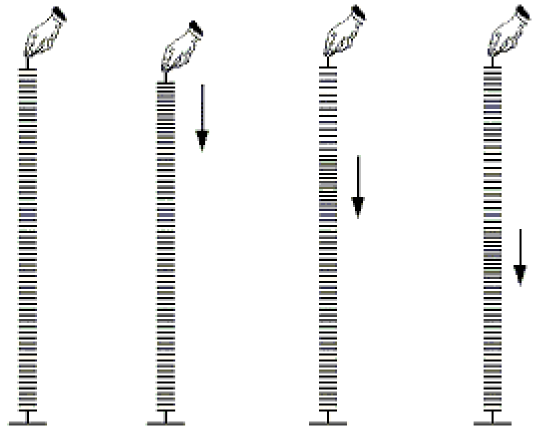
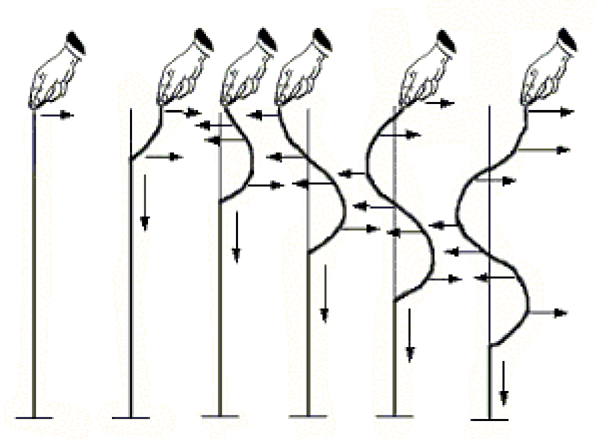
- **describe sound as the transmission of energy via longitudinal pressure waves and distinguish between sound intensity (W m^{-2}) and sound intensity level (dB)**
- **calculate sound intensity at different distances from a source using an inverse square law.**

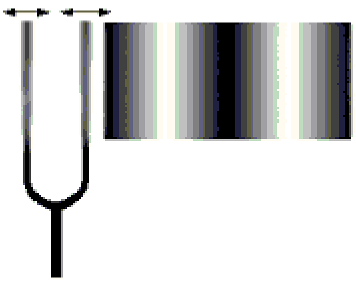
Sound

Sound is a form of energy that travels through any medium by causing parts of the medium to vibrate. Sound waves require a medium; they do not travel through a vacuum. All forms of wave motion allow the transfer of energy without the net transfer of matter. The energy travels by distorting the medium and so a measure of the wave's energy is the amount of distortion.





The sensation of sound is registered by our ears, vibrations of the air molecules adjacent to the eardrum causes tiny bones to move which is converted into an electrical signal which is interpreted as sound by the brain.

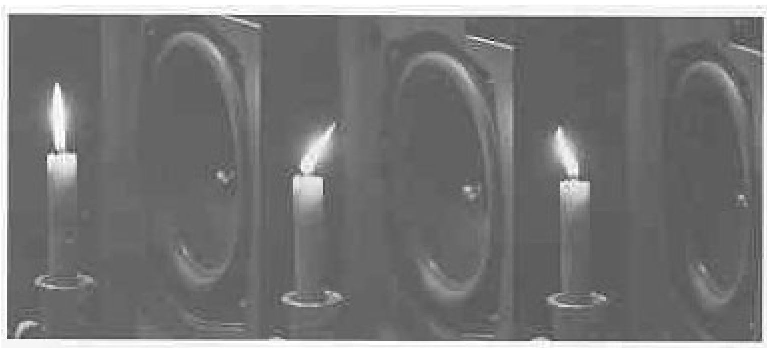


All sounds are produced by vibrations. When we speak, our vocal cords vibrate. Musical instruments produce vibrations in their strings, cavities and bodies when they are being played. A tuning fork causes the air surrounding it to vibrate.

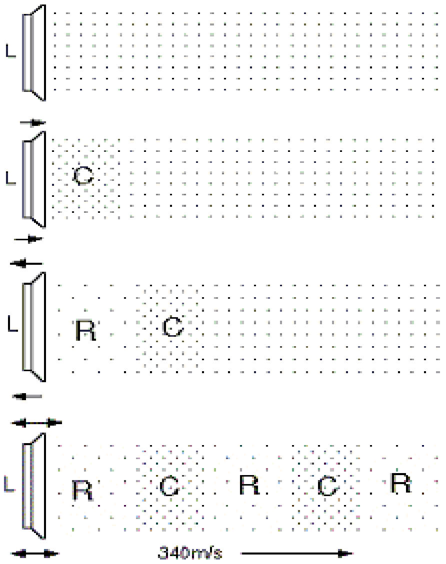
Vibrations also allow sounds to be detected, in microphones the sound vibration causes a diaphragm or crystal to vibrate slightly and produces an electrical signal that is sent to an amplifier.

Vibrations also allow sounds to be detected, in microphones the sound vibration causes a diaphragm or crystal to vibrate slightly and produces an electrical signal that is sent to an amplifier.

If we place a candle in front of a speaker, the sound from the speaker causes the flame to move backwards and forwards.



Sound is a wave, and waves are forms of energy that can travel through a medium. Sound is a longitudinal or compressive wave, this means that the medium vibrates along the same direction in which the wave is travelling. Below is an example of a loudspeaker “L” and the air particles in front of it.



When the loudspeaker is off, it does not vibrate. The air particles are evenly distributed; i.e. the pressure is constant.

When the loudspeaker is turned on, the speaker cone vibrates forwards. This causes the air particles near the cone to bunch up. A high pressure region, or compression results.

The compression moves forwards through the air. The speaker cone vibrates causing the air particles in front of the cone to spread out. A low pressure region, or rarefaction results.

As the speaker cone continues to vibrate forwards and backwards, a continuous series of compressions and rarefactions is created.

Transmission, Reflection and Absorption of Sound

Sound waves striking a solid object will be transmitted, reflected and absorbed to varying degrees. Thick materials transmit less sound than thin materials, this is why thick concrete sound barriers often line freeways passing through residential areas.

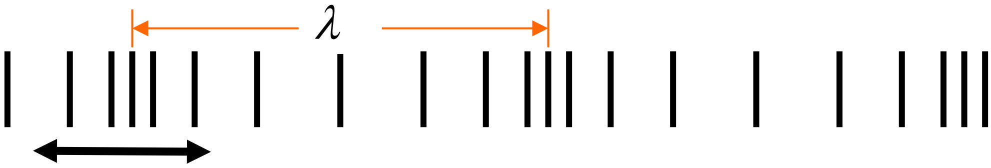
Porous materials such as carpet and curtains absorb sound. These materials reduce the amount of reflected sound and are often used in Concert Halls to limit reverberation of sound and so improve the sound quality. Classrooms that have carpet on the floor are a lot quieter than rooms with wooden floorboards. Smooth polished surfaces such as glass and tiles reflect a great deal of sound. People who sing in the shower often have an enhanced opinion of the strength of their voice due to the reflections off the glass and tiles of the shower recess.

