Chapter Nine

The Nature of Sound

Noise is fast becoming one of the more serious pollutants of our environment. At the same time, controlled sound is one of the most important means of communication, as well as providing a medium for enjoyment, relaxation and emotional fulfillment.

In recent years, the level of sensitivity of the average building user has greatly increased, so that both the control of unwanted sound (or noise) and the enhancement of speech and music have become important objectives in building design. Unfortunately, the line between noise and desirable sound cannot always be clearly defined. A sound such as might be produced by a new motor-bike can be at the same time ecstasy to its teenage owner and most objectionable to his next door neighbor. The acoustical design of a building cannot therefore be based wholly on the objective measurement of sound, but is also greatly influenced by the mental state and expectations of its occupants.

9.1 What is Sound?

Sound is the result of vibrations caused by a source that emits pressure fluctuations on a spherical front. The form of the vibrations is cyclic with a compression and a rarefaction completing one cycle (Figure 9.1). Furthermore, these vibrations are in the form of longitudinal waves and will therefore require some type of medium (i.e., solid, liquid or gas) to travel through.

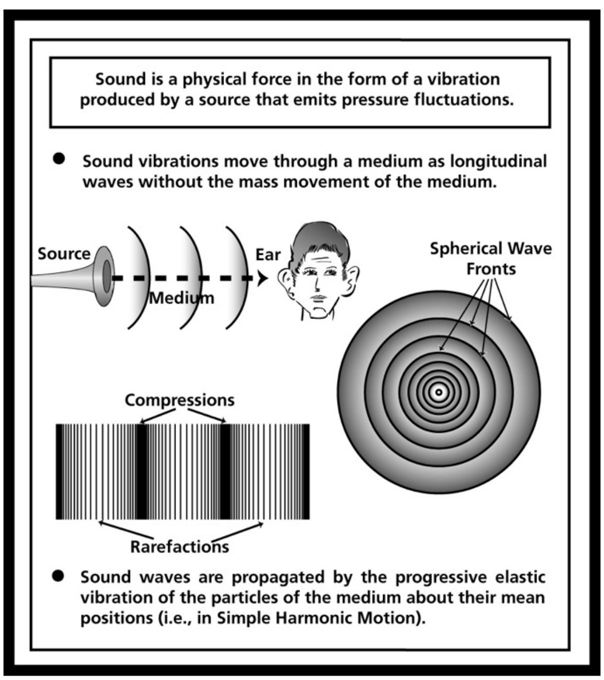

Figure 9.1: Sound as wave motion

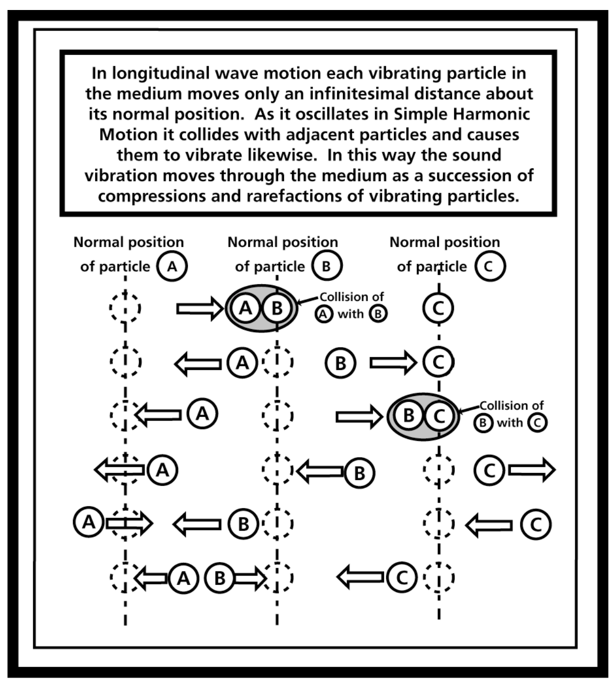

Figure 9.2: Propagation of sound

For example, let us consider the sound generated by the vibration of a guitar string. There will be a region of relative compression where the wave front or vibration is carried onto neighboring particles, followed by a rarefaction in the same region as the particles return to their former equilibrium positions. Accordingly, sound waves are propagated not by the mass movement of the medium, but by the progressive, elastic vibration of the particles of the medium about their mean positions. The particles themselves oscillate about their normal positions (Figure 9.2), setting other particles into similar oscillating motion as they collide with their neighbors. This kind of oscillating motion is very similar to the motion of a pendulum and is referred to in physics as Simple Harmonic Motion.

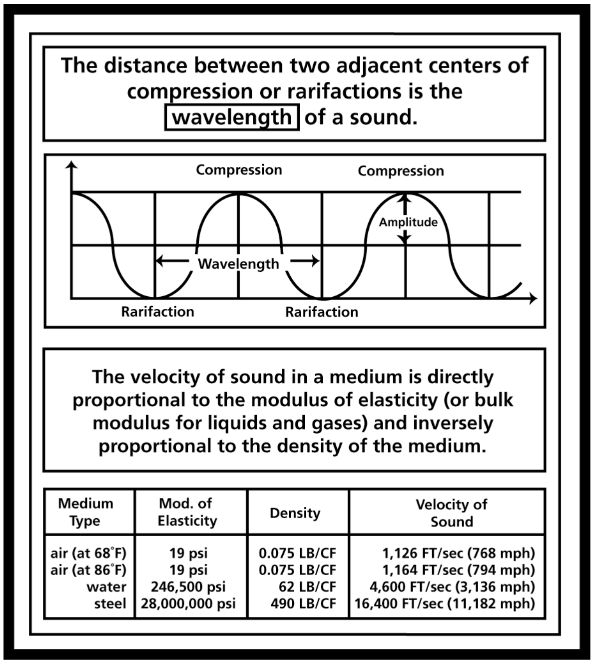

Figure 9.3: Wavelength and velocity of sound

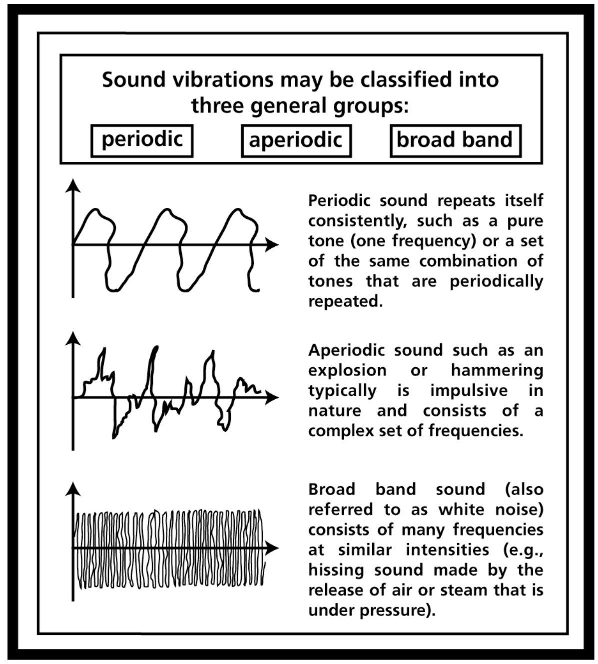

Figure 9.4: Types of sound vibration

As shown in Figure 9.3, the distance between two adjacent centers of compression (or centers of rarefaction) is defined as the wavelength (λ FT), while the rate at which the vibrations occur is called the frequency (f cycles per second (cps) or hertz). The wavelength and frequency of sound are related, so that high frequency sounds have a short wavelength and low frequency sounds have a long wavelength. The frequency range to which the human ear responds is about 30 cps to 20,000 cps, corresponding to wavelengths in the vicinity of 37 FT to 0.02 FT or ¼ IN, respectively.

The velocity of sound in a medium is directly proportional to the modulus of elasticity for solid materials or the bulk modulus for liquids and gasses, and inversely proportional to the density of the medium. Therefore, the velocity of sound will differ according to the medium and in the case of a gaseous medium also somewhat slightly with the temperature of the gas (Figure 9.3), as follows:

velocity of sound in air (at 68˚F) = 1,126 FT/sec or 768 mph
velocity of sound in air (at 86˚F) = 1,164 FT/sec or 794 mph
velocity of sound in water = 4,600 FT/sec or 3,136 mph
velocity of sound in mild steel = 16,400 FT/sec or 11,182 mph

This has implications for the control of noise in buildings, particularly in respect to mechanical equipment such as large fans and cooling equipment used in air-conditioning systems. The noise vibrations generated by this kind of equipment will be transmitted at 12 times the speed of air-borne noise through the structural frame of the building, unless precautions are taken to isolate the equipment by means of damping devices.

For convenience, sound vibrations may be classified (Figure 9.4) into the three general groups of *periodic*, *aperiodic*, and *broad band* sound.

- Periodic sound includes both simple tones of one frequency and complex tones of more than one frequency that are repeated over regular intervals. Examples include the ticking of a clock, the regular beat of a drum, the sound produced by an idling car engine, or the chime of a door bell.
- Aperiodic sound is impulsive in nature and typically produced by a single action such as hammer blows, falling objects, slamming of a door, or an explosion. Such impulsive sounds typically involve a complex combination of frequencies at different intensities.
- Broad band sound, also referred to as *white noise*, consists of many frequencies all of which are at approximately the same intensity. Typical examples include the rapid discharge of air from a car tire, the venting of steam from a pressure vessel, or the background sound in a large commercial aircraft produced by a combination of engine noise, the air-conditioning plant, and the movement of the plane's fuselage through the external air.

For each of these groups of sound vibrations the essential parameters are provided by intensity, frequency composition, and time distribution. The sources of sound that are of immediate concern to the environmental designer are related to the natural environment (e.g., thunder, water flow, and air movement), the artificial mechanical environment (e.g., traffic and industrial processes) and the various methods of communication (e.g., speech, music, radio, and television).

9.2 Objective Units of Sound Measurement

Although the pressures generated by vibrations are very small, the ear is able to respond to a very large range of these pressures. From physics we know that pressure is produced by the action of a force on an area.[1] In the study of acoustics this pressure or intensity is referred to as the amount of sound power either falling on or passing through a particular area. Since the unit of power in the metric system of units is the *watt,* it follows that sound intensity can be objectively measured in *watt* per *square meter* (w/m^2). The range of sound intensities that the human ear is sensitive to is enormous. Experiments have been conducted to show that we can just detect a sound intensity of 10^{-12} watt/m^2 and that our ears start to hurt when we are subjected to a sound intensity of around 10 watt/m^2.

