Physics notes – Sound

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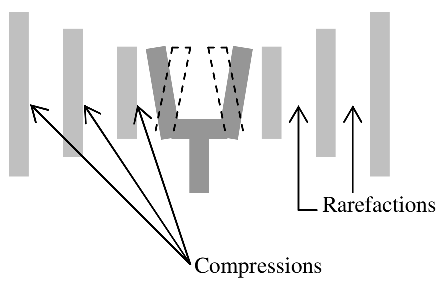
In the study of sound, we learn about the **source**, the **medium** and the **receiver**. Sound energy is transmitted from the source to the receiver via the medium.

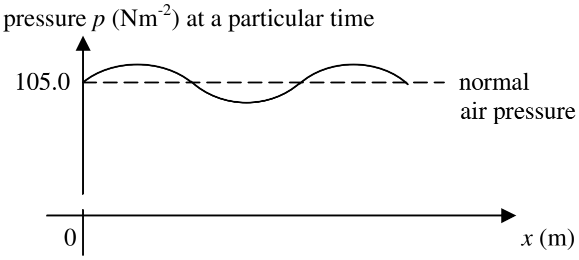
An example of the source is a tuning fork. It provides sound of a single **frequency** *f*. If the arms of the tuning fork vibrate 261 times in a second, we say its frequency of vibration is 261 Hz and the produced sound (vibration of the molecules in air, the medium) has the same frequency.

As the arms of the tuning fork vibrate, a series of high (**compression**) and low (**rarefaction**) pressure regions in the air (the medium) is generated and propagates outwards. This series of compressions and rarefactions constitutes a **travelling sound wave** in the air.

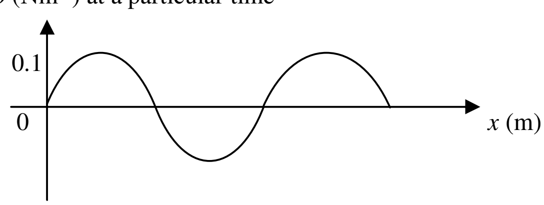


Sound wave in terms of air pressure variation

Air pressure p (Nm^{-2}) at a particular time



Δp (Nm^{-2}) at a particular time

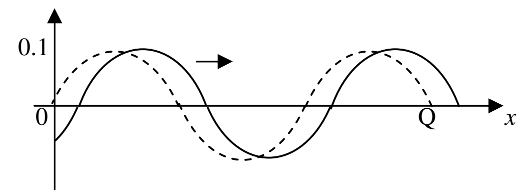


The last graph shows the pressure **variation** of the medium (air) versus distance in front of the source at a particular time.

The distance from one compression (or rarefaction) to the next is called a **wavelength** λ.

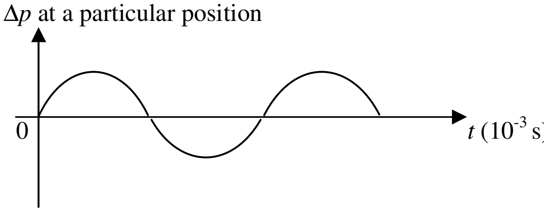
It is a travelling wave. The pattern moves away from the source as time progresses.

Δp at a latter time



Another way to describe the sound wave is to stand at a particular position (e.g. position Q) in front of the source and record the pressure variation as a function of time *t*.

Δp at a particular position



The interval from one high (or low) to the next is called the **period** *T* of the sound wave.

The relationship between *f* and *T* is $f = \frac{1}{T}$.

The speed of sound

Use the usual formula $v = \frac{d}{t}$ to find the speed of sound, where *d* (m) is the distance travelled by the sound and *t* (s) is the time taken. The unit for *v* is ms^{-1}.

Example The timekeeper at an athletics meeting is 100 m from the starter. The time lag between the timekeeper seeing the flash of the starter's pistol and hearing the sound is 0.29 s. Determine the speed of sound.

$v = \frac{d}{t} = \frac{100}{0.29} = 345$ ms^{-1}.

Speed of sound in different media

The speed of sound depends on temperature and the type of medium that it travels in.

E.g.	Air (20° C)	344ms^{-1}
	Water (20° C)	1498ms^{-1}
	Iron (20° C)	5120ms^{-1}

The relationship between speed *v* (ms^{-1}) and temperature *T* (K) is

$$v \propto \sqrt{T}\ .$$

The wave equation

A sound wave travels a distance of one wavelength in a time interval of one period, $\therefore v = \frac{\lambda}{T}$, and since $f = \frac{1}{T}$, $\therefore v = f\lambda$.

The last equation is known as the **wave equation**.

Longitudinal and transverse waves

Depending on the direction of motion of the particles of the medium relative to the direction of propagation, a wave can be classified as longitudinal or transverse.

It is a **longitudinal wave** when the particles of the medium oscillate parallel to the direction of propagation of the wave. If the oscillation is perpendicular to the direction of propagation, it is called a **transverse wave**.

Sound waves in air and under water are longitudinal. In a solid, a sound wave can be either longitudinal or transverse.

Sound intensity

Sound **intensity** I measures the amount of sound energy E arriving at a particular surface of area A, over a time interval Δt. It is defined as $I = \frac{E}{A\Delta t}$, and since power $P = \frac{E}{\Delta t}$, $\therefore I = \frac{P}{A}$.
The units are W or Js^{-1} for P, m^2 for A, and Wm^{-2} or $Js^{-1}m^{-2}$ for I.

Sound intensity is an **objective** measure. Anyone handling the measuring device correctly will record the same reading. The perceived loudness of the sound by a person is a **subjective** sensation. In general, the greater the intensity, the louder is the sound. But perceived loudness is not directly proportional to intensity.