Sound like all other waveforms, obeys the laws of reflection, the angle of incidence is equal to the angle of reflection. Angles are always measured between the direction of travel of the wave and the normal to the media boundary.

Wave types - Longitudinal waves

When the vibration of the waves is in the same direction as the line of travel, then the wave is **longitudinal**. In a longitudinal wave the motion is in the same direction as the motion of the wave, but the particles do not move forward, they vibrate around an equilibrium position.

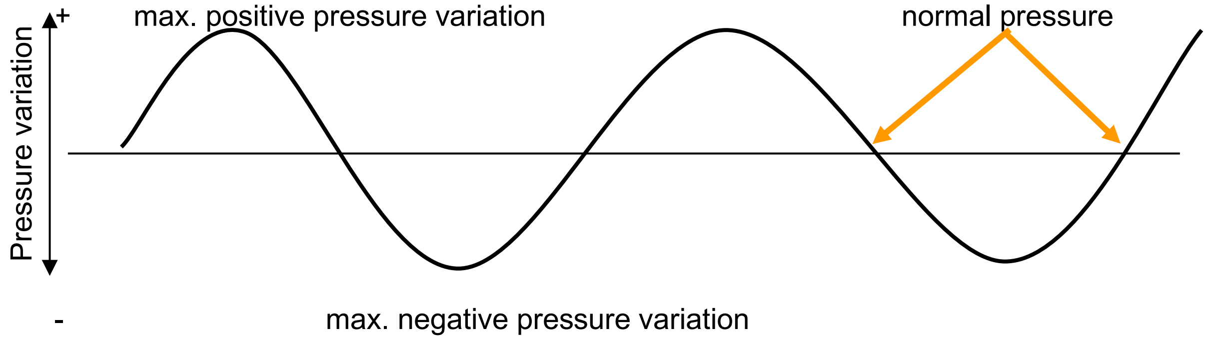
The distance between any two identical points is called the **wavelength** λ.



The movement of particle is in this direction.

compression rarefaction

It is possible to represent this particle movement as a pressure variation



Lines close together represent high pressure regions, where there is less shading there is a lower pressure region. At the two positions of maximum pressure variation (compressions and rarefactions) the molecules at these points are in their rest position. The minimum pressure variation occurs when particles are the furthest from their rest positions. The air pressure is normal, (pressure variation is zero) midway between the compressions and rarefactions. When sound travels through air it is a series of compressions and rarefactions

Frequency

The frequency of the wave determines the pitch of the sound. Frequency is a measure of how rapidly the source of the wave is vibrating. The **frequency** (f) is defined as the number of complete waves that pass a point in one second. The units for frequency are Hertz, Hz, which are cycles per second.

Wave equation

The wave equation links the velocity of the wave to the frequency and the wavelength.

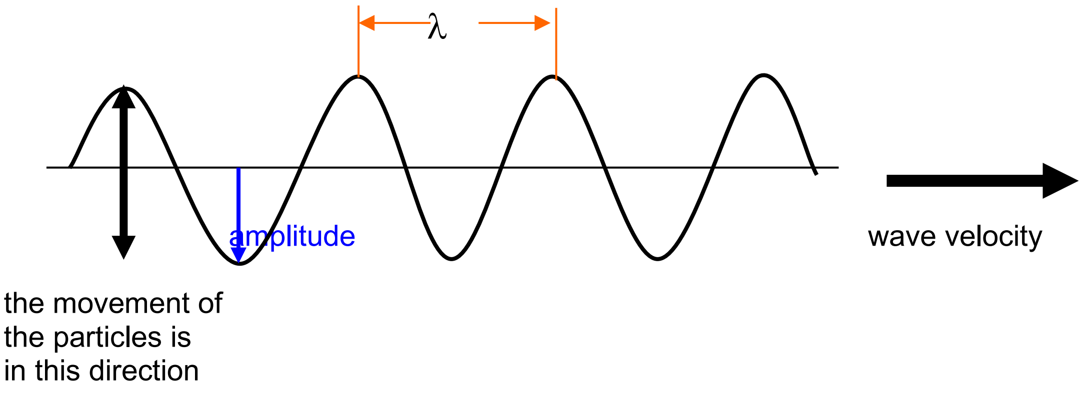
$$v = f\,\lambda$$

where v is the velocity in m s^{-1}, f is the frequency in Hz and λ is the wavelength in metres.

Transverse waves

When waves vibrate up and down in a direction perpendicular to the direction of motion of the wave, it is referred to as a **transverse** wave. e.g. water waves, where the motion of the water particles is at right angles (up and down) to the direction of the wave (forward).

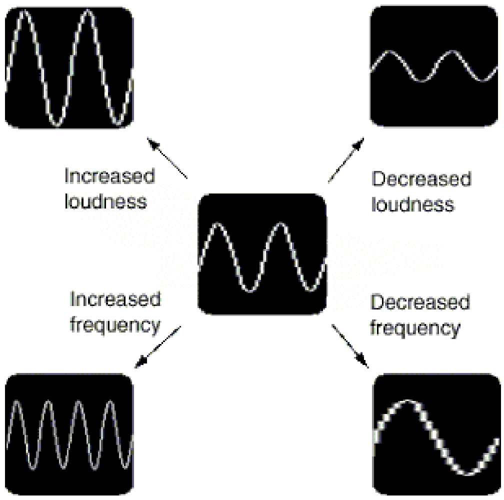
A transverse wave is modelled below



The **amplitude** of a wave is the distance from the rest position to the limit of a crest or trough; the total from crest to trough is **twice** the amplitude.

Loudness

The louder a sound is the greater the magnitude of the variation from normal pressure, this is represented in the following way. The loudness of a sound is related to the amplitude of the sound wave. A smaller amplitude wave sounds quieter than a larger amplitude sound wave.



Another way of representing this is:

Sound Intensity and Sound Level

A measure of the amount of sound energy passing through an area of one square metre each second is SOUND INTENSITY. Sound Intensity (I) is measured in Watts per square metre (W/m^2). The ear can detect intensities over a range from 1 to 10^{12}. In order to work with such a large range a logarithmic scale is used.

Even though the range of pressure changes is huge, what we sense as loudness is not proportional to intensity. The doubling of the loudness of a quiet whisper would be very noticeable, but adding the same sound to a louder source would not be noticed at all. This is because the human ear compares the loudness of two sounds in terms of the ratio of the two amplitudes rather than the difference in amplitudes.

A logarithmic scale is used to indicate sound intensity levels. The unit is the bel, or, more commonly the decibel (dB). 1 dB = 0.1 bel.

The sound loudness scale measures the ratio of the intensity of a sound wave against a standard intensity, called the "threshold of hearing" and = 10^{-12} W m^{-2}, I_0.

