Principles of Hearing Aid Audiology, 3rd Edition

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By

Maryanne Tate Maltby

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FOREWORD

Audiology continues to see major changes in technology, in hearing provision and in approaches to rehabilitation. It is therefore both timely and welcome that Maryanne Maltby's 3rd edition of the Principles of Hearing Aid Audiology is published.

The book introduces areas that were not in the last edition, for example: hyperacusis, synaptopathy, dementia, high frequency audiometry, laser scanning, 3D technology and music, all of which incorporate current modern thinking about audiology. The British Society of Audiology standards and HCPC codes of conduct have been updated.

Maryanne has a wealth of experience and knowledge, which is clearly evident in every line of text in this book. It is easy to read, clear, relevant and precise.

It is a great addition to any recommended reading list for students but also should be a useful, readable text to have in every audiology clinic.

Wendy Stevens
Audiologist of the year
Senior Lecturer in Audiology
De Montfort University, Leicester

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I also wish to express my gratitude to the British Library, UCL Ear Institute, Action on Hearing Loss library, Minerva Hearing, Starkey Laboratories, Sivantos, Widex and Otometrics.

If I have forgotten anyone, I apologise profusely.

PART 1:

BASIC SCIENCE

CHAPTER ONE

ACOUSTICS

1.1 Physical Properties of Sound

1.1.1 Sound Generation

Sound requires a source, a medium through which to travel and a detector. The detector is usually a listener but could be a sound measuring device (for example, a sound level meter).

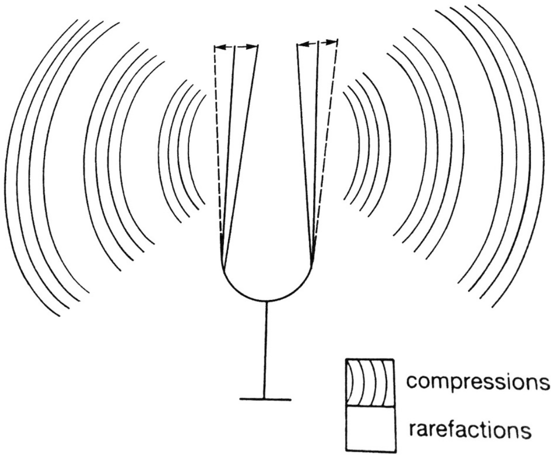

Figure 1-1. The tines of the tuning fork move alternately towards and apart from each other causing alternate regions of compression and rarefaction that move outwards through the air.

Sound is generated by a vibrating object and is transmitted through an elastic medium or substance: gas, liquid or solid. In air, the sound source sets the air particles into vibration, in the same back and forth motion as that of the vibrating sound source. The medium itself is not transferred to the detector; each particle is displaced only a very small distance from its resting position (equilibrium). The energy is passed across the medium as a series of compressions and rarefactions (Figure 1-1). This constitutes a sound wave. In compressions, the air particles move closer together and the air pressure is slightly higher than normal; in rarefactions, the particles move away from each other and the air pressure is slightly lower than normal.

The speed of sound varies with the density of the medium through which it travels. The denser the medium, the faster sound travels. For example, the speed of sound in air is approximately 340 metres per second but in water, which is a denser medium, sound travels at approximately 1450 metres per second. Sound becomes weaker with increasing distance from the source.

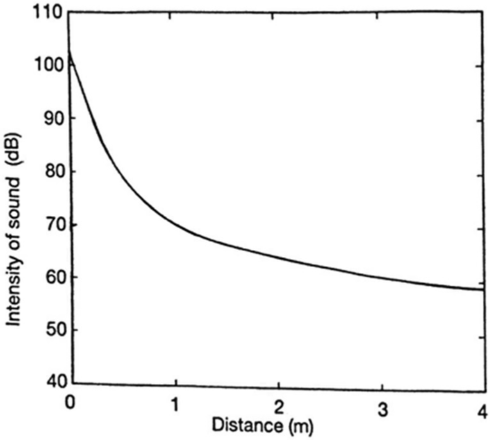

Figure 1-2. As distance away from the sound source increases, the sound level falls in accordance with the inverse square law**.**

If there are no obstacles to affect the progress of the sound waves, the decrease in intensity will be in accordance with the *inverse square law.* This states that the intensity varies inversely with the square of the distance from the sound source (Figure 1-2). In other words, doubling the distance decreases the intensity by a factor of 4 (2^2). This equates to a decrease of 6 dB for every doubling of distance.

