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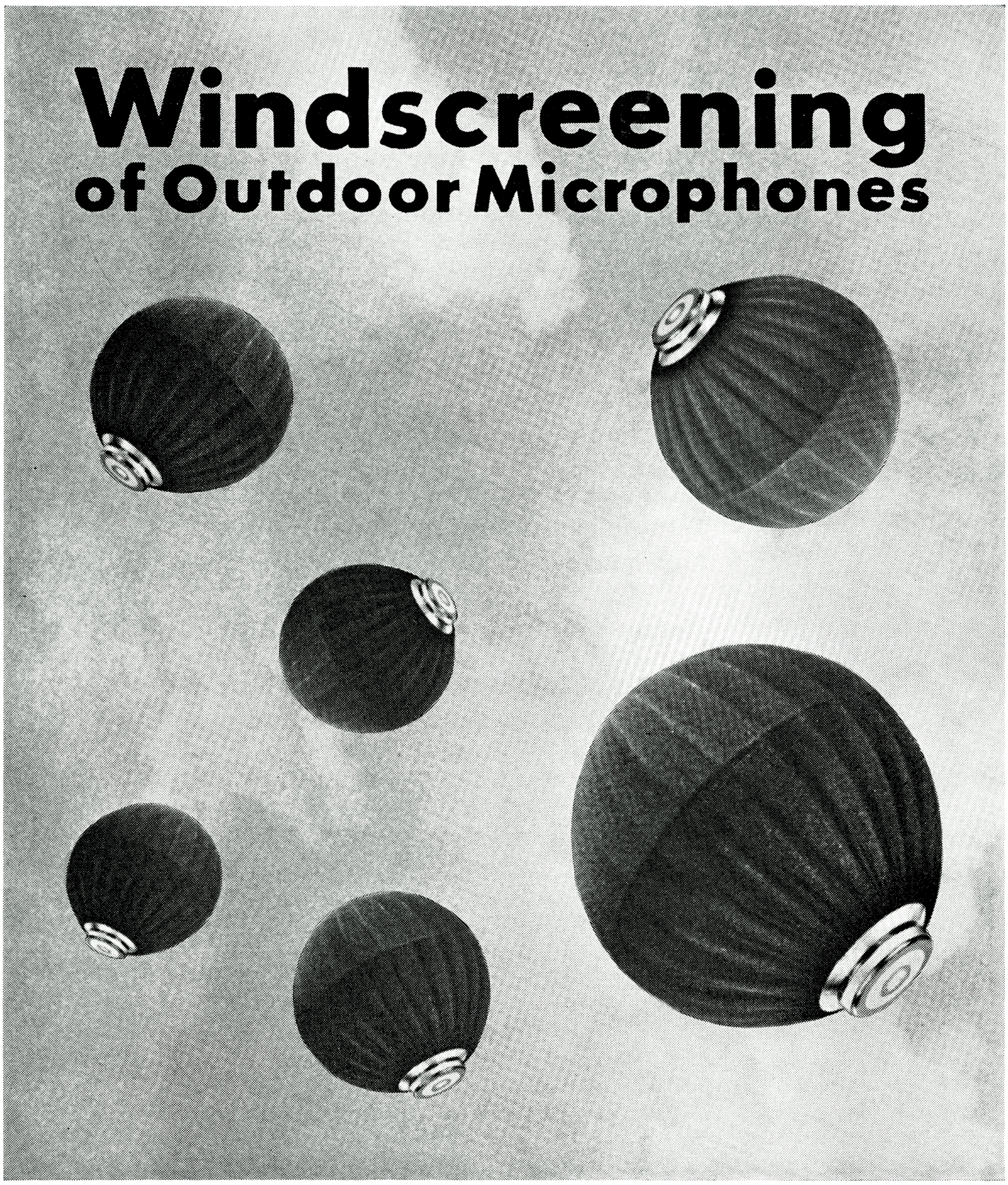
# Technical Review

To Advance Techniques in Acoustical, Electrical, and Mechanical Measurement




No. 1 1966

## PREVIOUSLY ISSUED NUMBERS OF BRÜEL & KJÆR TECHNICAL REVIEW

# TECHNICAL REVIEW

No. 1 – 1966

# Contents



K. LARSEN & SØN - LYNGBY

# Windscreening of Outdoor Microphones.*)

By

*Frede Skøde, Electronic Engineer,* A/S Brüel & Kjær

**ABSTRACT**

Noise level as a function of wind-screen diameter and type of covering material is reported. Various effects of windscreens are described.

**SOMMAIRE**

On rapporte quel est le niveau de bruit en fonction du diamètre de l'écran anti-vent et du matériau de couverture.
Divers effets des écrans anti-vent sont décrits.

**ZUSAMMENFASSUNG**

Der Geräuschpegel wird als Funktion des Schirmdurchmessers und der Art des Umkleidungsmaterials angegeben. Verschiedene Wirkungen der Windschirme werden beschrieben.

Firstly some considerations concerning the construction of wind screens for microphones are made. Secondly some measurements are made with different constant wind velocities and finally the measurements on a microphone placed outdoors and fitted with windscreens of different dimensions are discussed.

Where a microphone is exposed to wind, a noise will be generated in consequence of the variation of air pressure on the membrane. These variations of air pressure are caused by several things. Firstly: That the wind changes its velocity. Secondly: The turbulence generated around the microphone when it is placed in the wind. Thirdly: The interference between the variations in air pressure caused by wind velocity changes and turbulence.

The influence of the wind velocity variations can be reduced by mounting a windscreen made of cloth, or the like, around the microphone. The reason for this is that cloth presents a great resistance to the wind while the acoustic impedance of it is small. The windscreen must necessarily be bigger than the microphone and will therefore create more violent turbulences than the microphone itself, but because the turbulences are moved away from the membrane the noise will be decreased. Turbulence can be reduced by making the microphone and the windscreen respectively, as streamlined as possible. Thereby the interference will be further decreased, because the causes of it are reduced.

When working with constant windspeeds and fixed directions, as, for instance in windtunnels, the only problem is to reduce the turbulence as much as possible, for this purpose a nose cone having a parabolic surface turned into the wind is most suitable. In surroundings with random wind directions, in

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*( Paper given at the 5th International Congress on Acoustics, Liege, Belgium, 7—14 September 1965.

other words normal outdoor surroundings, you can no longer consider the wind a constant, as it will come in gusts. To reduce noise generated by this kind of wind the most suitable device is a screen having an aerodynamic shape and covered with a suitable cloth. To obtain optimum results from this screen you must be sure that it is always turned toward the wind direction. This creates no problem when dealing with measurements taken within a short time, where the wind direction can be considered as a constant.

In permanent set-ups you could consider mounting the screen on a ball bearing so that it would automatically turn toward the right direction. This has the disadvantage that the sensitivity of the microphone system will not only depend upon frequency, but also on the angle of incidence, and when the screen is unsymmetrical to a vertical axis the sensitivity will depend on the angle between the direction of the wind and the direction of the sound.

A good solution is to make the screen spherical, for instance a spherical wire frame covered with cloth. In order to avoid standing waves the clamping arrangement must have a surface as small as possible and as far as possible it must be avoided that the surface of the clamp is perpendicular to the radius of the sphere. Furthermore it must be in a position as far from the membrane as possible.

For measurements with constant wind velocity a Brüel & Kjær 1″ microphone Type 4131 and a cathode follower Type 2613 were used. The windscreen used was the Brüel & Kjær Type UA 0082, which consists of a spherical wire frame with a radius of 120 mm approximately and covered with two layers of nylon crepe 2/60 with 25 meshes per inch approximately. See figs. 1a, 1b and 2.

Microphone and cathode follower were placed on a microphone stand which in turn was mounted approximately one and a half meters above the roof of a car. The stand was constructed in a way which introduced as little turbulence as possible. Batteries were used for the power supply to the cathode follower, and a Sound Level Meter Type 2203 together with a Band-Pass Filter-Set Type 1612 were used as indicating instrument. The complete set-up was calibrated by means of the Pistonphone Type 4220.

In calm weather this set-up was driven at speeds from 30 to 120 km/h and the noise of the wind was measured as a function of speed over the frequency range 20 Hz to 20 kHz.

Fig. 3 curve A shows the noise with the wind direction parallel to the membrane while curve B shows the noise with the wind direction perpendicular to the membrane. It will be seen that the noise has been reduced 6—7 dB approximately from A to B. This is because the protecting grid itself acts as a wind screen, when the wind direction is perpendicular to the membrane. Measurements made by Dr. Brüel*) shows that if the measurements had been made without the protecting grid mounted, curve B would have been 1—2 dB higher than curve A and the direction of the wind relative to the membrane

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*) Ref: Per V. Brüel: Technical Review No. 2/1960.