The sound intensity level, β (in dB) = 10 log $\frac{I}{I_0}$ where I_0 = threshold of hearing = 10^{-12} W m^{-2}.

A tenfold increase in intensity will correspond to a level increase of 10 dB, which is heard as a doubling of the loudness. A soundwave of intensity 10^{-8} W m^{-2} sounds twice as loud as a wave of intensity 10^{-9} W m^{-2}.

A one hundred fold increase will correspond to a level increase of 20 dB, which is heard as an increase of ×4 in the loudness. This is a doubling, followed by another doubling.

The human ear can only detect changes as small as 3 dB. A sound level increase of 3 dB is equal to the doubling in the sound's intensity. A 3 dB decrease will halve the sounds intensity. This means that to make the least noticeable increase in the volume of a stereo, we need to double the power supplied.

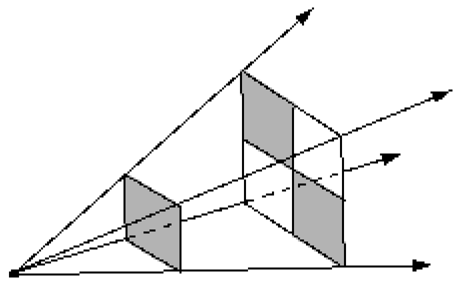
The loudness of a sound depends on how much energy is carried by the wave. The intensity of a sound wave is used to give a measurement of the rate at which the sound wave is carrying energy.

Power is energy per second, and intensity is power per unit area,

$\therefore$ **the units for Sound Intensity are W m^{-2}.**

As sound moves away from the source it spreads out over an increasingly larger area. As the energy is fixed, then the intensity must decrease.

The intensity varies $I \propto \frac{1}{r^2}$



source Intensity Intensity

I $\frac{I}{4}$

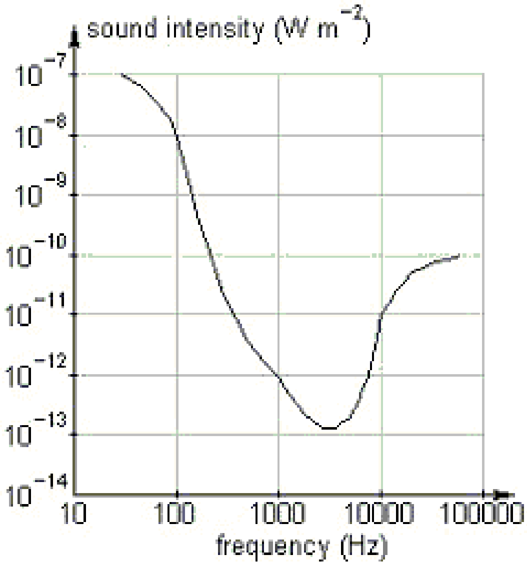
This means that if you double the distance the intensity will decrease by a factor of 4. This is known as an "Inverse Square Law".

Sound intensities and sound intensity levels of common sounds

Sound	**Intensity (W m^{-2})**	**Intensity level (dB)**
Threshold of hearing	10^{-12}	0
Rustle of leaves	10^{-11}	10
Whisper	10^{-10}	20
Quiet radio	10^{-8}	40
Normal conversation	10^{-6}	50 - 65
Busy street traffic	10^{-5}	70
Loud radio	10^{-4}	80
Noisy factory	10^{-4}	85
Motor car horn	$10^{-3} – 10^{-2}$	90 - 100
Jackhammer	10^{-2}	100
Thunder	10^{-1}	110
Rock concert	1	120
Threshold of pain	1	120
Jet aircraft	100	140

Hearing Sound intensity and level.

Human ears do not respond equally to all frequencies. Generally they respond best to frequencies around 3000 Hz. The graph shows the sound intensity that is required for a person with normal hearing to just hear each frequency. This means that the low point on the graph corresponds to the frequency that the ear is most sensitive to. Because the ear is sensitive to this frequency, it will appear louder. (Have you ever noticed that some ads on TV sound very loud. This is partly because they have a large component of around 3000 HZ, so the ear is very sensitive to them).



The curved line is a line representing all the frequencies that sound as loud as each other. So on this graph a 100 Hz sound will seem equally as loud as a 1000 Hz sound. This indicates that our ears are less sensitive to 100 Hz sounds, because we need more energy/m^2 to hear the 100 Hz sound.

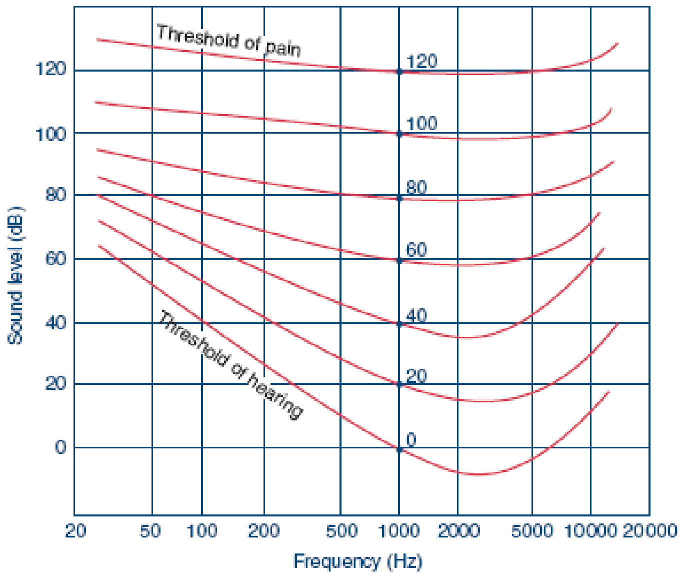
Phon

The phon is the unit of equivalent loudness. It measures how loud a sound is perceived to be compared to a reference sound – normally at 1 kHz (1000 Hz). Graphs can be drawn for an average human ear/brain combination. The phon level is read from the graph below. Hence 60 phons means “as loud as a 60 dB, 1000 Hz tone”.

To find what intensity level is required for a sound of frequency 5 kHz to have a loudness of 80phons, read along the 80-phon curve until it intersects with 5 kHz. $\therefore$ 80dB

The differences in loudness depend on the sensitivity of each individual ear, which obviously varies from person to person. What may be too loud for your parents is perfectly OK for you. The loudness also depends on the frequency. The "perceived loudness" of a sound depends on the response of a particular ear to a particular pitch.

Loudness and intensity are not the same, though they are related. Intensity is measurable, but loudness is subjective.