1.1.2 Properties of Sound

The simplest waveform is a sine wave or sinusoid. A sine wave is produced by simple harmonic motion, where each vibration is repeated back and forth. This motion repeats itself exactly in equal periods of time and is known as periodic motion. Sinusoidal sound waves are very clean or pure sounds and are therefore termed simple or pure tones. A pure tone can be described by three characteristics: frequency, intensity and duration.

(a) Frequency

The frequency of a sound is denoted by the number of cycles of vibration that occur in one second. A cycle consists of one compression and one rarefaction of air particles.

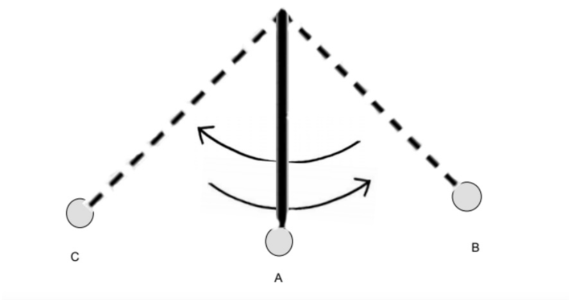

Figure 1-3. A complete cycle of the pendulum, shown by movement from A to B, from B to C and C back to A.

It can perhaps be better visualised as the movement of a pendulum. In a complete cycle, the pendulum would move from its resting position A to a position to one side B, then back through the resting position A to the other side C, and back to the resting position (Figure 1-3).

If it takes one second to complete one full cycle, the frequency is 1 cycle per second (cps) or 1 hertz (Hz); 10 Hz., therefore, means 10 cps. Cycles per second are termed hertz after Heinrich Hertz, a nineteenth century German physicist who was the first to demonstrate the existence of electromagnetic waves. The more cycles that occur in one second, the higher is the frequency of the sound. An electric motor, for example, can readily be heard to produce a higher frequency sound as its speed is increased. Pitch is the subjective attribute of frequency and is closely related to frequency. The higher the frequency, the higher is the pitch. However, as pitch is subjective, it cannot be measured directly.

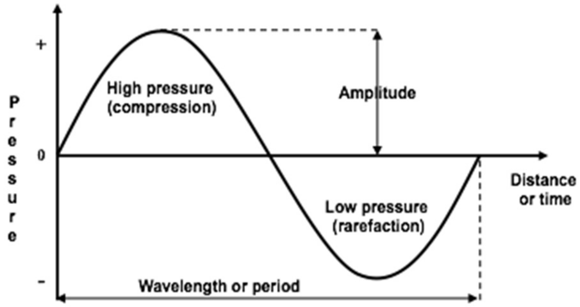

Figure 1-4. One cycle of a wave.

The piano produces its lowest note at 27.5 Hz and its highest at 4186 Hz; middle C is 261.63 Hz (Somerfield, 1987). The human ear can detect a much wider frequency range than this and the young healthy ear can perceive a frequency range from approximately 20 Hz to 20,000 Hz (20 kHz). Sounds below this frequency range are called infrasonic and those above this range are called ultrasonic.

The wavelength is defined as the distance covered by one complete cycle. Wavelength is inversely proportional to frequency and can be determined using the formula:

$$\text{wavelength } (\lambda) = \frac{\text{speed (v)}}{\text{frequency (f)}}$$

As frequency increases, wavelength decreases.

The speed of sound in air is approximately 340 metres per second (m/s). So, for example, if the frequency of a sound wave is 850 Hz, the wavelength in air will be:

$$\frac{340 \text{ m/s}}{850 \text{ Hz}}$$

$$= 0.4 \text{ m}$$

(b)Period

The time required for *one* cycle is known as the period (Figure 1-4). The relationship between frequency and period can be expressed as:

$$\text{Frequency} = \frac{1}{\text{period}}$$

(c) Intensity

Intensity is defined as the amount of energy transmitted per second per unit area. In the SI system, energy per second is measured in joules per second (J/s) or watts (W) and area is measured in square metres (m^2), therefore the unit of intensity is the watt per square metre (W/m^2).

Frequently, we measure the sound pressure level at a given point and not the intensity, although it can be shown that, in free-field conditions, the intensity is proportional to the square of the pressure:

$$\text{intensity} \propto (\text{pressure})^2$$

This is a little beyond the scope of this book so, to avoid confusion, we will consider only the sound pressure level. Sound pressure level is the amount of pressure variation about the mean (shown by "amplitude" in Figure 1-4). The greater the force or energy applied to the vibrating body, the more intense the vibrations. The intensity is governed by the distance the air particles move from their place of rest. With increased force, they will move further and thus cause increased compressions and rarefactions. In the SI system, pressure is expressed in pascals (Pa).

