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FUNDAMENTALS OF ACOUSTICS

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Fundamental aspects of acoustics are presented, as they relate to the understanding and application of a methodology for the recognition, evaluation and prevention or control of noise as an occupational hazard. Further information can be found in the specialised literature listed at the end of the chapter.

1.1. PHYSICS OF SOUND

To provide the necessary background for the understanding of the topics covered in this document, basic definitions and other aspects related to the physics of sound and noise are presented. Most definitions have been internationally standardised and are listed in standards publications such as IEC 60050-801(1994).

Noise can be defined as "disagreeable or undesired sound" or other disturbance. From the acoustics point of view, **sound** and **noise** constitute the same phenomenon of atmospheric pressure fluctuations about the mean atmospheric pressure; the differentiation is greatly subjective. What is **sound** to one person can very well be noise to somebody else. The recognition of noise as a serious health hazard is a development of modern times. With modern industry the multitude of sources has accelerated noise-induced hearing loss; amplified music also takes its toll. While amplified music may be considered as sound (not noise) and to give pleasure to many, the excessive noise of much of modern industry probably gives pleasure to very few, or none at all.

Sound (or noise) is the result of pressure variations, or oscillations, in an elastic medium (e.g., air, water, solids), generated by a vibrating surface, or turbulent fluid flow. Sound propagates in the form of longitudinal (as opposed to transverse) waves, involving a succession of compressions and rarefactions in the elastic medium, as illustrated by Figure 1.1(a). When a sound wave propagates in air (which is the medium considered in this document), the oscillations in pressure are above and below the ambient atmospheric pressure.

1.1.1. Amplitude, Frequency, Wavelength And Velocity

Sound waves which consist of a pure tone only are characterised by:

- the **amplitude of pressure changes**, which can be described by the maximum pressure amplitude, p_M, or the root-mean-square (RMS) amplitude, p_{rms}, and is expressed in Pascal (Pa). Root-mean-square means that the instantaneous sound pressures (which can be positive or negative) are squared, averaged and the square root of the average is taken. The quantity, $p_{rms} = 0.707\ p_M$;

- the **wavelength** (λ), which is the distance travelled by the pressure wave during one cycle;
- the **frequency** (f), which is the number of pressure variation cycles in the medium per unit time, or simply, the number of cycles per second, and is expressed in Hertz (Hz). Noise is usually composed of many frequencies combined together. The relation between wavelength and frequency can be seen in Figure 1.2.
- the **period** (T), which is the time taken for one cycle of a wave to pass a fixed point. It is related to frequency by: $T = 1/f$

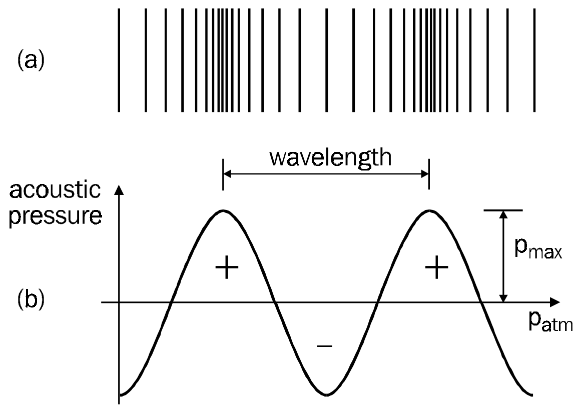


Figure 1.1. Representation of a sound wave.
(a) compressions and rarefactions caused in air by the sound wave.
(b) graphic representation of pressure variations above and below atmospheric pressure.

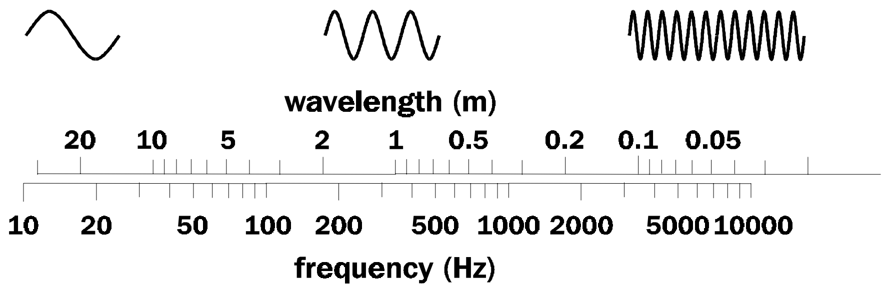


Figure 1.2. Wavelength in air versus frequency under normal conditions (after Harris 1991).

The speed of sound propagation, c, the frequency, f, and the wavelength, λ, are related by the following equation:

$$c = f\lambda$$

- the speed of propagation, c, of sound in air is 343 m/s, at 20°C and 1 atmosphere pressure. At other temperatures (not too different from 20°C), it may be calculated using:

$$c = 332 + 0.6T_c$$

where T_c is the temperature in $°C$. Alternatively the following expression may be used for any temperature and any gas. Alternatively, making use of the equation of state for gases, the speed of sound may be written as:

$$c = \sqrt{\gamma R T_k / M} \qquad (\mathrm{m\ s^{-1}}) \tag{1}$$

where T_k is the temperature in $°K$, R is the universal gas constant which has the value 8.314 J per mole$°$K, and M is the molecular weight, which for air is 0.029 kg/mole. For air, the ratio of specific heats, γ, is 1.402.

All of the properties just discussed (except the speed of sound) apply only to a pure tone (single frequency) sound which is described by the oscillations in pressure shown in Figure 1.1. However, sounds usually encountered are not pure tones. In general, sounds are complex mixtures of pressure variations that vary with respect to phase, frequency, and amplitude. For such complex sounds, there is no simple mathematical relation between the different characteristics. However, any signal may be considered as a combination of a certain number (possibly infinite) of sinusoidal waves, each of which may be described as outlined above. These sinusoidal components constitute the frequency spectrum of the signal.

To illustrate longitudinal wave generation, as well as to provide a model for the discussion of sound spectra, the example of a vibrating piston at the end of a very long tube filled with air will be used, as illustrated in Figure 1.3

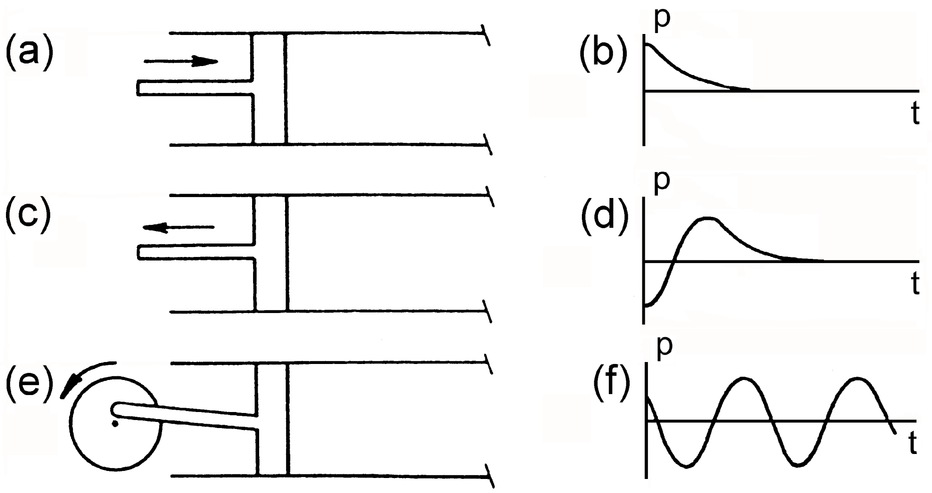


Figure 1.3. Sound generation illustrated. (a) The piston moves right, compressing air as in (b). (c) The piston stops and reverses direction, moving left and decompressing air in front of the piston, as in (d). (e) The piston moves cyclically back and forth, producing alternating compressions and rarefactions, as in (f). In all cases disturbances move to the right with the speed of sound.

Let the piston in Figure 1.3 move forward. Since the air has inertia, only the air immediately next to the face of the piston moves at first; the pressure in the element of air next to the piston increases. The element of air under compression next to the piston will expand forward, displacing the next layer of air and compressing the next elemental volume. A pressure pulse is formed which travels down the tube with the speed of sound, *c*. Let the piston stop and subsequently move backward; a rarefaction is formed next to the surface of the piston which follows the previously formed compression down the tube. If the piston again moves forward, the process is repeated with the net result being a "wave" of positive and negative pressure transmitted along the tube.