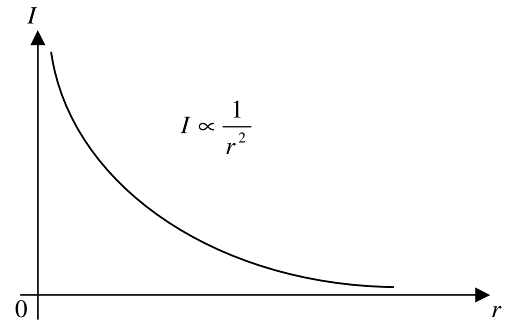
Example 1 If a person's eardrum had an effective cross-section area of 0.5 cm^2, what would be the power entering her ear given the intensity at her ear was 0.2 mWm^{-2}? In 5.0 s, how much sound energy would she receive?

$$I = \frac{P}{A},\ P = IA = (0.2\times10^{-3})(0.5\times10^{-4}) = 1.0\times10^{-8}\text{ W}.$$
$$P = \frac{E}{\Delta t},\ E = P\Delta t = (1.0\times10^{-8})(5.0) = 5.0\times10^{-8}\text{ J}.$$

Inverse square law for small sound sources

For a small (in size) sound source in the open, the sound energy spreads outwards spherically. Suppose the power P of the source is constant, at a distance of r metres away from the source, the intensity I is given by $I = \frac{P}{A} = \frac{P}{4\pi r^2}$, therefore, $I \propto \frac{1}{r^2}$. This is known as the **inverse square law**.

When the distance from the source is doubled, the intensity drops to a quarter ($\frac{1}{2^2} = \frac{1}{4}$) of the initial value. When the distance from the source is halved, the intensity becomes four times ($\frac{1}{\left(\frac{1}{2}\right)^2} = 4$) of the original value.



Example 1 At 2.4 m away from a bell, the intensity is 9.0×10^{-4} Wm^{-2}. What is the intensity at 7.2 m away? At 1.2 m away?

7.2 m is 3 times 2.4 m, $\therefore$ the intensity is $\frac{1}{3^2} = \frac{1}{9}$ of the original value, i.e. $I = \frac{1}{9}(9.0\times10^{-4}) = 1.0\times10^{-4}$ Wm^{-2}.

1.2 m is $\frac{1}{2}$ of 2.4 m, $\therefore$ the intensity is $\frac{1}{\left(\frac{1}{2}\right)^2} = 4$ times the original value, i.e. $I = 4(9.0\times10^{-4}) = 3.6\times10^{-3}$ Wm^{-2}.

If the distance is not a simple integral multiple of the original, use $\frac{I_f}{I_i} = \frac{r_i^2}{r_f^2}$.

Example 2 Refer to example 1. What is the intensity at 5 m away?

$$\frac{I_f}{I_i} = \frac{r_i^2}{r_f^2},\ I_f = \frac{r_i^2}{r_f^2}\times I_i = \left(\frac{r_i}{r_f}\right)^2 \times I_i$$
$$= \left(\frac{2.4}{5}\right)^2 \times (9.0\times10^{-4}) = 2.1\times10^{-4}\text{ Wm}^{-2}.$$

Range of sound intensity suitable for human hearing

	Wm^{-2}
Threshold of hearing for a young person	10^{-12}
Normal conversation	10^{-6}
Car alarm at one metre away	10^{-2}
Threshold of pain, rock concert	1
Jet engine at 30m away	10^{2}

Our ears can respond to a huge range of intensity, from $I = 10^{-12}$ wm^{-2} to $I = 1$ wm^{-2}. However, we do not perceive the loudest sounds are 10^{12} times as loud as the softest. To produce a sound that perceived as twice as loud requires a sound wave that has about 10 times the intensity. For example, a sound of 10^{-2} Wm^{-2} is about twice as loud as a sound of 10^{-3} Wm^{-2}, about 4 times as loud as 10^{-4} Wm^{-2}, 8 times as loud as 10^{-5} Wm^{-2}. Our ears are progressively less sensitive to sound as the intensity increases.

Because of this relationship between perceived loudness and intensity, this enormous range in intensity I can be converted to a more manageable range by defining another measure called **sound intensity level** L.

Sound intensity level L

By definition, $L = 10\times\log_{10}\left(\frac{I}{10^{-12}}\right)$.

The unit for L is decibel (dB).

In this definition, the intensity I of a sound is compared with the threshold of hearing, 10^{-12} Wm^{-2}, which is chosen as the reference intensity to define 0 dB.

The sound intensity level for the threshold of pain is 120 dB.

Example 1 Convert 5×10^{-8} Wm^{-2} to dB.

$$L = 10\times\log_{10}\left(\frac{I}{10^{-12}}\right) = 10\times\log_{10}\left(\frac{5\times10^{-8}}{10^{-12}}\right) \approx 47 \text{ dB}.$$

To convert sound intensity level L to sound intensity I, transpose the formula to $I = 10^{\frac{L}{10}-12}$.

Example 2 Convert 62 dB to Wm^{-2}.

$$I = 10^{\frac{L}{10}-12} = 10^{\frac{62}{10}-12} \approx 1.6\times10^{-6} \text{ Wm}^{-2}.$$

Everyday sounds and noises

	dB
Threshold of hearing for a young person	0
Whisper	20
Quiet radio in home	40
Normal conversation	60
City traffic	80
Car alarm at one metre away	100
Threshold of pain, rock concert	120
Jet engine at 30m away	140

Change (or difference) in sound intensity level, ΔL

The difference in sound intensity level between two given sources of intensities I_1 and I_2 is $\Delta L = 10\times\log_{10}\left(\frac{I_2}{I_1}\right)$.

The change in sound intensity level when the intensity changes from I_i to I_f is $\Delta L = 10\times\log_{10}\left(\frac{I_f}{I_i}\right)$.