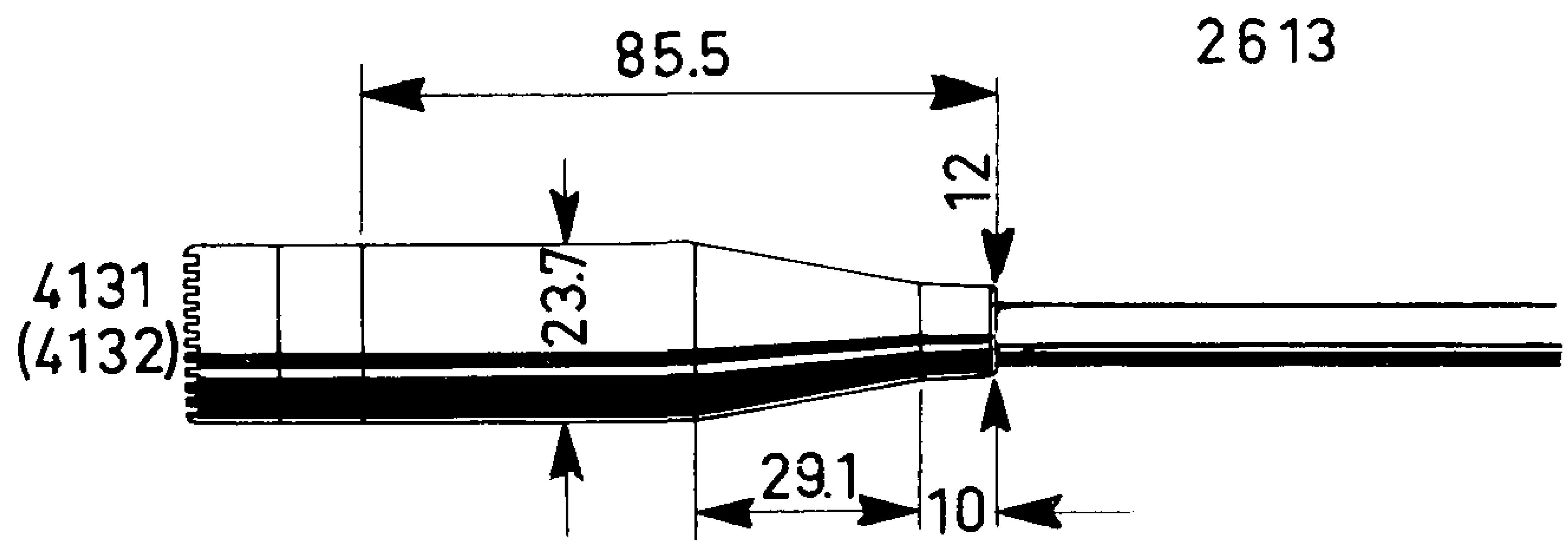


*Fig. 1a. Cathode Follower 2613 with 1" Microphone 4131.*

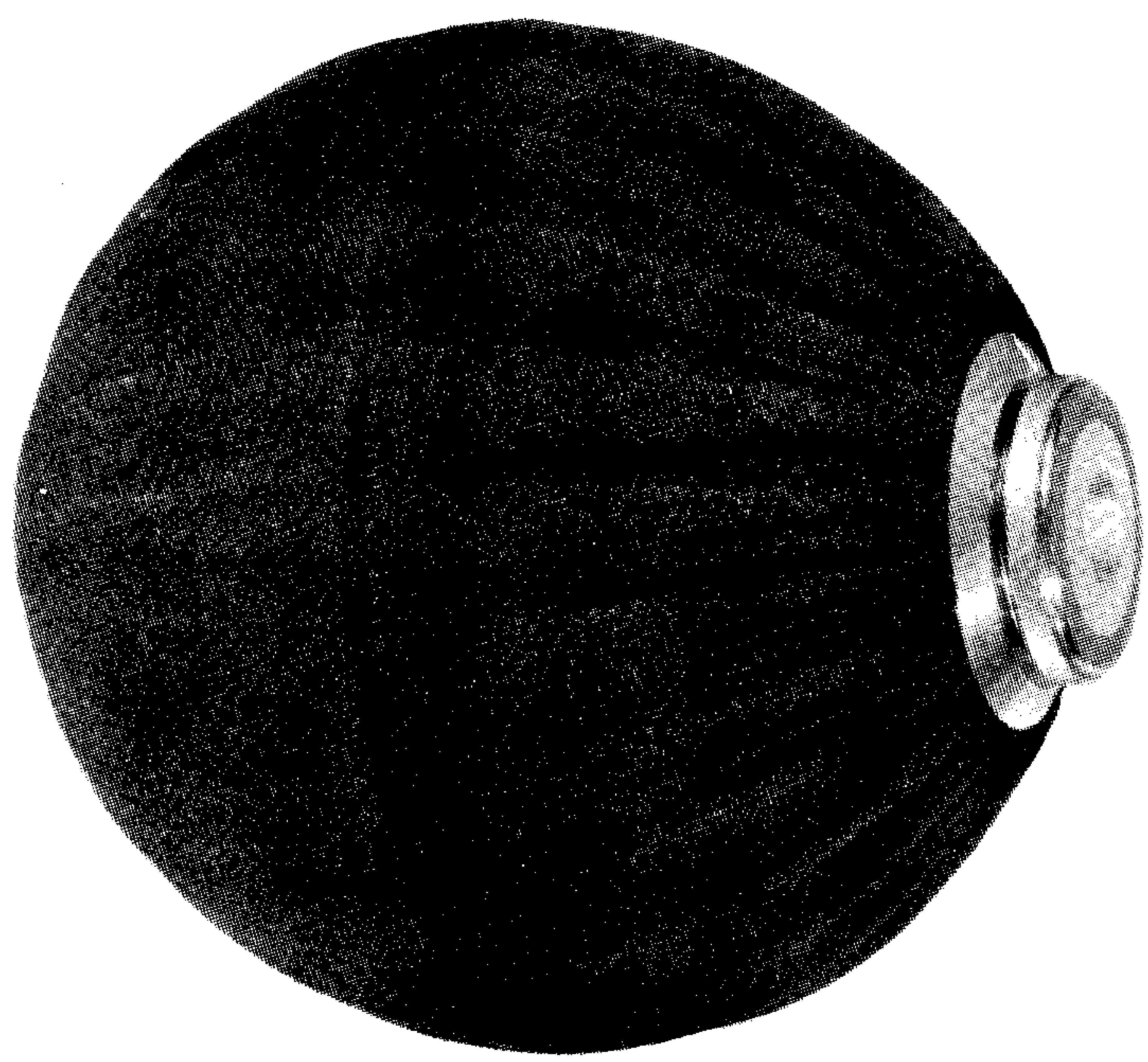

*Fig. 1b. Windscreen UA 0082.*

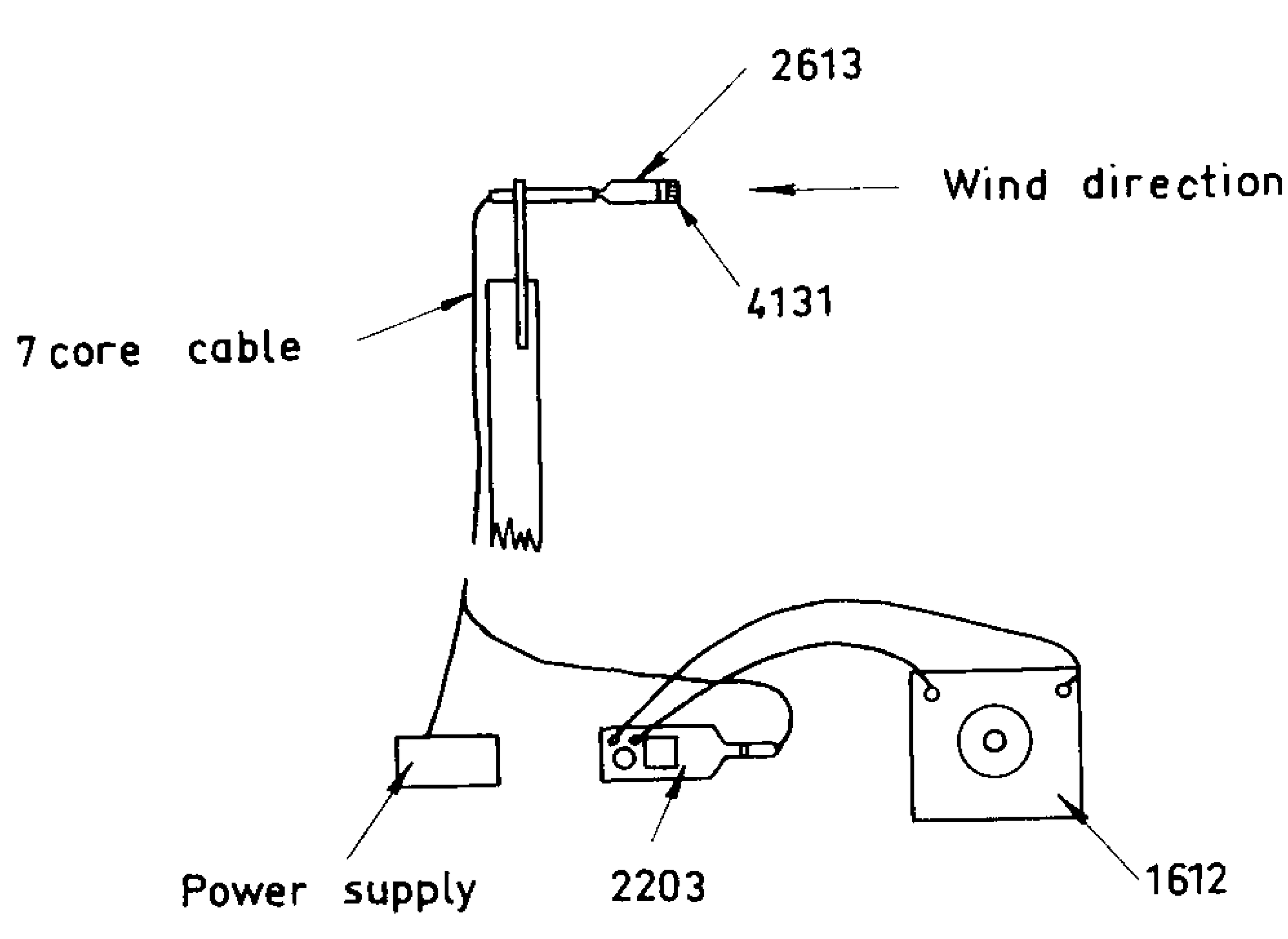


*Fig. 2. Measuring set-up for measuring noise at constant wind speed.*

would have had no important influence. The influence of the windscreen appears in the curves C and D. Here it is noticed that the reduction of noise is considerably greater when the wind direction is parallel to the membrane than when it is perpendicular to it. From this it can be seen that it is now the windscreen itself, which has reduced the noise of the wind. The reduction brought about by the protecting grid had influence only at higher speeds, i.e. above 60 km/h. An analysis of the noise was made by means of 1/3 octave filters (Fig. 4 and 5).

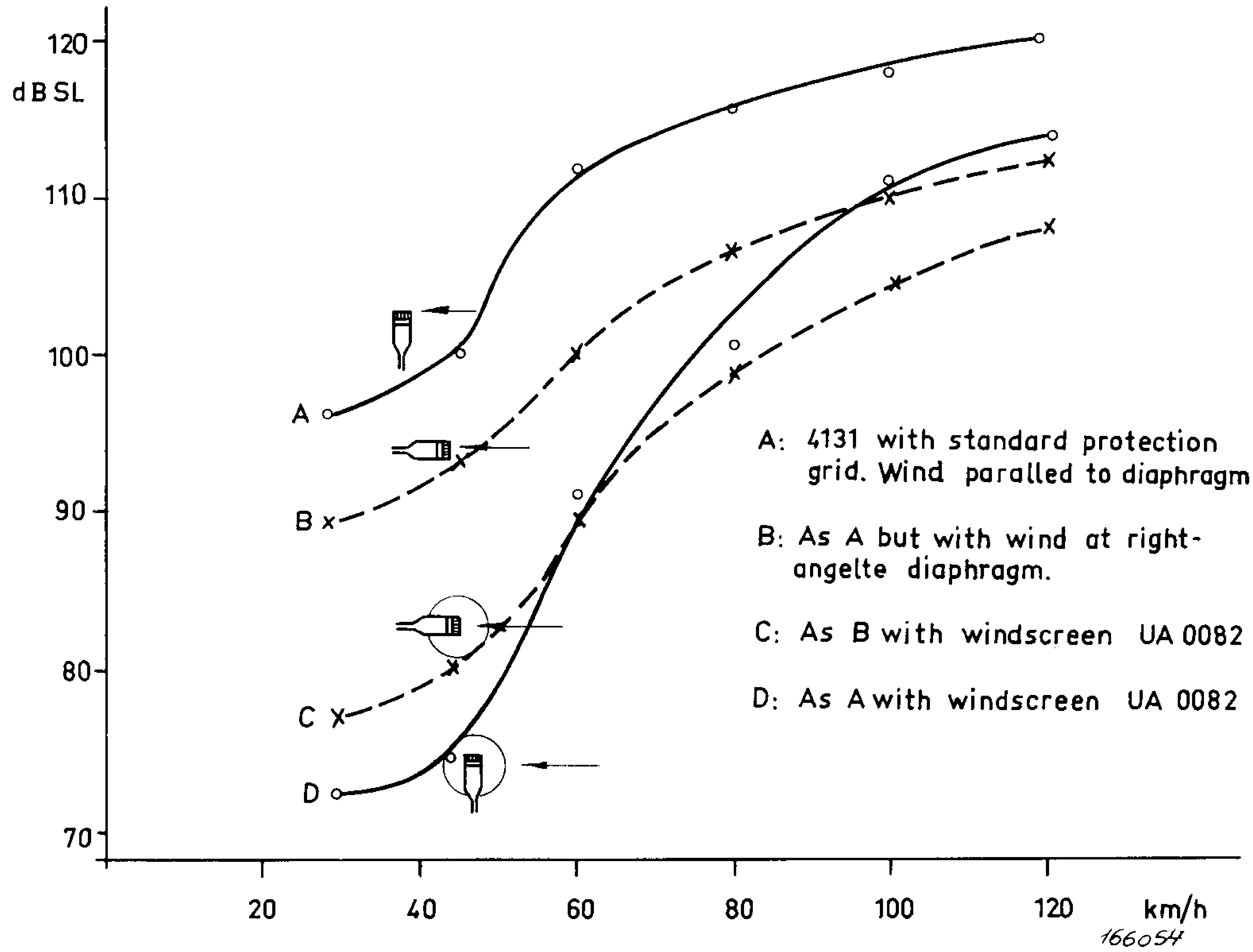


*Fig. 3. Wind noise as a function of wind speed in the range 20 Hz—20 kHz.*

From this it appears that the greater part of the noise is at the low frequencies and that the wind velocity has influence on which frequencies are attenuated most, so that the greater the velocity the higher the frequency where attenuation is most effective. This is the reason why, in measurements over the entire audible frequency range, noise reduction will be greatest at the low wind velocities, as it is here that the low frequencies are attenuated most. If the measurements were made for instance with a weighting network, curve A, where the low frequencies are filtered out, the total reduction would be approximately independent of the velocity.

Measurements, which perhaps in most cases correspond better with practice, were made on the roof of a building, where the microphone was placed together with an anemometer so that the noise level corresponding to a given wind velocity could be measured. The anemometer was so constructed that it indicated the mean value of the wind velocity.

In these measurements it is shown that the normal varying wind velocity generates a noise which has a higher level than that of a constant wind velocity. It is found that even if the wind velocity decreases from a certain level, the noise will increase until the wind velocity reaches a constant lower level.

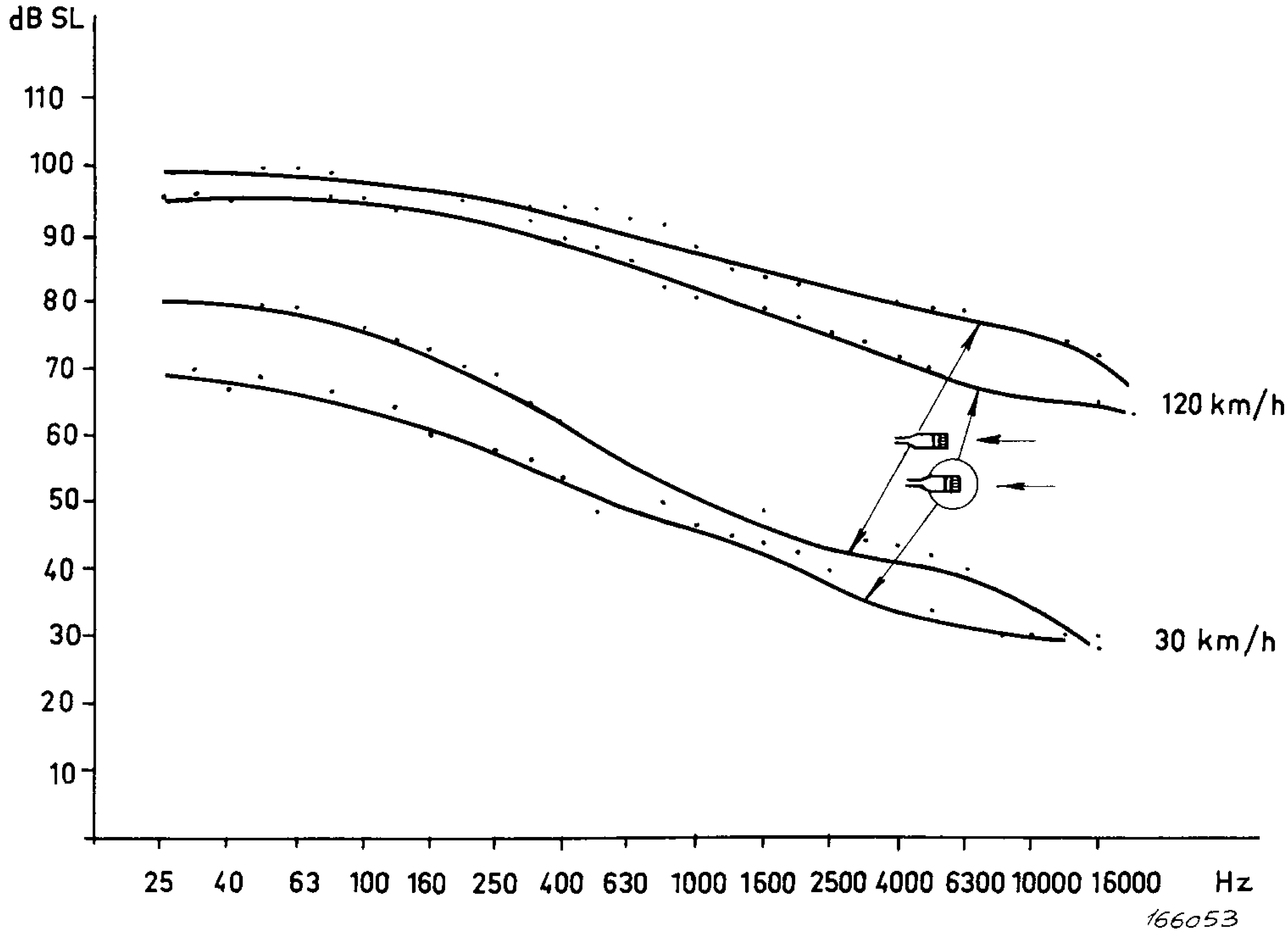


*Fig. 4. Wind noise as a function of frequency, measured in 1/3 octaves, with the wind direction at right angles to the membrane.*

With this set-up the microphone was mounted with windscreens of different dimensions and it was seen that if a windscreen of the same construction as that of the UA 0082 but with a diameter twice as big was mounted, the noise level was decreased approximately 6 dB more, while a screen that was four times as big as the UA 0082 gave a reduction of the noise level of approximately 8 dB (see Fig. 6).

