can even be hard to find a sharp limit between the hardware and software parts. They are both important to the complete system. If now this technology has to be implemented on the science of acoustics, it will be worth mentioning some of the reasons which could make today's acoustical scientists look for digital systems. The use of acoustical analysis in respect to maintainability analysis of a given type of equipment will need automatic data logging and evaluation. The higher demand for accuracy during noise analysis (sonic booms) has demanded higher rate of speed, and the wish to minimize the influence of human errors due to the higher speeds will require automatic logging.

The major point which seems to ask for digitalization is therefore speed. Slower requirements will still be able to rely on paper recordings and human

*) Paper presented at the 6th International Congress on Acoustics, Tokyo, Japan 21 – 28 August 1968.

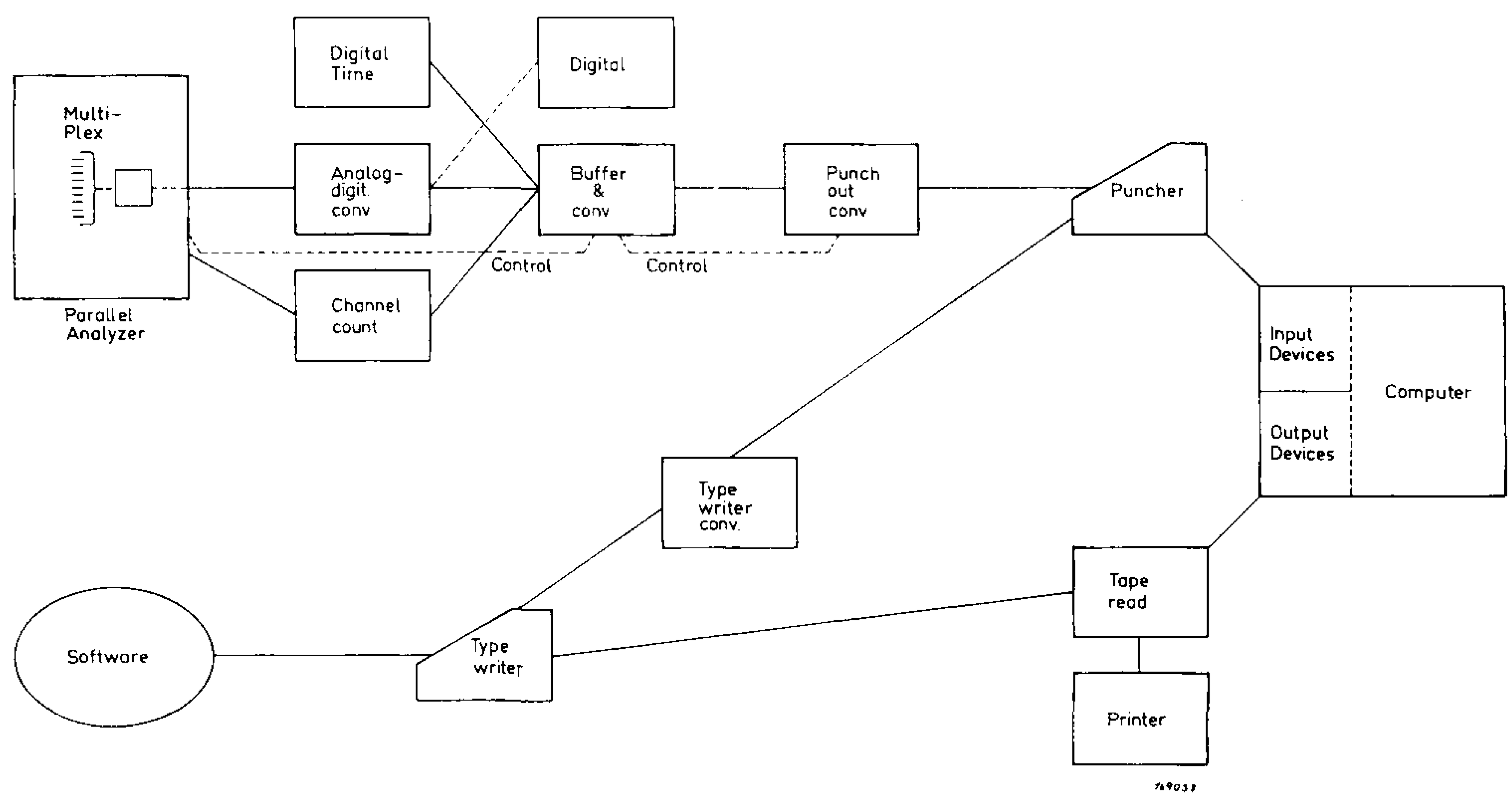

Fig. 1. Digital acoustical Analyzer System.

evaluation. Higher speeds will bring higher system costs, so a compromise between these two will normally be the result.

System Approach

The different requirements and the compromise between cost and speed therefore has demanded a system approach where flexibility is of major importance. The degree of speed will par example define the tape medium (paper or magnetic) or direct on-line connection to the computer. In B & K we have found that the system shown in Fig. 1 meets the above mentioned requirement of speed and flexibility.

The Parallelanalyzer is of the analog frequency analyzer type, with 50 band-pass filters, which can be scanned successively by a digital multiplexer circuit up to 40–50 complete cycles per second.

The multiplexer output is the analog representation of the filter now being scanned. Therefore a counter is needed to give the channel number information to the computer. For some use also a time reference may be needed. The analog signal comming out of the multiplexer, is fed to the A/D-converter, which converts the signal to max. five decimal digits, presented in the BCD- or the 8421-code. Codechange is easily obtained by changing a matrix card.

On the input side of the Buffer and codeconverter is now 9 digits in parallel ready to be transmitted to the tape medium. The paper puncher and paper tape will in this example represent the tape medium and no big change will be necessary in changing to magnetic tape or on-line type of operation. The buffer will be necessary to hold the 9 digit information during the read-out sequence, which is sequential digit by digit. No change in the 9 digits is allowed during this period. The codeconverter will change the code to the proper code for the punch-out converter, which in turn will control the mechanical punch-out arms in the puncher.

The flexibility is obtained when one tries to consider, that some application may want to get the digital display only. The display unit can be connected right after the A/D-converter. The time-unit can be interchangeable with other units (range no. information etc.).

Tape Organization

Fig. 2 shows as an example how the block-format on the tape may be organized. One block represents max. 9 digits, which might be defined as:

1. Range No. Shows the range of input selector.
2. 1' Digit of Channel no.
3. 2' Digit of Channel no. etc. as shown in Fig. 2.

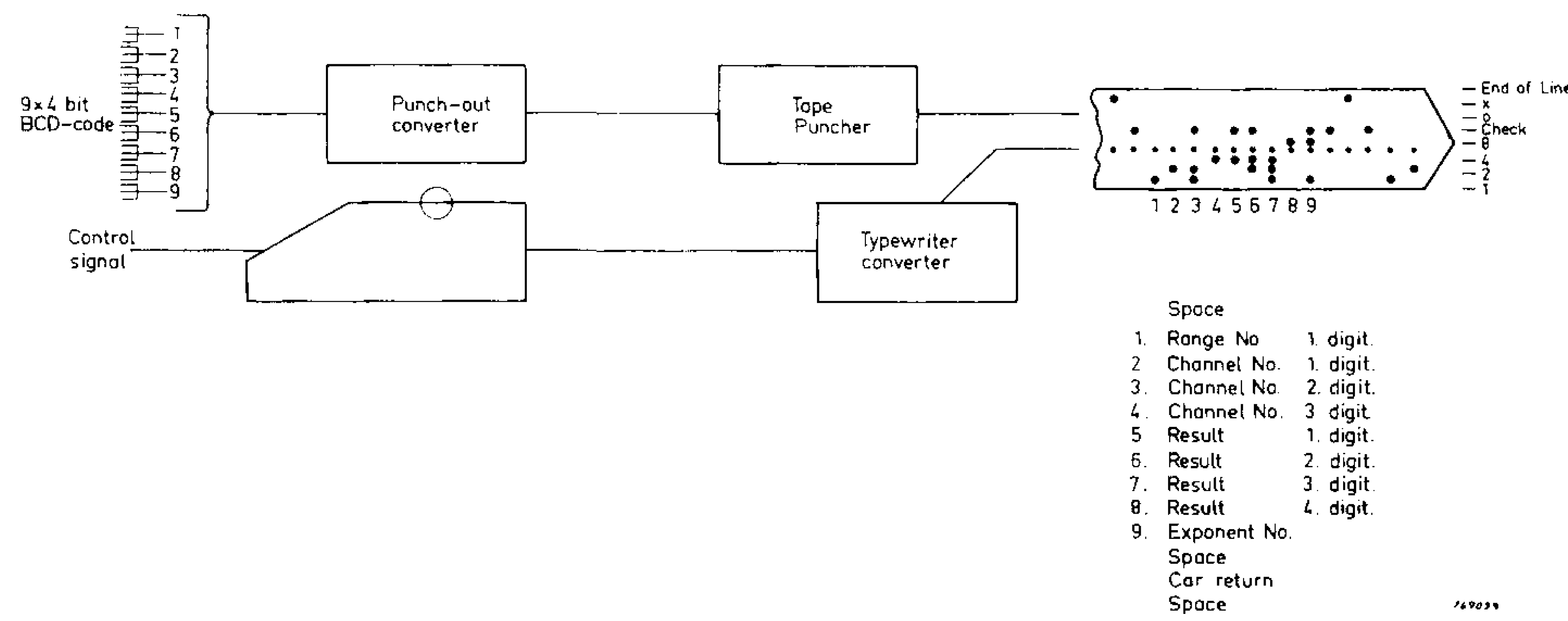

Fig. 2. Tape-System.

Using this kind of block definition, where every place in the block has the same meaning every time, will easily make the tape useful for the computer. The code shown on Fig. 2 is the standardized Flexiwritercode which is useful on most computer read equipment. Space and carreturn signals are necessary only if the tape is to be read directly to printer or typewriter to make a readable tabulation on the paper.

Software

Until now I have only talked about the hardware part of the system. But to use the computer to analyse, or to evaluate, the results, it is also necessary to work on the software side of the problem, that means to program the computer so it "knows" the meaning of the digits in the block format. Therefore Fig. 1 also shows the connection of the typewriter and typewriterconverter to the tape puncher. In this way the program and the data tape can be fed to the computer using part of the same equipment. Also comments in plain language can be printed on the tape. The computer program will later select between data and plain language, and only use the data.

Synchronization

Because speed of data handling was one of the prime reasons which resulted in the Fig. 1 system, a few comments on this aspect is worth mentioning. The

A/D converter and the paper puncher is normally working on different speed ranges. Therefore some kind of synchronization between the units is needed. The A/D-converter cannot accept any change in its input before the digitizing procedure is finished, and the puncher cannot accept any new 9 digit information from the buffer before the mechanical puncher arms have finished their punching. Every unit will give a ready signal when a process is ended. If now these signals are used as control signals this will make the system follow the slowest unit. In this case the slowest unit is the paper puncher, (max. 100 punched characters per second). If this speed is to be exceeded the flexibility of the system allows us to change the paper tape punch unit with the much faster magnetic tape units whereby the speed can be accelerated to more than 100,000 characters per second. And then if this speed still is not enough, on-line arrangements can be made directly to the computer.

Conclusion

We have seen the possibility to build up a digital system, which can be used to log and evaluate acoustical data in an automatic mode. The computer may be connected directly or may be fed with the tape at a later time. The future will probably show how special computers take over the entire system using Fast Fourier Transform arithmetics on samples taken directly at the acoustical waveform.

Impulse Noise Measurements*)

by

C. G. Wahrmann, M.Sc.

ABSTRACT

All averaging AC instruments have a problem as to the correct evaluation of short single impulses or repetitive pulses of low repetition rate. The behaviour of different meter circuits to simple impulse signals is compared. Finally some impulse sound level measurements of practical impulses noises are shown.

SOMMAIRE

Tous les appareils de mesure AC fournissant une indication à partir de la moyenne de valeurs successives présentent un problème en ce qui concerne l'évaluation correcte d'impulsions solitaires de courte durée ou d'impulsions à faible fréquence de répétition. On compare le comportement de différents circuits de mesure à des signaux impulsionnels simples. Finalement sont montrées quelques mesures de niveau de bruit à impulsion pour des bruits impulsionnels tels que l'on en rencontre en pratique.

ZUSAMMENFASSUNG

Die im Gleichrichterteil der Anzeigegeräte enthaltenen Integrationsglieder müssen so bemessen sein, daß kurze Einzelimpulse und Impulsfolgen mit langsamer Pulsfrequenz genau ausgewertet werden. Das Verhalten verschiedener Anzeigeschaltungen gegenüber impulshaltigen Signalen wird verglichen. Schließlich werden einige Impulsschallpegelmessungen praktischer, impulshaltiger Geräusche kommentiert.

It is the intention of this paper to show the importance of having high crest factor capability in instruments for impulse noise measurements. For measurements of alternating variables such as an acoustical signal the most useful quantity as a measure of the signal strength is the root mean square value. In some earlier articles (1) the squaring properties of such RMS instruments have been considered, but this paper will look more into the averaging process.

By measurements of stationary random or nonrandom signals, it is normally assumed that the averaging time is long compared to the longest period in the envelope of the signal. This assumption could be valid for repetitive pulses with reasonable repetition rate, but how should single impulses, or pulses with very low repetition rate be evaluated? Several authors (2) have worked on this problem and although there has been some discrepancies these works are now resulting in an IEC proposal for an Impulse Sound Level Meter (German standards for such an instrument already exist).

These standards prescribe that the mean square of the signal should be produced with an RC averaging time constant of 35 ms. The square root of the peak voltage on the averaging condenser should then be used as a measure of the impulse sound pressure in that the instrument in other respect should

*) Paper presented at the 6th International Congress on Acoustics, Tokyo, Japan 21 – 28 August 1968.

fulfil the Precision Sound Level Meter requirements.

These standards also prescribe some test signals of single tone bursts and repetitive tone bursts and the corresponding indications with tolerances whereby the averaging process can be checked.

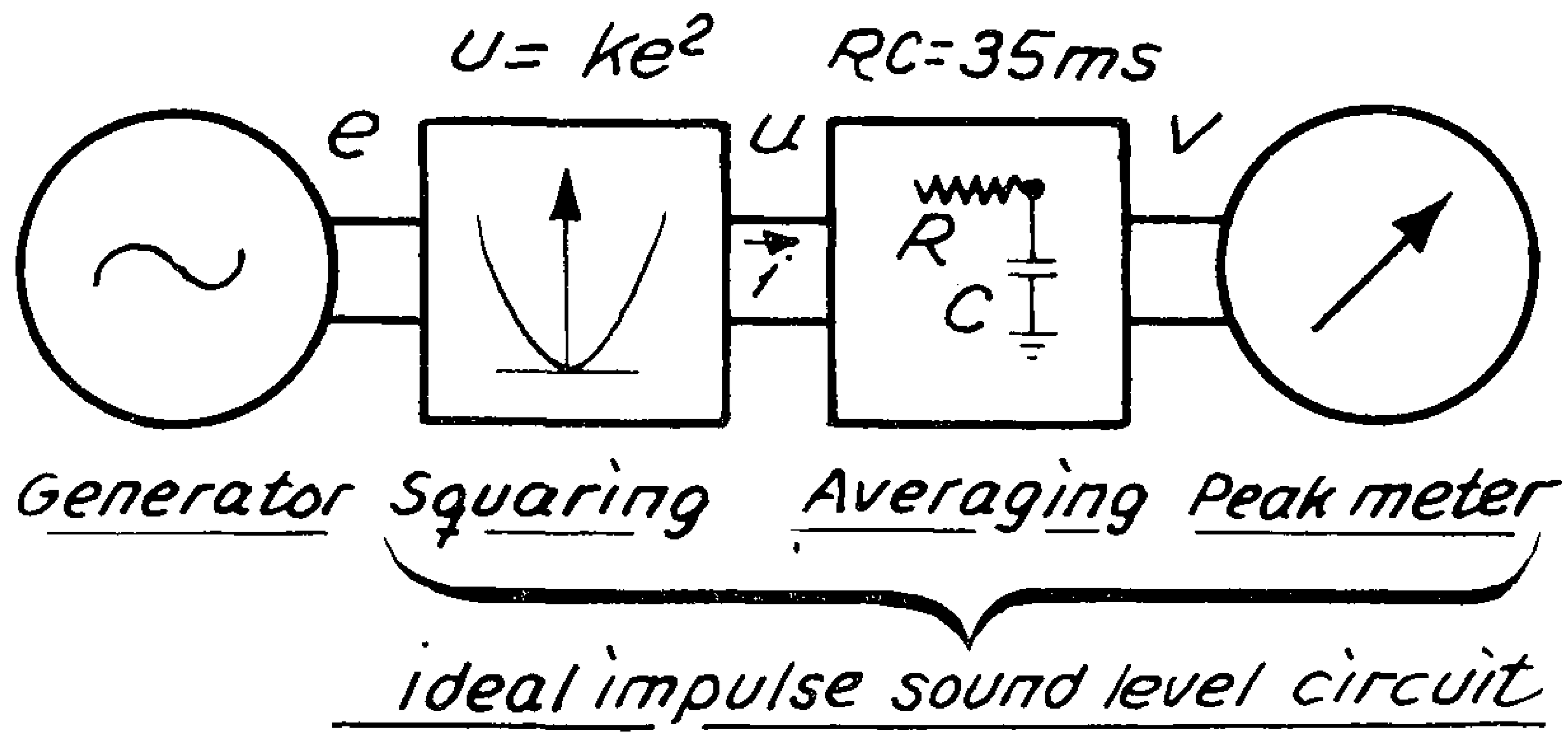

Fig. 1. Ideal impulse sound level circuit.