Alternatively, sound can also be measured directly in terms of pressure. The unit of pressure in the metric system of units is the *pascal* (Pa) and expressed in this unit the threshold of hearing and the threshold of pain are equal to 0.00002 Pa and 200 Pa, respectively. Due to this enormous range of audibility, neither sound intensity expressed in watt/m^2 nor sound pressure expressed in

[1] Expressed in more technical terms, pressure is equal to force per unit area.

Pa are a practical and useful measure of the volume of a sound. Imagine a sound meter scale with 10^{13} divisions.

However, once again mathematics comes to the rescue. A very wide numerical range can be conveniently reduced to a much smaller proportional range by applying the concept of logarithms. Expressed logarithmically as powers of 10, the number 10 is equivalent to the logarithmic value of 1, 100 becomes 2, 1000 becomes 3, and so on (Figure 9.5). Therefore, a sound pressure or sound intensity range of 10^{13} becomes a logarithmic range of just 13. The logarithmic equivalent of sound pressure and sound intensity is referred to as Sound Pressure Level (SPL). The decibel (dB) was chosen as the unit of SPL[2].

The SPL of any sound can be calculated as a logarithmic ratio based on the threshold of hearing expressed either in terms of sound pressure (10^{-5} Pa) or sound intensity (10^{-12} watt/m^2), as shown in equations 9.1 and 9.2 below (and Figure 9.6).

$$\textbf{SPL} = \textbf{20} \log_{10} \textbf{P / P}_\textbf{O} \textbf{ (dB)} \quad \ldots\ldots\ldots\ldots \quad \textbf{(9.1)}$$

Thus if the ambient sound pressure at any point in a room is 0.002 Pa, then the corresponding SPL can be calculated by applying equation 9.1 as follows: $20 \log_{10}(0.002 / 0.00002)$ is equal to $20 \log_{10}(100)$ is equal to 40 dB.

Similarly, the Sound Intensity Level (SIL) can be calculated as a logarithmic ratio of any ambient sound intensity compared with the threshold of hearing sound intensity of 10^{-12} watt/m^2.

$$\textbf{SIL} = \textbf{10} \log_{10} \textbf{I / I}_\textbf{O} \textbf{ (dB)} \quad \ldots\ldots\ldots\ldots \quad \textbf{(9.2)}$$

Prior to the availability of electronic calculators, logarithm tables provided a simplified method for the multiplication and division of large numbers.

- **Any number can be expressed in standard scientific form as a power of 10.**

e.g., $1000 = 10^3$ or $0.0001 = 10^{-4}$ or $892.3 = 8.923 \times 10^2$

- **Using 10 as a convenient base, the logarithm ($\log_{10}$) of any number is the exponent of that number expressed as a power of 10.**

$$\text{e.g., } \log_{10}(0.01) = \log_{10}(10^{-2}) = -2$$
$$\log_{10}(10.0) = \log_{10}(10^{1}) = 0$$
$$\log_{10}(100) = \log_{10}(10^{2}) = 2$$

- **Logarithm tables simplify calculation as follows:**

$$\text{i.e., } A \times B = \log_{10}(A) + \log_{10}(B)$$
$$A \div B = \log_{10}(A) - \log_{10}(B)$$
$$\log_{10}(A^n) = n \log_{10}(A)$$

Figure 9.5: The logarithmic scale

The human ear is sensitive to a very large range of pressures or sound intensities. This can be reduced to a more manageable *logarithmic* scale of Sound Pressure Level (SPL) or Sound Intensity Level (SIL).

- **In the Metric system the unit of *pressure* is *Pascal (Pa)* and the *reference sound pressure* (i.e., threshold of hearing) is *0.00002 Pa*:**

$$\text{SPL} = 20\log_{10}\left[\frac{P}{0.00002}\right] \text{(dB)}$$

- **Also, the unit of *power* is the *watt*. Since sound intensity is sound power falling on unit area it follows that the unit of sound intensity is *watt/square meter (w/m²)*. The *reference sound intensity* (i.e., threshold of hearing) is *10^{-12} w/m²*:**

$$\text{SIL} = 10\log_{10}\left[\frac{I}{10^{-12}}\right] \text{(dB)}$$

Figure 9.6: Sound Pressure Level (SPL)

We often refer to human hearing as perception on a logarithmic scale. As can be seen in Figures 9.7 and 9.8, the perception of loudness corresponds more realistically to a

[2] The decibel unit is named after Alexander Graham Bell (1847-1922) the inventor of the telephone.

logarithmic scale. The difference in SPL between a quiet office and a busy or noisy office is just 20 dB. The smallest difference in SPL that is just perceptible is 3 dB, while we require a difference of at least 5 dB for the change in loudness to be clearly noticeable.

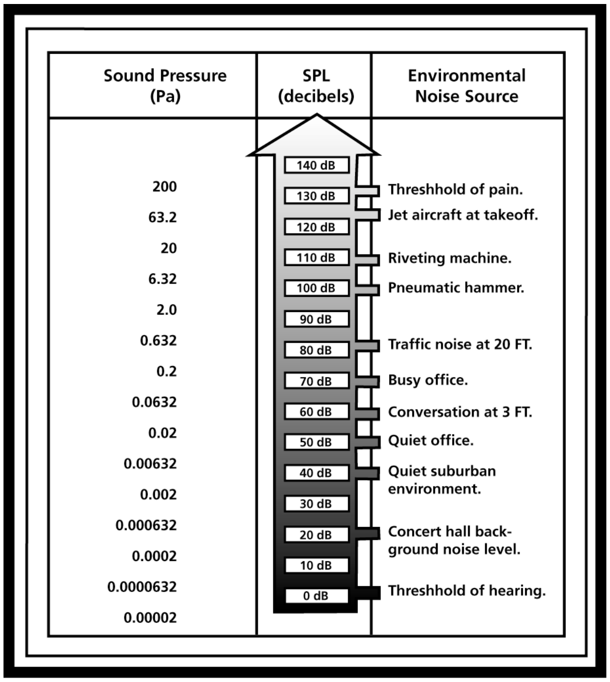

Figure 9.7: Range of sound pressures and equivalent Sound Pressure Levels (SPL)

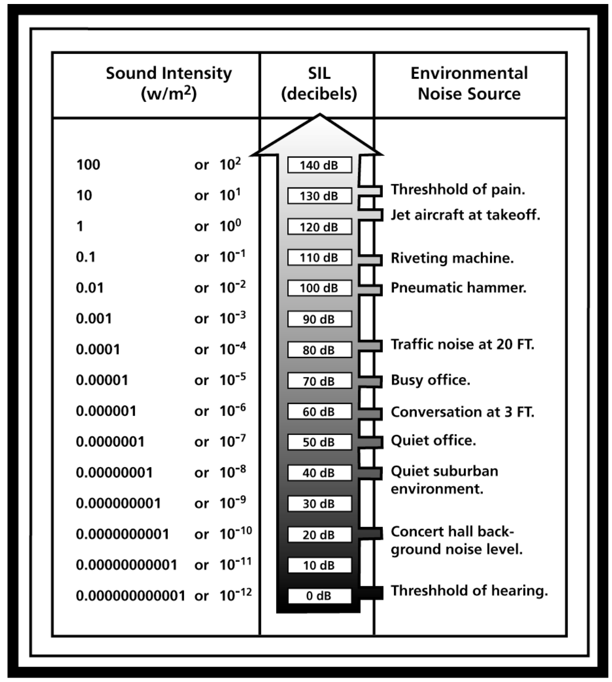

Figure 9.8: Range of sound intensities and equivalent Sound Intensity Levels (SIL)

The reader may well ask: What is the difference between SPL and SIL? In the field of building acoustics, they are considered to be numerically equal.[3] For example, if a sound pressure of 0.2 Pa is equal to a sound intensity of 0.0001 watt/m^2, then the following calculations using equations 9.1 and 9.2 will show that they both produce the same SPL of 80dB.

$$\mathbf{SPL} = \mathbf{20\ log_{10}\ P / P_O\ (dB)} \quad \mathbf{(9.1)}$$
$$= 20\ \log_{10} [0.2 / 0.00002]$$
$$= 20\ \log_{10} [10{,}000] = 20\ (4) = \mathbf{80\ dB}$$

$$\mathbf{SIL} = \mathbf{10\ log_{10}\ I / I_O\ (dB)} \quad \mathbf{(9.2)}$$
$$= 10\ \log_{10} [0.0001 / 10^{-12}]$$
$$= 10\ \log_{10} [100{,}000{,}000] = 10\ (8) = \mathbf{80\ dB}$$

The fact is that it is much easier to measure sound pressure than sound intensity, using a microphone. Such meters are relatively simple to construct and calibrate. Of course, they produce readings in SPL by automatically converting the measured sound pressure into its logarithmic equivalent dB value.

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3 Strictly speaking, from the point of view of a physicist they are not exactly equal.

9.3 Addition, Deletion, and Reduction of Sound Pressure Levels

When two sounds of equal or unequal intensities are added, the effective sound pressure leve1 is not simply the sum of the two individual sound pressure levels. This is due to the fact that the decibel scale is a logarithmic scale, and the calculation of a combined sound pressure level therefore requires the individual sound pressure levels to be converted to sound pressures before they can be added. After addition, the effective sound pressure is converted back to an equivalent sound pressure level in decibels. Accordingly, the addition of two sounds of SPLs L_1 and L_2 with corresponding sound intensities of SILs I_1 and I_2 occurring together will proceed as follows:

$$L_1 = 10 \log (I_1 / I_O)$$

$$I_1 / I_O = \text{antilog } (0.1\, L_1)$$

$$I_1 = I_O\, [\text{antilog } (0.1\, L_1)], \text{ and similarly}$$

$$I_2 = I_O\, [\text{antilog } (0.1\, L_2)]$$

Therefore, the composite SPL L, is given by:

$$L = 10 \log (I_1 + I_2) / I_O$$

$$\mathbf{L = 10 \log [\text{antilog } (0.1\, L_1) + \text{antilog } (0.1\, L_2)] \;\; (dB) \;\; \ldots\ldots\ldots\ldots\ldots \;\; (9.3)}$$