It is the ***ratio*** of the intensities (NOT the individual intensity values) that affects the change (or difference) in sound intensity level. For example,

when the intensity is doubled, i.e. $\frac{I_f}{I_i} = 2$, $\Delta L = +3$ dB,

when it is halved, i.e. $\frac{I_f}{I_i} = \frac{1}{2}$, $\Delta L = -3$ dB,

when the intensity is 10 times, i.e. $\frac{I_f}{I_i} = 10$, $\Delta L = +10$ dB,

irrespective of the values of I_i and I_f.

When the sound intensity level increases by about 10 dB, the perceived loudness of the sound is doubled.

For example, a 70 dB sound is about twice as loud as a 60 dB sound, 4 times as loud as 50 dB, 8 times as loud as 40 dB.

Example 1 Find the change in sound intensity level when the distance from the source is doubled.

According to the inverse square law, when the distance from the source is doubled, $\frac{I_f}{I_i} = \frac{1}{4}$.

$\therefore \Delta L = 10\times\log_{10}\left(\frac{I_f}{I_i}\right) = 10\times\log_{10}\left(\frac{1}{4}\right) = -6$ dB, i.e. decrease by 6 dB.

Alternative method:

$I_f = \frac{1}{4}I_i = \frac{1}{2}\times\frac{1}{2}I_i$, i.e. I_i is halved and halved again. So the intensity level decreases by 3 dB and 3 dB again, a total of 6 dB.

Example 2 Find the change in sound intensity level when you move closer to the source from 3.0 m to 1.0 m. What is the new sound intensity level if the original was 50.0 dB?

When you move closer to the source from 3m to 1m, your final distance is $\frac{1}{3}$ of your initial distance from the source. According to the inverse square law, $\frac{I_f}{I_i} = 9$.

$$\therefore \Delta L = 10\times\log_{10}\left(\frac{I_f}{I_i}\right) = 10\times\log_{10}(9) \approx +9.5 \text{ dB}.$$

$$\therefore L_f = L_i + \Delta L = 50.0 + 9.5 = 59.5 \text{ dB}.$$

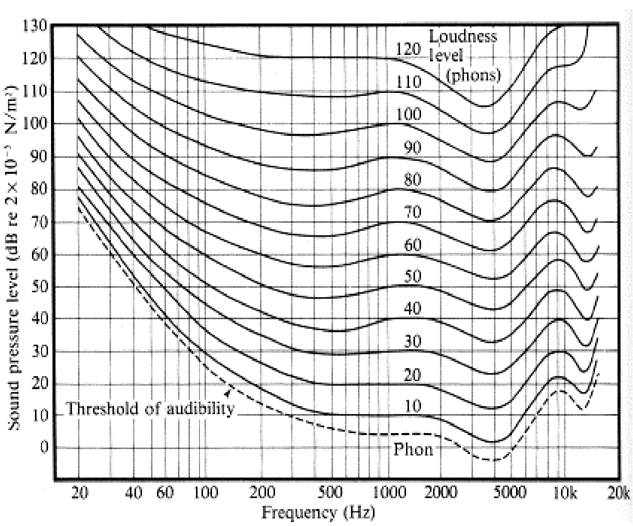
Frequency response of an 'average' human ear

The perceived loudness of a sound also changes with its frequency besides intensity. Human ear is most sensitive to sound of frequency around 4000 Hz.

Of the three sounds, 100 Hz, 4000 Hz and 10 000 Hz, at the same dB level at the ear, the 4000 Hz will sound louder to the listener.

To make the 100 Hz and 10 000 Hz the same loudness to the listener as the 4000 Hz, their dB levels have to be increased.

The frequency response of human ears is best shown by equal loudness curves. The following diagram shows a set of such curves.



Curves of equal loudness determined experimentally by Fletcher, H. and Munson, W.A. (1933) J.Acoust.Soc.Am. 6:59.

Loudness (measured in phons)

The loudness of a sound is usually compared with that of a 1000 Hz sound and it is measured in phons. The loudness of a 10 dB 1000 Hz sound is 10 phons, the loudness of a 80 dB 1000 Hz sound is 80 phons etc. Sounds at different frequencies, which are as loud as the 80 dB 1000 Hz sound, have a loudness of 80 phons.

Example 1   At what dB level do the 100 Hz and 4000 Hz sounds have the same loudness of 80 phons?

Read the 80-phon equal loudness curve. For 100 Hz, $L = 85$ dB. For 4000 Hz, $L = 70$ Hz.

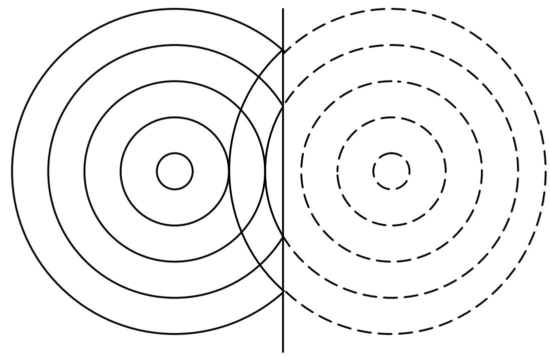
Example 2   Give the level and frequency of a sound that is (a) louder (b) softer than 80 phons.

(a) For example, 300 Hz at 90 dB.

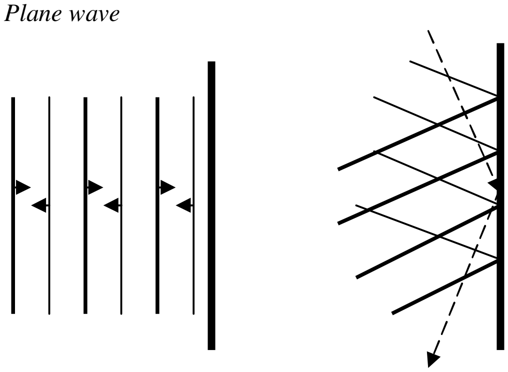
(b) For example, 30 Hz at 90 dB.

