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Contents

An Easy and Accurate Method of Sound Power Measurements
by H. Larsen 3

Measurement of Sound Absorption of rooms using Reference Sound Source
by O. Kramář 26

News from the Factory 37

An Easy and Accurate Method of Sound Power Measurements

by

Holger Larsen

ABSTRACT

An easy, quick and accurate method of determining sound power levels in a semi-anechoic chamber is outlined, where the microphone is traversed at a constant vertical speed. The errors caused by interference between the direct sound wave and the reflected sound wave reaching the microphone are minimized by this method, compared to the method using fixed microphone positions.

SOMMAIRE

L'article décrit une méthode commode, rapide et précise pour déterminer les niveaux de puissance acoustique dans une salle semi-anéchoïque, ou le microphone est translaté avec une vitesse verticale constante. Les erreurs causées par les interférences entre l'onde sonore directe et l'onde refléchie atteignant le microphone sont minimisées par cette méthode par rapport à la méthode utilisant des microphones fixes.

ZUSAMMENFASSUNG

Es wird eine einfache, schnelle und genaue Methode angegeben, mit der man Schalleistungspegel in einem halb-schalltoten Raum messen kann. Dabei wird das Mikrofon mit konstanter Geschwindigkeit senkrecht bewegt. Die Fehler aufgrund von Interferenz zwischen den directen und den reflektierten Schallwellen am Mikrofon sind bei dieser Methode verringert worden im Vergleich zu der Methode mit festen Mikrofonpositionen.

Introduction

The most accurate values of sound power levels of equipment can be determined in a free field environment achieved in anechoic chambers or in the open air. However, some sound sources are too heavy to be suspended or are associated with a reflecting plane, in which case measurements are carried out in what is termed a semi-anechoic chamber where the equipment is placed on a hard reflecting surface. In this case the spatial irregularity of the sound field may be increased due to the superposition of the sound field of the actual source and that of the image source, leading to a slightly lower accuracy of the measurement results.

Measurement methods for both these environments are outlined in Draft International Standard ISO 3745. For measurements over a hard reflecting surface, three different methods of measuring the mean spatial sound pressure level around the sound source are suggested.

Method 1

In the first method, the sound source is placed on a hard reflecting surface and the sound pressure level emitted by the source is measured at a number of microphone positions, placed on the surface of a hypothetical hemisphere with its centre preferably coinciding with the acoustical centre of the sound source. Since the acoustic centre is frequently unknown, the geometric centre is often chosen. The positions of the microphones on the surface of the hemisphere are uniformly spread out, so that each position is associated with an equal area. Fig.1 shows 10 microphone positions which is the minimum number required by the standard.

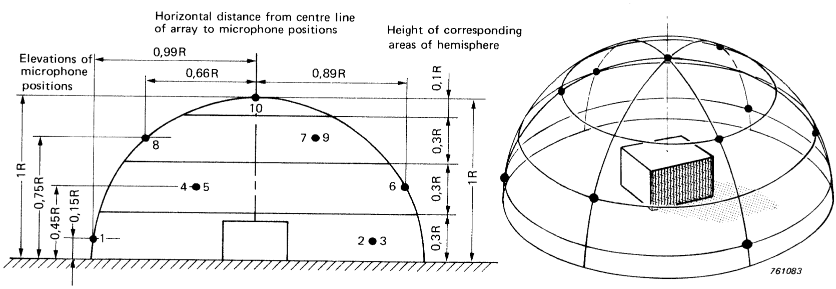


Fig. 1. Microphone positions for Method 1 as suggested by Draft ISO 3745

Method 2

Another method of determining the mean spatial sound pressure level is by traversing a single microphone successively along horizontal circular paths around the surface of the hypothetical hemisphere. A minimum of five circular paths must be used as shown in Fig.2 where the annular area of the hemisphere associated with each path is equal. The microphone should be traversed at constant speed. Alternatively, the microphone could be placed at five different heights as shown in the figure while the sound source is rotated at constant speed using a turntable.

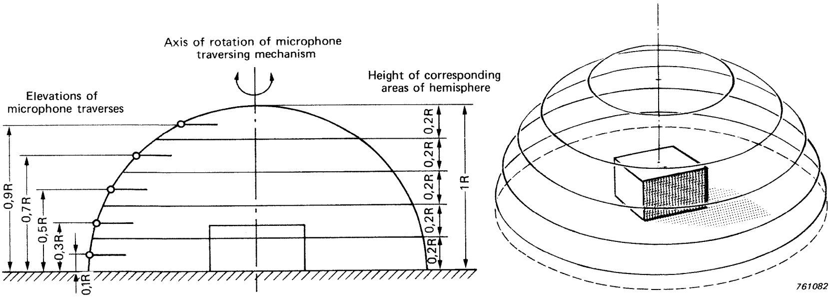


Fig.2. Microphone positions for circular arcs for Method 2 as suggested by Draft ISO 3745

Method 3

A further method outlined in the standard involves traversing a single microphone along a meridional arc, i.e. the microphone is moved along a quarter of a circular arc the axis of which lies in the reflecting plane. The traverse is shown in Fig.3. At least eight microphone traverses at equal azimuthal angles around the source must be carried out, which may be achieved by rotating the source. Since the microphone is traversed at a constant angular speed, equal areas of the surface of the hemisphere are not associated per unit of time of the microphone traverse. To compensate for this, a cosine function potentiometer must be included in the measuring chain.

On account of the directional characteristics of the sound source and the interference between the sound wave reaching the microphone directly and the sound wave reflected by the reflecting plane, the sound pressure level over the surface of the hypothetical hemisphere can vary significantly, see Appendix A. According to the standard the number of microphone positions for method 1 is sufficient, if the difference in decibels between the highest and the lowest sound pressure levels measured in any frequency band of interest is numerically less than half the number of measurement points. As will be shown later, the differences between the highest and the lowest sound pressure level could be up to 17 dB, whereby a considerable number of measurement points would be required to fulfil the standard.

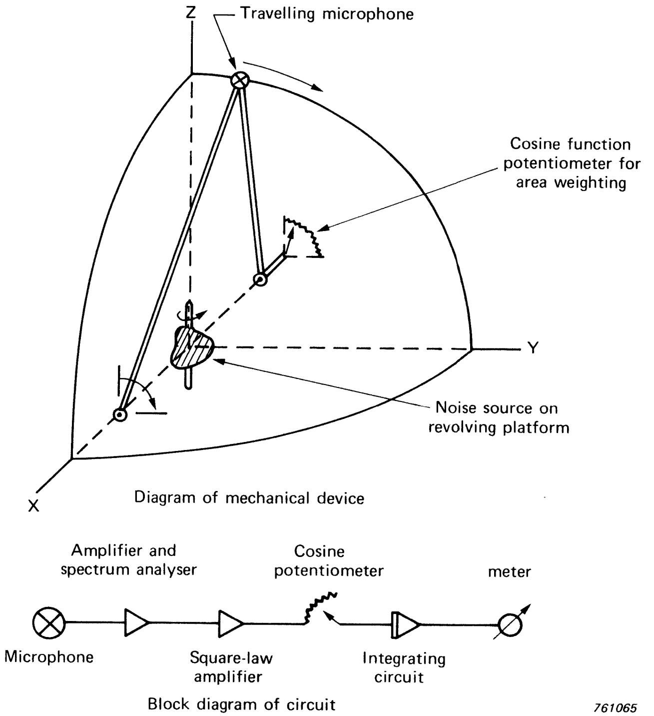


Fig.3. Microphone traverse for Method 3 as suggested by ISO 3745

Measurements according to method 2 are relatively easy to carry out. However, since the sound pressure level must be averaged over all the circular paths, the averaging process needs to be stopped each time the microphone is moved to a new position for the next circular path. The effect of interference between the direct and the reflected sound wave will generally be less in this method than in method 1, on account of the greater number of microphone positions in height (5 compared with 4 in method 1).

Method 3 is relatively difficult to realize in practice on account of the microphone sweep not representing equal areas per unit of time. On the other hand, the interference effect would have a minimum of influence on the results, as averaging is now carried out continuously over the height. Also in this method, however, the averaging process needs to be stopped each time the microphone is moved to the next azimuthal angle.