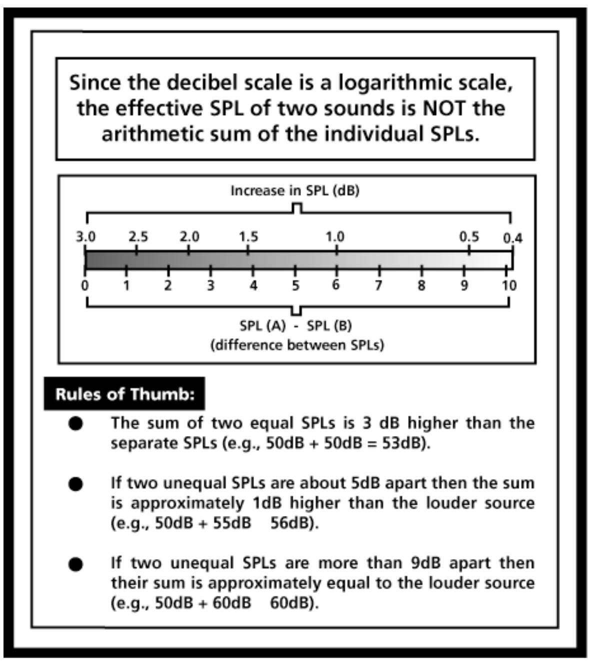

Figure 9.9: Addition of two SPLs

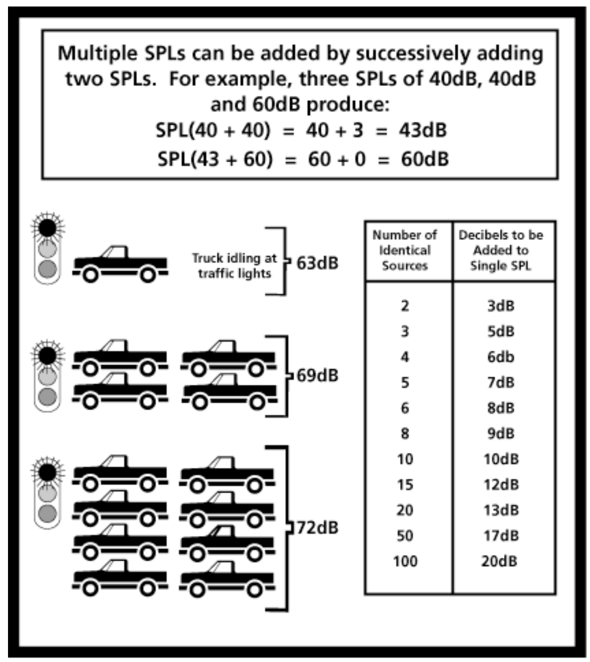

Figure 9.10: Addition of multiple SPLs

As a rule of thumb, the scale shown in Figure 9.9 indicates that the sum of two equal SPLs is always 3 dB higher than the separate levels.[4] In the case of unequal sound pressure levels we may adopt the general rule, that if one level is more than 9 dB higher than another the sum may simply be taken as the larger of the separate values (e.g., for two levels of 40 dB and 50 dB the sum is 50.4 dB, which is close enough to 50 dB).

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[4] This is readily verified as follows: if $L_1 = L_2 = 40$ dB, then $L_{1+2} = 10 \log_{10} (10{,}000 + 10{,}000) = 43$ dB.

As shown in Figure 9.10, the addition of multiple SPLs can proceed in pairs. Alternatively, the simple table shown in Figure 9.10 can be used. Therefore, the combined SPL produced by 20 trucks, each emitting 80 dB, would be 93 dB (i.e., 80 + 13 = 93 dB).

The deletion of SPLs can be dealt with by reversing the steps for addition. However, this task is greatly simplified by reference to the following two tables:

Table 9.1: Perception of SPL Changes

Human Perception of Change in SPLs	
Change in SPL (dB)	Human Perception
1 dB	imperceptible
3 dB	just perceptible
5 dB	clearly noticeable
10 dB	major change

Table 9.2: Deletion of Sound Sources

Difference in dB between total SPL and source to be deleted	Reduction in dB of the total SPL due to the deletion
1 dB	7 dB
2 dB	4 dB
3 dB	3 dB
4 to 5 dB	2 dB
6 to 9 dB	1 dB

We can see from Table 9.1 that a change in SPL of 3 dB is only just perceptible. The following example will illustrate the importance of this experimentally verified finding. Let us assume that there are four automatic riveting machines located in close proximity in a particular part of a factory. The ambient SPL in that part of the factory, mostly produced by these four machines, is 89 dB. Although all four of the riveting machines are of the same type, one is an older model and consequently generates more noise than the other three machines. According to recent sound measurements each of the new models generates 75 dB, while the old model generates 87 dB. Is it warranted on the basis of noise reduction alone to replace the older riveting machine with a new model? The difference between the SPL generated by the old machine and the ambient SPL is 2 dB (i.e., 89 dB – 87 dB). From Table 9.2 we see that the removal of the older model would decrease the overall SPL by 4 dB to 85 dB. According to Table 9.1, this would be a perceptible reduction in the overall SPL and therefore worthwhile.[5]

By how much a given sound is reduced over distance depends on whether the sound is produced by a *point source*, a *line source*, or an *area source*. In the case of a point source the precise reduction in SPL at a distance of *D* FT from a known source can be calculated on a proportional basis as follows. If SPL_1 is the sound pressure level at a distance of D_1 FT from the source then the sound pressure level SPL_2 at a distance of D_2 FT from the source is given by:

$$\mathbf{SPL_2 = SPL_1 - 20 \log_{10} [D_2 / D_1] \ (dB) \ \ldots\ldots\ldots\ldots \ (9.4)}$$

Using equation 9.4 we can, for example, calculate the SPL at 10 FT, 20 FT, and 40 FT from the source, if the SPL is 90 dB at a distance of 1 FT from the source.

$$SPL_{10} = 90 - 20 \log_{10} [10 / 1] = 90 - 20 [1.0] = 70 \text{ dB}$$

[5] It should be noted that according to Figure 9.10 the addition of the fourth car will increase the combined SPL generated by four cars by 1 dB (i.e., from 5 dB for three equal sound sources to 6 dB for four equal sound sources) to 81 dB.

$$SPL_{20} = 90 - 20 \log_{10} [20 / 1] = 90 - 20 [1.3] = 64 \text{ dB}$$

$$SPL_{40} = 90 - 20 \log_{10} [40 / 1] = 90 - 20 [1.6] = 58 \text{ dB}$$

Or, what would be the SPL at 10 FT from the source if the SPL of 90 dB had been recorded at a distance of 2 FT from the source?

$$SPL_{10} = 90 - 20 \log_{10} [10 / 2] = 90 - 20 [0.7] = 76 \text{ dB}$$

From a practical point of view, definitions of a *point source* and a *line source* are provided in Figure 9.11. As a rule of thumb (Figure 9.12) the SPL of a point source is reduced by about 6 dB with each doubling of distance and, in the case of a line source such as a traveling car, by about 3 dB for each doubling of distance.

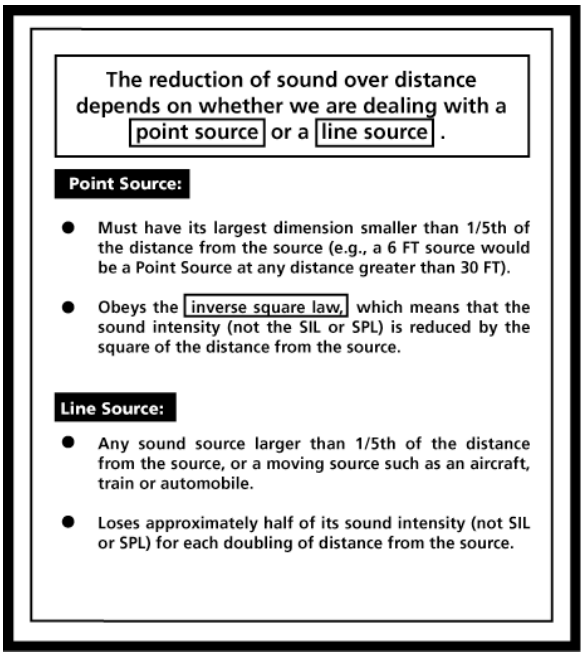


Figure 9.11: Reduction of sound in the Environment

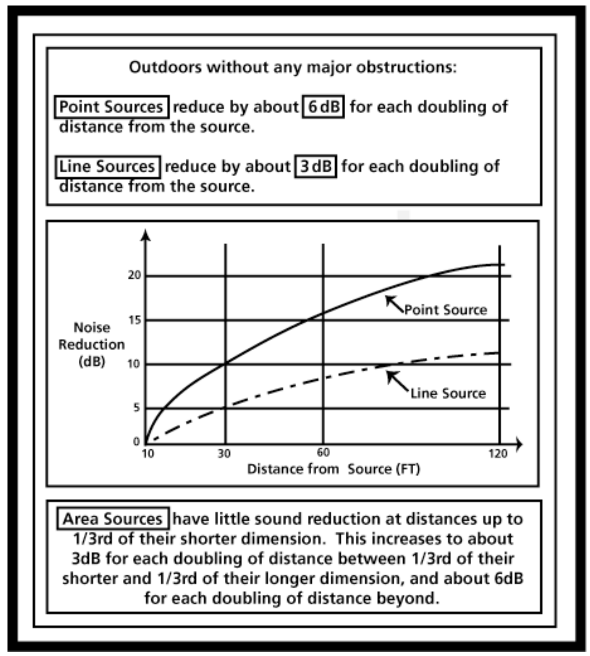


Figure 9.12: Comparison of *point*, *line*, and *area* sound sources

In comparison, an *area source* experiences little sound reduction at distances of up to one third of its shorter dimension. However, this increases to about 3 dB for distances between one third of the shorter dimension to one third of the longer dimension, and to about 6 dB for each doubling of distance beyond.

9.4 The Concept of Octave Bands

As briefly mentioned previously in Section 9.1, the human ear is sensitive to a very wide range of frequencies ranging from around 30 Hz to 20,000 Hz. Within this range the perception of approximately equal changes in frequency is based on the ratio between two frequencies and not on the arithmetic difference between them. In other words, the perceived difference between two sounds of frequencies 100 Hz and 200 Hz is not the same as our perception of the difference between 200 Hz and 300 Hz or between 400 Hz and 500 Hz. In each of these cases we would judge the differences in frequency to be significantly less than. However, we would judge the frequency interval between 200 Hz and 400 Hz to be about the same as that between 100 Hz and 200 Hz, and likewise between 400 Hz and 800 Hz, and so on. For this reason, architectural acoustics has borrowed the concept of *octave bands* from the field of music. As shown in Figure 9.13, an *octave band* or simply an *octave* is defined as a range of frequencies whose upper limit is twice its lower limit.

