CHAPTER 16

Sound, Music and Audio Technology

16.1 INTRODUCTION

What is sound? One frequently-played TV advertisement for a popular drink showed people whistling and causing glasses and bottles to shatter. A measure of greatness of singers is that they have the ability to shatter crystal glasses by reaching a certain high-pitched note. The renowned opera singer Maria Callas (1923–77) was reputed to be able to do this.

- Why does this happen? Does it happen or is it one of those exaggerated myths?

After completing this chapter you will be able to answer this question and others such as these:

- Can you hear space ships explode in space?
- How is sound used by dentists, doctors, bats, the blind, and fishermen?
- Why do you hear a siren differently as a police car comes toward you and goes away? How does an understanding of this allow you to measure the speed of a cricket ball?
- Why can you hear so well in the 'bush' at night?
- Do you know how insects can hear bats? Can any animal 'jam' a bat's sonar?

16.2 WHAT IS SOUND AND HOW IS IT PRODUCED?

In Chapter 13 we suggested that if a tuning fork is tapped and held beside another one of the same frequency, the second fork also begins to vibrate. Sound waves also cause our ear drums to vibrate and microphones to produce small alternating voltages.

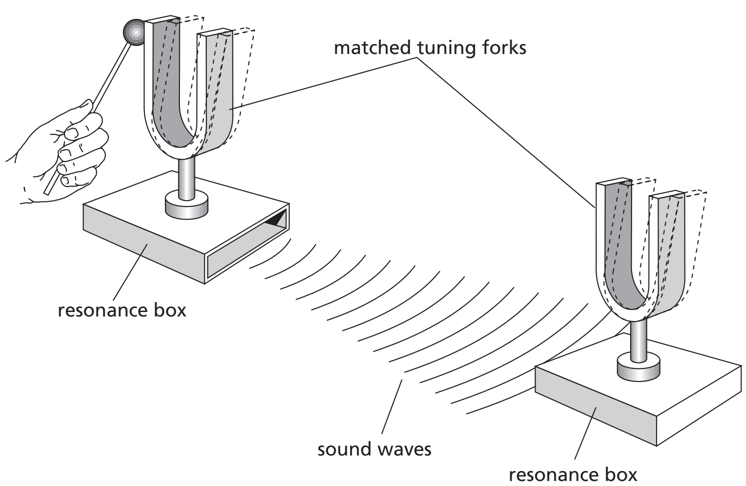


Figure 16.1
One vibrating tuning fork will cause another close by to vibrate.

Sound is a form of energy — one that travels from the source to the receiver by means of waves. In Chapter 13 it was indicated that sound waves were longitudinal mechanical waves — waves that require a medium for transmission.

Activity 16.1 SOUND ENERGY

Research an area that supports the proposition that sound is a form of energy, and be prepared to explain your evidence to the class.

Activity 16.2 SOUND IN A VACUUM

If your school possesses a vacuum pump, the following demonstration should indicate that sound requires a medium for its propagation.

1 Place a ringing electric bell in a bell jar and extract the air from the jar using the vacuum pump.
2 What do you notice as the amount of air in the jar is reduced?
3 Predict what you would notice if the air pressure was increased.

Activity 16.3 SOUND VIBRATIONS

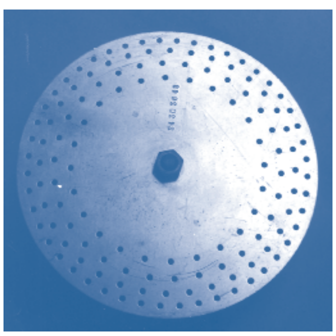
Part A

1 Place a metal ruler on a desk with about three-quarters of its length overhanging the edge of the desk. Pull the overhanging end down and let it vibrate.
2 What happens? What do you hear?
3 Place your ear on the desk and you will hear the sound with astonishing clarity.
4 What do you notice when you rest the bone behind your ear on the desk?
5 As the ruler is vibrating move it in so less overhangs. What do you hear now?
6 What happens if you use a softer surface?