In this case the optimum screen size in relation to the reduction of the wind noise seems to be a diameter of approximately 24 cm. Measurements on changes in the sensitivity of the microphone when it is mounted with a windscreen proves that for frequencies up to 2 kHz, no difference can be measured. At higher frequencies the sensitivity will vary around the sensitivity of the microphone without the windscreen mounted (see Fig. 7).

This proves that it is not because the sound pressure is damped by passing the cover of the windscreen, but rather because of defraction around the screen and standing waves inside it.

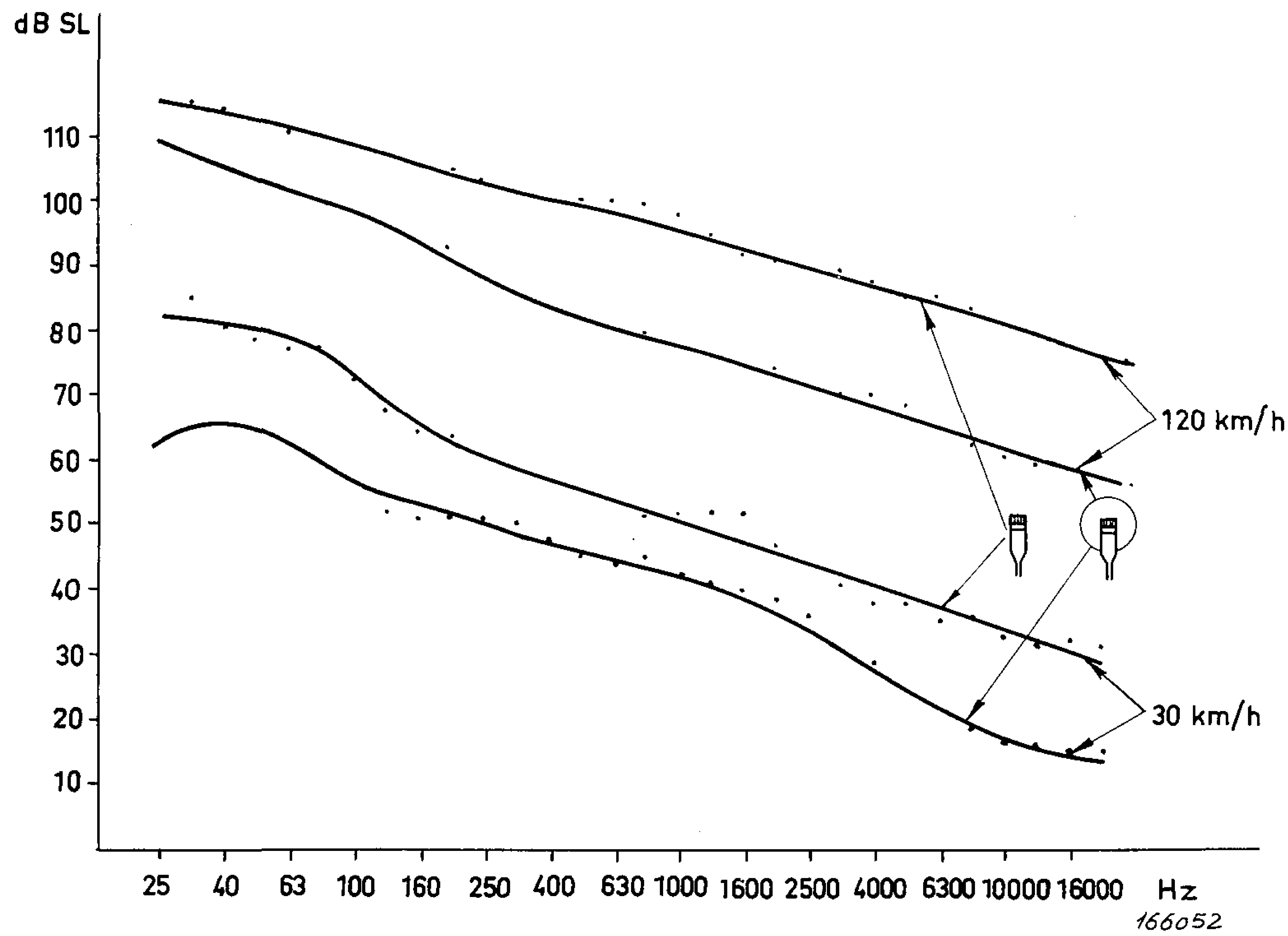


*Fig. 5. As 4, but with the wind direction parallel to the membrane.*

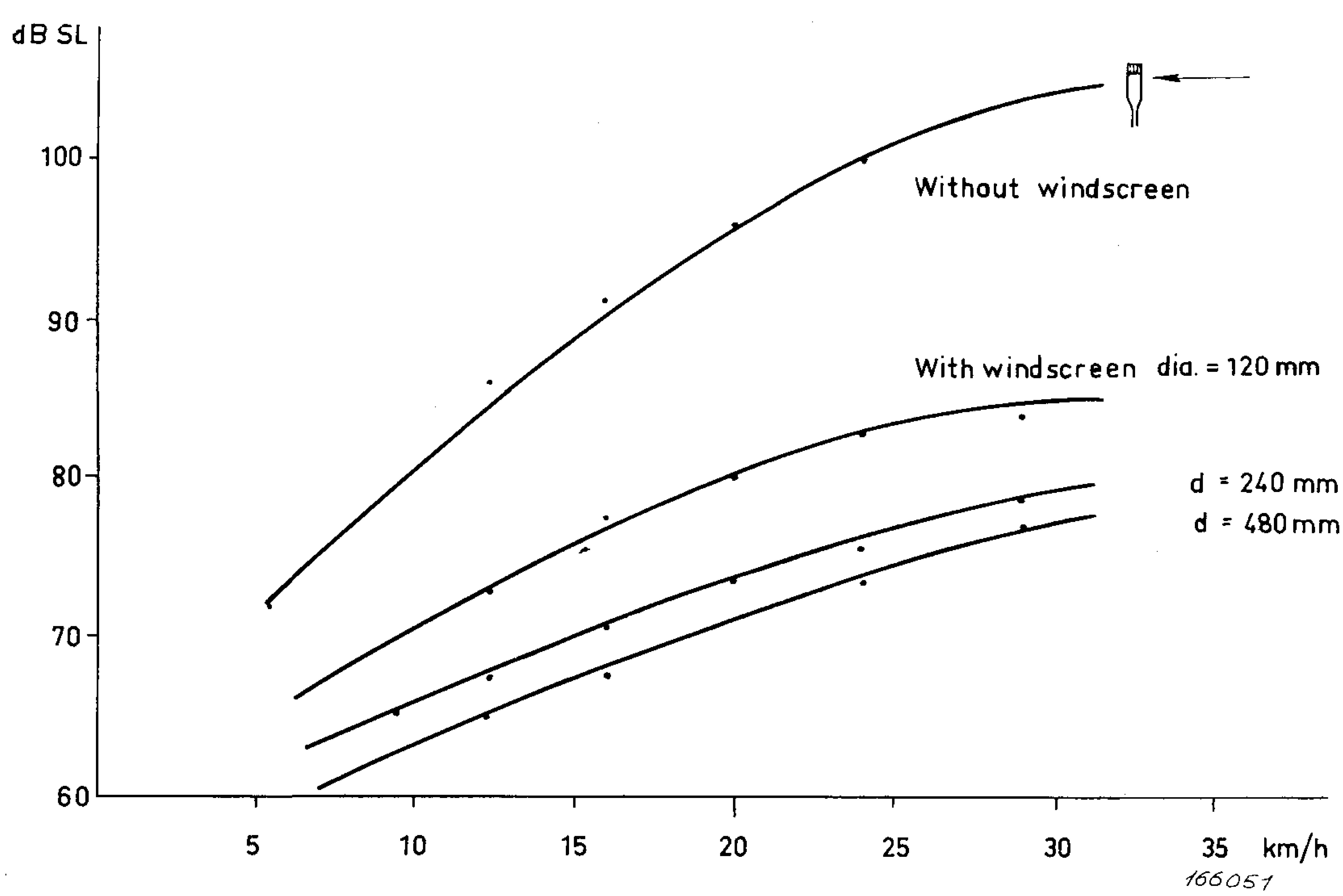


*Fig. 6. Wind noise, measured with different size of windscreens.*

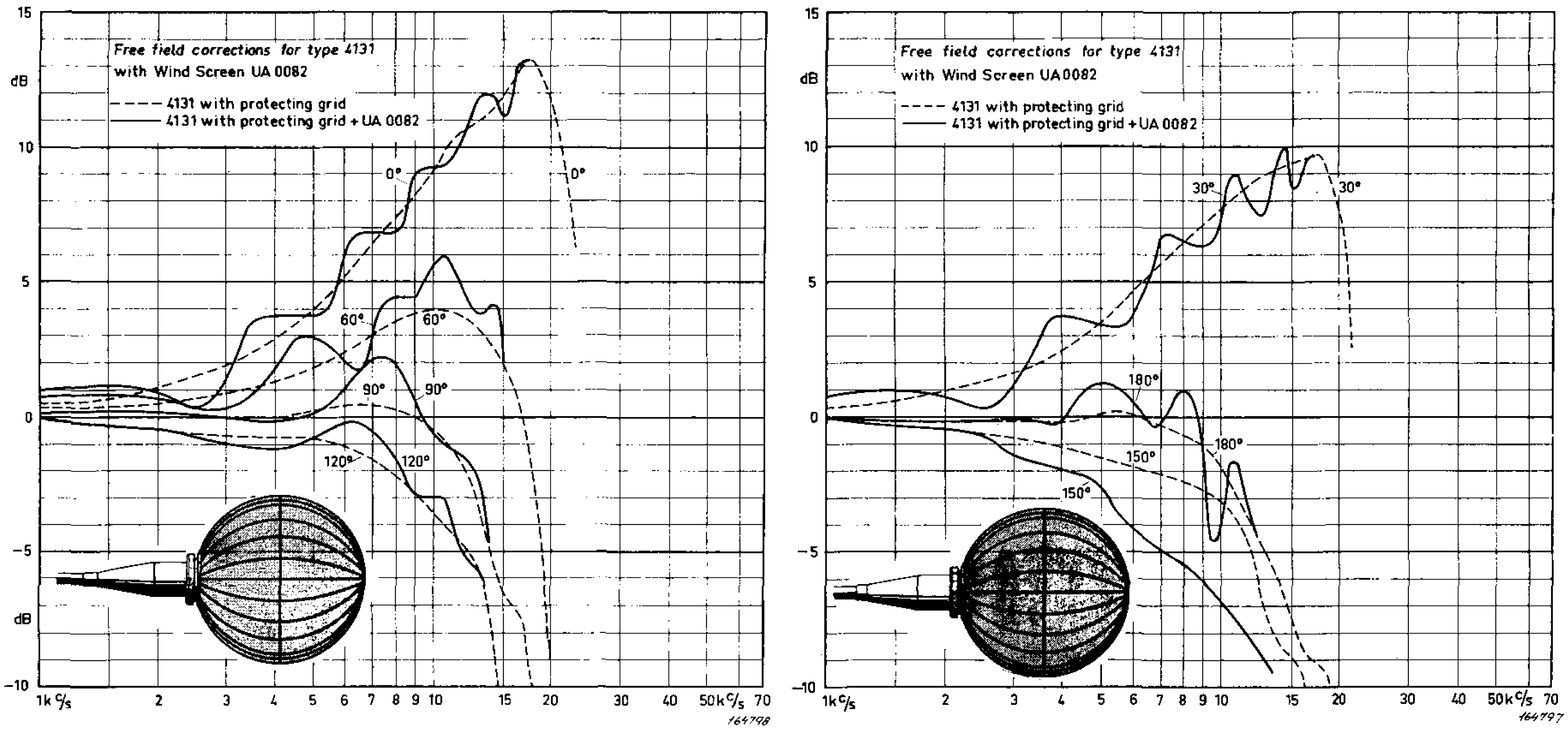


*Fig. 7. Free field correction curves for 4131 with UA 0082.*

The fact mentioned above is an argument for making the windscren unsymmetrical since this to some extent will eliminate the influence of standing waves, but as mentioned before it will be considerably more difficult to achieve a well defined sensitivity.
Considering the size of the windscreen, it must be a compromise between the higher degree of noise reduction obtained when using a large windscreen, and the difficulties and disadvantages in design caused by the dimensions required for the clamping arrangement of a large windscreen, just as in many cases it can be very inconvenient to work with windscreens of large dimensions.

# A New Artificial Mouth.*)

By

*Peter Willhjelm, M. Sc.*, A/S Brüel & Kjær

**ABSTRACT**

The paper gives a brief introduction to the operating principles of a regulated artificial mouth, with the main emphasis on the influence of an obstacle in the near field of the mouth. It will be demonstrated how this influence can be measured, and that it is negligible under normal conditions.

**SOMMAIRE**

L'article donne une brève introduction aux principes de fonctionnement d'une voix artificielle régulée et met en évidence l'influence d'un obstacle dans le champ proche de la bouche. On montre comment cette influence peut être mesurée et l'on démontre qu'elle est négligeable dans les conditions normales.

**ZUSAMMENFASSUNG**

Der Artikel führt kurz in die Arbeitsweise eines geregelten künstlichen Mundes ein und legt besonderen Nachdruck auf den Einfluß eines Hindernisses im Nachfeld des Mundes. Es wird gezeigt, wie dieser Einfluß gemessen werden kann und daß er unter normalen Bedingungen vernachlässigt werden kann.

Investigation of microphones used for the transmission of the human voice is often carried out using subjective test methods, however these methods can be very troublesome and time wasting. This leads to a desire for the substitution of an objective sound source, or more correctly an artificial mouth. If however we adopt this substitution method we must realize that a complete substitute for a human voice is not possible. The objective result one obtains from the artificial mouth has to be investigated every time with respect to the desired accuracy and resemblance to the subjective test.

The value of using objective tests instead of subjective tests is basically the accuracy of reproducibility, and in this way it is very easy to carry out reference measurements.

With the artificial mouth described here, the long stability and reproducibility is only dependent on the regulating microphone which is used within the mouth (as long as you are able to obtain complete regulation). The microphone in this case is a 1/2″ condenser microphone suited for sound pressure measurements and having long time stability, ± 1 %/year.