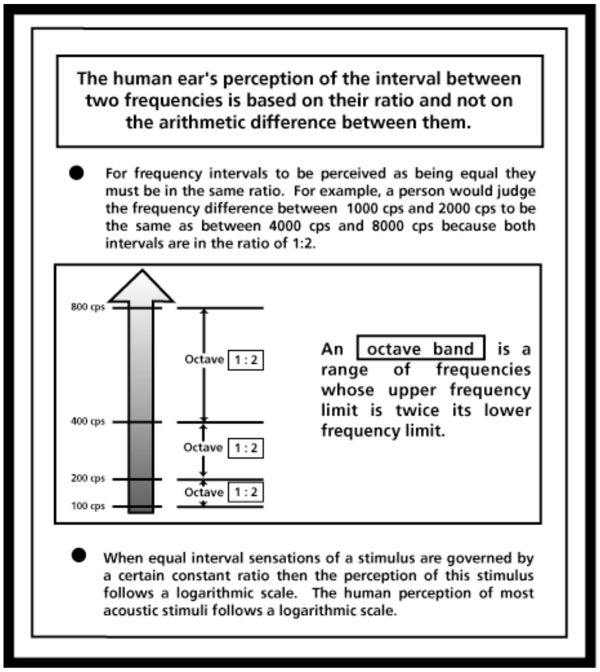


Figure 9.13: The concept of *octave bands*

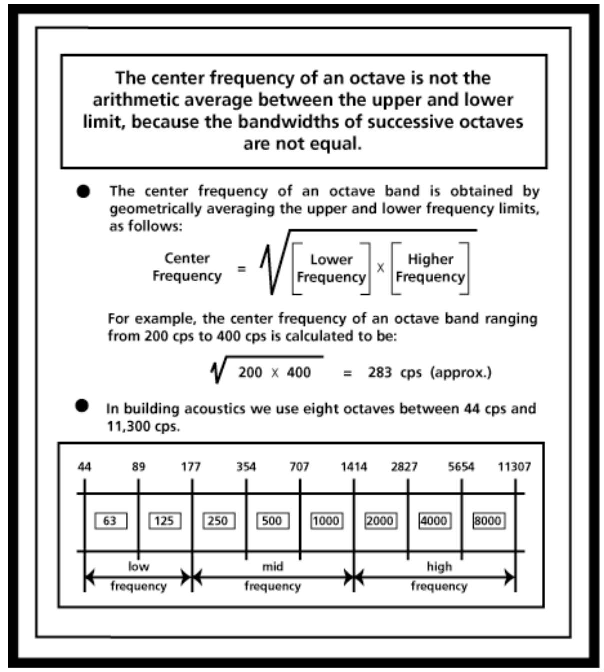


Figure 9.14: Center frequency of an *octave*

Since frequencies below 50 Hz and above 10,000 Hz are rarely of any consequence in buildings, the range of 44 Hz to 11,307 Hz has been divided into eight frequency bands (Figure 9.14), as follows:

low frequencies:	44	to	89	mid-frequency =	[44 x 89]$^{1/2}$	=	62.6 or 63 Hz
	89	to	177	mid-frequency =	[89 x 177]$^{1/2}$	=	125.5 or 125 Hz
middle frequencies:	177	to	354	mid-frequency =	[177 x 354]$^{1/2}$	=	250.3 or 250 Hz
	354	to	707	mid-frequency =	[354 x 707]$^{1/2}$	=	500.3 or 500 Hz
	707	to	1,414	mid-frequency =	[707 x 1414]$^{1/2}$	=	999,8 or 1,000 Hz
high frequencies:	1,414	to	2,827	mid-frequency =	[1414 x 2827]$^{1/2}$	=	1,999.3 or 2,000 Hz
	2,827	to	5,654	mid-frequency =	[2827 x 5654]$^{1/2}$	=	3,998.0 or 4,000 Hz
	5,654	to	11,307	mid-frequency =	[5654 x 11307]$^{1/2}$	=	7,995.6 or 8,000 Hz

At times when a more detailed sound analysis is called for, such as in the determination of building material properties or the exploration of room acoustics, a more finely grained one-third octave band analysis may be performed. However, in the same way that the mid-frequency of an active band is not the arithmetic mean between the lower and upper boundary frequencies, the mid-frequency of a third-octave band cannot be determined by simple arithmetic. If we know the mid-frequency of a particular octave band, such as 63 Hz for the first octave band (i.e., 44 to 89 Hz), then the mid-frequencies of its three one-third octave bands are calculated as follows:

mid-frequency of first third-octave band = $62.6 \div 2^{1/3} = 62.6 \div 1.26 = 49.7$ or 50 Hz
mid-frequency of second third-octave band = 62.6 or 63 Hz
mid-frequency of third third-octave band = $62.6 \times 2^{1/3} = 62.6 \times 1.26 = 78.9$ or 80 Hz

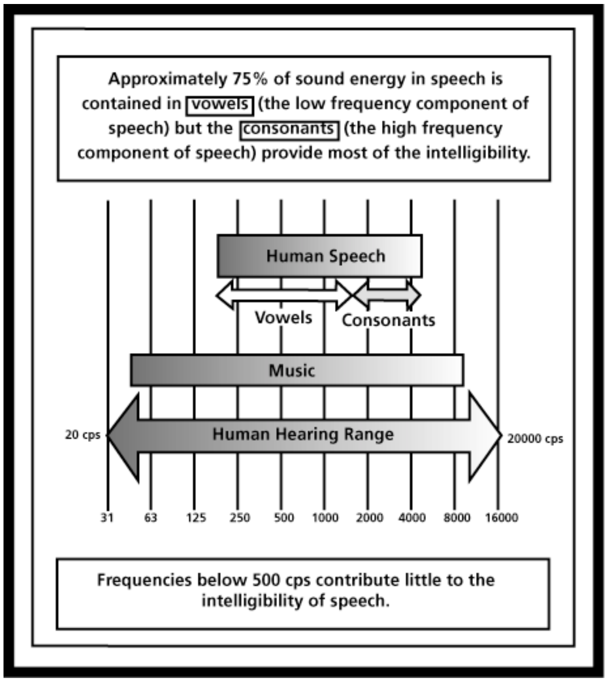


Figure 9.15: Human speech

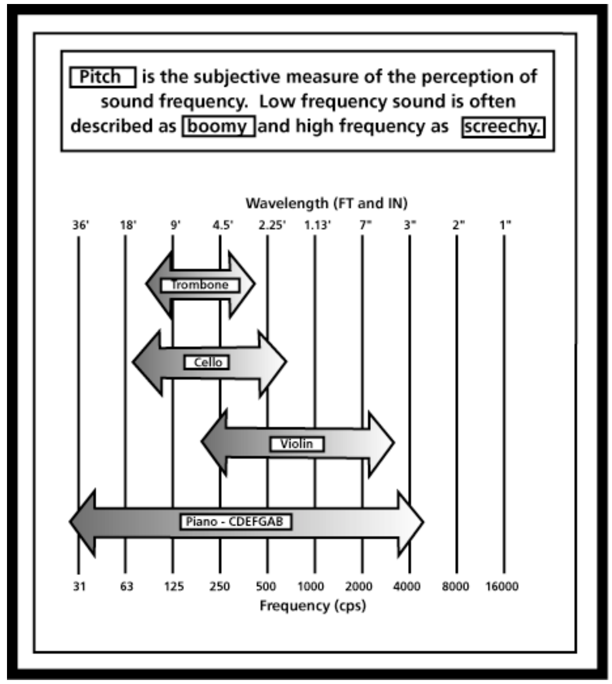


Figure 9.16: Pitch and frequency

It is interesting to note that virtually all of the sound energy in speech is contained within the five octave bands that extend from 177 Hz to 5,654 Hz, with close to three-quarters of that sound energy being provided by vowels (Figure 9.15). However, it is the relatively high frequency consonants that are largely responsible for the intelligibility of speech.

9.5 Subjective Units of Sound Measurement

The perception of sound, like the perception of heat and light, is a subjective mechanism, and it is therefore necessary to relate the objective measurements of vibration (i.e., sound pressure, SPL, sound intensity, and SIL) with the ability of the human ear to distinguish and measure the resulting audible sound.

The situation is complicated by the fact that the human ear not only varies in its response to high and low sound pressure differences, but also with frequency. For example, if notes of 50 Hz and 1,000 Hz frequencies are played alternately at a sound pressure level of 40 dB, then the higher frequency note will appear to be louder. The greatest sensitivity of the ear occurs at approximately 4000 Hz, but this falls off quite sharply for higher and lower frequencies. It is therefore to be expected that the threshold of hearing has different SPL values at different frequencies, because the perceived loudness of a sound depends on both its SPL and frequency. At higher SPLs (e.g., above 100 dB) the sensitivity of the ear is much more evenly distributed, with the result that the threshold of pain tends to be fairly constant at about 130 dB to 140 dB for the audible frequency range. To allow for this subjective characteristic of the hearing mechanism a scale of equal loudness levels has been devised (Figure 9.17). A pure tone of 1,000 Hz frequency has been adopted as a reference standard and a sound pressure of 0.00002 Pa (i.e., 0 dB SPL) chosen as the zero level of a loudness scale of units called *phons*. Since 1,000 Hz serves as the reference frequency, SPLs in *decibels* and subjective loudness levels in *phons* are identical at that frequency.