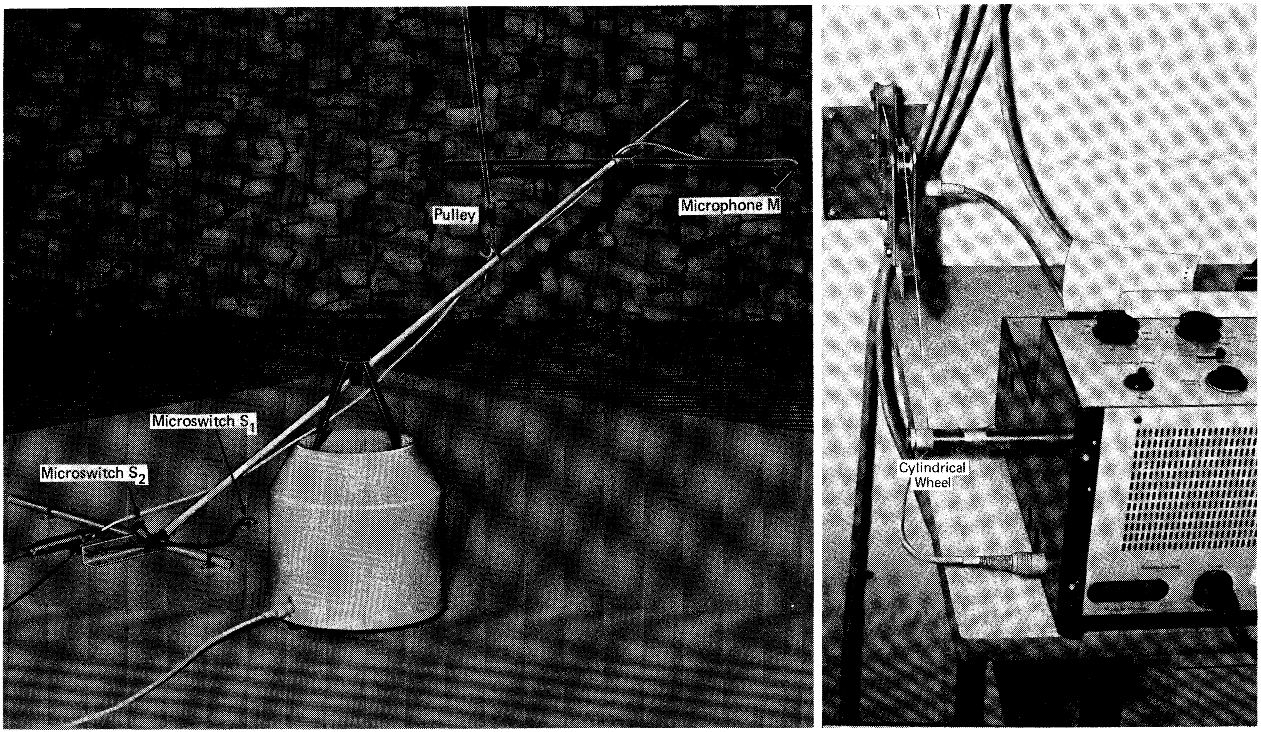


Fig. 4. Measuring arrangement for test object with rotational symmetry of directional characteristics

A Simplified Measurement Method

a) Measurement arrangement and instrumentation

The method suggested here is similar to method 3, except that the microphone traverse is modified such that it moves with a constant vertical speed (instead of constant angular speed) and therefore, is associated with equal areas per unit of time, (see appendix B).

Fig.4 shows a microphone M attached to a swing arm which is mounted on the reflecting surface on two bearings. The swing arm is attached to a string which, via a pulley, is lead to the roof of the anechoic chamber and then around a wheel attached to the mechanical drive of a Level Recorder. The reason for attaching the pulley on the swing arm is simply to halve the tension in the string on account of the relatively low torque available on the mechanical drive of the Level Recorder Type 2307. A microswitch S_1 is fixed to the reflecting plane such that it is activated when the swing arm is in its lowest position. Another microswitch S_2, shown in the figure, is activated when the swing arm is in the vertical position.

The measuring instrumentation used is shown in Fig.5. The signal from the microphone is fed to the Measuring Amplifier Type 2607 connected to a 1/3 Octave Filter Set Type 1615. The filtered AC signal from the Filter Set is fed to the Noise Level Analyzer and Statistical Processor Type 4426, while the output from the Measuring Amplifier may be fed to the Level Recorder. Both the Level Recorder and the Noise Level Analyzer are activated by a relay (when it is in the off position) which is again activated by the microswitches S_1 and S_2.

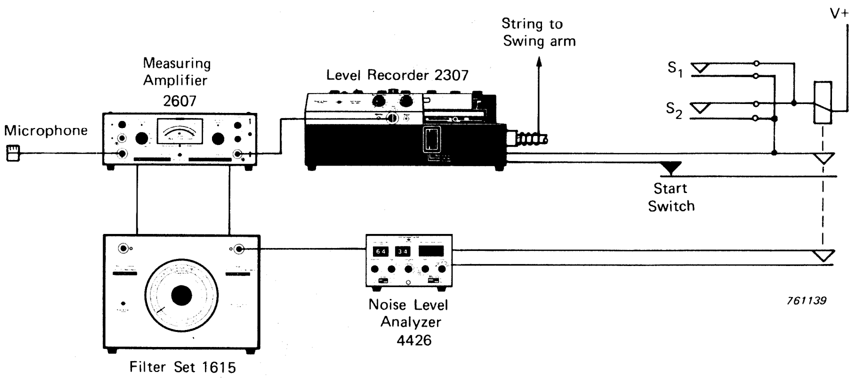


Fig.5. Measuring instrumentation for determination of Sound Power Levels

b) Measurement Procedure

The swing arm is placed in its horizontal position activating microswitch S_1. The start switch in Fig.5 is depressed, whereby the Level Recorder is started and moves the swing arm upwards. Switch S_1 is thus released, activating the Noise Level Analyzer and measurements are taken until the swing arm reaches its vertical position activating microswitch S_2 and stopping the Level Recorder and the Noise Level Analyzer. During the microphone traverse, the Noise Level Analyzer evaluates the L_{eq} value of the noise given by the equation

$$L_{eq} = 10 \log_{10} \left[\frac{1}{T_s} \int_0^{T_s} \left(\frac{p(t)}{p_o} \right)^2 dt \right]$$

where T_s is the observation period

$p(t)$ is the sound pressure at time t

and p_o is the reference sound pressure 2×10^{-5} N/m^2.

It can thus be seen that L_{eq} is the mean spatial average of the sound pressure level measured along the microphone traverse over the surface of the hemisphere. If the directional characteristics of the sound source are rotationally symmetrical in the horizontal plane, a single vertical microphone traverse is sufficient and the sound power level can be calculated from

$$L_P = L_{eq} + 10 \log_{10} \left(\frac{2\pi R^2}{S_o} \right)$$

where R is the radius of the hemisphere

and S_o is reference area of 1 m^2.

When R is chosen to be 1,26 m the second term on the right hand side of the equation gives 10 dB. The Noise Level Analyzer can be calibrated so that this term is taken into consideration, whereby the sound power level can be directly read on the digital display of the instrument.

For sound sources with non-symmetrical directional characteristics the measurements must be repeated for eight microphone traverses uniformly distributed around the vertical axis.

A simpler method, however, is to place the measurement object on a turntable and rotate it during the microphone traverse. As shown in Fig.6, the microphone would then move in a spiral path relative to the sound source. The microphone traverse must be slow enough so that the measurement object can be rotated at least five times during one traverse of the microphone. The greater the number of rotations during the microphone traverse the higher will be the accuracy achieved. However, the measurement time would also correspondingly increase. Ideally an integer number of rotations of the test object during the traverse must be aimed at. This can be easily achieved as the scanning speed of the microphone can be adjusted by moving the fixing position of the string on the swing arm.

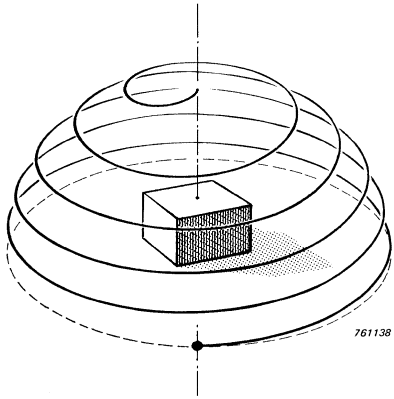


Fig. 6. Spiral traverse of the microphone relative to the test object

c) Sweep Time

The observation time T_s required for a single sweep depends on the statistical error ϵ in the measurements that can be tolerated. The averaging time T is determined from the formula

$$\epsilon = \frac{1}{2\sqrt{BT}}$$

where B is the bandwidth in Hz.

For an error of 2% and the lowest 1/3 octave frequency band of centre frequency 100 Hz the averaging time would be

$$T = \frac{1}{4\,\epsilon^2 B} = \frac{1}{4 \times 0{,}02^2 \times 23} = 27\ \text{s}$$

If the measurement object emits noise uniformly in all directions, as is normally the case at low frequencies, the observation time T_S can be made equal to the averaging time T. If the object, however, has directional characteristics, the observation period must be 2 to 5 times the averaging time.

At higher frequencies, the noise sources often become more directive, but on the other hand the bandwidths get larger so that the same observation period can be used.

Measurement Results

a) Measurement object with rotational symmetry of directional characteristics

Fig.4 shows a picture of the measurement object, which is built up of two loudspeakers (one dome tweeter and one woofer) mounted concentrically in a cylindrical cabinet. The directional characteristics have a rotational symmetry within 0,2 dB in the horizontal plane.