A vibrating source sets up small, localised changes in the air pressure. These changes can be so small they are measured in terms of micro pascals (μPa), which are millionths of a pascal. The human ear can accommodate pressure changes from approximately 20 μPa to approximately 20 Pa. In other words, the greatest pressure change the human ear can withstand is a million times greater than that which is just audible.

The range of hearing is so great that the numbers involved are very large and it is more convenient to describe intensity using a logarithmic scale - the decibel scale. Decibels (dB) are units of relative intensity. The number of dB describes how much greater is the intensity of a measured sound than a fixed reference level. The dB SPL scale (decibels sound pressure level), for example, uses 0.00002 Pa as its fixed reference level, so that 0 dB SPL= 0.00002 Pa. Some everyday sound levels are given in Table 1-1 with the approximate dB SPL equivalent. Decibels are considered in more detail in section 1.2.

Sound pressure level (dBSPL)	Sound pressure (Pa)	Equivalent to
120	20.0	Discomfort
100	2.0	Pneumatic drill
80	0.2	Shouting
60	0.02	Quiet conversation
40	0.002	Loud whispering
20	0.0002	Rustling leaves
0	0.00002	(Standard audiological reference)

Table 1-1. The relationship between dBSPL and pascals.

1.1.3 Complex Sounds

Pure tones can be produced artificially using a tuning fork or an electronic sine-wave generator, but they rarely occur naturally. Most sounds consist of a number of different tones, although these can be broken down into the component pure tones using frequency analysis. Each component pure tone may vary in amplitude, frequency and phase.

1.1.4 Phase

When we consider the phase of a wave, we are simply referring to the position at any point in its cycle. Often it is of more relevance to consider phase differences between waves. Phase and phase difference are expressed in degrees (°), analogous to the mathematical relationship between circles and sine waves.

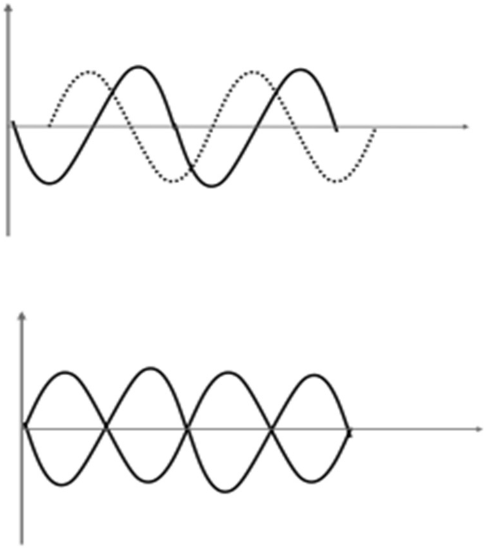

Figure 1-5. Tones that are out of phase.

Any circle corresponds to 360° rotation. The degrees of rotation are used to denote the position in the cycle, where 0° is the starting point, 90° is a quarter of the way around the circle, 180° is half way around and so on. Thus, a quarter of a cycle of a (sine) wave is denoted by 90°, half a cycle by 180° and so forth. Tones are exactly out of phase if they differ from the standard starting point by 180°. Figure 1-5 illustrates tones that are out of phase.

1.1.5 Periodic Sounds

Complex sounds may have a waveform that repeats itself over time. Such sounds are called periodic. Periodic sounds are musical or harmonic.

Musical notes are an example of complex periodic sounds. In such sounds, all frequencies present are harmonics. These are whole number multiples (integers) of the fundamental frequency. The lowest frequency of the tones presented determines the fundamental frequency.

A 1 kHz tone has harmonics at 2 kHz, 3 kHz, 4 kHz and so on, although not all the harmonics are necessarily present. Two instruments playing the same note sound different because their sound contains different harmonics, although the fundamental frequency remains the same. The pitch of a note is recognised by its fundamental frequency. Thus middle C, for example, can be recognised regardless of whether it is played on a piano or a violin.

Harmonics are denoted by number. The fundamental frequency (f) is the first harmonic (1 x f). The second harmonic is twice the fundamental (2 x f); the third harmonic is three times the fundamental (3 x f). The fundamental frequency does not necessarily have to be heard for recognition. Our minds can determine the fundamental from the pattern of harmonics.

1.1.6 Aperiodic Sounds.

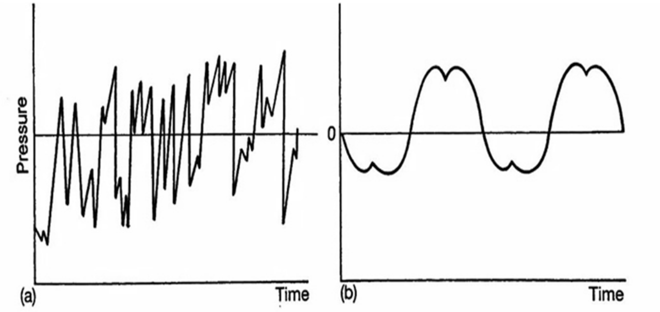

Figure 1-6. Complex waveforms: (a) noise and (b) periodic sound.