Let us consider first the ideal circuit: The prescribed check signals consists of tone bursts of a 2 kHz sinusoidal signal. As the RC timeconstant is long compared to a half period of the signal, the sinewave can be replaced with a squarewave with same RMS value without introducing essential differences. As the squaring circuit is symmetrical, the input can just as well be considered as a rectangular pulse of same duration as the tone burst. The output of the squaring circuit $u = ke^2$ will thus also be a rectangular pulse and the voltage v on the condenser will rise and fall exponentially with timeconstant RC according to the differential equation: $ke^2 - v = iR = RC \frac{dv}{dt}$.

As the indication should be proportional to the square root of v peak, a fictious instantaneous indication $E = \sqrt{\frac{v}{k}}$ is introduced: $e^2 - E^2 = RC \frac{dE^2}{dt}$.

This means that E^2 will rise and fall exponentially, while E will rise faster than the exponential and fall exponentially with timeconstant 2 RC as shown on Fig. 2.

As the rising curve is of special interest to the following this curve is also shown in log-log scales on Fig. 3.

In this ideal circuit the root extraction is done on the meter scale. Let us now consider a circuit where the root extraction is done in the squaring circuit itself. This is done by feeding back the voltage on the smoothing condenser to the squaring circuit, thereby changing the size of the parabola as shown on Fig. 4 (see also (1)). Actually the parabola is approximated by a polygon, but as the errors are less than 5 % and in most cases will average out to less

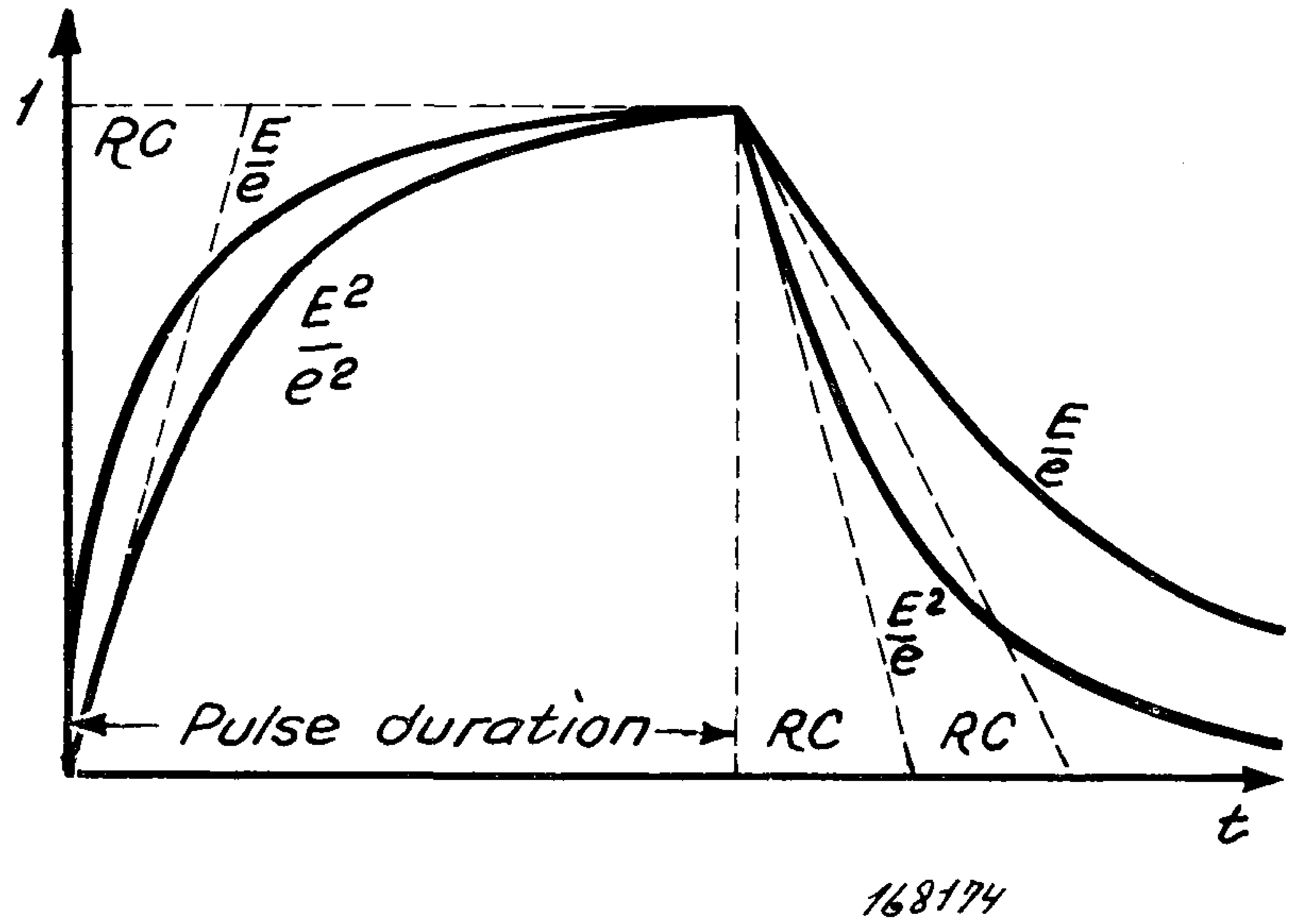

Fig. 2. Rise and fall curves.

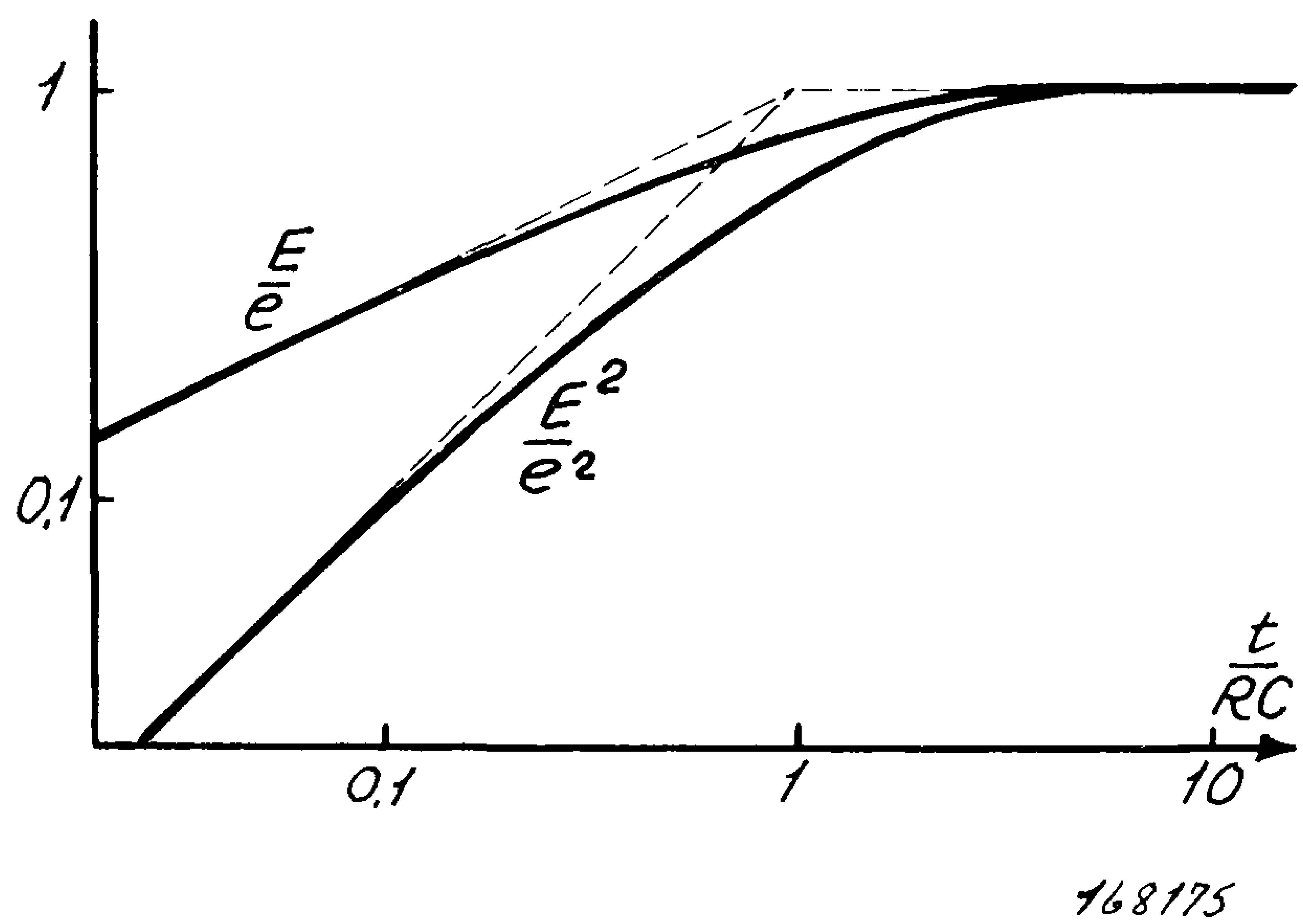

Fig. 3. Rise curves in log-log scales.

than 1 or 2 % the curve will in the following be considered as a true parabola. In Fig. 5 is shown such a circuit.

For this circuit:

$$i = \frac{h}{v}\ e^2 - \frac{v}{R} = C\,\frac{dv}{dt}$$

Again a fictuous instantaneous indication E is introduced:

$$E = \frac{v}{\sqrt{h \times R}}$$

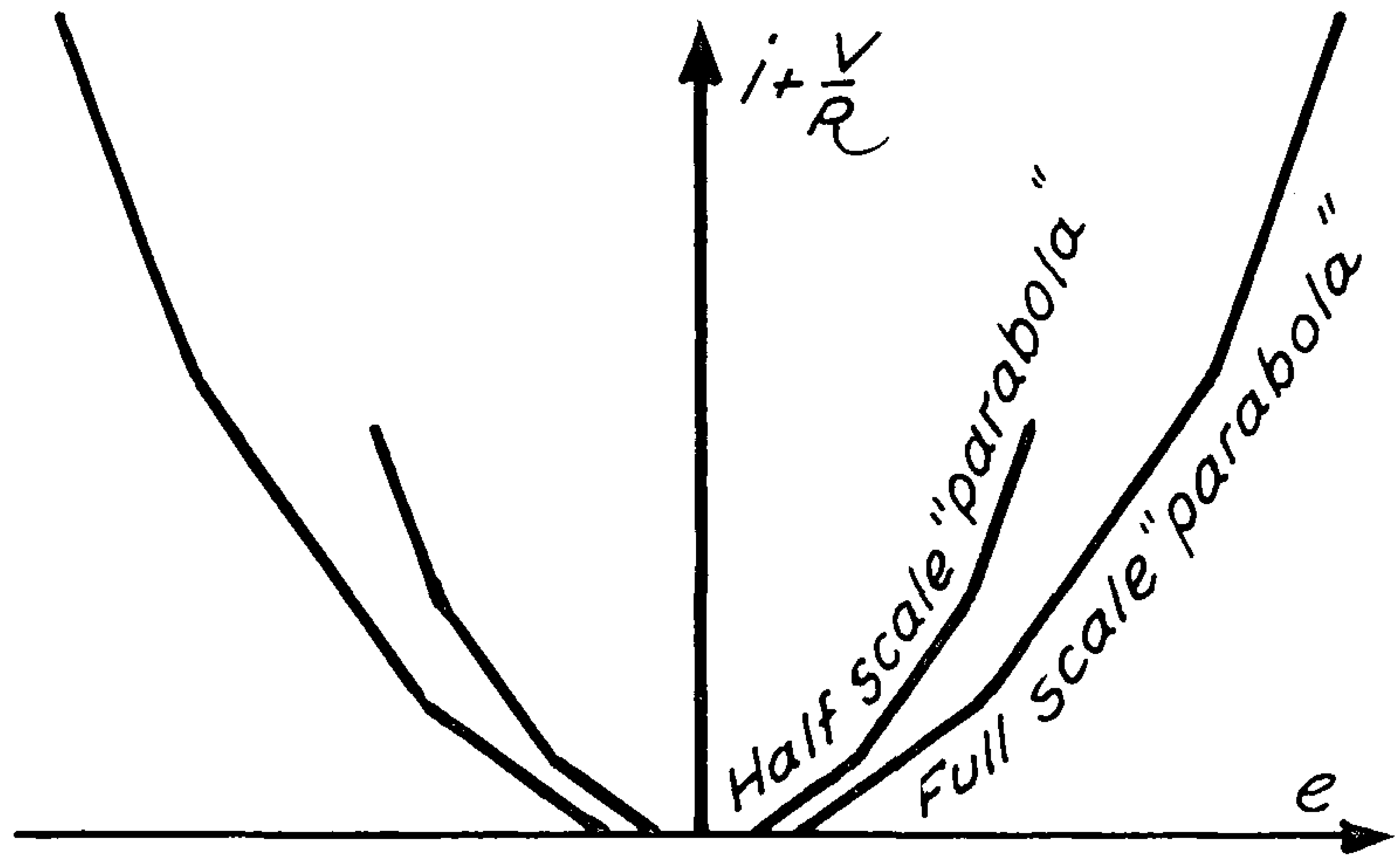

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Fig. 4. Variable squaring function.

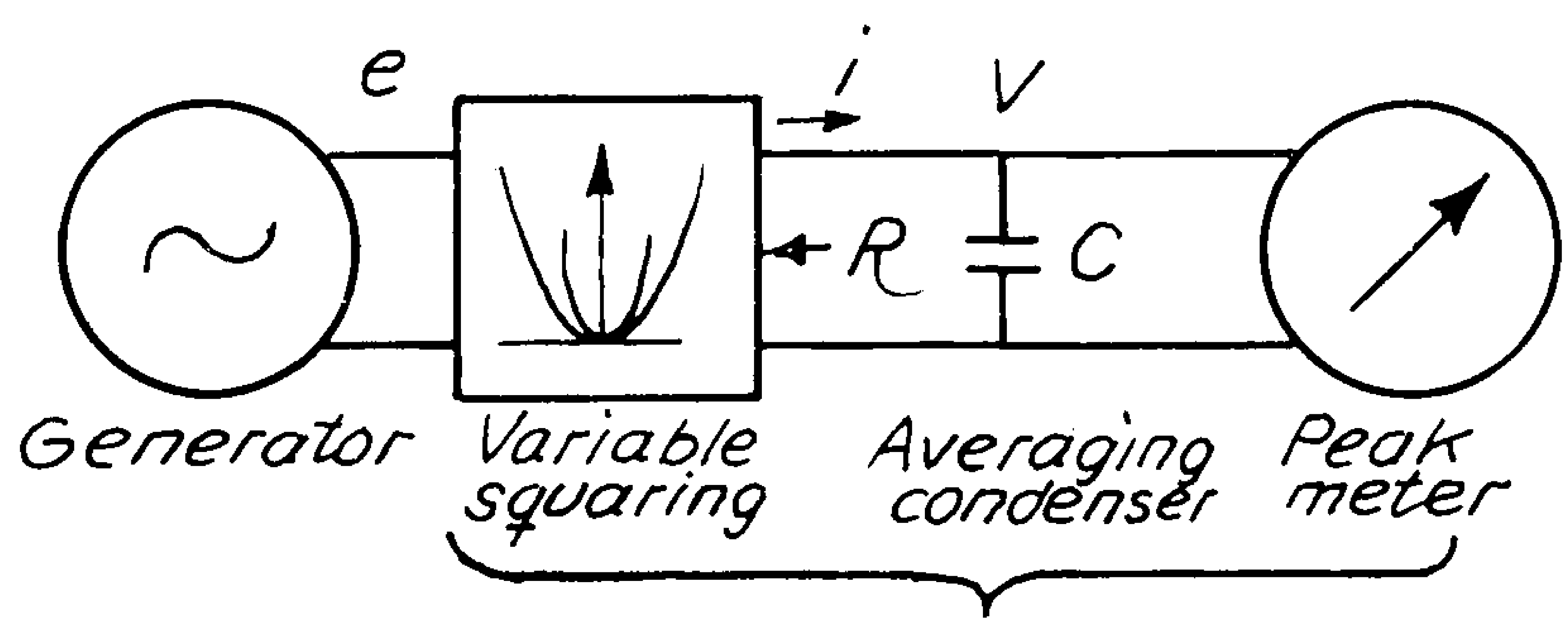

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Fig. 5. Special peak RMS circuit.

One thereby gets:

$$e^2 - E^2 = \frac{RC}{2} \frac{dE^2}{dt}.$$

This is exactly the same differential equation as for the first circuit except for the factor 1/2 on the time constant, and the rise and fall curves will therefore be the same if the time constants are adapted to each other.