Fig.7 shows an example of the curve obtained on the Level Recorder during the microphone traverse in the vertical plane. The curve shows 1/3 octave sound pressure level, centre frequency 2 kHz, as a function of the microphone height above the reflecting plane, when the test ob-

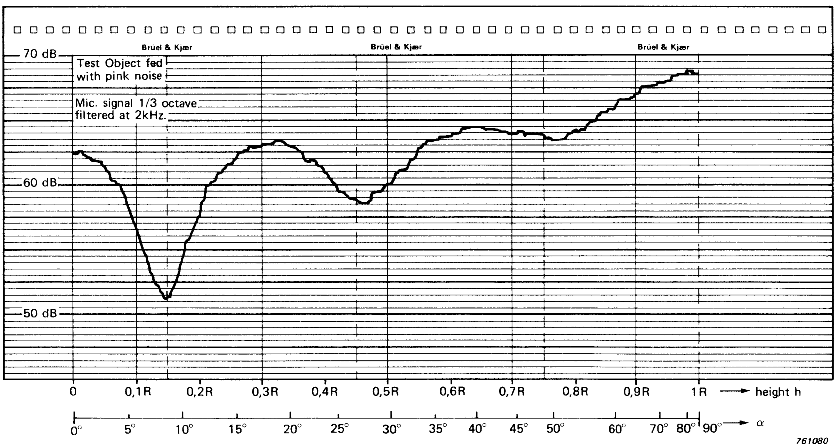


Fig.7. Sound pressure level (1/3 octave, centre freq. 2 kHz) as a function of height and angular position of the microphone relative to the reflecting plane. Test object fed with Pink Noise

ject was fed with pink noise. Also the angular position α of the swing arm, relative to the reflecting plane, is shown on the abscissa. The values from this curve are transferred to Fig.8 which illustrates the directional characteristics in the vertical plane.

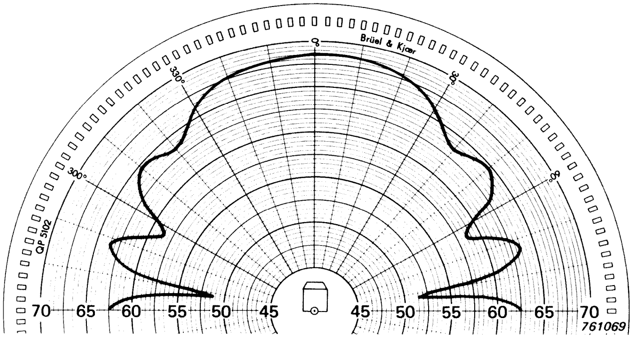


Fig.8. Directional characteristics of the test object in the vertical plane

Figs.7 and 8 reveal sharp minimums in the characteristics due to the interference phenomenon as explained in Appendix A. From Fig.7 the sound pressure level at different heights can be read off to determine the sound power levels obtained by different methods. The results are shown in Table 1.

Measurement Method	Sound Power Level dB	
	for pink noise input	for sinusoidal signal at 2 kHz input
Method 1. 10 fixed microphones at 4 heights	72,5	84 dB
Method 2. 5 Horizontal circular paths	73,4	86,3 dB
Method 2. 10 Horizontal circular paths	73,7	86,4 dB
Microphone sweep in vertical plane	73,8	86,5 dB

Table 1

It can be seen that the value obtained by Method 1 with 10 fixed positions is slightly lower than the other methods. A difference of 17 dB is found between the maximum and minimum values of sound pressure, wherefore more measurement positions are required for this method according to the ISO standard.

When the test object is fed with a sinusoidal signal at 2 kHz, the curve obtained is shown in Fig.9, while the results evaluated are again given in Table 1. From the figure the dips in the curve are seen to be more predominant and a difference of 22 dB between maximum and minimum value is found.

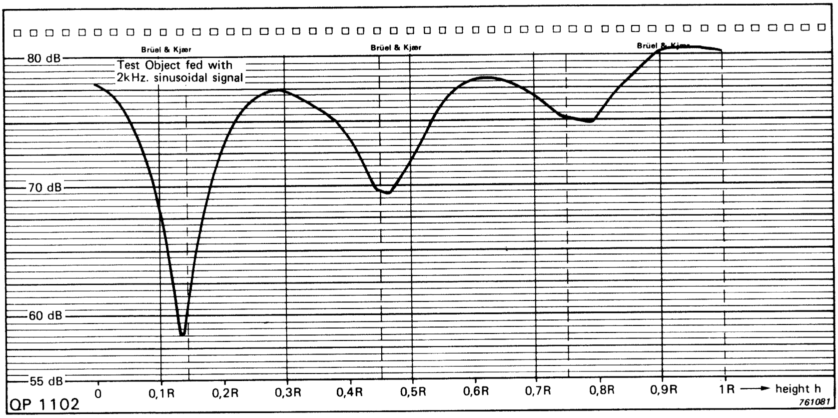
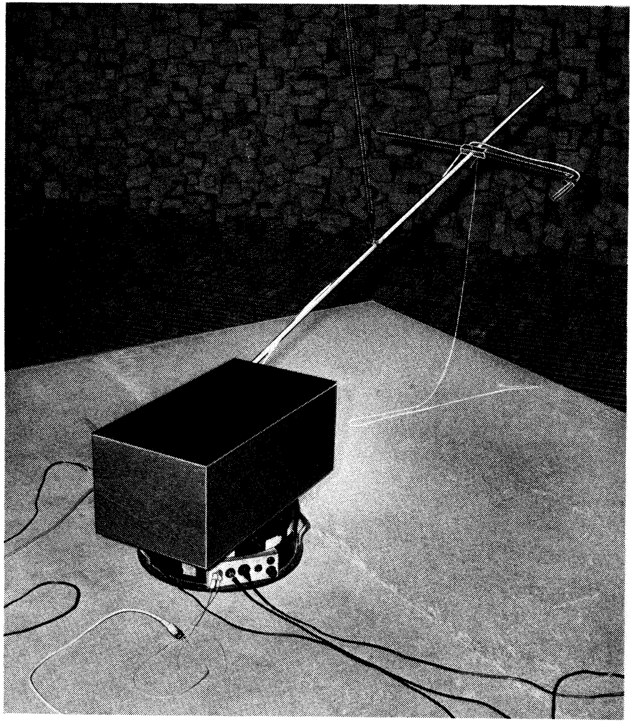


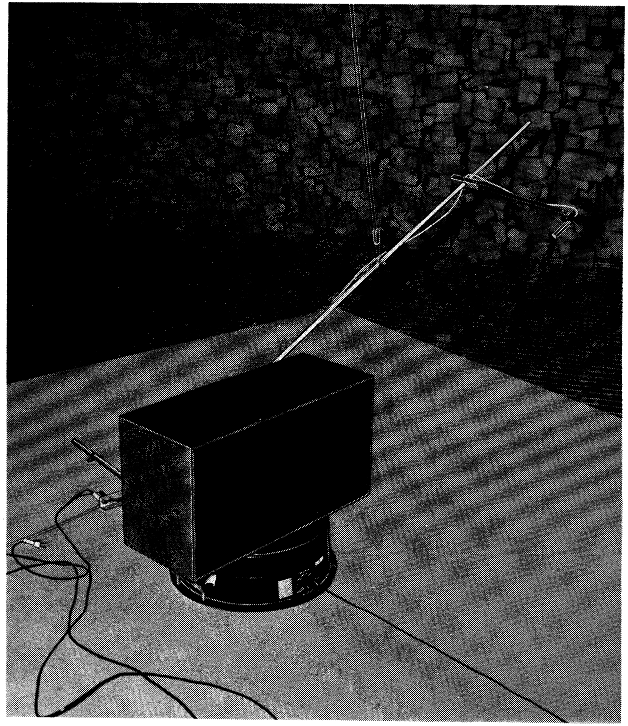
Fig.9. Sound pressure level (1/3 octave, centre freq. 2 kHz) as a function of height of the microphone. Test object fed with 2 kHz sinusoidal signal

b) Measurement object with non-symmetrical directional characteristics

The measurement object used was a conventional 3 way loudspeaker system shown in Fig.10. Pink noise was fed to the loudspeaker and the output was filtered with a 1/3 octave filter with centre frequency 2 kHz. The frequency of 2 kHz was chosen on account of the pronounced directivity of the loudspeaker at this frequency and also because of the tighter tolerances required by the standard for measurement of sound power around this frequency. Sound power was evaluated for two cases of the loudspeaker system:

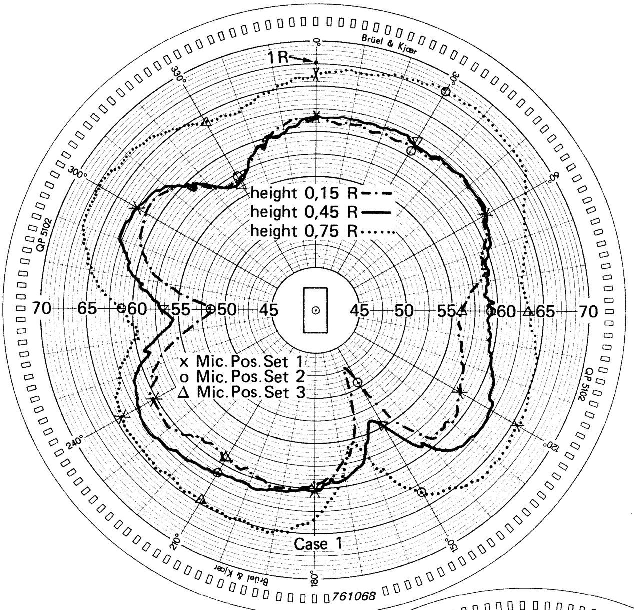


Case 1



Case 2

Fig.10. Measuring arrangement of test object with non-symmetrical directional characteristics