Reflection of sound waves

Spherical sound wave



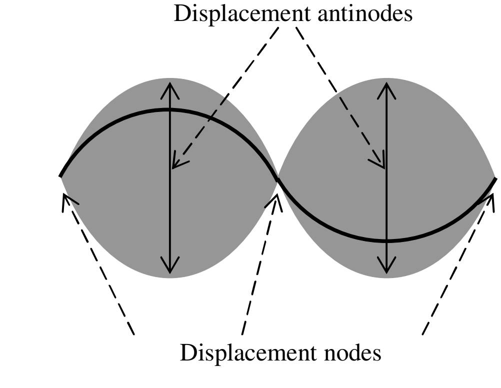
Plane wave



After reflection, the frequency, the wavelength and the wave speed remain the same.

Interference of sound waves

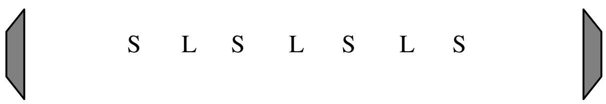
Interference refers to the crossing of two or more waves. For waves of the same frequency, a definite pattern called **interference pattern** is produced,
For example, consider two travelling waves of the same frequency moving in opposite direction in a stretched string (or spring). The pattern created is called a **standing wave**. It is formed by the **superposition** of the two travelling waves at different stages of their propagation. The following diagram shows a standing wave.

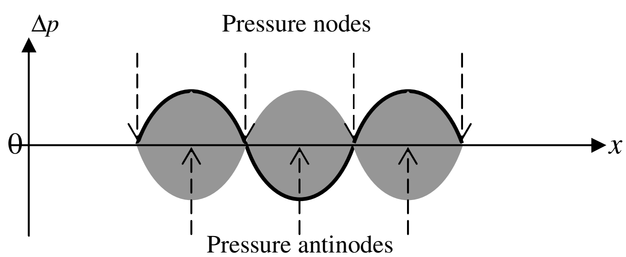


The positions where the string is always at rest are called **displacement nodes**. At a displacement node the superposition of the two travelling waves in the string always gives rise to **destructive** interference (cancellation). Maximum vibration occurs at the **displacement antinodes**, which is the result of **constructive** interference (addition) of the two waves.

A standing sound wave can also be generated by placing two sound sources facing each other sending out travelling sound waves at the same frequency; or a single source facing a solid wall so that the forward travelling wave interferes with the reflected wave.

A series of soft S (**pressure nodes**, destructive interference) and loud L (**pressure antinodes**, constructive interference) spots are formed between the two sources. If the two sources are **in phase**, the mid-point is a pressure antinode.





Note that the nodal separation (i.e. distance between two adjacent nodes) is $\frac{\lambda}{2}$.

The vibration of a medium that is caused by the formation of a sustained standing wave in the medium is called **resonance**.

Every object has its own **natural frequencies** of vibration (modes of vibration). If an energy source at one of these frequencies interacts with the object, the latter will be set into vibration with constant or increasing amplitude, i.e. a standing wave will be formed. We say the object resonates when it is forced into vibration by an energy source at a matching frequency. The natural frequencies of vibration are called the **resonant frequencies** of the object.

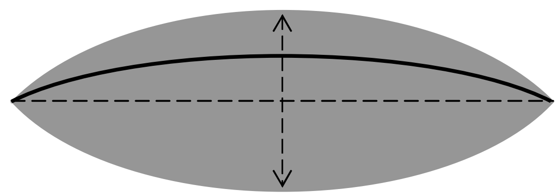
Example 1 A stretched guitar string on its own produces very soft sound because of its small area of contact with the air. When it is connected to the guitar box, its frequency matches one of the resonant frequencies of the box, causing the box to resonate. The large contact area of the box with the air gives a louder sound.

Example 2 A high pitch note sung by a soprano could cause a thin wine glass to shatter. The frequency of the note matches one of the resonant frequencies of the glass and sets it into vibration with increasing amplitude. The glass shatters if the amplitude is too large for the glass to withstand.

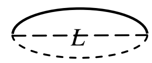
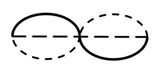
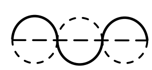
Standing waves and stringed instruments

For a stretched string, the frequency of vibration (standing wave) depends on the tension, the length and the type of string used. If these quantities remain constant, only vibrations of certain frequencies are possible. These frequencies are **integral multiples** of the lowest one. The lowest frequency of vibration is called the **fundamental frequency**.

Consider a stretched string of length *L*. The diagram below shows the vibration of the string at its fundamental frequency. It is a standing wave with displacement nodes at the ends and an antinode in the middle.



Other modes of vibrations are possible at higher frequencies.

Modes	Overtones	λ	$f = v/\lambda$	Harmonics
	Fundamental	$2L$	$1\left(\frac{v}{2L}\right)$	First har.
	First o'tone	L	$2\left(\frac{v}{2L}\right)$	Second har.
	Second o'tone	$\frac{2L}{3}$	$3\left(\frac{v}{2L}\right)$	Third har.

v is the speed of the *travelling wave in the stretched string*.

The term **harmonics** is used to describe the different modes of vibration <u>only</u> when they are integral multiples of the fundamental frequency, $f_n = n\left(\frac{v}{2L}\right)$, $n = 1, 2, 3, \ldots$..

Note that the frequency of vibration is directly proportional to the speed and inversely proportional to the length of the string.

The speed of the travelling wave depends on the tension and the type of string used.