Part B

It is not easy to see a tuning fork vibrating but it can be felt. Touch the stem to your lips, your teeth, your head. Touch the prongs on to the surface of water. What do you notice?

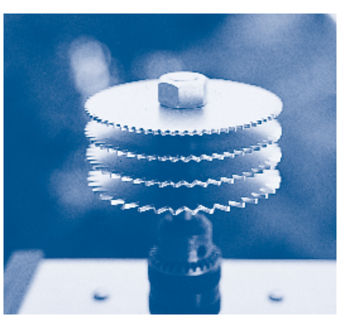
Photo 16.1
Siren disc.



It can easily be seen that the vibrations cause the sound. Music students in the class may be able to tell you why guitars, flutes or trombones produce sound. Sound is produced by something vibrating — a string, a reed or an air column. The different sounds are produced by the different frequencies of vibrations. This can be shown by tapping two tuning forks of different frequencies. They produce different sounds due to the different rates of vibration of the arms of the forks.

We speak and hear due to the vibration of membranes. The vocal cords in our throats vibrate at different rates to produce sounds. The tension of the vocal cords controls the rate of vibrations. The energy carried by the sound waves causes the ear drum to vibrate. These vibrations are transposed into discernible noises by the brain. When you get a throat infection your vocal cords become inflamed and swollen. This gives you a husky voice.

Photo 16.2
Savart's disc.



Sound is a form of energy produced by the vibrations of objects and carried by longitudinal mechanical waves.

A siren disc (see Photo 16.1) has concentric circles of holes in an aluminium disk. When it is spun at high speed and air blown through the holes, the chopped jet of air produces a sound wave.

Savart's toothed wheel can also be used to produce 'musical' sounds. By holding a piece of card against the spinning toothed wheel, different frequencies can be produced. The number of teeth per wheel are in a set ratio and, when sounded at the right speed, produce what music students call the notes of the major triad: C_4, E_4, G_4, and C_5.

NEI Activity 16.4 WAVE MOTION

1. Mount a tiny mirror on one prong of a tuning fork held upright in a stand. Shine a laser beam onto the mirror so that it reflects onto a wall. Set the tuning fork vibrating and slowly rotate it back and forth.
2. What pattern appears on the wall?

16.3 PROPAGATION OF SOUND WAVES

Recall from Chapter 13 that sound waves are longitudinal waves producing compressions and rarefactions of the air particles in the direction the wave propagates. Therefore without air (in a vacuum) no sound can propagate. So do not believe those space movies in which spaceships explode with a large noise.

- Would you be able to hear the explosion?
- Would you be able to see it?

Then how does a Space Shuttle communicate with Earth? This is done by means of radio waves — a form of electromagnetic waves that do not require a medium for their propagation. Therefore sound can propagate in all media that have particles — air, water, wood and the ground. You may have seen Indian scouts in old Westerns fall to their knees and press their ears to the ground to detect distant and unseen riders. They relied on the fact that sound travels through the ground very well and doesn't get scattered as it does in the air.

NOVEL CHALLENGE

If you put your head underwater while having a bath, you can hear sounds from all over the house that you wouldn't normally hear. Why is that?

Speed of sound

The more rigid the particles in a medium, the faster the sound will travel through it. This is shown in Table 16.1.

Table 16.1 THE SPEED OF SOUND IN VARIOUS MEDIUMS

MEDIUM	SPEED OF SOUND (m s^{-1})
Air	342
Water	1410
Copper	3560
Aluminium	5100

In air, where the particles are loosely connected, the speed of sound is approximately 340 m s^{-1}. However, this varies with the atmospheric conditions such as temperature, humidity, and air movement. As air temperature rises, the speed of sound increases by about 0.6 m s^{-1} for each degree. At 0°C sound travels at 331 m s^{-1}, whereas at 20°C it travels at $331 + 20 \times 0.6 = 343$ m s^{-1}.