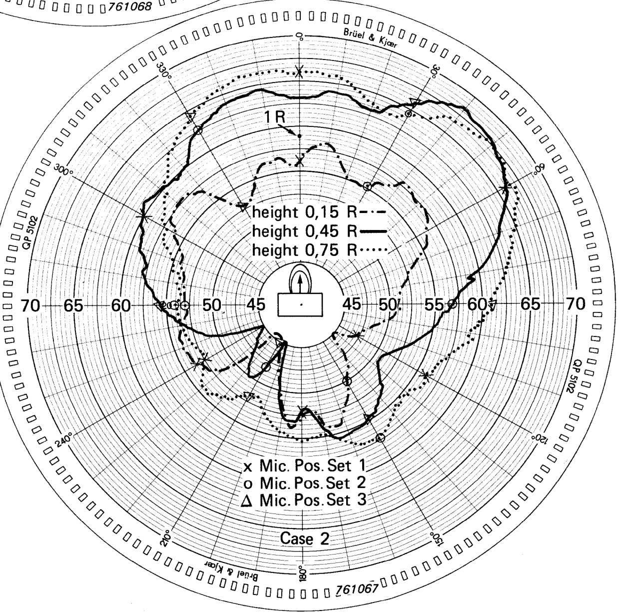


Fig. 11. Directional characteristics of the loudspeaker in the horizontal planes at the three heights suggested for Method 1.

Case 1: Loudspeaker facing upwards towards the vertex of the hemisphere, see Fig.10.

Case 2: Loudspeaker facing sideways.

Fig.11 shows the directional characteristics of the loudspeaker in the horizontal plane for the 2 cases. The heights of these planes correspond to the heights of the microphone positions above the reflecting plane as suggested by method 1. Since the *set* of the microphone positions can be chosen arbitrarily relative to the angular position of the loudspeaker, the sound power levels were evaluated for 3 different sets of microphone positions (marked x o Δ in Fig.11) around the source for both the cases. The results are shown in Table 2.

Measurement Method	Case 1 Loudspeaker facing upwards	Case 2 Loudspeaker facing sideways
Method 1. 10 fixed microphone positions. Set 1	73,7	70,4
10 fixed microphone positions. Set 2	73,1	68,4
10 fixed microphone positions. Set 3	72,5	69,4
Method 2. Five circular paths	72,4	72,9
"Spiral" Method	72,6	72,5

Table 2

Also given in Table 2 are the sound power levels evaluated by method 2 (averaging over five circular arcs) as well as those obtained by the "spiral" method, in which case the loudspeaker was rotated 8 times during the microphone traverse. For illustration purposes, the curve obtained on the level recorder during the "spiral" traverse of the microphone for Case 1 is shown in Fig.12. The value of (72,6—10)dB is in fact the mean value of this curve.

Discussion of Results

It can be seen from the results that the agreement between the values obtained by Method 2 and the "spiral" method is excellent. However, the values obtained by Method 1 are generally higher for Case 1 and lower for Case 2 than those obtained by the other two methods. To investigate the reason for this, the directional characteristics of the loudspeaker in one vertical plane (corresponding to 0° in Fig.11) were obtained for both the cases and are shown in Fig.13. The loudspeaker

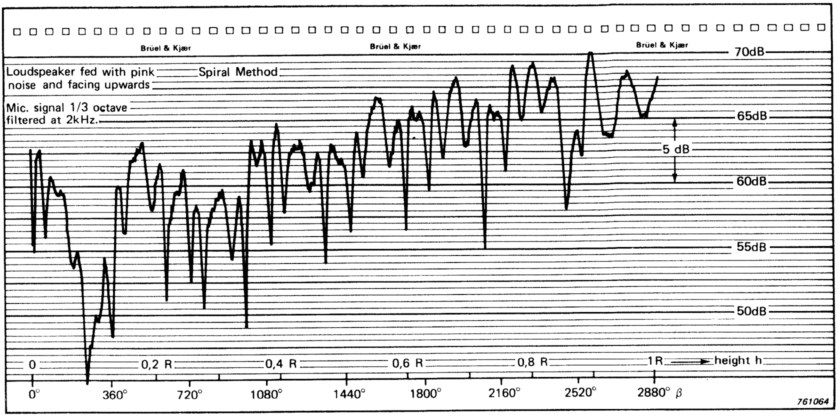


Fig. 12. Sound pressure level as a function of microphone height for the "spiral" method

was fed with pink noise and the microphone signal was 1/3 octave filtered at 2 kHz.

It has been shown in Appendix A that the minima in these curves occur at heights given by

$$h_n \approx n\lambda \frac{R}{4\,b}$$

where λ = the wavelength

R = the radius of the hemisphere

b = the height of the acoustic centre above the reflecting plane

n = 1, 3, 5....

For Case 2 the height of the acoustic centre of the loudspeaker above the reflecting plane is approximately 34 cms. Therefore for 2 kHz centre frequency the heights at which minima occur are

$$h_n = n \frac{345}{2000} \times \frac{R}{4 \times 0{,}34}$$

$$h_n = n\; 0{,}126\ R$$

$$\therefore\ h_n = 0{,}126\ R,\ \ 0{,}38\ R,\ \ 0{,}63\ R.$$

From Fig.13 it can be seen that the first two minima agree fairly well with the calculated values.

The reason why method 1 gives lower values for Case 2 than the values obtained by the spiral method is because the sound pressure levels are around the minimum values at the microphone heights of 0,15R, 0,45R, 0,75R and 1R suggested by the standard. For Case 1, on the other hand, the values obtained by method 1 are higher because the sound pressure levels at the heights suggested by the standard are away from the minima, as the acoustic centre of the source is higher in Case 1 than in Case 2.

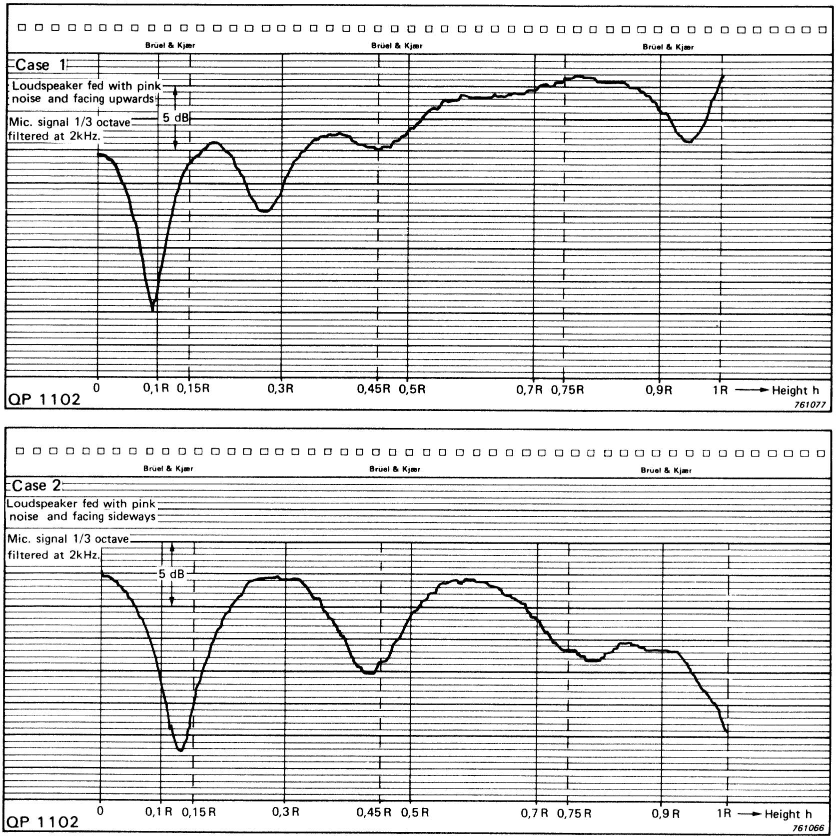


Fig.13. Sound pressure level as a function of microphone height in the vertical plane corresponding to 0° in Fig.11

From the equation given above it can be seen that for a particular height b of the acoustic centre above the reflecting plane, there will always be a frequency where the minima in the curves occur at the heights suggested by the standard.

In other words, if the measurement was carried out at a lower frequency (1250 Hz) for Case 1 the minima would have fallen at the heights suggested by the standard assuming the acoustic centre of the loudspeaker was at the same position.

Conclusion

From the discussion above, it would seem that the choice of the microphone positions for method 1 are rather unfortunate in the sense that the heights are at multiples of 1, 3, 5, the same factors at which the direct and reflected waves are in antiphase. The same would apply for method 2 where the circular areas in the horizontal plane are at heights which are again at multiples of 1, 3, 5, etc.

