LEVEL METERS

APPLIED ACOUSTICS: BASIC PRINCIPLES

Acoustics is the science that studies sound (1). Sounds are vibration phenomena (in gas, liquids, and solids) originally limited to the sense of hearing and subsequently extended to ultrasounds and infrasounds (2–5). In the following, the term *noise* will be used as both an undesired sound and as a statistically random oscillation, according to American Standard Acoustical Terminology (6).

Sound Pressure

A sound exists when a perturbation propagates in an elastic medium and causes a pressure variation and particle movement in the medium. When a particle of air is displaced from its mean position, there is a temporary variation in pressure. The pressure variation acts in two ways: by restoring the particle to its original position, and by passing on the disturbance to the next particle. The cycle of pressure increases (compressions) and decreases (rarefactions) propagates through the medium as a longitudinal sound wave, characterized by two important parameters: the pressure (the local increases and decreases with respect to the environment) and the velocity of the air particles that oscillate around a fixed position. The basic definitions of sound are in terms of the magnitude of the fluctuating component of pressure in a fluid medium. The sound pressure $p(t)$ is a scalar quantity, characteristic of the measurement points; it has a dimension of newtons per square meter or pascals. The sound pressure level (L_p) is measured in decibels (dB):

$$L_p = 20 \log_{10} p/p_0 \, \mathrm{dB} \tag{1}$$

where p is the root-mean-square (rms) sound pressure, expressed in micro Pascal, and p_0 is equal to 20 μPa.

The rms value of the fluctuating component of pressure is used because most sound consists of random signals rather than pure sine waves. The value 20 μPa is an accepted standard reference value of pressure against which other pressures are compared by Eq. (1). Note that when p equals 20 μPa, the sound pressure level is 0 dB. This value was selected somewhat arbitrarily, but it represents the average threshold of audibility for human beings if a 1000 Hz tone is used. That is, the 0 dB level was selected as the lowest pressure fluctuation normally discernible by human beings. The decibel (logarithmic) scale is used as a convenience because of the human ear, sensitivity to noise, which follows an approximately logarithmic law, and the great ranges of sound pressure level of interest in ordinary work (from 10^{-6} Pa to 10^3 Pa). For example, an office with tabulating machines may have an L_p of 60 dB to 65 dB. The average human threshold of pain is about 140 dB. Sound pressure close to large rocket engines is on the order of 160 dB. One atmosphere is 194 dB. The span from the lowest to the highest pressure of interest is thus on the order of 1 to 10^9.

Sound Intensity

Sound pressure must not be confused with sound intensity, since the first is a scalar quantity, characteristic of the measurement point, and the second is a vector with its direction of propagation. Sound intensity $I(t)$ is a fundamental quantity for sound description (7): It is the product of particle velocity and pressure. As can be seen from the transformation of Eq. (2), it is equivalent for a given point and direction to the temporal average of the energy flux transmitted through a unitary surface perpendicular to the assigned direction in the considered point, and it is then measured in watts per square meter.

$$\begin{aligned}\text{Intensity} &= \text{Pressure} \times \text{Particle Velocity}\\ &= \text{Force/Area} \times \text{Distance/Time}\\ &= \text{Energy/(Area} \times \text{Time)} = \text{Power/Area}\end{aligned} \tag{2}$$

The acoustic intensity level L_{I} expressed in decibels is

$$L_{\mathrm{I}} = 10 \log_{10} I/I_0 \text{ decibels (dB)} \tag{3}$$

where I is the time-averaged intensity, expressed in W m^{-2}, and I_0 is equal to 1 pW m^{-2}.

The relationship between intensity and pressure is known precisely only in the case of *free fields* and *diffuse fields.* Given that a sound field is a region where there is sound, fields are classified according to the manner and the environment in which the sound waves travel.

Free Field. The term *free field* describes sound propagation in an idealized free space where there are no reflections. These conditions hold in the open air or in an anechoic room where the sound striking the walls is totally absorbed. Free field propagation is well characterized in terms of pressure and intensity level drop versus the distance from the source in the direction of sound propagation, as well as in terms of the relationship between sound pressure and acoustic intensity (8):

$$|I| = p^2/\rho c \tag{4}$$

where ρc is the acoustic resistance of the medium (ρ is the density and c the speed of sound in the medium). Having defined that, the acoustic impedance of the medium on a given surface lying in a wave front is the complex quotient of the sound pressure on the surface divided by the flux (volume velocity, or linear velocity multiplied by the area) through the surface; the acoustic resistance is defined as the real component of the impedance.

Diffuse Field. In a diffuse field, sound is reflected so many times that it travels in all directions with equal magnitude and probability. This field is approximated in a reverberant room. Although the net intensity $|I|$ is zero, there is a theoretical relationship that relates the pressure in the room to the one-side intensity, I_x (the intensity in one direction, ignoring the equal and opposite component) (88):

$$I_x = p^2/4\rho c \tag{5}$$

Sound propagation involves energy flow, but there can still be sound pressure even when there is no propagation. An *active field* is one where there is energy flow. In a pure *reactive field* there is no energy flow. A perfect correspondence, which gives rise to the use of the terms *active* and *reactive,* can be stated between pressure, particle velocity, and intensity, on the one hand, and voltage, current, and electrical power on the other hand. In an active field, pressure $p(t)$ and particle velocity $u(t)$ vary simultaneously. A peak in the pressure signal occurs at the same time as a peak in the particle velocity signal. They are therefore said to be in phase, and the product of the two signals gives a time-averaged intensity $I(t)$ (Fig. 1). In a pure reactive field the pressure and the particle velocity are 90° out of phase. One is shifted a quarter of wavelength with respect to the other. Multiplying the two signals together gives an instantaneous intensity $i(t)$ varying sinusoidally around zero. Therefore, the time-averaged intensity $I(t)$ is zero. In a diffuse field the pressure and particle velocity phase vary at random, so the time-averaged intensity is zero.

Sound Power

The sound power W is the energy emitted in the time unit, measured in watts, while the power level L_{W} expressed in decibels is

$$L_{\mathrm{W}} = 10 \log_{10} W/W_0 \text{ decibels (dB)} \tag{6}$$

where W_0 is equal to 1 pW.

According to the law of energy conservation, the energy flow per time unit that crosses a surface that completely covers a source must be equal to the sound power of that source (except for eventual dissipation losses). The total sound power for a spherical propagation in an elastic media is linked to the sound intensity at distance r by the equation

$$W = 4\pi r^2 I = 4\pi r^2 (p_r^2/\rho c) \tag{7}$$

where p_r is the sound pressure at a distance r.