As an orchestra warms up, the pitch of wind instruments becomes higher because the speed of sound in air increases and this affects the frequency of the standing waves inside the instruments. String instruments, on the other hand, go lower in pitch because the friction of the fingers rubbing over the strings heats them up and causes them to lengthen.

In other gases, the speed is different, as Table 16.1 shows. You may have seen people take a lung-full of helium gas from party balloons and when they speak they sound like Donald Duck. The speed of sound in helium is about 965 m s^{-1}, which causes the resonant frequency of the throat and mouth cavity to rise.

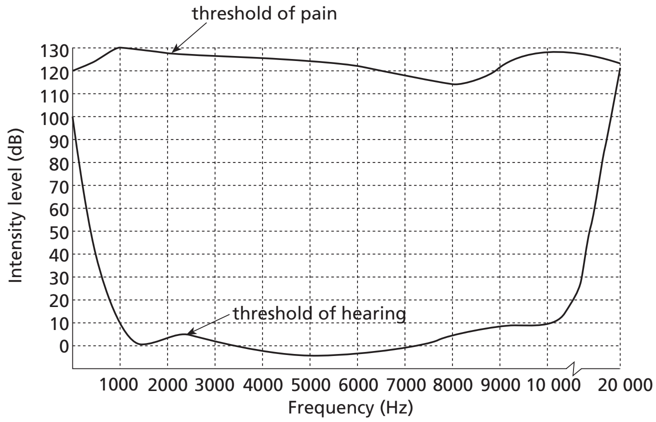
NOVEL CHALLENGE

In the *Song of the White Horse* by David Belford, the lead soprano is required to breathe in helium to reach the extremely high top note.
Question: if you released some of the helium into the middle of the orchestra, what would happen to the pitch of the following instruments: wind, brass, strings, percussion?

Range of hearing

The frequency and therefore wavelength of sound waves is controlled by the frequency of the vibrating source. The human ear can detect frequencies from 20 Hz to 20 000 Hz; however, the ear does not respond in the same way to all frequencies. The ear is most sensitive to frequencies of about 3000 Hz (Figure 16.2). Age also affects our hearing. As people age, they find high frequencies more difficult to hear. At about 65 years old, the highest frequency heard is about 5000 Hz. Human speech produces frequencies from 600 Hz to 4800 Hz.

Figure 16.2
The graph shows that the human ear does not respond equally to all frequencies. Some frequencies need to be more intense to be heard.



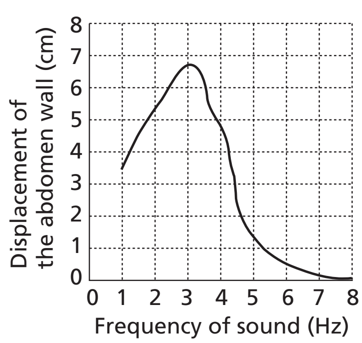
NOVEL CHALLENGE

The air bladder of a fish serves another purpose besides buoyancy: it enhances the fish's hearing. How does it do that?

Infrasound

Our ears are not the only detectors of sound. Other body organs can detect sound waves, especially vibrations of low frequencies. For example, the intestines and stomach are susceptible to low vibrations, with a maximum response at about 3 Hz. Your stomach wall actually moves in and out in response to such vibrations (Figure 16.3). This property is used in cinemas to produce an effect called 'sensurround' in which low frequencies are generated by banks of large 'woofer' speakers to make the effects of bomb blasts and earthquakes more realistic as you feel the effect as well as hear it. Physicists call such low frequencies **infrasound** (Latin *infra* = 'below') and while it may be safe under controlled conditions, infrasound can also cause nausea and dizziness, such as in car sickness. Death can occur in extreme cases when internal organs rub against each other and haemorrhage (rupture). Enjoy your movie.

Figure 16.3
The abdomen responds to low frequencies. The abdominal wall can move an incredible 6.5 cm in response to frequencies of about 3 Hz.