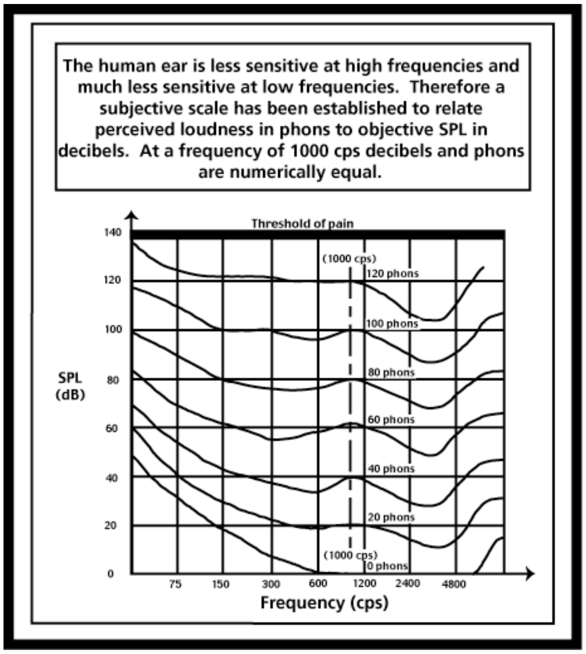


Figure 9.17: Subjective loudness (phon)

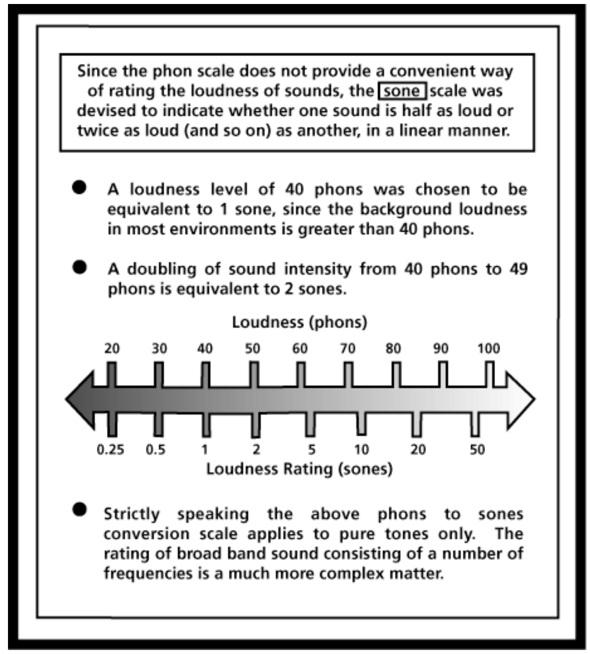


Figure 9.18: Rating of loudness (sone)

It can be seen in Figure 9.17 that for the 40 phon contour a pure tone of 100 Hz at a SPL of 52 dB will appear equally loud as a tone of 4,000 Hz, at 30 dB. However, the phon scale does not provide information in regard to the manner in which persons normally rate one sound with another. At times, it would be convenient to be able to specify one sound as being twice or three times as loud as another. The *sone* scale has been devised to achieve this aim, by rating the loudness of sounds in a linear manner. It was suggested that since a loudness level of about 40 phons normally exists in quiet environments, that this loudness level be equivalent to one sone. Accordingly, a noise which is twice as loud would be 2 sones or 49 phons. On the basis that there exists sufficient correlation between the phon scale and the intuitive judgments of large groups of people, it has been agreed internationally that a doubling of sound will produce approximately a 10 phons increase in loudness (Figure 9.18).[6]

9.6 How Do We Hear Sound?

The structure of the human ear has adapted over millions of years to receive and transmit to the brain those vibrations that constitute the hearing stimulus. Sound pressures set the ear drum (Figure 9.19) in vibration and this movement is transmitted by a system of levers to the inner ear where nerves are stimulated. It is therefore apparent that the ear incorporates an effective transducing mechanism (i.e., the middle ear) for transferring the vibration of air (i.e., the outer ear) to the fluid system surrounding the hair cells of the hearing nerve (i.e., the inner ear). Due to its particular shape, the outer ear matches over a wide range of frequencies the impedance of the ear drum on the external air.

[6] In comparison, a doubling of sound energy produces only a 3 dB increase in SPL, which is the objective equivalent of loudness in phons (see Section 9.3).

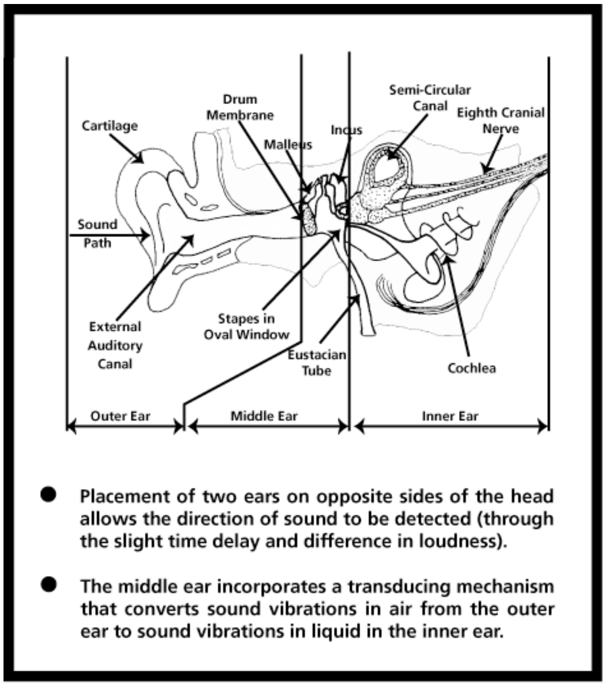


Figure 9.19: The anatomy of the ear and the biological hearing mechanism

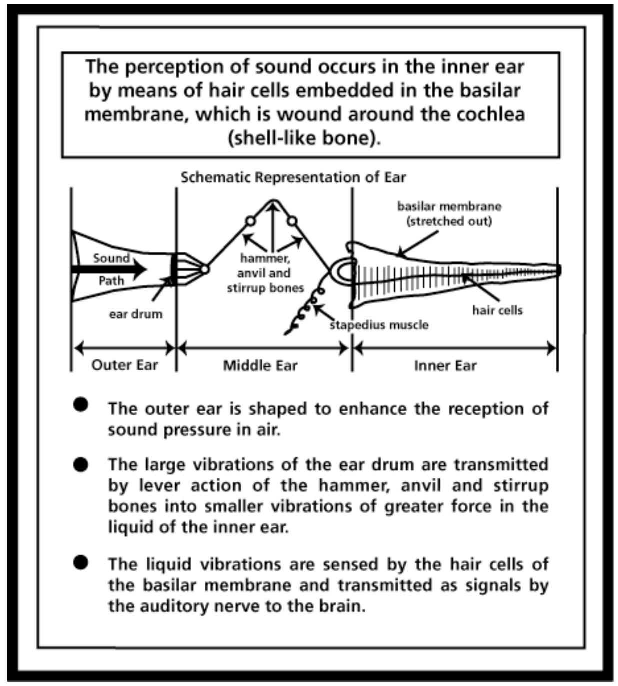


Figure 9.20: Mechanical simulation of the human hearing mechanism

As shown by the mechanical simulation of the hearing mechanism depicted in Figure 9.20, the middle ear changes by multiple lever action the relatively large movements of the ear drum to smaller movements of greater force more suitable for activating the fluid system of the inner ear. This transducing mechanism is necessary because a gas such as air is compressible and a liquid is not. The sound vibrations transmitted through air represent a relatively small force acting over a relatively large distance. This has to be converted in the middle ear to a relatively large force acting over a very small distance.

The perception of the sound occurs by means of hair cells embedded along the basilar membrane, which transversely divides the spiral wound cochlea bone (Figure 9.19). Although it is known that the basilar membrane is able to discriminate between frequencies, this innate capability is thought to be insufficiently accurate to explain the extremely high degree of frequency discrimination achieved by man.[7] We must, therefore, assume that the brain interprets the electrical impulses that flow from the nerve endings of the hair cells.

Hearing sensitivity, particularly at the higher frequencies, diminishes with age even in healthy persons. This aging condition is referred to as presbycusis and can reduce the sensitivity of the human ear to frequencies below 5,000 Hz. However, long-term and repeated exposure to noise levels above 90 dB are known to cause permanent hearing damage regardless of age, a condition referred to as sociocusis.

[7] The German physicist Zwicker (1961) has been able to identify some 24 critical sections of the basilar membrane of the cochlea with individual frequency perception characteristics. His work laid the foundations for various applications in the field of psycho-acoustics. For example, the concept of critical sections or bands in the cochlea have practical applications in the design of hearing aids, sound compression algorithms (like MP3), and audio watermarking (the addition of a distinctive sound pattern to an audio signal that is undetectable to the human ear).

9.7 Hearing Conservation in the Environment

Whereas it has been clearly demonstrated on the basis of statistical data that an adverse thermal building environment will have a detrimental effect on its occupants, no such definitive large-scale studies appear to have been conducted in respect to the impact of noise on the productivity of building occupants. However, there is a great deal of medical evidence that directly links high level noise to hearing loss.

Hearing is easily damaged by exposure to very high SPLs over a short period of time or elevated SPLs over an extended period of time.

Short Term Exposure (120 (+) dB):

Can cause abrupt physical damage to the middle ear. For example, rupture of the tympanic membrane at about 160 dB and permanent damage to the cochlea accompanied by pain, nausea, nystagmus (i.e., shifting of the visual field) and loss of orientation at SPLs above 135 dB.

Long Term Exposure (90 (+) dB):

Can cause gradual physical damage over a period of months or years to the hair cells in the basilar membrane. Persons working in noisy environments (e.g., factories, aircraft servicing areas in airports, and dancehalls) are particularly susceptible to hearing loss.

Figure 9.21: Physical hearing damage

The subjective impact of noise on mental and motor behavior may be categorized as interference with activities and disturbance of attitudes.

Interference with Activities:

Even slightly elevated noise levels may interfere with activities such as speech communication (above 50 dB), working, reading, relaxing, convalescing, and sleeping.

Disturbance of Attitudes:

Annoyance is caused in particular by higher frequency sound and noise that the listener considers to be unnecessary. A noisy environment can certainly accentuate nervous stress and disorders, particularly if associated with a fear of economic and status loss.

Figure 9.22: Subjective considerations

Research has indicated that for relatively short periods of time (i.e., 1 to 2 hours) persons undertaking tasks requiring a high degree of concentration do not appear to be significantly affected by moderately high background noise levels involving indistinguishable or musical sound. Nevertheless, it is certainly true that very high noise levels and in particular certain types of noise will cause annoyance and, after lengthy exposure, permanent hearing damage. Attempts have been made to classify the possible effects of noise on man into main groups. Foremost among these are the effects on anatomical structures and principal physiological processes, such as the vibration of bodily structures, damage to the middle ear (i.e., rupture of the tympanic membrane at a SPL of about 160 dB), temporary or permanent damage to the cochlea, and pain due to SPLs in excess of 135 dB that may also be accompanied by nausea and apparent shifting of the visual field (i.e., nystagmus) or loss of orientation. Less clearly defined are the subjective effects on mental and motor behavior, which may be categorized as follows:

Category A - Physical impact leading to interference with the following kinds of activities:

- Speech communication (usually above 50 dB).
- Working (depending on the kind of work performed).
- Relaxing.
- Sleeping (usually above 35 dB).
- Reading.

- Convalescing.
- Traveling.