Loudness

Loudness is a subjective quantity. It is defined as that aspect of auditory sensation in terms of which sounds may be ordered on a scale running from *soft* to *loud.* Loudness is chiefly a function of the level of sound pressure, but it is also dependent on the frequency and the composition of the sound. The range of loudness is divided subjectively into equal-unit steps called *sones.* The loudness of a sound at a given sound pressure level at one frequency may be quite different from the loudness of a sound of the same level at a different frequency. Nevertheless, listeners can adjust the level of one tone to match the loudness of another, and fair agreement among observers is usually obtained. Such experiments provide a useful objective scale of loudness, called loudness level. The loudness level of a tone in *phons* is numerically equal to the sound pressure level of a 1000 Hz tone that sounds equally loud. The loudness in sones (S) can be related to the loudness level in phons (P) by the empirical formula

$$S = 2^{(P-40)/10} \tag{8}$$

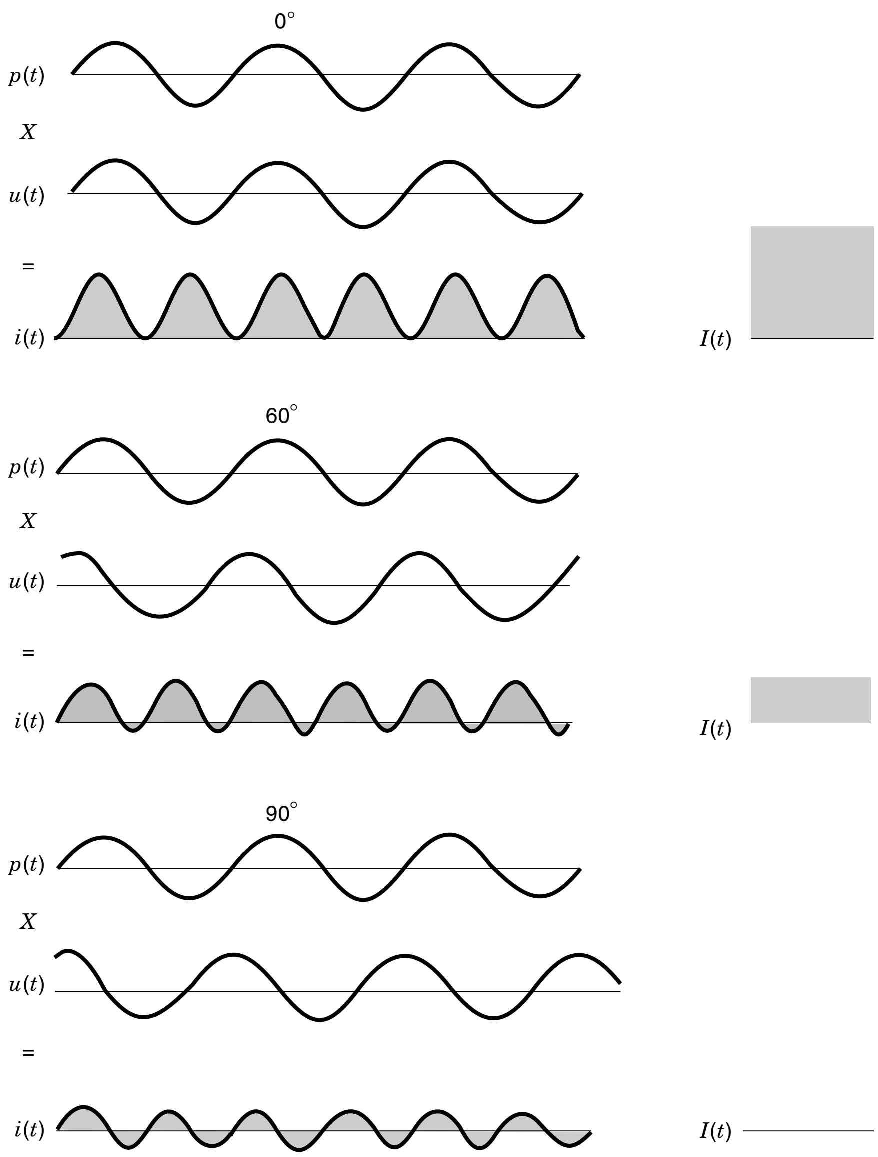

Figure 1. Instantaneous intensity $i(t)$ and time-averaged intensity $I(t)$ for different phase shifts between pressure $p(t)$ and particle velocity $u(t)$ highlight the differences between the two quantities and the effect of the phase shift on the $I(t)$.

The equal loudness contours for pure tones of Fig. 2 were obtained from measurements on human beings and show that the frequency response of the human ear is both nonflat and nonlinear (9). The numbers on the contour indicate the loudness level in phons, with 0 phon corresponding to the threshold of audibility. These curves were obtained under free-field conditions, and the ordinate of the curve is free-field sound pressure level. Thus, the ordinate indicates what pressure amplitude must be applied at any given frequency so that the human observer will perceive a sensation of equal loudness. For example, at a 50 phon loudness, a sound level of 58 dB L_p at 100 Hz sounds as loud as one of 50 dB L_p at 1000 Hz. This demonstrates the nonflatness of the ear's frequency response. Its nonlinearity is manifested by the need for a family of curves for various loudness levels, rather than just a single curve.

The calculation of loudness is crucial in characterizing loudspeakers, whose quality depends on how much the transduced music or speech is loud with respect to background noise. Namely, the loudness efficiency rating is defined as the ratio of the total effective acoustic power produced by the loudspeaker to the available electrical power; here the total effective acoustic power is measured so that it is nearly proportional to the loudness produced by the loudspeaker in the free field (10).

Reference Levels

Sound pressure, intensity, power, and particle velocity levels (L_p, L_I, L_W, and L_u, respectively), are all measured in decibels. Decibels are a ratio of the specified quantity measured against some reference. As previously stated, the pressure reference level is chosen so that it corresponds approximately to the threshold of audibility. Other reference levels have been approximately related to this by using the free field rela-

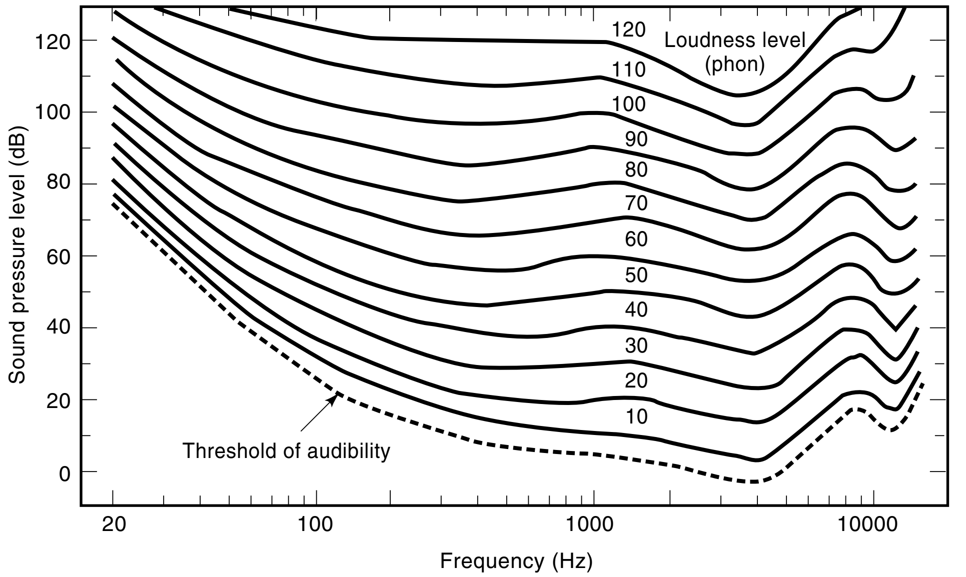

Figure 2. Normal equal loudness contours for pure tones. The levels were measured after the listener left the free plane-wave sound field. (ISO R226-1961)