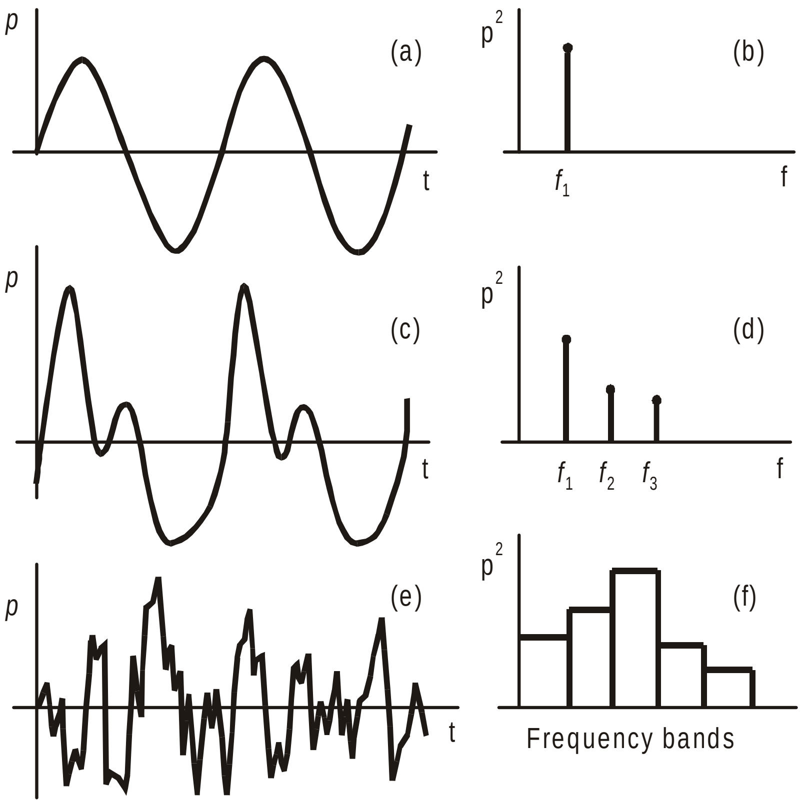


Figure 1.4. Spectral analysis illustrated. (a) Disturbance *p* varies sinusoidally with time *t* at a single frequency f_1, as in (b). (c) Disturbance *p* varies cyclically with time *t* as a combination of three sinusoidal disturbances of fixed relative amplitudes and phases; the associated spectrum has three single-frequency components f_1, f_2 and f_3, as in (d). (e) Disturbance *p* varies erratically with time *t*, with a frequency band spectrum as in (f).

If the piston moves with simple harmonic motion, a sine wave is produced; that is, at any instant the pressure distribution along the tube will have the form of a sine wave, or at any fixed point in the tube the pressure disturbance, displayed as a function of time, will have a sine wave appearance. Such a disturbance is characterised by a single frequency. The motion and corresponding spectrum are illustrated in Figure 1.4a and b.

If the piston moves irregularly but cyclically, for example, so that it produces the waveform shown in Figure 1.4c, the resulting sound field will consist of a combination of sinusoids of several frequencies. The spectral (or frequency) distribution of the energy in this particular sound wave is represented by the frequency spectrum of Figure 1.4d. As the motion is cyclic, the spectrum consists of a set of discrete frequencies.

Although some sound sources have single-frequency components, most sound sources produce a very disordered and random waveform of pressure versus time, as illustrated in Figure

1.4e. Such a wave has no periodic component, but by Fourier analysis it may be shown that the resulting waveform may be represented as a collection of waves of all frequencies. For a random type of wave the sound pressure squared in a band of frequencies is plotted as shown; for example, in the frequency spectrum of Figure 1.4f.

It is customary to refer to spectral density level when the measurement band is one Hz wide, to one third octave or octave band level when the measurement band is one third octave or one octave wide and to spectrum level for measurement bands of other widths.

Two special kinds of spectra are commonly referred to as white random noise and pink random noise. White random noise contains equal energy per hertz and thus has a constant spectral density level. Pink random noise contains equal energy per measurement band and thus has an octave or one-third octave band level which is constant with frequency.

1.1.2. Sound Field Definitions *(see ISO 12001)*

1.1.2.1. Free field

The free field is a region in space where sound may propagate free from any form of obstruction.

1.1.2.2. Near field

The near field of a source is the region close to a source where the sound pressure and acoustic particle velocity are not in phase. In this region the sound field does not decrease by 6 dB each time the distance from the source is increased (as it does in the far field). The near field is limited to a distance from the source equal to about a wavelength of sound or equal to three times the largest dimension of the sound source (whichever is the larger).

1.1.2.3. Far field

The far field of a source begins where the near field ends and extends to infinity. Note that the transition from near to far field is gradual in the transition region. In the far field, the direct field radiated by most machinery sources will decay at the rate of 6 dB each time the distance from the source is doubled. For line sources such as traffic noise, the decay rate varies between 3 and 4 dB.

1.1.2.4. Direct field

The direct field of a sound source is defined as that part of the sound field which has not suffered any reflection from any room surfaces or obstacles.

1.1.2.5. Reverberant field

The reverberant field of a source is defined as that part of the sound field radiated by a source which has experienced at least one reflection from a boundary of the room or enclosure containing the source.

1.1.3. Frequency Analysis

Frequency analysis may be thought of as a process by which a time varying signal in the time domain is transformed to its frequency components in the frequency domain. It can be used for quantification of a noise problem, as both criteria and proposed controls are frequency dependent. In particular, tonal components which are identified by the analysis may be treated somewhat differently than broadband noise. Sometimes frequency analysis is used for noise source identification and in all cases frequency analysis will allow determination of the effectiveness of controls.

There are a number of instruments available for carrying out a frequency analysis of arbitrarily time-varying signals as described in Chapter 6 . To facilitate comparison of measurements between instruments, frequency analysis bands have been standardised. Thus the International Standards Organisation has agreed upon "preferred" frequency bands for sound measurement and analysis.

The widest band used for frequency analysis is the octave band; that is, the upper frequency limit of the band is approximately twice the lower limit. Each octave band is described by its "centre frequency", which is the geometric mean of the upper and lower frequency limits. The preferred octave bands are shown in Table 1.1, in terms of their centre frequencies.

Occasionally, a little more information about the detailed structure of the noise may be required than the octave band will provide. This can be obtained by selecting narrower bands; for example, one-third octave bands. As the name suggests, these are bands of frequency approximately one-third of the width of an octave band. Preferred one-third octave bands of frequency have been agreed upon and are also shown in Table 1.1.

Instruments are available for other forms of band analysis (see Chapter 6). However, they do not enjoy the advantage of standardisation so that the inter-comparison of readings taken on such instruments may be difficult. One way to ameliorate the problem is to present such readings as mean levels per unit frequency. Data presented in this way are referred to as spectral density levels as opposed to band levels. In this case the measured level is reduced by ten times the logarithm to the base ten of the bandwidth. For example, referring to Table 1.1, if the 500 Hz octave band which has a bandwidth of 354 Hz were presented in this way, the measured octave band level would be reduced by $10 \log_{10} (354) = 25.5$ dB to give an estimate of the spectral density level at 500 Hz.

The problem is not entirely alleviated, as the effective bandwidth will depend upon the sharpness of the filter cut-off, which is also not standardised. Generally, the bandwidth is taken as lying between the frequencies, on either side of the pass band, at which the signal is down 3 dB from the signal at the centre of the band.

There are two ways of transforming a signal from the time domain to the frequency domain. The first involves the use of band limited digital or analog filters. The second involves the use of Fourier analysis where the time domain signal is transformed using a Fourier series. This is implemented in practice digitally (referred to as the DFT - digital Fourier Transform) using a very efficient algorithm known as the FFT (fast Fourier Transform). This is discussed further in the literature referenced at the end of the chapter.

1.1.3.1. A convenient property of the one-third octave band centre frequencies

The one-third octave band centre frequency numbers have been chosen so that their logarithms are one-tenth decade numbers. The corresponding frequency pass bands are a compromise; rather than follow a strictly octave sequence which would not repeat, they are adjusted slightly so that they repeat on a logarithmic scale. For example, the sequence 31.5, 40, 50 and 63 has the logarithms 1.5, 1.6, 1.7 and 1.8. The corresponding frequency bands are sometimes referred to as the 15th, 16th, etc., frequency bands.

Table 1.1. Preferred octave and one-third octave frequency bands.

Band number	Octave band center frequency	One-third octave band center frequency	Band limits Lower	Upper
14		25	22	28
15	31.5	31.5	28	35
16		40	35	44
17		50	44	57
18	63	63	57	71
19		80	71	88
20		100	88	113
21	125	125	113	141
22		160	141	176
23		200	176	225
24	250	250	225	283
25		315	283	353
26		400	353	440
27	500	500	440	565
28		630	565	707
29		800	707	880
30	1000	1000	880	1130
31		1250	1130	1414
32		1600	1414	1760
33	2000	2000	1760	2250
34		2500	2250	2825
35		3150	2825	3530
36	4000	4000	3530	4400
37		5000	4400	5650
38		6300	5650	7070
39	8000	8000	7070	8800
40		10000	8800	11300
41		12500	11300	14140
42	16000	16000	14140	17600
43		20000	17600	22500

NOTE: Requirements for filters see IEC 61260; there index numbers are used instead of band numbers. The index numbers are not identical, starting with No.“0" at 1 kHz.

When logarithmic scales are used in plots, as will frequently be done in this book, it will be well to remember the one-third octave band centre frequencies. For example, the centre frequencies given above will lie respectively at 0.5, 0.6, 0.7 and 0.8 of the distance on the scale between 10 and 100. The latter two numbers in turn will lie at 1.0 and 2.0 on the same logarithmic scale.

1.2. QUANTIFICATION OF SOUND

1.2.1. Sound Power (*W*) and Intensity (*I*) (*see ISO 3744, ISO 9614*)

Sound intensity is a vector quantity determined as the product of sound pressure and the component of particle velocity in the direction of the intensity vector. It is a measure of the rate at which work is done on a conducting medium by an advancing sound wave and thus the rate of power transmission through a surface normal to the intensity vector. It is expressed as watts per square metre (W/m^2).