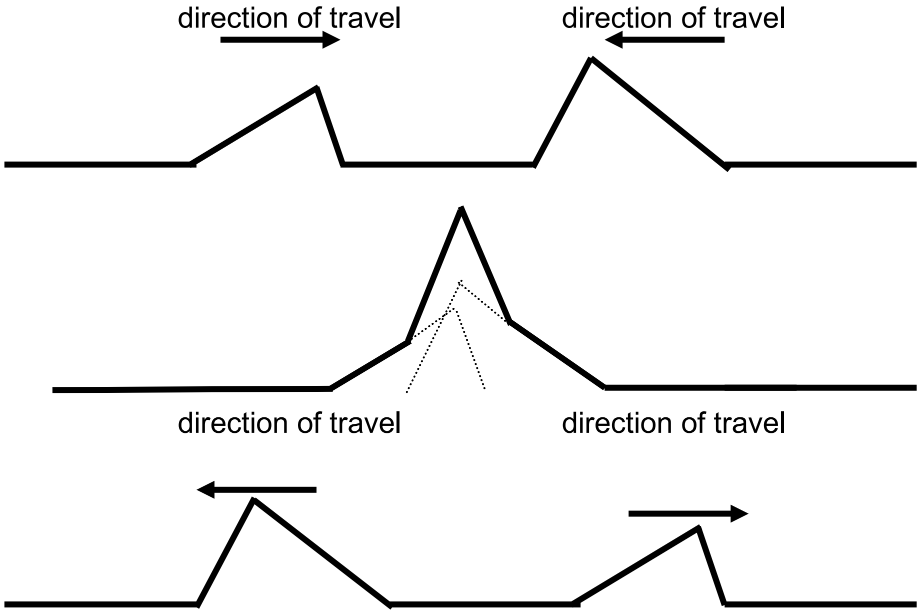
- **explain resonance and identify it as related to the natural frequency of an object, and analyse the unique sound of an instrument as a consequence of multiple resonances created by the instrument and described as timbre.**
- **analyse, for strings and open and closed resonant tubes, the fundamental and subsequent harmonics and apply this analysis to selected musical instruments.**

Superposition

The displacement of two waves combing with each other is calculated by the vector addition of the two components. The displacement of the combined pulse is the sum of the separate displacements.

The two pulses pass through each other without being altered. To find the total wave disturbance at any time, the individual displacements of each wave are added at each point.

When different sound waves pass through the same region of space, the individual waves add together to produce the resultant sound wave. This is called superposition.

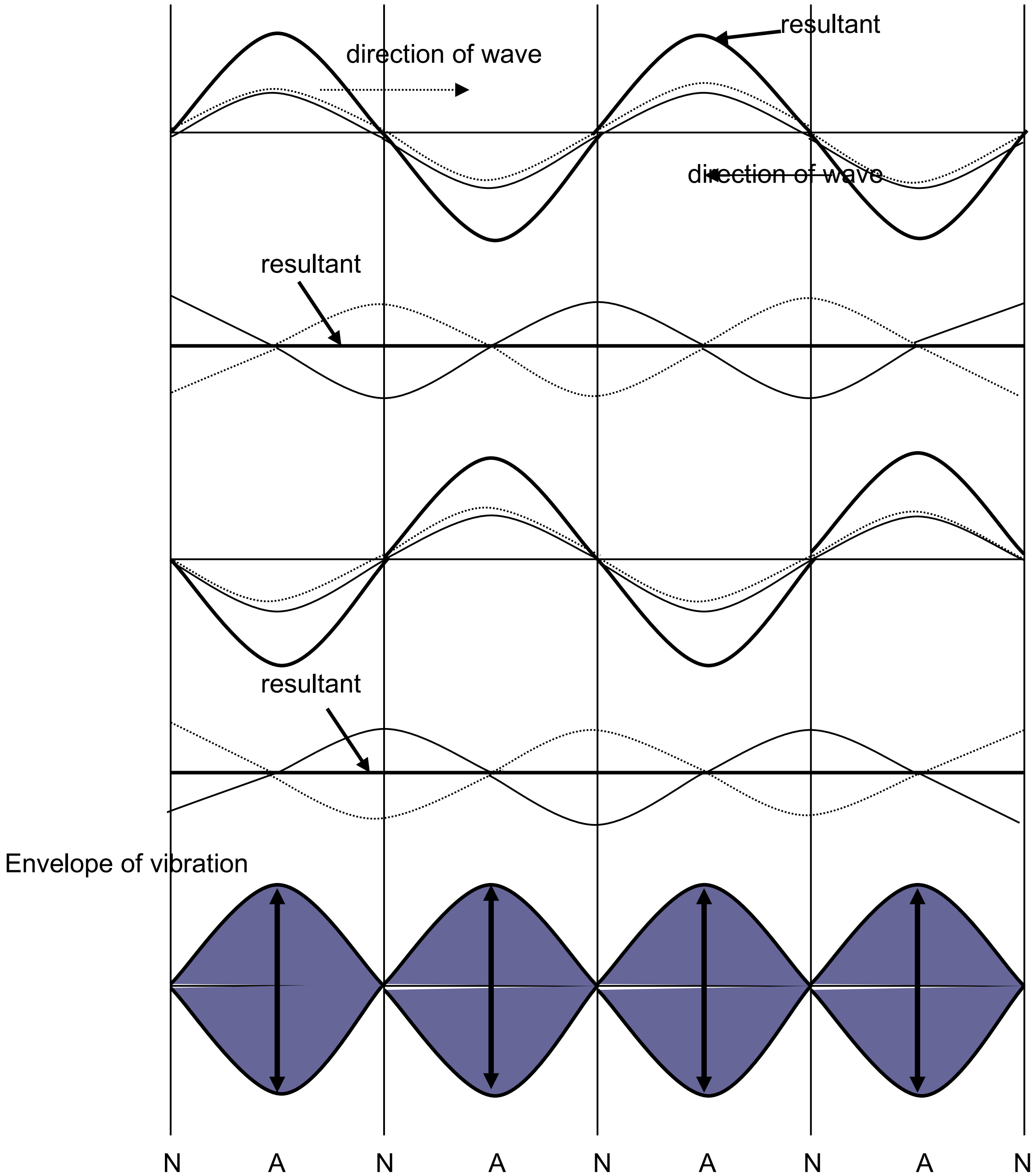


If we have two identical waves travelling in the opposite directions in one medium we get a **standing** or **stationary** wave. The superposition principle is used to obtain the waveform.

- Certain points marked N = node.
- Loops or antinodes, marked A, midway between the nodes.

The wave does not progress through the medium.

- Wavelength is the same as that of the components.
- Maximum amplitude of the resultant wave is twice that of the components.
- The distance between adjacent nodes or antinodes is $\frac{\lambda}{2}$.
- They can only be produced by the superposition of two waves of equal amplitude and frequency travelling in the opposite direction.
- They are the result of resonance and occur only at the natural frequencies of the vibration.