The following section gives a brief description of the mouth itself and its operating principles. A selected 3″ dia. loudspeaker with paper cone is used as the electro-acoustic transducer. In order to avoid resonances, the mechanical construction of its housing has been made very rigid, and the air volume behind the loudspeaker has been heavily damped. The mouth opening is 20

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*) Paper given at the 5th International Congress on Acoustics, Liege, Belgium, 7—14 September 1965.

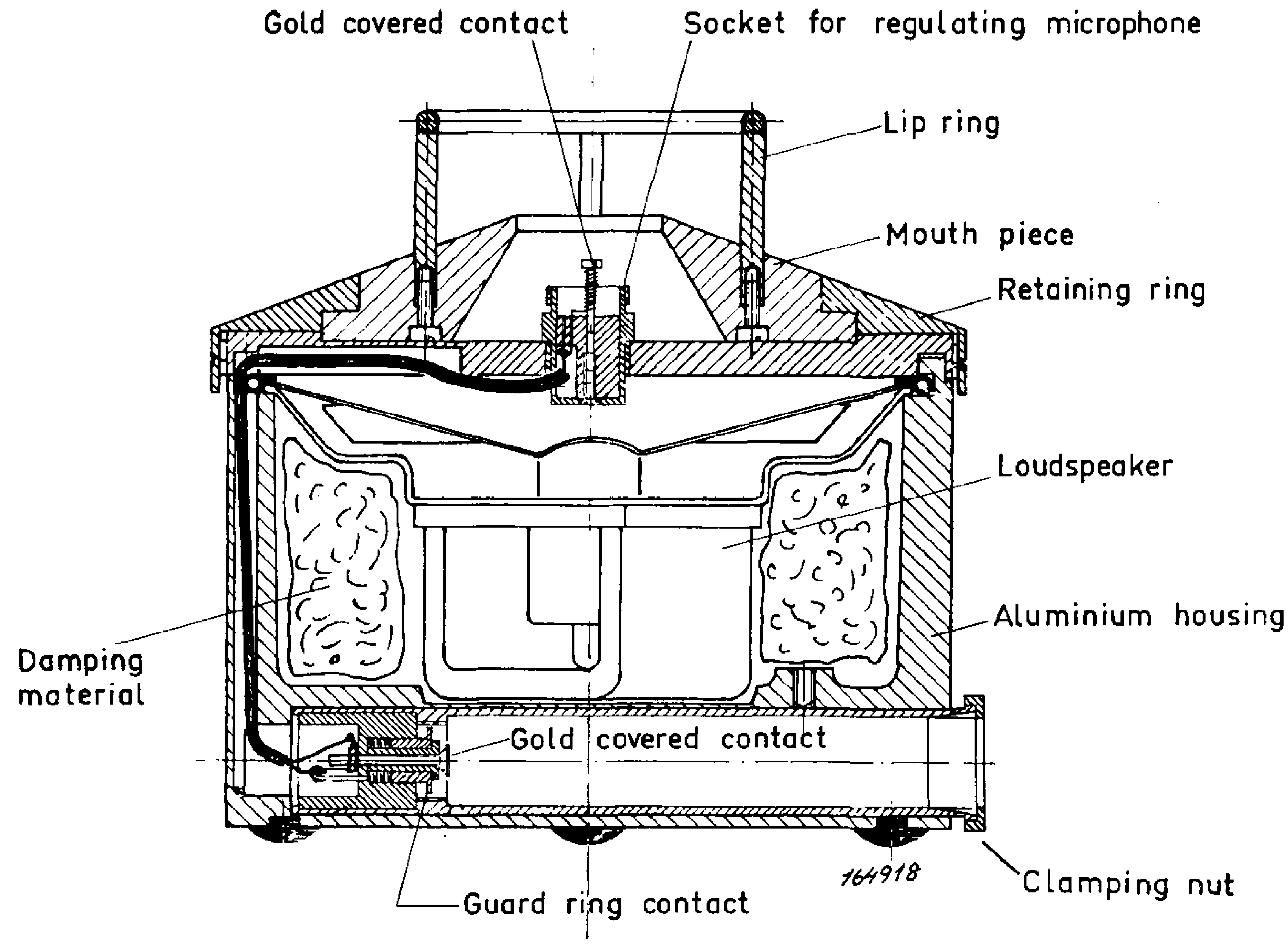



*Fig. 1. Sectional view of the Artificial Mouth.*

mm in diameter, but could very easily be changed as one only requires enough space for the regulating microphone. Positioning of the microphone is such that the membrane is situated in the plane of the mouth opening. Electrical signals from the microphone are fed, via a screened cable, to the cathode follower which is screwed into the base of the housing. It is necessary that the drive oscillator has the facility of a built-in regulating circuit.

The operating principles are as follows. Output power from the oscillator is fed to the loudspeaker which then radiates acoustic energy. The microphone situated in the mouth opening picks up the produced sound pressure and via the cathode follower feeds an amplified electrical signal back to the input

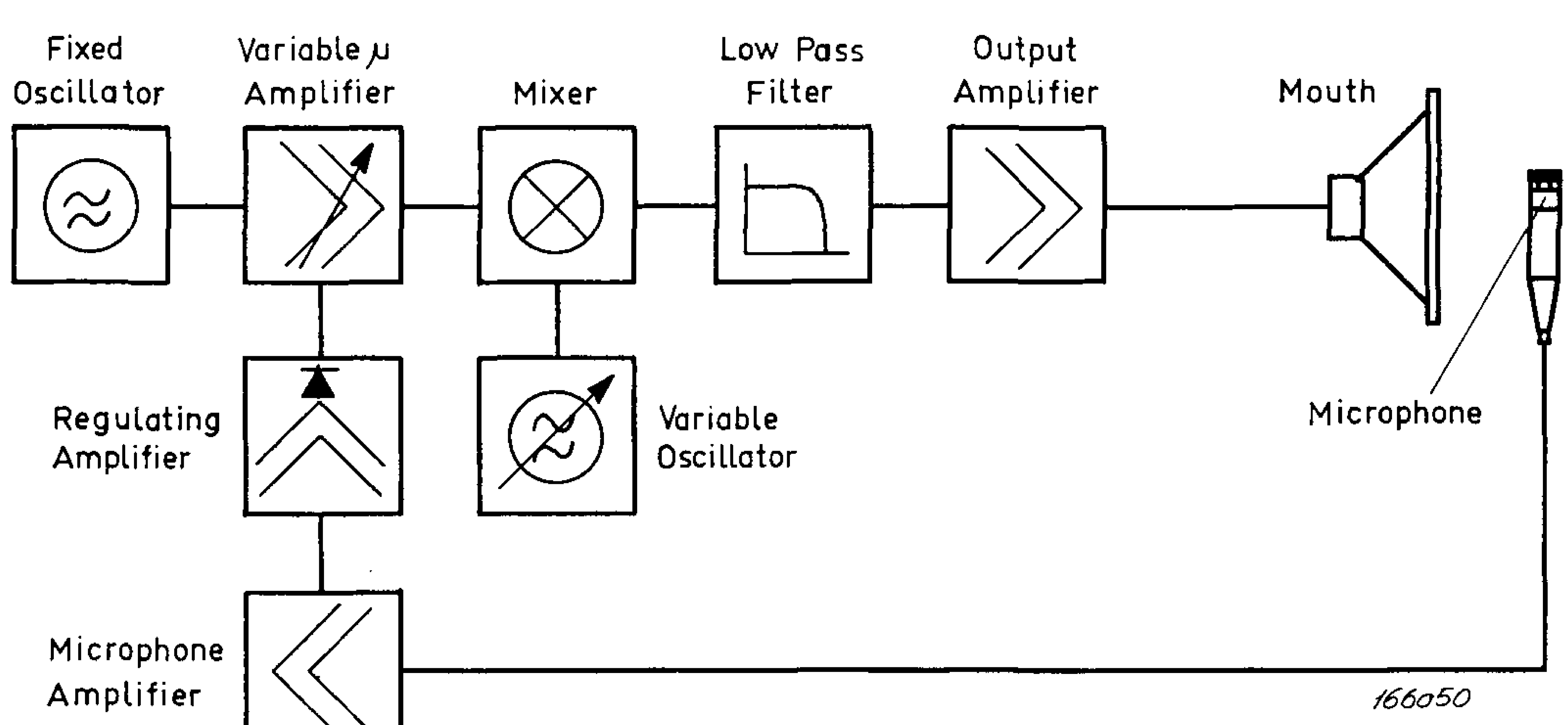



*Fig. 2. Function diagram.*

socket of the regulating circuit in the oscillator. Here in this circuit the AF signal is rectified in a full wave rectifier designed to give a DC output voltage proportional to the average value of the AF control voltage. Following this is an AC filtering network which can be varied in order to change the regulating speed. Finally the DC output voltage controls the grid bias of a pentode situated in the output circuit of the fixed oscillator in the heterodyne oscillator before the mixer stage. This gives a regulated output to the loudspeaker, dependent on the amplification in the regulating loop, and in this way a definite constant sound pressure level in the mouth opening is obtained. Should the sound pressure tend to increase due to non linearities in the loudspeaker characteristic, then the output from the regulating microphone will also increase and bring about, by means of the regulating circuit in the oscillator, a decrease in drive signal to the loudspeaker. We can from this assume that at any point in the regulating circuit there will be a constant level, but at the loudspeaker drive circuit one can investigate the changes in level brought about by the regulating circuit.
The pressure microphone used has a flat pressure response from 20 Hz—20 kHz, which means that a sound pressure can be kept constant within this frequency range provided that the radiated sound pressure level is below the radiated sound pressure level at the lowest points of the unregulated frequency characteristic.

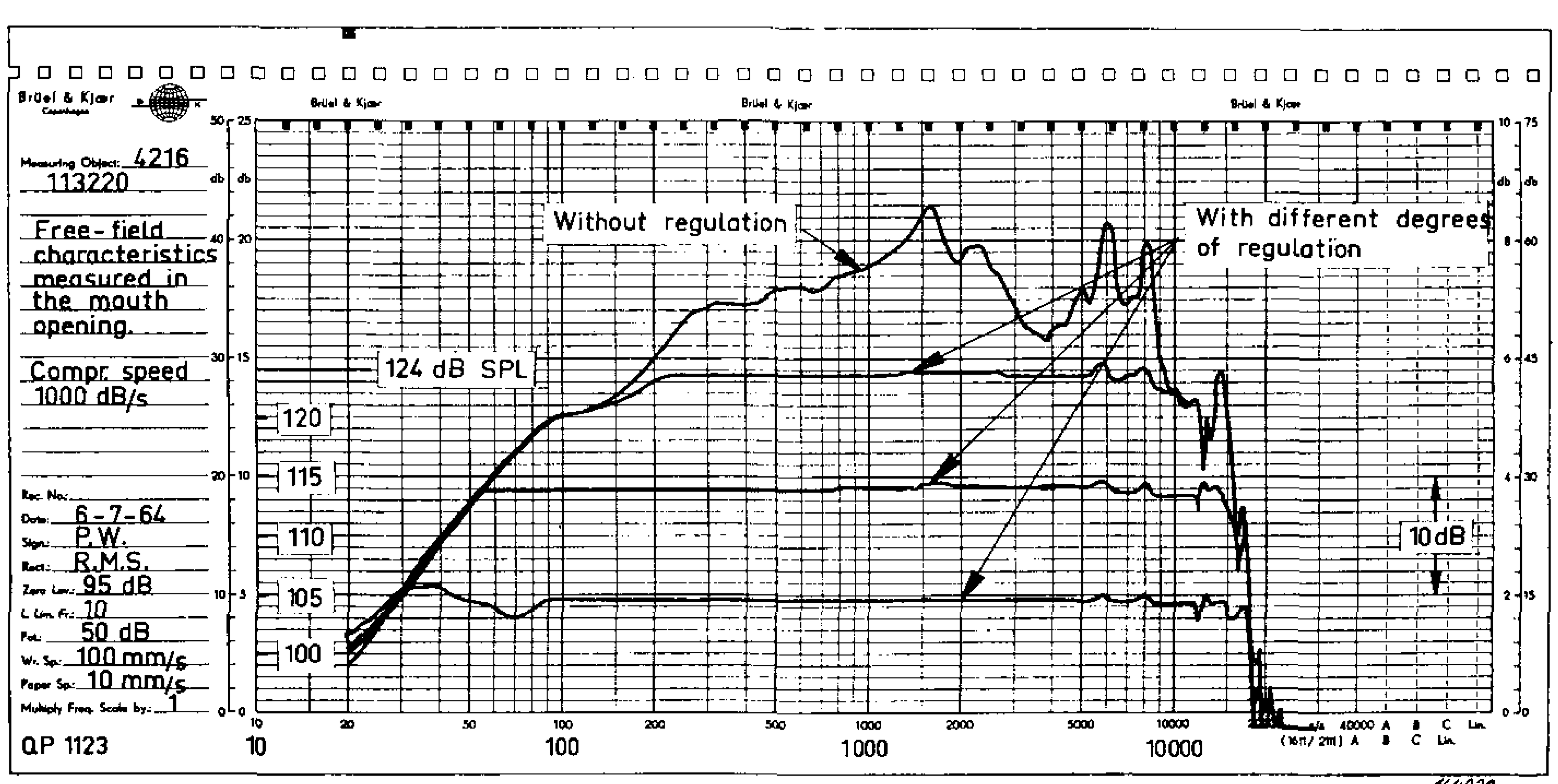



*Fig. 3. Free-field characteristics measured in the mouth opening.*

With the regulating circuit switched out and the built-in microphone used as measuring microphone, Fig. 3, curve I shows one possible pressure distribution over the frequency range. Curve II indicates that with the regulating circuit switched on, the sound pressure can be kept constant from 200 Hz—8 kHz, when 124 dB SPL is required in the mouth opening. By increasing the compression by 10 dB the frequency range will be extended to cover constant sound pressure from 50 Hz—15 kHz, curve III.