In a free-field environment, i.e., no reflected sound waves and well away from any sound sources, the sound intensity is related to the root mean square acoustic pressure as follows

$$I = \frac{p_{rms}^2}{\rho c} \tag{2}$$

where ρ is the density of air (kg/m^3), and c is the speed of sound (m/sec). The quantity, ρc is called the "acoustic impedance" and is equal to 414 Ns/m^3 at 20°C and one atmosphere. At higher altitudes it is considerably smaller.

The total sound energy emitted by a source per unit time is the sound power, *W*, which is measured in watts. It is defined as the total sound energy radiated by the source in the specified frequency band over a certain time interval divided by the interval. It is obtained by integrating the sound intensity over an imaginary surface surrounding a source. Thus, in general the power, *W*, radiated by any acoustic source is,

$$W = \int_A \boldsymbol{I} \cdot \boldsymbol{n} \, \mathrm{d}A \tag{3}$$

where the dot multiplication of I with the unit vector, $\boldsymbol{n}$, indicates that it is the intensity component normal to the enclosing surface which is used. Most often, a convenient surface is an encompassing sphere or spherical section, but sometimes other surfaces are chosen, as dictated by the circumstances of the particular case considered. For a sound source producing uniformly spherical waves (or radiating equally in all directions), a spherical surface is most convenient, and in this case the above equation leads to the following expression:

$$W = 4\pi r^2 I \tag{4}$$

where the magnitude of the acoustic intensity, *I*, is measured at a distance *r* from the source. In this case the source has been treated as though it radiates uniformly in all directions.

1.2.2. Sound Pressure Level

The range of sound pressures that can be heard by the human ear is very large. The minimum acoustic pressure audible to the young human ear judged to be in good health, and unsullied by too much exposure to excessively loud music, is approximately 20 x 10^{-6} Pa, or 2 x 10^{-10} atmospheres (since 1 atmosphere equals 101.3 x 10^3 Pa). The minimum audible level occurs at about 4,000 Hz and is a physical limit imposed by molecular motion. Lower sound pressure levels would be swamped by thermal noise due to molecular motion in air.

For the normal human ear, pain is experienced at sound pressures of the order of 60 Pa or 6 x 10^{-4} atmospheres. Evidently, acoustic pressures ordinarily are quite small fluctuations about the mean.

A linear scale based on the square of the sound pressure would require 10^{13} unit divisions to cover the range of human experience; however, the human brain is not organised to encompass such a range. The remarkable dynamic range of the ear suggests that some kind of compressed scale should be used. A scale suitable for expressing the square of the sound pressure in units best matched to subjective response is logarithmic rather than linear. Thus the Bel was introduced which is the logarithm of the ratio of two quantities, one of which is a reference quantity.

To avoid a scale which is too compressed over the sensitivity range of the ear, a factor of 10 is introduced, giving rise to the decibel. The level of sound pressure p is then said to be L_p decibels (dB) greater or less than a reference sound pressure p_{ref} according to the following equation:

$$L_p = 10 \log_{10} \frac{p_{rms}^2}{p_{ref}^2} = 20 \log_{10} \frac{p_{rms}}{p_{ref}} = 20 \log_{10} p_{rms} - 20 \log_{10} p_{ref} \qquad \text{(dB)} \qquad (5)$$

For the purpose of absolute level determination, the sound pressure is expressed in terms of a datum pressure corresponding to the lowest sound pressure which the young normal ear can detect. The result is called the sound pressure level, L_p (or SPL), which has the units of decibels (dB). This is the quantity which is measured with a sound level meter.

The sound pressure is a measured root mean square (r.m.s.) value and the internationally agreed reference pressure p_{ref} = 2 x 10^{-5} N m^{-2} or 20 μPa . When this value for the reference pressure is substituted into the previous equation, the following convenient alternative form is obtained:

$$L_p = 20 \log_{10} p_{rms} + 94 \qquad \text{(dB)} \qquad (6)$$

where the pressure p is measured in pascals. Some feeling for the relation between subjective loudness and sound pressure level may be gained by reference to Figure 1.5, which illustrates sound pressure levels produced by some noise sources.

1.2.3. Sound Intensity Level

A sound intensity level, L_I, may be defined as follows:

$$L_I = 10 \log_{10} \frac{\text{(sound intensity)}}{\text{(ref. sound intensity)}} \qquad \text{(dB)} \qquad (7)$$

An internationally agreed reference intensity is 10^{-12} W m^{-2}, in which case the previous equation takes the following form:

$$L_I = 10 \log_{10} I + 120 \qquad \text{(dB)} \qquad (8)$$

Use of the relationship between acoustic intensity and pressure in the far field of a source gives the following useful result:

$$L_I = L_p + 10 \log_{10} \frac{400}{\rho c} \qquad (8a)$$

$$L_I = L_p + 26 - 10 \log_{10}(\rho c) \qquad \text{(dB)} \qquad (9)$$

At sea level and 20°C the characteristic impedance, ρc, is 414 kg m^{-2} s^{-1}, so that for both plane and spherical waves,

$$L_I = L_p - 0.2 \qquad \text{(dB)} \qquad (10)$$

1.2.4. Sound Power Level

The sound power level, L_w (or PWL), may be defined as follows:

$$L_w = 10 \log_{10} \frac{\text{(sound power)}}{\text{(reference power)}} \qquad \text{(dB)} \qquad (11)$$

The internationally agreed reference power is 10^{-12} W. Again, the following convenient form is obtained when the reference sound power is introduced into the above equation:

$$L_w = 10 \log_{10} W + 120 \qquad \text{(dB)} \qquad (12)$$

where the power, W, is measured in watts.

For comparison of sound power levels measured at different altitudes a normalization according to equation (8a) should be applied, see ISO 3745.

1.2.5. Combining Sound Pressures

1.2.5.1. Addition of coherent sound pressures

Often, combinations of sounds from many sources contribute to the observed total sound. In general, the phases between sources of sound will be random and such sources are said to be incoherent. However, when sounds of the same frequency are to be combined, the phase between the sounds must be included in the calculation.

For two sounds of the same frequency, characterised by mean square sound pressures $p_{1\ rms}^2$ and $p_{2\ rms}^2$ and phase difference $\beta_1 - \beta_2$, the total mean square sound pressure is given by the following expression (Bies and Hansen, Ch. 1, 1996).

$$p_{t\ rms}^2 = p_{1\ rms}^2 + p_{2\ rms}^2 + 2[p_1 p_2]_{rms} \cos(\beta_1 - \beta_2) \qquad (13)$$