Category B - Physiological impact leading to the disturbance of attitudes:

- Annoyance (typically caused by unwanted or unnecessary noise.
- Fear of bodily injury or economic and social loss of status.
- Accentuation of nervous stress and disorders.

In recent years experience has shown that the construction of new or the extension of existing public complexes that produce a great deal of noise (e.g., regional airports and sporting complexes) near residential neighborhoods, can lead to severe public reaction. It has become apparent that blatant disregard of the acoustic aspects of city and regional planning considerations may lead to time consuming and costly political action, if the provocation is directed at a sufficiently wide sector of the population. Such examples have been related in particular to the large-scale expansion of existing airports. The public response tends to escalate in stages with increasing severity, as follows:

Stage 1: Attempts to provide noise barrier or buffer.

Stage 2: Leaving or avoiding the noise field.

Stage 3: Action against the party blamed for the noise, progressing in stages:

- Discussion within localized groups.
- Complaints to local authorities.
- Complaints to State and National authorities.
- Public demonstrations.
- Threats of legal action.
- Vigorous legal action.

Stage 4: Illegal actions such as sabotage.

It is incumbent on architects and environmental planners to take into account the external and internal sound environments throughout the design process in an effort to avoid the likely occurrence of noise interference with the activities of building occupants. Desirable sound environments fall into two main categories; namely, those that satisfy existing laws and those that enable the full enjoyment of work, recreation and sleep. Over the past several decades an increasing number of national, state and local government authorities have adopted zoning ordinances in an attempt to shield the public from noisy environments. These ordinances typically specify performance standards for industrial and vehicular traffic noise sources and require measurements to be made at the boundaries of industrial areas and where highways intrude on residential neighborhoods. While the World Health Organization is concerned with all threats to health and efficiency, the International Standards Organization (ISO) and the International Electrotechnical Commission (IEC) deal respectively with criteria for hearing conservation, annoyance and speech communication, and with instrumentation.

One of the foremost design aims should be to provide noise levels that will promote efficiency and safety while maintaining good relationships among neighboring communities. The fact that sound that is annoying to one person may be highly desirable to another (e.g., music) further complicates the achievement of this elusive aim. Over recent years three parameters have been established for desirable sound environments.

1. ***Hearing conservation:*** Protection from physical damage.
2. ***Satisfactory communication:*** Sufficiently low speech interference level.
3. ***Comfort and enjoyment:*** Uninhibited concentration, relaxation, and sleep.

Early studies of the relationship between deafness and noise were aimed at the prediction of the average hearing loss to be expected after a certain exposure time to a specific noise environment. The resultant range of hearing loss was found to vary widely, depending on the individual susceptibility to hearing damage of the subject. This led to the establishment of Damage Risk Criteria based on critical noise levels likely to cause the onset of hearing damage in average ears. However, since it is not possible to gauge accurately the threshold of noise-induced deafness of an individual until some permanent hearing loss has occurred, this approach has now been largely superseded by the formulation of Hearing Conservation Criteria.

In general Hearing Conservation Criteria allow SPLs that are some 10 dB below the Damage Risk Criteria. Investigations in several countries have shown that about 85 dB in the critical octave frequency bands between 300 Hz and 4,800 Hz should be chosen as an upper limit necessary for the protection of persons whose hearing is susceptible to damage. In environments where higher noise levels are likely to occur it will be necessary to provide the occupants with some form of hearing protection. There is some evidence to suggest that the duration of exposure is also a significant factor. It is generally accepted that a reduction of exposure by 50% would allow the Hearing Conservation Level to be raised by some 3 dB.

The precise measurement of noise-induced deafness is somewhat complicated by the natural loss of sensitivity of the ear that occurs with increasing age. This phenomenon is described in medical terms as presbycusis and symptomized in particular by the loss of sensitivity at higher frequencies. Some years ago, when television first became available in Australia, a number of manufacturers received an inordinate number of complaints from customers who had bought television sets. In each case these complaints were traced back to the alleged presence of a high frequency background sound that allegedly originated from the set and disturbed the viewer. The manufacturers were puzzled that these complaints came almost exclusively from young people, and that their service inspectors consistently reported that they were unable to detect the offending sound. After a certain amount of confusion, it was found that the complaints had been justified in as much as the frequency and SPL of the reported sound placed it in a range affected by presbycusis.

One of the most common detrimental effects of noise is interference with communication. The requirements for unimpeded speech communication are related to the frequency components and dynamic properties of speech, as well as the background noise level. Research has been undertaken to study the contribution to intelligibility of small sections of the speech spectrum and the masking effect of background noise, as a means of predicting the effectiveness of communication in a given environment. These studies have shown that the intelligibility of normal conversation is reduced whenever the sound pressure level of speech is less than 10 dB above the level of ambient, broad-band, background noise. If we assume the ordinary speaking voice to reach a level of about 60 dB at a distance of 3 FT, then it is apparent that the background noise level in any room to be useful for discussions should not exceed 50 dB.

During the 1950s the procedure for predicting the effectiveness of communication in the office environment was considerably simplified by the notion of Speech Interference Levels (SIL).[8] The SIL is the average in decibels (dB) of the SPLs in the three octave bands 600-1,200 Hz, 1,200-2,400 Hz, and 2,400 to 4,800 Hz. On the basis of questionnaire-rating studies the well-known American acoustics pioneer Leo Beranek found that, although SILs are a necessary measure of satisfactory speech communication there are some noises for which the loudness level (LL) must also be considered. He therefore devised a set of Noise Criteria (NC) curves to relate SILs and LLs on the assumption that the noise is continuous (Beranek 1960). He recommended the following maximum background noise levels (i.e., SIL) to allow reliable speech communication at the distances indicated (Table 9.3):

Table 9.3: Maximum background noise levels (i.e., SIL) for reliable speech communication

Distance Between Speaker & Listener	Speaker's Voice Level: Normal	Raised	Loud	Shouting
1 FT	66 dB	72 dB	78 dB	84 dB
2 FT	60 dB	66 dB	72 dB	78 dB
4 FT	54 dB	60 dB	66 dB	72 dB
6 FT	50 dB	56 dB	62 dB	68 dB
12 FT	44 dB	50 dB	56 dB	62 dB
24 FT	38 dB	44 dB	50 dB	56 dB

9.8 Sound Measurement Instruments

Precise measuring instruments are required for the comparison of sound levels taken at different places and different times. The basic instrument of noise measurement is the Sound Level Meter, consisting of a microphone, amplifier with attenuators calibrated in decibels, a meter calibrated in decibels for reading the amplifier output, and weighted networks (Figure 9.23). Performance standards for Sound Level Meters are typically established in individual countries, generally based on recommendations compiled by the International Electrotechnical Commission (IEC 1961).

Most modern Sound Level Meters use transistor circuits with large amounts of feedback and temperature compensation to stabilize performance. The transistor amplifier has the significant advantages of low weight, negligible heat output and small power consumption. High quality Sound Level Meters, referred to as Precision Sound Level Meters, normally incorporate a convenient electrical system for calibrating the amplifier gain and a calibrated control for adjusting this gain to suit microphones of varying sensitivity.

Sound Level Meters also incorporate a number of filter-networks that modify the overall frequency response of the instrument to approximately simulate the response of the human ear. The selection of any network is controlled by a switch that usually provides a choice of three

[8] Unfortunately, the acronym for Speech Interference Level (SIL) is identical to the acronym for Sound Intensity Level (SIL) referred to previously in Section 9.2. However, this should not cause too much confusion since SPL is mostly used in preference to Sound Intensity Level in the field of building acoustics.

filter-networks (i.e., A, B, and C). Precision Sound Level Meters have a linear network (L_m) without filters, in addition to the three filter-networks. This fourth network provides a linear (flat) frequency response for use in conjunction with analyzers and recorders.

The degree of rating and tolerance at each frequency of these weighting networks are specified by national standards according to recommendations of the International Electrotechnical Committee. With C-weighting the response is practically unaltered in the region of 60 Hz to 4,000 Hz. However, at 31.5 Hz the response is -3 dB, which represents a halving of the input. B-weighting is virtually unmodified between 300 Hz and 5,000 Hz, while the A-network is unmodified between 1,000 Hz and 2,000 Hz only (Figure 9.25). For medium noise levels, the ear responds approximately in accordance with the B-network, while for low levels the A-network should be used, and for traffic noise the C-network is the closest simulation. It follows that if the appropriate weighting network is used, the reading will be approximately in *phons* (i.e., a measure of loudness). On the recommendation of the International Electrotechnical Committee most measurements are now taken with an A-weighting and therefore designated as *dBA* irrespective of the noise level.

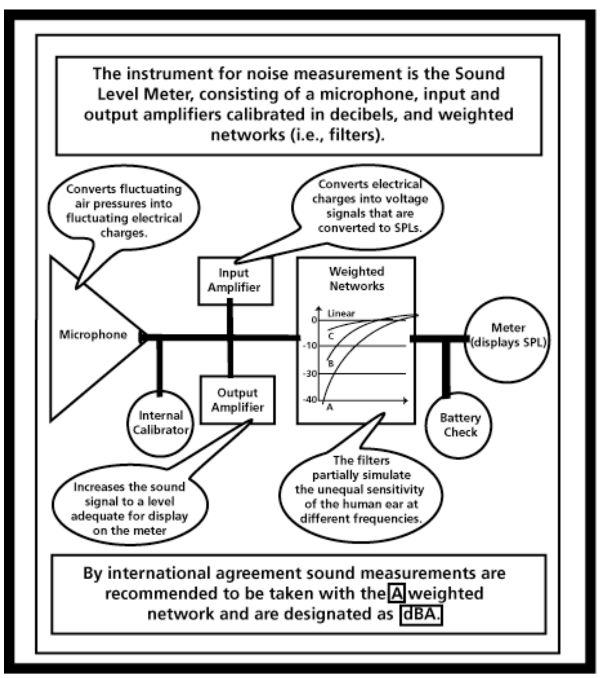


Figure 9.23: Sound Level Meter Components

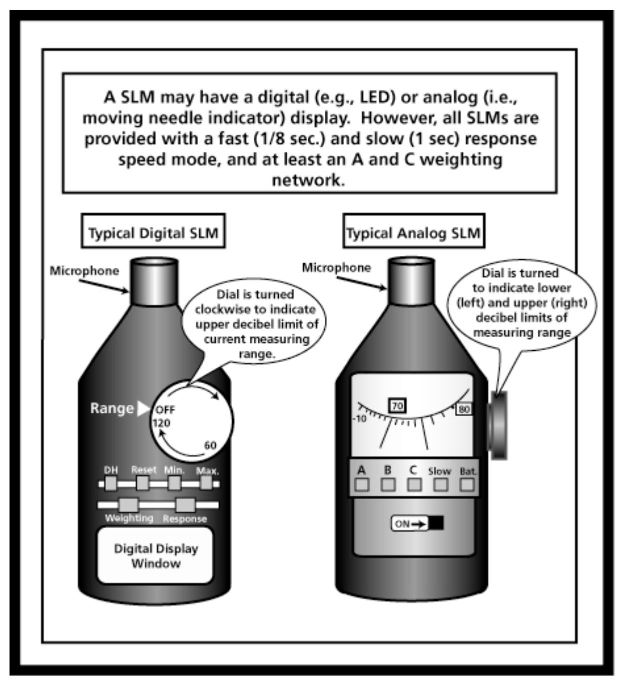


Figure 9.24: Typical Sound Level Meters

Sound Level Meters incorporate a calibrated attenuator in the form of a switch calibrated in 10 dB steps and synchronized to provide a reading of the sound field in conjunction with a display meter. A typical implementation of this switch mechanism is shown in Figure 9.24. It is important to check the validity of the attenuator switch at regular intervals and measure the signal to noise ratio in all positions. The display meter is normally calibrated from -6 to +10 dB. Simple addition of the display meter reading to the attenuator switch calibration provides the SPL measurement.