However, this equality between the two circuits is correct only as long as the parabolic approximation of the squaring circuit is correct. This is most certainly not the case during the first part of the rise where the parabola is very small. In Fig. 6 is shown the variable parabola in log-log scale as a set of straight lines with slope 2 and it is shown how at the high end these lines bend over to a common line with slope 1, parallel to the line for the instantaneous indication E. If the "knee" is at $F\,E$ where F is the crest factor of the parabola, then for this part of the rise:

$$i = \frac{h}{v} F^2E^2 \frac{e}{FE} - \frac{v}{R} = C \frac{dv}{dt} \quad \text{which gives:} \quad Fe - E = RC \frac{dE}{dt}.$$

On Fig. 7 is shown the resultant total rise curve in log-log scales.
It starts as a straight line with slope 1 corresponding to an exponential rise of E with the time constant RC against the value Fe. At a value $\frac{e}{F}$, however, the curve bends over in the ideal curve where E^2 rises exponentially with the time-constant 1/2 RC against the value e^2, only is the curve delayed $\frac{1}{F^2} \times \frac{RC}{2}$. It is seen that the rise curve is reasonably near to the ideal curve at $t = \frac{RC}{2} \times \frac{4}{F^2}$. Thus to measure a single square wave tone burst of duration t msec.: $F \geqq \sqrt{\frac{140}{t}}$. For sine wave tone bursts: $F \geqq \sqrt{\frac{280}{t}}$ and for practical impulse noise signals F should probably be even a factor 1.5 to 2 higher. For the shortest test tone burst of 5 msec. $F = 7.5$.

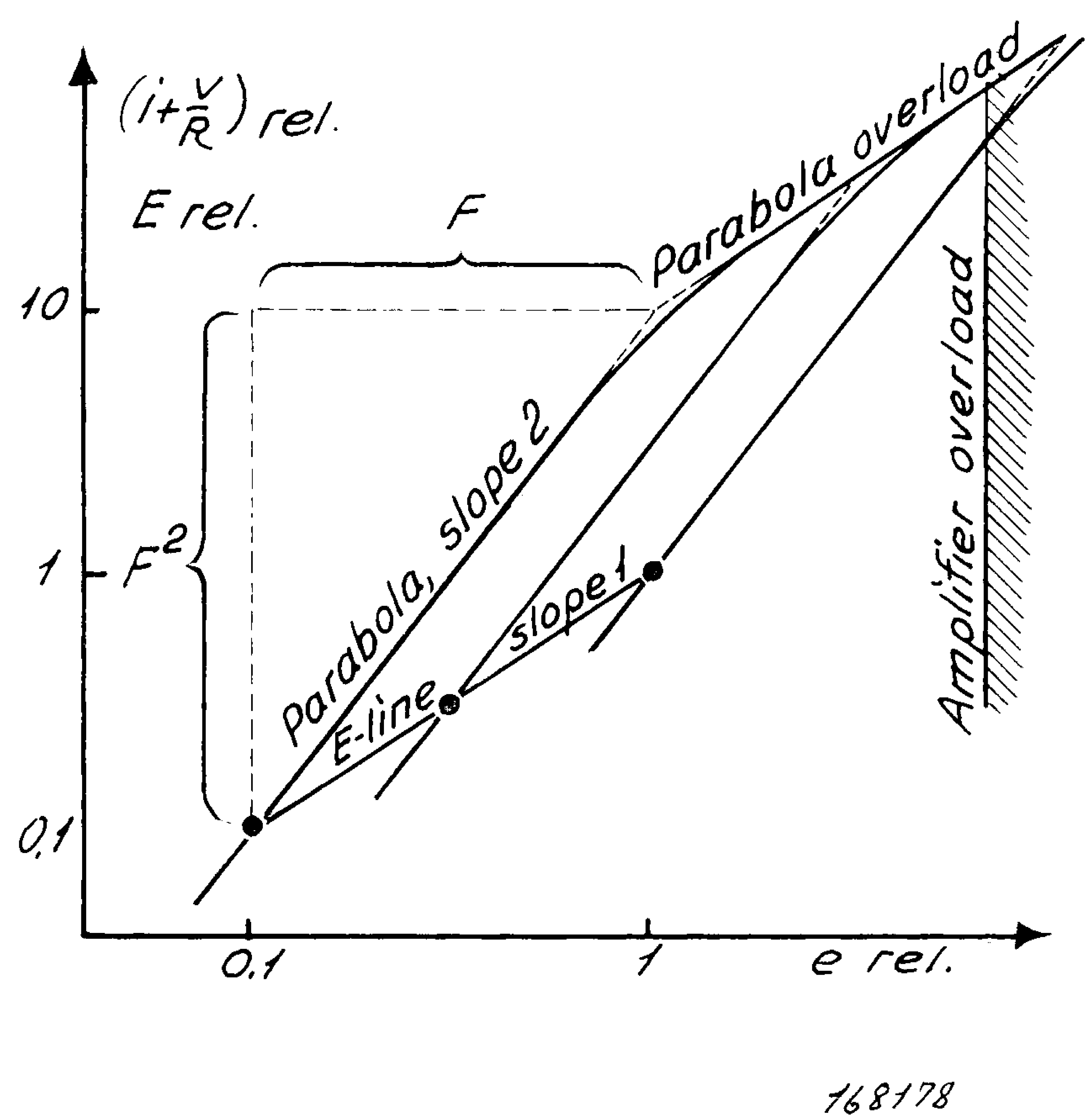


Fig. 6. Log-log plot of variable "parabola".

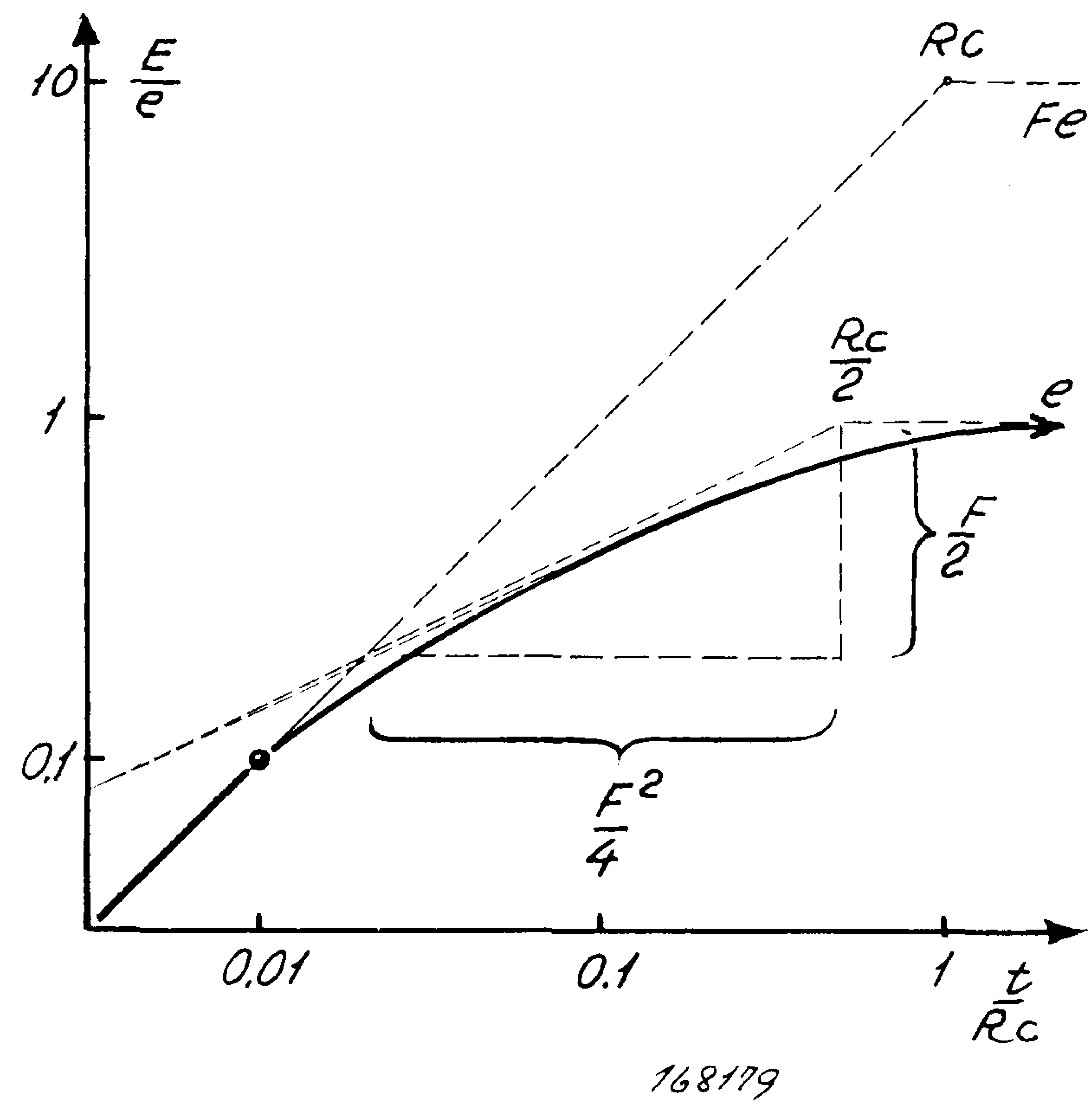
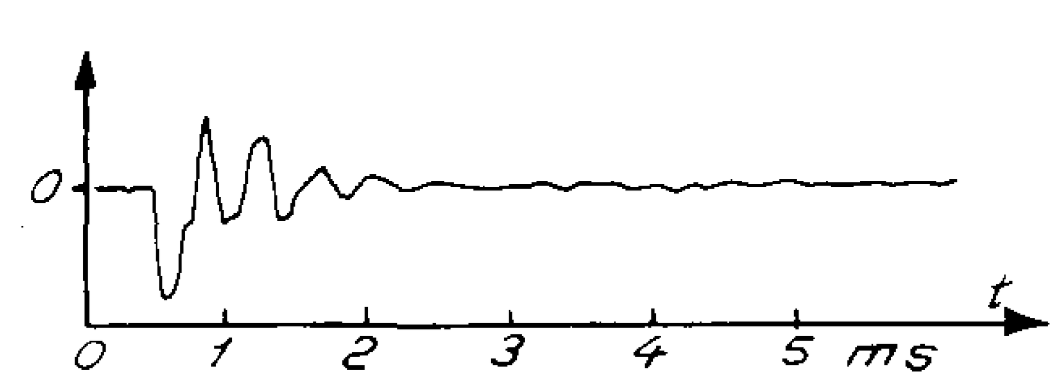


Fig. 7. Rise curve with overloaded "parabola".

Finally are shown some impulse noise measurements on practical noise sources made with a B & K impulse sound level meter, Type 2204.



Meter / Weight	Hold	Imp	Fast	Slow	Peak
N	101	100	88 *	86 *	122
A	93	92	80	76 *	115
B	91	90	78 *	75 *	115
C	91	90	78 *	75 *	115
Lin	92	91	81	76 *	116

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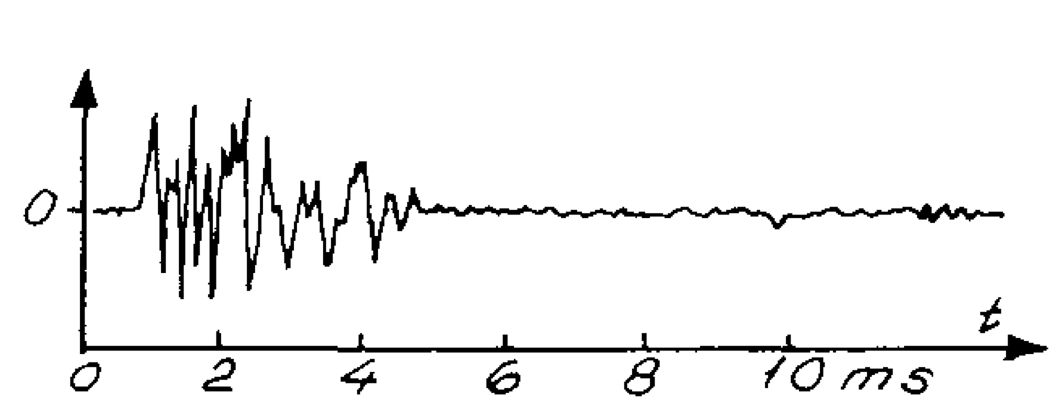
Fig. 8. Hammering metal against metal. Repetition rate: 2/sec.
** Overload indicated.*



Meter / Weight	Hold	Imp	Fast	Slow	Peak
N	94	93	84	83	112
A	85	84	75	75	103
B	84	83	74	73	102
C	84	83	74	73	102
Lin	84	84	75	74	103

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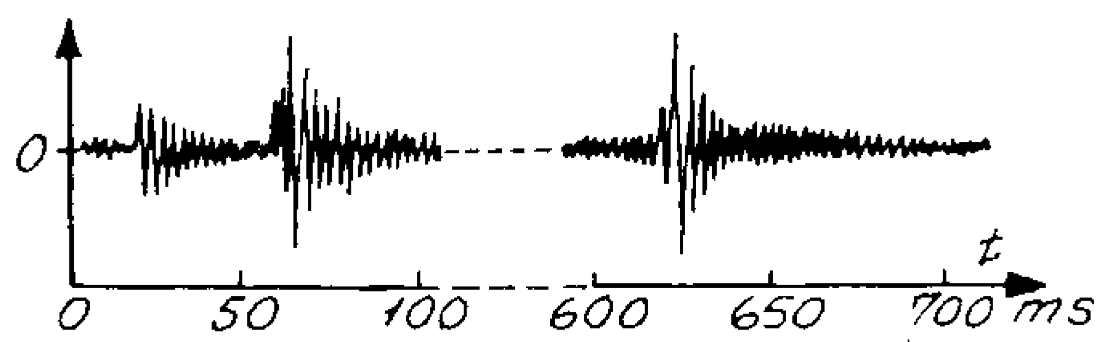
Fig. 9. Typewriter noise. Repetition rate 3/sec. Distance 0.5 meter.



Meter / Weight	Hold	Imp	Fast	Slow	Peak
N	101	101	93	90	118
A	94	94	86	84	111
B	95	95	88	85	111
C	96	95	89	86	112
Lin	96	96	90	87	112

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Fig. 10. Punch press noise. Repetition rate 1/sec. Distance 1.5 meter.

The oscillograms show that pulse durations can easily be essentially shorter than 5 msec. down to 0.2 msec. and the tables show that the peak value can be more than 20 dB above the impulse sound level reading. This means that a parabola crest factor capability of 20 to 40 is needed to give correct evaluation of these signals. The 2204 has a crest factor capability of 10 at full scale and inversely proportional to the deflection up to about 40 at 1/4 scale. The measurements also show that it is recommendable to check impulse sound level meters with tone bursts shorter than 5 msec. probably down to 0.2–0.5 msec.

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Low Frequency Measurements Using Capacitive Transducers*)

by

Frede Skøde

ABSTRACT

Techniques for detection of low frequency signals, such as encountered in sonic booms and thunder storms, are discussed briefly. A self balancing carrier system with a choice of low frequency limit going from 0.01—0.1 to 1 Hz among other features, is discussed more thoroughly.

SOMMAIRE

On discute brièvement les techniques pour la détection de signaux à basse fréquence, tels ceux que l'on rencontre dans les bangs soniques et les orages. Un système auto-équilibrant, à signal porteur, est discuté plus en détail, qui présente entre autres un choix de limites inférieures en fréquence allant de 0,010,1 à 1 Hz.

ZUSAMMENFASSUNG

Die Methoden zum Erfassen tieffrequenter Signalverläufe, beispielsweise bei Stoßwellen und Gewitter, werdenkurz diskutiert. Ein Mikrofon-Trägerfrequenzsystem mit automatischem Nullabgleich, wählbarer unterer Frequenzgrenze ab 0,01 — 0,1 — 1 Hz und weiteren Merkmalen wird ausführlicher behandelt.

The measurement of slowly varying air-pressures e.g. sonic booms, raises the problem of transducers and electrical systems which are stable and sensitive enough to convert the pressure variations into electrical signals which can be detected by indicating instruments or stored on tape for later analysis.

A typical shock-wave produced by a supersonic aircraft at altitudes above 20,000 feet has a nature at ground level as shown in Fig. 1. The peak pressure change p may be as high as 2–5 lbs/foot^2 (1000–2500 μbar) and several orders higher at low altitudes. The duration t varies from 0.03 to 0.4 seconds. A Fourier-integral of the N-wave shows that to get a good transformation of the signal the system must have a linear frequency response in the range from 2–3 decades below the lowest and above the highest fundamental frequency $1/t$. That is 0.01 Hz to 3 kHz to cover the entire range.