The method suggested in this article, with a microphone traverse at a constant vertical speed is easy to realize in practice and is considerably time saving and accurate. For a measurement object with rotational symmetry of directional characteristics, a single sweep is required. For test objects with non-symmetrical directional characteristics, a turntable could be used to rotate the test object while traversing the microphone. Alternatively 8 microphone traverses at equal increments of azimuthal angle around the source could be made. Measurement time can be further reduced by replacing the Noise Level Analyzer Type 4426 by a Digital Frequency Analyzer Type 2131, whereby all the frequency bands could be analyzed simultaneously in real time.

References

I.S.O.	Draft International Standard 3745
BERANEK, L.	Noise Reduction. McGraw-Hill
WELLS, R. J.	Apparatus Noise Measurement, AIEE, Power Apparatus and Systems, December, 1955

APPENDIX A

Mirror Effect of Sound Source

Fig.A1 shows the acoustic centre of the sound source S at a height b above the reflecting plane and the image source at I.

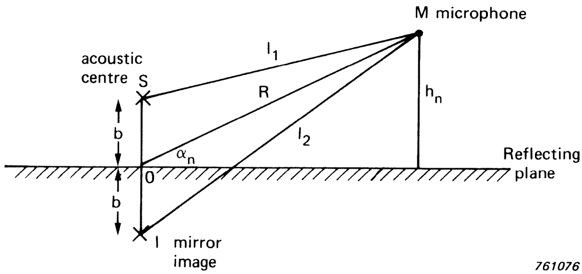


Fig.A1. Schematic diagram showing acoustic centre, position of its mirror image and microphone position

When $l_2 - l_1 = n\,\lambda/2$. $\quad n = 1, 3, 5, 7....$

the sound pressure level at the microphone position becomes minimum. The corresponding height of the microphone h_n above the reflecting plane is sought for.

From

$$\frac{n\lambda}{2} = \sqrt{R^2 \cos^2 \alpha_n + (R \sin \alpha_n + b)^2} - \sqrt{R^2 \cos^2 \alpha_n + (R \sin \alpha_n - b)^2}$$

it can be shown that

$$h_n = R \sin \alpha_n = n\lambda \frac{R}{4b} \sqrt{1 + \left(\frac{b}{R}\right)^2 - \left(\frac{n\lambda}{4R}\right)^2}$$

i.e. $$h_n = R \sin \alpha_n \approx n\lambda \frac{R}{4b} \quad \text{for} \quad R \gg b \text{ and } R \gg \frac{n\lambda}{4}$$

When $\lambda = 0{,}6\,b$

for $n = 1$ $\quad h_1 = 1 \times 0{,}6\,b \times \dfrac{R}{4b} = 0{,}15\ R$

$n = 3$ $\quad h_3 = 3 \times 0{,}6\,b \times \dfrac{R}{4b} = 0{,}45\ R$

$n = 5$ $\quad h_5 = 5 \times 0{,}6\,b \times \dfrac{R}{4b} = 0{,}75\ R$

It can thus be seen that at a particular frequency f_1 for which $\lambda = 0{,}6\,b$, three minima occur in the sound pressure level curve at exactly the heights given for method 1 in Draft ISO 3745. The sound power level evaluated will, therefore, generally have a lower value at this frequency.

APPENDIX B

Vertical Speed of Microphone Traverse

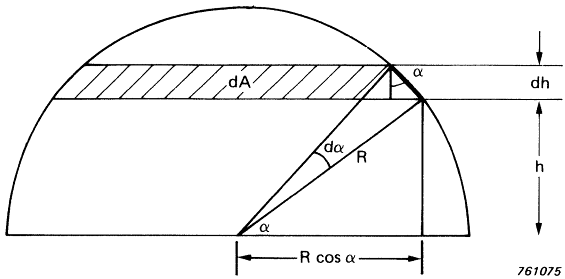


Fig.B1. Traverse of microphone over hemisphere

For correct measurement of sound power the microphone traverse must be such, that equal areas are represented per unit of time, i.e.

dA/dt must be constant

From Fig.B1

$$dA = 2\pi R \cos\alpha \; Rd\alpha = 2\pi R \cos\alpha \; \frac{dh}{\cos\alpha} = 2\pi R \; dh$$

Division by dt gives

$$\frac{dA}{dt} = 2\pi R \frac{dh}{dt} \qquad (1)$$

For the microphone to represent equal areas per unit of time, it can be seen that dh/dt must be constant. Fig.B2 shows schematically the swing arm OM being pulled up by a string BC fixed at a point B, a distance a above the reflecting plane. The string is pulled up at a constant speed V m/s. At time $t = 0$ the swing arm is in the horizontal position and in the vertical position at $t = T_s$

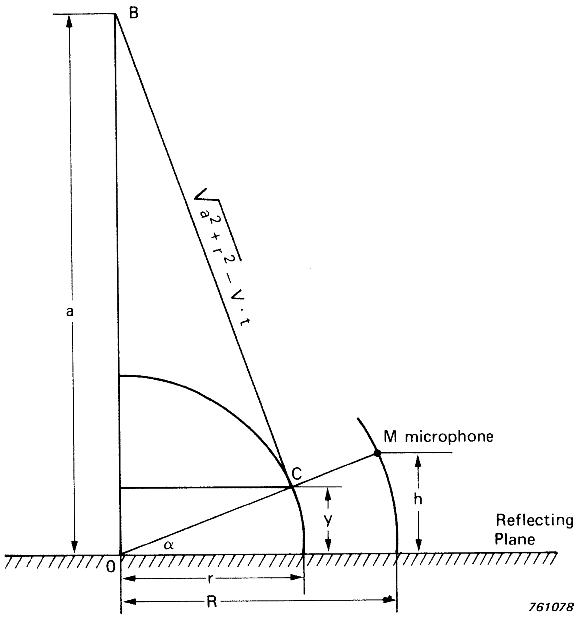


Fig.B2. String BC lifting the swing arm OM

From the diagram it can be seen that at time t

$$\left(\sqrt{a^2 + r^2} - Vt\right)^2 = (a - y)^2 + (r \cos \alpha)^2$$

and

$$r^2 = y^2 + (r \cos \alpha)^2$$

from which we obtain

$$y = \frac{V\sqrt{a^2 + r^2}}{a} \; t - \frac{V^2}{2a} \; t^2$$

Since

$$\frac{h}{R} = \frac{y}{r}$$

$$h = \frac{R}{r} y = \frac{R\,V\sqrt{a^2 + r^2}}{r\,a}\; t - \frac{RV^2}{r2a}\; t^2$$

$$\frac{dh}{dt} = \frac{R\,V\sqrt{a^2 + r^2}}{ra} - \frac{R\,V^2}{ra}\; t \qquad (2)$$

When the swing arm is in the vertical position

$$a = r + \sqrt{a^2 + r^2} - V\,T_s$$

or

$$V = \frac{\sqrt{a^2 + r^2} + (r - a)}{T_s} \qquad (3)$$

Choosing $n = a/r$ and substituting equation (3) in (2) and then in (1) we obtain

$$\frac{dA}{dt} = \frac{2\pi R^2}{T_s} \left(\frac{(1 - n)\sqrt{n^2 + 1} + n^2 + 1}{n} - \frac{(1 - n)\sqrt{n^2 + 1} + n^2 + 1 - n}{n}\, 2\frac{t}{T_s} \right)$$

When $n \to \infty$ i.e. when the fixed point B of the string is brought infinitely high over the reflecting plane, the quantity in the parenthesis becomes 1, indicating a constant rate of change of area. For a finite value of n the rate of change of area varies linearly as shown in Fig.B3, where a value of $n = 5$ is used.

Since dA/dt is not constant, an error occurs in the measurement of sound power, the magnitude of which depends on the value of n and the directional characteristics of the sound source. Theoretically the largest error would occur when a sound source emits all of its power in a narrow ray either vertically upwards or horizontally in which case it would be + 0,5 dB and —0,5 dB respectively for $n = 5$.

When the directional characteristics of the sound source in the vertical plane is known, the error caused on account of dA/dt not being constant, can be evaluated. This was carried out for the sound source shown in Fig.7 and was found to be 0,18 dB.

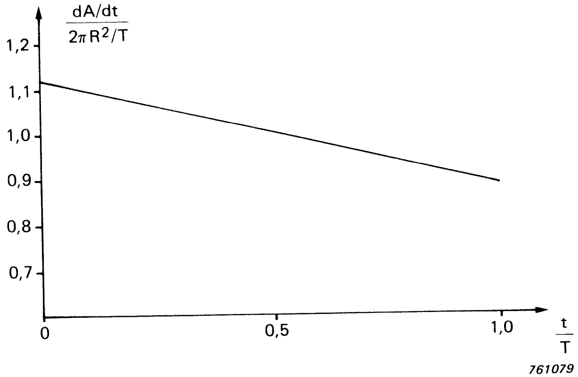


Fig.B3. Relative change in dA/dt as a function of time t for n = a/r = 5

It is, however, possible to compensate for this error by allowing the velocity of the string to be time dependent instead of constant. It can be shown that for dA/dt to be constant the velocity of the string must vary in time according to the equation

$$V(t) = \frac{r}{T_s} \frac{n}{\sqrt{n^2 + 1 - 2n\frac{t}{T_s}}}$$

The velocity of the string can be made to vary approximately as the above function by wrapping it around a conical axle instead of a cylindrical one shown in Fig.4. The error can then be reduced from a maximum of 0,5 to 0,06 dB for n = 5.