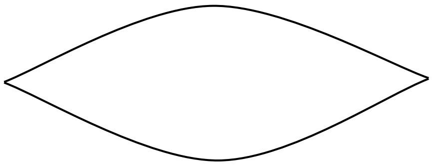


Harmonics

The standing wave frequencies are referred to as **harmonics**. The simplest mode of vibration, which has only one antinode, is called the **fundamental.** Higher level harmonics are referred to as overtones. The first overtone is the second harmonic. The fundamental and overtones form the harmonic series.

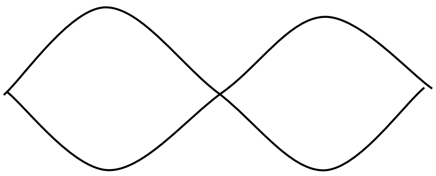
first harmonic

$\lambda_1 = 2L \qquad f_1 = \dfrac{v}{\lambda_1} = \dfrac{v}{2L}$



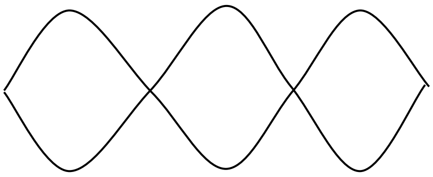
second harmonic (first overtone)

$\lambda_2 = L \qquad f_2 = \dfrac{v}{\lambda_2} = \dfrac{v}{L} = 2f_1$



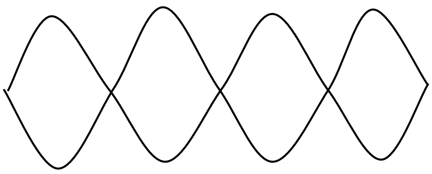
third harmonic (second overtone)

$\lambda_3 = \dfrac{2L}{3} \qquad f_3 = \dfrac{v}{\lambda_3} = \dfrac{3v}{2L} = 3f_1$



fourth harmonic (third overtone)

$\lambda_4 = \dfrac{L}{2} \qquad f_4 = \dfrac{v}{\lambda_4} = \dfrac{2v}{L} = 4f_1$



In **strings** the wavelength of the standing waves corresponding to the natural harmonics is

$$\lambda_n = \frac{2L}{n} \text{ or } f = \frac{nv}{2L}.$$

All harmonics (n = 1, 2, 3, …) may be present, and the ratio of frequencies $f_1 : f_2 : f_3 = 1 : 2 : 3$.

Wind instruments and air columns

Many instruments produce sound from the vibrations of standing waves in a column of air within the tube or pipes of the instrument eg. woodwinds, brasses and the pipe organ. Amplification of the sound comes through resonance within the air column. Sometimes a vibrating reed or the performer's lips set up the initial vibration in the air column. In the flute and organ, air is directed over an opening, creating turbulence to set up the vibration.

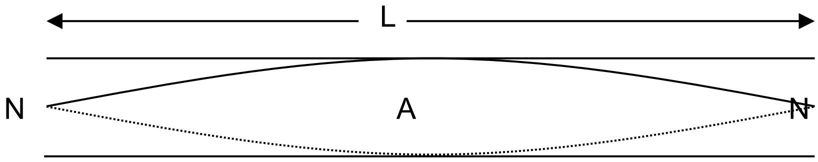
The air in the tube vibrates with a variety of frequencies but only certain frequencies persist. These correspond to the standing waves established in the tube. The lowest frequency standing wave is called the fundamental.

Pressure variation (Open at both ends)

First harmonic (fundamental frequency)

$\lambda_1 = 2L$

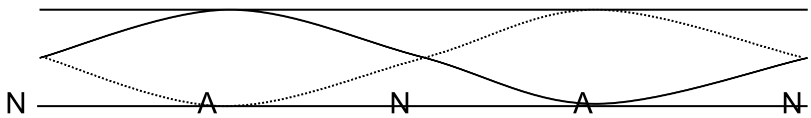
$f_1 = \dfrac{v}{2L}$



Second harmonic (first overtone)

$\lambda_2 = \dfrac{2L}{2} = L$

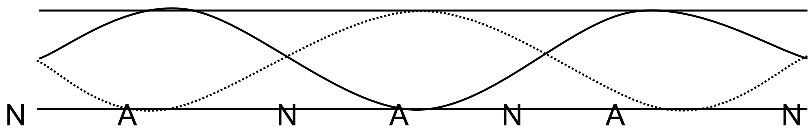
$f_2 = \dfrac{2v}{2L} = 2f_1$



third harmonic (second overtone)

$\lambda_3 = \dfrac{2L}{3}$

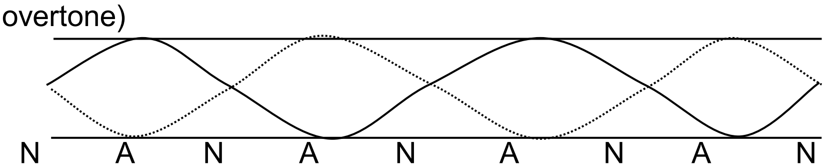
$f_3 = \dfrac{3v}{2L} = 3f_1$



fourth harmonic (third overtone)

$\lambda_4 = \dfrac{2L}{4} = \dfrac{L}{2}$

$f_4 = \dfrac{4v}{2L} = 4f_1$



It can be shown that the fundamental frequency of a stretched wire depends on 3 things: length, tension and mass per unit length. Combining these gives a single formula from which the fundamental frequency of a stretched wire can be found:

$f = \dfrac{1}{2L}\sqrt{\dfrac{T}{m}}$ where: f = fundamental frequency in Hertz (Hz), L = length in metre (m), T = tension in Newton (N), and m = mass per unit length in kg m^{-1}.

Often T and m are constant so $f_2L_2 = f_1L_1$ and when L and m are constant $\dfrac{f_2}{\sqrt{T_2}} = \dfrac{f_1}{\sqrt{T_1}}$

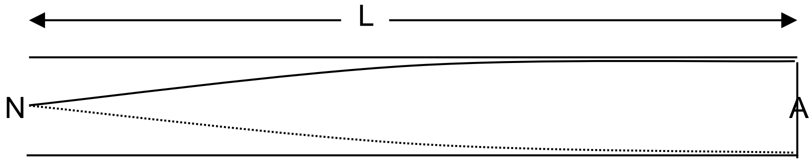
If we combine $v = f\lambda$ and $\lambda = \dfrac{2L}{n}$ we can get $v = \sqrt{\dfrac{T}{m}}$ where v is the velocity of propagation (m s^{-1}).