As transducer one of the best choices is a condenser microphone but as a standard microphone normally has an air equalization hole, which gives a lower limiting frequency of 1–3 Hz, something must be done to lower it. One way is to tighten the microphone absolutely, but to sensitive devices this will cause problems when it is exposed to very low pressure e.g. by air transport another possibility is to connect the volume behind the diaphragm to a larger volume with constant pressure. This will give an increase in sensitivity to the microphone around 10 % at frequencies lower than that determined by the air equalization hole. A new type of microphone, Fig. 3, can be modified for

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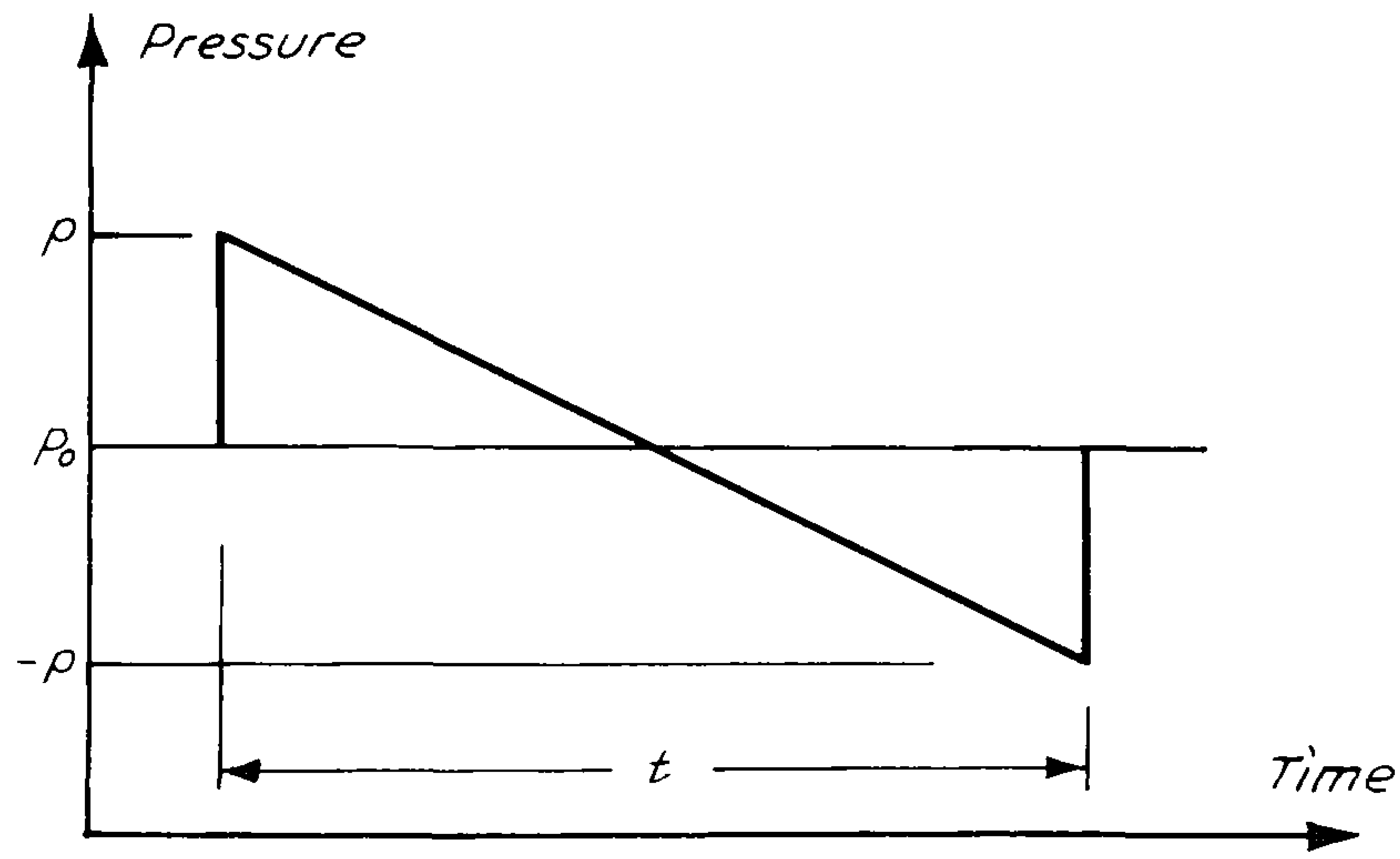

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Fig. 1. Idealized shock wave.

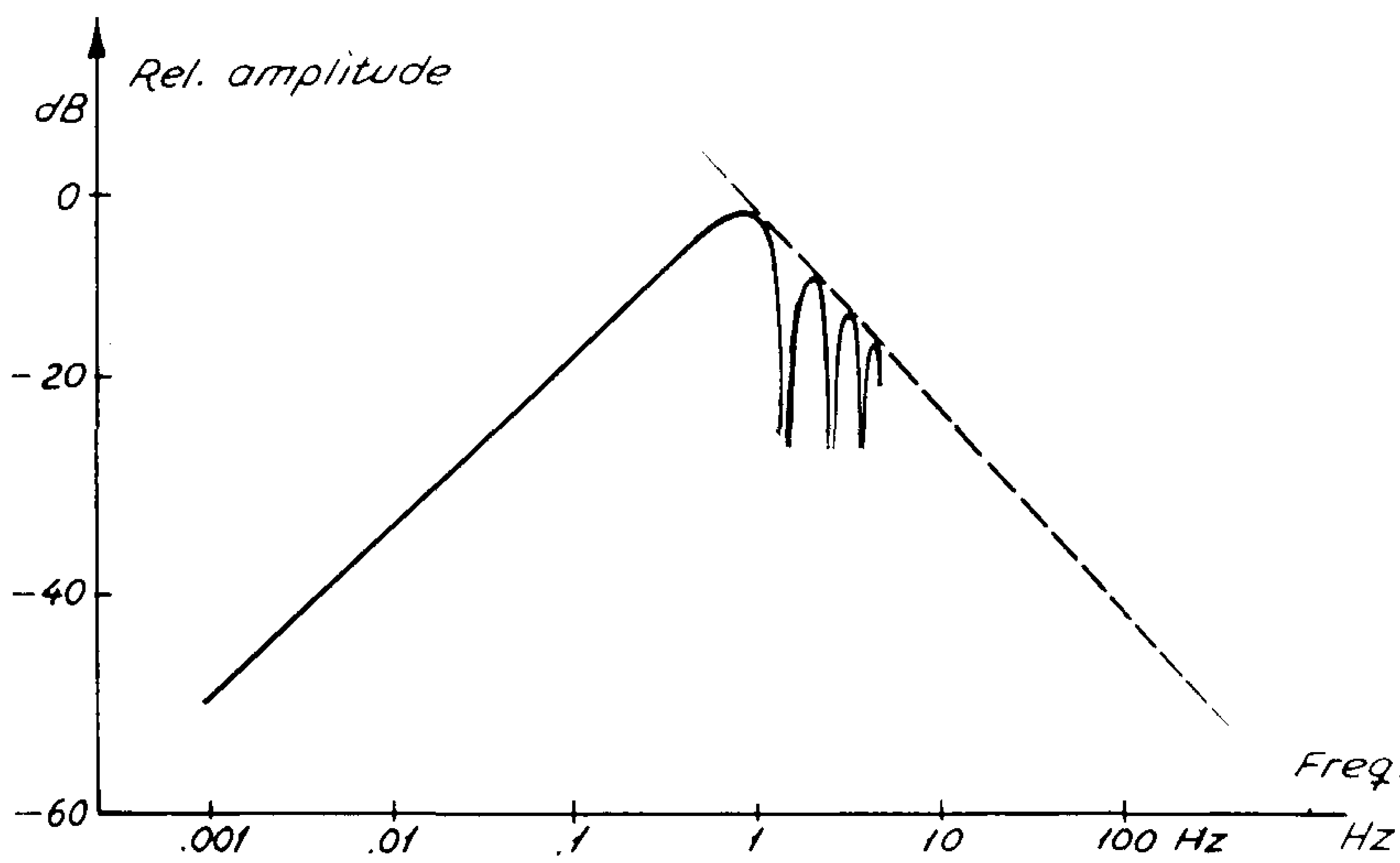

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Fig. 2. Fourier integral of N-wave.

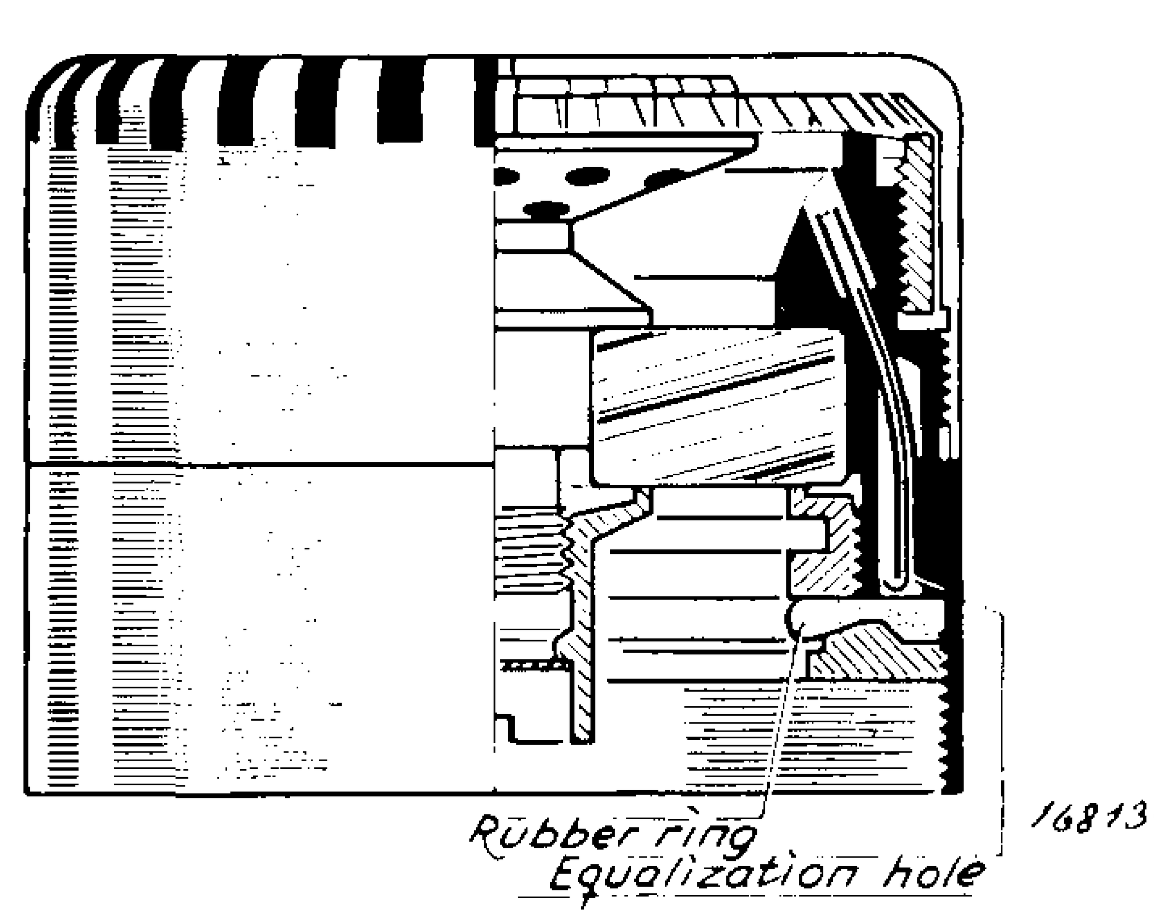


Fig. 3

Low frequency microphone

Fig. 3. Low frequency microphone.

low frequency operation by mounting a silicone-rubber ring which closes the equalization hole. The lower limiting frequency will then be in the order of 0.01 to 0.02 Hz.

To detect the slowly varying capacitance changes of the microphone ordinary preamplifiers must have a very high input impedance which is difficult to obtain and difficult to maintain under humid conditions. To make the microphone act as a low impedance, the capacitance of it can be measured at a frequency which is well above the frequency range of the microphone itself. This may be done in several ways. The microphone may be compared to a well known capacitor in a bridge circuit. The advantage of this is that the circuit can be made very stable and independent of the frequency of the generator, on the other hand it is difficult to detect the signals from such a bridge over a wide dynamic range.

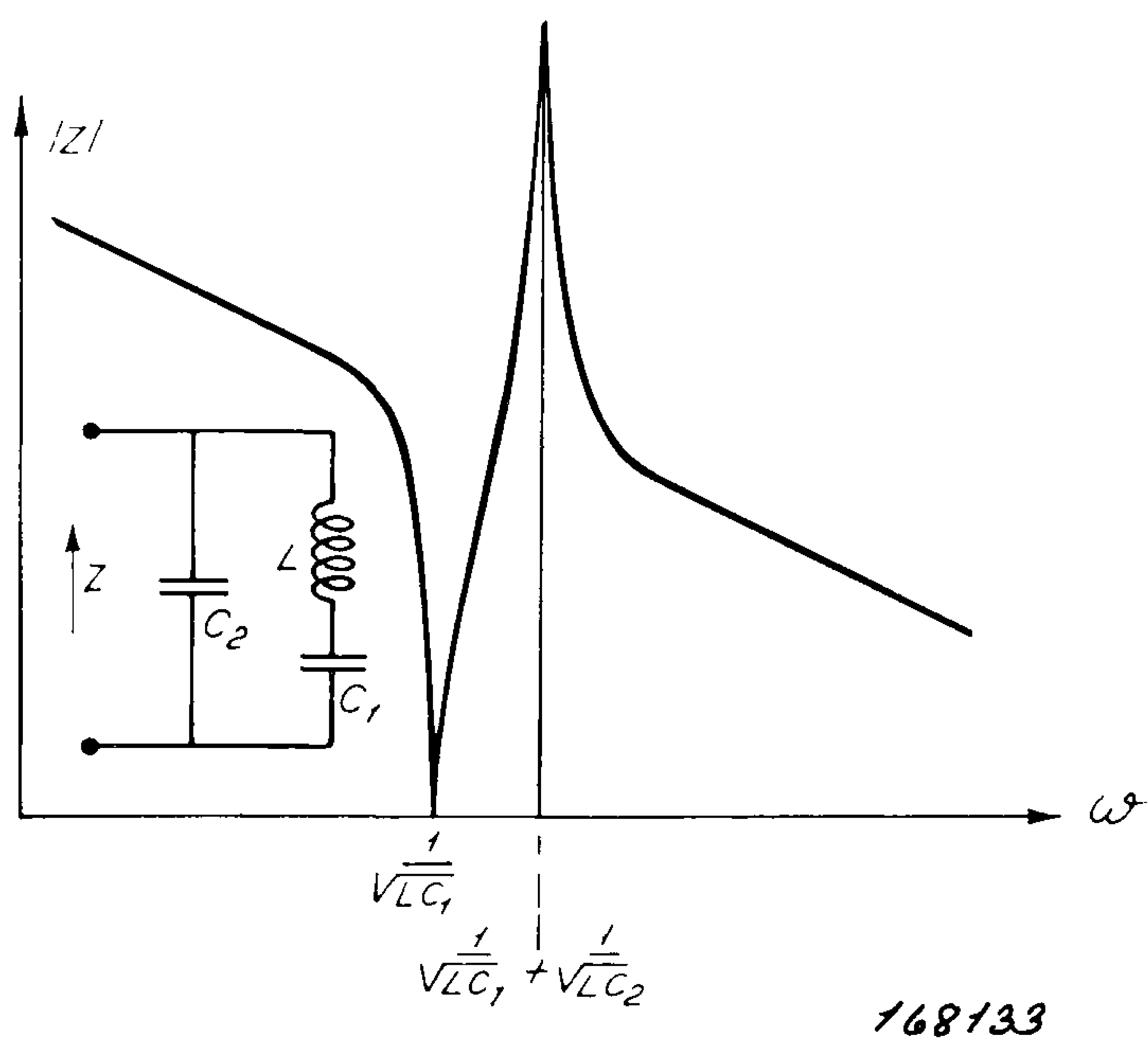


Fig. 4. Series circuit with parallel capacitor.

A very often used solution is a circuit as shown in Fig. 4 where C_1 is the microphone and C_2 a rather large condenser, including cable. This circuit can be used in two ways, the impedance around the resonant points can be detected at a constant frequency, or the circuit can be used as a frequency determining part of an oscillator the output signal of which can then be detected in a FM detector. The main advantage of this system is that the microphone and a coil adaptor can be connected through a two-wire cable to detector and power supply. The disadvantage is that it is difficult to make the system sensitive as the quality factor of the circuit will be rather low. A parallel circuit can be used in the same ways as the series circuit but requires that signal-detection is close to the microphone. The main features

of this system is that it is possible to make it very sensitive and get a large dynamic range and a good stability. A system using this principle was built and will be discussed more thoroughly.
It was decided that the system should fit direct to standard 1″ condenser microphones with a capacitance ranging from 50–80 pF therefore the microphone adaptor should have a diameter of 15/16″ and a length in the order of 3″. To avoid transmission of high frequencies in a cable, the adaptor should contain the oscillator and the detector, to be able to drive long cables it should also contain a preamplifier. Tuning of the circuit should be done by an external voltage.
The limitation in space forces the use of simple circuits and small components. A schematic drawing of the system is shown in Fig. 5.

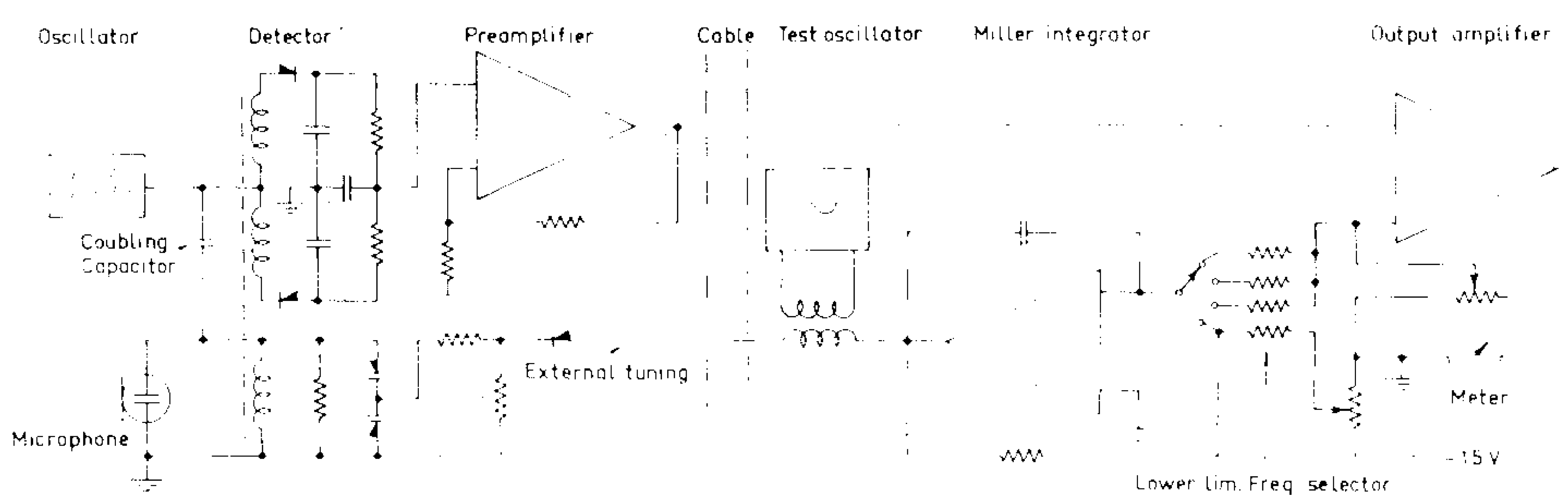


Fig. 5. Schematic drawing of the system.