Naturally, the microphone is the vital component of a Sound Level Meter since it converts the ambient sound energy into electrical energy. The condenser, crystal and dynamic microphones are most commonly used for sound measurement. Each suffers from some defect in performance, and it is therefore necessary to investigate a number of critical characteristics before a selection can be made for any particular noise measurement situation. The physical size of a microphone is directly related to its frequency response. Especially when the microphone faces a high frequency sound source, pressure build-up across its face can give rise to inordinately high readings. Some Sound Level Meters are designed to be used in a manner that will allow the sound field to pass across the diaphragm of the microphone, so that a flat response is obtained. The same flat response may be achieved for sound coming from any direction, by using a small microphone. Other factors that lead to poor frequency response include resonance of air cavities, or critical components such as diaphragms and microphone shapes that in themselves disturb the sound field.

The difference between the highest level at which a microphone will operate without distortion and the electrical noise that is generated within its various circuits is defined as the dynamic range. Since microphones are easily damaged by inadvertent exposure to high level sound fields, it is also important to know the upper safe limit of exposure.

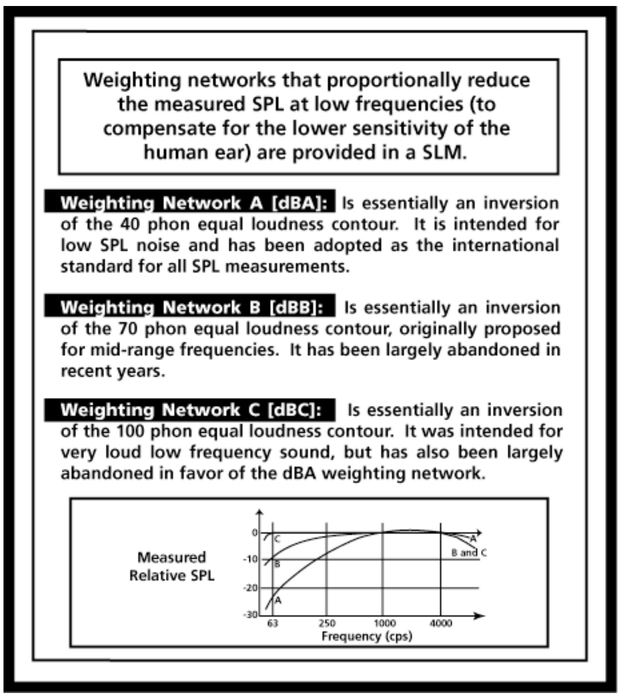
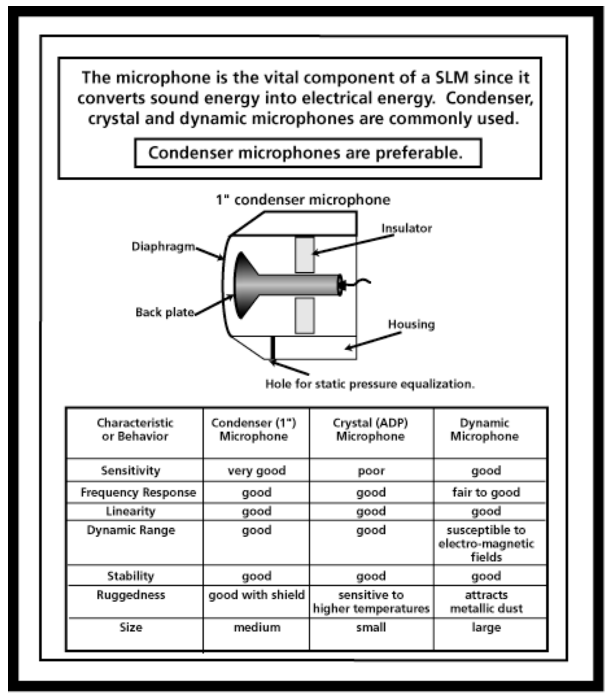


Figure 9.25: The weighting networks



Characteristic or Behavior	Condenser (1") Microphone	Crystal (ADP) Microphone	Dynamic Microphone
Sensitivity	very good	poor	good
Frequency Response	good	good	fair to good
Linearity	good	good	good
Dynamic Range	good	good	susceptible to electro-magnetic fields
Stability	good	good	good
Ruggedness	good with shield	sensitive to higher temperatures	attracts metallic dust
Size	medium	small	large

Figure 9.26: The condenser microphone

Crystal microphones were commonly used in conjunction with Sound Level Meters before the development of high-quality condenser microphones. A common type of crystal material is ammonium dihydrogen phosphate (ADP), which has a low specific output but is operative for temperatures up to 257˚F. These two properties do not conflict, because a material of low sensitivity is useful only in high energy sound fields with their attendant heat. Some less expensive Sound Level Meters have hearing-aid type Rochelle-salt microphones with rather poor humidity and temperature characteristics (upper limit of about 113˚F, which can be easily reached during field noise measurements). A further type of crystal microphone commonly in use with Sound Level Meters is the PZT microphone. The letters PZT refer to lead zirconate titanate ceramic, which has good high temperature characteristics (up to 212˚F).

Dynamic microphones sometimes referred to as moving coil microphones, operate on the principle that a coil of wire mounted behind a diaphragm induces a voltage as it moves in the annular space of a magnetic system. By virtue of its design the dynamic microphone will normally assume larger dimensions than its condenser or crystal counterparts, and it is therefore likely that its frequency response will be affected by pressure buildup at frequencies normally encountered in sound measurements. For this reason, high-quality dynamic microphones employ controlled damping behind the diaphragm to level the response characteristics. Although inherently low electrical impedance makes these microphones suitable for use on long extension cables, they have two shortcomings. They are susceptible to induced stray electrical noise and the attraction of metallic dust particles.

Condenser microphones (Figure 9.26) consist basically of a thin metallic, pressure sensitive diaphragm stretched in front of an insulating metal disc. They are high quality, stable microphones now normally prescribed for Sound Level Meters. Since the choice of a particular Sound Level Meter follows directly from the selection of the most suitable microphone system, a comparison of the performance characteristics of condenser, crystal and dynamic microphones is provided in the bottom half of Figure 9.26 in respect to the following characteristics: sensitivity; frequency response; linearity; dynamic range; stability, ruggedness; and, physical size.

Sound Analysis with Sound Analyzers: Experience with Sound Level Meters has shown that overall decibel readings do not provide a true indication of how the measured sound will be perceived by human ears even when weighting networks are used. For example, a Sound Level Meter adds noise levels according to the objective rule that a doubling of the source adds 3 dB to the ambient noise level, while the ear may add complex sound levels of equal intensity (subjectively) and recognize an increase of 6 phons or more. It is therefore common practice to analyze sound measurements into their component frequencies before accurate predictions of subjective response are made. The most commonly used type of Sound Analyzer divides the sound spectrum into eight bands of center frequencies (i.e., 63, 126, 250, 500, 1000, 2000, 4000, and 8000 Hz)

Most Sound Level Meters have provisions for attaching a Sound Analyzer directly to the body of the Sound Level Meter. In fact, often the Sound Analyzer is designed to draw its electrical power from the power source (i.e., batteries) of the Sound Level Meter. .

Calibration, Testing and Measurement: Calibration and testing are an essential aspect of dependable sound measurement. Standards based on IEC recommendations as a rule require periodic testing of a considerable number of characteristics, including overtaxing, influence of temperature, relative humidity, as well as electric and magnetic fields, which are determined either by the assembly as a whole or its individual components. Most of these characteristics will be similar for instruments of the same type, and it is therefore convenient to divide calibration tests for Sound Level Meters into two categories, namely, type-tests and individual-tests. These were defined by the German DIN 45633 Standard (DIN 1966) in 1966, as follows:

Type Tests:
1. Influence of static pressure.
2. Testing of dynamic characteristics in all measurement ranges and for all weighting-networks.
3. Influence of mechanical vibration.
4. Influence of magnetic and electrostatic fields.

5. Temperature range.
6. Influence of relative humidity.
7. Over-taxing.
8. Effect of externally connected circuits or instruments.
9. Directional properties.
10. Difference between the weighting-network ratings for diffuse sound fields and plane, propagating waves (for a number of frequencies).
11. Influence of large sound fields.

Individual Tests:

1. The ratings of the weighting-networks in open sound fields.
2. Dynamic characteristics of the indicator mechanism.
3. Effective-value registration.
4. Graduation of the scale of the indicator mechanism.
5. Accuracy of the overall-measuring-range switch.
6. Absolute value of the reference frequency.
7. Inherent noise level of the instrument.

In the case of Precision Sound Level Meters, which are used for very accurate acoustical analyses and legal disputes, it is normal practice to calibrate the Precision Sound Level Meter before every use and at frequent intervals during anyone period of use. The *+3 dB* level on the display meter is recommended as a reference calibration level, since it represents the average of the range most often used during measuring. When coupled to a tape recorder, the meter reading and attenuator settings should be stated so that the analyzing equipment can be adjusted to the same setting during replay. If these calibration and testing instructions are closely adhered to, then the quality of the instrument alone will determine the error incurred in measurements of the ambient noise level.

