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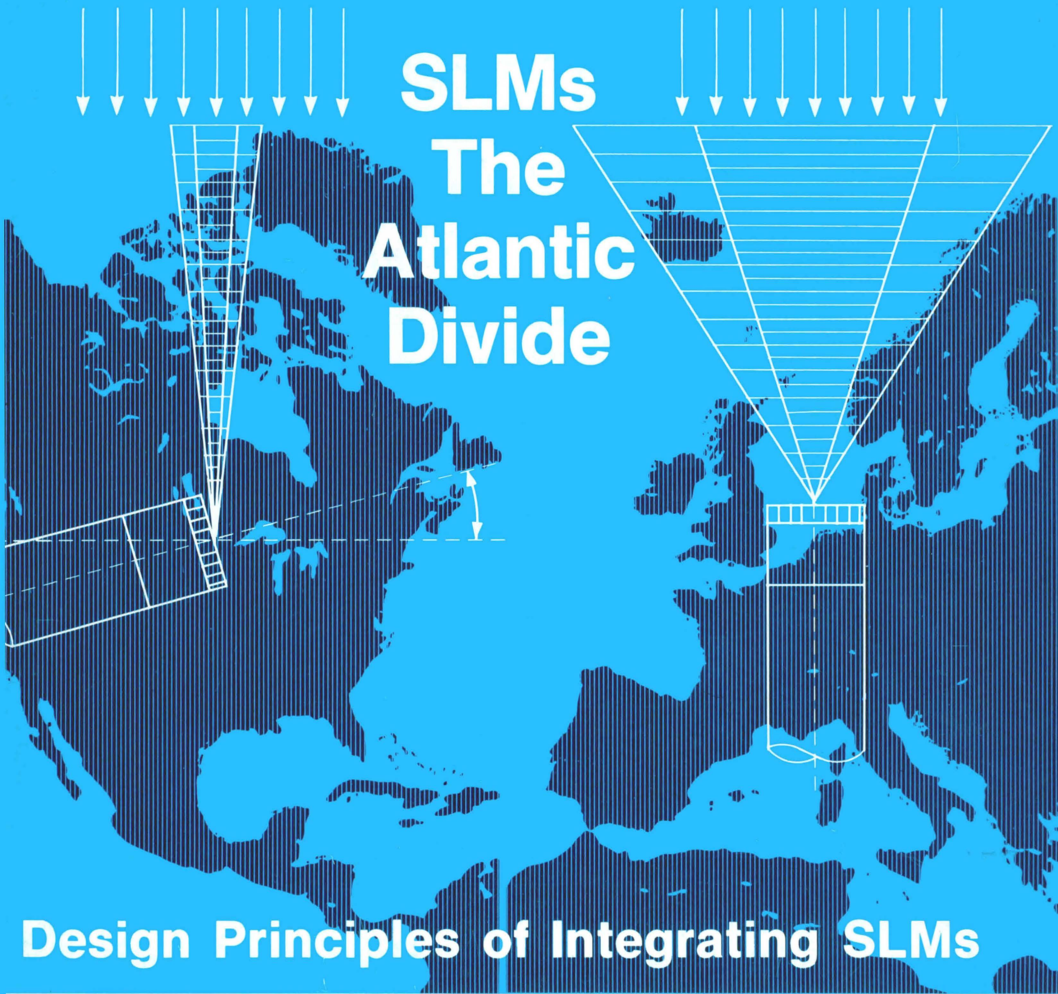

## SLMs The Atlantic Divide

## Design Principles of Integrating SLMs

Brüel & Kjær

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# TECHNICAL REVIEW

No. 4 — 1983

# Contents

# SOUND LEVEL METERS — THE ATLANTIC DIVIDE*

by

*Per V. Brüel, D.Sc.*

## ABSTRACT

The IEC Publication 651 (1979) and the American National Standard ANSI S 1.4 (1983) for sound level meters are completely alike, except for one important point. While the IEC standard requires sensitivity calibration in a reference direction specified by the manufacturer, the ANSI standard prescribes the random incidence sensitivity. Unfortunately, discrepancies can occur between results obtained by the two sound level meters. However, there are well documented reasons why IEC and ANSI have chosen their respective philosophies. There are merits and disadvantages behind both philosophies. It will be shown that the use of a sound level meter made to IEC 651 gives more accurate results for free-field measurements where the direction of sound is known; whereas the sound level meter made according to ANSI S 1.4 is the better choice for measurements in random fields, where the sound arrives from several directions. It is therefore hoped that the IEC and ANSI committees will permit both concepts to be used. As the problem is even more acute with the advent of integrating sound level meters, it is imperative that this hope becomes a reality, so that measurements results achieved on both sides of the Atlantic are in conformity.

## SOMMAIRE

Les normes pour sonomètres CEI 651 (1979) et ANSI S 1.4 (1983) sont entièrement similaires, à l'exception d'un point important. Alors que la norme CEI demande un étalonnage de sensibilité suivant une direction de référence spécifiée par le constructeur, la norme ANSI demande un étalonnage en incidence aléatoire. Malheureusement, des différences peuvent apparaître entre les résultats des deux sonomètres. Cependant, les raisonnements qui ont conduit à choisir des méthodes différentes sont bien étayés; chaque méthode a ses avantages et ses inconvénients. Cet article montre que l'utilisation d'un sonomètre conforme aux normes CEI donne des résultats plus précis pour les mesures en champ libre lorsque la direction du son est connue; alors qu'un sonomètre

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conforme à ANSI S1.4 est un meilleur choix pour les mesures en champ diffus, lorsque le son arrive suivant plusieurs directions. Il est par conséquent souhaité que les comités CEI et ANSI permettent d'utiliser les deux méthodes. Comme ce problème est encore plus critique avec l'apparition des sonomètres intégrateurs, il est impératif que ce souhait devienne réalité afin que les résultats des mesures effectuées des deux côtés de l'Atlantique soient conformes.


## ZUSAMMENFASSUNG

Die IEC-Veröffentlichung 651 (1979) und die nationale amerikanische Norm ANSI S 1.4 (1983) für Schallpegelmesser stimmen, mit Ausnahme eines wesentlichen Punktes überein. Während IEC die Kalibrierung des Übertragungsfaktors in einer bestimmten, vom Hersteller angegebenen Bezugsrichtung verlangt, schreibt ANSI den Übertragungsfaktor in einem Diffusfeld vor. Unglücklicherweise führen Messungen mit diesen zwei Arten von Schallpegelmessern zu unterschiedlichen Ergebnissen. Es gibt jedoch wohlbegründete Ursachen für die Philosophien von sowohl ANSI als auch IEC. Es wird gezeigt, daß eine Schallpegelmessung nach IEC zu genaueren Ergebnissen unter Freifeldbedingungen (d.h. die Schallrichtung ist bekannt) führt; während Schallpegelmessungen nach ANSI S 1.4 besser für Messungen in Diffusfeldern sind, in denen der Schall aus verschiedenen Richtungen kommt. Es bleibt zu hoffen, daß die ANSI und IEC-Ausschüsse die Anwendung beider Konzepte erlauben. Da das Problem bei integrierenden (mittelnden) Schallpegelmessern noch dringlicher ist, ist es notwendig, daß diese Hoffnung zur Realität wird, damit sich Meßergebnisse von beiden Seiten des Atlantiks miteinander vergleichen lassen.


## Introduction

The waters of the Atlantic divide the two controversial schools of thought that prevail with regard to the construction of sound level meters (SLM). In the USA, Canada, Mexico and most of South America, the sound level meters used are built in accordance with American National Standard ANSI S 1.4, Ref. [1]. In Europe, Asia and Japan, the SLM are made in accordance with IEC Publication 651, or the forerunner Publication 179, Ref. [2, 3]. The two standards, for practical purposes, are completely alike – except for one important point. IEC 651 (1979) requires calibration in a reference direction of incidence specified by the manufacturer (generally the 0° incidence is chosen, as it is found to be most convenient when the sound level meter is used in practice). On the other hand, the reference prescribed by ANSI S 1.4 is the random incidence sensitivity, which is the root-mean-square sensitivity to sound arriving with equal probability from all directions.

In a given measurement situation the results obtained using an IEC and an ANSI instrument are generally not the same, or at least the accuracy of the results is different. Since measurement of sound pressure level is rather simple, most do not understand why there should be any difference in the results. The problem is outlined in the following. However, it is not easy to comprehend the problem in its entirety. It will be shown that the SLM, made according to IEC 651, gives more accurate results if the direction of sound is known; whereas the SLM made in accordance with ANSI S 1.4 is the best choice in situations where the sound arrives from many directions, for example, indoors.

**Microphone in a Free Progressive Wave**

Fig.1 illustrates schematically reflections around a microphone when it is placed in a free progressive wave with sound pressure *p*. Figure 1.a shows 0° incidence where the reflections cause an increase of sound pressure level $\Delta p$, as much as 10 dB for certain frequencies. Sound striking an infinitely large reflecting wall will experience a doubling of sound pressure, an increase of 6 dB sound pressure level at the wall surface. The reason the end of a small cylinder can produce an even higher pressure increase is because of diffraction or scattering at the ends of cylinders of finite size. If the microphone has a cavity in front, or is covered with an unsuitable protecting grid, the pressure increase can be even higher. The pressure increase due to reflection is highly dependent on the ratio between the wavelength of the sound and the cylinder diameter, that is, for a particular microphone diameter it is frequency dependent, as shown in Fig.2.