Pressure variation (Closed at one end)

First harmonic (fundamental frequency)

$\lambda_1 = 4L$

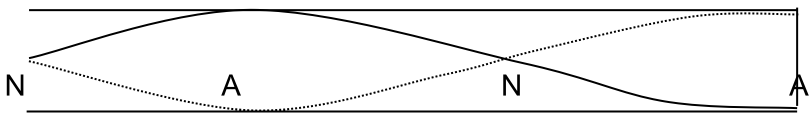
$f_1 = \frac{v}{4L}$



third harmonic (first overtone)

$\lambda_2 = \frac{4L}{3}$

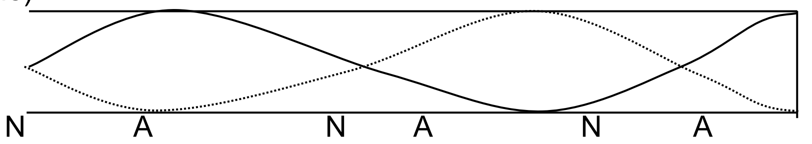
$f_2 = \frac{3v}{4L} = 3f_1$



fifth harmonic (second overtone)

$\lambda_3 = \frac{4L}{5}$

$f_3 = \frac{5v}{4L} = 5f_1$



The term *overtone* is applied to harmonics other than the fundamental frequency.

Summary

The main features of standing waves in air columns can be summarised

- at open ends of the tube there is always a node
- at closed ends there is always an antinode
- the wavelength of the sound must fit the length of the pipe. The length of the air column that vibrates is slightly longer than the length of the pipe, since nodes form just outside the end of the pipe
- the fundamental vibration in a closed pipe has a wavelength twice as long as the fundamental in an open pipe of the same length. This makes the frequency of the sound produced by a closed pipe half that of the same length of an open pipe, so it is an octave lower.

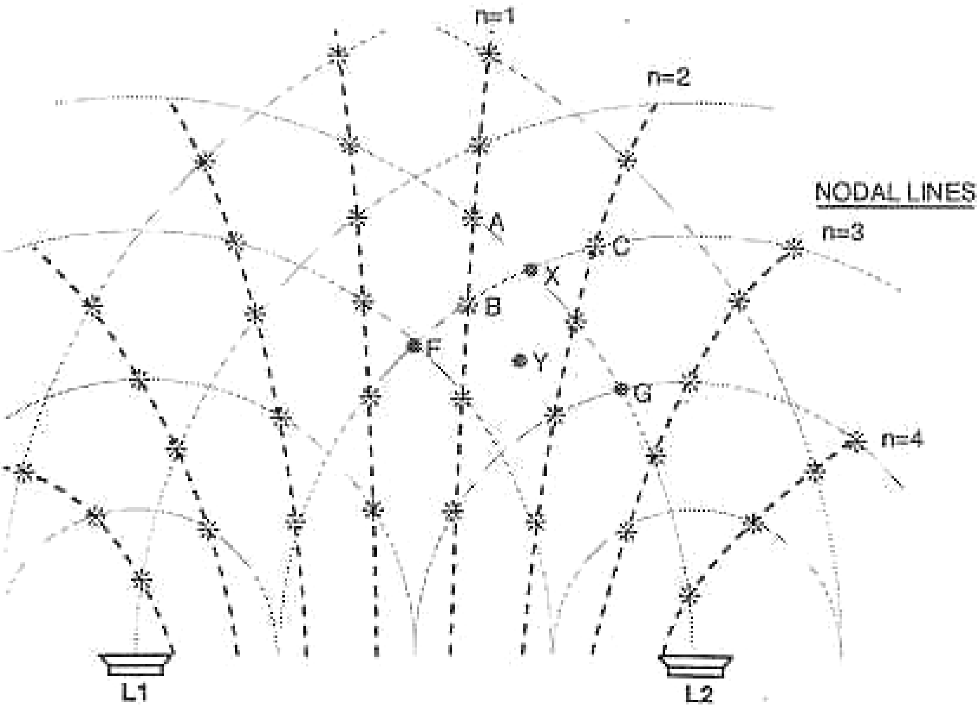
Resonance

Blowing air across the mouth of a bottle causes the air inside to vibrate and a sound is produced. Resonance occurs when a forcing frequency, the same as the natural frequency, is applied.
Eg. The oscillations of a person on a swing have a certain natural frequency. The amplitude of these oscillations will decrease unless the lost energy is replaced. To keep the swing moving it is best to be pushed at exactly the right times. The frequency of the push must be the same as the natural frequency to get the best response. This is called resonance.

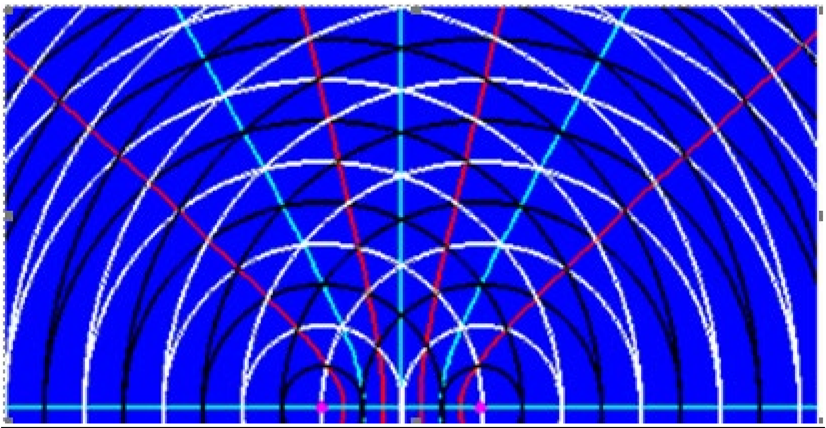
The high frequency sound shatters the glass.

Interference

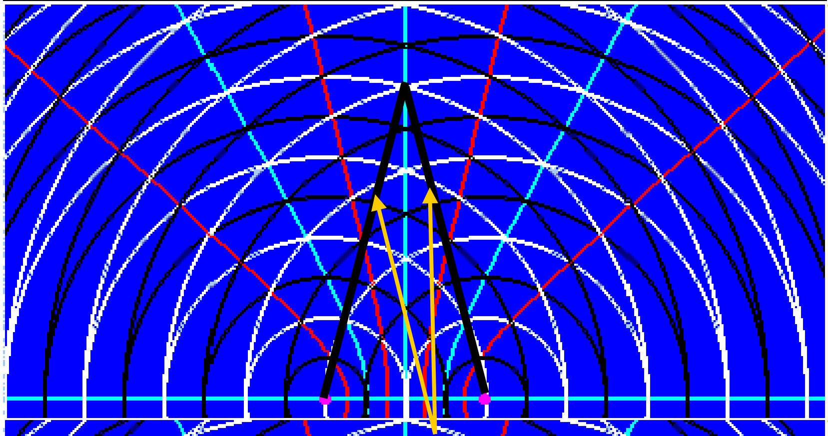
When sound waves pass through each other, they can add together so that the reinforce each other; or they can cancel each other out. When the waves add together this is called **constructive interference**, and will lead to a louder sound. When the cancel each, this is called **destructive interference** and this leads to a quieter sound.