tions between pressure and intensity, and pressure and particle velocity. In the free field we will obtain the same decibel reading irrespective of whether we measure pressure, intensity, or particle velocity (measured in the direction of propagation). Actually, because round numbers have been chosen for the reference levels, there is a slight difference in levels. The actual difference depends on the value of the acoustic impedance, ρc, of the medium in which the difference is measured. The difference is usually negligible in the air except at high altitudes. To avoid possible confusion with pressure levels, sound power levels are sometimes given in bels (10 dB equals 1 bel).

In the free field the pressure and intensity levels in the direction of propagation are numerically the same. However, intensity measurements in the free field are not needed. In practice, measurements are not performed in the free field, so there will be a difference between the pressure and intensity levels. This difference is an important quantity. It is known variously as the reactivity index, pressure-intensity index, or phase index.

INSTRUMENTS FOR SOUND-LEVEL MEASUREMENTS

Sound-Level Meter

The most commonly utilized instrument for routine sound measurement is the sound-level meter (11,12). This is basic to all sound- and noise-level measurement, particularly that done outside the laboratory. It is actually a measurement system made up of a number of interconnected components. Figure 3 shows a typical arrangement.

The sound pressure $p(t)$ is at first transduced to a voltage by means of the microphone. Microphones generally employ a thin diaphragm to convert pressure to motion. The motion is then converted to voltage by suitable transducers, usually a capacitance, piezoelectric, or electret type.

The output voltage of the microphone generally is quite small and at a high impedance level; thus an amplifier of high input impedance and gain is used at the output of the microphone. This can be a relatively simple ac amplifier, since re-

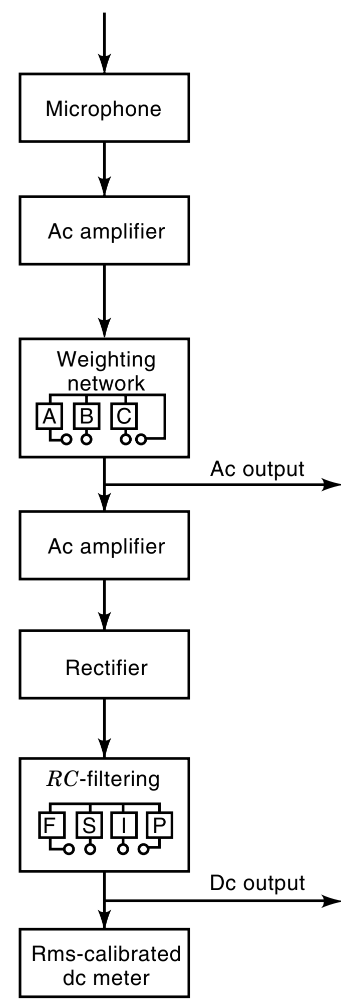

Figure 3. Simplified block diagram of a sound-level meter.

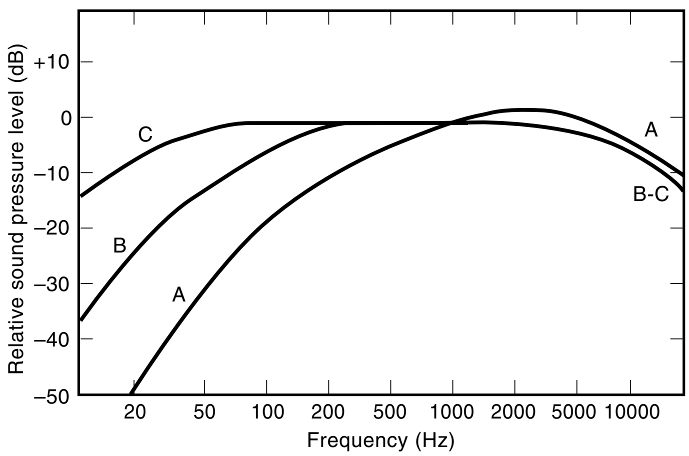

Figure 4. Normalized A, B, and C weighting curves.

sponse to static or slowly varying voltages is not required. Capacitor microphones often use for the first stage a field-effect transistor (FET)-input amplifier built directly into the microphone housing. This close coupling reduces stray capacitance effects by eliminating cables at the high-impedance end.

The weighting networks follow the first amplifier. They are electrical filters whose frequency response is tailored to approximate the frequency response of the average human ear. In fact, the main use of a sound-level meter is not the accurate measurement of pressure but rather the determination of the loudness perceived by human beings; for this reason a flat instrument frequency response is not really desired. The weighting networks of Fig. 3 are electrical filters suitably designed to approximate the human ear's response at different loudness levels so that the instrument reading will respect perceived loudness. Usually three filters are provided: A, B, and C. Figure 4 shows the frequency response of these filters. Some meters also provide a *flat* setting if true pressure measurements are wanted; if a flat setting is not available, the C network is a good approximation of a flat response. Actually, many practical measurements are made by employing the A scale since it is a simple approach that has given good results in many cases and has been written into many standards and codes. The level of sound pressure measured with a sound-level meter and its weighting network is called sound level to distinguish it from the original sound pressure levels. When A-weighting is used, the sound level is given in dB(A) instead of dB.

The output of the weighting network is further amplified, and an output jack is provided to lead this ac signal to an oscilloscope (if observation of the waveform is desired) or to a wave analyzer (if the frequency content of the sound is to be determined). If only the overall sound magnitude is desired, the rms value of the voltage signal must be found. While true rms voltmeters are available, their expense is justifiable only in the highest-grade sound-level meters. Rather, the average value of the ac signal is determined by rectifying and filtering; then the meter measures the obtained dc value on a scale usually calibrated to provide the rms value. This procedure is exact for pure sine waves since there is a precise and known relationship between the average value and the rms value of a sine wave. For nonsinusoidal waves this is not true, but the error is generally small enough to be acceptable for relatively unsophisticated work. The filtering is accomplished by both a simple low-pass *RC* filter and low-pass meter dynamics. Moreover, the filtering time constant also acts on the meter response time. In particular, most meters have a switch that changes the filtering time constant. The *slow* position (1 s time constant) gives a steady, easy-to-read needle position but masks any short-term variations in the rms of the signal. If these short-term variations in the signal are of interest, they may be visually observed on the meter by switching it to the *fast* (125 ms), *impulse* (35 ms), or *peak* (20 μs to 50 μs) position. Figure 5 shows the effect of the different time constant on a sound-level measurement.