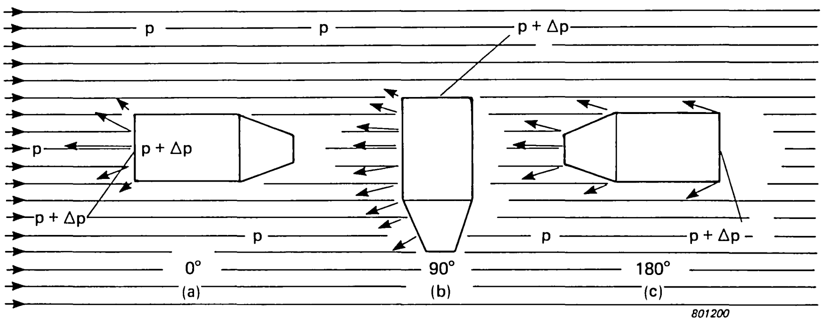



*Fig. 1. Pressure increase at the microphone's diaphragm when it is placed in a free sound field:*
*(a) direction of the sound source perpendicular to the diaphragm (0° incidence),*
*(b) diaphragm at grazing incidence (90°),*
*(c) sound waves arriving from behind (180°)*

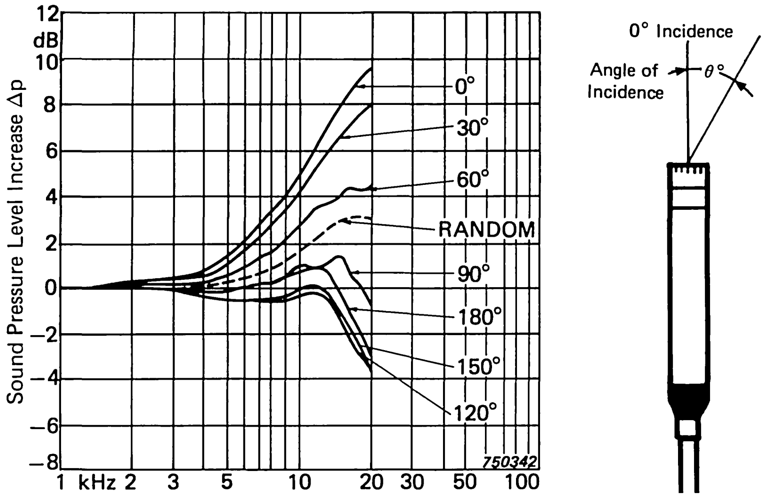



*Fig. 2. Pressure increase level $\Delta p$ as a function of frequency for different angles of incidence shown in Fig.1. The curves are valid for 1/2 inch B & K microphone type 4165*

When the microphone is small compared with the wavelength, the pressure increase is negligible. For a 1/2 inch microphone around 3000 Hz, $\Delta p$ starts to build up and reaches a maximum around 22 kHz. At higher frequencies, the variation of $\Delta p$ is highly irregular and is fortunately outside the usable frequency range of the microphone. We will therefore, for the most part, discuss the problems up to the first maximum of $\Delta p$.

If the microphone is rotated, for example 90° (grazing incidence) as shown in Fig.1.b, the pattern of the reflections changes dramatically. The pressure increase is mainly on the side of the cylinder and there is only a minor variation in the pressure at the diaphragm; in other words, $\Delta p$ is small and shows both positive and negative values at different frequencies (see curve for 90° in Fig.2).

If the microphone is rotated around 180°, the diaphragm is in the "shadow" of the cylinder and $\Delta p$ is mainly negative. It should be emphasized that the pressure increase curves $\Delta p$ shown in Fig.2, have almost no relation to the frequency response of the microphone. By variation of the diaphragm mass and stress, gap between the diaphragm and the back plate, holes in the back plate, and form and size of the internal air volume, the frequency response can be varied in shape over a very large range. The curves of $\Delta p$ in Fig.2 are dependent on only the outer size and shape of the microphone. In other words, microphones of the same size and similar outer geometry always have the same shape of $\Delta p$ curves, depending only on the angle of incidence and frequency. It is therefore necessary to obtain $\Delta p$ curves only once for a certain type of microphone with its protecting grid and not for each individual microphone, as is the case for the frequency response.

It can be seen that if the microphone could be made very small, for example 1,5 – 3 mm in diameter, the pressure increase for varying angles of incidence would be above the interesting frequency range for SLM (10 – 20 kHz). Unfortunately, this is impractical since a very small microphone would have poor sensitivity, making it unable to measure low level sounds, and a small capacity, making low frequency measurements difficult. A miniature microphone has an extremely thin membrane which makes the microphone expensive and mechanically fragile. The thin membrane is also sensitive to chemical erosion, which would shorten the life of the microphone.

The diameter of a microphone for a sound level meter is usually between 10 and 25 mm; such a diameter lies just in the range where there are considerable variations in $\Delta p$ due to reflections in the important frequency regions.

**Microphone Sensitivity**

In as much as a sound level meter is intended to measure the sound pressure $p$ of the free sound field of Fig.1 (that is, before the SLM is brought into the field), the sensitivity of the SLM microphone is defined as:

$$\text{Free-Field Sensitivity } S_F = \frac{\text{output voltage } e}{\text{sound pressure } p}$$

From Fig.2 it can be seen that it is also necessary to state the angle of incidence to define a figure for the sensitivity. The IEC Standard leaves the choice of angle of incidence to the manufacturer; however, the best choice is 0° incidence because:

1. At 0° incidence the change in sensitivity with the angle of incidence is a minimum, permitting very accurate measurements to be made over a wide angle of incidence.

2. The upper end of the useful frequency range can be extended by nearly an octave compared to other incidence angles

3. A higher maximum sound pressure level is measurable.

For other applications of the microphones, for instance in artificial ears, pressure feedback in artificial mouth etc., the so-called:

$$\text{Pressure Sensitivity } S_p = \frac{\text{output voltage } e}{\text{pressure at the diaphragm } (p + \Delta p),}$$

is of great importance.

Thus, the pressure sensitivity is not dependent on the angle of incidence but only on frequency. This curve is normally called the pressure response curve and has to be determined individually for each microphone cartridge. By adding the $\Delta p$ curve to the pressure response curve, the free-field response for the microphone is obtained for that particular direction.

Fig. 3 shows the pressure and free-field response for two different microphones. The two microphones are the same size (1/2 inch) and geometrical form, and consequently have the same $\Delta p$ curves.

The microphone in Fig.3.a is a *pressure microphone* since the pressure response is fairly flat; for example, the sensitivity is almost independent of frequency up to 10 kHz. The free-field response at 0° incidence shows a significant increase in the sensitivity starting around 5 kHz. At grazing incidence (90°) the free-field curve is close to the pressure response curve, since $\Delta p$ at 90° is rather small.

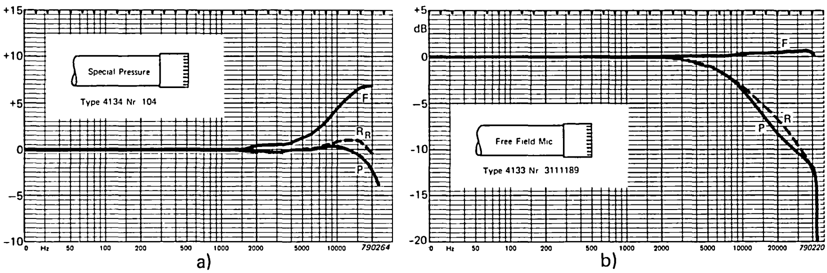



*Fig. 3. Pressure response P and free-field response F for 0° and random incidence for:*
*(a) pressure microphone type 4134 and*
*(b) free-field microphone type 4133*

A pressure microphone is ideal for use in artificial ears where the pressure at the microphone diaphragm is important. The pressure microphone is also a good choice for a SLM made to ANSI S 1.4, as will be shown later.

Over the major part of the usable frequency range, the pressure sensitivity of a condenser type microphone is determined by the stiffness of the diaphragm, which is contributed by the stress in the membrane itself and the stiffness of the internal air volume in the microphone cartridge. This frequency range is shown (range *s*) in Fig.4. At low frequencies a condenser microphone normally has a drop in sensitivity, either due to acoustical leakage in the internal volume, or electrical leakage in the preamplifier (range *l*). At higher frequencies (range *r*), the microphone has its first resonance caused by the mass of the membrane; in addition, the tiny air cushion on both sides of the membrane contributes to the mass and total stiffness. Normally, this resonance can be damped effectively by keeping a very small gap between the membrane and the backplate (electrode), and drilling holes in the backplate at strategic positions. Thus, only minute ripples show up on the response curve in the frequency region *r.* At higher frequencies the movement of the diaphragm is entirely controlled by the mass of the diaphragm and consequently the response curve falls off at 12 dB/octave (range *m*).

It is important to note that the sensitivity of the microphone in range *s* can be changed by altering the stiffness of the system, and in range *l* by blocking both the acoustical and electrical leakage. In the range *r*, large variations can be introduced by changing the damping of the system.

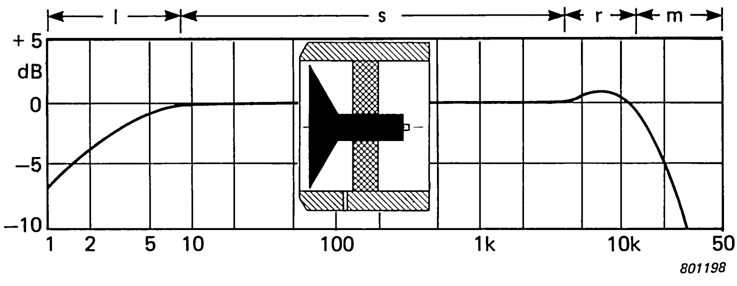



*Fig. 4. Drawing illustrating characteristic frequency bands of a condenser microphone: (l) low frequency with cutoff due to acoustical or electrical leakage, (s) main frequency band – stiffness controlled, (r) region of damped resonance, (m) high frequency region with 12 dB/octave fall-off*

However, only small changes can be achieved in range *m*, since the mass of the diaphragm can only be changed slightly in practice.