Measurement of Sound Absorption of rooms using Reference Sound Source

by

O. Kramář *

ABSTRACT

The sound absorption of rooms, measured with the aid of a reference sound source, is found to be different from that determined by reverberation time measurements or using the sound impulse integration method. The sound absorption values obtained using the reference sound source lead to better agreement between the semi-reverberant room and free field sound power measurements of sound sources, than using the absorption values obtained by reverberation time measurements.

SOMMAIRE

L'absorption sonore des salles mesurée à l'aide d'une source de bruit de référence s'avère différente de celle déterminée par les mesures de temps de réverbération ou en utilisant la méthode de l'impulsion sonore intégrée. Les valeurs de l'absorption sonore obtenues avec la source de bruit de référence conduisent à un meilleur accord entre les mesures acoustiques de puissance acoustique en salle semi-réverbérante et en champ libre que les valeurs de l'absorption obtenues par la mesure du temps de réverbération.

ZUSAMMENFASSUNG

Die Schallabsorption von Räumen, die mithilfe einer Bezugsschallquelle gemessen wird, unterscheidet sich von den Werten, die man aus den Messungen der Nachhallzeit oder nach der Methode von der Integration der Schallimpulse gewinnt. Die Werte für die Schallabsorption, die man mithilfe einer Bezugsschallquelle bekommt, stimmen besser mit Schalleistungsmessungen von Schallquellen in halb-nachhallenden Räumen und im Freifeld überein als mit den Absorptionswerten aufgrund von Nachhallzeitmessungen.

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Introduction

One of the data used in the evaluation of sound power measurements or the sound field in reverberant rooms is the sound absorption. This value is normally evaluated from a measurement of the reverberation time determined either from a sound pressure level decay, or by using the method of sound impulse integration, known as the Schroeder—Kuttruff method. For both of these methods the response of the room is measured for transient or generally nonstationary character of the sound field in the room.

Another means of determining the sound absorption of a room is by the use of a reference sound source, the sound power of which is determined by calibration in a free field. The calibrated sound source is now used to excite a stationary sound field in the room of interest, where the sound pressure level and sound power level are related by the following expression:

$$L_p = L_P + 10 \log \left(4 \frac{S - A}{S \times A} + \frac{Q(\theta, \phi)}{4 \pi r^2} \right)$$

where

L_p = is the mean sound pressure level in the room at a distance r, dB

L_P = the sound power level of the sound source dB

S = surface area of enclosing walls (m^2)

A = total absorption (m^2)

$Q(\theta, \phi)$ = directivity factor in (θ, ϕ) direction

r = distance between measuring point and sound source (m)

When measurements are performed at a reasonably large distance from the sound source where the reverberant field is predominant, the second term in the bracket can be neglected and the absorption in the room can be evaluated from the simplified formula:

$$A = \frac{4 \times S}{4 + S\, 10^{(L_p - L_P)/10}}$$

Description of Instrumentation

To investigate to what extent calibrated sound sources could be used for determination of the sound absorption of rooms, three sound sources were used for the experiments; an aerodynamic sound source, a vacuum cleaner and a mechanical sound exciter.

The aerodynamic sound source was designed in the Technical Acoustic Department of the National Research Institute for Machine Design in Běchovice in 1972. This apparatus operates on the same principle and has similar features as the Brüel & Kjær Reference Sound Source Type 4204. The sound power levels emitted in each octave band are shown in Fig.1. They were determined according to ISO Draft Proposal 3745 with measurements on 10 circles 1 meter from the machine surface in free field conditions over a reflecting plane.

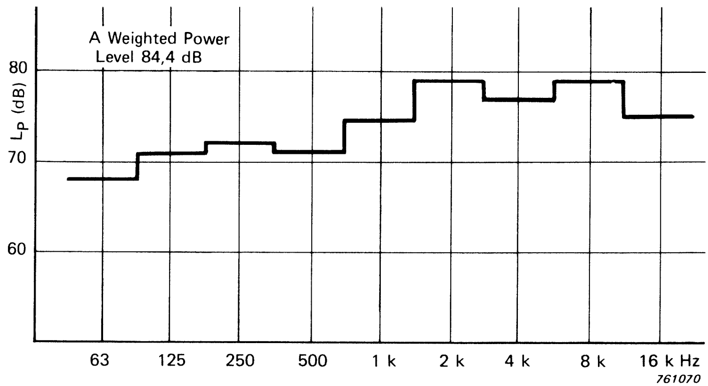


Fig. 1. Octave Band Sound Power Levels in dB ($P_0 = 10^{-12}$ W) of the aerodynamic sound source measured in a free field over a reflecting plane

The directivity index of the sound source was measured in the vertical plane in individual octave bands and is given in Table 1. In the frequency range of interest the maximum deviations are within —4,8 and + 3,5 dB. The directional characteristics in the horizontal plane are not given in detail, as the directivity index is within ± 0,2 dB in all frequency bands.

The reference sound source was designed to operate at 220 V, 50 Hz power supply. To test the stability of the sound source, its sound power was measured for 180, 200, 220 and 240 Volts power supply and was found to vary 0,1 dB for a 10% deviation of the supply voltage.

[Hz] \ Θ °	0	10	20	30	40	50	60	70	80	90
31,5	+ 3	+ 2,7	+ 1,9	+ 1	–0,5	+ 1,8	–3,2	–5,7	–7,3	–8,6
63	+ 3,2	+ 2,8	+ 1,8	+ 0,6	–0,3	–0,2	–3	–4	–4,2	–5,7
125	+ 2,6	+ 2,1	+ 1,5	+ 1	+ 0,2	–1	–1,8	–2,4	–3,1	–3,8
250	+ 2,6	+ 2,2	+ 1	+ 0,4	–0,4	–1,1	–1,7	–1,7	–1,9	–2,1
500	+ 3,5	+ 2,4	+ 0,6	–1,2	–1,7	–2	–1,6	–1,3	–1,2	–1,2
1000	+ 2,5	–1,4	–2,3	–2,6	–1,2	–0,7	+ 0,1	+ 1,1	+ 1,5	+ 1,3
2000	+ 1,9	–1,1	+ 2	+ 0,1	–1,7	–1,1	–0,6	–0,5	+ 0,2	–0,3
4000	–1,2	+ 3,5	+ 2,4	+ 1,4	+ 0,7	–0,4	–1,5	–2,9	–4,7	–4,4
8000	+ 2,3	+ 2,6	+ 1,8	+ 1,1	+ 0,3	–0,6	–1,7	–3	–4,1	–4,8
16000	+ 1,8	+ 1,8	+ 1,9	+ 1,6	+ 1	–0,3	–1,6	–3,2	–4,3	–4,2

Table 1. Directivity index of the aerodynamic reference source in a vertical plane

Table 2 stipulates the sound power levels emitted by the vacuum cleaner and the mechanical sound exciter. They were also measured in a free field over a reflecting plane according to ISO Draft Proposal 3745. However, the directional characteristics and stability of these sources were not investigated as they were considered only for auxiliary purposes.

For sound absorption measurements, six rooms were investigated differing in volume and acoustic treatment (furniture, equipment) under the

Device	Sound power levels in dB in octave bands							
	63	125	250	500	1 K	2 K	4 K	8 K
vacuum cleaner	55,1	64,8	69,8	71,0	72,8	78,5	80,4	76,1
mechanical exciter		67,8	76,2	82,1	83,7	82,3	79,4	72,8

Table 2. Sound power levels in dB ($P_0 = 10^{-12}$ W) of the vacuum cleaner and mechanical exciter

Room No.	Dimensions (m)	Volume (m^3)	Surface (m^2)	Description
1	not rectangular	80,25	112,9	Reverberation room
2	5,1 x 4,35 x 3,4	75,5	109	Room with absorbing ceiling
3	6,95 x 3,5 x 4,75	116	147,9	Reverberation room
4	6,95 x 3,5 x 4,75	116	147,9	Special reverberation test room
5	6,75 x 4,75 x 3,75	120	150,3	Reverberation room
6	15 x 9,2 x 7,5	1035	639	Factory hall

Table 3. Characteristics of the six rooms investigated

same meteorological conditions. The dimensions of the rooms are given in Table 3. The instrumentation required for these measurements is shown in Fig.2 and could be either analogue or digital.