The quality factor of the tuned circuit is determined by the phase shift, that gives usable linearity and the max. capacitance variation of the microphone which is about 6 pF for linear operation. L is about 2.5 H and the capacitance of microphone and tuning capacitance diodes is 105 pF. Quality factor is in the order of 15. The generator is a 10 Mc crystal controlled oscillator with a single transistor and it is driven to bottom to give a constant voltage, which is in the order of 5 V_{RMS}. The preamplifier is a monolitic circuit. The tuning voltage is applied through a diode to compensate for thermal capacitance variations of the capacitance diodes.
The adaptor containing these items is connected through a 5 wire cable to a power supply which also contains the rest of the system. The signal is amplified in an output amplifier and part of the output signal is applied to a Miller-integrator with three different time constants, the output from the integrator is used to tune the microphone circuit. The time constant is set to give the system a lower limiting frequency of 0.01–0.1 and 1 Hz. To obtain this with a reasonable condenser the biggest resistor should be 20 MΩ, and to avoid current off-set it is necessary to use a differential pair of field effect transistors in the input of the integrator.

In the fourth position the circuit can be tuned by a potentiometer to make the electrical system DC sensitive. It is possible to apply a 1 kHz signal to the tuning voltage and in this way produce a capacitance variation which can be detected at the output and in this way test the entire sysem. The meter indicates overload and balance in the DC mode.

Data Obtained	*System Alone*	*System with Microphone*
Capacitance range	50–80 pF	
Frequency range	DC – 200 kHz	0.01 Hz – 18 kHz
Linear range	± 6 pF	± 2000 μbar
Noise: 0.01 Hz – 200 kHz		0.8 μbar_{peak}
2 Hz – 200 kHz		0.2 μbar_{peak}
2 Hz – 20 kHz		0.15 μbar_{peak}
Stability in DC-mode	0.4 pF/day 0.015 pF/°C	30 μbar/day, 1 μbar/°C
Output of adaptor	± 12 V, ± 10 mA	
Output of amplifier	± 12 V, ± 12 mA	
Power	+ 15 V, 20 mA	– 15 V, 22 mA

It is pcssible to cover other ranges than described. The microphone here is made for a polarization voltage of 200 Volts. As this is not applied, the microphone may be made more sensitive and the quality factor of the tuned circuit may be higher in order to be able to detect lower levels. For detection of higher levels a less sensitive microphone may be used.

Details in the Construction of a Piezo-electric Microphone*)

K. Styhr Hansen

ABSTRACT

Some of the problems in constructing a piezoelectric microphone are the suspension of the piezoelectric element and the damping of resonances. A simple method of realizing a hinged suspension is shown and a construction is described in which the air damping can be adjusted to any value within a wide range.

SOMMAIRE

Parmi les problèmes rencontrés dans la construction d'un microphone piézoélectrique il y a la suspension de l'élément piézo-électrique et l'amortissement des rsonances. On montre une méthode simple de réaliser une suspension pivotante et l'on décrit une construction dans laquelle l'amortissement de l'air peut être ajust à n'importe quelle valeur dans une grande étendue.

ZUSAMMENFASSUNG

Bei der Konstruktion eines piezoelektrischen Mikrofons sind die Aufhängung des piezoelektrischen Elementes und die Dämpfung der Resonanz problematisch. Dieser Artikel zeigt und beschreibt eine einfache Ausfshrung einer gelenkigen Aufhängung und einen Mikrofonaufbau, bei dem die Luftdämpfung auf jeden beliebigen West innerhalb weiter grenzen einstellbar ist.

Two of the basic problems facing the designer of a piezoelectric microphone using a bender type ceramic element are how to mount the bender, and how to attenuate its natural resonance.

This paper describes some of the considerations involved and shows how the problems have been solved in the practical construction of a microphone.

Mounting of the Bending Element

Basically two different methods of mounting seem appropriate: Simple support of both ends of the bending element, or cantilever mounting. For a given microphone diameter the simply supported bender can be twice as long as that used for cantilever mounting.

Fig. 1 shows the shape of the bending moment curve for the two cases when the bending element is loaded by the same force, *P*. If the bending moment curves are integrated, the value of the integral will be the same in both cases. This again means that the charge sensitivity of the design is also equal in both cases. However, the compliance of the cantilever construction is twice that of the simply supported bender. To reduce losses from air stiffness and from edge stiffness of the microphone diaphragm the compliance of the bending element construction should be small, a fact which favours the simply supported construction with a factor of 2.

*) Paper presented at the 6th International Congress on Acoustics, Tokyo, Japan 21 – 28 August 1968.

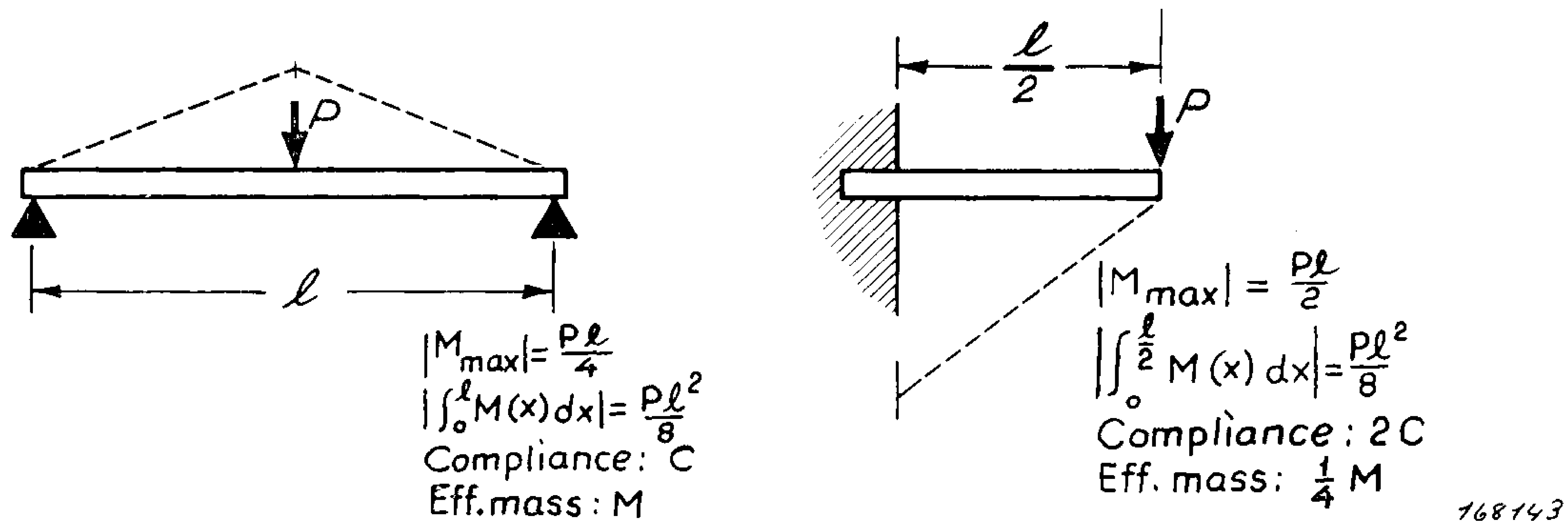
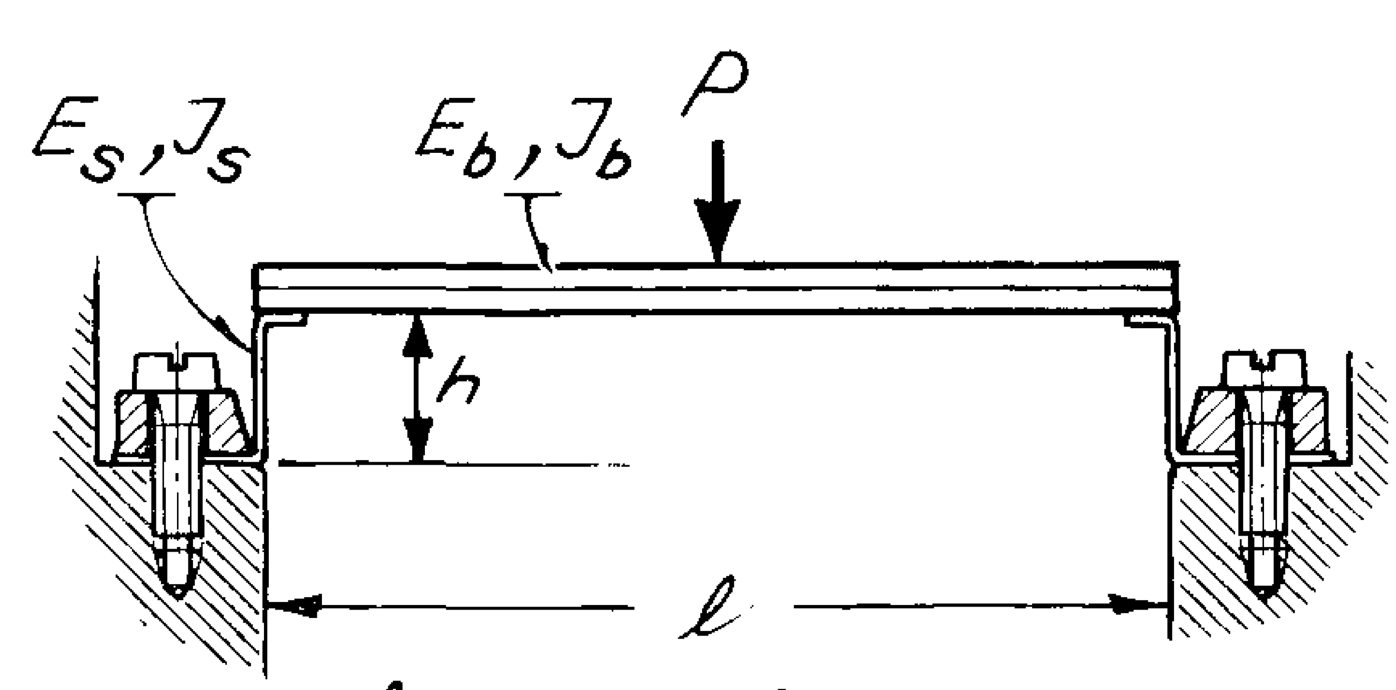


Fig. 1. Bender on two simple supports and a "cantilever" arrangement.

Another important fact to be considered is the effective mass of the construction. This should be small to reduce the sensitivity to vibrations of the microphone. For the simply supported bender case the effective mass is four times larger than for the cantilever type construction. It should, on the other hand, be pointed out that in the microphone design considered the mass of the diaphragm is of the same order of magnitude as the mass of the bender. Thus the sensitivity to vibration is only a factor 2 larger in the simply supported bender case than in the cantilever case.

However, a fact which actually decided the type of support finally chosen for the practical construction was that the angular displacement of the bender at the point where this is loaded by the microphone diaphragm is zero for the simply supported system, while it is maximum for the cantilever type construction. If an angular displacement is transferred to the diaphragm, this will no longer describe pure translational movements only but it will also be twisted, which is highly undesirable for the achievement of sufficient attenuation of the system resonance.

Fig. 2 shows how the simply supported construction was realized in practice. On both ends of the bending element a bronze band has been attached, bent at right angles and fastened by means of a ring. The integral of the bending moment curve is then:



$$\int_0^\ell M(x)dx = \frac{P\ell^2}{8} \cdot \frac{1}{1+2ab}$$

$$a = \frac{E_s \cdot I_s}{E_b \cdot I_b}, \text{ and } b = \frac{\ell}{h}$$

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Fig. 2. Realization of the simple support.

$$\int_{0}^{l} M(x)\,dx = \frac{Pl^2}{8} \times \frac{1}{1 + 2\,ab}$$

where

$$a = \frac{E_s \times I_s}{E_b \times I_b}\ , \text{ and } b = \frac{l}{h}$$

It can be seen that this expression tends towards $Pl^2/8$, as in the case of the simply supported bender, when $a \times b$ goes to 0, i.e. when long, thin bronze bands are used. Also it is seen that when $a \times b$ tends towards infinity the above expression and thus also the microphone sensitivity, tends towards zero. In the actual case the bronze bands are 0.078 mm thick and 2 mm long. This results in a reduction in sensitivity of approximately 5 % relative to the maximum achieveable sensitivity. One of the advantages of this realization of a simple support is that it is very stable because it consists of fixed joints only.

Attenuation of the Natural Resonance

One of the requirements to the microphone was that it should have a flat free-field response up to around 10 kHz for sound incidence perpendicular to the diaphragm.

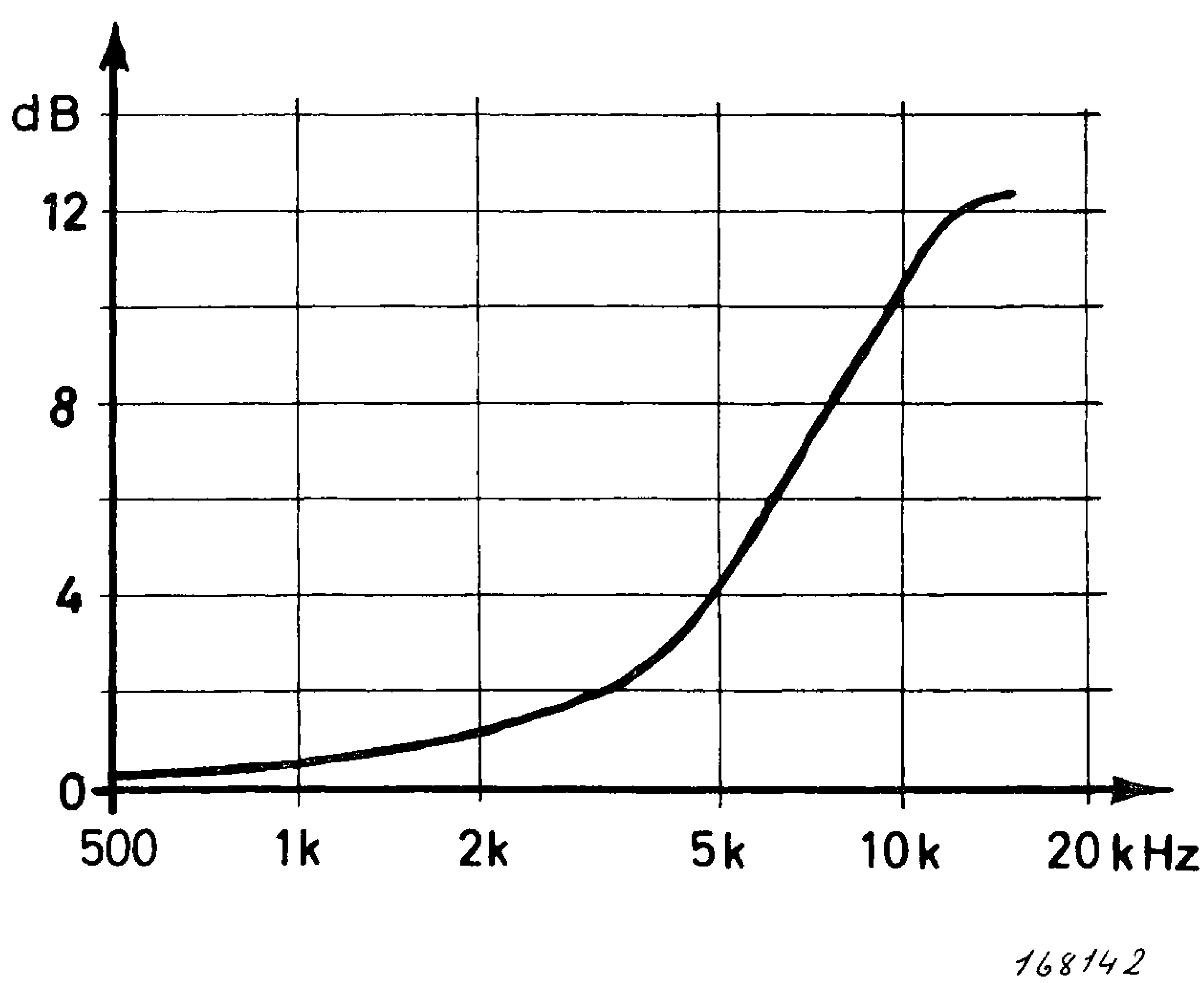

Fig. 3. Pressure increase for incidence of sound perpendicular to diaphragm.