Referring back to Fig.3.b, which shows a microphone cartridge similar to that in Fig.3.a, but with high damping in frequency range *r*, it can be seen that the pressure response falls off from around 2 kHz, thus compensating for the increase due to $\Delta p$ at 0° incidence. In other words, this microphone has a flat response curve for sound waves arriving perpendicular to the diaphragm. A microphone of this type is called a free-field microphone. The over-riding advantages of this system are that not only is a flat response obtained for sound waves coming from a known direction, but also that the upper end of the usable frequency range has been extended by about an octave compared with the pressure microphone in Fig.3.a. This extra octave band is only achievable in practice if 0° incidence is used as the reference direction "specified by the manufacturer" of the SLM. Consequently, for most SLMs made to IEC 651, 0° incidence is specified as reference.

**Random Incidence Sensitivity**

As mentioned earlier, ANSI S 1.4 requires the reference of the sound level meter to be the random incidence sensitivity, which is the root-mean-square sensitivity to sound arriving with equal probability from all directions in space. As it is almost impossible to create such a sound field, the random incidence sensitivity is calculated from free-field sensitivities in specific directions. The presentation of the measured results can be in a form as shown in Fig.2 (curve random) and Fig.5, where the random incidence response is recorded as a function of frequency. For calculation of the random incidence response, either $\Delta p$ can be plotted

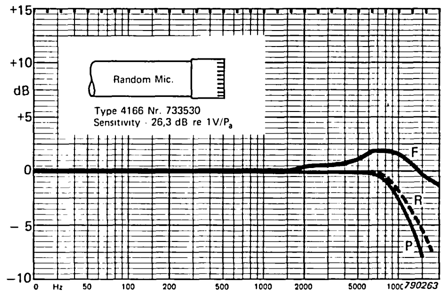



*Fig. 5. Random incidence response (dotted), the spatial average of equally distributed free-field responses. The figure is valid for 1/2 inch microphone type 4166*

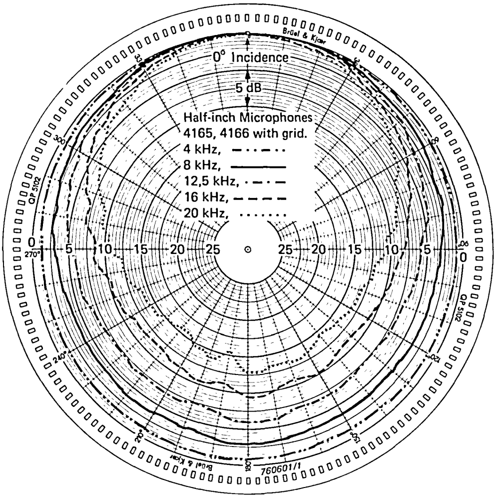



*Fig. 6. Polar diagram for microphone types 4165 and 4166 with protecting grid. The diagram shows the sensitivity relative to that at 0° incidence, for different angles of incidence with frequency as parameter*

as a function of frequency for various angles of incidence, or $\Delta p$ can be plotted as a function of the angle of incidence on a polar diagram at various frequencies (see Fig.6).

When the random incidence sensitivity is calculated, it should be remembered that it is the space average which must be determined and not the average in a plane. Consequently, the 90° incidence has a much higher significance than 0° and 180°. It is assumed that the microphone is completely rotationally symmetrical around its longitudinal axis. The random incidence response determined in this way is drawn as a dotted curve in Fig.5. By adding the dotted curve in Fig.2 to the microphone's pressure response, the random response is obtained for the individual cartridge. Microphones with a nearly flat random incidence response are called *random incidence microphones.* It should be emphasized that each microphone has a pressure response, 0° response, and a random incidence response, and is named after which of its responses is nearly flat as a function of frequency. Pressure, free-field and random incidence response curves are all well defined physical properties, which can be determined very accurately. The problems and possible inaccuracies arise only in connection with the use of the SLM in practice.

**Random Incidence Microphone in a Free-Field**

If a microphone with a flat random incidence response (as required by ANSI S 1.4) is placed in a free-field and pointed towards the sound source (see Fig.7.a), the sound pressure measured will, in this case, be much too high, namely the difference between the random response and the 0° incidence. For a 1/2 inch microphone the error would be 1 dB at 4 kHz, 2,5 dB at 8 kHz, and for higher frequencies still greater errors up to 7 dB can occur. In other words, a SLM according to ANSI S 1.4 should not be pointed towards the sound source as is common practice for the SLM according to IEC 651.

The question now being: Is there a particular angle of incidence for which a random incidence microphone, without any correction for frequency, can measure correctly in a free-field? In other words, is there among the response curves in Fig.2. a curve which corresponds exactly to the random response curve? The answer is no, however, the 75° curve is very close. This would mean that a SLM according to ANSI S 1.4, with a microphone of the type shown in Fig.2, should be rotated to 75° incidence when measuring outdoors and also indoors if the sound is expected to arrive predominantly from a certain direction, as shown in Fig.7.b.

It should be pointed out that practical conditions for correct use of both a pressure and a free-field microphone are very often met in our daily life. Outdoors, it is very often possible to determine the direction of the sound source and consequently the free-field microphone can be used without introducing any error. The same cannot be said about the use of

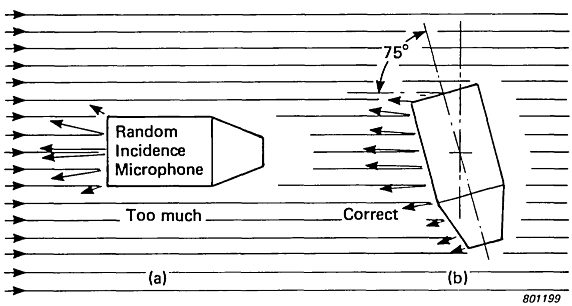



*Fig. 7. Random incidence microphone in a free-field:*
*(a) pointing towards the source will give too high a reading,*
*(b) nearly grazing incidence, or more precisely 75° incidence, will give correct free-field curve*

random incidence microphones, as the random sensitivity is valid only where the sound field is random. In practice it is difficult to find a room where a nearly true random field exists. If there are a few sound sources in a room which generate an uneven distribution, an error of several decibels is very likely to occur.

### Free-Field Microphone in a Random Field

If a microphone with a Flat free-field response (0°) is placed in a sound field where the sound arrives from several directions, the sound pressure measured will be too low, as the sound from any direction other than perpendicular to the diaphragm will give too low a contribution. If the sound field is completely random, the resulting response can be obtained by adding the pressure curve in Fig.3.b to the random curve in Fig.2. The result is shown in Fig.8, where all the higher frequencies are measured too low, as indicated by the hatched area. This would mean that a free-field microphone tuned for flat response for 0° incidence is not suitable for measurement indoors where the sound arrives from several widely different directions.

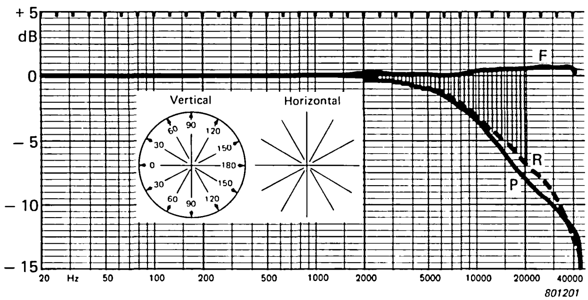



*Fig. 8. Free-field microphone type 4165 used in a random sound field will indicate too low a value, as indicated by the hatched area*

### Directional Sensitivity for Free-Field Measurements

Since IEC 651 prescribes calibration of a SLM for free-field measurements in the direction specified by the manufacturer, this would imply, in practice, that SLMs with a free-field microphone should be pointed towards the sound source, whereas a SLM with a random incidence microphone should be oriented at 75° from the direction of the sound source. Some recommend the use of grazing incidence (90°) but this is rarely optimum, as can be seen from Fig.2. It will be shown that in many situations it is impossible to determine exactly from which direction the sound arrives, or whether there are reflections; that is, the sound one wishes to measure arrives from different directions. It would, therefore, be desirable to determine an angular range of incidence over which all sound waves could be measured correctly.

From Fig.6 it can be seen that a free-field microphone, when pointed towards the sound source, can measure between ± 33° with an error between 0 and −1 dB at 12,5 kHz. For a random incidence microphone, which must be oriented at 75° from the sound source, the conical angle between which a maximum error of ± 0,5 dB is obtained is only ± 6°, which can also be inferred from Fig.6. Fig.9 shows the limits for the angles of incidence, for an error of 0 to −1 dB for a ½ inch and 1 inch free-field microphone, and for an error of ± 0,5 dB for a ½ inch and 1 inch random incidence microphone. It can be seen that the usable angular range for the free-field microphone is much larger than for the random incidence microphone.