Aperiodic or non-periodic sounds do not have regularly repeating waveforms. They consist of more than one frequency and are not harmonically related. This random and unstable waveform has no stated period or repetition and is perceived as noise (Figure 1-6). Certain particular types of noise may be used in audiometry, for example:

- *White noise* consists of all the frequencies of the audible spectrum present at equal intensities.
- *Pink noise* consists of equal energy per octave (an octave interval is given by the doubling of frequency). This is perceived as a lower frequency noise than white noise since each octave contains equal energy but the octave intervals are wider at higher frequencies resulting in less energy at any single frequency. For example, the octave 250 Hz to 500 Hz contains 250 discrete frequencies, whereas the octave 2 kHz to 4 kHz contains 2000 discrete frequencies; yet each of these contains the same amount of energy.

1.1.7 Wave Motion

A wave is basically a disturbance that is repeated as it spreads out from the source. Waves are sometimes categorised according to how the disturbance or vibration travels.

Transverse waves (electromagnetic waves, e.g. light) are those where the vibrations are perpendicular to the direction of energy transfer. It may be helpful to think of a pebble being dropped in a pond (see Figure 1-7a).

Longitudinal waves (mechanical waves, e.g. sound) are those where the vibrations are parallel to the direction of energy transfer. It may be helpful to think of a plucking a stretched spring (see Figure 1-7b).

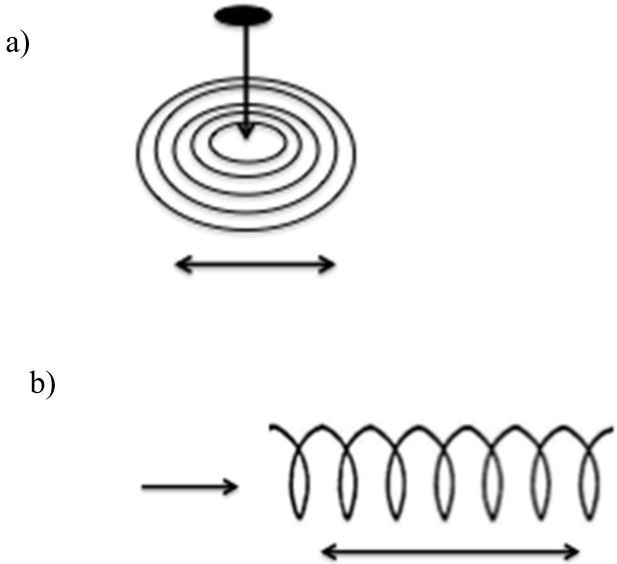

Figure 1-7. Diagrams to illustrate (a) transverse vibrations, and (b) longitudinal vibrations**.**

In an environment where there is no interference from reflections (a free sound field or "free-field"), sound waves will spread in all directions and gradually lose energy as they move away from the sound source.

The intensity of the sound decreases proportionally to the square of the distance from the sound source. This is the inverse square law (Figure 1-2) and in practice it means that a doubling of distance will produce a reduction of 6 dB in the sound pressure level.

In an enclosed field, hard surfaces will reflect sound waves. Sound reflections can increase the sound level within a room, making it easier to hear, as long as the reflections follow the original sound very rapidly. When the time between the incident wave (sound source) and the reflection is prolonged to a degree that is noticeable, it is termed reverberation.

A reflected sound, that occurs some time after the original, is heard as an echo. When a reflection follows closely, it adds to the direct sound. Reverberation follows with a short time delay, which may be destructive to intelligibility; with a greater delay, a distinct echo may be heard.

In a large, empty room, the reverberation effect may last for several seconds, which will make speech indistinct. Soft furnishings will absorb sound and help to reduce reverberation. Reverberation is particularly destructive to intelligibility for hearing aid users. It is not, however, desirable to cut out all reflection (except for research purposes, as in an anechoic chamber) as this makes the room very quiet and "dead".

Standing waves are caused when incident waves and reflected waves meet and combine to affect the sound level at that particular point. The interference that this creates may be constructive (increase the sound level) or destructive (decrease the sound level) (Figure 1-5).

When a standing wave is formed within the confines of an object, resonance occurs as, for example, in a pipe or in the ear canal. The length of the pipe or canal will affect the frequency of the standing waves formed within it. For example, if the length of a pipe is halved, the fundamental frequency of the air column within it is doubled and the resonant frequency will rise by one octave.