This regulating principle means then, that interactions caused by an object in the near field of the mouth, would be sensed by the regulating microphone and thus regulated out. However this interaction is, under normal practical conditions, so small that it can be considered negligible. Even so we are always able to determine its size.
The artificial mouth described here has an output impedance which is zero due to the regulating system. The impedance of the human voice is neither zero nor infinite as it changes considerably with articulation. Pronouncing a "P", one has the feeling that the impedance is high, and with an "A" the impedance is low. Impedance measurements were carried out under pressure conditions, but the DC components of speech made these measurements very difficult. The results showed, however, that different people gave different results and most likely great variations in impedance.
The human mouth is not a regulated system of the type described here, but one has the idea that we do in fact have some regulating effect incorporated in our vocal mechanism. If for instance, an object is placed close to the human mouth it would cause a pressure increase at the ear, and in some way the speaker would reduce the speech level until the original level was obtained.
Let us return to the artificial regulating system. As previously mentioned, the interaction from an object under test is inevitable and can be divided into two principal effects. By coming very close to the mouth opening with an object the acoustical impedance changes from the 41 acoustical ohms in the free field. This impedance change will increase the pressure at the regulating microphone which in turn brings about a decrease in the output from the oscillator so that the original sound pressure is still present. If however it is desired to retain this effect then it can be incorporated into the regulating circuit by altering the amplification.
The second effect of interaction is caused by wave movements between the object and the mouth, this occurs when the distance between the two is 1/4

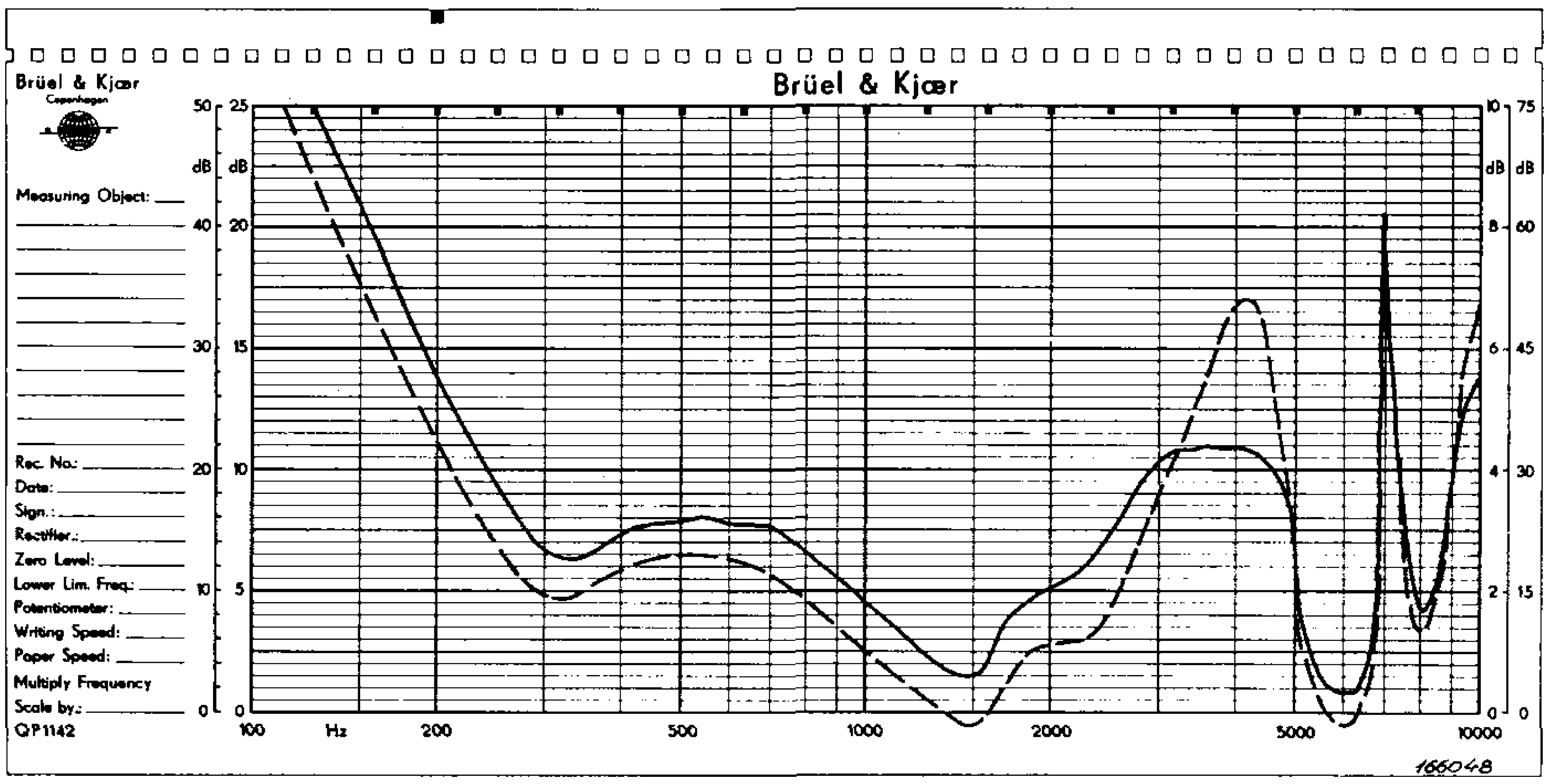



*Fig. 4. Drive voltage for the loudspeaker with and without a test object in the near field.*

wavelength or longer. This effect can to a certain degree be reduced by using a smaller regulating microphone.
If we now measure the drive voltage level to the loudspeaker both with and without an object in the near field of the mouth, then the difference between the two curves will show the exact size of the interaction effect.
From the two curves shown in Fig. 4 the difference is taken and plotted against a constant level indicating no interaction (mouth looking into a free field). This curve is shown in Fig. 5. It can be seen that in the low frequency

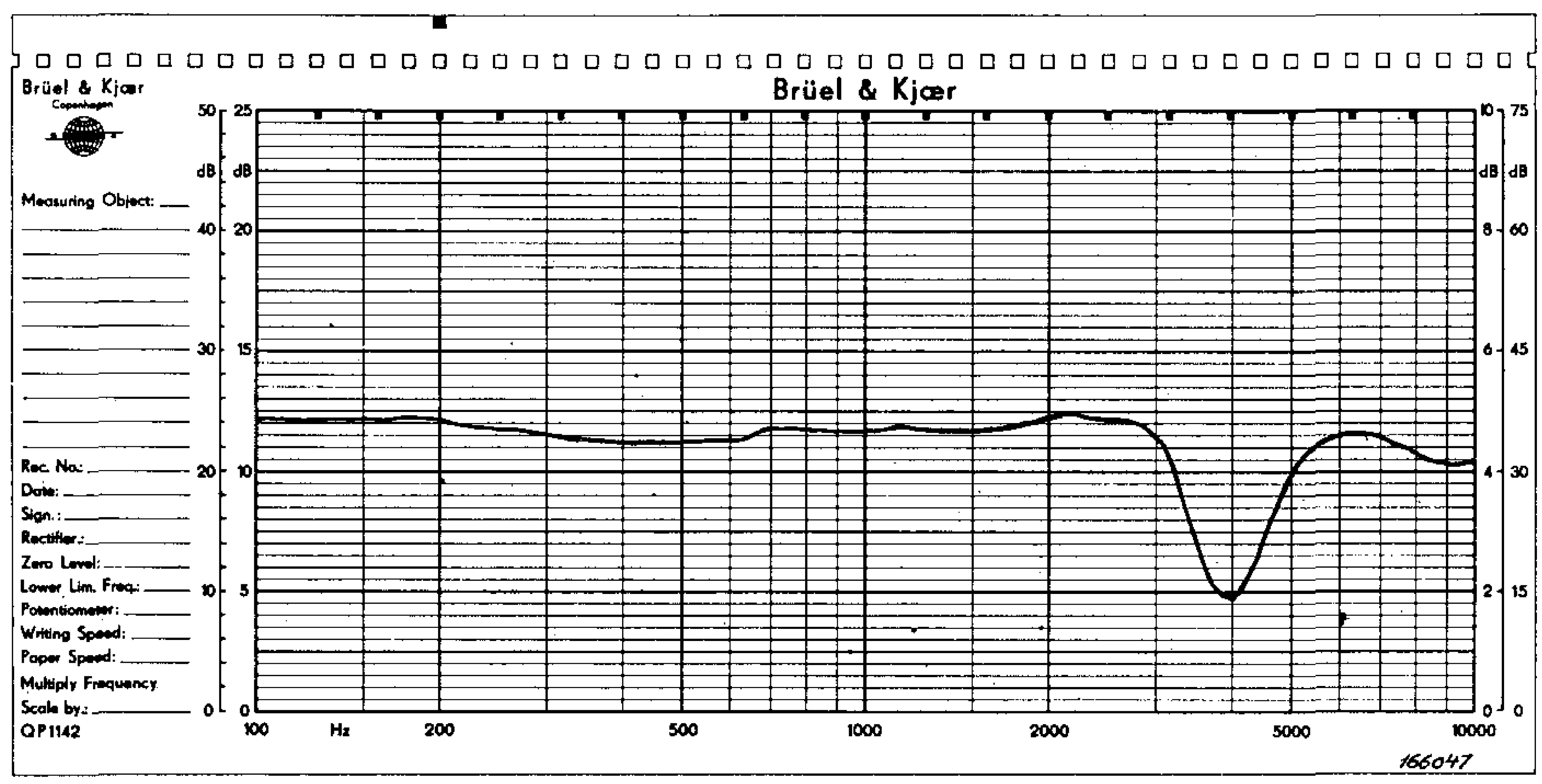



*Fig. 5. The difference curve for Fig. 4 plotted against a constant level (no interaction).*

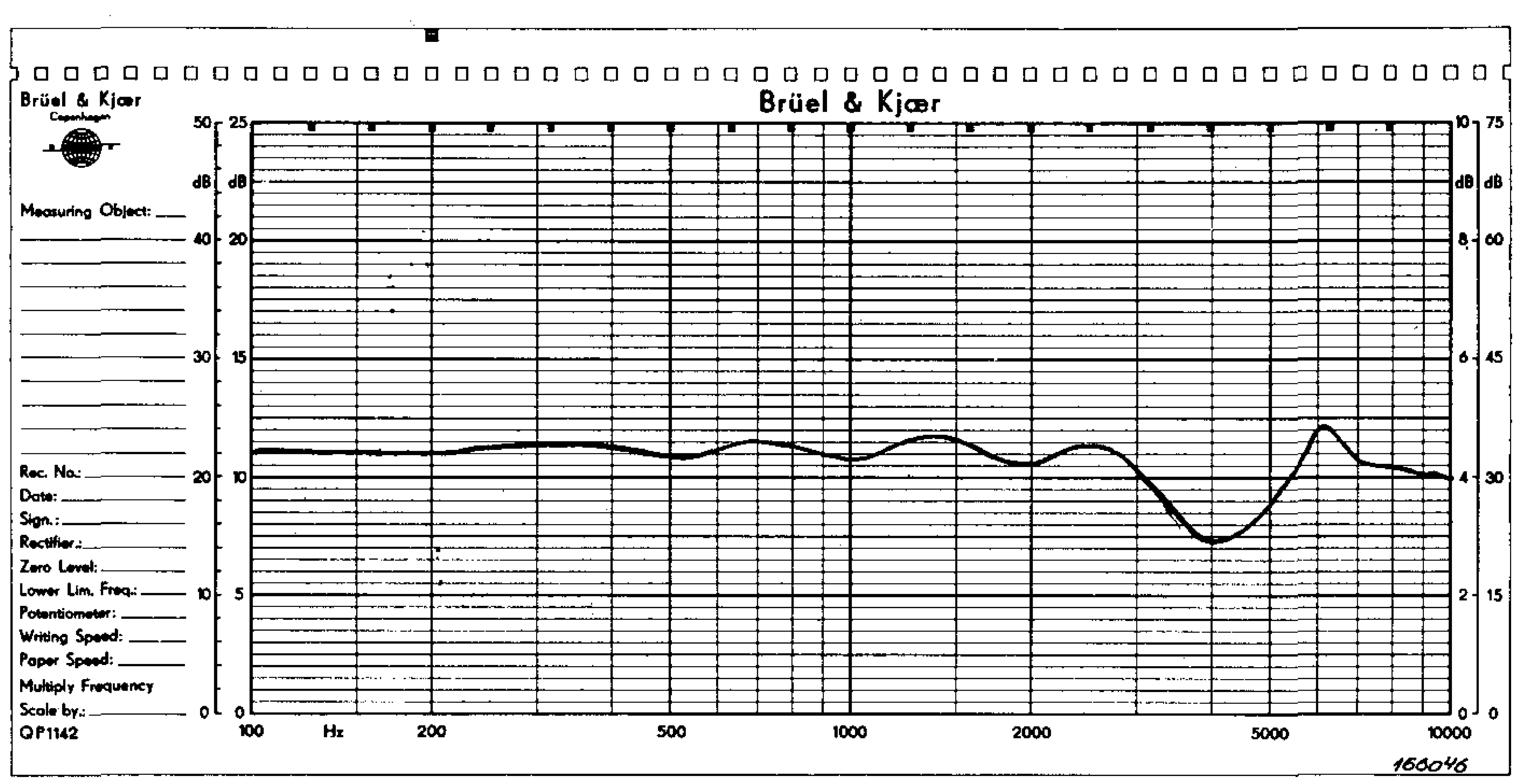



*Fig. 6. As Fig. 5 but with a 1/4″ microphone replacing the 1/2″.*

range the previously described constant influence from changing the impedance in front of the mouth, is present. At 4 kHz a standing wave between the microphone face and the obstacle is present. The curve in Fig. 6 is for the same conditions but with a 1/4″ microphone the reduction in interaction is clearly seen. In the examples given here, the object has been extremely close