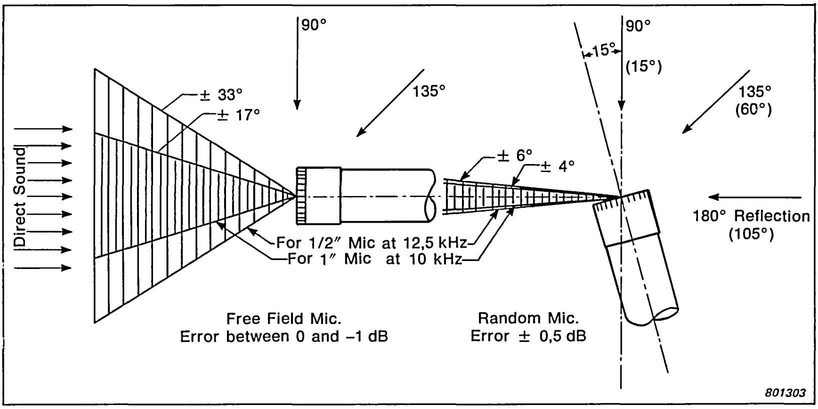



*Fig. 9. Angular range for an error of (0 to −1) dB for a ½ inch and 1 inch free-field microphone and for an error of ± 0,5 dB for a ½ inch and 1 inch random incidence microphone*

**Maximum Measurable Sound Pressure Level**

As mentioned earlier, when a microphone is brought into a sound field it will alter the pressure in the field. From Fig.2 it can be seen that the pressure rise, $\Delta p$, at 12,5 kHz for 0° is 7 dB. For microphones having a flat free-field response for 0° incidence the sensitivity is reduced correspondingly. This means that the microphone is able to measure the same sound pressure level at higher, as well as lower frequencies. Furthermore, this sound pressure level will not overload the microphone at any angle of incidence, as the pressure increase $\Delta p$ is lower at all other angles than for 0° incidence.

If a random incidence microphone is placed in a free-field, the same alteration in the field would occur around the microphone. However, at higher frequencies the sensitivity is only reduced corresponding to the pressure increase in a random field, which is lower than for 0° incidence. Thus, the maximum sound pressure to which a random incidence microphone can be subjected, is less than that for the free-field microphone. Table 1 shows the drop in the maximum measurable sound pressure level to which 1 inch and 1/2 inch random incidence microphones can be subjected, relative to corresponding free-field microphones.

| **Frequency** | **1** | **2** | **5** | **10** | **12.5** | **16** | **kHz** |
|---|---|---|---|---|---|---|---|
| 1/2 inch microphone (4166) | 0 | 0.3 | 1.3 | 3.4 | 4.6 | 5.6 | dB |
| 1 inch microphone (4144) | 0.5 | 1.1 | 3.0 | 6.8 | 8.0 | 12 | dB |

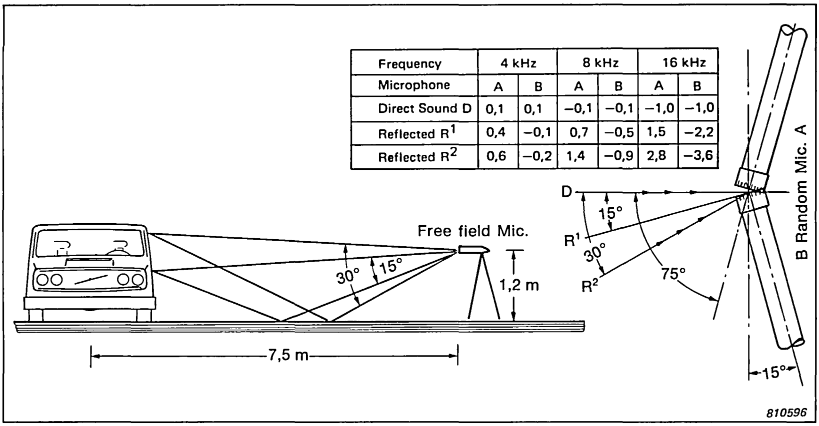
T00210GB0

*Table 1. Drop in the maximum measurable sound pressure level of 1 and 1/2 inch random incidence microphones relative to corresponding free-field microphones*

Fig.3 shows the necessary attenuation (damping) of the pressure response to obtain a flat random incidence response and free-field response for a random incidence and free-field microphone respectively. The values in Table 1 are obtained from the difference between the attenuation of the two microphones. It should be noted, however, that due to the higher attenuation of the free-field microphone, the thermal noise of the free-field microphone is increased by about 2,5 dB relative to the random incidence microphone.

**IEC 651 contra ANSI S 1.4**

There are well documented technical reasons why free-field and random incidence responses are specified in IEC 651 and ANSI S 1.4 respectively. Since the position of the sound source is often known when measurements are made outdoors, one can point the microphone of the SLM towards the source. A typical example is shown in Fig.10, which illustrates the standardized method of measuring the noise emitted by a car in accordance with ISO Recommendation 362 Ref. [4]. It can be seen that




| Frequency | 4 kHz | | 8 kHz | | 16 kHz | |
|---|---|---|---|---|---|---|
| Microphone | A | B | A | B | A | B |
| Direct Sound D | 0,1 | 0,1 | −0,1 | −0,1 | −1,0 | −1,0 |
| Reflected $R^1$ | 0,4 | −0,1 | 0,7 | −0,5 | 1,5 | −2,2 |
| Reflected $R^2$ | 0,6 | −0,2 | 1,4 | −0,9 | 2,8 | −3,6 |

*Fig. 10. Measurement of car noise according to ISO Recommendation 362. Direct and reflected waves measured with IEC 651 SLM (left) and with ANSI S 1.4 SLM (right). Table indicates errors for variation in angle of incidence for a random incidence microphone*

the angles between the direct and reflected waves arriving at the SLM are about 20° to 25°. As the road must be fully reflective, the sound pressures of the two waves are of the same order of magnitude. From Fig.6 it can be seen that within ± 25° around 0° incidence, the variation in the microphone sensitivity is minimal and consequently both the dominant sound waves are measured correctly. If an ANSI S 1.4 SLM is used the microphone should be held such that the direct wave is at 75° incidence and the reflected wave would then arrive 20° to 25° from the 75°. It can be seen from Fig.6 that near grazing incidence there is a large change in sensitivity for only a small change in the angle of incidence. As a result, the reflected wave will contribute too little to the overall level. It is possible to rotate the ANSI S 1.4 meter further and minimize the error, however, from the polar diagrams in Fig.6 it can be seen that further rotation will also increase the contribution from the direct wave. Furthermore, the direct and the reflected waves being in opposite phase at some frequencies does not help the situation.

If the noise emitted by the car consists mainly of low frequencies, the error caused by using a random incidence microphone would be relatively small. However, the test laid down by the ISO Recommendation requires acceleration at full power, making the fan noise very dominant due to the high engine speed. The intake noise is also high as the throttle is fully open. Because of the high acceleration in low gear, the squealing of the tires can make a significant contribution to the overall noise. Often the exhaust noise, contrary to what many believe, is negligible. As the dominant noise in this test consists of high frequencies, the importance of the microphone response between 5 and 12 kHz is evident. As standardized noise tests on cars, motorcycles, airplanes, etc., have to be carried out with the highest possible accuracy, the IEC microphone philosophy is far superior for these outdoor uses.

For indoor measurements, however, many are of the opinion that the situation is just the opposite, which is also true with some modifications. When measuring the overall level in a large highly reverberant workshop, where the sound arrives from many different and often unknown directions, the use of the ANSI S 1.4 meter with its random incidence microphone is preferable. For the same situation the IEC 651 meter will measure too low a value. When the noise has to be measured in the same workshop at the operator's position near a particular machine, it is highly probable that most of the sound arrives from one direction, and therefore, the ANSI S 1.4 meter will indicate too high a level. Also, if the workshop is absorbent and has a short reverberation time, it is most likely that the dominant sound arrives from a single direction. In these last two cases the use of an IEC 651 meter would be more correct.

Measurement of sound power emitted by machines and equipment is described in a series of ISO standards, many of which have now been adopted as national standards in several countries. Basically two different methods are described: free-field measurement and measurement in reverberation rooms. Ref.[5, 6]. Undoubtedly, all free-field measurements, both in anechoic rooms and outdoors, should be made with free-field microphones (IEC 651), and it is just as clear that a random incidence microphone (ANSI S 1.4) is the right choice for the reverberation room methods. If a free-field microphone is used in the reverberation room, errors up to 3 – 5 dB can easily occur at high frequencies.

Table 2 shows for which applications the free-field and random incidence microphones should generally be used. It can be seen that the free-field microphone is the most applicable in the majority of cases. Measurements for the noise certification of automobiles, airplanes, building equipment, chain saws, lawn mowers, as well as determination of noise limits in neighbouring grounds, outdoor shooting ranges, and radiated power from machinery and household equipment, all require an accuracy of a fraction of a decibel, which can be achieved. The only

| Measuring Object & Purpose | Normal SLM | |
|---|---|---|
| | **Free-Field Microphone IEC 651** | **Random Microphone ANSI S 1.4** |
| Outdoor | X | |
| Automobiles | X | |
| Motorcycles | X | |
| Traffic – general | X | |
| Airplanes – outside | X | |
| inside | | X |
| Ships – outside | X | |
| inside | | X |
| Industries – outside | X | |
| Workshop – reverberant | | X |
| Machines in workshop | X | |
| Sound power measurements in reverberation room ISO 3741 | | X |
| Sound power measurements in anechoic room ISO 3745 | X | |
| Workshop – absorbent | X | |
| Ventilation system | | X |

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*Table 2.*

situation where an ANSI S 1.4 meter is preferable is for noise measurements indoors. However, these measurements can never be very accurate since only a few centimeters change in the position of the SLM would give a difference of 1 – 2 dB due to resonances in the room. One can, therefore, appreciate why the free-field concept was chosen for IEC 651.