NOVEL CHALLENGE

The frequencies of insects' wing beats has been measured: butterfly 12 Hz, bumblebee 130 Hz, honeybee 225 Hz, mosquito 600 Hz. Which can you hear, and why? Spiders know when an insect has been caught in their web by the vibrations emitted. Spiders will run to a 256 Hz tuning fork held in their web, but not if you just stick your finger in it. Don't you want to have a go?

Questions

1. Sound travels through media containing particles. Explain how this occurs.
2. People can not only hear sound travelling in materials, they can also feel the sound vibrations. Apart from the above situations can you think where this is used?
3. Describe what is meant by the terms 'compression', and 'rarefaction'.
4. Use Figure 16.3 to determine what range of frequencies is most suitable for 'sensurround' speakers to produce.

16.4 PROPERTIES OF SOUND WAVES

As sound is a wave, those properties common to all waves apply to sound. Sound waves can be reflected, refracted, diffracted and can interfere. The wave equation $\boldsymbol{v} = f\lambda$ that you have used for water waves and light waves also applies to sound waves.

The following are a few examples of where properties of sound waves are experienced every day. You can probably think of many more.

Reflection

When you hear an echo you are hearing the reflected sound from a distant mountain, cliff or wall. This is very noticeable when you shout in an empty room as there are no materials to absorb the wave energy. Careful design and placement of curtains and furniture is essential in concert halls to absorb sound and stop reflected waves interfering with sounds produced by the artists. Theatre design and construction utilises computer simulations to show where reflections will interfere and where absorbing materials are essential. People's bodies also assist in absorbing sounds. Modern theatre acoustics now include designs that allow unoccupied seats to retain the same sound-absorbing qualities as human beings, so that the sound reinforcement is similar whether the theatre is full or empty.

Reflection of sound can be used to measure the speed of sound.

EI Activity 16.5 THE SPEED OF SOUND BY REFLECTION

Clap your hands hard (or hit a piece of wood with a mallet) at a distance from a wall in an open area. When you hear the echo clap your hands again. Continue to do this until 10 echoes are heard. The speed of sound can be calculated by dividing the total distance the sound covered in the 10 'trips' by the time taken from the first clap till the last echo.

1. How closely did your measurement match the stated value for the speed of sound?
2. Discuss the sources of error in your measurement, and how the experiment could be improved.

NOVEL CHALLENGE

Thunder from lightning refracts upwards, and at distances greater than 24 km you can't hear it. Draw a diagram to illustrate this phenomenon.

Refraction

Have you ever wondered why it is easier to hear sound at night than in the day? It is particularly noticeable in open spaces, where fewer reflections occur. This is due to the refraction of sound, which results from the fact that the speed of sound changes with temperature — sound travels faster in warmer air. During the day the air directly above the ground is warm whereas higher up it is cooler. Therefore sound waves, instead of propagating parallel to the ground, are refracted upward. This refraction will not be an abrupt change but a gradual change, as there is no distinct boundary between the warm and the cool air. At night the ground is cooler than the air and the sound waves will be refracted downward. So during the day sound waves are refracted upward away from observers while at night they are refracted downward towards the observer. In the city this effect is not as noticeable due to reflections from buildings.

Air movement can also cause the refraction of sound, as wind affects the motion of the particles that are necessary for the propagation of the sound waves. (See Figure 16.4.)

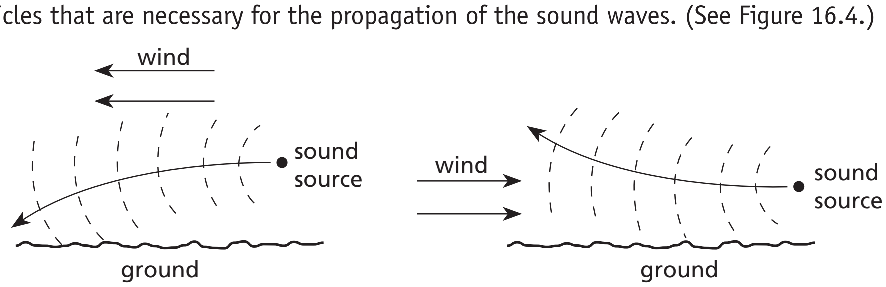


Figure 16.4
Air movement also causes the refraction of sound waves.