1.1.8 Warble Tones

Warble tones are often used in free-field audiometry to reduce the risk of standing waves affecting test results.

A warble tone is a frequency-modulated tone. The tone has a base or centre frequency, around which it varies. For example, a 1 kHz warble tone might vary from 950 Hz to 1050 Hz, which would be a frequency deviation of ±50 Hz or 5%. The tone therefore consists of frequencies above and below its centre frequency and it changes rapidly between these limits. The number of frequency changes occurring in one second is termed the "modulation rate" and is itself quoted in hertz (Hz).

1.2 The Measurement of Sound

1.2.1 Decibel Scales

The decibel scales express a ratio between two numbers.

Logarithms are used as a convenient method of expressing a ratio. A logarithm (log) tells us how many times the base number is multiplied by itself, 10^2 means 10 to the power of 2, that is, 10 is multiplied by itself once, similarly 10^3 is multiplied by itself twice, as follows:

$10^1 = 10$ therefore $\log_{10}(10) = 1$
$10^2 = 10 \times 10$ therefore $\log_{10}(100) = 2$
$10^3 = 10 \times 10 \times 10$ therefore $\log_{10}(1000) = 3$

The wide range of intensities involved is compressed by transforming it to a logarithmic scale. The unit of relative intensity is the bel, named after Alexander Graham Bell, who first patented the telephone. One bel equals ten decibels. The bel is rather too large a unit to reflect the accuracy required for audiometry. In measuring sound intensity or sound pressure level, the decibel scale is therefore used.

Any tenfold increase in sound pressure corresponds to 20 dB. For example, a noise level of 80 dB has a sound pressure that is 1000 times greater (10^3) than a noise of 20 dB. Subjectively, 10 dB appears as a doubling of loudness, whereas 1 dB is equivalent to the smallest change of intensity we can detect in ideal conditions.

In mathematical terms, for a measured sound pressure P (in pascals or Pa), the dB SPL value is given by the formula:

$$20 \log_{10}\left(\frac{P_1}{P_{ref}}\right)$$

where the audiological reference pressure = 0.00002 Pa.

So, for example, if:

$P_1 = 0.002$ Pa

using the formula:

$$20 \log_{10} \left(\frac{P_1}{P_{ref}} \right)$$

gives us:

$$20 \log_{10} \left(\frac{0.002}{0.00002} \right)$$

$= 20 \log_{10} 100$
$= 20 \times 2$ (since the log of 100 to base 10 = 2)
$= 40$ dB SPL.

For those who are less able mathematically, Table 1-1 shows the relationship between dB SPL and pascals.

The range of pressures to which the average normally hearing person is sensitive starts at just above 0 dB SPL (0.00002 Pa) and has a limit, where sound becomes uncomfortably loud, of 120 dB SPL (20 Pa). This range is called the dynamic range of hearing.

Since decibels express a ratio between two sound pressures, their values in decibels cannot simply be added together. Two sound sources of 40 dB SPL, for example, do not equal 80 dB SPL; in fact, this results in an increase of 3 dB, producing 43 dB SPL overall.

To be meaningful, a ratio must have a reference level. It is pointless to say that a sound is 10 times greater unless we also state what it is greater than. In audiometry, a number of different decibel scales are used and each has its own reference level. For example, the dB SPL scale (dB sound pressure level) expresses the pressure of a sound in relation to the standard audiological reference pressure, 0.00002 Pa, which is the minimum pressure required to cause the sensation of hearing in the mid-frequency region. The dB SPL scale compares with a flat or “absolute” reference: it takes no account of the way hearing varies with frequency.

Frequency (Hz)	RETSPL dB (Reference 20 μPa)
125	45.0
250	25.5
500	11.5
1k	7.0
2k	9.0
3k	10.0
4k	9.5
6k	15.5
8k	13.0

Table 1-2. RETSPL values for audiometric zero.

Other decibel scales include dB(A), dB(B), dB(C), dB(D) dB(Z) and dB HL. A range of scales is used because the human ear is not equally sensitive at all frequencies or intensities. The different decibel scales attempt to reflect these changes. For audiometric purposes dB(A) and dB HL (dB hearing level) scales are important. The dB HL scale is used when sounds are presented monaurally through air conduction headphones or bone conduction transducers. The dB(A) scale is used when the sound level is presented "free-field" in a room. The two scales are very similar but the dB(A) scale provides values that are about 4 dB greater than dB HL values.