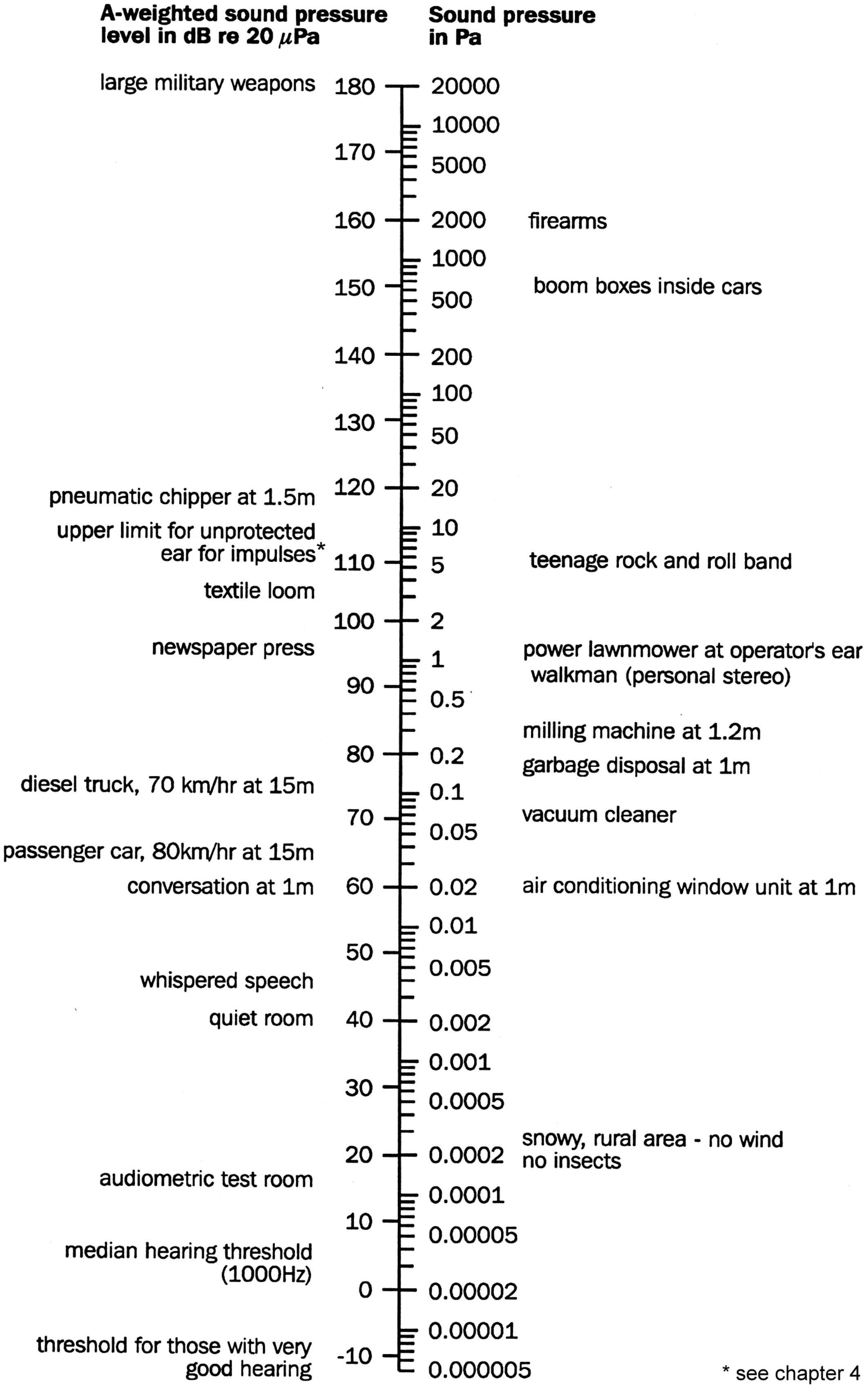


Figure 1.5. Sound levels produced by typical noise sources

When two sounds of slightly different frequencies are added an expression similar to that given by the above equation is obtained but with the phase difference replaced with the frequency difference, Δ, multiplied by time, t. In this case the total mean square sound pressure rises and falls cyclically with time and the phenomenon known as beating is observed, as illustrated in Figure 1.6.

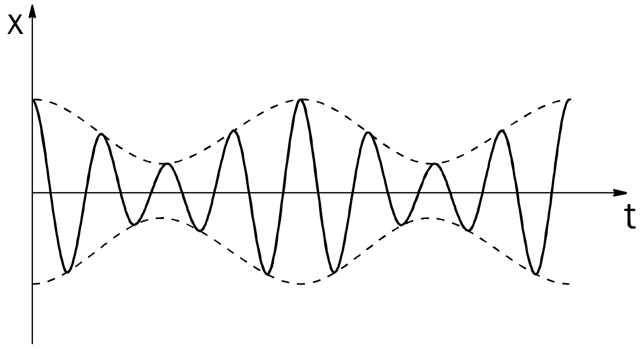


Figure 1.6. Illustration of beating.

1.2.5.2. Addition of incoherent sound pressures (logarithmic addition)

When bands of noise are added and the phases are random, the limiting form of the previous equation reduces to the case of addition of incoherent sounds; that is (Bies and Hansen, Ch. 1, 1996),

$$p_{t\ rms}^{2} = p_{1\ rms}^{2} + p_{2\ rms}^{2} \tag{14}$$

Incoherent sounds add together on a linear energy (pressure squared) basis. A simple procedure which may easily be performed on a standard calculator will be described. The procedure accounts for the addition of sounds on a linear energy basis and their representation on a logarithmic basis. Note that the division by 10 in the exponent is because the process involves the addition of squared pressures.

It should be noted that the addition of two or more levels of sound pressure has a physical significance only if the levels to be added were obtained in the same measuring point.

EXAMPLE

Assume that three sounds of different frequencies (or three incoherent noise sources) are to be combined to obtain a total sound pressure level. Let the three sound pressure levels be (a) 90 dB, (b) 88 dB and (c) 85 dB. The solution is obtained by use of the previous equation.

Solution:

For source (a):

$$p_{1\ rms}^{2} = p_{ref}^{2} \times 10^{90/10} = p_{ref}^{2} \times 10 \times 10^{8}$$

For source (b):

$$p_{2\ rms}^{2} = p_{ref}^{2} \times 6.31 \times 10^{8}$$

For source (c):

$$p_{3\ rms}^{2} = p_{ref}^{2} \times 3.16 \times 10^{8}$$

The total mean square sound pressure is,

$$p_{t\ rms}^{2} = p_{1\ rms}^{2} + p_{2\ rms}^{2} + p_{3\ rms}^{2} = p_{ref}^{2} \times 19.47 \times 10^{8}$$

The total sound pressure level is,

$$L_{pt} = 10 \log_{10}[p_{t\ rms}^{2} / p_{ref}^{2}] = 10 \log_{10}[19.47 \times 10^{8}] = 92.9 \text{ dB}$$

Alternatively, in short form,

$$L_{pt} = 10 \log_{10}\left(10^{90/10} + 10^{88/10} + 10^{85/10}\right) = 92.9 \text{ dB}$$

Table 1.2 can be used as an alternative for adding combinations of decibel values. As an example, if two independent noises with levels of 83 and 87 dB are produced at the same time at a given point, the total noise level will be 87 + 1.5 = 88.5 dB, since the amount to be added to the higher level, for a difference of 4 dB between the two levels, is 1.5 dB.

Table 1.2. Table for combining decibel levels.

Difference between the two db levels to be added										**dB**
0	1	2	3	4	5	6	7	8	9	10
3.0	*2.5*	*2.1*	*1.8*	*1.5*	*1.2*	*1.0*	*0.8*	*0.6*	*0.5*	*0.4*
Amount to be added to the higher level in order to get the total level										***dB***

As can be seen in these examples, it is only when two noise sources have similar acoustic powers, and are therefore generating similar levels, that their combination leads to an appreciable increase in noise levels above the level of the noisier source. The maximum increase over the level radiated by the noisier source, by the combination of two random noise sources occurs when the sound pressures radiated by each of the two sources are identical, resulting in an increase of 3 dB over the sound pressure level generated by one source. If there is any difference in the original independent levels, the combined level will exceed the higher of the two levels by less than 3 dB. When the difference between the two original levels exceeds 10 dB, the contribution of the less noisy source to the combined noise level is negligible; the sound source with the lower level is practically not heard.

1.2.5.3. Subtraction of sound pressure levels

Sometimes it is necessary to subtract one noise from another; for example, when background noise must be subtracted from total noise to obtain the sound produced by a machine alone. The method used is similar to that described in the addition of levels and will be illustrated with an example.

EXAMPLE

The noise level measured at a particular location in a factory with a noisy machine operating nearby is 92 dB(A). When the machine is turned off, the noise level measured at the same location is 88 dB(A). What is the level due to the machine alone?

Solution

$$L_{pm} = 10 \log_{10}\left(10^{92/10} - 10^{88/10}\right) = 89.8 \text{ dB}(A)$$

For noise-testing purposes, this procedure should be used only when the total noise exceeds the background noise by 3 dB or more. If the difference is less than 3 dB a valid sound test probably cannot be made. Note that here subtraction is between squared pressures.

1.2.5.4. Combining level reductions

Sometimes it is necessary to determine the effect of the placement or removal of constructions such as barriers and reflectors on the sound pressure level at an observation point. The difference between levels before and after an alteration (placement or removal of a construction) is called the insertion loss, *IL*. If the level decreases after the alteration, the *IL* is positive; if the level increases, the *IL* is negative. The problem of assessing the effect of an alteration is complex because the number of possible paths along which sound may travel from the source to the observer may increase or decrease.

In assessing the overall effect of any alteration, the combined effect of all possible propagation paths must be considered. Initially, it is supposed that a reference level L_{pR} may be defined at the point of observation as a level which would or does exist due to straight-line propagation from source to receiver. Insertion loss due to propagation over any other path is then assessed in terms of this reference level. Calculated insertion losses would include spreading due to travel over a longer path, losses due to barriers, reflection losses at reflectors and losses due to source directivity effects (see Section 1.3).

For octave band analysis, it will be assumed that the noise arriving at the point of observation by different paths combines incoherently. Thus the total observed sound level may be determined by adding together logarithmically the contributing levels due to each propagation path.

The problem which will now be addressed is how to combine insertion losses to obtain an overall insertion loss due to an alteration. Either before alteration or after alteration, the sound pressure level at the point of observation due to the *i*th path may be written in terms of the *i*th path insertion loss, IL_i, as (Bies and Hansen, Ch. 1, 1996)

$$L_{pi} = L_{pR} - IL_i \tag{15}$$

In either case, the observed overall noise level due to contributions over *n* paths is

$$L_p = L_{pR} + 10 \log_{10} \sum_{i=1}^{n} 10^{-(IL_i/10)} \tag{16}$$

The effect of an alteration will now be considered, where note is taken that, after alteration, the propagation paths, associated insertion losses and number of paths may differ from those before alteration. Introducing subscripts to indicate cases *A* (before alteration) and *B* (after alteration) the overall insertion loss $(IL = L_{pA} - L_{pB})$ due to the alteration is (Bies and Hansen, Ch. 1, 1996),

$$IL = 10 \log_{10} \sum_{i=1}^{n_A} 10^{-(IL_{Ai}/10)} - 10 \log_{10} \sum_{i=1}^{n_B} 10^{-(IL_{Bi}/10)} \tag{17}$$

EXAMPLE

Initially, the sound pressure level at an observation point is due to straight-line propagation and reflection in the ground plane between the source and receiver. The arrangement is altered by introducing a barrier which prevents both initial propagation paths but introduces four new paths. Compute the insertion loss due to the introduction of the barrier. In situation *A*, before alteration, the sound pressure level at the observation point is L_{pA} and propagation loss over the path reflected in the ground plane is 5 dB. In situation *B*, after alteration, the losses over the four new paths are respectively 4, 6, 7 and 10 dB.