Diffraction

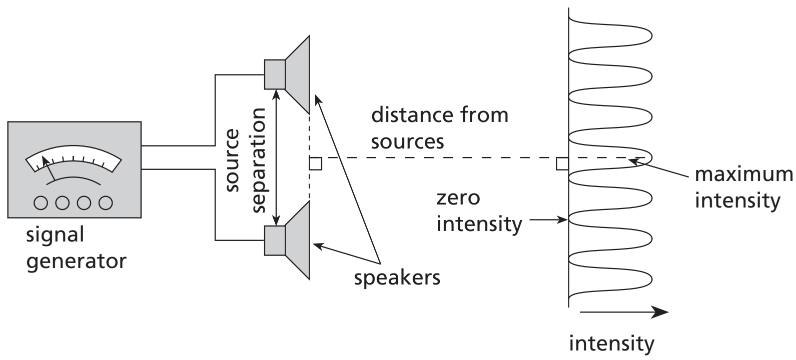
Imagine if sound waves were not diffracted around corners of objects but travelled in straight lines. It would mean that when you spoke, sound waves produced by your vocal cords would come out of your mouth and would go straight ahead. You would have to stand directly in front of a person to be heard. However, we know that people at your side can hear you because the waves diffract around the edges of your mouth. For the same reason, sound can be heard around edges of open doors. If you are put outside the classroom you can still hear the teacher's voice even though you are around the edge of the door and cannot see the teacher. Why is this?

You may also have noticed how echoes have a higher pitch than the original sound. Recall that long wavelengths are diffracted more than short ones, so the low frequencies (large wavelength) are diffracted more whereas the higher ones are reflected back to you.

Interference

As sound is a wave, interference abides by the same rules as for all waves. This can be demonstrated in the classroom by using a signal generator connected to two speakers, as shown in Figure 16.5.

Figure 16.5
The interference of sound can be observed using two speakers connected to the same signal generator making sound waves from the speakers in phase.



If you walk across the room in front of the speakers, constructive and destructive interference producing maximums and minimums of loudness will be detected. Constructive interference (loud) will produce sound when the path difference from the two speakers to the detector is $n\lambda$. Destructive interference will be detected, and therefore no sound heard, when there is a path difference of $(n - \frac{1}{2})\lambda$.

Pitch of sound

Pitch is our perception of whether a musical note is high like a soprano or low like a bass singer. The pitch of a sound refers to its frequency. If the sound has a high frequency it is said to be of a high pitch. The pitch or frequency of sounds emitted by humans is controlled by the tension in the vocal cords.

NEI Activity 16.6 PRE-PUBESCENT BOYS' VOICES

1 It would be interesting to find out why boys, before puberty, have higher-pitched voices than after puberty. What is happening to cause this 'cracking'? Do some research to discover what is happening to their vocal cords.

2 Why doesn't this happen to girls?

3 At what age, if any, does a dog's bark 'crack' and become deeper?

EI The 'snickometer'

Here's an idea for an experiment. TV broadcasts of test cricket are often accompanied by the use of a 'snickometer' to study the waveform of various noises, so that the commentators and third umpire can judge whether a ball made contact with the bat, pads, gloves etc. How do the following waveforms differ: bat on ball, bat on pads, bat on pad buckle? Propose a justifiable hypothesis before you begin. Taping the sounds and studying the waveforms on a CRO may be the way to go.