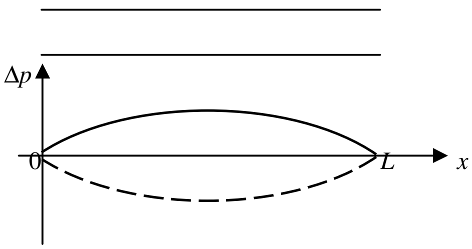
In the case of a guitar string, a short thin string in high tension produces a high pitch sound.

Other types of musical instruments that produce harmonics are the wind instruments. They can be modeled as **open** or **closed resonant tubes**.

Standing waves and wind instruments

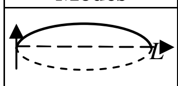
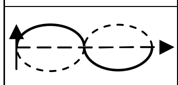
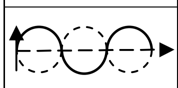
In a wind instrument it is the vibration of the air column in the resonant tube forming a standing sound wave.

Consider an **open resonant tube** of length L. The fundamental frequency of vibration of the air column is shown below in terms of pressure variations.



The open ends are always pressure nodes and in the middle is a pressure antinode where maximum variation in air pressure occurs.

Other modes of vibrations are possible at higher frequencies.

Modes	Overtones	λ	$f = v/\lambda$	Harmonics
	Fundamental	$2L$	$1\left(\frac{v}{2L}\right)$	First har.
	First o'tone	L	$2\left(\frac{v}{2L}\right)$	Second har.
	Second o'tone	$\frac{2L}{3}$	$3\left(\frac{v}{2L}\right)$	Third har.

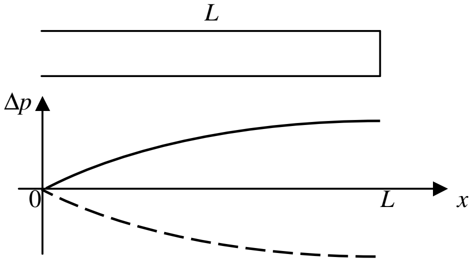
v is the speed of the *travelling sound wave in the air*, and

$$f_n = n\left(\frac{v}{2L}\right), n = 1, 2, 3, \ldots\ldots$$

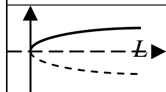
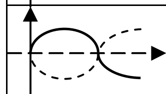
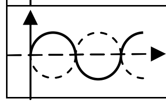
Since the frequency of vibration of the air column depends on the speed of sound, it may be necessary to retune the musical instrument when there is a change in temperature.

Like stringed instruments, open resonant tube wind instruments can produce **all** harmonics, $n = 1, 2, 3, \ldots\ldots$

Closed resonant tube wind instruments can produce only the **odd** harmonics, $n = 1, 3, 5, \ldots\ldots$



It is always a pressure antinode at the closed end.

Modes	Overtones	λ	$f = v/\lambda$	Harmonics
	Fundamental	$4L$	$1\left(\frac{v}{4L}\right)$	First har.
	First o'tone	$\frac{4L}{3}$	$3\left(\frac{v}{4L}\right)$	Third har.
	Second o'tone	$\frac{4L}{5}$	$5\left(\frac{v}{4L}\right)$	Fifth har.

$f_n = n\left(\frac{v}{4L}\right)$, $n = 1, 3, 5, \ldots\ldots$, i.e. only the odd harmonics.

Example 1 What will be the fundamental frequencies and first three overtones for a 26cm long organ pipe at 20° C (speed of sound 343ms^{-1}) if it is (a) open and (b) closed?

(a) $f_n = n\left(\frac{v}{2L}\right)$, $f_1 = \frac{343}{2\times 0.26} = 660$ Hz,

$f_2 = 2\times 660 = 1320$ Hz, $f_3 = 3\times 660 = 1980$ Hz.

(b) $f_n = n\left(\frac{v}{4L}\right)$, $f_1 = \frac{343}{4\times 0.26} = 330$ Hz, $f_3 = 3\times 330 = 990$ Hz,

$f_5 = 5\times 330 = 1650$ Hz.

Example 2 A flute is designed to play the middle C (264Hz) as the fundamental frequency when all the holes are covered. How long should the length be from the mouthpiece to the end of the flute at 20° C? (A flute can be modelled as a closed resonant tube with the closed end at the mouthpiece.)

$f = \frac{v}{4L}$, $264 = \frac{343}{4L}$, $L = 0.325$ m.

Example 3 If the temperature is only 10° C (speed of sound 337 ms^{-1}), what will be the frequency of the note played when all the openings are covered in the flute of the previous example?

$f = \frac{v}{4L} = \frac{337}{4\times 0.325} = 259$ Hz.

Example 4 The voice of a person who has inhaled helium sounds very much like Donald Duck. Why?
(Speed of sound in helium ≈ 1005 ms^{-1})

The tract configuration remains the same when one tries to pronounce the same vowel as before with a throat full of helium. The vocal chord still vibrates the same way, so the harmonics generated occur at the same frequencies. But the speed of sound is greater, so resonances occur for those harmonics (of the vocal chord) at higher frequencies. $f = \left(\frac{v}{4L}\right)$, L is constant.

For some musical instruments, the overtones are not harmonics because the higher frequencies are not integral multiples of the fundamental.

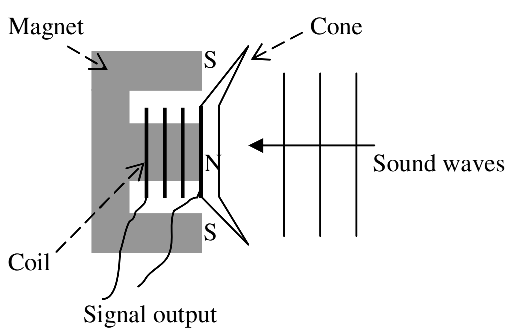
E.g. Percussion instruments like drums, the frequency ratios of each overtone to the fundamental are: 1.59, 2.14, 2.30, 2.65, 2.92, 3.16, 3.50 etc, which show the drum is far from harmonic.