Due to the diffractions around the microphone a frequency dependent pressure increase is formed at the microphone diaphragm as shown in Fig. 3. To achieve a flat free-field characteristic it is therefore necessary that the

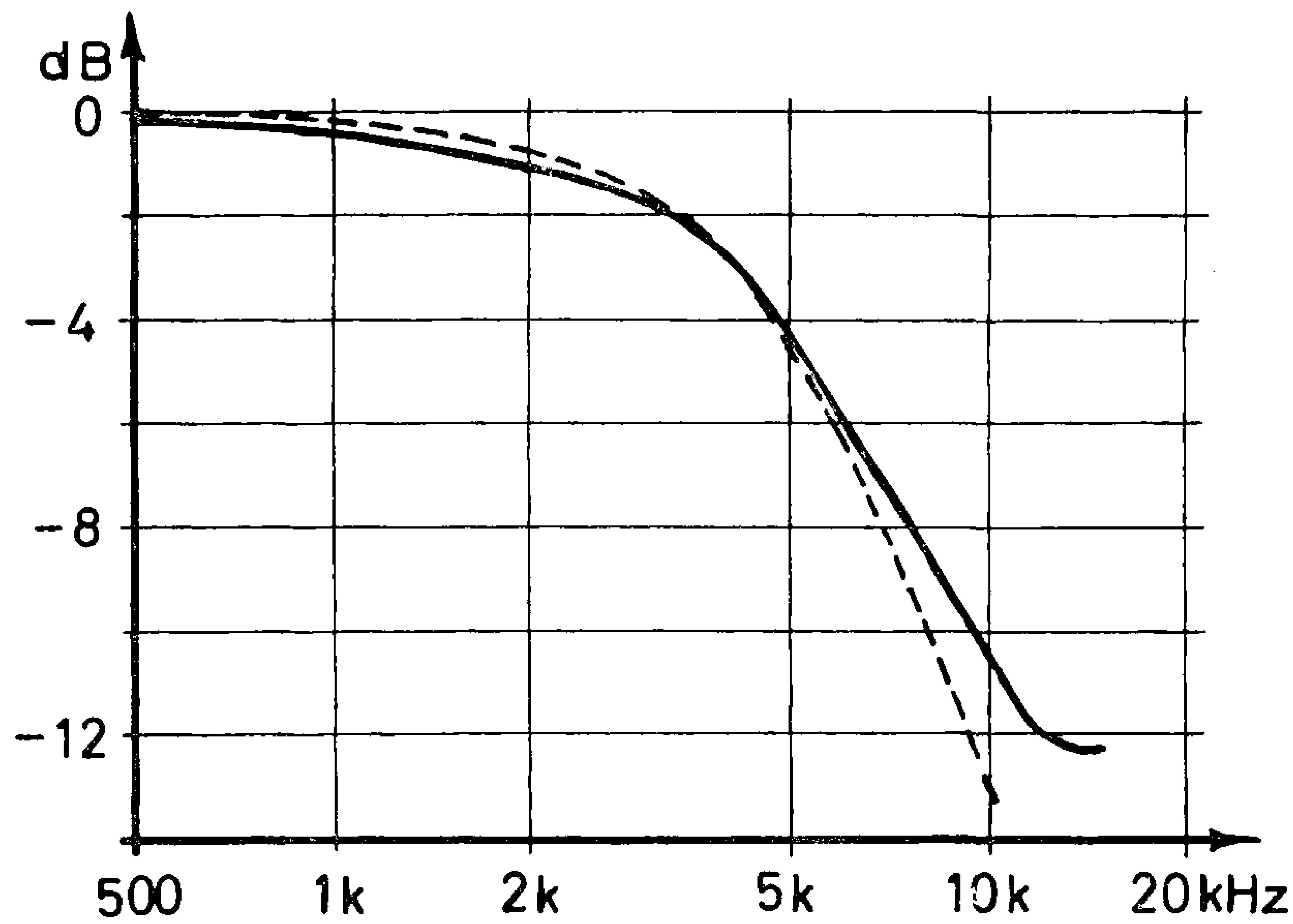

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Fig. 4. Solid curve: Desired pressure response to give flat free-field response. Dotted curve: Voltage across capacitor in series resonant circuit, $f_0 = 5$ kHz, $d = 1.7$.

pressure reponse of the microphone shows a fall-off with frequency which corresponds to the above mentioned increase in pressure, see Fig. 4. On the figure is also shown the voltage across the capacitor in a series resonance circuit having a resonance frequency of 5 kHz and a loss-factor of 1.7. As can be seen, this curve is very close to the desired pressure characteristic.

Originally a microphone construction similar to that shown in Fig. 5 was suggested. Here a number of holes, covered by acoustic damping material, e.g. tightly woven cloth, connects the cavity behind the microphone diaphragm with a somewhat larger cavity. An approximately equivalent circuit for this

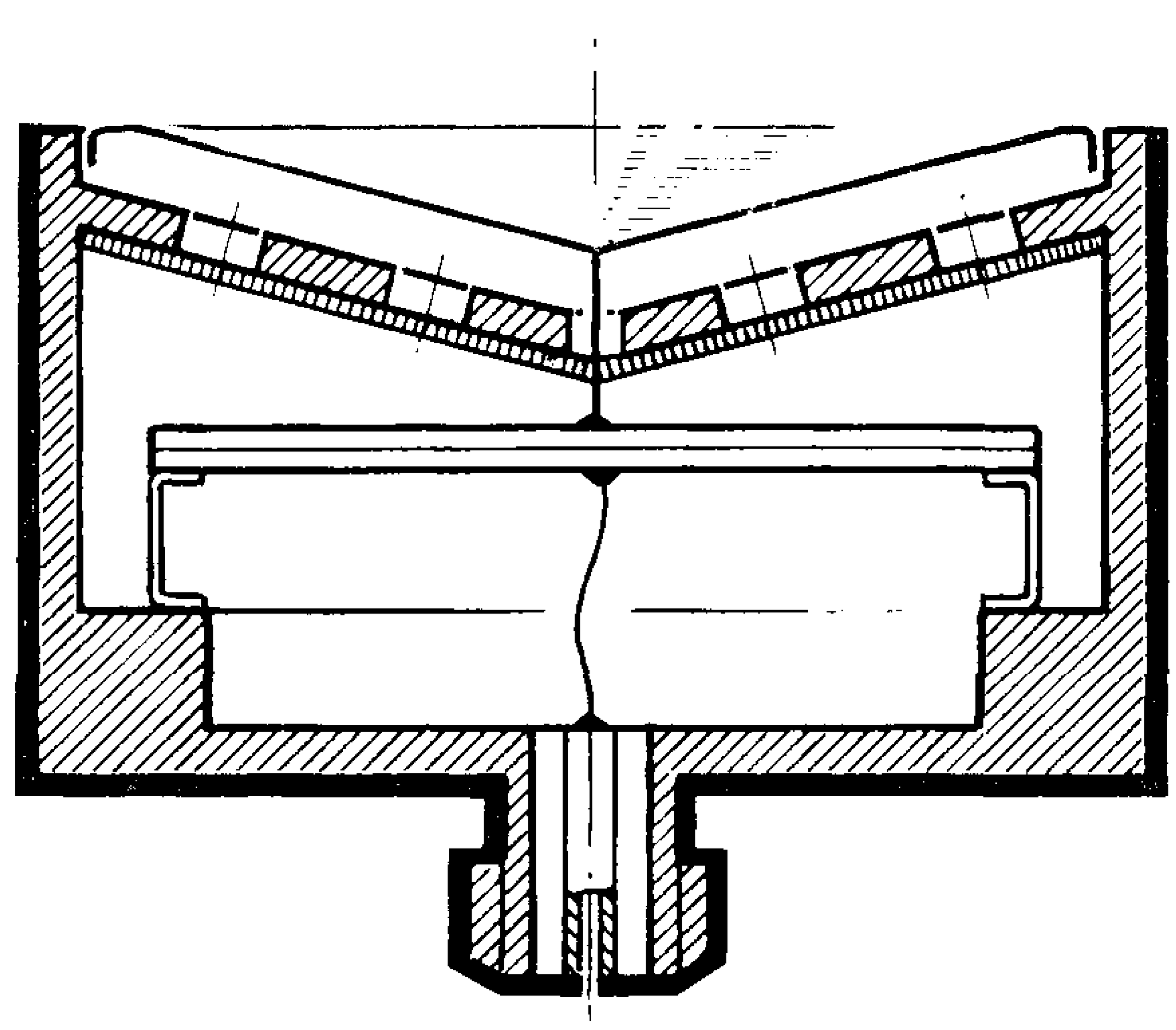

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Fig. 5. First sketch of the microphone.

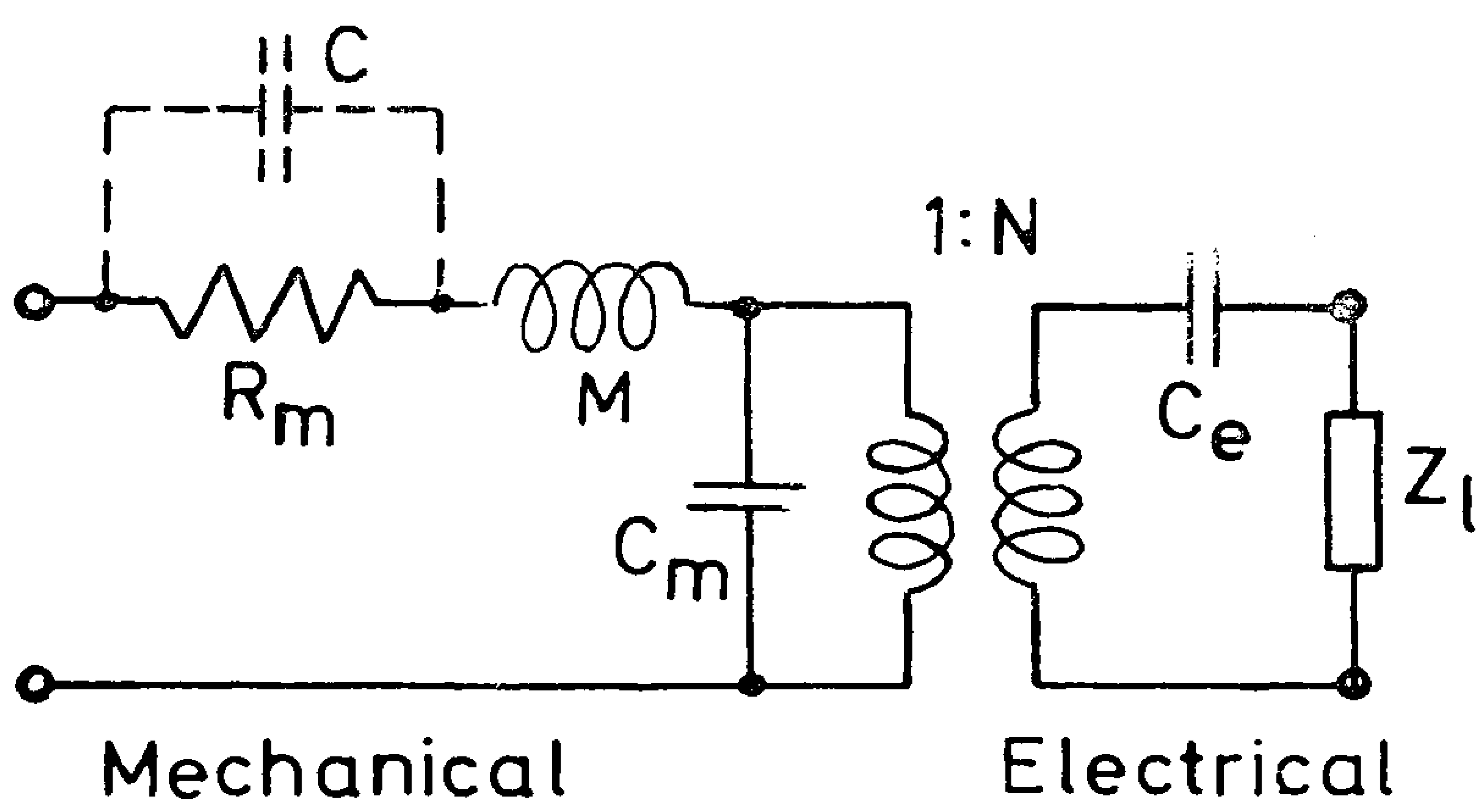

Fig. 6. Simplified equivalent circuit.

type of design is shown in Fig. 6. The "dashed" capacity represents the cavity just behind the diaphragm. It can be shown, quite easily, that with the ceramic bending elements presently available, and with a desirable loss-factor of 1.7, the cavity behind the diaphragm must be smaller than some 0.03 cm^3. For a microphone diameter of 23.77 mm this corresponds to a distance between the diaphragm and the back-"wall" of roughly 0.1 mm. A distance of this order of magnitude requires, however, an extreme conformity of diaphragm and back-"wall". This again requires extreme care in the manufacture and

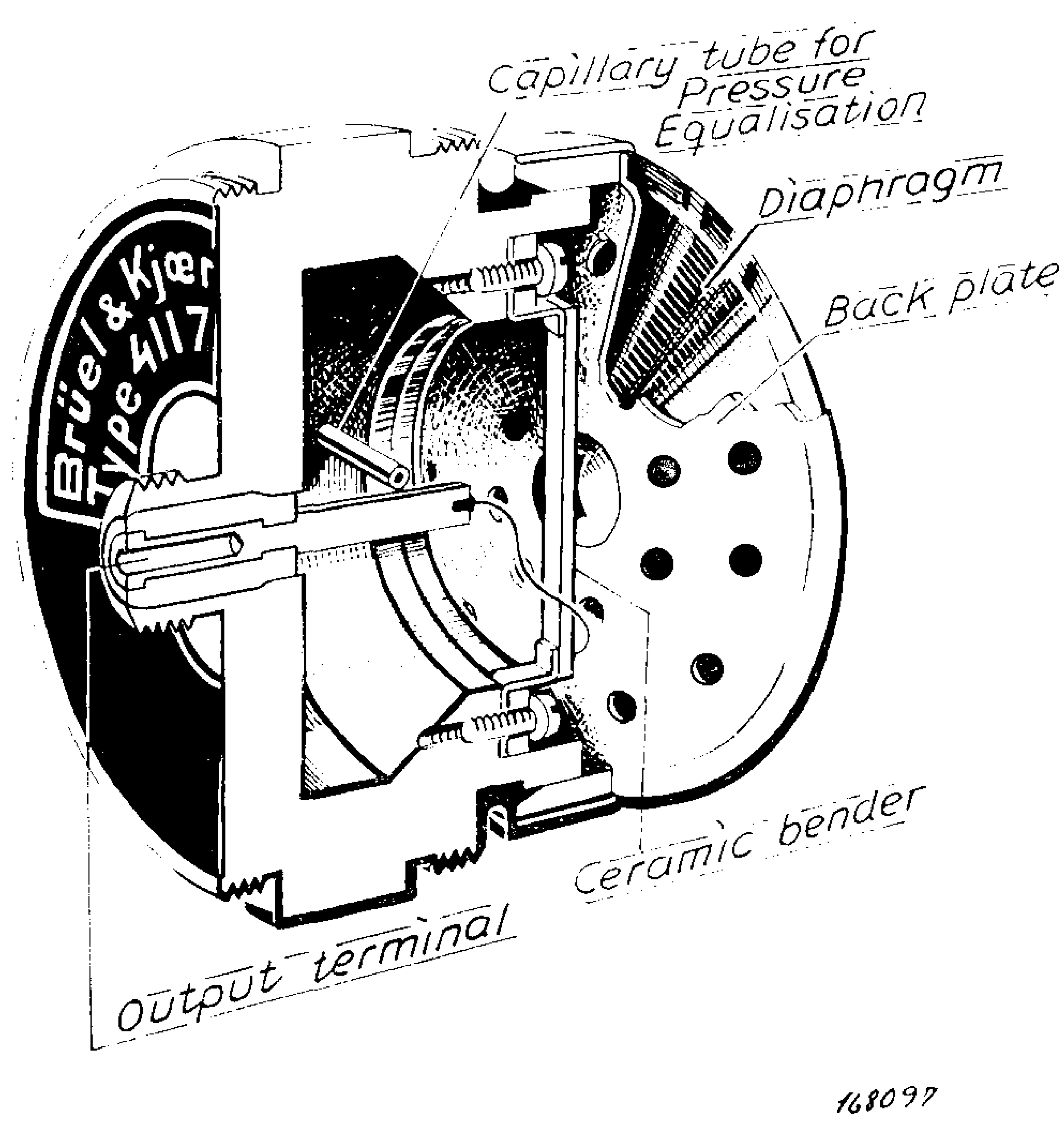

Fig. 7. Exploded view of the microphone.

assembly of the microphone, and it was therefore decided to try to obtain the desired loss-factor solely by means of the air viscosity in a narrow cavity between the diaphragm and a back-plate.

The design is shown in Fig. 7. As can be seen, no acoustic damping material has been used, whereby tightening problems have been avoided and the diaphragm has been connected directly to the bending element. A further advantage of this construction is that the loss-factor can be adjusted to the desired value simply by varying the distance between the diaphragm and the back-plate. In this way it is possible to vary the loss-factor within very wide limits.

Fig. 8 shows the pressure characteristic of the microphone for different distances between the diaphragm and the back-plate, measured by means of an electrostatic actuator. The loss-factor varies here between some 0.35 and 5. On the curves some irregularities can be seen at the high frequency end. These are caused by resonances in the back-plate.