Finally, a dc meter calibrated in rms gives the measurement result. Since Eq. (1) establishes a definite relation between sound pressure in μPa and dB, the meter is often directly calibrated in decibel.

Integrating Sound-Level Meter

Integrating sound-level meters are designed to measure continuously the *continuous equivalent level* in a defined time interval T, $L_{\mathrm{eq},T}$ (13). The aim of this parameter is to characterize, with a unique measurement data, a variable noise. The $L_{\mathrm{eq},T}$ of a signal is the level of a hypothetical constant

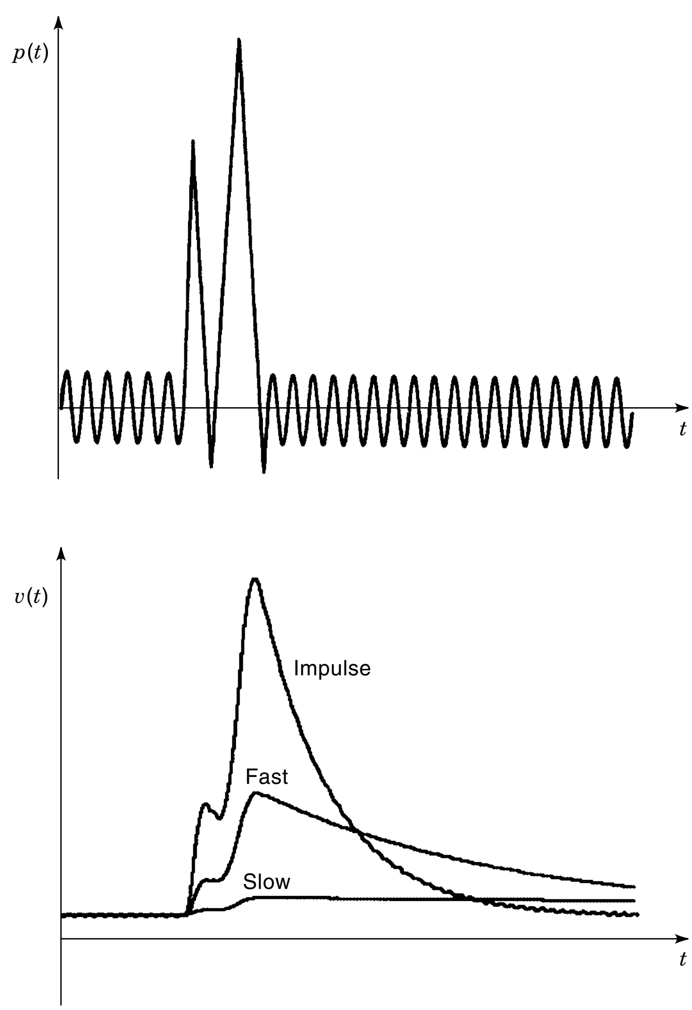

Figure 5. The *RC*-filtering outputs for different time constants in the case of an impulse noise.

level noise that has the same acoustic energy as the original signal in the same period of time and, namely,

$$L_{\mathrm{eq},T} = 10 \log \frac{1}{T} \int_0^T \left(\frac{p(t)^2}{p_o^2} \right) dt \text{ (dB)} \tag{9}$$

where p_0 is the reference pressure (20 μPa) and $p(t)$ is the sound pressure level of the noise under analysis. Usually A-weighted sound levels, rather than sound pressure levels, are considered and then a $L_{\mathrm{Aeq},T}$ is obtained.

If the noise under analysis is constituted by a sequence of constant levels L_{A1}, L_{A2}, . . ., L_{AN}, each one present for a time interval Δt_1, Δt_2, . . ., Δt_N, respectively, or if the available measures are in this form, it is possible to evaluate the $L_{\mathrm{Aeq},T}$, with $T = \Sigma_i \, \Delta t_i$ by the equation

$$L_{\mathrm{Aeq},T} = 10 \log \frac{1}{T} \sum_{i=1}^{N} 10^{L_{\mathrm{A}i}/10} \Delta t_i \text{ [dB(A)]} \tag{10}$$

In the field of ear protection, a more complex formula was employed for a long time and can sometimes still be implemented; this formula has no physical explanation but allows the parameters q [exchange factor, representing the level increment in dB(A) that requires halving the exposure time to obtain a constant risk] and L_{t} (threshold level, namely, a level under which the noise damage can be considered absent) to be introduced:

$$L_{\mathrm{eq},T} = L_{\mathrm{t}} + \frac{q}{\log 2} \log \frac{1}{T} \sum_{i=1}^{N} 10^{\log 2 (L_{\mathrm{A}i} - L_{\mathrm{t}})/q} \Delta t_i \text{ (dB)} \tag{11}$$

which gives back the previous equation for $q = 3$ and $L_{\mathrm{t}} = 0$. The value of q is usually fixed on the basis of the reference standards: In the United States a $q = 5$ is suggested. In Europe a $q = 3$ is suggested. The value of L_{t} is often assumed to be equal to 70 dB.

It is important to emphasize that even if the $L_{\mathrm{eq},T}$ is often referred to as an average noise measurement, a simple arithmetic mean cannot be used to compute its value starting from partial values, since it is a logarithmic average. For example, an 8 h exposure time subdivided into 4 h at 100 dB and 4 h at 80 dB is characterized by an $L_{\mathrm{eq,8h}}$ of 97 dB rather than 90 dB. As a further example, two 80 dB sources give a total $L_{\mathrm{eq},T}$ of 83 dB—namely, 3 dB higher than the one produced by each source.

Another useful parameter found on more elaborated integrating sound-level meters is the *sound exposure level,* $L_{\mathrm{EA},T}$, also measured in dB(A) and often referred to as SEL. This is defined as the level that, lasting for 1 s, has the same acoustic energy as a given noise event lasting for a chosen period of time T. As a measure of acoustic energy, the SEL can be used to compare unrelated noise events, since the time element in its definition is always normalized to one second. The value of T is often chosen as the time interval in which the sound level is no lower than 10 dB of the maximum value, in order to avoid an increase in the measurement time that is not justified by a significant SEL variation.

Some integrating sound-level meters measure the continuous equivalent level over a fixed period of 60 s, $L_{\mathrm{Aeq,60s}}$. There is a certain similarity in use between the quantities $L_{\mathrm{Aeq,60s}}$ and $L_{\mathrm{EA},T}$, since both are based on measurement performed at fixed time intervals. In addition, for any noise event lasting no more than 60 s, there is a fixed difference between its SEL and its $L_{\mathrm{Aeq,60s}}$ (the former is greater by 17.8 dB).