Recording and reproducing sound –

Microphones

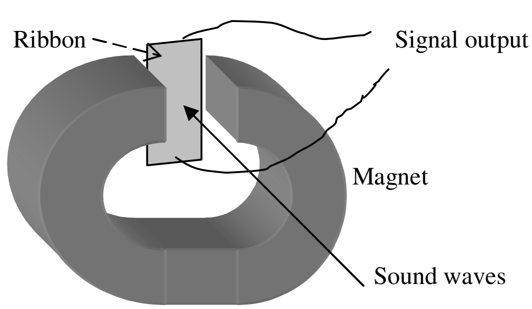
Microphones are input transducers in which sound energy is transformed to electrical energy. The most common microphones for musical use are **dynamic**, **ribbon** (**velocity**) or **condenser** microphones because of their relatively good frequency response. **Crystal** microphones have a larger electric output, but the frequency response is poorer in comparison with a good dynamic microphone.

Dynamic microphones



Sound moves the cone and the attached coil of wire in the magnetic field forwards and backwards. This causes the coil of wire to cut across the magnetic field lines. Electromagnetic induction produces an emf (signal output) at the terminals of the coil.
A dynamic microphone works without any power supply and provides a reliable signal under a wide range of environmental conditions. It is quite rugged and can accept very loud sound levels. It has a slower transient response than the other microphones and therefore can be used to soften the fine detail that other microphones would pick up. It is a good choice for woodwinds or brass, if you want to take the edge off the sound.

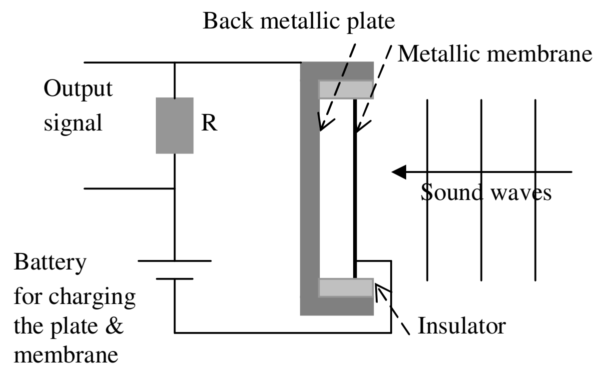
Ribbon microphones



The air movement due to the sound waves moves the metallic ribbon in the magnetic field. This causes the ribbon to cut across the magnetic field lines. Electromagnetic induction generates an emf (signal output) between the ends of the ribbon. The name *velocity* microphone comes from the fact that the induced emf is proportional to the velocity of the ribbon.

Ribbon microphones are more delicate than the dynamic type. They are often prized for their warm, smooth tone quality. Typically they are used on brass instruments to mellow the tone.

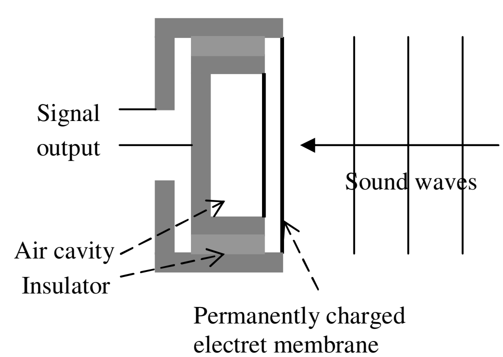
Condenser microphones



The back plate and the membrane form a capacitor (condenser). Sound waves cause the metallic membrane to vibrate and change the spacing between the membrane and the back metallic plate. A change in spacing results in a change in the amount of charge on the plate and the membrane, and thus forces a varying current through resistor R to produce a signal output.

Electret condenser microphones

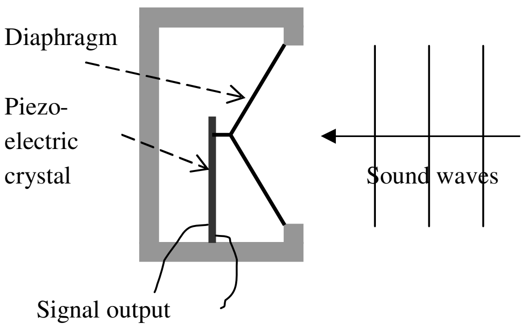
Electret microphones are condenser microphones, which use a permanently charged electret material for their membranes, thus eliminate the necessity for the charging battery. The operation of an electret microphone has the same principle as for a condenser microphone.



Because of its lower membrane mass and higher damping, a condenser microphone responds faster than a dynamic microphone to transients. It provides a smooth, detailed sound with a wide frequency response. It is especially suitable for micing cymbals, acoustic instruments and studio vocals.

Crystal microphones

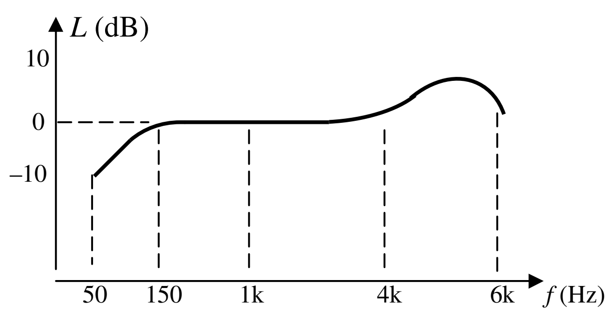
Piezoelectric crystals produce voltages when they are deformed. The crystal microphone employs a thin strip of piezoelectric crystal attached to a diaphragm. The two sides of the crystal become oppositely charged when it is deflected by the diaphragm that is sent into vibration by sound waves.