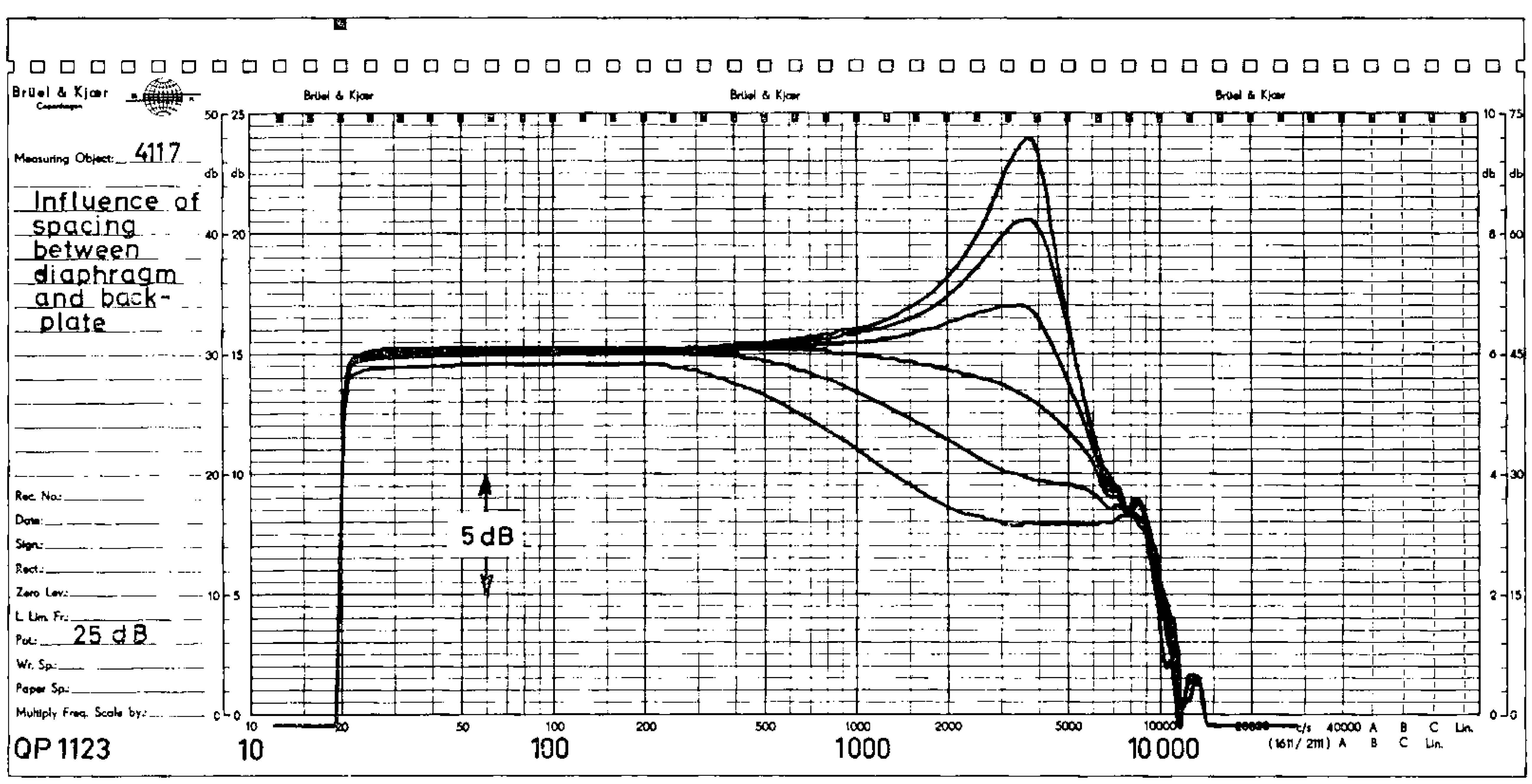

Fig. 8. Pressure response for different spacing between the diaphragm and back plate.

Finally, Fig. 9 shows an example of the frequency characteristic achieved by means of the above mentioned measures. The lower curve represents the microphone pressure response measured by means of an electrostatic actuator while the upper curve is the free-field response to sound incidence perpendicular to the diaphragm.

The following data have been obtained for the microphone:

Voltage sensitivity: 0.3 mV/μbar
Capacity: 4000 pF
Resonance frequency: (90° phase shift) 4.8 kHz
Lower Limiting Frequency: 3 Hz
Sensitivity to vibrations: 1 g represents 100 dB S.P.L.

Components in the Equivalent Circuit:

$$M = 45 \times 10^{-6}\ \mathrm{Kg}$$

$$C_m = 24.4 \times 10^{-6}\ \frac{m}{N}$$

$$R_m = 2.31\ \frac{N \times s}{m}$$

$$N = 10\ \frac{V}{N}$$

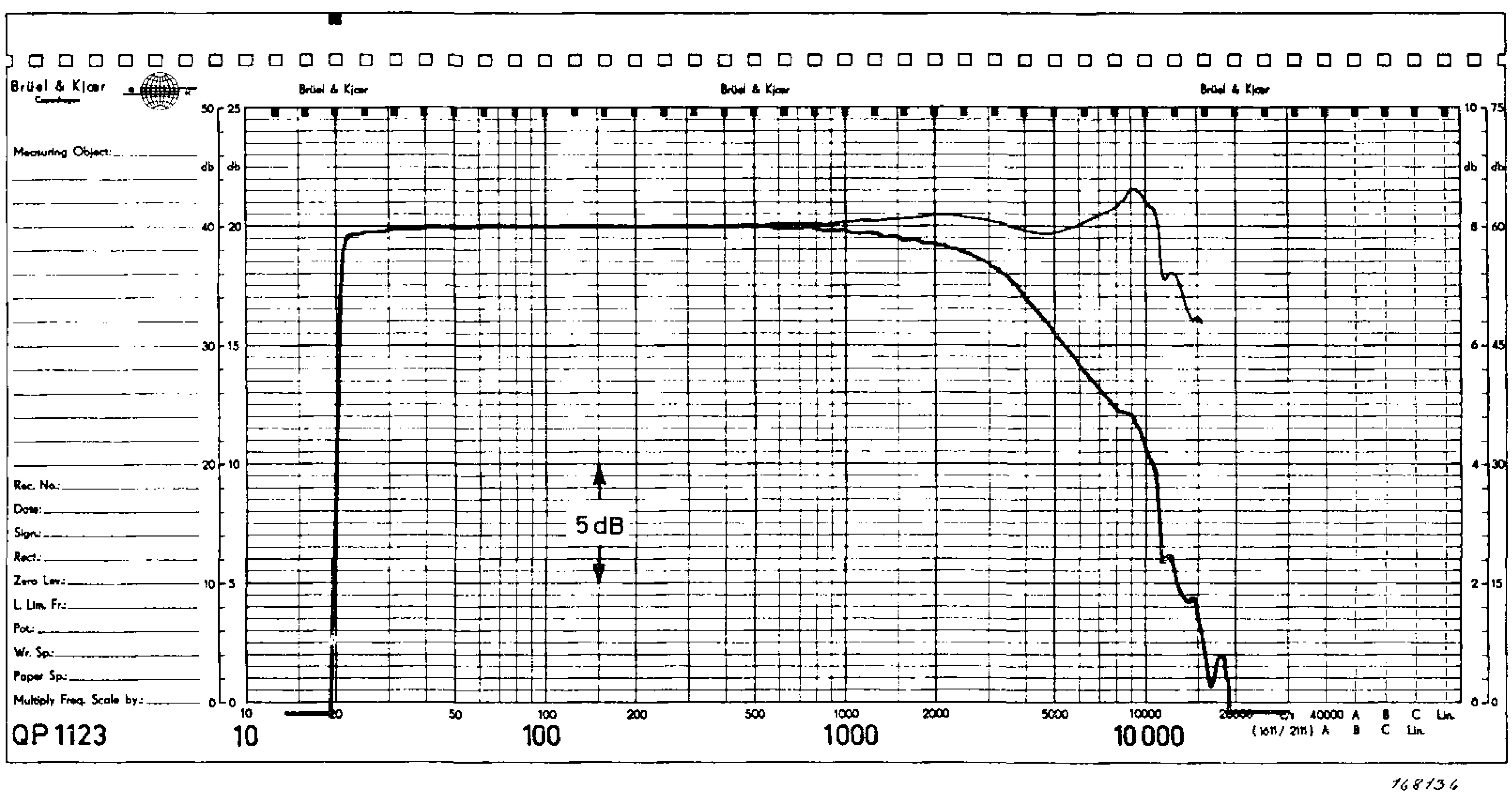

Fig. 9. Example of frequency response.
Upper curve: Free-field response.
Lower curve: Pressure response.

A New Method in Stroboscopy*)

by

Ole Nielsen

ABSTRACT

The design of a new stroboscopic phase-locking system is briefly described. The system allows any acoustical-mechanical system to be studied at any desired phase or in slow motion. Automatic tracking with frequncy is possible keeping phase conditions constant. Typical applications of the system in medico-acoustical investigations as well as dynamical mechanics research are presented and discussed.

SOMMAIRE

On décrit brièvement la conception d'un nouveau système stroboscopique à verrouillage de phase. Le système permet d'étudier tout système mécano-acoustique à n'importe quel décalage de phase ou en mouvement ralenti. Un asservissement automatique de la fréquence est possible qui maintient constantes les conditions de phase. Des applications typiques du système dans les investigations médico-acoustiques comme en recherche en mécanique dynamiue sont présentés et discutées.

ZUSAMMENFASSUNG

Der Entwurf eines neuen Stroboskopes wird kurz vorgestellt, mit de meine bestimmte Bewegungsphase periodischer Abläufe stroboskopisch festgehalten wird. Jedes akustisch-mechanische System kann in jeder gewünschten Phasenlage oder in Zeitlupe betrachtet werden. Automatischer Gleichlauf mit der Meßfrequenz ermöglicht konstante Phasenbedingungen. Typische Anwendungen des Systems in medizinisch-akustischen als auch in dynamisch-mechanischen Untersuchungen werden aufgezeigt und diskutiert.

Introduction

A new working principle for stroboscopes, designed mainly around pulse circuits and offering exceptionally high long term stability, has been developed and incorporated into the new B & K Stroboscope. This design provides two *new essential features in stroboscopy,* namely:

1. The instrument will accept *any periodic waveform* at its input.
2. The phase in which the moving object is observed *remains the same,* once set, *even if the input signal varies.*

In this way a very useful tool for the visual inspection of vibrating and rotating phenomena is obtained.

Description of the Instrument

To allow for the good features mentioned in the introduction the input signal to the Stroboscope is converted into rectangular pulses which again controls a signal sawtooth generator.

The purpose of the sawtooth generator is to obtain a linear phase deviation range between 0° and 360°, and this can be achieved simply by comparing the instantamous values of the sawtooth waveform with a variable DC. Each time the two voltages are of equal magnitude a pulse releases a flash, see

*) Paper presented at the 6th International Congress on Acoustics, Tokyo, Japan 21 – 28 August 1968.

Fig. 1. If the DC voltage is replaced by a sawtooth wave, a slow-motion effect takes place. The degree of slow-motion is directly controlled by this sawtooth wave, and varies between 0.5 Hz and 2 Hz. With vibrating objects it seems as though 2 Hz slow-motion frequency is the most convenient, whereas 0.5 Hz is the better frequency for rotating objects.

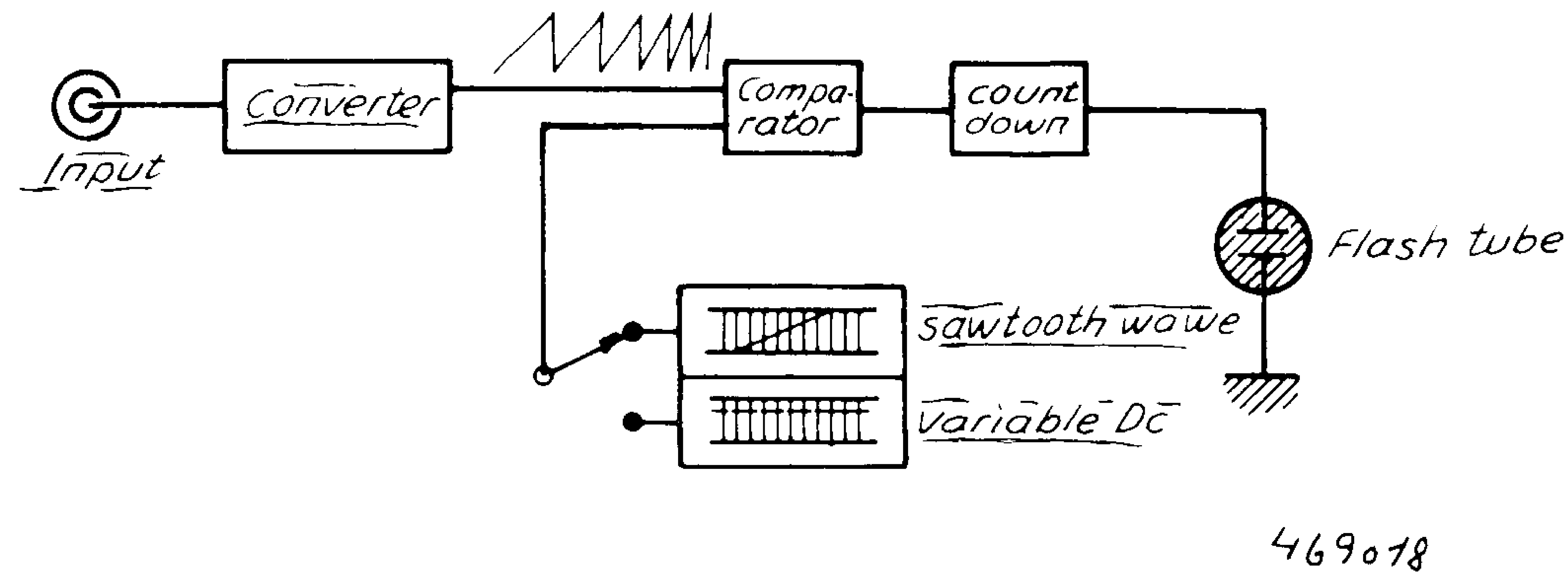

Fig. 1. Simplified block diagram of the stroboscope.

When increasing the signal frequency a built-in count-down circuit will automatically count-down the flashing rate in such a way, that the phase in which the moving objects is observed will not be disturbed. Thereby the mean light output appears as practically constant in the entire frequency spectrum. The formula governing the flash rate is:

$$\text{Flash frequency: } f_f = \frac{f_s \pm \Delta f}{n}$$

f_s = signal frequency, Δf = slow-motion frequency, n = positive integral number.

Some Applications of the Stroboscope

Calibration. During the calibration of accelerometers it is very important to be able to determine the vibration amplitude exactly. With the stroboscope the maximum displacement of the object can be measured simply by turning the phase-deviation knob 180° from one peak of excursion to the other. If extremely high accuracy is required the external synchronizing signal can be rectified by means of a full wave rectifier thus obtaining an input signal of twice the frequency. The stroboscope lamp thereby flashes with the double frequency.

A point on the test specimen illuminated by the flash, will now appear as two static points the distance between which can be adjusted by turning the phase deviation knob so as to indicate the maximum displacement, see Fig. 2. By means of a microscope it will be possible to measure displacements of the order of micrometers. The use of a flashing light source gives a better determination of the peak amplitude than can be obtained with a constant light source.

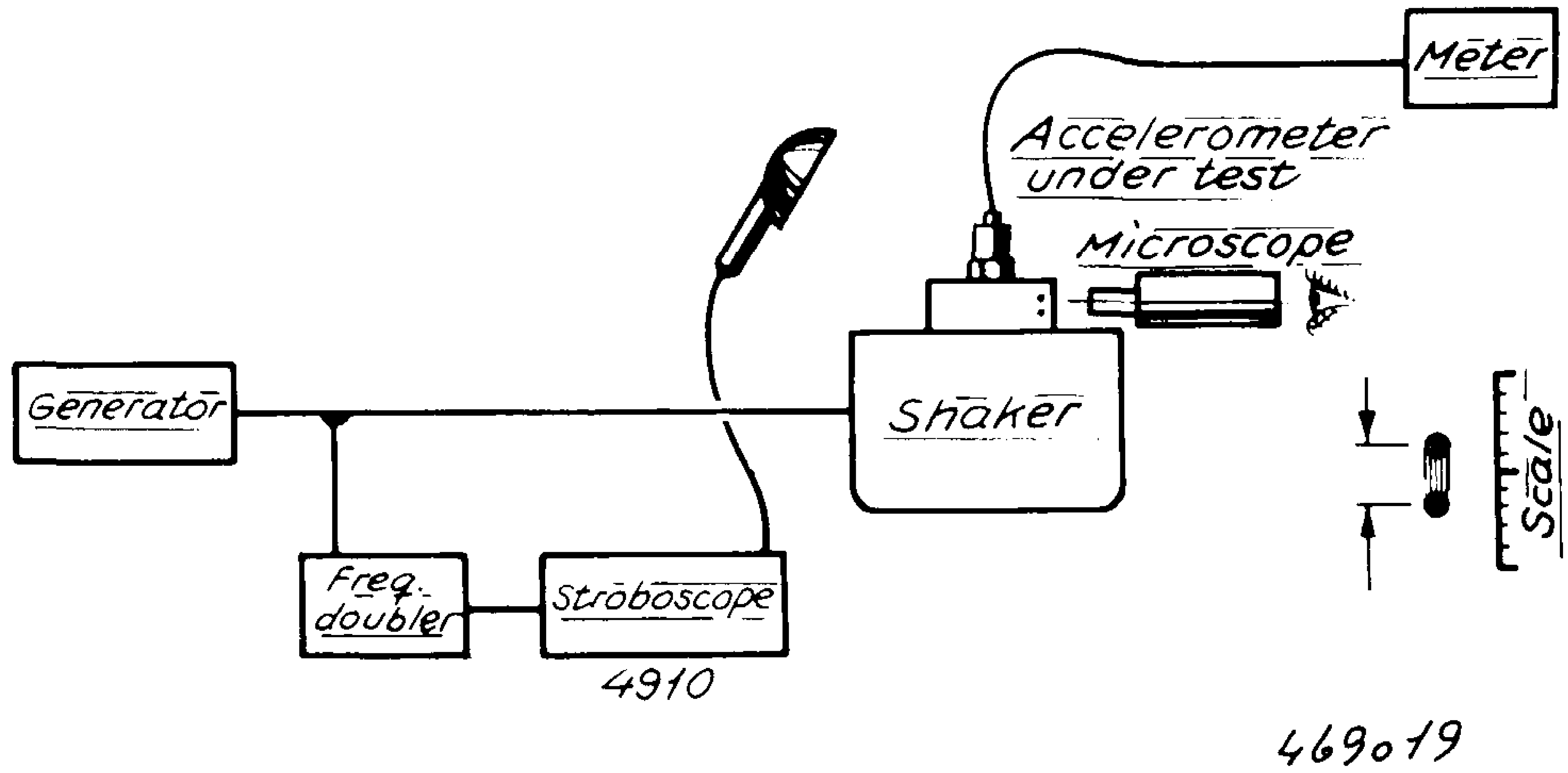

Fig. 2. Calibration set-up.