A simplified block diagram of an integrating sound-level meter is shown in Fig. 6. It adds an integrating section fed by the signal already amplified and weighted to the already described features of a sound-level meter (see Fig. 6). This section performs the numeric evaluation of the equivalent sound level on the basis of the previously introduced formulas. It is important here to stress the differences among the average operations performed by this integrating section and those performed by sound-level meters. At first, traditional sound-level meters can average signals for prefixed and, in any case, short time intervals (maximum 1 s for slow op-

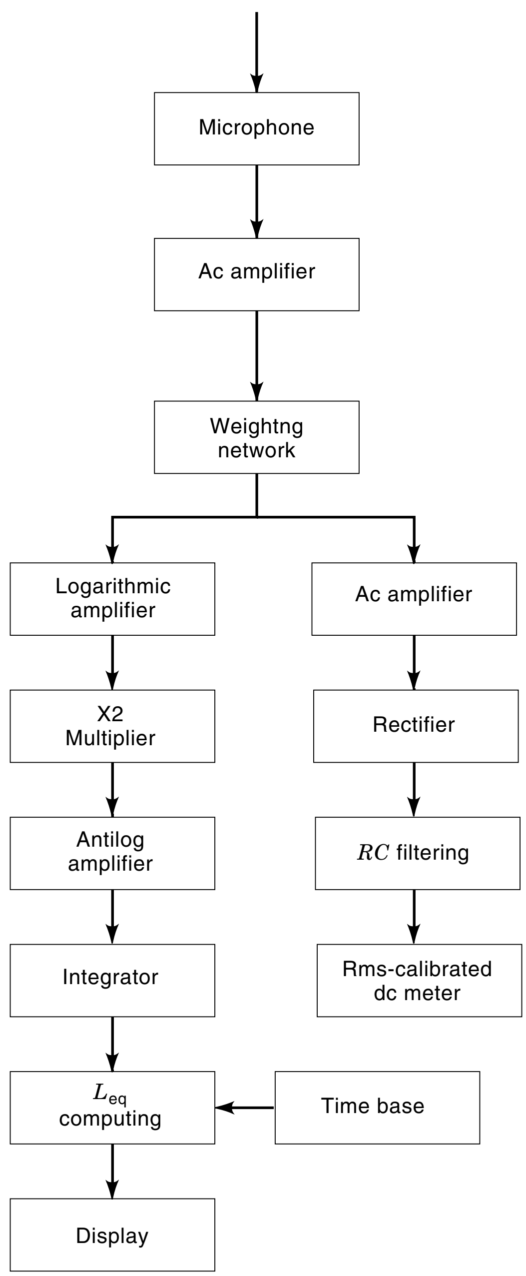

Figure 6. Simplified block diagram of an integrating sound-level meter.

erating), whereas integrating instruments can average for hours. Furthermore, integrating sound-level meters equally weight all the sounds occurring in the considered time window, while traditional sound-level meters give a greater weight to more recent sounds. In fact, the time weighting of traditional sound-level meters decreases exponentially; for example, having chosen a slow 1 s time constant, greater weights are given to sounds occurring less than 1 s previously while sound occurring 10 s previously has very little influence on the meter output. The integrating section has both a better capability in following rapid time evolution of the sound pressure and a higher crest factor (ratio between peak and rms values), which can reach 60 dB.

Noise Dose Meter

Noise dose meters are used to measure the continuous equivalent level of randomly fluctuating noise on an 8 h or longer time interval (14). Namely, a noise dose meter measures D, the percentage of daily noise dose that is allowed by the standards. This instrument is a miniature integrating sound-level meter that uses the A-weighting network and allows the desired exchange factor to be selected. The noise dose meter measures continuously and at the same time reads out the dose as a percentage of the maximum allowable (100%) over an exposure period (usually 8 h). When representative data can be obtained in less time, the reading can be converted easily to an equivalent 8-h exposure.

Two kinds of noise dose meters can be used: personal dose meters, carried by a person, and noise dose meters, installed in a fixed place. The former are usually provided with a small microphone located near the ear; the latter, used to control noisy areas, are usually exposed to a sound level lower than that of a microphone installed on a person immersed in the same sound field. This difference, usually contained within 2 dB, is due to the different sound pressure on a microphone located near a reflective surface: The actual increase depends on the noise spectrum, the noise direction, and the installation.

Sound-Level Statistical Analyzer

This kind of instrument allows the sound level to be analyzed statistically in a defined time interval. It was designed to deal with urban noise, especially traffic noise, but it can be used to measure noise in working areas as well. Its measurement principle is conceptually the following: Suppose we want to evaluate the evolution of the sound level in a 50 dB(A) range—for example, from 40 dB(A) to 90 dB(A)—and that this range is subdivided into 25 contiguous classes of 2 dB(A) each. By sampling the sound level at a constant rate (e.g., every 0.1 s) and by assigning each sample to the corresponding class, the measurement result is obtained as a number of counts for each class or, since the number of counts can be linked to a time interval, as the time percentage of the total investigation time in which the sound level was contained in each dB(A) interval. For example, if we have a 600 s measurement time in which 6000 samples were acquired (0.1 s sampling time) and the class 58 dB(A) to 60 dB(A) was counted 1200 times, the sound level was contained within 58 and 60 dB(A) for 20% of the total measurement period. The result of the statistical distribution is shown in Fig. 7(a). A cumulative frequency distribution [see Fig. 7(b)], is almost always preferred, thus considering the time percentage during which a certain sound level was exceeded.

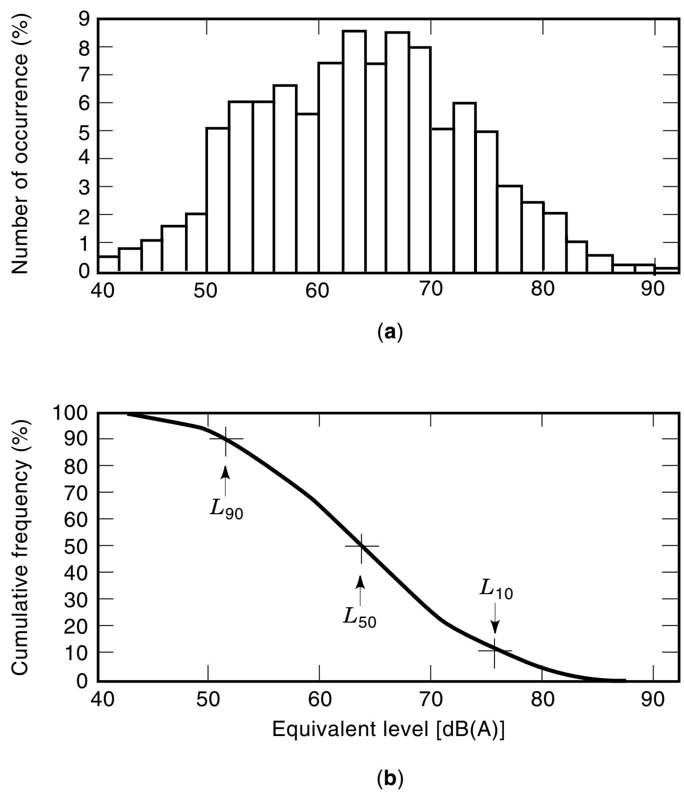

Figure 7. Example of sound-level statistical distribution obtained by using a sound-level statistical analyzer: (a) absolute frequency distribution; (b) cumulative frequency distribution [in this example, L_{90} = 51.7 dB(A); L_{50} = 63.9 dB(A); L_{10} = 75.8 dB(A)].