The SPL (sound pressure level) values for audiometric zero (0 dB HL) for people aged 18 to 25 years at each frequency are given in BS EN ISO 389-1: 2018. The values are the recommended equivalent threshold sound pressure levels (RETSPL) in a 9A acoustic coupler. The RETSPL values provide average minimum audible pressure (MAP) under headphones, which may be illustrated as a hearing threshold curve (Figure 1-8) or a table providing values at the various audiometric frequencies (Table 1-2).

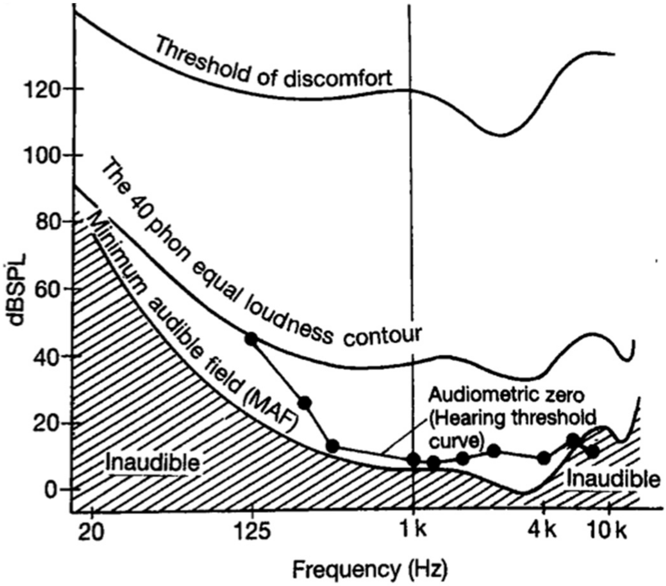

Figure 1-8. A diagrammatic representation of the dynamic range of human hearing.

The values were established on the basis of a statistical average obtained by testing many otologically normal people between the ages of 18 and 25 years. A similar method was used to obtain the minimum audible field (MAF), which sets out thresholds obtained binaurally, in response to sound pressures presented through loudspeakers. Thresholds taken from the MAF curve are an average of about 6 dB more sensitive than the normal MAP thresholds.

The normal increase observed in the threshold of hearing with age can be found in BS EN ISO 7029: 2017 "Statistical distribution of hearing thresholds as a function of age". This gives thresholds, for pure tones via earphones, at 2 kHz for ages from 18 to 80 years and from 3 kHz to 8 kHz for ages up to 70 years. Gender differences are included. Individuals can be compared to the norms for their age group. For extended high frequencies an expected median is given for 9 kHz to 12.5 kHz for ages 22 to 80 years.

1.2.2 The Sound Level Meter.

A sound level meter (Figure 1-9) is a precision instrument used for sound measurement. A precision microphone converts the sound signal to an electrical signal, which is amplified by a pre-amplifier before being processed.

The signal may be displayed in dB SPL or it may pass through a weighting network. This is an electronic circuit that varies in sensitivity across the frequency range to simulate the sensitivity of the human ear.

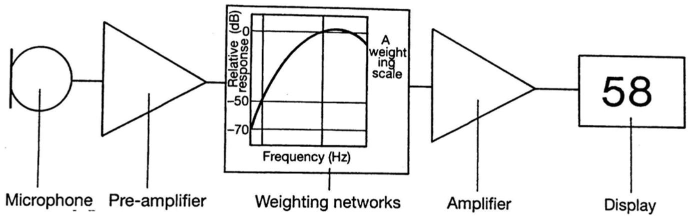

Figure 1-9. Constituent parts of a sound level meter using the A weighting network.

The weighting networks are termed A, B, C and D.

The A weighting network corresponds to an inverted equal loudness contour (see section 1.4.2) at low sound pressure levels. The B network corresponds to medium SPLs and the C network to high SPLs. The D network is used for aircraft noise measurements.

1.3 The Audiometer

1.3.1 Introduction

Hearing is a subjective sensation and therefore cannot truly be "measured". Audiometry measures the *stimulus* that causes the sensation of hearing.

An audiometer is an instrument of comparison, which indicates the difference between the sound pressure level required to produce hearing in the individual under test and that required to produce hearing in an average normal young person. An audiometer generates pure tones at specified points within a restricted range of frequencies considered important for communication, usually the octave intervals from 125 Hz to 8 kHz.