Solution:

Using the preceding equation gives the following result.

$$IL = 10\log_{10}\left[10^{-0/10} + 10^{-5/10}\right] - 10\log_{10}\left[10^{-4/10} + 10^{-6/10} + 10^{-7/10} + 10^{-10/10}\right]$$

$$= 1.2 + 0.2 = 1.4 \text{ dB}$$

1.3. PROPAGATION OF NOISE

1.3.1. Free field

A free field is a homogeneous medium, free from boundaries or reflecting surfaces. Considering the simplest form of a sound source, which would radiate sound equally in all directions from a apparent point, the energy emitted at a given time will diffuse in all directions and, one second later, will be distributed over the surface of a sphere of 340 m radius. This type of propagation is said to be spherical and is illustrated in Figure 1.7.

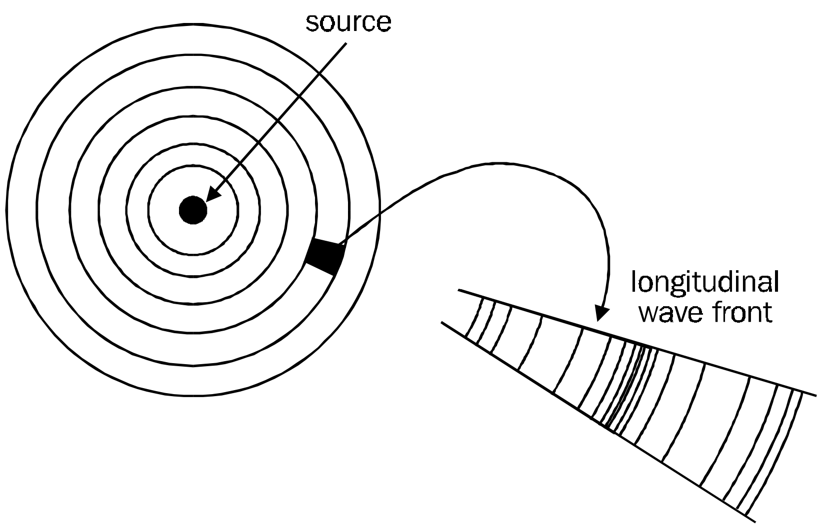


Figure 1.7. A representation of the radiation of sound from a simple source in free field.

In a free field, the intensity and sound pressure at a given point, at a distance r (in meters) from the source, is expressed by the following equation:

$$p^2 = \rho c I = \frac{\rho c W}{4\pi r^2} \tag{18}$$

where ρ and c are the air density and speed of sound respectively.

In terms of sound pressure the preceding equation can be written as:

$$L_p = L_w + 10\log_{10}\left(\frac{\rho c}{400}\right) - 10\log_{10}(4\pi r^2) \tag{19}$$

which is often approximated as:

$$L_p = L_w - 10\log_{10}(4\pi r^2) \tag{20}$$

Measurements of source sound power, L_w, can be complicated in practice (see Bies and Hansen, 1996, Ch. 6). However, if the sound pressure level, L_m, is measured at some reference distance, r_m, from the noise source (usually greater than 1 metre to avoid source near field effects which complicate the sound field close to a source), then the sound pressure level at some other distance, r, may be estimated using:

$$L_p = L_m - 20\log_{10}\left(\frac{r}{r_m}\right) \tag{21}$$

From the preceding expression it can be seen that in free field conditions, the noise level decreases by 6 dB each time the distance between the source and the observer doubles. However, true free-field conditions are rarely encountered in practice, so in general the equation relating sound pressure level and sound power level must be modified to account for the presence of reflecting surfaces. This is done by introducing a directivity factor, Q which may also be used to characterise the directional sound radiation properties of a source.

1.3.2. Directivity

Provided that measurements are made at a sufficient distance from a source to avoid near field effects (usually greater than 1 meter), the sound pressure will decrease with spreading at the rate of 6 dB per doubling of distance and a directivity factor, Q, may be defined which describes the field in a unique way as a function solely of direction.

A simple point source radiates uniformly in all directions. In general, however, the radiation of sound from a typical source is directional, being greater in some directions than in others. The directional properties of a sound source may be quantified by the introduction of a directivity factor describing the angular dependence of the sound intensity. For example, if the sound intensity I is dependent upon direction, then the mean intensity, I_{av}, averaged over an encompassing spherical surface is introduced and,

$$I_{av} = \frac{W}{4\pi r^2} \tag{22}$$

The directivity factor, Q, is defined in terms of the intensity I_θ in direction (θ,ψ) and the mean intensity (Bies and Hansen, Ch. 5, 1996):

$$Q_\theta = \frac{I_\theta}{I_{av}} \tag{23}$$

The directivity index is defined as (Bies and Hansen, Ch. 5, 1996),

$$DI = 10 \log_{10} Q_\theta \tag{24}$$

1.3.2.1. Reflection effects

The presence of a reflecting surface near to a source will affect the sound radiated and the apparent directional properties of the source. Similarly, the presence of a reflecting surface near to a receiver will affect the sound received by the receiver. In general, a reflecting surface will affect not only the directional properties of a source but also the total power radiated by the source (Bies, 1961). As the problem can be quite complicated the simplifying assumption is often made and will be made here, that the source is of constant power output which means that its output sound power is not affected by reflecting surfaces (see Bies and Hansen, 1996 for a more detailed discussion).

For a simple source near to a reflecting surface outdoors (Bies and Hansen, Ch. 5, 1996),

$$W = I \frac{4\pi r^2}{Q} = p_{rms}^2 \frac{4\pi r^2}{\rho c Q} \tag{25a,b}$$

which may be written in terms of levels as

$$L_p = L_w + 10\log_{10}\left(\frac{Q}{4\pi r^2}\right) = L_w + 10\log_{10}\left(\frac{1}{4\pi r^2}\right) + DI \tag{26a,b}$$

For a uniformly radiating source, the intensity I is independent of angle in the restricted region of propagation, and the directivity factor Q takes the value listed in Table 1.3. For example, the value of Q for the case of a simple source next to a reflecting wall is 2, showing that all of the sound power is radiated into the half-space defined by the wall.

Table 1.3. Directivity factors for a simple source near reflecting surfaces.

Situation	Directivity factor, Q	Directivity Index, *DI (dB)*
free space	1	0
centred in a large flat surface	2	3
centred at the edge formed by the junction of two large flat surfaces	4	6
at the corner formed by the junction of three large flat surfaces	8	9

1.3.3. Reverberant fields

Whenever sound waves encounter an obstacle, such as when a noise source is placed within boundaries, part of the acoustic energy is reflected, part is absorbed and part is transmitted. The relative amounts of acoustic energy reflected, absorbed and transmitted greatly depend on the nature of the obstacle. Different surfaces have different ways of reflecting, absorbing and transmitting an incident sound wave. A hard, compact, smooth surface will reflect much more, and absorb much less, acoustic energy than a porous, soft surface.

If the boundary surfaces of a room consist of a material which reflects the incident sound, the sound produced by a source inside the room - the direct sound - rebounds from one boundary to another, giving origin to the reflected sound. The higher the proportion of the incident sound reflected, the higher the contribution of the reflected sound to the total sound in the closed space. This "built-up" noise will continue even after the noise source has been turned off. This phenomenon is called reverberation and the space where it happens is called a reverberant sound field, where the noise level is dependent not only on the acoustic power radiated, but also on the size of the room and the acoustic absorption properties of the boundaries.

As the surfaces become less reflective, and more absorbing of noise, the reflected noise becomes less and the situation tends to a "free field" condition where the only significant sound is the direct sound. By covering the boundaries of a limited space with materials which have a very high absorption coefficient, it is possible to arrive at characteristics of sound propagation similar to free field conditions. Such a space is called an anechoic chamber, and such chambers are used for acoustical research and sound power measurements.

In practice, there is always some absorption at each reflection and therefore most work spaces may be considered as semi-reverberant.

The phenomenon of reverberation has little effect in the area very close to the source, where the direct sound dominates. However, far from the source, and unless the walls are very absorbing, the noise level will be greatly influenced by the reflected, or indirect, sound. The sound pressure level in a room may be considered as a combination of the direct field (sound radiated directly from the source before undergoing a reflection) and the reverberant field (sound which has been reflected from a surface at least once) and for a room for which one dimension is not more than about five times the other two, the sound pressure level generated at distance r from a source producing a sound power level of L_w may be calculated using (Bies and Hansen, Ch. 7, 1996),

$$L_p = L_w + 10\log_{10}\left(\frac{Q}{4\pi r^2} + \frac{4(1-\bar{\alpha})}{S\bar{\alpha}}\right) \qquad (27)$$

where $\bar{\alpha}$ is the average absorption coefficient of all surfaces in the room.

These principles are of great importance for noise control and will be further discussed in more detail in Chapter 5 and 10.

1.4. PSYCHO-ACOUSTICS

For the study of occupational exposure to noise and for the establishment of noise criteria, not only the physical characteristics of noise should be considered, but also the way the human ear responds to it.