On the basis of the information gathered according to this criterion, the meter is able to produce, as a final result, together with the sound-level statistical distribution, the equivalent sound level L_{eq} and the so-called statistical levels, typically L_{90} (sound level exceeded for 90% of the measurement time and consequently representative of background noise), L_{50} (sound level exceeded for 50% of the measurement time and consequently representative of the average level), and L_{10} (sound level overcome for 10% of the measurement time and consequently representative of the maximum sound levels).

A statistical analyzer can be used instead of an integrating sound-level meter to evaluate L_{eq}, taking into consideration that this value is not estimated by means of a real integration but by summing the values obtained by sampling, at a constant rate, the sound level that varies according to the meter time constant (slow or fast). This can create problems when dealing with variable or impulsive noises.

Sound-Level Spectrum Analyzer

In some applications a frequency analysis of the acoustic signal is required, in terms of separation of the different frequency components present in the overall signal (15). In fact, the spectrum represents an additional element to the global equivalent sound level expressed in dB(A) to characterize noise correctly, by highlighting the presence of pure tones or high frequencies. Frequency analysis is indispensable for the design of noise control techniques since it allows the main noise sources to be identified and noise-control techniques, materials, and structure to be optimized. This task is carried out by instruments called *spectrum analyzers.*

As far as measurement principles, characteristics, and problems of usage of general-purpose spectrum analyzers as well as of FFT-based digital spectrum analyzers are concerned, the related articles of this Encyclopedia, together with other specialized texts, can be used. In the field of spectrum analysis of acoustic signals, a further possibility is available: performing a sequential frequency analysis, according to Fig. 8. In this case the signal is conceptually applied in parallel to a set of suitable filters. The output for the different bands is read sequentially by an output unit. It is evident that this approach is correct only when a stationary noise has to be measured, or at least a noise that is stationary during the overall measurement time (usually a few minutes). In any case this kind of approach, even if not rigorous, can be used in the absence of more sophisticated instruments to obtain useful qualitative information about the noise spectrum. The frequency range of each filter defines its band or bandwidth; the most commonly used bandwidth in acoustics is the octave (16). An octave is the interval between two frequencies having a ratio of 2, [e.g., from 707 Hz to 1414 Hz (central frequency 1000 Hz, determined as the geometric average between the two frequencies)]. An analyzer that uses this bandwidth is called an octave-band analyzer. Other analyzers use narrower bandwidths to allow a more detailed frequency analysis (e.g., third-octave-band analyzers). Other analyzers use filters with a constant bandwidth in hertz. An ideal filter has a uniform response within its passband and no response out of its passband. Of course this behavior is only approximated by real filters, since their response is not uniform within the passband and not zero outside the passband, giving a meaningful output when relevant frequency components are present immediately out of the passband. The *effective bandwidth* of a filter is the bandwidth of an ideal filter that has the same maximum response and the same output of the real filter when a white noise is fed into both filters. Most analyzers are designed to have an effective bandwidth that is very similar, practically equal, to the filter bandwidth.

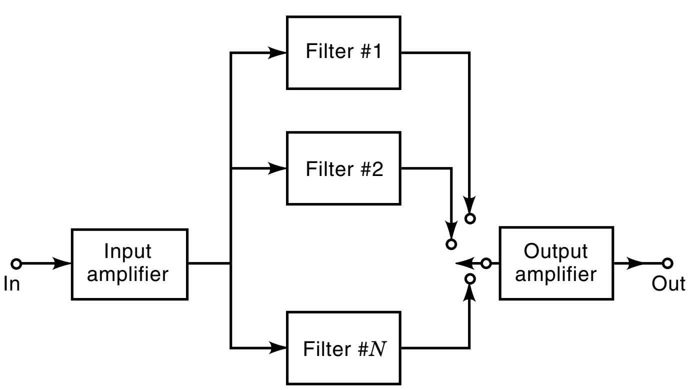

Figure 8. Filtering section of a sequential frequency analyzer.

SOUND INTENSITY ANALYZING SYSTEM

A sound intensity analyzing system consists of a suitable probe and an analyzer. The probe measures the pressure at a pair of microphones, while the analyzer performs the integration and calculation necessary to evaluate the sound intensity.

Sound intensity is the time-averaged product of the pressure and particle velocity. While instantaneous pressure can easily be measured by using a microphone, a direct measurement of particle velocity is not common and requires the use of devices like hot wire anemometers or delicately suspended mica disks (17). However, as the particle velocity is related to the pressure gradient (the rate at which the instantaneous pressure changes with distance), a simpler measurement method can be set up using two identical pressure microphones. The method is based on Newton's second law:

$$\rho \frac{\partial \boldsymbol{u}}{\partial t} = -\nabla p \tag{12}$$

also called Euler's relation, where ρ is the density of air and $\boldsymbol{u}$ the particle velocity.

In one direction, r, we have

$$\rho \frac{\partial u_r}{\partial t} = -\frac{\partial p}{\partial r} \tag{13}$$

Since the pressure gradient is proportional to particle acceleration, particle velocity can be obtained by integrating the pressure gradient with respect to time.

$$u_r = -\frac{1}{\rho} \int \frac{\partial p}{\partial r}\, dt \tag{14}$$

In practice, the pressure gradient can be approximated by measuring the pressures, p_A and p_B, at two closely spaced points, A and B, and dividing the pressure difference $p_A - p_B$ by the transducer separation distance Δr, thus giving the following estimate for the particle velocity component u_r in the direction r:

$$u_r = -\frac{1}{\rho \Delta r} \int (p_A - p_B)\, dt \tag{15}$$

This approximation is valid as long as the separation is small compared with the wavelength ($\Delta r \ll \lambda$).

Practical sound intensity probes therefore consist of two closely spaced pressure microphones, allowing measurement of both pressure and the component of the pressure gradient along a line joining the centers of the microphones. Hence, the magnitude and the direction of the component of the intensity vector along this line is measured.