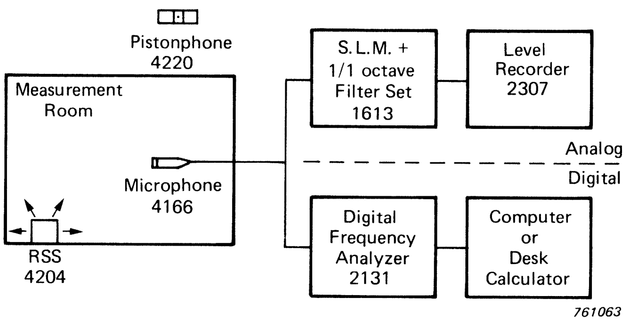


Fig.2. Instrumentation required for absorption measurements using a reference sound source

Measurement Results

For this investigation four sets of experiments were carried out:

a) The sound absorption of rooms No.4 and No.6 were determined using the three calibrated sound sources.

b) The influence of the number of source and microphone positions on accuracy of absorption measurements was studied in rooms Nos.1 and 5.

c) The sound absorption of rooms Nos.1-6 were measured

1) by reverberation time measurements
2) by sound impulse integration technique
3) by the use of the aerodynamic reference sound source.

d) The difference in the sound power levels of the three sources measured under free field conditions and measured in rooms Nos.4 and 6 using absorption values obtained 1) from reverberation time measurements and 2) with the use of the aerodynamic reference sound source, were determined.

a) The sound absorption of rooms Nos.4 and 6 were determined for each octave band using the three calibrated sound sources and are shown in Fig.3. In room No.4 the sound sources were located in one position and the microphone in five positions, while in room No.6 the sound sources were located in two positions and the microphone in five positions for each position of the sound source. Due to the background noise not all the data are given for room No.6 for the frequency bands of 63 and 125 Hz.

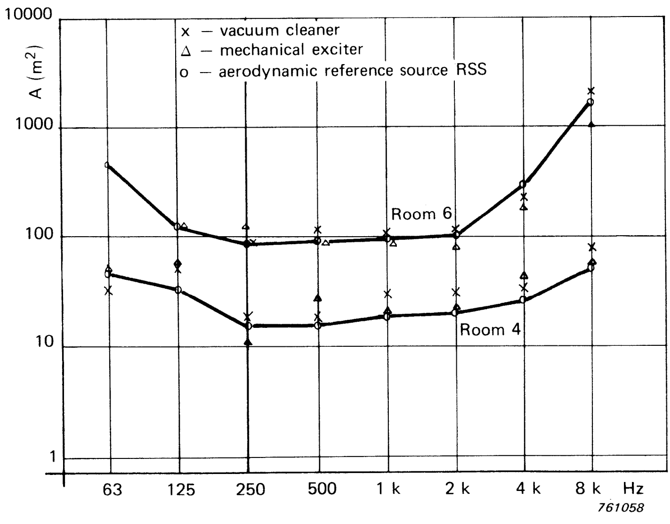
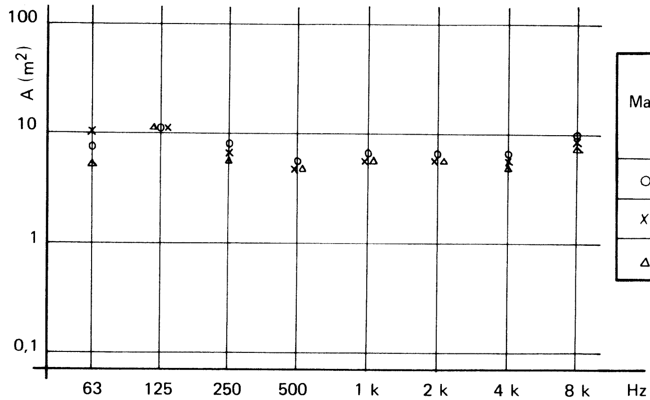


Fig.3. Sound absorption A (m²) of rooms Nos.4 and 6 measured with various calibrated sound sources

From the figure it can be seen that the sound absorption measured by means of the different sources of stationary sound do not vary significantly and have very similar character. The absorption determined with the use of the vacuum cleaner and the aerodynamic reference sound source are almost identical, though wider deviations are apparent for the mechanical exciter. This was because the mechanical exciter was found to be less stable than the other sound sources. It was, however, calibrated under free field conditions each time before its use in reverberant rooms.

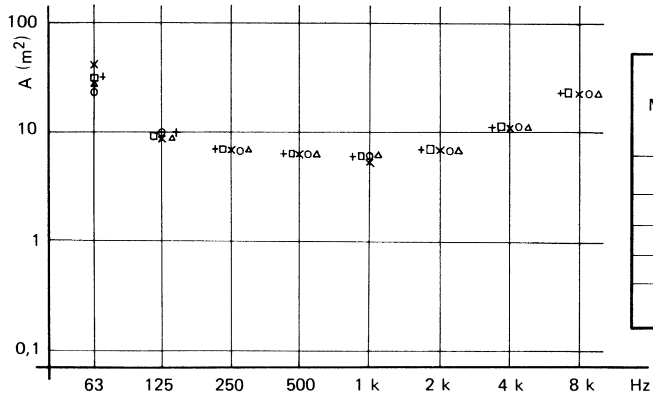
b) The influence of the number of sound source and microphone positions was investigated in rooms Nos. 1 and 5. The results are



Mark	Number of positions	
	Microphone	RSS
o	4	1
x	4	2
Δ	2	2

Room 1

760259



Mark	Number of positions	
	Microphone	RSS
x	1	4
o	4	1
Δ	2	2
□	3	3
+	4	4

Room 5

761060

Fig. 4. Influence of the number of microphone and sound source positions on the absorption of rooms Nos. 1 and 5

shown in Fig.4 where the dependence on the number of positions is seen to be greater for room No.1 than for room No.5. For higher accuracy the number of positions should ideally be decided from standard deviations of the results.

c) The sound absorption of the six rooms was now measured by three methods and the results are given in Fig.5. Curve 1 indicates results obtained by reverberation time measurements while curve 2 indicates results obtained with the use of the Brüel & Kjær Reverberation Processor Type 4422 utilizing the sound impulse integration technique for determining the reverberation time. Curve 3 gives results obtained with the use of the calibrated aerodynamic sound source. The number and situation of the measurement positions was identical in the rooms for all sound sources.

From the curves it can be seen that the absorption values obtained by different methods can vary significantly. The values obtained by means of the reverberation decay rate methods are found generally to be lower than those obtained with the integrated impulse method which again are lower than those obtained with the use of the aerodynamic reference source. The differences are more significant at the lower frequencies. The reason for these differences could be due to the fact that reverberation time measurements are carried out during a transient sound field while with the reference sound source the measurements are made during excitation of a stationary sound field.

d) To investigate whether the absorption values obtained by reverberation time measurements or those obtained with the use of the reference sound source were more suitable for determining the sound power levels of sound sources, the following investigation was carried out. The sound power levels of the three sources measured in a free field over a reflecting plane were compared to those obtained in rooms Nos.4 and 6 using the absorption values determined in section c) from reverberation time measurements and from the use of the reference sound source.

i.e. $\delta = L_{PFF} - L_{PRR}$

where L_{PFF} = sound power level measured in a free field, dB
L_{PRR} = sound power level determined in rooms Nos.4 and 6 using absorption values obtained by the two methods in section c), dB

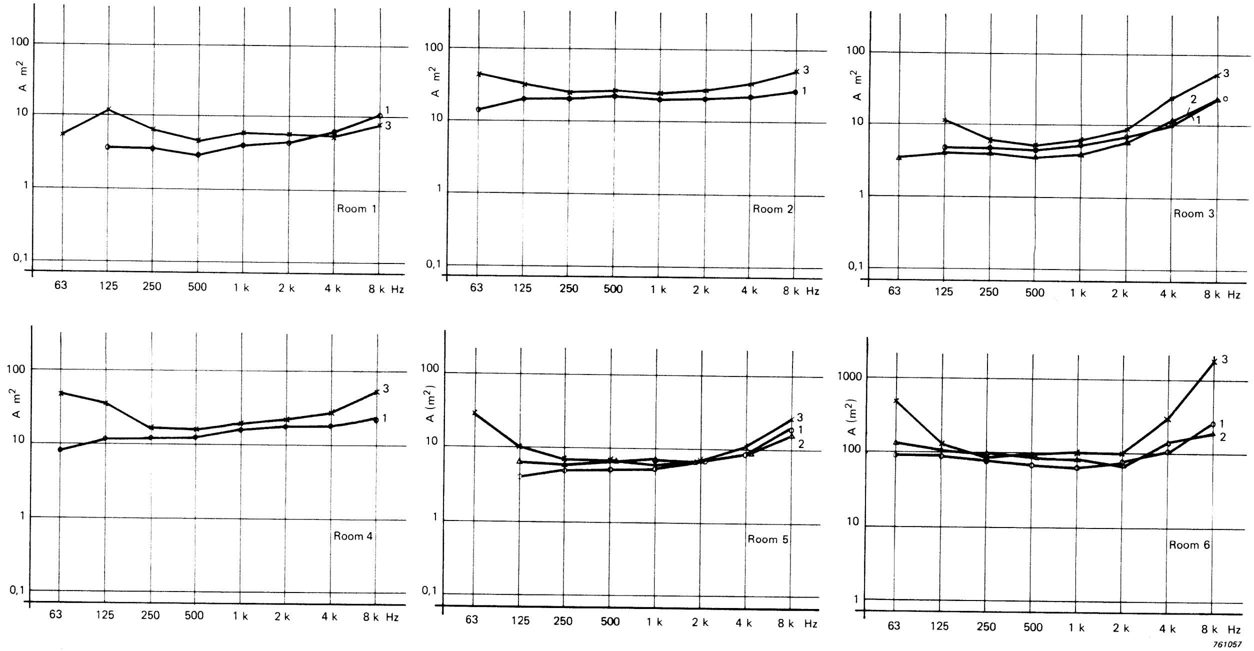


Fig.5. *Sound absorption of rooms Nos.1-6*
Curve 1 — Reverberation Time Method
Curve 2 — Sound Impulse Integration Method
Curve 3 — With the use of aerodynamic reference source

The difference in sound power levels δ is shown for the three sound sources for each octave band for rooms No.4 and No.6 in Fig.6. It can be seen that δ is significantly lower for cases where the absorption values obtained with the use of the reference sound source are used than those obtained using reverberation time measurements.