The signal output of crystal microphones is comparatively large, but the frequency response is not comparable to a good dynamic microphone. They are usually used as build-in microphones in, for example, tape-recorders.

Frequency response curves of microphones

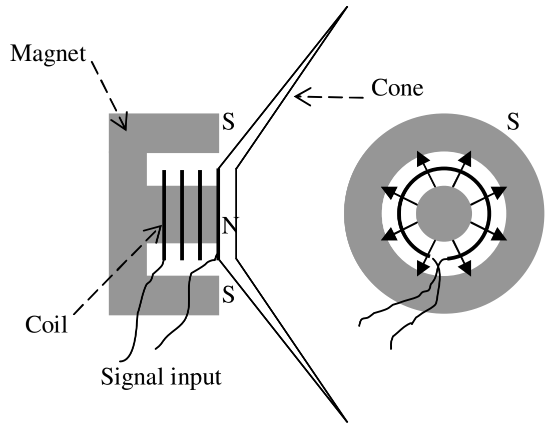
The frequency response curves of microphones are different from the equal loudness curves for human ears. Such a frequency response curve is a graph of output intensity level versus frequency for a constant level input. Zero dB is assigned to 1000-Hz sound as the reference level. The output level for other frequencies will be measured against this reference level. If for some frequencies $L < 0$, it means the device does not respond well to these frequencies. The following graph shows the frequency response of a workhorse microphone like the Shure SM-57.



This microphone can reliably detect frequencies from 150 Hz to 6 kHz.

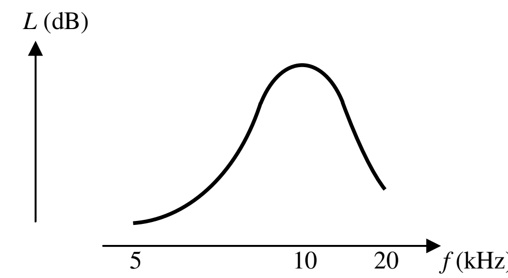
Loudspeaker systems

Loudspeakers are output transducers that change electrical signals to sound energy. A common type of loudspeakers for reproduction of sound is the dynamic type.
A **dynamic loudspeaker** works on the principle that a magnet exerts a force on a current-carrying wire. The input signal changes the current in the wire and results in a varying magnetic force on the coil, which is attached to the cone.



Dynamic loudspeakers and microphones have the same basic construction. In fact a dynamic loudspeaker can be used as a microphone.

A single loudspeaker on its own tends to 'colour' the sound it produces. This is because it reproduces some frequencies louder than others due to resonance. The following graph shows the frequency response of a typical loudspeaker.



An ideal loudspeaker would need to have the same loudness at all frequencies in the audible range. This can be achieved by using more than one loudspeaker and put them together in a box.

Baffles and enclosures for loudspeakers

A direct radiating loudspeaker by itself has a few problems:
(a) The sound from the back of the speaker cone tends to cancel the sound from the front – interference.
(b) The cone resonates when the signals are at or near its natural frequency of vibration – resonance.
(c) The free cone speaker is inefficient at producing sound with wavelengths longer than the diameter of the cone – diffraction (refer to next page).

Enclosures formed by baffles are used to rectify some of these problems. Baffles are used to control the flow of sound out of an enclosure. The sound quality will improve greatly by placing a single dynamic loudspeaker in an **enclosure**, i.e. closed baffle (diagram 1 below). The loudspeaker is mounted on the front baffle of the enclosure. This will suppress the low-frequency cancellation found in a naked loudspeaker. Why low frequency?

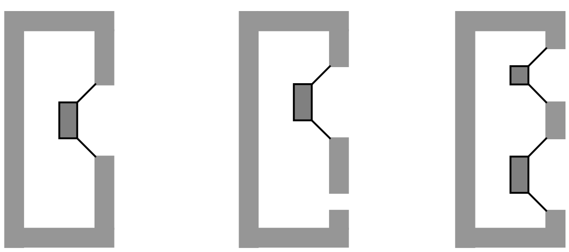
At low frequencies the wavelengths of sound are much longer than the diameter of the cone; the sound waves radiate beyond the outer edge of the cone due to diffraction. When the front and rear sound waves meet at the outer edge of the cone, destructive interference occurs because they are **out of phase** with each other. This results in little low frequency output from the loudspeaker.

Usually the enclosure has an acoustic resonant frequency that is lower than the resonant frequency of the loudspeaker. This effectively lowers the low cutoff frequency of the entire system.

Bass reflex baffle (diagram 2) is used to extend the useful bass (low frequency) range of the loudspeaker.

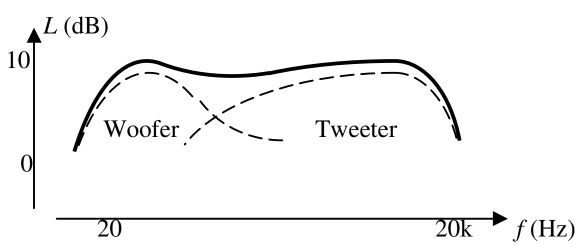
Modern loudspeaker boxes involve multiple loudspeakers to provide a more uniform frequency response across the audio frequency range. Most common systems are those with two loudspeakers (diagram 3), woofer (low frequency, 20 – 3000 Hz) and tweeter (high frequency, 1500 – 20 kHz). Low-pass filters are used to ensure low-frequency signals are sent to the woofer and high-pass filters for high-frequency signals to the tweeter. If a mid-range loudspeaker is included as well in the system, then a band-pass filter is used for the mid-range signals. This combination of filters is known as a **crossover**. The frequencies where the two filters overlap are called **crossover frequencies**.