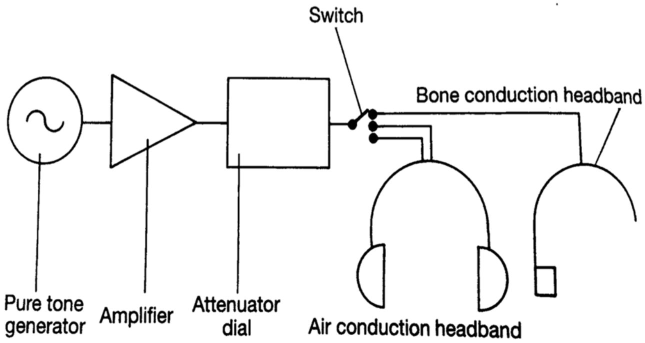

Figure 1-10. Constituent parts of an audiometer.

An octave is a doubling of frequency. Most audiometers also produce tones at the half-octave intervals, 750 Hz, 1.5 kHz, 3 kHz and 6 kHz. The sound level can also be varied, usually in 5 dB steps, from -10 dB HL to ll0dB HL or more. There should be a non-auditory warning indicator for levels above 100 dB HL. An interrupter switch controls the duration of the tones. The audiometer is provided with two types of transducer: a pair of headphones and a bone vibrator. A block diagram of a basic audiometer is shown in Figure 1-10.

1.3.2 Audiometer Calibration

An audiometer must be accurate to be of value and, to this end, all audiometers are calibrated to British Standards when new and the calibration should be re-checked at least once a year. The standard BS EN 60645-1: 2017 defines the levels of pure tone audiometer accuracy with which manufacturers should comply.

The main points of this standard, for our purposes, can be summarised as follows:

- The attenuator 5 dB steps must be correct within ± 1 dB.
- Unwanted sound from the audiometer should be inaudible up to and including the dial setting 50 dB HL over the main frequency range.
- The narrow band noise filters must be centred on the frequency (according to a given table).
- The rise and fall times of the signal tone for air conduction should not exceed 200 ms and for bone conduction should be at least 20 ms.

- The hearing level (reference tone) must be accurate to within ±3 dB from 500 Hz to 4 kHz and to within ±5 dB at the remaining frequencies.
- Total harmonic distortion (THD) should not exceed 2.5% for air conduction and 5.5% for bone conduction.

Laboratory calibration is carried out for the complete audiometer. The transducers (headphones and bone vibrator) are a part of the audiometer and should not be exchanged unless the audiometer is recalibrated with the new transducers.

In addition to calibration, the audiometer should be checked daily as in the following section.

1.3.3 Daily Audiometer Checks

- Check for wear and damage.
- Straighten any tangled leads. Ensure that all connections are firm and have good contact. Flex the leads for possible intermittency.
- Check that all knobs and switches are secure and function in a silent, smooth and click-free manner over their full range.
- Check function of response button.
- Check tension of AC and BC headbands.
- Check output levels for all tones at 10-15 dB above own known threshold for AC (each earphone) and BC. Your own audiogram should be used for this approximate calibration check.
- Check at 60 to 70 dB HL for all tones and functions on AC (each earphone) for noticeable distortion, intermittency and so forth. Repeat at 40 dB HL for BC.
- Check masking noise over a range of outputs, also loudness balance and other facilities, if they are to be used.
- Check battery condition, if appropriate.
- Generally ensure that the audiometer and all its attachments are clean. Wipe earphones and BC receiver with clean, dry tissues. Alcohol wipes may also be used as necessary.

1.3.4 Rise and Fall Times

The audiometer output signal is not an instantaneous event. The signal rises until it reaches its maximum and falls off in a similar manner when the tone is switched off (Figure 1-11).

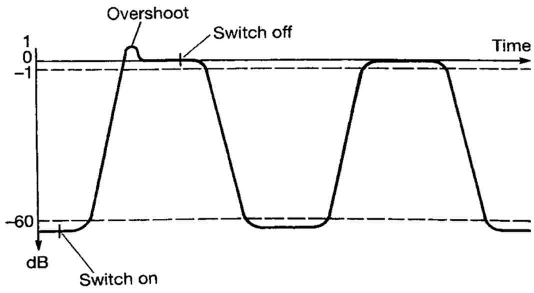

Figure 1-11. The rise and fall envelope of test tones. Between the dotted lines the sound should rise or fall in a progressive manner.

Rise time is defined as the time taken for the signal to rise from -60 dB to within 1 dB of its steady state.

Fall time is defined as the time taken for the signal to decay by 60 dB from its steady state.

This rise and fall (decay) should be correctly timed or it may affect the results of the audiometric test.

Too slow a rise time may result in erroneously poor thresholds, because this does not elicit the maximum on-effect of the ear. (The on-effect is the phenomenon whereby the initial onset of a sound is, subjectively, the loudest).

Too brief a rise time produces an overshoot that may be heard as a click. This may result in erroneously good thresholds if the patient responds to the click rather than the test tone. An overshoot should never be greater than 1 dB.