The response of the human ear to sound or noise depends both on the sound frequency and the sound pressure level. Given sufficient sound pressure level, a healthy, young, normal human ear is able to detect sounds with frequencies from 20 Hz to 20,000 Hz. Sound characterised by frequencies between 1 and 20 Hz is called infrasound and is not considered damaging at levels below 120 dB. Sound characterised by frequencies in excess of 20,000 Hz is called ultrasound and is not considered damaging at levels below 105 dB. Sound which is most damaging to the range of hearing necessary to understand speech is between 500 Hz and 2000 Hz.

1.4.1. Threshold of hearing

The threshold of hearing is defined as the level of a sound at which, under specified conditions, a person gives 50% correct detection responses on repeated trials, and is indicated by the bottom line in Figure 1.8.

1.4.2. Loudness

At the threshold of hearing, a noise is just "loud" enough to be detected by the human ear. Above that threshold, the degree of loudness is a subjective interpretation of sound pressure level or intensity of the sound.

The concept of loudness is very important for the evaluation of exposure to noise. The human ear has different sensitivities to different frequencies, being least sensitive to extremely high and extremely low frequencies. For example, a pure-tone of 1000 Hz with intensity level of 40 dB would impress the human ear as being louder than a pure-tone of 80 Hz with 50 dB, and a 1000 Hz tone at 70 dB would give the same subjective impression of loudness as a 50 Hz tone at 85 dB.

In the mid-frequency range at sound pressures greater than about 2×10^{-3} Pa (40 dB re 20 μPa SPL), Table 1.4 summarises the subjective perception of noise level changes and shows that a reduction in sound energy (pressure squared) of 50% results in a reduction of 3 dB and is just perceptible to the normal ear.

Table 1.4. Subjective effect of changes in sound pressure level.

Change in sound level (dB)	Change in power Decrease	Increase	Change in apparent loudness
3	1/2	2	just perceptible
5	1/3	3	clearly noticeable
10	1/10	10	half or twice as loud
20	1/100	100	much quieter or louder

The loudness level of a sound is determined by adjusting the sound pressure level of a comparison pure tone of specified frequency until it is judged by normal hearing observers to be equal in loudness. Loudness level is expressed in **phons**, which have the same numerical value as the sound pressure level at 1000 Hz. Attempts have been made to introduce the **sone** as the unit of loudness designed to give scale numbers approximately proportional to the loudness, but it has not been used in the practice of noise evaluation and control.

To rate the loudness of sounds, "equal-loudness contours" have been determined. Since these contours involve subjective reactions, the curves have been determined through psycho-acoustical experiments. One example of such curves is presented in Figure 1.8. It shows that the curves tend to become more flattened with an increase in the loudness level.

The units used to label the equal-loudness contours in the figure are called phons. The lines in figure 1.8 are constructed so that all tones of the same number of phons sound equally loud. The phon scale is chosen so that, at 1 kHz, the number of phons equals the sound pressure level. For example, according to the figure a 31.5 Hz tone of 50 phons sounds equally as loud as a 1000 Hz tone of 50 phons, even though the sound pressure level of the lower-frequency sound is 30 dB higher. Humans are quite "deaf" at low frequencies. The bottom line in Figure 1.8 represents the average threshold of hearing, or minimum audible field (*MAF*).

1.4.3. Pitch

Pitch is the subjective response to frequency. Low frequencies are identified as "low-pitched", while high frequencies are identified as "high-pitched". As few sounds of ordinary experience are of a single frequency (for example, the quality of the sound of a musical instrument is determined by the presence of many frequencies other than the fundamental frequency), it is of interest to consider what determines the pitch of a complex note. If a sound is characterised by a series of integrally related frequencies (for example, the second lowest is twice the frequency of the lowest, the third lowest is three times the lowest, etc.), then the lowest frequency determines the pitch.

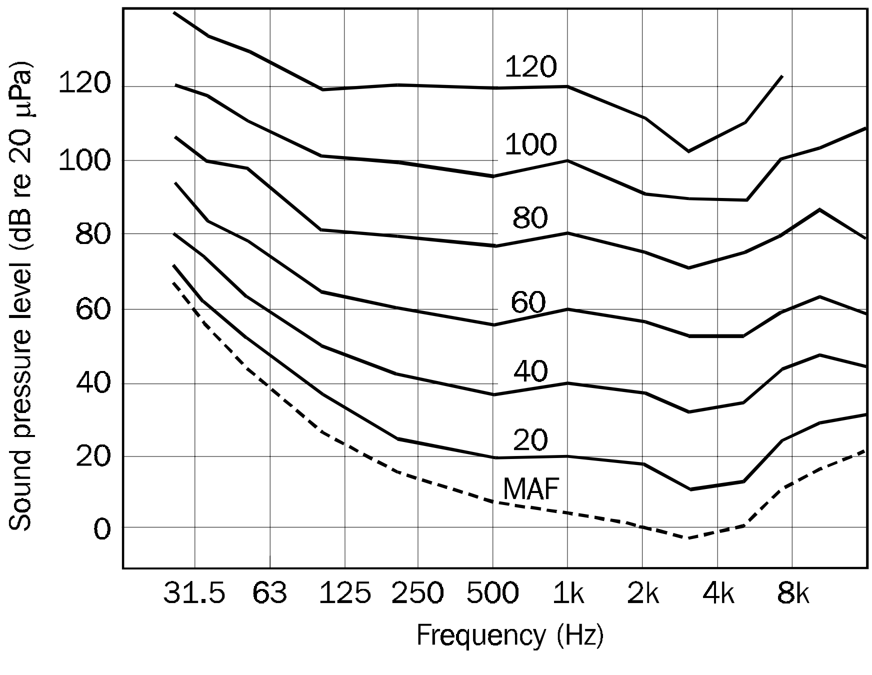


Figure 1.8. Loudness level (equal-loudness) contours, internationally standardised for pure tones heard under standard conditions (ISO 226) . Equal loudness contours are determined relative to the reference level at 1000 Hz. All levels are determined in the absence of the subject, after subject level adjustment. MAF means minimum audible field.

Furthermore, even if the lowest frequency is removed, say by filtering, the pitch remains the same; the ear supplies the missing fundamental frequency. However, if not only the fundamental is removed, but also the odd multiples of the fundamental as well, say by filtering, then the sense of pitch will jump an octave. The pitch will now be determined by the lowest frequency, which was formerly the second lowest. Clearly, the presence or absence of the higher frequencies is important in determining the subjective sense of pitch.

Sense of pitch is also related to level. For example, if the apparent pitch of sounds at 60 dB re 20 µPa is taken as a reference, then sounds of a level well above 60 dB and frequency below 500 Hz tend to be judged flat, while sounds above 500 Hz tend to be judged sharp.

1.4.4. Masking

Masking is the phenomenon of one sound interfering with the perception of another sound. For example, the interference of traffic noise with the use of a public telephone on a busy street corner is probably well known to everyone.

Masking is a very important phenomenon and it has two important implications:

- speech interference, by which communications can be impaired because of high levels of ambient noise;
- utilisation of masking as a control of annoying low level noise, which can be "covered" by music for example.

In general, it has been shown that low frequency sounds can effectively "mask" high frequency sounds even if they are of a slightly lower level. This has implications for warning sounds which should be pitched at lower frequencies than the dominant background noise, but not at such a low frequency that the frequency response of the ear causes audibility problems. Generally frequencies between about 200 and 500 Hz are heard most easily in the presence of typical industrial background noise, but in some situations even lower frequencies are needed. If the warning sounds are modulated in both frequency and level, they are even easier to detect.

Other definitions of masking are used in audiometry and these are discussed in Chapter 8 of this document.

1.4.5. Frequency Weighting

As mentioned in the previous section, the human ear is not equally sensitive to sound at different frequencies. To adequately evaluate human exposure to noise, the sound measuring system must account for this difference in sensitivities over the audible range. For this purpose, frequency weighting networks, which are really "frequency weighting filters" have been developed.

These networks "weight" the contributions of the different frequencies to the **over-all sound level**, so that sound pressure levels are reduced or increased as a function of frequency before being combined together to give an overall level. Thus, whenever the weighting networks are used in a sound measuring system, the various frequencies which constitute the sound contribute differently to the evaluated over-all sound level, in accordance with the given frequency's contribution to the subjective loudness of sound, or noise.

The two internationally standardised weighting networks in common use are the "A" and "C", which have been built to correlate to the frequency response of the human ear for different sound levels. Their characteristics are specified in IEC 60651.

Figure 1.9 and Table 1.5 describe the attenuation provided by the A, and C networks (IEC 60651).

The "A" network modifies the frequency response to follow approximately the equal loudness curve of 40 phons, while the "C" network approximately follows the equal loudness curve of 100 phons, respectively. A "B" network is also mentioned in some texts but it is no longer used in noise evaluations.

The popularity of the A network has grown in the course of time. It is a useful simple means of describing interior noise environments from the point of view of habitability, community disturbance, and also **hearing damage**, even though the C network better describes the loudness of industrial noise which contributes significantly to hearing damage. Its great attraction lies in its direct use in measures of **total noise exposure** (Burns and Robinson, 1970).