**Interfering Reflections and Noise**

Since the pressure on the membrane of a free-field microphone increases for sounds arriving at 0° incidence, it would imply that the sounds arriving from all other directions would be measured too low. This is of great advantage for free-field measurements in practice, since the interfering reflections and noise, consisting of high frequency components, can be suppressed.

For example, when the noise of automobiles is measured according to ISO 362 (1964) (see Fig.10), the measurement site must be chosen far from buildings and other large surfaces which could reflect some of the sounds one wishes to measure. In general this is possible; however, it can be difficult to avoid reflections from smaller objects such as trunks of trees, tripods, an instrument operator, a hand behind a microphone, etc. These small objects do not disturb the measurements at low frequencies, but at high frequencies they can contribute significant interfering reflections.

For traffic noise or noise in residential areas coming from a particular industry, outdoor shooting ranges, airports, etc., it is desirable to isolate and thus evaluate the noise coming from the individual source. The different codes of practice, in general, require suppression of interfering noise from other sources and reflections. In other words, *only* the direct sound should be measured. The free-field microphone is therefore of considerable help in suppressing the interfering sounds. Table 3 lists the suppression in decibels of reflected sounds arriving at 90°, 135° and 180° *relative to the direct sound*, as shown in Fig.9.

| | Microphone Type with Grid | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | 1/2" Microphone Free Field Type 4165–F Random Type 4166–R | | | | | | 1 Microphone Free Field Type 4145–F Random Type 4144–R | | | | | |
| Direction of reflections relative to direct sound | 90° | | 135° | | 180° | | 90° | | 135° | | 180° | |
| Frequency | F | R | F | R | F | R | F | R | F | R | F | R |
| 1 kHz | 0 | 0 | 0 | 0 | 0 | 0 | –0,5 | +0,5 | –1,0 | +0,5 | –0,5 | 0,0 |
| 2 kHz | –0,5 | 0 | –0,5 | 0 | –0,5 | 0 | –1,0 | +1,0 | –1,5 | +0,5 | –1,0 | –0,5 |
| 4 kHz | –1,0 | +1,0 | –1,5 | +0,5 | –1,0 | –0,5 | –3,0 | +2,0 | –4 | +1,0 | –3,0 | –1,0 |
| 8 kHz | –3,0 | +2 | –4 | +1,0 | –3,0 | –1 | –7,0 | +5 | –10 | +4,5 | –8,0 | –2,5 |
| 16 kHz | –8 | +5,5 | –10,5 | +1,5 | –9,5 | –4 | –20 | +14 | –22 | +3,5 | –21 | –8 |

– indicates suppression + indicates amplification 810001

*Table 3. Suppression of reflected waves or noise arriving at various angles relative to the direct wave valid for 1 and 1/2 inch free-field microphones F and random incidence microphones R*

It can be seen that when a random incidence microphone is used for free-field measurements, reflections arriving at angles greater than 180° are suppressed insignificantly, while sounds such as secondary reflections arriving at angles less than 180° and noise are in fact amplified. Furthermore, a 1 inch microphone naturally has a greater suppression effect than a 1/2 inch microphone. This is one of the few situations in acoustical measurements where a 1 inch microphone is superior to a 1/2 inch microphone.

The advantages of a free-field microphone (IEC 651) over a random incidence microphone (ANSI S 1.4) can be concluded in the following:

1. The sensitivity does not alter signficiantly for the angle of incidence between ± 30°, ensuring high accuracy.

2. Larger frequency range (approximately one octave).

3. Interfering reflections and noise arriving from behind are significantly suppressed (2 to 5 dB)

4. A higher maximum sound pressure level is measurable.

On the other hand, a free-field microphone will systematically measure too low a level in a reverberation room. Here a random incidence microphone is undoubtedly preferable.

**Integrating SLM**

With respect to the above discussion, the only instrument on the minds of both the IEC and ANSI committees when formulating the standards was the ordinary SLM, but with the advent of the integrating SLM changes are imminent, Ref.[7]. The integrating sound level meter is just a normal SLM, but with the capability of storing a large number of samples and calculating a linear average (instead of exponential) for any length of time.

By holding the instrument in a fixed position and direction, an average value over a time period can be obtained, which is a very useful feature, especially when measuring noise of a fluctuating character. As the variation in the momentary sound level is often large when measuring over long times, a very large dynamic range is required. It is therefore suggested in the Standard IEC 29C WG 11, dealing with integrating sound level meters, that the instrument should have a dynamic range of at least 60 dB for type 1, Ref.[7]. For a fixed position and direction all the previous arguments concerning free-field contra random incidence microphones would be the same. Thus, the recommendations given in

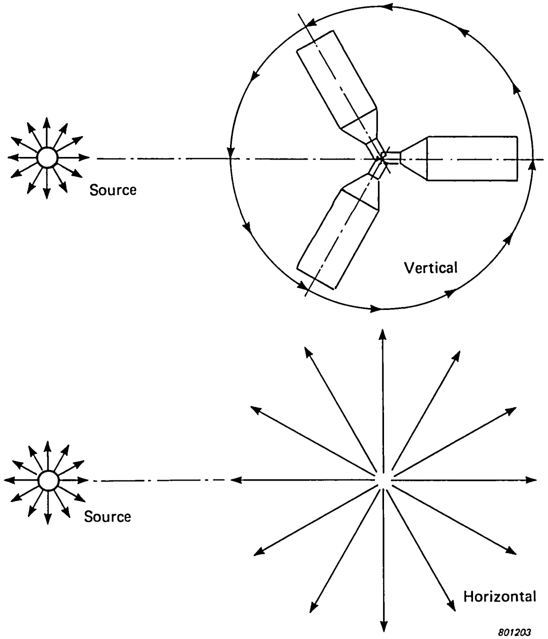



*Fig. 11. Rotation of a sound level meter in several planes*

Table 2 are also valid for integrating SLMs, indicating that free-field microphones (IEC 651) would be the best choice in the majority of situations.

By moving an integrating SLM around, the instrument averages over the different directions and will now indicate too low a value if it is supplied with a free-field microphone. In this case, it is more sensible to mount a random incidence microphone on the instrument; however, only if the instrument is pointed an equal length of time in all directions in space. If the temptation to move the instrument around in only one plane is not overcome, errors in the order of 2 to 5 dB can be introduced. It is therefore imperative to rotate the instrument through several planes, covering all directions in space (see Fig.11).

For measurement of the average sound level in a workshop or in any other room, the integrating SLM helps considerably in achieving accurate results. By going around the room equally in all spaces while pointing the instrument in all directions, a temporal and spatial average

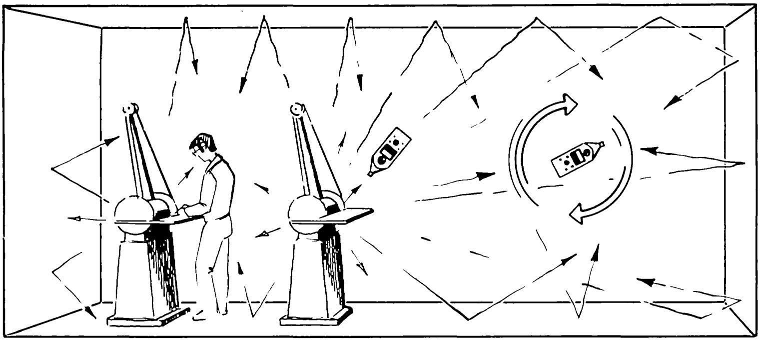



*Fig. 12. Noise measurement in a workshop:*
*(a) illustration of rotating an integrating SLM with a random incidence microphone, rotation in several planes gives the correct average for all directions*
*(b) a SLM with a free-field microphone should be directed towards the source over the entire measurement period when determining the sound at the operator's position*

of the sound level can be obtained. Normally, measurement for around a minute is enough to get a very reliable result (see Fig.12.a).

It is surprising that with an integrating SLM, repeatability in the results of a fraction of a decibel can be achieved, compared to a conventional SLM where variations of several decibels can occur.

**Conclusions**

It can be seen that generally for measurements indoors, especially with an integrating SLM, and for sound power measurements in a reverberation room, it is necessary to supply the SLM with random incidence microphones. It is just as important for outdoor measurements and measurements in an anechoic chamber to equip the SLM with a free-field microphone. Consequently, an integrating SLM should either:

1. Be supplied with two microphones, one free-field and one random incidence microphone, or

2. Have a free-field microphone and a random incidence corrector (small resonator in front of the microphone) for indoor use, or

3. The amplifier should be supplied with a switch which can change between free-field and random incidence response electronically.

In the above-mentioned standards, as well as forthcoming standards describing noise measurements, mention should be made of the situation in which each type of microphone is preferable. Furthermore, it is hoped that both IEC – TC 29 and ANSI S 1.4 will specify both the free-field and random incidence microphones as standards and give an indication as to where each should be used.

### References


[1] ANSI S 1.4 — *American National Standard.* Specification for Sound Level Meters, S 1.4 1983.

[2] IEC 651 — Sound Level Meters, *International Electrotechnical Commission Standard,* Publication 651 (1979).

[3] IEC 179 — Precision Sound Level Meters, *IEC Publication 179* (1965 and 1973).

[4] ISO R 362–1964 — Measurement of noise emitted by vehicles, *International Standards Organization.*

[5] ISO 3745–1977 — Determination of sound power levels of noise sources –Precision methods for anechoic and semi-anechoic rooms (Also 3744).

[6] ISO 3741–1975 — Determination of sound power levels of noise sources – Precision methods for broadband sources in reverberation rooms.