TWO DIMENSIONAL REPRESENTATIONS

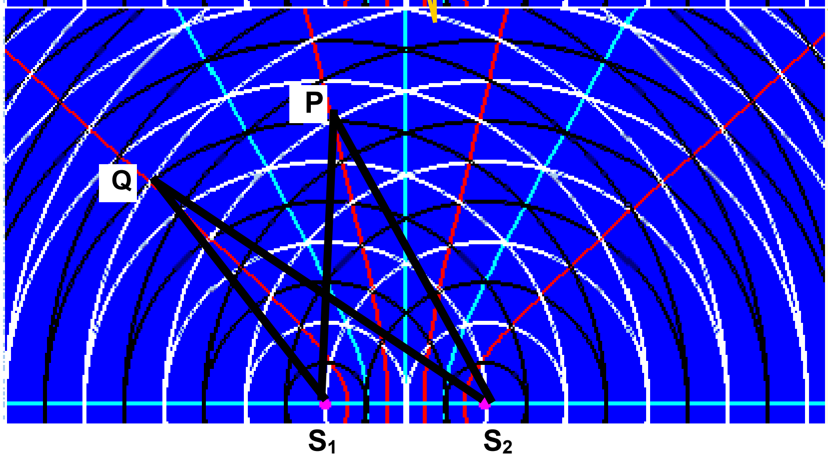


This is a diagram of two point sources in phase. The dark lines represent troughs and the light lines represent crests, coming from the sources. The lighter line in the central position is the central antinode, because everywhere along this line, there are two waves in phase. The 'red' lines are the nodal lines. It is along these that the path difference between the two sources is ½ λ



The path difference between these two lines is zero. i.e. they are the same length, both are 4λ

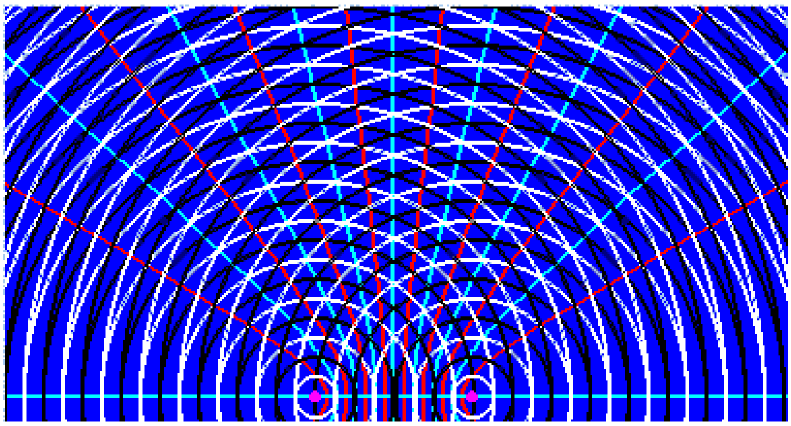
The distance from S_1 to P is 3.5λ while the distance from S_2 to P is 4λ. So the difference between these two lengths, called the path difference is 0.5λ. Everywhere along the first nodal line (on either side of the central maximum) the path difference will always be 0.5λ.



If we consider the point **Q,** the path difference is

$S_2Q - S_1Q = 5\lambda - 3.5\lambda$
$= 1.5\lambda.$

Effect of wavelength



When the sources have a shorter wavelength they produce more nodal lines.

For any **nodal line** the path difference is summarise by

P.D. = (n - ½)λ

where n = 1, 2, 3….

For the **anti-nodal lines** the path difference

P.D. = nλ

where n= 0, 1, 2…..

- **investigate factors that influence natural frequency including shape and material and explain how this relates to instruments.**
- **investigate and explain a variety of musical instruments with reference to the similarities and differences of sound production between instrument types (brass, string, woodwind and percussion) and how they compare with the human voice.**

Timbre (or Quality)

A note played on a piano can be easily distinguished from notes of the same pitch played on the organ, the violin, the flute, etc. The attribute which enables us to distinguish between notes of the same pitch but from different sources is termed quality or **timbre**.

When a note is sounded on a particular instrument the practised ear can perceive that the sound is really a number of superimposed tones. The constituent tones into which a given musical note can be analysed are called partial tones. The partial tone of lowest frequency, the one which determines the pitch of the composite note, is known as the first partial or the **fundamental**. The other partial tones are known collectively as **overtones**. The term harmonic is given to any partial whose frequency is a whole number multiple of that of the fundamental.

The fundamental is the first harmonic and if its frequency is f, then the second, third fourth harmonics etc. are those partials of frequencies $2f$, $3f$, $4f$, and so on. It should be noted, however, that overtones are not always harmonics: for example, such frequency ratios as 1.0: 1.5: 2.7: 3.3 or 1.00: 1.65: 2.10: 3.00: 3.54: 4.97: 5.33 have been found for the partial tones of the notes emitted from particular bells. Other sources which emit notes with inharmonic partials are xylophones and kettle drums.

The timbre of a tone is determined by the relative intensities and frequencies of its partial tones. The timbre of a sound of given pitch from a particular instrument can be varied by changing the method of excitation. For instance, a given violin string, when plucked, emits a note of quite different quality from that obtained when it is bowed.

The origin of partial tones

The partial tones of a musical note arise from the fact that sound sources such as strings, air columns and plates have a number of possible modes of vibration. For instance, a string can vibrate in multiple harmonic modes at the one time. Each partial tone in the sound emitted by a source is due to a particular mode of vibration of that source.

Timbre of notes from Strings

The timbre of the note from a stretched string is found to depend on (i) the method of excitation, (ii) the point of excitation, and (iii) the modes of vibrations of the associated vibrators.

(i) *Method of excitation.* The fact that the method of excitation affects the timbre of the emitted note is obvious even to the untrained ear. The difference in quality between the note from a piano string (excited by striking), a violin string (excited by bowing) and that from a harp string (excited by plucking) is easy to detect.

(ii) *Point of excitation.* The timbre of the note from a stretched string is found to alter when the point at which the vibration is excited is changed. For example, if a string is plucked at its midpoint, then any vibratory mode requiring the midpoint to be a node must be absent, i.e., the even numbered harmonic partials will be absent from the emitted sound. If, on the other hand, the string is plucked at one-quarter of its length, then the fourth, eight, twelve, etc., harmonic partials will be absent, since the corresponding vibratory modes require nodes at the point of excitation. This difference in quality is quite noticeable.

In a piano the hammer strikes the string at about one-seventh of its length, and this considerably weakens the seventh, ninth and eleventh partials which, if strongly present, produce a displeasing quality when chords are played.

(iii) *Effect of associated vibrators*. The third factor determining the timbre of the note emitted by a stretched string is the forced vibrations of associated vibrators: for example, the quality of a violin note is determined, in part, by the vibratory modes of the belly and the air in it, whilst the sounding board of a piano, by executing forced vibrations, makes its contribution to the quality of the notes from the instrument.

Musical instruments

Conventional musical instruments can be grouped into one of three classes: those in which the sound is produced by vibrating strings, those in which the sound is produced by vibrating air columns, and those in which the sound is produced by *percussion* – the vibrating of a two-dimensional surface.