When frequency weighting networks are used, the measured noise levels are designated specifically, for example, by dB(A) or dB(C). Alternatively, the terminology A-weighted sound level in dB or C-weighted sound level in dB are often preferred. If the noise level is measured without a "frequency-weighting" network, then the sound levels corresponding to all frequencies contribute to the total as they actually occur. This physical measurement without modification is not particularly useful for exposure evaluation and is referred to as the linear (or unweighted) sound pressure level.

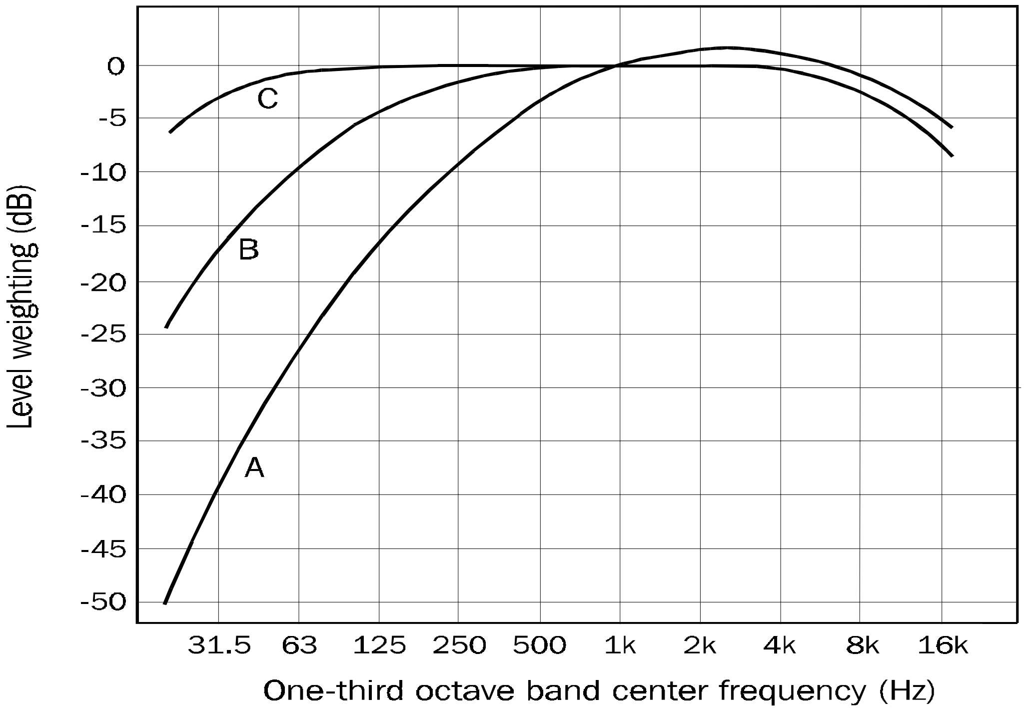


Figure 1.9. Frequency weighting characteristics for A and C networks.

1.5. NOISE EVALUATION INDICES AND BASIS FOR CRITERIA

To properly evaluate noise exposure, both the type and level of the noise must be characterised. The type of noise is characterised by its frequency spectrum and its variation as a function of time. The level is characterised by a particular type of measurement which is dependent on the purpose of the measurement (either to evaluate exposure or to determine the optimum approach for noise control).

Table 1.5. Frequency weighting characteristics for A and C networks (*).

Frequency Hz	Weighting, dB	
	A	C
31.5	- 39	- 3
63	- 26	- 1
125	- 16	0
250	- 9	0
500	- 3	0
1,000	0	0
2,000	1	0
4,000	1	- 1
8,000	- 1	- 3

*This is a simplified table, for illustration purposes. The full characteristics for the A, B and C weighting networks of the sound level meter have been specified by the IEC (IEC 60651).

1.5.1. Types of Noise (*see ISO 12001*)

Noise may be classified as steady, non-steady or impulsive, depending upon the temporal variations in sound pressure level. The various types of noise and instrumentation required for their measurement are illustrated in Table 1.6.

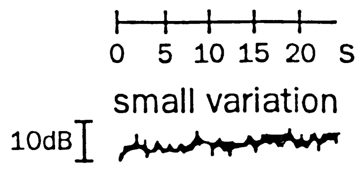
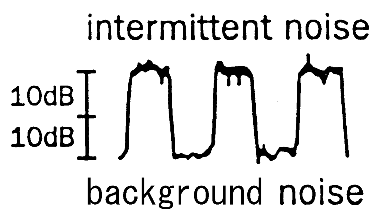
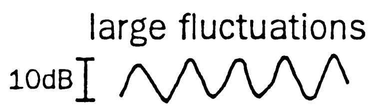
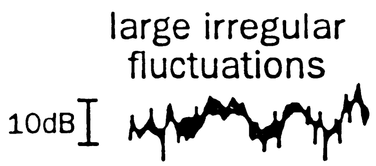
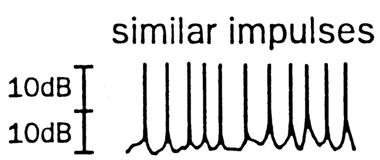
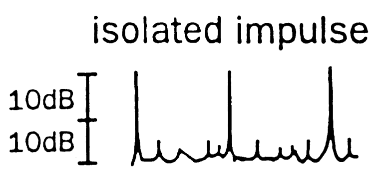
Steady noise is a noise with negligibly small fluctuations of sound pressure level within the period of observation. If a slightly more precise single-number description is needed, assessment by NR (Noise Rating) curves may be used.

A noise is called **non-steady** when its sound pressure levels shift significantly during the period of observation. This type of noise can be divided into intermittent noise and fluctuating noise.

Fluctuating noise is a noise for which the level changes continuously and to a great extent during the period of observation.

Tonal noise may be either continuous or fluctuating and is characterised by one or two single frequencies. This type of noise is much more annoying than broadband noise characterised by energy at many different frequencies and of the same sound pressure level as the tonal noise.

Table 1.6. Noise types and their measurement.

	Characteristics	Type of Source
	Constant continuous sound	Pumps, electric motors, gearboxes, conveyers
	Constant but intermittent sound	Air compressor, automatic machineryduring a work cycle
	Periodically fluctuating sound	Mass production, surface grinding
	Fluctuating non-periodic sound	Manual work, grinding, welding, component assembly
	Repeated impulses	Automatic press, pneumatic drill, riveting
	Single impulse	Hammer blow, material handling, punch press, gunshot, artillery fire

Noise characteristics classified according to the way they vary with time. Constant noise remains within 5 dB for a long time. Constant noise which starts and stops is called intermittent. Fluctuating noise varies significantly but has a constant long term average ($L_{Aeq,T}$). Impulse noise lasts for less than one second.

Type of Measurement	Type of Instrument	Remarks
Direct reading of A-weighted value	Sound level meter	Octave or 1/3 octave analysis if noise is excessive
dB value and exposure time or L_{Aeq}	Sound level meter, Integrating sound level meter	Octave or 1/3 octave analysis if noise is excessive
dB value, L_{Aeq} or noise exposure	Sound level meter Integrating sound level meter	Octave or 1/3 octave analysis if noise is excessive
L_{Aeq} or noise exposure Statistical analysis	Noise exposure meter, Integrating sound level meter	Long term measurement usually required
L_{Aeq} or noise exposure & Check "Peak" value	Integrating sound level meter with "Peak" hold and "C-weighting"	Difficult to assess. More harmful to hearing than it sounds
L_{Aeq} and "Peak" value	Integrating sound level meter with "Peak" hold and "C-weighting"	Difficult to assess. Very harmful to hearing especially close

Intermittent noise is noise for which the level drops to the level of the background noise several times during the period of observation. The time during which the level remains at a constant value different from that of the ambient background noise must be one second or more. This type of noise can be described by

- the ambient noise level
- the level of the intermittent noise
- the average duration of the on and off period.

In general, however, both levels are varying more or less with time and the intermittence rate is changing, so that this type of noise is usually assimilated to a fluctuating noise as described below, and the same indices are used.

Impulsive noise consists of one or more bursts of sound energy, each of a duration less than about 1s . Impulses are usually classified as type A and type B as described in Figure 1.10, according to the time history of instantaneous sound pressure (ISO 10843) . Type A characterises typically gun shot types of impulses, while type B is the one most often found in industry (e.g., punch press impulses). The characteristics of these impulses are the peak pressure value, the rise time and the duration (as defined in Figure 1.10) of the peak.

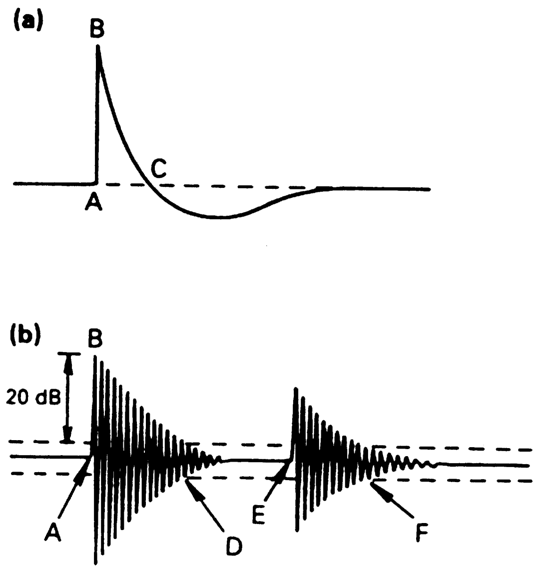


Figure 1.10. Idealised waveforms of impulse noises. Peak level = pressure difference AB; rise time = time difference AB; A duration = time difference AC; B duration = time difference AD (+ EF when a reflection is present).