[7] IEC Draft, Integrating Sound Level Meters, *29C (Secretariat)* 37 September 1980.

# DESIGN PRINCIPLES FOR INTEGRATING SOUND LEVEL METERS

by

*Peter Hedegaard*

**ABSTRACT**

Several methods may be used for direct measurement and calculation of the equivalent continuous sound pressure level, $L_{eq}$.
From the measuring application, desired measuring accuracy, speed as well as the cost frame for the actual design, the instrument designer has to select his circuits and decide on the compromises necessary.
Practical examples of direct measuring / calculating methods used, including their advantages and disadvantages are given.
Two test methods and their influence on judgement of the quality of different instrument designs are illustrated.

**SOMMAIRE**

Plusieurs méthodes peuvent être utilisées pour la mesure et le calcul directs du niveau de pression sonore continu équivalent, $L_{eq}$.
Suivant les applications, la précision de mesure souhaitée, la vitesse et le prix de l'appareil, le constructeur doit choisir les circuits et décider des compromis nécessaires.
Cet article monte des exemples pratiques de méthodes de mesure et de calcul directs utilisées, ainsi que leurs avantages et désavantages. Il montre aussi l'influence de deux méthodes d'essai sur l'évaluation des qualités des différentes conceptions des appareils.

**ZUSSAMMENFASSUNG.**

Verschiedene Methoden lassen sich für die direkte Messung und die Berechnung des energieäquivalenten Schallpegels $L_{eq}$ benutzen.
Der Konstrukteur muß für Anwendung, gewünschte Genauigkeit, Geschwindigkeit sowie Kostenrahmen der jeweiligen Konstruktion seine Schaltungen auswählen und die notwendigen Kompromisse eingehen.
Es werden praktische Beispiele mit ihren Vor- und Nachteilen sowohl für die direkte Messung als auch für Berechnungsmethoden gegeben.
Zwei Prüfmethoden und ihr Einfluß auf die Beurteilung der Qualität verschiedener Geräte werden gezeigt.

## Introduction

For several years, modern technology has made possible the direct measurement and calculation of the equivalent continuous sound pressure level, $L_{eq}$. The straightforward method, with fast sampling of the

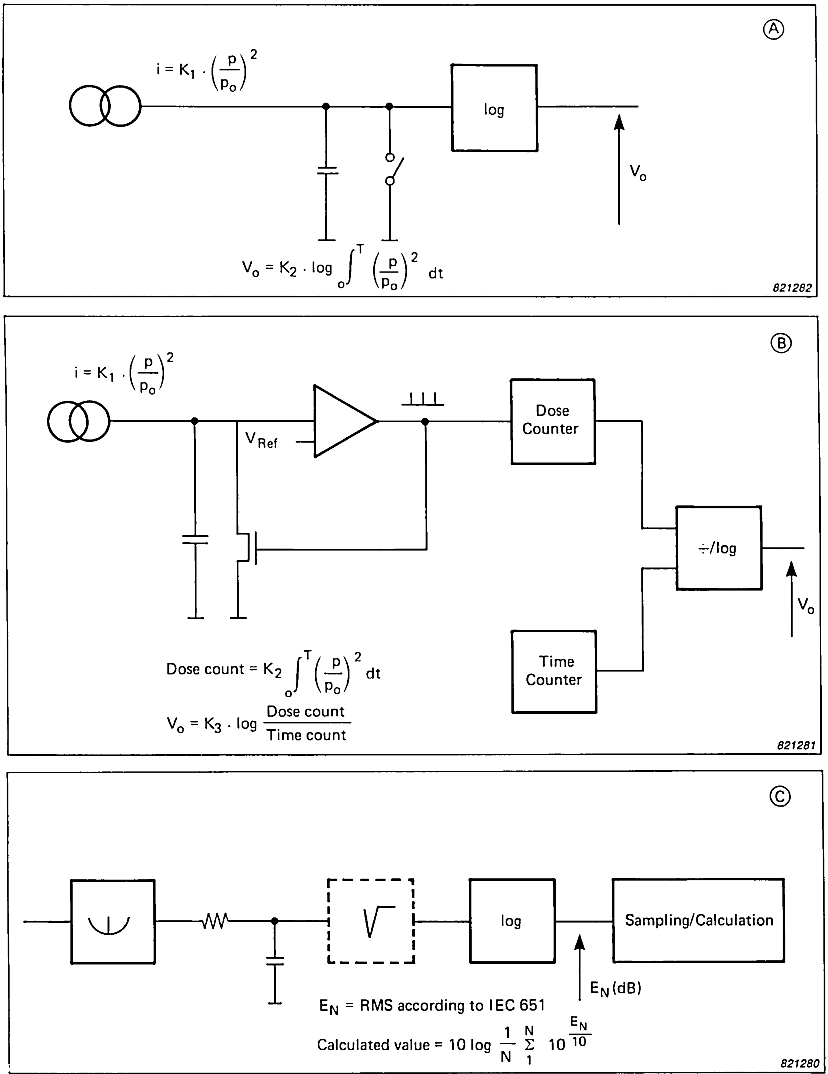



*Fig. 1. Measurement principles for the described integrating SLM types*

"raw" signal and calculation of the required parameters from the sampled values is still only possible by use of relatively large processor systems, and is therefore not yet suitable for small hand-held instruments. Therefore, combined analog and digital techniques are most often used. Three measuring methods are selected here, partly because they are commonly used in small instruments and partly because they differ in their working principles.

The first principle, shown in Fig.1.a, is the simplest and cheapest. A current proportional to the square of the input signal charges the capacitor, which acts as the integrator. After a fixed integrating time the voltage across the capacitor represents the $L_{eq}$ over the integration time.

It should be pointed out that the figure shows the principle of operation and not the practical design, where a feed-back mechanism is often used to make the capacitor voltage equal to the logarithm of the integral. As the fixed integration time often will be in the order of magnitude of 1 minute, the currents handled in such circuits are very low.

The second principle, shown in Fig.1.b, works partly on the same principle with a current source, proportional to the square of the input quantity, charging a capacitor. But in contrast to the previous example, the charging time is very short (down to a few microseconds) for maximum charge at the highest input. When the charge reaches a certain maximum, an electrical switch discharges the capacitor very rapidly and thereby "resets" the integrator and prepares it for a new integration. Each "reset" represents a fully charged capacitor and thereby equal parts of the total integral which will be proportional to the counted reset events. The $L_{eq}$ is then found simply by dividing the content of the dose counter by the measuring time.

The third principle, Fig.1.c, works on the basis of the commonly used RMS detector system described in the International Standard for Sound Level Meters, IEC 651. The output value (often logarithmic) from the detector is sampled, and the sampled values are then used as the basis for the calculation for the $L_{eq}$. In contrast to the previously mentioned principles, the circuit does not follow the integration formula a) but the formula b)

a) $L_{eq} = 10 \times \log \frac{1}{T} \int_0^T \left(\frac{p}{p_0}\right)^2$ b) $L_{eq} = 10 \times \log \frac{1}{N} \sum_1^N 10^{\frac{L_N}{10}}$

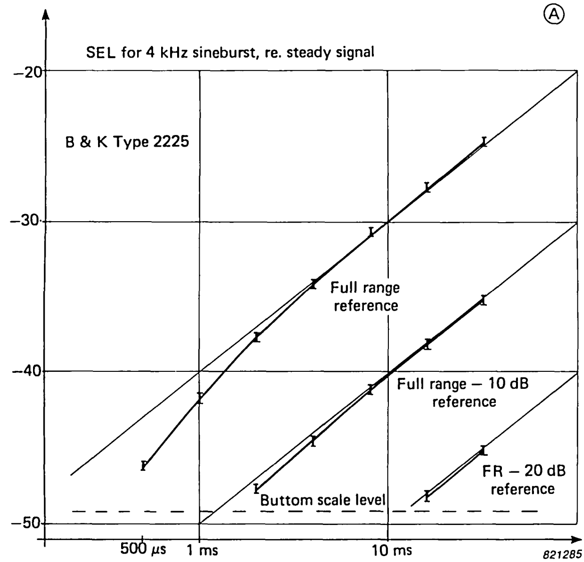
as the sampled values are the squared and exponentially averaged detected values. But as the measuring time for the $L_{eq}$ is large, often several minutes or hours, compared with the time constant for the detector circuit, the circuit principle works very well and can easily meet all requirements proposed for integrating sound level meters.

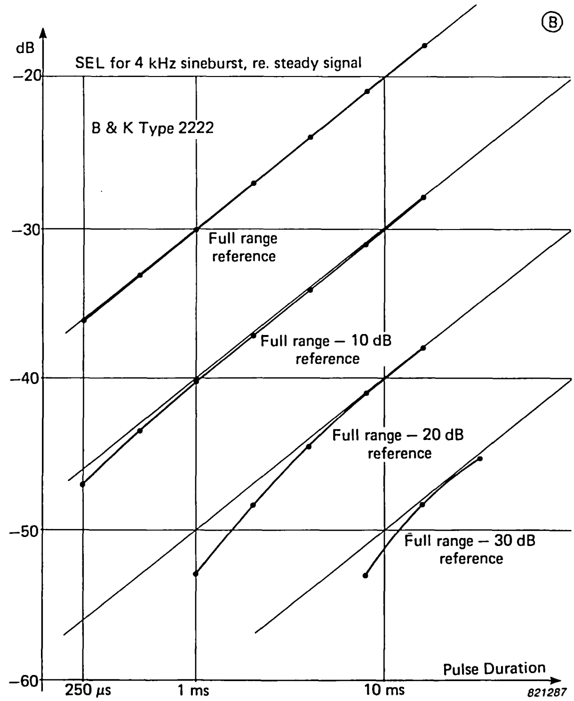
The following will only deal with the integrating part of the described principles, because the characteristics of the circuits and the influence of the test methods are most clearly reflected here.