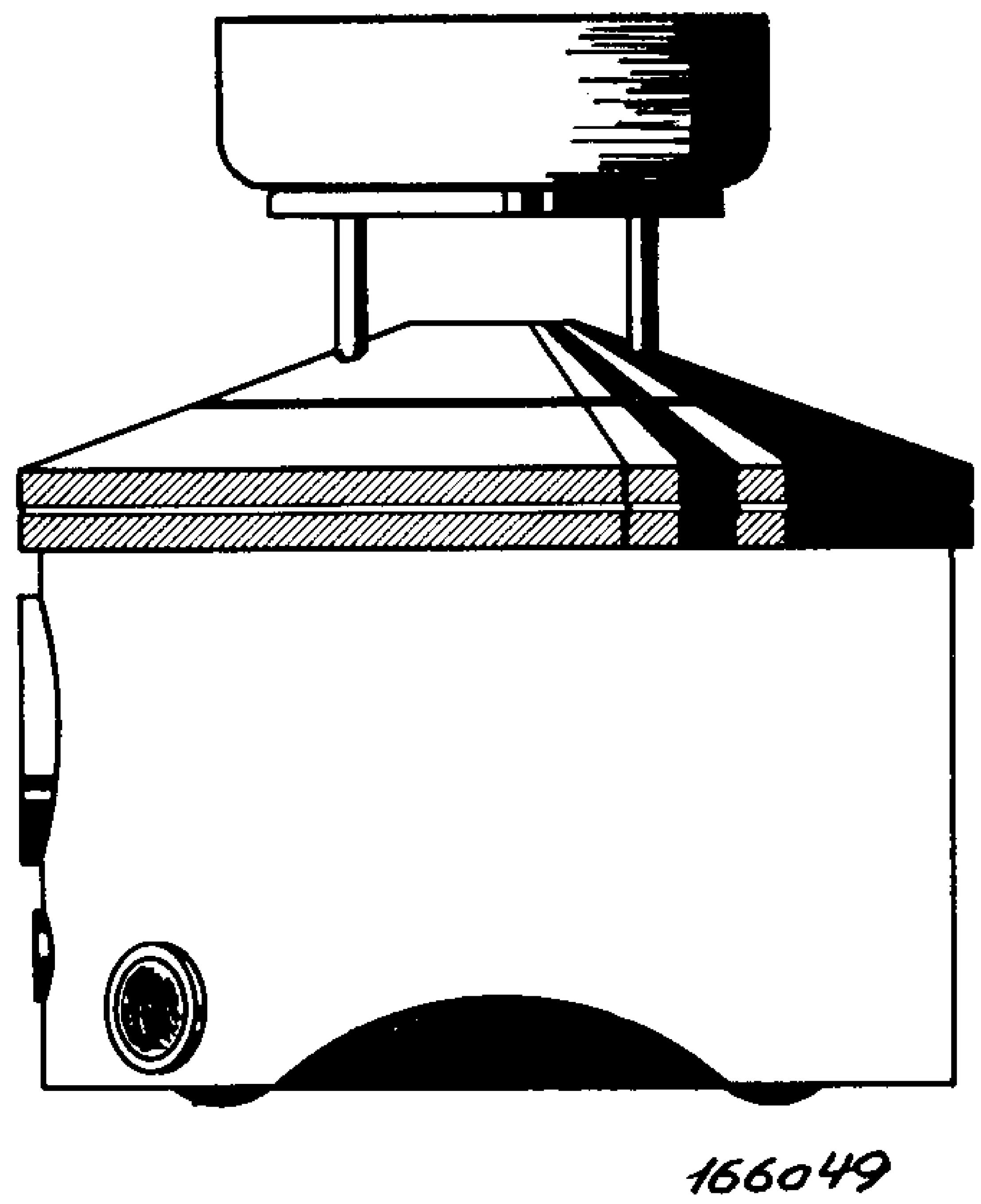



*Fig. 7. Sketch showing the position of the telephone microphone cap on the artificial mouth.*

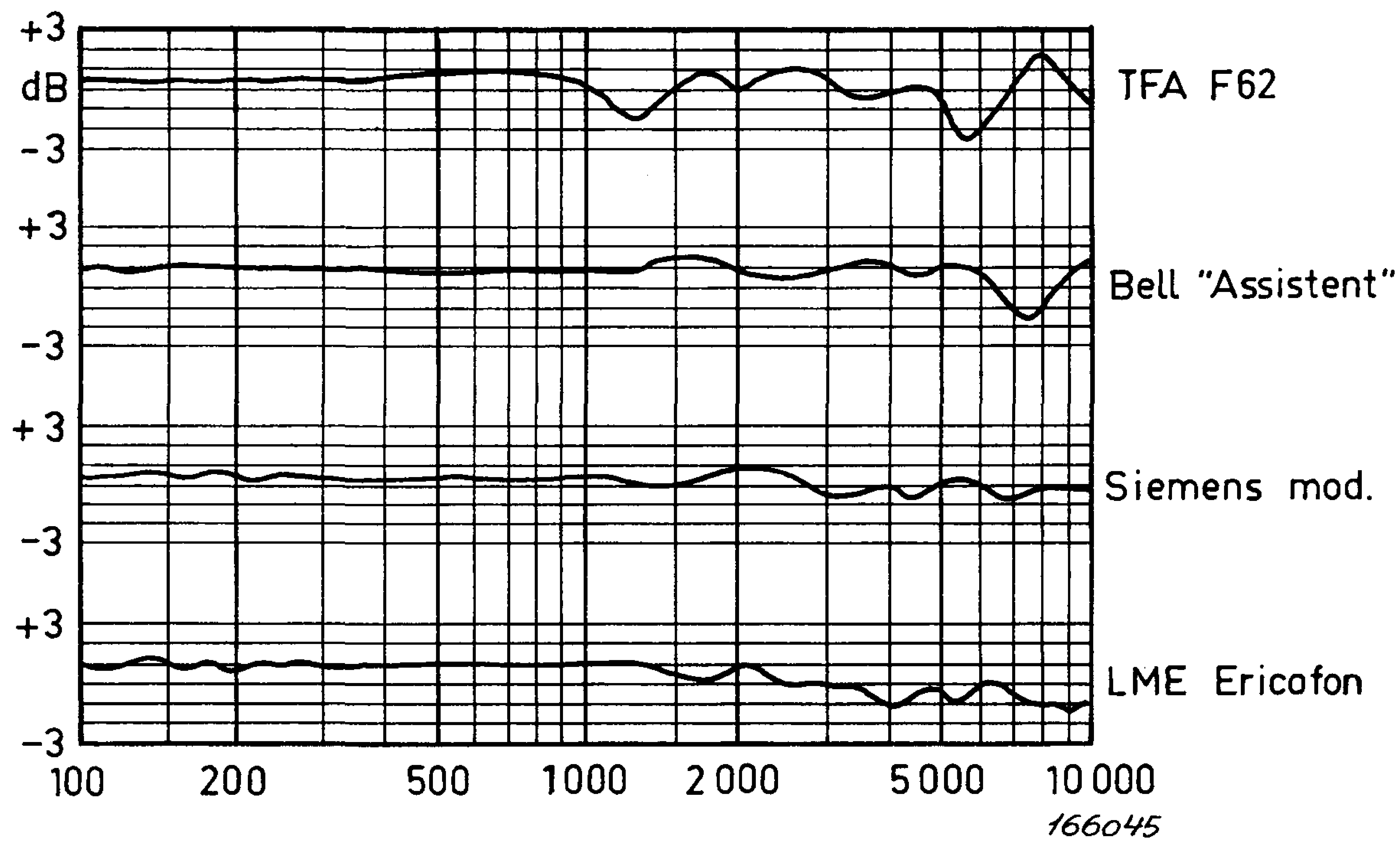



*Fig. 8. Practical measurements of the interaction for different types of handsets fixed according to the Standardized Reference Head.*

to the mouth, in fact it was fixed in the "lip plane" perpendicular to the axis of symmetry of the mouth, as shown in Fig. 7 the object was a microphone cap from an ordinary telephone hand-set.
By replacing the normal microphone grid with a nose cone, the effect of interaction can be reduced by about 5 %. This reduction is probably caused by the influence of the nose cone on the wave motions by changing the Q of the resonance.
During normal practical testing of close-vicinity microphones such as those used as transmitters in telephone hand-sets the position in front of the mouth would have much less influence on the regulating microphone. Fig. 8 shows the interaction for different types of hand-sets fixed in front of the mouth according to the standardized reference head. Some hand-sets are so short that their microphones are placed inside the lip but inclined against the axis of symmetry of the mouth and thus cause less influence.
*Conclusion:* When using a regulating system in an artificial mouth, one has a good long term, reproduceable sound source having constant free field characteristics over a very wide frequency range (50 Hz—15 kHz). Furthermore it is proved that errors introduced by the regulating system under normal conditions, are negligible, and under extreme conditions these errors can be investigated and in turn compensated for.

## Brief Communications.

*With this issue of the Brüel & Kjær Technical Review we are introducing a new section which we have called "Brief Communications". It is the intention with this innovation to cover the more practical aspects of the use of our instruments, and it is hoped to be an "open forum" for communication between the readers and our development and application laboratories.*
*We hope that this section will fill a long-felt need and kindly invite you to take part in the "Communications". Contributions should be as short as possible and under no circumstances longer than 3 typewritten pages (A 4).*

### Method for Recording Latency Times.


By

*Sverre Gran, Cand. real.,* Institute of Physics. University of Oslo.


Several investigations in physics and experimental psychology are concerned with the latency time of a response due to a given stimulus. An experiment following the block diagram Fig. 1a, may be representative:
A signal from a stimulus generator is fed first to a switch and then presented to the object under investigation through a stimulus transducer. The response to this signal is picked up by a response transducer and may for instance be recognized as a modulation of an AC test signal.
The idea of the suggested method is to operate the switch by suitable perfora-

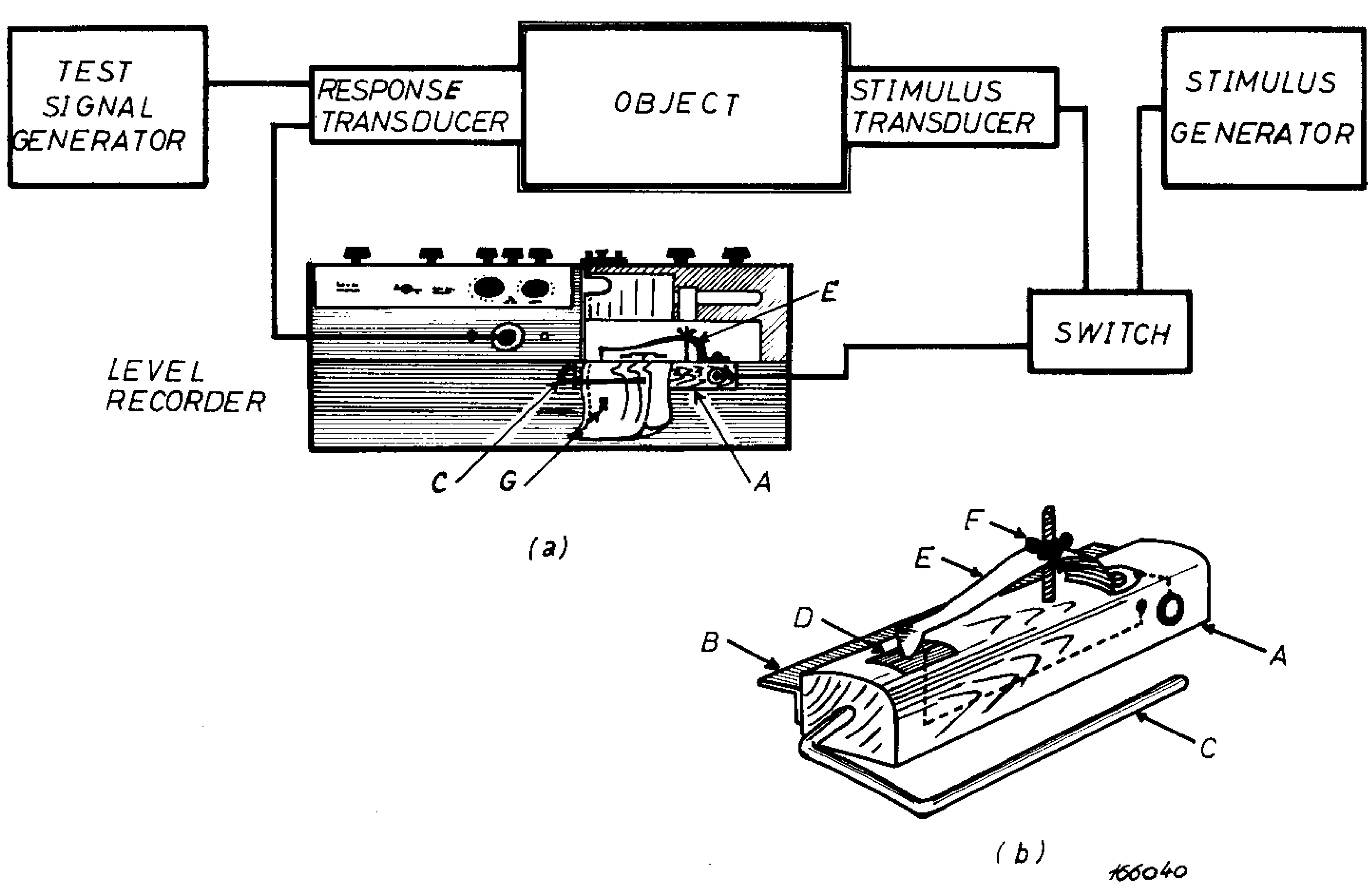



*Fig. 1a) Block diagram showing apparatus recording latency time. Contact arrangement mounted on recorder.*
*Fig. 1b) Contact arrangement.*

tions in the recording paper. A simple device is all that is required to make this method function, in the case of the Brüel & Kjær Level Recorder Type 2305 this may be constructed as shown in Fig. 1b:
A wooden block "A" is fixed to the recorder by a blade "B" which replaces the paper cutting blade. An arm "C" guides the paper over a metal plate "D" embedded in the upper side of the block. The pressure of a phosphor bronze steel spring "E" is regulated by a nut "F" to sweep gently over the paper. The spring makes contact with the metal plate whenever a perforation "G" in the paper passes. One may pass a long cut in the paper between the spring and the plate to give direct contact to the entire stimulus, or one may pass rather short cuts which will operate the switch by pulses.
The same chart length can be used many times thus enabling one to trace several repetitions of the measurement on one single chart and on the same time axis. Spacing between the individual curves is obtained by changing the reference level of the recorder. The time unit is given by the paper speed, and the origin (stimulus on-off) is readily determined by calibration:—The measurement repeated with the actual object replaced by a calibration object which gives a similar response to the same stimulus with zero or a well defined latency time. It will be convenient to use waxed paper and sapphire writing stylus, and it seems possible to obtain 30—40 measurements on the same chart without introducing errors due to damaged paper. The limits of resolution are highly dependent on the smoothness and the rise-time of the signals, and will at best be of the order of 5 ms.
This method should, when usable, be well suited for studying *variations* in latency time, e.g. stochastic variations when a standard stimulus is used, or systematic variations due to variations of stimulus parameters.