The probe arrangement of two microphones mounted face to face with a solid spacer in between has been found to have better frequency response and directivity characteristics than side-by-side, back-to-back, or face to face without solid spacer arrangements. As for directivity characteristics of the sound intensity analyzing system, as already mentioned, it is an intensity vector component and not the intensity vector that is measured by this technique. The consequence is that the the-

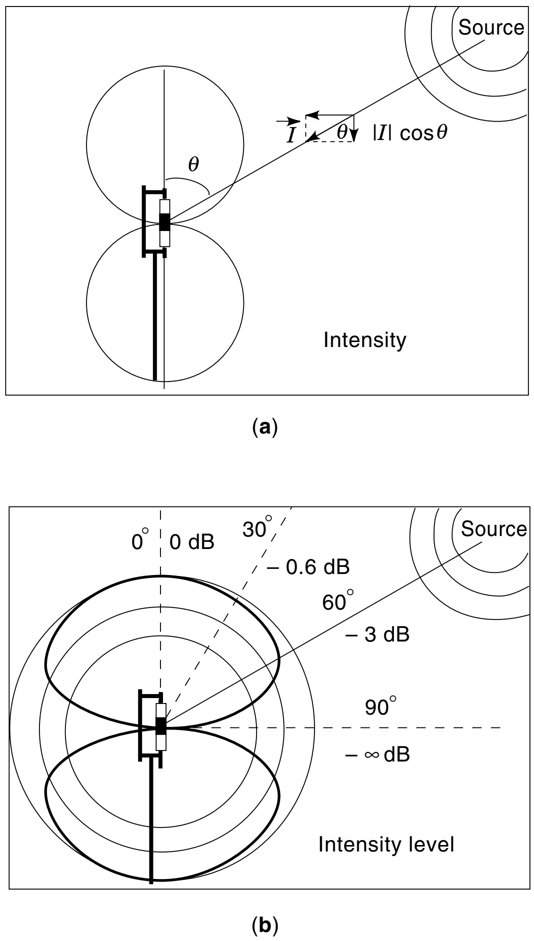

Figure 9. Intensity measurement by two-microphone sound-intensity probe. The difference between intensity (a) and intensity level (b) is shown. The directivity characteristic (cosine characteristic) is also highlighted in (b), where the differences between the measured intensity component and the intensity module are reported in decibels for different angles.

oretical directional characteristic of the sound intensity probe is a cosine function

$$|I_r| = |\boldsymbol{I}|\cos\alpha \tag{16}$$

where α is the angle between the direction of the sound intensity vector and the orientation of the probe (Fig. 9).

For processing the signals from the two microphones, two approaches are in current use today. One approach is a direct method that can be implemented by analog as well as digital techniques. The second approach, the indirect method, can only be implemented by use of a digital technique.

Direct Method

The sound intensity vector component in the direction r is calculated from

$$I_r = -\frac{1}{2\rho\Delta r}\overline{(p_A + p_B)\int(p_A - p_B)\,dt} \tag{17}$$

where sound pressure is taken as being the mean pressure $(p_A + p_B)/2$ between the two microphones, and where the velocity is calculated from Eq. (15) (the superior line indicates au averaging).

Figure 10 shows a block diagram of a practical real-time sound intensity meter (including third-octave digital filters) that follows the equation step by step. Instruments like this have an analysis range from a few hertz to about 10 kHz.

Indirect Method

A dual channel FFT analyzer can be used for intensity calculations within the well-known FFT-limitations. It can be shown (8) that the intensity can be calculated from the imaginary part of the cross-spectrum G_{AB} between the two microphone signals.

$$I_r = -\frac{1}{\omega\rho\Delta r}\,\mathrm{Im}(G_{AB}) \tag{18}$$

Today, this forms a commonly used method of calculating sound intensity. However, a computer is required to carry out the final calculations. Unfortunately, this method has certain disadvantages. One of these is that sound measurement is normally specified in octaves and third octaves, and the calculation of these from narrow-band spectra is a time-consuming procedure, usually requiring multipass analysis and synthesis, which cannot be performed easily in real time.

MICROPHONES

The microphone is the first element of each acoustic measurement chain, since it transduces sound pressure variations into corresponding variations of an electric signal (17). While microphone design is a specialized and complex field with a large amount of technical literature, some of the main considerations will be presented in the section. As for many other

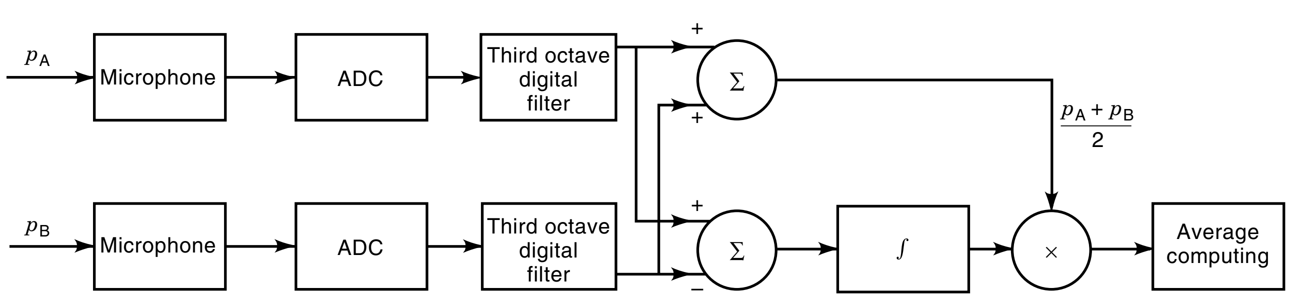

Figure 10. Block diagram of a real-time digital sound intensity meter.

transducers, sensitivity, amplitude, and frequency response are characteristics of major interest. However, sound wavelength and the direction of propagation produce effects on transducer amplitude and frequency response that are aspects of dynamic behavior not regularly encountered in other measurements.

The *pressure response* of a microphone refers to the frequency response relating a uniform sound pressure applied at the microphone diaphragm to the output voltage of the microphone. The pressure response of a given microphone may be estimated theoretically or measured experimentally by one of a number of accepted methods (18).

What is usually desired is the *free-field response* of the microphone (namely, the relation between the microphone output voltage and the sound pressure that existed at the microphone location before the microphone was introduced into the sound field). As a matter of fact, the microphone distorts the pressure field because its acoustical impedance is radically different from that of the medium (air) in which it is immersed. For most purposes the microphone (including its diaphragm) may be considered a rigid body. Sound waves impinging on this body give rise to complex reflections that depend on the sound wavelength (frequency), the direction of propagation of the sound wave, and the microphone size and shape. When the wavelength of the sound wave is very large compared to the microphone dimensions (low frequencies), the effect of reflections is negligible for any angle of incidence between the diaphragm and the wave-propagation direction, and the pressure response equals the free-field response. At very high frequencies, where the wavelength is much smaller than the dimension of the microphone, it acts as an infinite wall, and the pressure at the microphone surface (for waves propagating perpendicular to the diaphragm [0° angle of incidence]) is twice what it would be if the microphone were not there. For waves propagating parallel to the diaphragm (90° incidence angle), the average pressure over the diaphragm surface is zero, giving no output voltage.

For simple geometric shapes, such as spheres and cylinders, theoretical results are available; otherwise an experimental characterization of the actual microphone has to be carried out. Note that for sufficiently low frequencies (below a few thousand hertz) there is little change in pressure because the presence of the microphone and the angle of incidence have little effect. This flat frequency range can be extended by reducing the size of the microphone; however, a smaller size tends to reduce sensitivity. The effect of size is directly related to the relative size of the microphone and the wavelength of the sound. The wavelength λ of sound waves in air is roughly $330/f$ m, where f is the frequency in hertz. When λ becomes comparable to the microphone diaphragm diameter, significant reflection effects can be expected.