(Ref: Geoff Martin 2002)



Closed baffle | Bass-reflex baffle | Multi-speaker baffle

Frequency response of multi-speaker systems

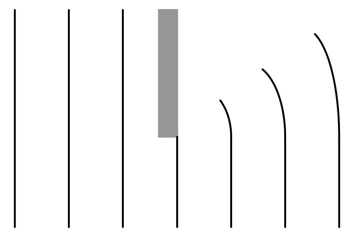


Loudspeaker designers take a lot of care when they put loudspeakers together. When the effects from all of them are added together they should give a fairly **flat** response curve, usually ± 3 dB. Each loudspeaker has to give just the right amount of each frequency. Some loudspeaker enclosures have tubes (called **ports**) put in them. The size and depth of the port can be changed to absorb sounds of a particular frequency in order to produce a flat frequency response.

Note: What is important in a frequency response curve for microphone or loudspeaker system is not what the actual numbers are, but how much they vary from frequency to frequency.

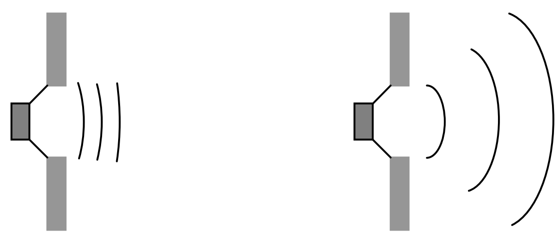
Diffraction of sound waves

Sound **diffracts** when it passes by the edge of a barrier, i.e. it spreads out and bends behind the barrier.



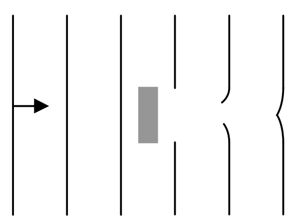
Diffraction through an opening

E.g. an opening in a baffle



Diffraction around an obstacle

E.g. a column in the path of a sound wave



The extent of diffraction $\propto \frac{\lambda}{w}$, where λ is the wavelength and w, the width of the opening or obstacle.

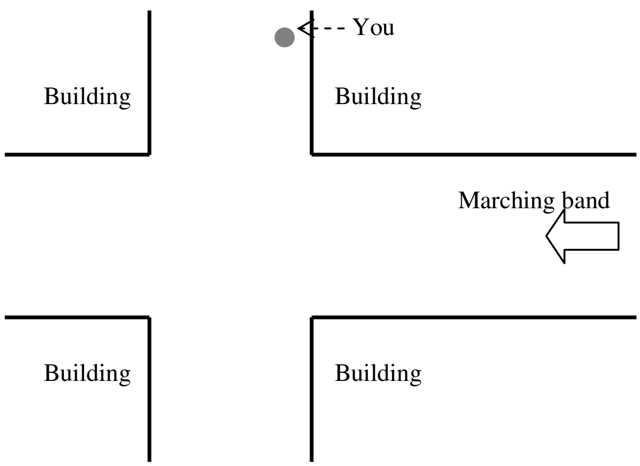
Diffraction is significant when the value of $\frac{\lambda}{w}$ is about 1 or greater than 1.

Low pitch (frequency) sound diffracts more than high pitch sound because its wavelength is longer.

After passing through (by) an opening (obstacle), frequency, wavelength and wave speed of sound remain the same.

A loudspeaker is **omni-directional**, i.e. it radiates sound energy spherically in all directions, when the diameter *w* of the speaker cone is less than about ¼ of the wavelength of the frequency being produced, i.e. $\frac{\lambda}{w} > 4$. The higher the frequency (the shorter the wavelength), the more directional is the loudspeaker. This is a problem because if your ear is not **on axis** (i.e. your ear is not more or less directly in front of the loudspeaker), you are likely to get some high-frequency **roll-off** (i.e. miss the high-pitch sound).
(Ref: Geoff Martin 2002)

Example 1 If a marching band is approaching an intersection, which instruments, piccolo or bass drum, will you hear first?

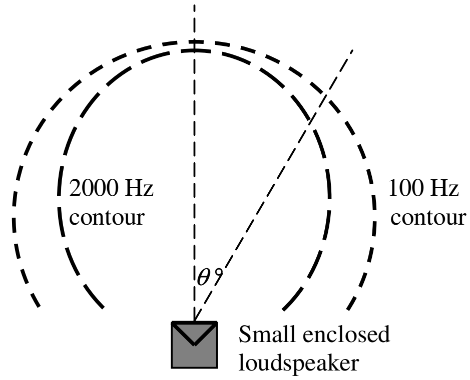


Bass drum produces low-frequency (long-wavelength) sound that diffracts more than the sound of piccolo. Thus you will hear the base drum earlier.

Example 2 Unless you want more low-frequency sound, don't place the loudspeaker in the corner of the room. Explain this in terms of diffraction and reflection of sound coming from the loudspeaker.

Low-frequency (long-wavelength) sound from the loudspeaker diffracts more due to the value of the ratio $\frac{\lambda}{w} << 1$. The diffracted sound will hit the corner walls and reflect to the listener in addition to the low-frequency sound directly from the loudspeaker.

Example 3 The two curves below represent equal intensity level (90 dB) contours of sounds (100 Hz and 2000 Hz) from a small-enclosed loudspeaker.



In terms of diffraction explain (a) the general shape of the contours and (b) the 100-Hz contour has a larger radius than that of the 2000-Hz contour.

(a) As θ increases the spread (diffraction) of sound wave decreases. $\therefore$ you need to be closer to the loudspeaker to be at the same intensity level of 90 dB.

(b) 100-Hz sound diffracts more than 2000-Hz sound. At the same spot at larger θ's the intensity level of the 100-Hz sound will be higher than that of the 2000-Hz sound. $\therefore$ the 90 dB contour for the 100-Hz sound is further from the loudspeaker than the 90 dB contour for the 2000-Hz sound as θ increases.