(a) explosive generated noise.

(b) impact generated noise.

1.5.2. A-weighted Level

The noise level in dB, measured using the filter specified as the A network (see figure 1.9) is referred to as the "A-weighted level" and expressed as dB(A). This measure has been widely used to evaluate occupational exposure because of its good correlation with hearing damage even though the "C" weighting better describes the loudness of industrial noise.

1.5.3. Equivalent Continuous Sound Level (*see ISO 1999*)

Very often industrial noise fluctuates. This can be easily observed as the oscillations in the visual display of a sound level meter in a noisy environment. The equivalent continuous sound level (L_{eq}) is the steady sound pressure level which, over a given period of time, has the same total energy as the actual fluctuating noise. The A-weighted equivalent continuous sound level is denoted L_{Aeq}. If the level is normalised to an 8-hour workday, it is denoted $L_{Aeq,8h}$. If it is over a time period of T hours, then it is denoted $L_{Aeq,T}$, and is defined as follows:

$$L_{Aeq,T} = 10\log_{10}\left(\frac{1}{T}\int_0^T \left(\frac{p_A(t)}{p_0} \right)^2 \mathrm{d}t \right) \qquad (28)$$

where $p_A(t)$ is the time varying A-weighted sound pressure and p_0 is the reference pressure (20µPa). A similar expression can be used to define $L_{Ceq,T}$, the equivalent continuous C-weighted level.

The preferred method of measurement is to use an integrating sound level meter averaging over the entire time interval, but sometimes it is convenient to split the time interval into a number (M) of sub-intervals, T_i, for which values of $L_{Aeq,Ti}$ are measured. In this case, $L_{Aeq,T}$ is determined using,

$$L_{Aeq,T} = 10\log_{10}\left(\frac{1}{T}\sum_{i=1}^{M} T_i \times 10^{(L_{Aeq,T_i})/10} \right) \quad \text{dB} \qquad (29)$$

1.5.4. A-weighted Sound Exposure

Sound exposure may be quantified using the A-weighted sound exposure, $E_{A,T}$, defined as the time integral of the squared, instantaneous A-weighted sound pressure, $p_A^2(t)$ (pa^2) over a particular time period, $T = t_2 - t_1$ (hours). The units are pascal-squared-hours (Pa^2.h) and the defining equation is,

$$E_{A,T} = \int_{t_1}^{t_2} p_A^2(t)\,\mathrm{d}t \qquad (30)$$

The relationship between the A-weighted sound exposure and the A-weighted equivalent continuous sound level, $L_{Aeq,T}$, is

$$E_{A,T} = 4T \times 10^{(L_{Aeq,T} - 100)/10} \qquad (31)$$

A noise exposure level normalised to a nominal 8-hour working day may be calculated from $E_{A,8h}$ using

$$L_{Aeq,8h} = 10\log_{10}\left(\frac{E_{A,8h}}{3.2 \times 10^{-9}} \right) \qquad (32)$$

1.5.5. Noise Rating Systems

These are curves which were often used in the past to assess steady industrial or community noise. They are currently used in some cases by machinery manufacturers to specify machinery noise levels.

The Noise Rating (*NR*) of any noise characterised in octave band levels may also be calculated algebraically. More often the family of curves is used rather than the direct algebraic calculation. In this case, the octave band spectrum of the noise is plotted on the family of curves given in Figure 1.11. The NR index is the value of that curve which lies just above the spectrum of the measured noise. For normal levels of background noise, the NR index is equal to the value of the A-weighted sound pressure level in decibels minus 5. This relationship should be used as a guide only and not as a general rule.

The NR approach actually tries to take into account the difference in frequency weighting made by the ear, at different intensity levels. NR values are especially useful when specifying noise in a given environment for control purposes.

NR curves are similar to the NC (Noise criterion) curves proposed by Beranek (Beranek, 1957). However, these latter curves are intended primarily for rating air conditioning noise and have been largely superseded by Balanced Noise Criterion (NCB) curves, Fig. 1.12**.**

Balanced Noise Criterion Curves are used to specify acceptable noise levels in occupied spaces. More detailed information on NCB curves may be found in the standard ANSI S12.2-1995 and in the proposals for its revision by Schomer (1999). The designation number of an NCB curve is equal to the Speech Interference Level (SIL) of a noise with the same octave band levels as the NCB curve. The SIL of a noise is the arithmetic average of the 500 Hz, 1 kHz, 2 kHz and 4 kHz octave band levels.

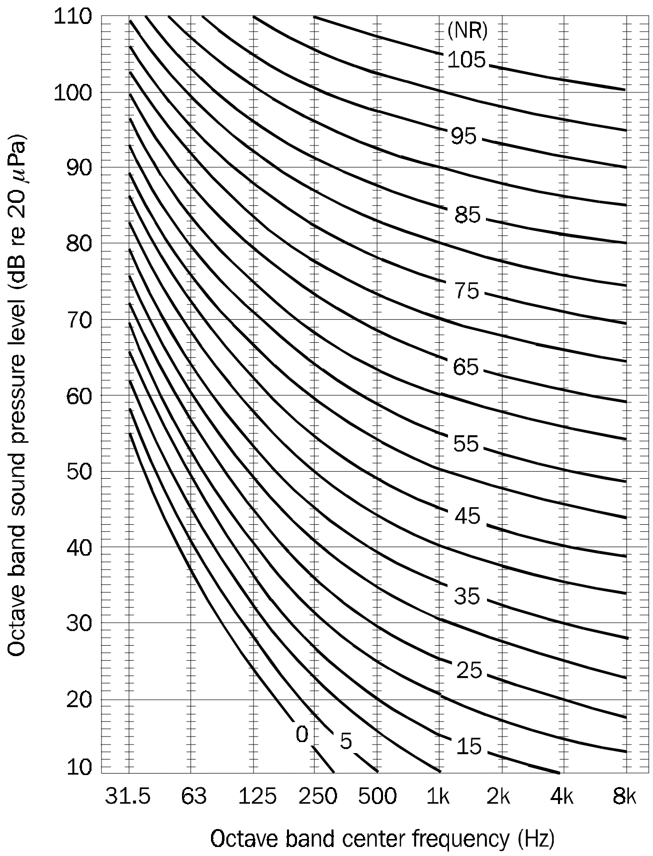


Figure 1.11. Noise rating (NR) curves

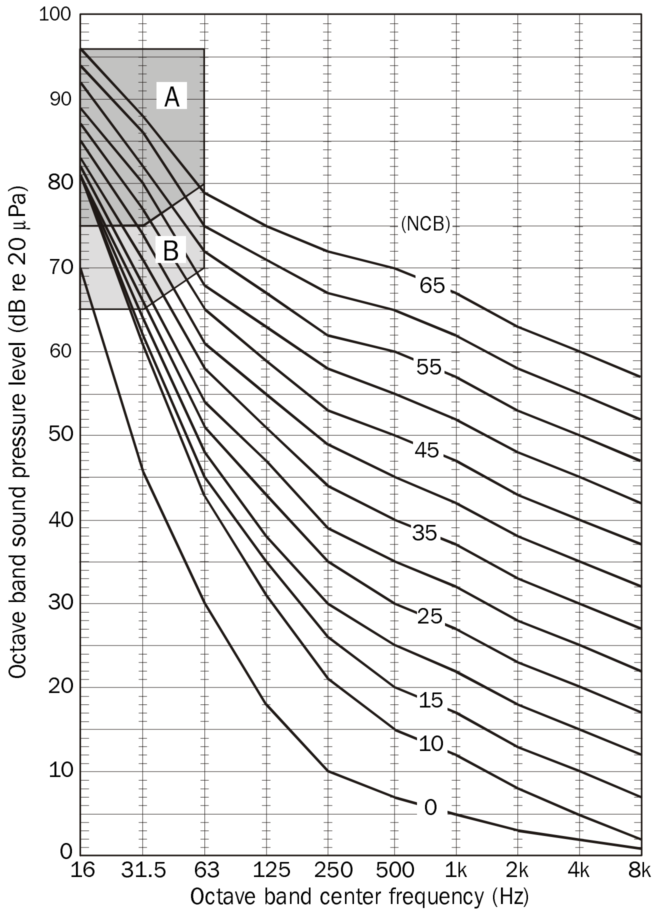


Figure 1.12. Balanced Noise Criterion (NCB) curves. Region A represents exceedance of criteria for readily noticeable vibrations and Region B represents exceedance of criteria for moderately (but not readily) noticeable vibrations.

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INTERNATIONAL STANDARDS

Titles of the following standards related to or referred to in this chapter one will find together with information on availability in chapter 12:

ISO 226, ISO 1999, ISO 2533, ISO 3744, ISO 9614, ISO 12001, ISO 10843,
IEC 60651, IEC 60804, IEC 60942, IEC 61043, IEC 61260.

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