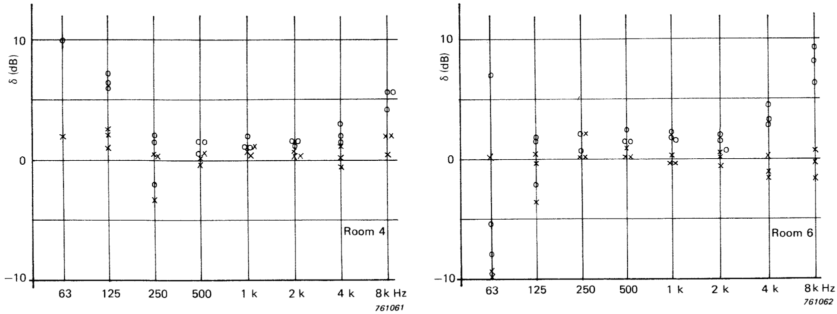


Fig.6. Difference in sound power levels δ measured in free field and in rooms Nos.4 and 6
o — absorption values determined using reverberation time method
x — absorption values determined using aerodynamic reference source

Conclusion

The results of this investigation show that the absorption values obtained with the use of stable calibrated sound sources do not vary significantly, though give different values than those obtained by reverberation time measurements. Also, the absorption values determined from a stationary sound field using the reference source leads to better agreement between the reverberant room and free field sound power measurements. Another advantage of using the reference source for absorption measurements is that it can be used in rooms where the background noise is relatively high.

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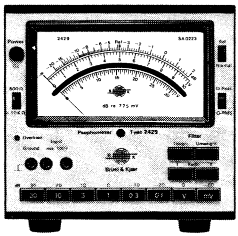
News from the Factory

Psophometer Type 2429

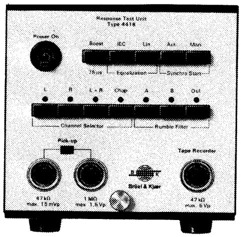


In general, psophometers are used for measurements on telephone circuits and radio broadcasting equipment, to determine the subjective signal-to-noise ratio in communication channels where the final receiver is the human ear. The Psophometer Type 2429 is designed for measurements in accordance with CCITT P53 and CCIR 468—1 standards. It has both Q-peak and Q-RMS detectors and the four weighting networks described in the standards. The amplification of the instrument is calibrated and adjustable in 10 dB steps making it suitable for use in set-ups where a calibrated amplifier with balanced input is needed. For this purpose AC and DC outputs are available.

Among other features of the instrument are a very high overload margin and two balanced inputs, $600\,\Omega$ or $>100\,k\Omega$, both of which are floating. To further minimize the influence of AC voltages the cabinet and signal ground are isolated.

The instrument is also simple to operate. If an illogical attenuator setting is accidentally chosen, indication will be given by the overload detector. With the aid of the built-in reference oscillator, the instrument can be calibrated quickly, since the attenuator logic automatically selects the correct range and returns to the measurement settings after calibration.

Response Test Unit Type 4416



When testing and recording the frequency response of gramophone pick-ups, it is necessary to feed the signal from the cartridges via a response test unit such as the Brüel & Kjær Type 4416. The main function of the instrument is to deemphasize the output signal from pick-ups and give a "start" signal to a Level Recorder for synchronisation between its frequency calibrated paper and the frequency sweep on the test record. The test unit has a built-in electronic switch which allows separation and balance measurements on stereo systems to be made.

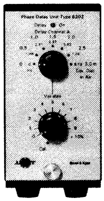
The instrument has three inputs, designed for measurements on dynamic pick-ups, piezoelectric pick-ups and tape recorders, respectively. The pick-up inputs are led after amplification to one of three equalization networks, "IEC", "Boost 75 μs" and "Lin.". "IEC" network is used for unequalized signals using test records cut to the IEC standard. "Boost 75 μs" network is used in connection with B & K test records on account of their special preemphasis, while "Lin." network is used where a preamplifier with built-in equalization is incorporated in the device under test. The third input for tape recorder testing bypasses these equalization networks.

After the equalization networks, the signals are led to an adder and a chopper whereby four different signals can be selected, L, R, L + R and "Chop". The chopper operates at approximately 1 Hz and is suitable for detailed analysis of the stereo balance as a function of frequency.

After passing the adder and chopper the signal may be led to the output for frequency response, crosstalk, balance or other measurements on pick-up cartridges. The signal may also be led to the output via the built-in Rumble Filters A and B which are in accordance with IEC 98 A and DIN 45 539 for measurements of turntable quality.

For automatic swept distortion measurements, an output to the Tracking Frequency Multiplier Type 1901 is available via a noise filter which is necessary for avoiding disturbance of the tracking due to clicks and surface noise.

Phase Delay Unit Type 6202



Errors in the measurement of phase characteristics of loudspeakers would normally occur, if the time delay, due to the time required for the sound to propagate from the loudspeaker to the microphone, is not compensated for. Also in the case of tape recorders, a time delay arises on account of the physical separation of the record and replay heads along the tape. Compensation for these time delays can be made by introducing electronically a similar delay period in the measuring circuit, with the use, for example, of a Phase Delay Unit Type 6202.

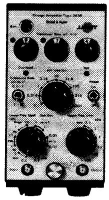
The Delay Unit is connected to the Phase Meter Type 2971 by means of a multicable as it operates on a digital signal. The actual delay occurs in a shift register which has six output tappings to give different delay times, and is controlled by feeding in a clock signal derived from two alternative sources. The internal clock circuit gives calibrated time delays up to 8,73 ms corresponding to equivalent propagation distances between 0 and 3 metres.

Normally, the distance between the loudspeaker and microphone is measured and the corresponding delay time is selected by a multiposition switch on the front panel. Another control knob operates a potentiometer giving a 10% increase in the delay time to permit accurate matching of distance to delay.

The delay times possible with the internal clock are adequate for most near-field measurements on loudspeakers. However, when intermediate values or longer distances have to be used, or when measurements are made on tape recorders where time delays are typically between 80 and 200 ms, an external generator may be used as a clock source. The external clock input can accept voltages between 2 V and 15 V peak-to-peak, sine or square wave, in the frequency range 1 kHz to 1,5 MHz.

The Delay unit also contains a built-in phase correction filter to compensate for the changes in phase that can occur between the input and output of B & K Measuring Amplifiers Types 2606, 2607 and 2608, when used with the Slave Filter Type 2020 in measuring arrangements.

Vibration Pick-up Conditioning Amplifier Type 2635



This conditioning charge preamplifier is a comprehensive portable instrument for general purpose vibration measurements in the field and in the laboratory. A three digit sensitivity adjustment network enables the preamplifier to be conditioned to suit transducer sensitivities between 0,1 and 11 pC/unit facilitating calibration, especially when transducers of different sensitivities are used. The overall gain of the preamplifier can be adjusted to provide a rated output level between 0,1 mV/unit and 1 V/unit in 10 dB steps.

The low frequency limit can be switched either to 1 Hz or 0,1 Hz (—3 dB) when low frequency measurements are carried out with force transducers or B & K "Delta-Shear" accelerometers which exhibit very low temperature transient sensitivities. Integrating networks incorporated in the instrument permit velocity and displacement measurements with low frequency cut-off switchable to either 1 or 10 Hz (—10%) to suppress low frequency noise.

Six switchable upper cut-off frequencies can be chosen between 0,1 kHz and 100 kHz (—10%) with the low pass filter having a 12 dB/octave roll-off. The upper limiting frequency of 200 kHz (—3 dB) is provided for underwater sound measurements using hydrophones.

The preamplifier is equipped with a push-button activated test oscillator that applies a 160 Hz sinusoidal signal to the input. The oscillator level is factory adjusted to give an output level of 1 V RMS (for prescribed sensitivity settings) which facilitates calibration of recording levels on tape recorders.