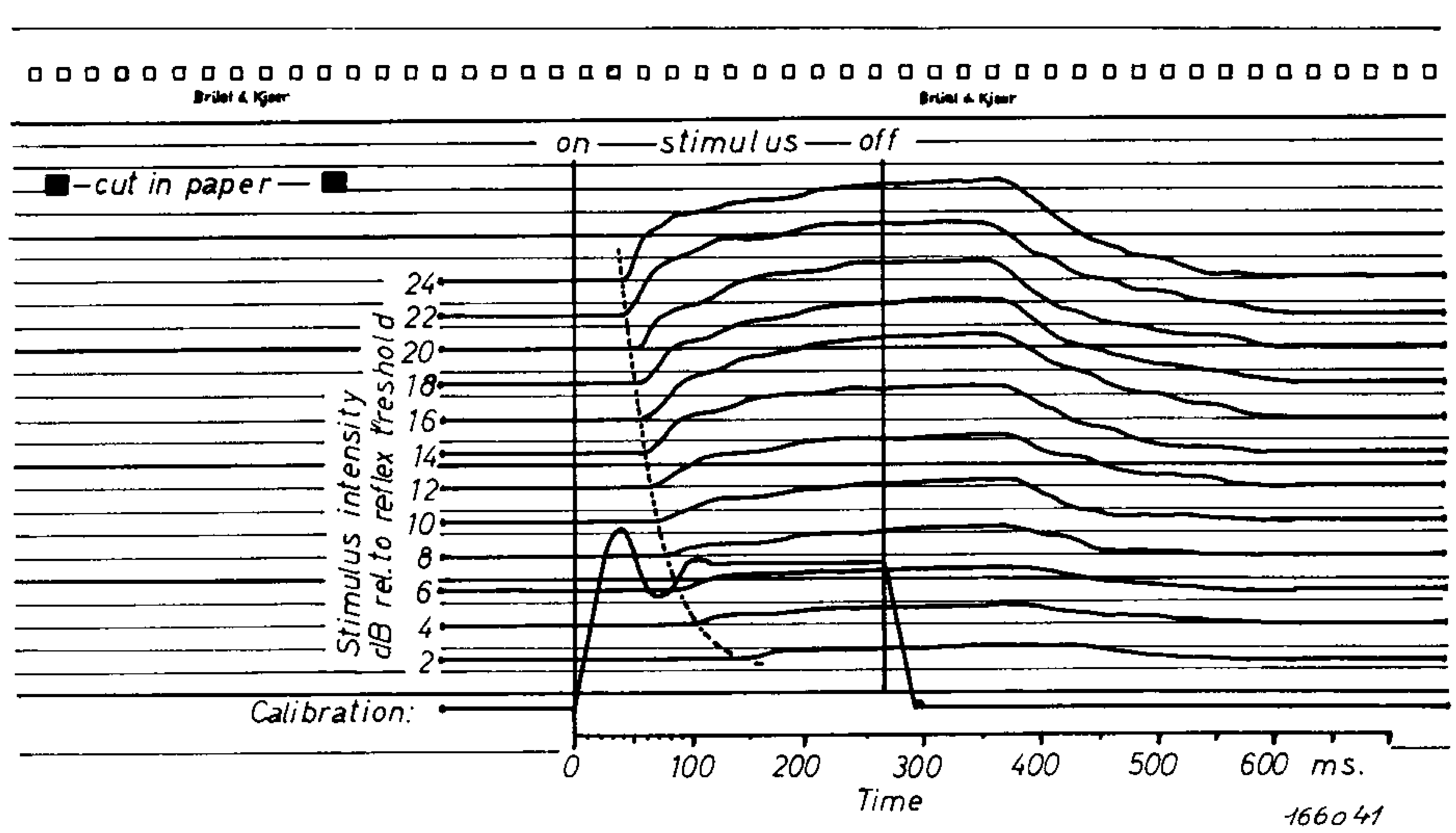



*Fig. 2. Chart showing latency time of contralateral middle-ear reflex as a function of stimulus intensity (dotted line).*

As an example Fig. 2 shows the latency time of the contralateral acoustic middle-ear reflex in a man, and its variations with stimulus intensity. The stimulus is an acoustic signal presented by an ear-phone. The response is a change in the acoustic impedance in the opposite ear, detected by a telephone-microphone probe transmission system. Calibration of the time scale is performed by close acoustic coupling between stimulus transducer and response microphone probe.

| | |
|---|---|
| Stimulus: | Pure tone of 2000 cps, intensity varying in 2 dB steps and rise-time 10 ms. |
| Test signal: | Pure tone of 450 cps. |
| Paper speed: | 100 mm/sec. |
| Writing speed: | 800 mm/sec. |
| Potentiometer: | 10 dB Logarithmic Potentiometer. |

## A Measuring Set-up for Carrier Frequency Equipment and Narrow Band Filters in the Frequency Range 50 kHz - 200 kHz.

By

*S. E. Fauerskov,* A/S Brüel & Kjær

During the period that the Frequency Response Tracer Type 4709 has been in production there have been requests, from various sources, for a logarithmic amplitude scale covering a dynamic range of 5 dB together with a relatively narrow frequency range.

In response to these requests for a 5 dB range, a modified 4709 is being produced where the 25 dB range is replaced by a 5 dB range. In this form the Response Tracer will be known as Type 4711, and as a standard accessory the 4711 will be supplied with a plug-in unit covering the linear frequency range of 150 Hz—4150 Hz.

Fig. 1 shows a set-up for checking 4 kHz bandpass filters in the frequency range 60—108 kHz.

The Tracer has a separate X-input that is only sensitive to frequency. This then, with the help of a mixer and low-pass filter, is used to indicate the difference between the frequency of the test signal and a fixed (crystal) frequency. The fixed frequency is therefore only used as a reference which must be altered when changing to another frequency range.

The advantages with this measuring set-up are:

1. The scale covers a definite frequency band whose width is independent of the frequency. This is exactly what is required when measuring on channel filters for carrier frequency telephony and the like.
2. With the above application it is possible to utilize the same plug-in unit, scale and tolerance curves for all channel filters.
3. The accuracy of the frequency indication (X-scale) is the same (in Hz) as it is for the direct measurement of the low frequency, this assumes that the reference frequency is ultra stable.

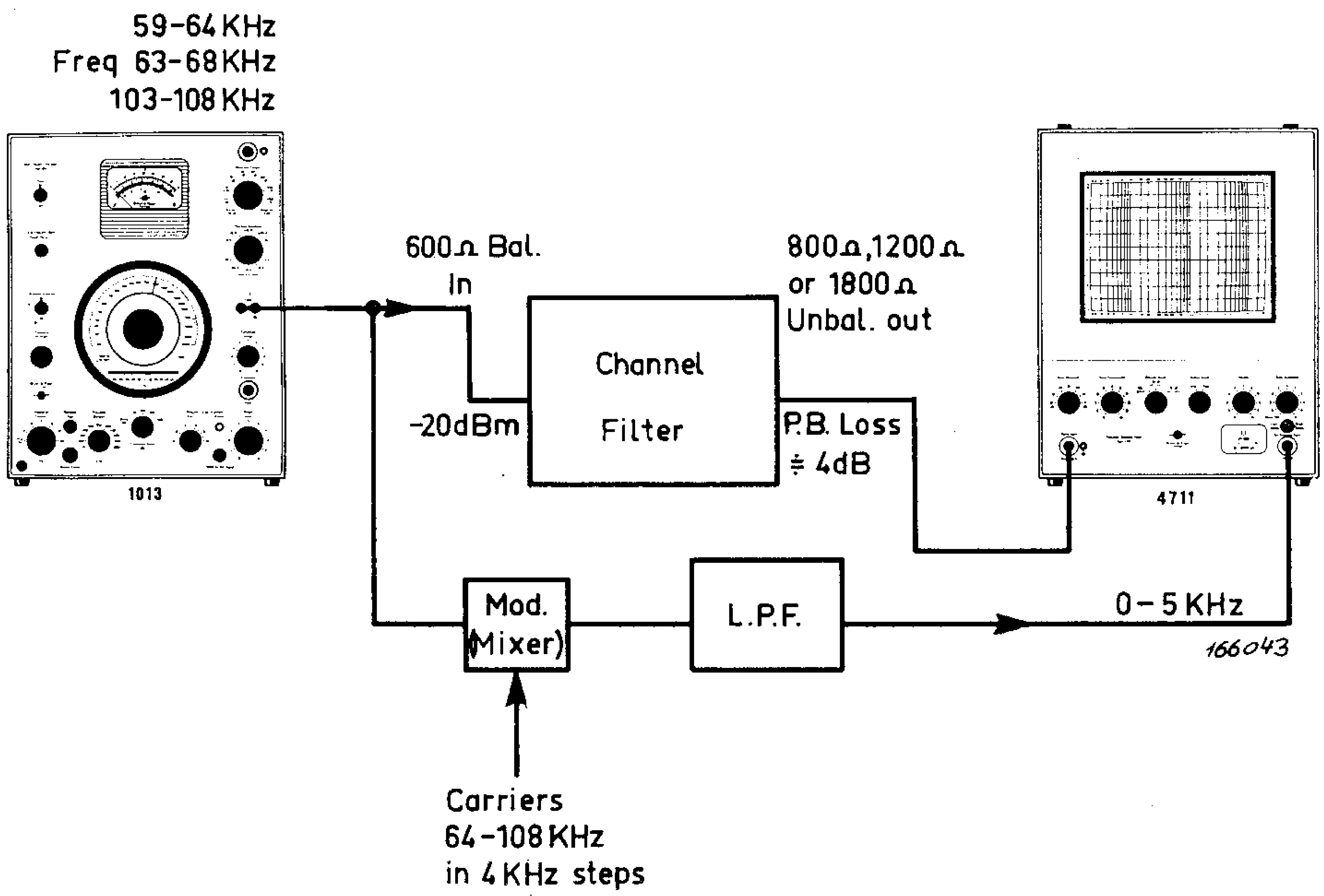



*Fig. 1.*

The measuring set-up shown in Fig. 1 is used, among others, by STC in England for the measurement of the frequency characteristic of channel filters in the range 60—64 kHz, 64—68 kHz etc. up to 104—108 kHz. The available fixed frequencies of 64, 68 - - - 108 kHz are used as the "reference" frequencies for the mixer. As the oscillator Type 1013 is swept (by means of the built-in wobbler) through the frequency range 64—59 kHz and using 64 kHz as reference, 0—5 kHz is obtained at the output of the mixer and in turn fed to the external X-input of 4711.

By using the plug-in unit "150—4150 Hz", 63,850 Hz—59,850 Hz will now be indicated from left to right on the linear frequency scale. Thereby amply covering the frequency range of interest, namely $F_c$—200 Hz to $F_c$—3400 Hz (SSB modulation with audio frequency 200—3400 Hz).

Fig. 2 shows the method of measuring the frequency characteristics of the so called "channel modems" on the transmitter side.

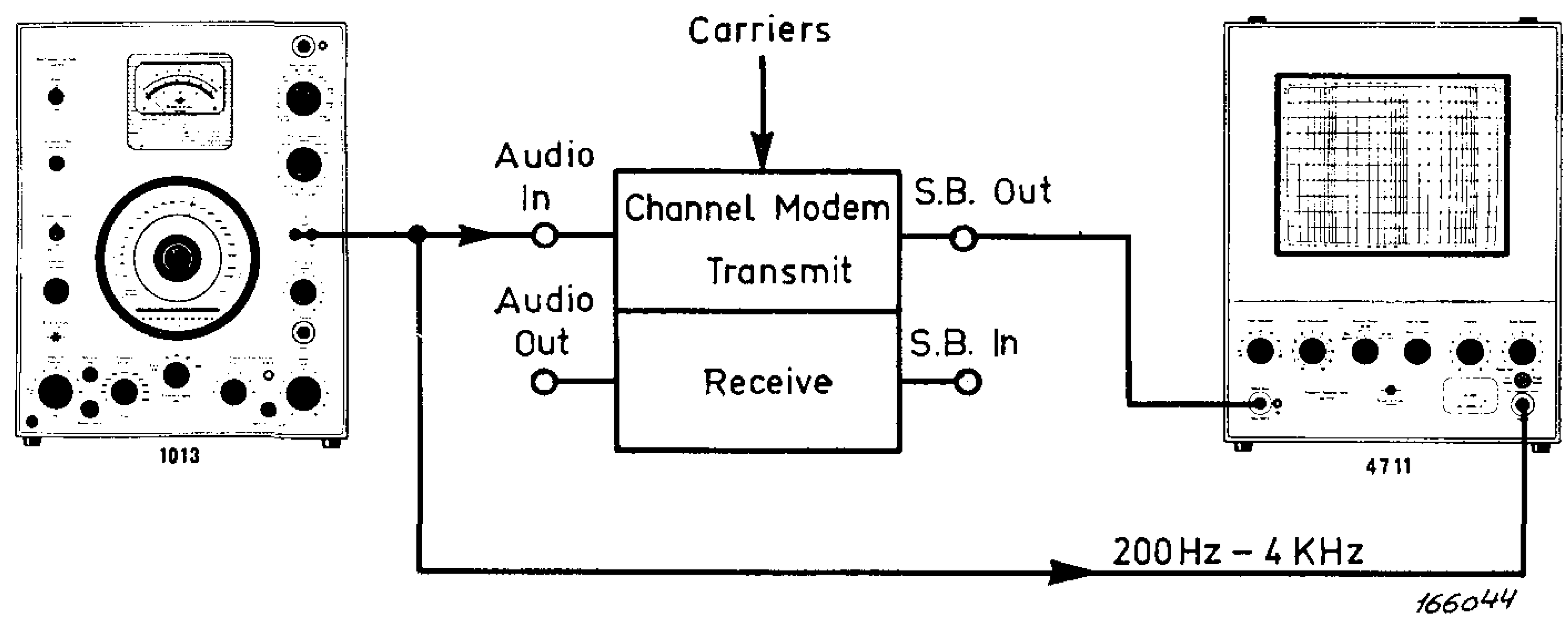



*Fig. 2.*

Fig. 3 shows the measurement carried out on the receiver side. In this case, as we have a low frequency measurement, we use a logarithmic frequency scale and a corresponding plug-in unit in the tracer.