Product Control. In the checking or adjustment of *mechanical relays* the stroboscope has been used. When the repetition rate of the relay switching is adequately high (> 15 Hz) the movement can be controlled in each phase, e.g. if two contacts are activated from one solenoid, cooperation between contacts can be visually inspected and adjusted. Furthermore, the proper operation of the spring can be checked in order to avoid contact chatter.
Running gearwheels may be inspected by means of the stroboscope detecting insufficient mesh due to wrong shape of the cogs. To do so a magnetic transducer is placed close to the gearwheel and the signal is fed to the stroboscope.

Research
To study phenomena around the *building of drops* in fluids the set-up, shown in Fig. 3, can be advantageously used. When the drops pass the photosensitive device with regular intervals the phenomenon can easily be examined.

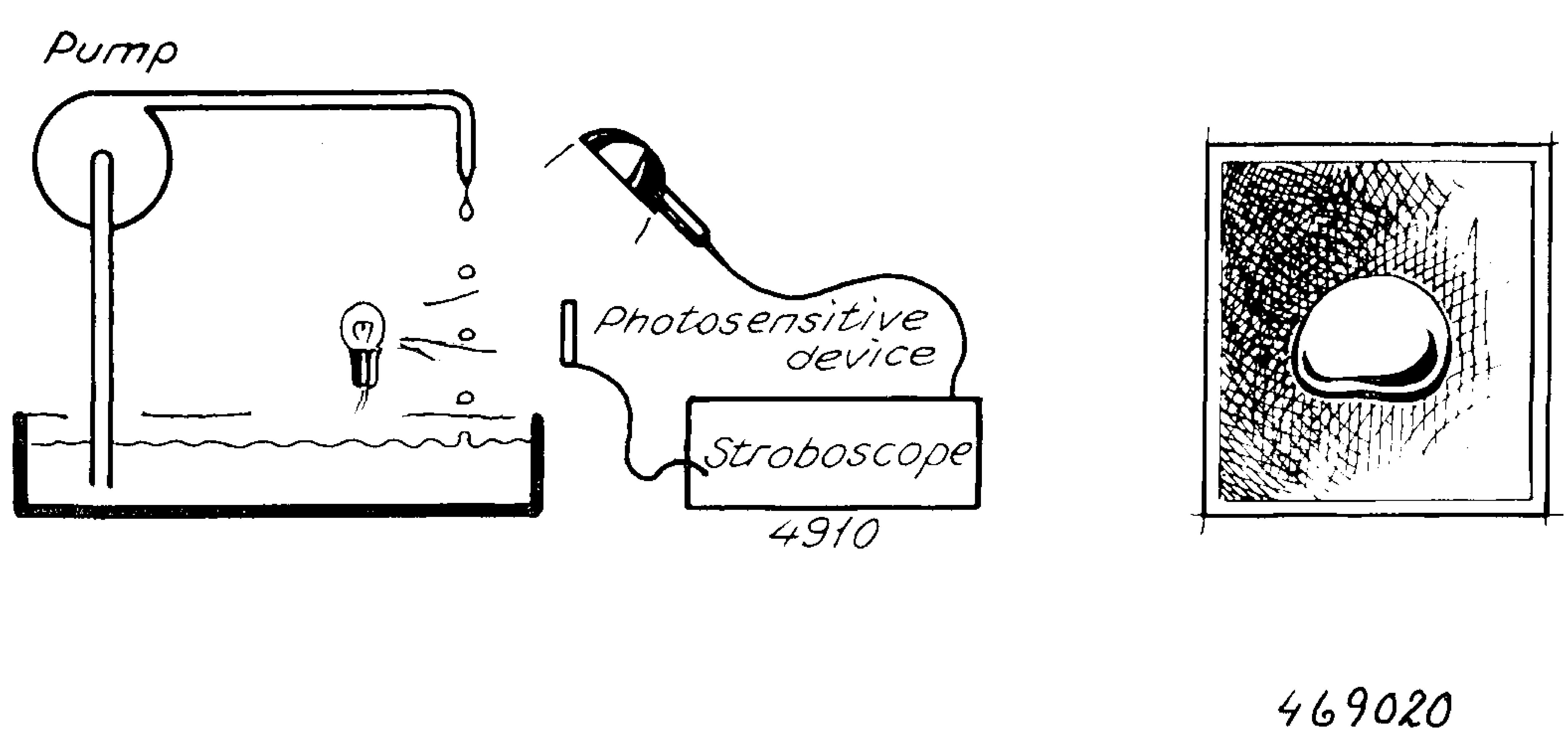

Fig. 3. Set-up for study of drops.

The electrical signal from this set-up is especially suitable for the stroboscope because of its impulsive character. The input circuit is so designed that the triggering level can be varied around the zero crossing. In the case of uncertain triggering it may therefore help to adjust the triggering level.
Another interesting application of stroboscopy is the examination of musical instruments. The *dynamic behaviour of a pianostring* has, for instance, been watched when struck by the hammer, and so has the following decay of the string vibration. An elliptical transversal movement of the string was observed, Fig. 4.

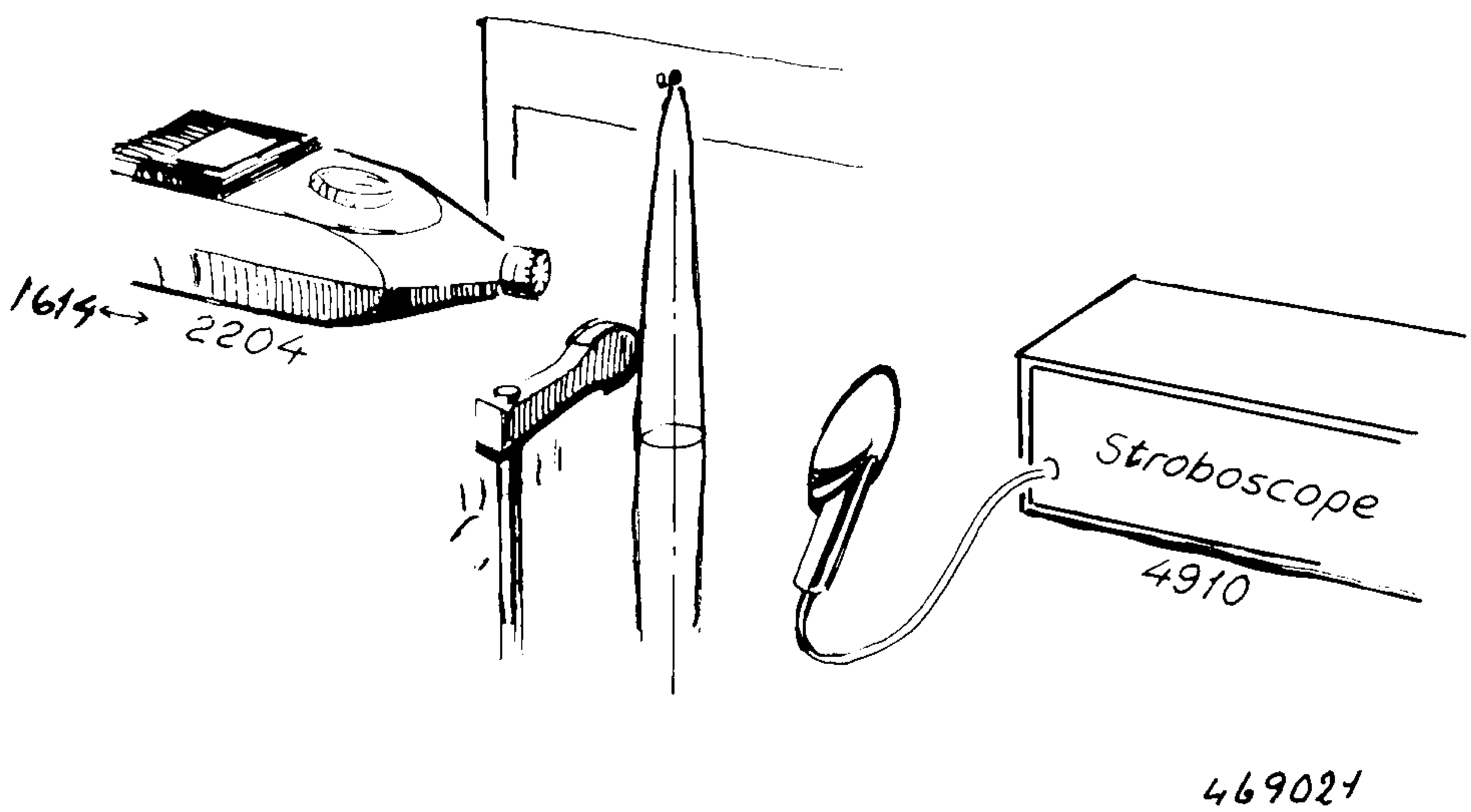

Fig. 4. Set-up for observing pianostring-movement.

To avoid disturbances at the input of the stroboscope caused by harmonics (or back-ground noise) a one octave filter was connected between the microphone and the stroboscope.
Had a filter not been used the zero crossings originating from the second harmonic would in this case have caused false triggering.
The movement of a violinstring and *the surface of the violins body* can be examined in the way above. When the amplitudes are small, however, a microscope must be used to allow the movements to be studied.
Experiments with Organ-pipes in which smoke is mixed with the emerging air can give an indication of edge-tone effects. Here again the sound synchronizes the stroboscope triggering via a microphone and a filter.
In the Field of Medicine some applications of stroboscopy worth mentioning are *Laryngscopy* and *studies of the operation of the internal parts of the human ear*. As an actual example a complex equipment currently used on a hospital in Copenhagen is shown in Fig. 5.
It consists of a microphone, a frequency analyzer and a stroboscope. For each patient the professor determine the typical speech spectrum, both before and after the treatment. Furthermore, it is of very great interest to combine these frequency determinations with the visual observation of the functioning of the

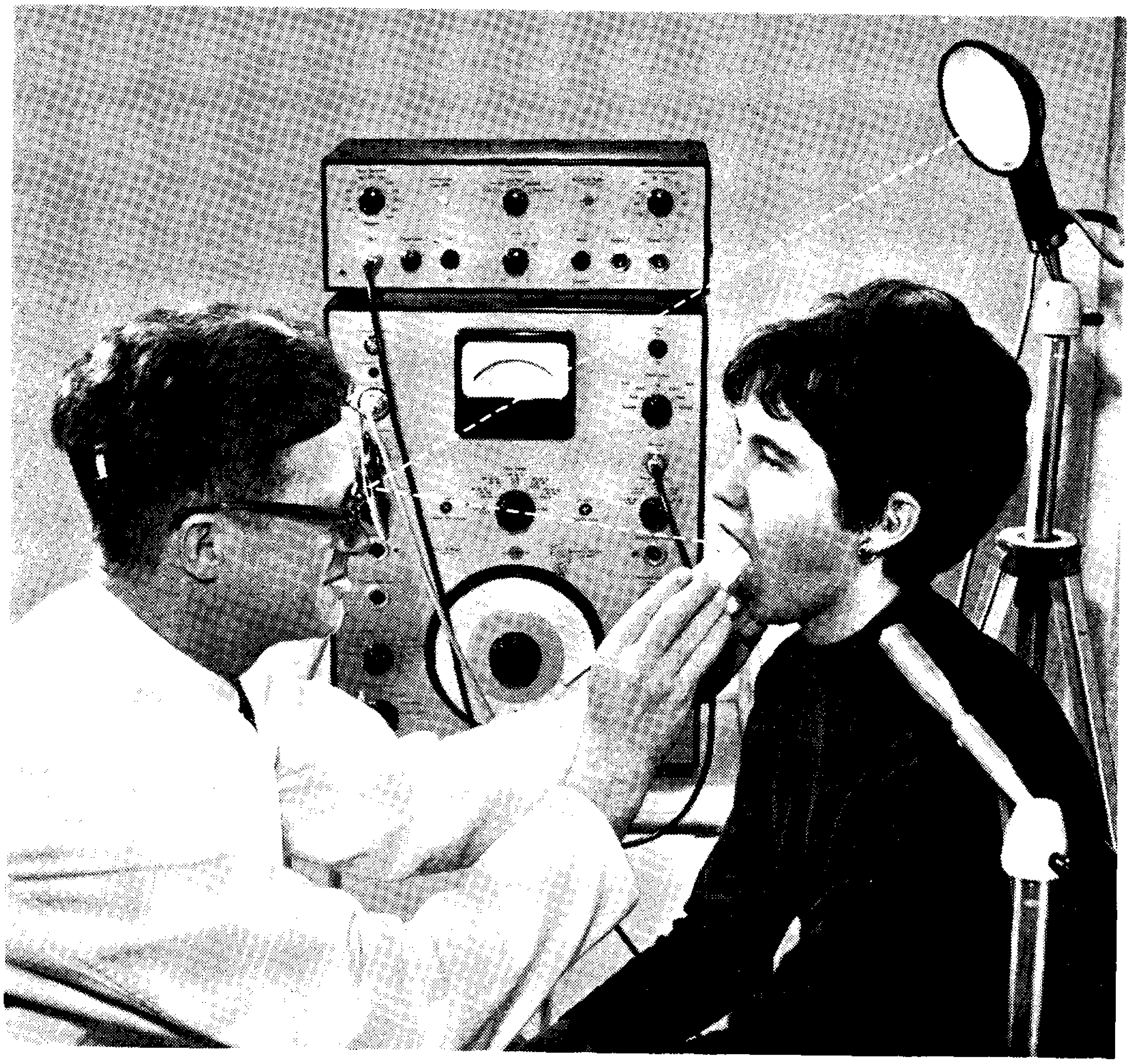

Fig. 5. Set-up for examination of Larynx.

vocal cords. For men the relevant frequency range is 63–200 cps and for women 200–630 cps. The most important diagnostic possibilities offered by the above arrangement are:

1 Early determination of Larynx-cancer.

2. Detection of tumours on the vocal cords.

3. Description of the laryngeal function after lesions of the recurrent nerve.

Also in the *zoological field* stroboscopy may be used to study wing movements and other periodic phenomena like the production of sound by grasshoppers etc.

Conclusion

There are numerous other problems *in research* as well as *in education* where the principles of stroboscopy can be advantageously applied. It has unfortunately only been possible within the frame of this paper to mention but a few of the possibilities that this technique offers, but it is deemed that the new principles in the design of stroboscopes, will make stroboscopy a very important tool in the science and technology of the future.

News from the Factory

Heterodyne Slave Filter Type 2020

Designed to be used as external filter for the B & K measuring amplifiers the Heterodyne Slave Filter gives *new possibilities for highly selective measurements with automatic tuning in the audio-frequency range.* The filter is tuned from the high frequency signals of the B & K generators 1022 and 1024 and will instantly follow their low frequency output signal over the whole frequency range. The bandwidth has been variable 3.16, 10, 31.6 and 100 Hz for optimum speed of analysis. Automatic compensation for bandwidth according to $1/\sqrt{B}$ is included.

The Slave Filter will be found extremely useful in electro-acoustical measurements which are difficult because of background noise, such as measurements in anechoic chambers with insufficient sound insulation, measurement of low level signals, and in production line measurements where adequate isolation is not available.

A *rejection output* is available for *automatic measurement of harmonic distortion.*

The *phase difference* between two slave filters is better than 1°, making the filters ideal for mechanical impedance measurements or otherwise where the phases of two signals from the same source are compared. A 90° phase shift position facilitates measurements of co- and quad-spectra in *cross-spectrum* analysis.

The Slave Filter is also used with advantage for automatic selection of the fundamental *in feedback loops* in frequency response measurements or vibration testing.

Photo of the Heterodyne Slave Filter Type 2020.

Brüel & Kjær

ADR.: BRÜEL & KJÆR
NÆRUM - DENMARK

TELEPHONE: 800500
BRUKJA, Copenhagen

TELEX 5316

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