Each microphone has a random incidence curve that refers to its response to a diffuse sound field (namely, where the sound is equally likely to come to the microphone from any direction, the waves from all directions are equally strong, and the phase of the waves is random at the microphone position). Such a field may be approximated by constructing a room with highly irregular walls and placing reflecting objects of various sizes and shapes in it. A source of sound placed in such a room gives rise to a diffuse sound field at any point in the room. Microphones calibrated under such conditions are of interest because many sound measurements take place in enclosures that, while not giving perfect random incidence, certainly do not give pure plane waves. Microphone calibrations usually give the pressure response and the free-field response for selected incidence angles, usually 0° and 90°.

Microphones used for engineering measurements are usually capacitor, electret, or piezoelectric types.

A capacitor microphone is constituted by a thin metallic membrane (about 5 μm thickness) that represents the sensing element, mounted in parallel (at about 25 μm) from a rigid posterior plate (back plate), thus forming a capacitor (Fig. 11). The capacitor charge is maintained constant by a constant polarization voltage. When the membrane is solicited by sound pressure, a capacitance and, consequently, a voltage variation occur proportional to the sound pressure. Microphones often have a slow leak (capillary tube) connecting the two sides of the diaphragm, to equalize the average pressure (atmospheric pressure) and prevent bursting of the diaphragm. This is necessary because the (slow) hour-to-hour and day-to-day changes in atmospheric pressure are much greater than the sound pressure fluctuations to which the microphone must respond. (Note that the Eustachian tube of the human ear serves a similar function.) The presence of this leak dictates that microphones will not respond to constant or slowly varying pressures. This is usually not a problem since many measurements involve a human response to the sound, and this is known to extend down to only about 20 Hz. Thus the microphone frequency response need only reach this value, not zero frequency. The typical sensitivity is of about 50 mV/Pa.

Electret-type microphones are related to the capacitor types; however, they require no polarizing voltage since their charge is permanently built into the polymer film that forms the diaphragm. Since the unsupported polymer film would sag and creep excessively, a backup plate with raised points is used. Such microphones are less expensive than the capacitor type, can be used in high-humidity conditions (where the capacitor type may arc over), and result in smaller instruments with lower power consumption. A version that preserves the desirable features of an all-metal diaphragm has also been developed.

Piezoelectric microphones use PZT (or lead zirconate titanate) as a bending beam coupled with the center of a conical

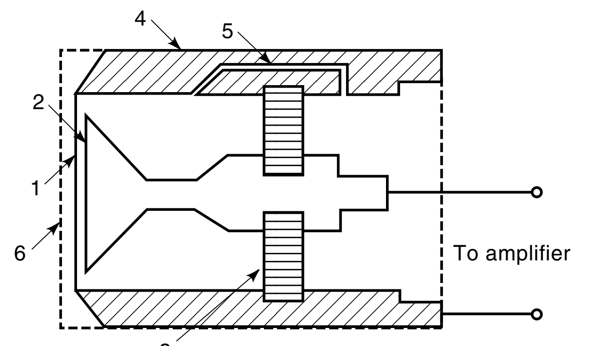

Figure 11. Simplified diagram of a capacitor microphone. (1) Diaphragm; (2) back plate; (3) insulator; (4) case; (5) capillary tube for average pressure equalization; (6) protection grid.

diaphragm of thin metal foil. This kind of microphones offers the following advantages compared to a capacitor microphone: less sensitivity to high humidity; greater robustness; response in a wide frequency range, even with a lower regularity; no polarization voltage required; and excellent stability. On the contrary, it has greater vibration sensitivity; greater sensitivity drift with temperature; and electrical characteristics that vary with temperature.

The selection and use of microphones for critical applications require some background in acoustics, which is beyond the scope of this article; fortunately, useful references are available (2,19).

METER CALIBRATION

Calibration of Level Meters

Traditional and integrating sound-level meters, as any other measurement instrument, must be calibrated periodically in order to verify the stability of their functionality (20). In particular, it is strongly recommended to calibrate a sound-level meter at the beginning and at the end of a set of measurements and at least before and after each day's measurement. If the calibration levels do not coincide, measured data can be corrected for differences contained within 1 dB, but data with greater differences will be discarded.

The apparatus used for the calibration is the calibrator (namely, a sound source that can operate on the basis of different principles but represents the transfer standard between national standards and the meter under calibration). Electromechanical and electroacoustic calibrators are normally used; in the former, sound pressure is generated by the oscillation in a phase opposition of two small pistons, driven by a disk cam; in the latter, a stable oscillator supplies a metallic membrane of a piezoelectric drive. The reference temperature (20°C) and pressure (101.3 kPa) must be assured or the appropriate correction must be applied.

Calibration of Sound Intensity Meters

One of the advantages of using the two-pressure-microphone technique is the ease with which very accurate calibration can be carried out using a pistonphone, which provides a known sound pressure level at a known frequency. As already detailed, the reference values for sound pressure levels and for intensity levels are 20 μPa and 1 pW/m^2, respectively. These reference values have been chosen so that for a freely propagating plane wave, a 0 dB sound pressure level corresponds to a 0 dB sound intensity level.

A calibrated barometer is also necessary to determine the necessary correction for the ambient atmospheric pressure. In fact, both the sound pressure of the pistonphone and the air density are proportional to the ambient pressure. Keeping in mind Eq. (4), the correction term is

$$\begin{aligned}\Delta L_1(p_{\text{amb}}) &= 20 \cdot \log_{10}(p_{\text{amb}}/p_0) - 10 \cdot \log_{10}(p_{\text{amb}}/p_0) \\ &= 10 \cdot \log_{10}(p_{\text{amb}}/p_0)\end{aligned} \tag{19}$$

where p_0 equals 0.101 MPa (1 atm). Therefore, when calibrating the system for use in sound intensity mode, only half the atmospheric pressure correction indicated on the barometer scale must be applied to each microphone.

The air density is inversely proportional to the absolute temperature T, which leads to the temperature correction term:

$$\Delta L_1(T) = 10 \cdot \log_{10}(T/T_0) \tag{20}$$

where T_0 = 293 K (20°C).

In general, these correction terms can often be ignored since they are relatively small; for example, for a temperature of 40°C and an ambient pressure of 75 kPa (Mexico City, 2300 m above sea level) the correction term is only 1.0 dB.

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LEVITATION, MAGNETIC. See MAGNETIC LEVITATION.
LEVITATION, SUPERCONDUCTING. See SUPERCONDUCTING LEVITATION.
LF IONOSPHERIC WAVE PROPAGATION. See SKY WAVE PROPAGATION AT LOW FREQUENCIES.
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