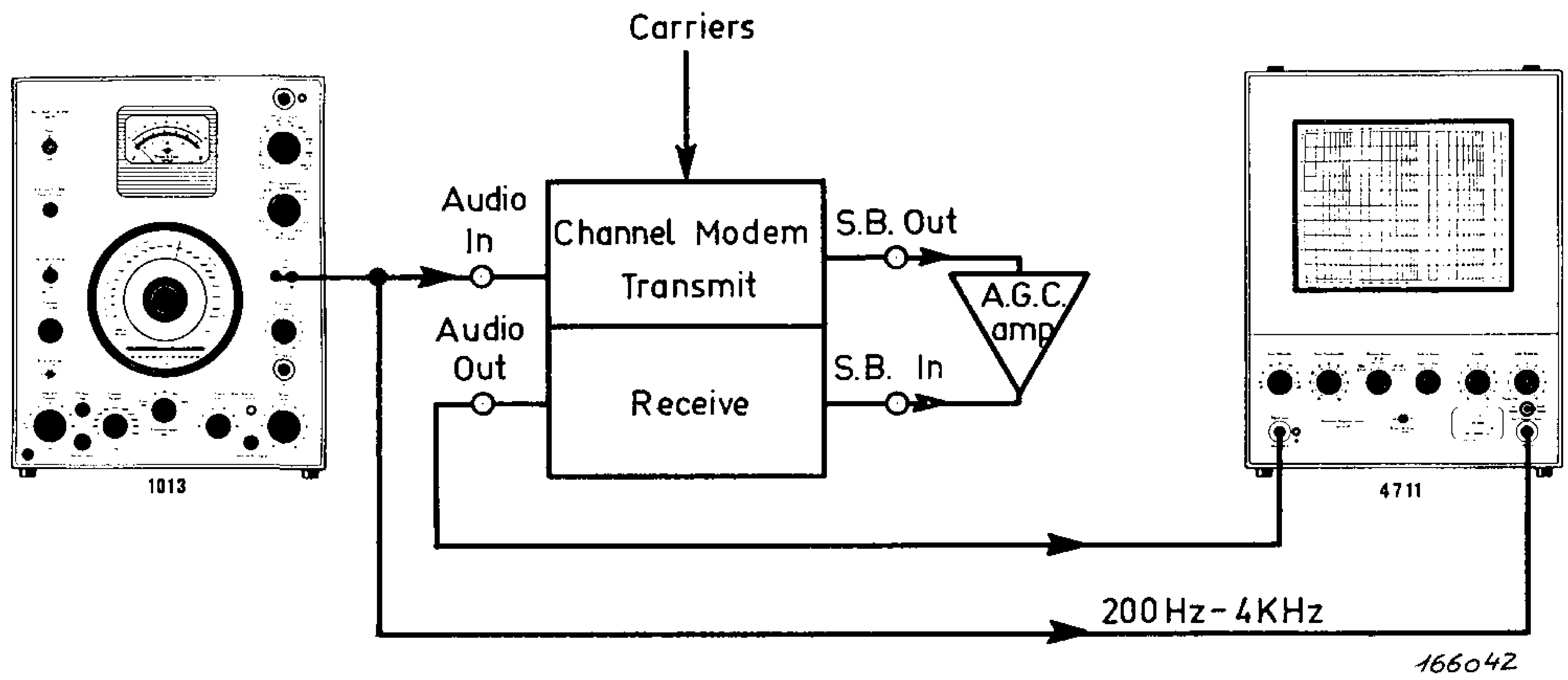



*Fig. 3.*

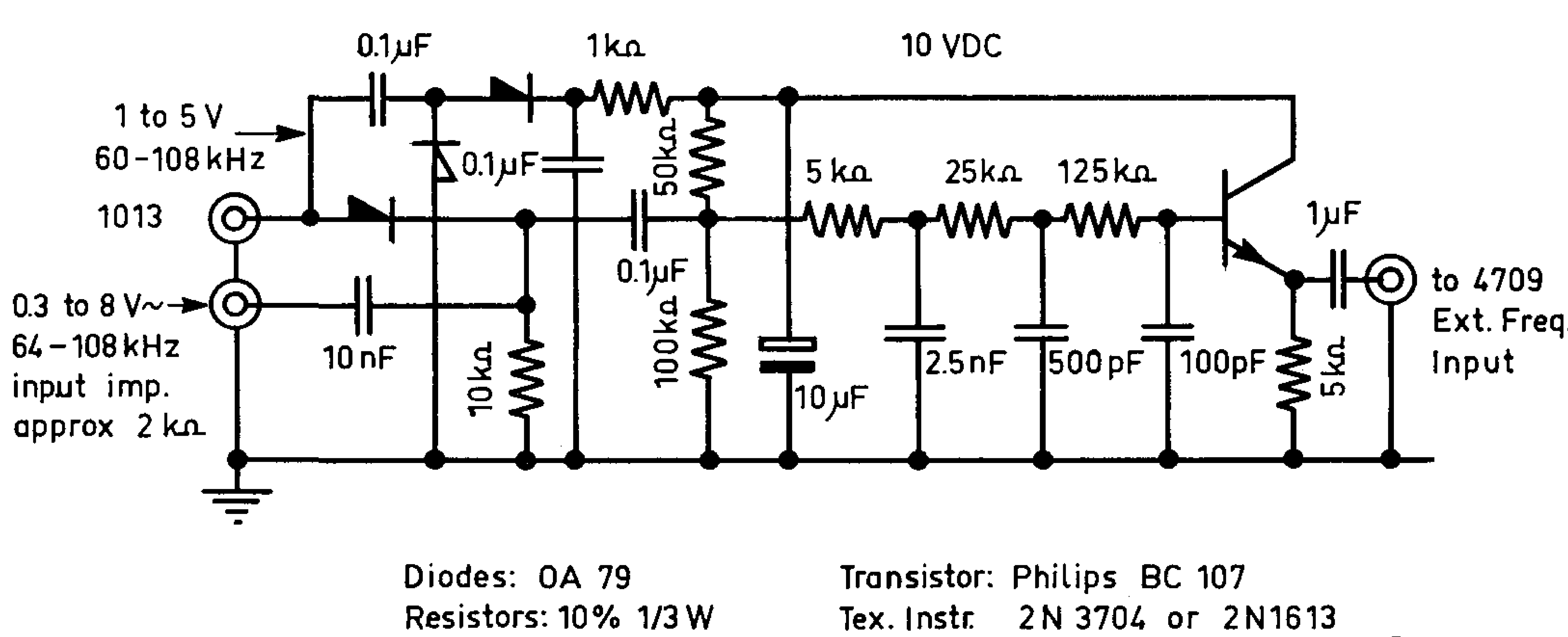



*Fig. 4.*

*Mixer stage and filter for narrow band measurements.*

In the set-ups of Figs. 1 and 2 is shown a mixer stage and a filter. The diagram Fig. 4 shows a simple example of a mixer stage followed by a low pass filter. The emitter follower after the filter is supplied with power by rectifying one of the signals.

*Modification and improvement of Type 1013.*

From serial no. 159322 the wobbler has been improved, so that the amplitude modulation at maximum frequency swing ($\pm$ 2500 Hz) is less than o.2 dB. When making use of the built-in compressor circuit this amplitude modulation becomes less than o.02 dB.

*Improvement of Type 4709.*

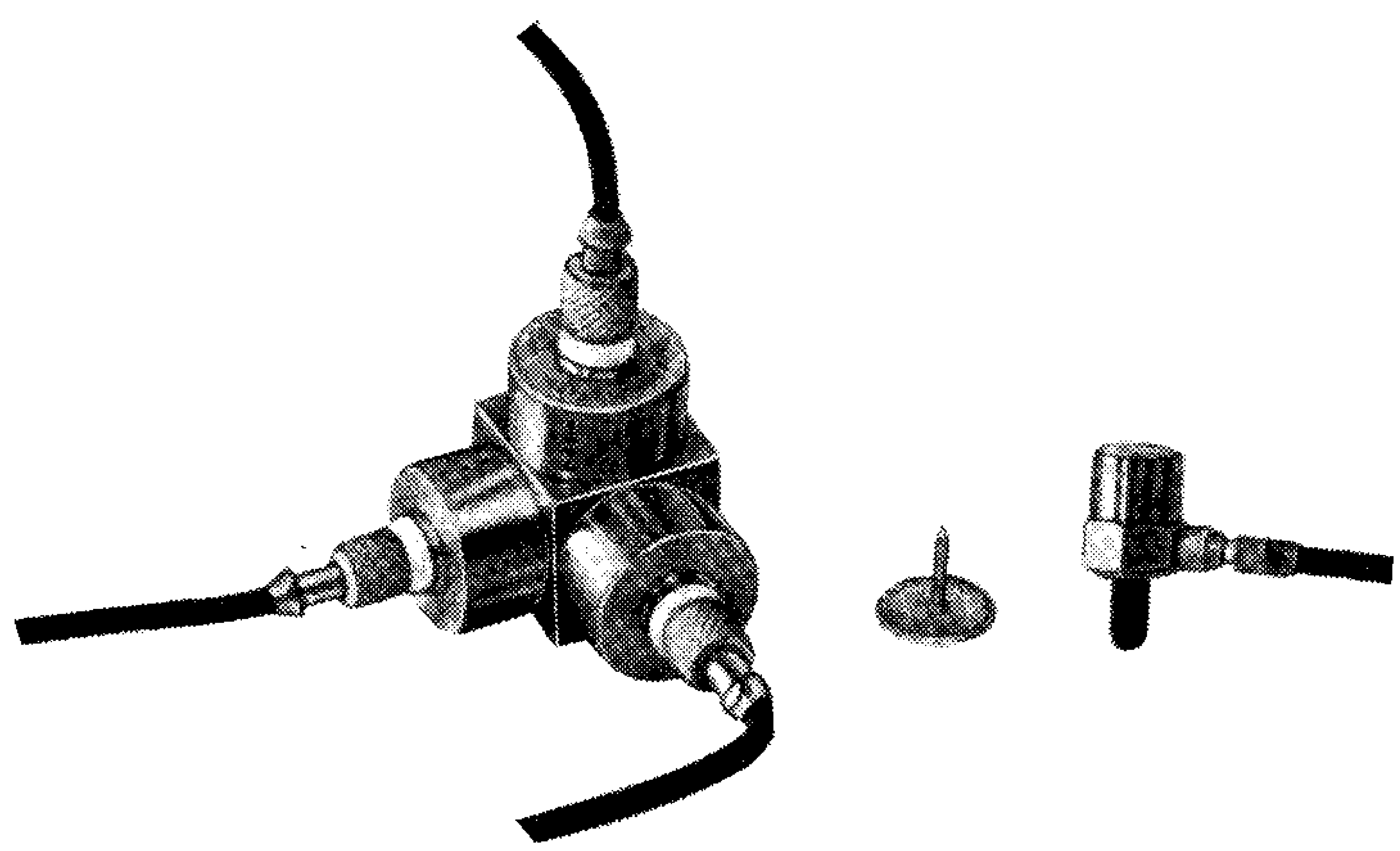
The stability of the "Sweep Limit" adjustment (motor control) is now better than 1 % of full scale over the course of one day, after normal warm up time. The frequency range of the logarithmic amplifier has been extended upwards by 1 octave, i.e. approx. 1 dB drop at 200 kHz.

# News from the Factory.

**New Accelerometers.**

Two new types of accelerometers are now available as extensions to the B & K acceleromter program.

One of these is a Miniature Accelerometer Type 4336 (shown to the right in the picture). The weight of the Accelerometer is only 2 grams, making it especially well suited for measurements on relatively light weight structures where otherwise the mass of the accelerometer might greatly influence the measured results. It has a resonance frequency of 125 kHz (kc/s) and the sensitivity is 4—6 mV/g. As all the B & K accelerometers, each Miniature Accelerometer is supplied with full individual calibration data upon delivery. The base material is titanium and the mounting thread is M3.



The second new accelerometer is a Triaxial Accelerometer Type 4340 which is intended for use where the vibrations of a point on a structure are to be determined simultaneously in more than one direction. Its useful frequency range (to within 2 %) is 2—5000 Hz (c/s). The sensitivity is 14—20 mV/g and the total weight is 35 grams.

This acceleromter also has a titanium base, but the mounting thread is 10—32 NF. Provision is made for watercooling when measurements are to be made at very hot points.

Both of the new acceleromters are delivered in the form of Accelerometer Sets containing not only the accelerometer itself but also various mounting accessories, thus the ordering type numbers are:

*Miniature Accelerometer Set Type 4316.*

*Triaxial Accelerometer Set Type 4320.*

**New Accelerometer Preamplifiers.**

The Accelerometer Preamplifiers Type 2616 and Type 2623 have been developed with special emphasis on small size and battery-operation for use outside the laboratory. They are both rugged, solid state devices and can easily be


placed very close to the accelerometer. The latter is important as the preamplifier's main electrical function in vibration measurements is as impedance transformer, transforming the high output impedance of the accelerometer into the low impedance output of the preamplifier. This transformation allows long cables to be used between the preamplifier and the actual amplifying and indicating instruments (up to some 100 m cable with a capacity of 100 pF/m).
The Type 2616 Preamplifier contains a peak-indicating overload indicator, a 40 dB attenuator and a signal adjustment potentiometer. Power is drawn from 6 Mercury Cells (1.35 V each) which are contained in a battery compartment attached to the Preamplifier (black, see also photo above). The battery compartment is removeable and if so desired the Preamplifier can be connected to an external power supply of any voltage from 6 to 35 Volts D.C. Provision is made for battery voltage checks.
The input resistance is greater than 1200 M$\Omega$, the input capacity less than 10 pF. Output impedance $< 100\ \Omega$. The inherent noise level is 15 $\mu$V (for 2 Hz (c/s) — 40 kHz (kc/s) and 1000 pF across the input terminal).
The Preamplifier Type 2623 does not contain an overload indicator and must be powered from an external battery (28 Volts, current consumption 2 mA). It has, however extremely small mechanical dimensions and may, if desired, be mounted directly on the accelerometer. In case the size (or weight) of the accelerometer then becomes too great it can also be mounted separately by means a clamping arrangement supplied with the Preamplifier.
The input resistance is greater than 2000 M$\Omega$, and the input capacity is in the order of 3.5 pF. Output resistance is 40 $\Omega$. The inherent noise is 15 $\mu$V (for 2 Hz (c/s) — 40 kHz (kc/s) and 300 pF across the input terminal).
Both Preamplifiers utilize field-effect transistors in the input stages to obtain the required high impedance, wide frequency range and low inherent noise level.

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