1.4 The Psychological Properties of Sound

1.4.1 Psychoacoustics

The study of the psychological properties of sound is known as psychoacoustics. The way in which we hear sound is subjective and cannot be directly measured. Subjective qualities are compared with reference levels obtained by averaging the judgements of a large number of normally hearing people.

Sensorineural hearing loss may reduce the ability to discriminate pitch. It may also increase the sensation of loudness, such that a small increase in objective intensity results in a large increase in subjective loudness. This abnormal loudness growth is termed *recruitment*.

1.4.2 Loudness

Loudness is the subjective perception of sound in terms of intensity. The relative loudness of pure tones is normally expressed as equal loudness contours. Each contour represents sounds that appear equally loud. Each contour or curve has a loudness value given in phons, which can be defined as the sound pressure level of a 1 kHz tone judged to be of equal loudness.

The 40 phon curve, for example (see Figure 1-8), represents pure tones across the frequency range, which are judged equally loud as a 1 kHz tone at an intensity of 40 dB SPL. An inverted 40 phon curve is used for the A weighting scale of a sound level meter. BS ISO 226: 2003 gives normal equal loudness level contours from 10 to 100 phons (although the two extremes are based on limited data).

Another unit of loudness, the sone, is used for the purpose of providing a scale to define loudness. This numerical scale of loudness is based on judgements of average listeners as to when sounds are "twice as loud" or "half as loud". One sone is taken to be the loudness of a 1 kHz tone at 40 dB SPL. Thus, 1 sone has a loudness level of 40 phons. A sound that has a loudness of 2 sones is one that is judged to be twice as loud.

In terms of sound pressure level, an increase of 10 dB corresponds to a doubling of loudness.

1.4.3 Pitch

Pitch is the subjective perception of sound in terms of frequency. The unit of pitch used is the mel. One thousand mels represents the pitch of a 1 kHz tone presented at 40 phons. The ability to recognise the pitch of sounds of different frequencies is known as "frequency resolution". The average normal person can discriminate a frequency difference where a change of 3 Hz occurs, although this varies across the audible spectrum.

1.4.4 Temporal Integration

The effect of duration on the recognition of sound is known as “temporal integration”. Tone presentations that are very brief may not be heard. The shorter a tone presentation, the more intense must be that sound if it is to be perceived. Duration also has an effect on the apparent pitch of a sound. Tones that are of very short duration will be perceived as a click.

1.5 Summary

For sound to exist it must be in our audible frequency range, which is normally 20 Hz to 20,000 Hz, and there must be an elastic medium to convey the vibrations.

Simple harmonic motion produces a pure tone, which is a sound of one frequency. Sound is created by a vibrating object, which produces alternate compressions and rarefactions in the medium, which is usually air.

The frequency of vibration is calculated by the number of vibrations or cycles in 1 second. The time taken for 1 cycle is known as the period. The unit of frequency is hertz; 1 Hz = 1 cycle per second.

The decibel scale is used to describe the intensity of sound. Decibels are not absolute units of measurement but provide a ratio between a measured quantity and an agreed reference level, which must be specified. An audiometer provides a comparison between the hearing of the subject under test and average normal hearing as set out in the relevant British Standard.

Audiometers must be regularly checked and calibrated to ensure accuracy.

Sound can be measured objectively in terms of frequency, intensity or pressure and duration. Hearing is subjective and may be described in terms of pitch and loudness.

CHAPTER TWO

ANATOMY AND PHYSIOLOGY OF THE EAR

2.1 Prenatal Development of the Ear

Pregnancy is divided into three trimesters; each consisting of about three months. The embryo develops rapidly during the first trimester; the primary germ layers, (endoderm or internal layer, mesoderm or middle layer and ectoderm or external layer) are formed and organized. The ectoderm forms the exoskeleton, the mesoderm forms the organs and endoderm forms the lining of the organs.

The pharyngeal arches are the structures that contribute to the head and neck formation. Each arch consists of the three primary germ layers. Each arch is separated externally by a groove or cleft and internally by a pouch or pocket.

In the second trimester, the embryo is known as a foetus or fetus.

(a) Development of the outer ear

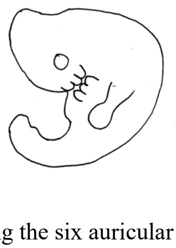

Figure 2-1. The embryo, showing the six auricular hillocks.

The pharyngeal arches form the main structures of the head and neck, including the outer and middle ears. Between weeks 4 to 8, the ear develops (Figure 2-1) from six auricular hillocks, which form the first and second arches.