**Design Principles**

As mentioned earlier, the first circuit principle (Fig.1.a) has to work with very low currents. It is therefore to be expected that some error will occur at low levels or for a short impulse response. From Fig.1.a. it can be seen that the charging current will be proportional to the square of the input quantity, which for a 60 dB dynamic range means a ratio of 1:1000000. This, in connection with the relatively long integration time, makes a practical solution which strictly follows Fig.1.a almost impossible. However, it is possible in the current generation to include the log mechanism before the integration capacitor and still have electrically identical circuits. This means that the charging current now, within the given dynamic range, will only vary proportionally with the logarithm of the input quantity. In Fig.2.a. is shown the impulse response for an instrument using this principle. From the figure it can be seen (as also could be expected) that some error occurs for short impulse responses and that the response error depends on the impulse width and not on the level.

If we look at the second circuit principle (Fig.1.b), it can be seen that the analog part of the integration circuit looks similar to the circuit shown in Fig.1.a. The integration mechanism consists here of the analog integration and a counting of the "full charges" on the integration capacitor. As the integration time for the analog part does not need to be long, the problem with the large current ratio can be managed. However, concerning the impulse response from the whole system, two problems can be expected. First, because of the large current ratio (1:1000000 for a 60 dB dynamic range) we have to deal with very small currents at low levels and some error in the response to short impulses will therefore probably occur. Second, the frequency of the reset impulses will vary proportionally with the charging current and therefore, for the mentioned 60 dB, will have a similar ratio namely 1:1000000. For the practical solution of this measuring principle, the frequency range for


A
SEL for 4 kHz sineburst, re. steady signal
B & K Type 2225
Full range reference
Full range – 10 dB reference
FR – 20 dB reference
Buttom scale level
–20
–30
–40
–50
500 μs
1 ms
10 ms
821285


B
dB
SEL for 4 kHz sineburst, re. steady signal
B & K Type 2222
Full range reference
Full range – 10 dB reference
Full range – 20 dB reference
Full range – 30 dB reference
–20
–30
–40
–50
–60
250 μs
1 ms
10 ms
Pulse Duration
821287

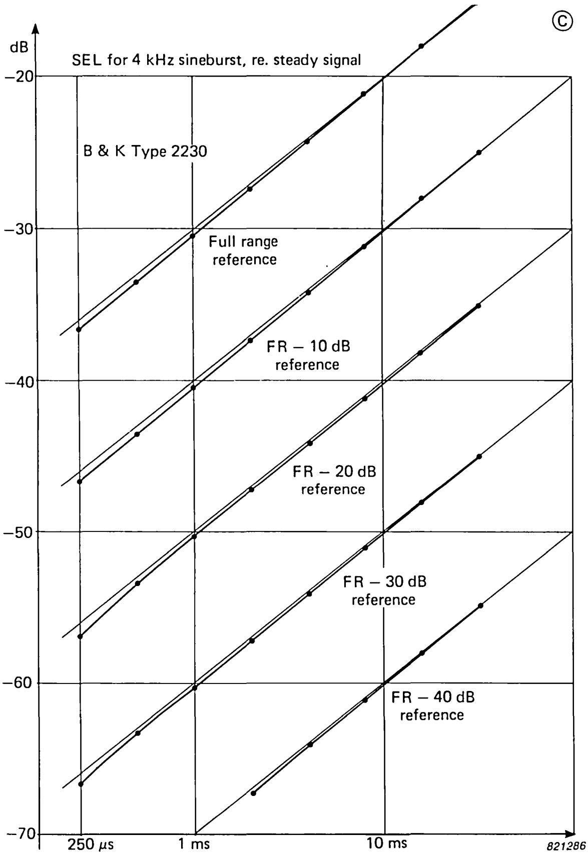



*Fig. 2. Theoretical and measured sound exposure level for a 4 kHz tone burst*

impulses to be counted is 0,25 Hz to 250 kHz. In Fig.3 is shown the connection between the level, the time to produce a full charge on the integration capacitor, and the pulse frequency. It is now seen that if the time to fill the integration capacitor is bigger than the width of the signal impulse, the system will not respond at all. This situation can occur for short impulses and low levels. In other words, the condition for precise measurements depends on the fact that the integration capacitor has to be "filled up" several times. In Fig.2.b is shown the impulse response for an instrument built according to the mentioned principle. It is seen that at low levels the error will depend on level and

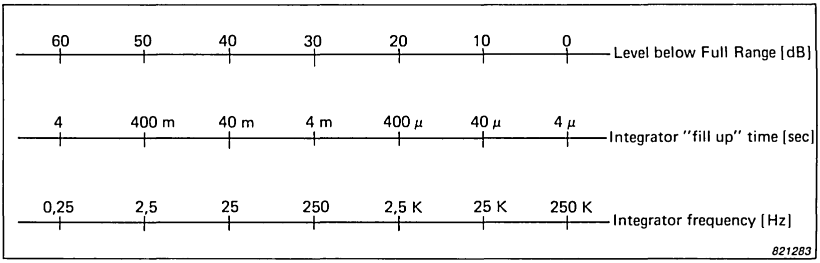



*Fig. 3. Connection between signal level and integration time/frequency. (B & K Type 2222)*

impulse width. However, for the purpose of measuring single short impulses, the response is only required to be a few dB's above the bottom level for reliable results. The advantage of the system is that it allows long time integration and (apart from the situation concerning measurement of single short impulses) it gives very accurate integration.

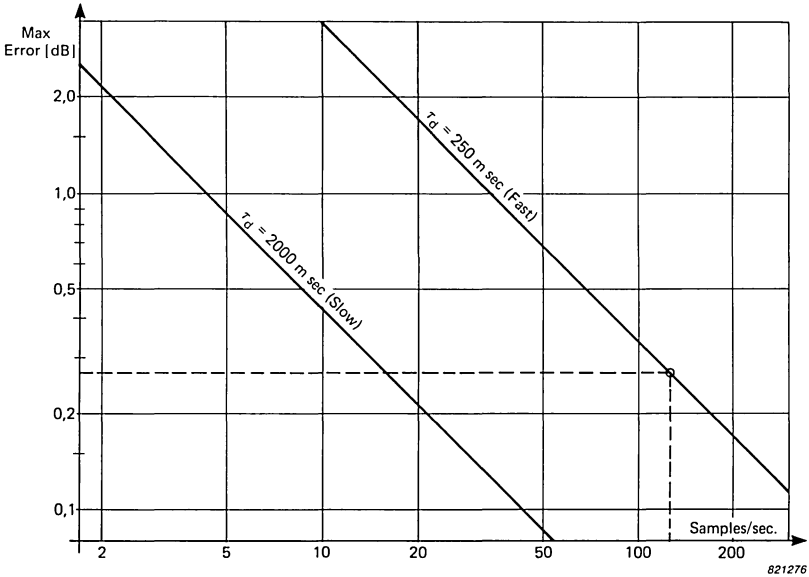



*Fig. 4. Error introduced by sampling on an exponentially averaging RMS detector*

The third principle, Fig.1.c, works on the basis of sampled data taken from a detector system which works according to the requirements stated in the IEC 651 Standard for Sound Level Meters. Again, if we look at the impulse response, two sources of error can be found. The first is the impulse response for the detector itself and the second is the speed of the sampling. A measure of the sampling error may be the relative change of the detector's decay function during a sampling interval. Fig.4 shows a maximum error as a function of the number of samples/second for the two standardized time constants "Fast" and "Slow". Fig.5 shows the response for the detector itself. From Figs.4 and 5 it can be seen that a more precise impulse response is obtained if the time constant is short, but that a short time constant requires a faster sampling speed. The compromise has to be chosen on the basis of the requirements stated for response to short impulses, the time constant and the possible speed of the following system. In Fig.2.c. is shown the impulse response for a practical system built according to the described principle. The detector time constant is chosen to be 125 ms, corresponding to IEC "Fast". This choice is made because the processor system then, besides the values used for calculation of the $L_{eq}$, can "pick up" values corresponding to measurements according to IEC 651 (max, min, max in 1 s periods etc.). The sampling speed, which is 128/s, will give rise to a max. sampling error of 0,27 dB. From Fig.2.c it can be seen that the error for short impulses is independent of level.