Sound Level Meters are commonly used to measure widely varying sound fields under changing conditions and for different purposes. The noise levels under investigation will differ in frequency spectrum and may originate from more than one source. Measurements may be taken in the open or between reflecting walls, in a reverberation room or anechoic chamber.[9] The purpose of the measurements may be, for example, to: estimate the danger of hearing-damage; assess the effectiveness of acoustic remedies; compare building materials or systems of construction; or, control legally prescribed maximum noise levels. In every case the method of measurement must suit the problem, since the mode of application could have considerable influence on the final interpretation of the test results.

[9] Reverberation rooms and anechoic chambers are special acoustic laboratories designed to produce maximum reflection or absorption, respectively. Accordingly, in a reverberation room all of the surfaces are acoustically hard surfaces to maximize the interreflection of sound within the room, while in an anechoic chamber the walls and ceiling are typically sprayed with a highly sound absorbing finish such as a porous plastic foam. A similar type of treatment is applied to the floor of an anechoic chamber, with the addition of a wire mesh screen on top to support light pedestrian traffic.

During the actual reading of the instrument scale the observer must be positioned in such a way that he or she does not noticeably disturb the sound field in the vicinity of the microphone. The avoidance of interference due to reflections from the observer is of particular importance for the measurement of pure tones and narrow-band noise. This requirement will generally be satisfied if the observer is at least 4 FT distant from the microphone at one side of a tripod-mounted Sound Level Meter. The German standard DIN 45633 recommends an optimum distance of 3 meters (i.e., 10 FT) between the microphone and the observer. This would normally require the microphone to be connected to the Sound Level Meter by means of an extension cord.

In the particular case of sound insulation measurements, if a constant noise source of 90 dBA exists on one side of a partition and the Sound Level Meter reading on the A-network is 50 dBA on the other side, then the insulation provided by the partition is calculated to be 40 dBA (i.e., 90 - 50 = 40 dBA). This is naturally on the assumption that the background noise level on the Sound Level Meter side of the partition is less than about 40 dBA (which can be neglected because 40 dBA + 50 dBA = 50 dBA). Measurements on both sides of the partition should be taken for at least three distances from the partition and the average calculated. This procedure will tend to avoid the possibility that the microphone may have been influenced by reflections, standing waves, or interference patterns.

9.9 Questions Relating to Chapter 9

Answers to the following multiple-choice questions with references to the appropriate text (by page number) may be found at the back of the book.

1. **If sound is the result of vibrations caused by a source that emits pressure fluctuations on a spherical front, which of the following statements (if any) is *not* correct:**
   A. The form of the vibrations is cyclic.
   B. Sound vibrations are composed of compressions and rarefactions.
   C. Sound vibrations are in the form of transverse waves.
   D. Sound waves cannot travel through a vacuum.
   E. All of the above statements (i.e., A, B, C, and D) are correct

2. **Which of the following statements best describes the propagation of sound waves:**
   A. Sound waves are propagated by the mass movement of any medium.
   B. The essential parameters of the propagation of sound waves are wavelength, velocity, and frequency.
   C. The velocity of propagation of sound waves in air (at 68˚F) is 1,126 FT/sec.
   D. Sound waves are propagated by the progressive, elastic vibration of particles about their mean position.
   E. None of the above statements (i.e., A, B, C, and D) are correct.

3. **If the velocity of sound in air (at 68°F (≈ 20˚C)) is 1,126 FT/sec, then the velocities of sound in mild steel and water will <u>most likely</u> be:**

A. 16,400 FT/sec (mild steel) and 4,600 FT/sec (water).
B. 4,600 FT/sec (mild steel) and 16,400 FT/sec (water).
C. 0 FT/sec, since sound cannot travel through either.
D. 15,000 FT/sec (mild steel) and 10,000 FT/sec (water).
E None of the above statements (i.e., A, B, C, and D) are correct.

4. **The frequency range to which the normal human ear responds is approximately:**
   A. 5 Hz to 1,000 Hz.
   B. 15 Hz to 10,000 Hz.
   C. 100 Hz to 30,000 Hz.
   D. 20 Hz to 20,000 Hz.
   E. None of the above statements (i.e., A, B, C, and D) are correct.

5. **Sound vibrations may be classified into three general groups as follows:**
   A. Periodic, aperiodic and broad band.
   B. Cyclic, periodic and aperiodic.
   C. Cyclic, impulsive and white noise.
   D. Cyclic, impulsive and broad band.
   E. None of the above statements (i.e., A, B, C, and D) are correct.

6. **In what respect is the perception of sound similar to the perception of heat and light:**
   A. The normal human being is sensitive to heat, light and sound.
   B. The human being requires time to perceive heat, light and sound.
   C. The perception of sound is a subjective mechanism.
   D. The perception of sound is both subjective and objective.
   E. The perception of sound is a very complicated objective mechanism.

7. **The following are objective measures of sound:**
   A. phon, sound pressure, decibel
   B. phon, sone, sound pressure level
   C. sound pressure, sound pressure level
   D. sound pressure, sone, decibel
   E. None of the above statements (i.e., A, B, C, and D) are correct.

8. **The decibel (db) was chosen as the unit of sound pressure level because it:**
   A. Was the only one that could be agreed upon internationally.
   B. Is a reasonable guide to the measurement of subjective response.
   C. Expresses a wide range of sound pressures on an objective logarithmic scale.

D. Expresses a small range of sound pressures on a wide range of logarithmic measurements.
E. Is a reasonable guide to the objective measurement of a physical phenomenon.

9. **The reference pressure (P_0 Pascal) that is used in the calculation of the sound pressure level of any sound is equal to:**

A. 0.0003 Pascal
B. 2.0×10^{-4} Pascal
C. 0.0002 Pascal
D. 0.00002 Pascal
E. None of the above statements (i.e., A, B, C, and D) are correct.

10. **The normal human ear varies in its response to:**

A. Sound pressure level and frequency.
B. Frequency only.
C. Sound pressure only.
D. Sound pressure level only.
E. Loudness, wavelength and velocity.

11. **We are able to identify the <u>direction</u> of a sound source by virtue of:**

A. Visual perception.
B. Pressure differences identified by hair cells embedded in the basilar membrane.
C. The placement of two ears on opposite sides of the head.
D. Secondary reflections which occur in the outer ear.
E. None of the above statements (i.e., A, B, C, and D) are correct.

12. **Which (if any) of the following statements is <u>not</u> correct.**

A. The perception of sound occurs by means of hair cells embedded along the basilar membrane, which transversely divides the spiral wound cochlea in the middle ear.
B. The basilar membrane is able to discriminate between frequencies.
C. The ear incorporates an effective transducing mechanism for transferring the vibration of air to the fluid system surrounding the hearing nerve.
D. The middle ear changes by multiple lever action the relatively large movements of the ear drum to smaller movements of greater force more suitable for activating the fluid system of the inner ear.

E. All of the above statements (i.e., A, B, C, and D) are correct.

13. **Very *high noise levels* and in particular certain *types of noise* will cause after lengthy exposure:**

    A. Permanent damage to the cochlea in the inner ear.
    B. Annoyance and temporary physical damage to the hearing mechanism.
    C. Annoyance but no physical damage to the hearing mechanism.
    D. A temporary change of state of the fluid in the inner ear.
    E. A feeling of pleasure followed by slight pain.

14. **Which (if any) of the following statements is <u>not</u> correct:**

    A. One of the foremost design aims of an architect should be to provide noise levels that will promote efficiency and safety while maintaining good relationships among neighboring communities.
    B. Investigations in many countries have shown that about 95 db in the critical frequency bands between 300 Hz and 4,000 Hz should be chosen as an upper limit necessary for the protection of persons whose hearing is susceptible to damage.
    C. Over recent years three parameters have been established for desirable sound environments:
        1. Hearing conservation.
        2. Satisfactory communication.
        3. Comfort and enjoyment.
    D. There is some evidence to suggest that the duration of exposure to high noise levels is also a significant factor in determining the likelihood of hearing damage. Accordingly, a reduction of exposure by 50% would allow the Hearing Conservation Level to be raised by some 3 db.
    E. All of the above statements (i.e., A, B, C, and D) are correct.

15. **One of the most common detrimental effects of noise is interference with communication. The requirements for unimpeded speech communication are most closely related to:**

    A. The frequency spectrum and sound level of the speech and the background noise level.
    B. The sound pressure level of the speech and the frequency components of the background noise level.
    C. The relative sound pressure levels of the speech and the background noise.
    D. The frequency spectrum, pitch and sound pressure level of the speech.
    E. None of the above statements (i.e., A, B, C, and D) are correct.

16. **The intelligibility of normal speech communication is substantially reduced whenever the sound pressure level of the speech is less than <u>*X* dB</u> above the level of the ambient background noise.**

    A. X = 3 dB

B. X = 40 dB
C. X = 15 dB
D. X = 10 dB
E. X = 20 dB

17. **The <u>correct definition</u> of Speech Interference Level (SIL) is given by:**
   A. SIL is the average in phons of the sound pressure levels in the octave bands between 600 Hz and 4,800 Hz.
   B. SIL is the average in decibels of the sound pressure levels in the three octave bands: 300 Hz to 1,200 Hz; 1,200 Hz to 2,400 Hz; and 2,400 Hz to 4,800 Hz.
   C. SIL is the sum of sones of the sound pressure levels in the octave bands between 300 Hz and 9,600 Hz.
   D. SIL is the average in decibels of the sound pressure levels in the three octave bands: 600 Hz to 1,200 Hz; 1,200 Hz to 2,400 Hz; and 2,400 Hz to 4,800 Hz.
   E. None of the above statements (i.e., A, B, C, and D) are correct.

18. **Noise Criteria Curves (NC) derived by Beranek are a good measure of satisfactory speech communication, because they <u>relate:</u>**
   A. Speech Interference Levels with Loudness Levels.
   B. Sound Pressure Levels with Loudness Levels.
   C. Sones with phons.
   D. Speech Interference Levels with frequency.
   E. Speech Interference Levels with the Sound Pressure Levels of the ambient background noise.

19. **The <u>addition</u> of two individual sound pressure levels of 55 dB each will yield a resultant sound pressure level of:**
   A. 58 dB
   B. 55 dB
   C. 66 dB
   D. 68 dB
   E. 78 dB

20. **The <u>addition</u> of two individual sound pressure levels of 40 dB and 48 dB will yield a resultant sound pressure level of approximately:**
   A. 53.6 dB
   B. 58 dB
   C. 40.3 dB
   D. 48 dB
   E. 62 dB