Musical instruments consist broadly of three main parts:

(i) the principal vibrator;

(ii) the exciter which starts and, if necessary, maintains the vibration;

(iii) the associated vibrators which reinforce the intensity and sometimes modify the quality of the notes emitted by the principal vibrator.

In a stringed instrument the principal vibrator is a stretched string or wire and is excited by striking (piano), bowing (violin), or plucking (guitar).

In some stringed instruments only a few strings are available (e.g., the violin or guitar) and the pitch of the note from a given string is varied by altering its vibrating length by pressing at the appropriate point. In the piano and the harp, there is a separate string (or group of strings) for each note. The lengths of the strings progressively increase from the short ones for the high pitch notes to the long ones for the low pitch ones. The strings vary in mass per unit length and the tension of each is adjusted until its note is of correct pitch. In the piano all strings are steel wires of suitable gauges; the bass strings are loaded by additional wire wound spirally around them to obtain the necessary mas per unit length.

The direct sound from a vibrating wire is of weak intensity and it must therefore be reinforced by some associated vibration. In the piano one of the two bridges which determine the vibrating length of a wire, is attached to a large flat board, termed the sounding board, which is therefore set in vibration, but in the violin, it is the belly of the instrument and its enclosed air which reinforces the sound from the string. If any of the free vibratory modes of the belly or its associated volume of air have frequencies approximating to those of the fundamental or partials of the notes emitted by the strings, then those frequencies are selectively reinforced by resonance. It is these resonances that give each instrument its characteristic quality or timbre. Any group of frequencies which is thus selectively strengthened for a particular instrument is called the formant of that instrument.

For wind instruments the principal vibrator is the air column but two classes can be distinguished. One type is exemplified by the pipe organ in which various notes come from a series of pipes of suitable lengths (compare with piano); each pipe is separately excited. The factors affecting the quality of the notes is whether the pipe is stopped or open, i.e., whether the open end is closed or open to the air. The vibratory modes of a pipe open at both ends form a complete harmonic series. And therefore, such a pipe can emit all partial tones corresponding to this complete series of modes. In contrast, a pipe closed at one end can emit only the odd numbered harmonic partials and for this reason the quality of the note from a stopped (closed) pipe is different from one of the same pitch from an open pipe.

Other factors which affect the relative intensities of the partial tones and therefore determine the quality of the notes from a flue type of organ pipe are (i) the material in the pipe; (ii) the wind pressure used to operate it; high pressure is associated with pronounced upper partials; (iii) the lateral dimensions and shape of the pipe; (iv) the height from the slit to the sharp edge of the upper lip; (v) the width of the mouth.

The other type of wind instrument includes instruments like the flute, the trombone, the trumpet, the cornet. In each of these the principal vibrator is a single air column, which can be either straight (flute) or coiled (trombone). The desired frequencies are derived by varying the length of the column. In the flute this variation is obtained by opening and closing holes drilled at suitable intervals along the pipe. The exciter is an edge tone generated by blowing over the bowing hole. The oboe, the clarinet, and the saxophone also have straight pipes whose effective length is varied by holes in their tubes, but they differ from the flute in that the exciter is a reed which is set in forced vibration by blowing over it.

For brass instruments, like the cornet, trumpet or trombone, the exciter is the lips of the blower. The two lips, which are inserted into the mouthpiece of the instrument, function like a double reed system. In these instruments the principal vibrator is an air column whose effective length can be varied either by the movement of a sliding piece, as in the trombone, or by inserting additional lengths of tube as in the trumpet and cornet. The quality of the notes from brass instruments is determined partly by the shape of the main tube and partly by the shape of its open end.

In brass instruments such as trumpets, French horns, and trombones, vibrations of the players lips interact with standing waves that are set up by the acoustic energy reflected within the instrument by the flared bell. The lengths of the vibrating air columns are manipulated by pushing valves that add or subtract extra segments or by extending the tube. In woodwinds such as clarinets, oboes, and saxophones, a stream of air produced by the musician sets a reed vibrating, whereas in fifes, flutes, and piccolos, the musician blows air against the edge of a hole to produce a fluttering stream of air that sets the air column into vibration.

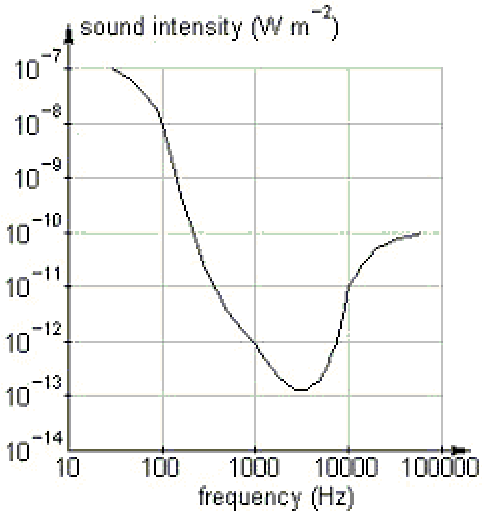
In a stringed instrument, the vibration of the strings is transferred to a sounding board and then to the air, but with low efficiency. To compensate for this, we find relatively large string sections in orchestras. A smaller number of high efficiency wind instruments sufficiently balances a much larger number of violins.

In percussion instruments such as drums and cymbals, a two-dimensional membrane or elastic surface is struck to produce sound. The fundamental tone depends on the geometry, the elasticity and, in some case, the tension of the surface. Changes in pitch result from changing the tension in the vibrating surface; depressing the edge of a drum membrane with the hand is one way of accomplishing this. Different modes of vibration can be set up by striking the surface in different places. In the kettle drum, the shape of the kettle changes the frequency of the drum. As in all musical sounds, the quality depends on the number and relative loudness of the partial tones.

Human hearing

Human ears do not respond equally to all frequencies. Generally they respond best to frequencies around 3000 Hz. The graph shows the sound intensity that is required for a person with normal hearing to just hear each frequency. This means that the low point on the graph corresponds to the frequency that the ear is most sensitive to. Because the ear is sensitive to this frequency, it will appear louder. (Have you ever noticed that some ads on TV are very loud? This is partly because they have a large component of around 3000 HZ, so the ear is very sensitive to them).

The curved line is a line representing all the frequencies that sound as loud as each other. So on this graph, a 100 Hz sound will seem equally as loud as a 1000 Hz sound. This indicates that our ears are less sensitive to 100 Hz sounds, because we need more energy/m^2 to hear the 100 Hz sound.



The differences in loudness depend on the sensitivity of each individual ear, which obviously varies from person to person. What may be too loud for your parents is perfectly OK for you. The loudness also depends on the frequency. The "perceived loudness" of a sound depends on the response of a particular ear to a particular pitch. Loudness and intensity are not the same, though they are related. Intensity is measurable, but loudness is subjective.

Audibility range for the human ear