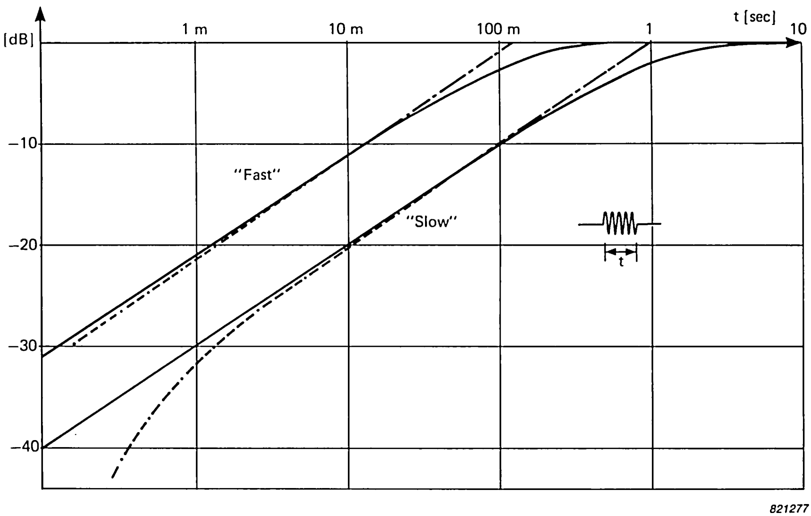



*Fig. 5. Theoretical and measured response to 6 kHz tone bursts for an exponentially averaging RMS detector (B & K Type 2230)*

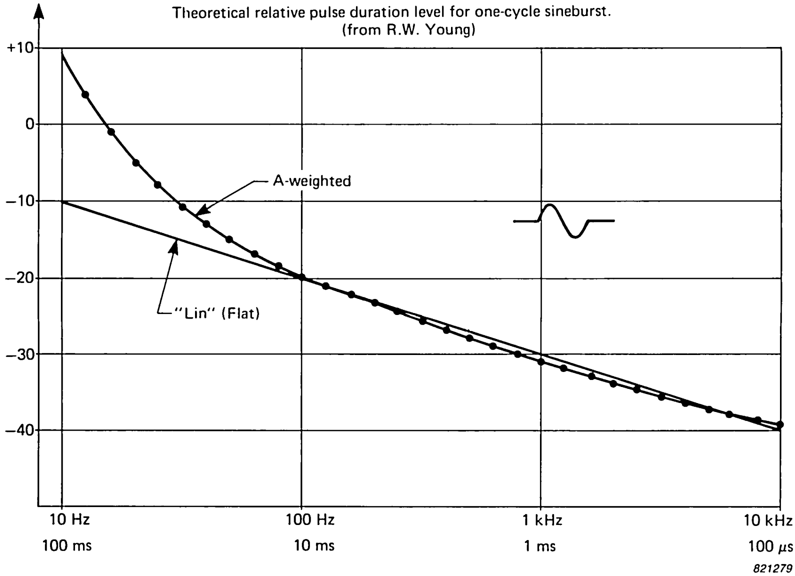



*Fig. 6. Theoretical relative pulse duration level for one-cycle sine burst. (From R.W. Young)*

The previous tests have all been based on the response to 4 kHz tone bursts of varying width. Practically occurring impulses are often characterized by their peaks or, after some filtering, by transients stimulated by the on-set. Transients stimulated by on-set are often masked by the steady signal that follows, for which reason the multi-cycle toneburst test may be insufficient. An interesting test method to overcome this problem has been suggested by Dr. R.W. Young, San Diego, CA, USA. As test pulses, one-cycle sine bursts of varying frequency are used. The test deals with a quantity "Pulse duration level", which is defined as the sound exposure level of a one cycle sine burst minus the sound pressure level of the steady sine wave of the same frequency and amplitude. Fig.7 shows the results of practical measurements carried out on the three instruments used above. One difficulty found by using this method, is that it combines requirements for several characteristics i.e. detector impulse response, level linearity and frequency weighting. There may therefore be some difficulties in interpreting the measurement results.

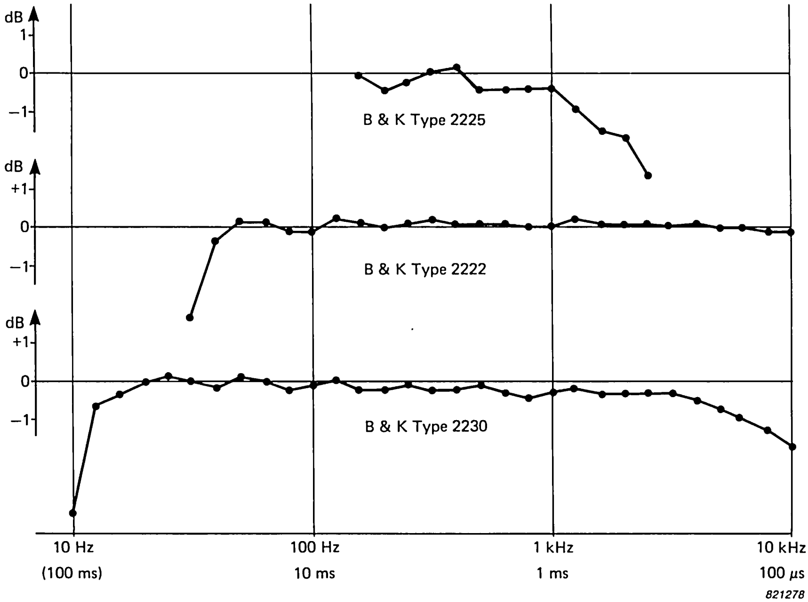



*Fig. 7. Error in measured A-weighted pulse duration level*

To make thorough tests on integrating sound level meters may be a difficult and time consuming task. Results of simplified tests, which do not take into account the design principles, will hardly give a measure of the quality of the instrument. On the other hand, provided that instruments are well specified and designed to fulfil their measurement application and not just a certain standard, even relatively simple tests may lead to reasonable results.

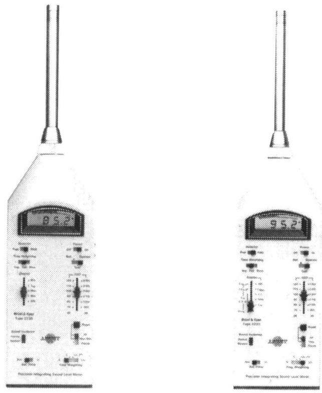
**References**

[1] Draft. Specifications for Integrating Sound Level Meters IEC TC29–WG11 1981

[2] YOUNG, R.W. et.al. Draft. Specifications for Integrating Averaging Sound Level Meters. Nov. 1980.

# News from the Factory

## Precision Integrating Sound Level Meters Types 2230 and 2233

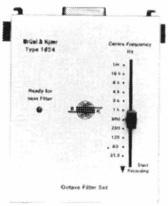


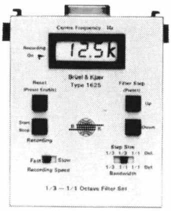
Two new Type 1 Impulse Precision Integrating Sound Level Meters Types 2230 and 2233 have been introduced by Brüel & Kjær. Their comprehensive facilities and high precision make them ideal for all kinds of sound level measurements, including octave and 1/3 octave frequency analysis using clip-on filter sets. The meters comply with IEC 651 Type 1 (Impulse) and ANSI S1.4. 1983 Type 1.

Type 2230 carries out five independent measurements in parallel, and enables the user to switch between the various parameters without interrupting the measurement. These are: Current, Max., and Min. SPL; $L_{eq}$ and SEL. The Type 2233 is similar to the 2230 but measures the following parameters according to regulations applicable in the Federal Republic of Germany: $L_{max}$, $L_T$ (1, 3 or 5 s), all appropriate $L_m$ (according to TA-Lärm and VDI 2058), SEL, and finally, the elapsed measuring time.

For both instruments a choice between two detector modes (RMS and Peak), three time weightings (Slow, Fast, Impulse), and four frequency weightings (A, C, Lin and All-pass) are provided. A free or diffuse field frequency response is obtained for the microphone by electronic frequency weighting. A partial ($L_{max}$) reset for 2233 and partial (Max/Min) reset for 2230, together with a total reset and a pause function increase the usefulness of the instrument. Measurements are displayed with a 0,1 dB resolution on a large liquid crystal display together with various warnings. The SPL is continuously monitored on a quasi-analogue 60 dB scale. AC and DC outputs allow chart or tape recordings and audio monitoring. Despite their comprehensive facilities and sturdy design the new meters are easily held in one hand and weigh only 860 gr.

**Octave Filter Set Type 1624 and Third Octave/Octave Filter Set Type 1625**






Octave Filter Set Type 1624 and Third Octave/Octave Filter Set Type 1625 are primarily designed for use with the Precision Integrating Sound Level Meters Types 2230 and 2233 for in situ octave and 1/3 octave acoustic analyses.

The overall frequency range of both filter sets covers the entire audio-frequency range with the centre frequencies arranged according to the preferred frequencies of ISO R 266, DIN 45 401 and ANSI S 1.6–1960.

The Type 1624 contains 10 active octave filters with centre frequencies from 31,5 Hz to 16 kHz. Each octave filter satisfies the requirements of IEC Recommendation R 225–1966, DIN 45 651 and ANSI S1.11–1966 Class II. The total frequency range is from 22 Hz to 22 kHz. A linear position is also available covering the frequency range from 5 Hz to 75 kHz (–1 dB).

The Type 1625 contains 31 active $^1/_3$ octave filters and 31 overlapping $^1/_1$ octave filters at $^1/_3$ octave intervals covering 11 octaves. Centre frequencies are from 20 Hz to 20 kHz. Each filter fulfils the requirements of IEC recommendation 225 – 1966, DIN 45 652 and ANSI S1.11–1966 Class III for $^1/_3$ octave filters; and DIN 45 651 and ANSI S1.11–1966 Class II for $^1/_1$ octave filters. The total frequency range is from 18 Hz to 22 kHz ($^1/_3$ oct.) and from 14 Hz to 28 kHz ($^1/_1$ oct.). A linear position is also available covering the frequency range from 3 Hz to 75 kHz (–1 dB).

The filters may be switched successively manually (Type 1624 or 1625), or automatically (Type 1625). The level in each frequency band is displayed on the Sound Level Meter; however, for greater convenience the analysis can be recorded in situ, either semi-automatically (Type 1624) or automatically (Type 1625) using one of the portable B & K Level Recorders Type 2306 (one channel) or Type 2309 (two channels).

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