Loudness and energy

The loudness of a sound is related to the intensity of the sound, which depends on the energy carried by the wave. The energy is a function of the amplitude of the wave. Loud sounds carry large amounts of energy and cause large vibrations in the particles of the medium in which they are moving. This causes large vibrations in detecting equipment — ear drum, or microphone membranes. Refer to Section 16.9 for further discussion.

Sound quality

The quality of a sound produced by a musical instrument is dependent on the waveform associated with that instrument. The waveform is made up of a combination of frequencies and not just one frequency like that emitted by a 256 Hz tuning fork. The waveform produced by a musical instrument consists of a combination of the **fundamental frequency**, the lowest natural one, and a number of other less intense frequencies called **overtones** or **harmonics**. Overtones are whole number multiples of the fundamental frequency. This will be discussed in Section 16.6 where diagrammatic representations will help to explain the production of fundamental frequencies and overtones. Figure 16.6 shows examples of the waveforms produced by some common musical instruments.

Notes are said to differ in pitch by one **octave** when the frequency of one is double that of the other. In the frequency table of musical notes (Table 16.2), doh′ is one octave above middle C. In Latin, *octa* means 'eight', referring to the fact that there are eight notes from C to C′ inclusive.

Figure 16.6
The quality of the sound from an instrument depends on the mixture of frequencies emitted by the instrument. A tuning fork emits only one frequency.

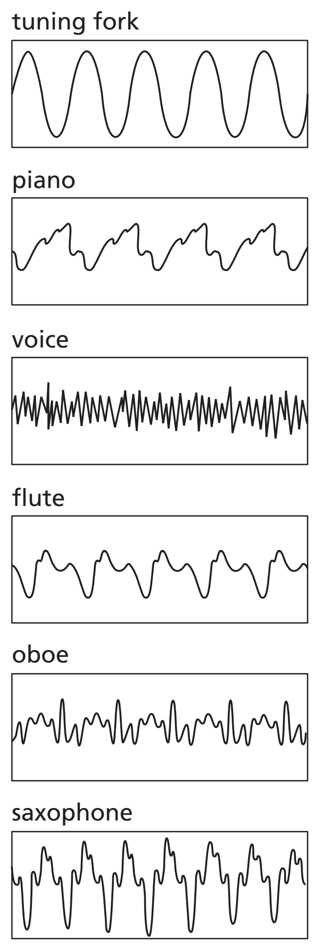


Table 16.2 MUSICAL NOTES AND THEIR FREQUENCIES

C	doh	256 Hz
D	ray	288 Hz
E	me	320 Hz
F	fah	340 Hz
G	soh	384 Hz
A	lah	427 Hz
B	te	480 Hz
C′	doh′	512 Hz

Questions

5 A student makes a noise in the back of the classroom with a frequency of 1000 Hz. Calculate how long it will take to reach the teacher in the front of the room 3.0 m from the student. What will be the wavelength of this sound?

6 A scuba diver taps one end of a 50 m copper pipe under water. Who will be first to know this has occurred — a scuba diver 50 m from the source or another diver holding the other end of the pipe? How much earlier will it be recorded?

7 Calculate the speed of a sound wave that has a frequency of 800 Hz and a wavelength of 42 cm. Can this sound be heard? Explain your answer.

8 When watching the fireworks at the local show you observe that you see the flashes from the exploding fireworks in the sky 0.50 s before you hear them. How far are they exploding from you?

9 A person standing on an observation platform in the mountains shouts and hears the echo off a cliff 1.5 s later. How far is the cliff from the 'outlook'?

10 A signal generator produces sound waves of frequency 1700 Hz, which are fed into two speakers placed 1.0 m apart. A student walks across the room 5.0 m from, and parallel to, the speakers. (Refer back to Figure 16.5.) How far apart will the student hear minimums of intensity? (The speed of sound is 340 m s^{-1}.)

11 Students sitting in the stands at an athletic competition 200 m from the start see the smoke from the starting pistol 0.6 s before hearing the sound of the gun. Find the speed of sound in air on this day.

ULTRASOUND 16.5

NOVEL CHALLENGE

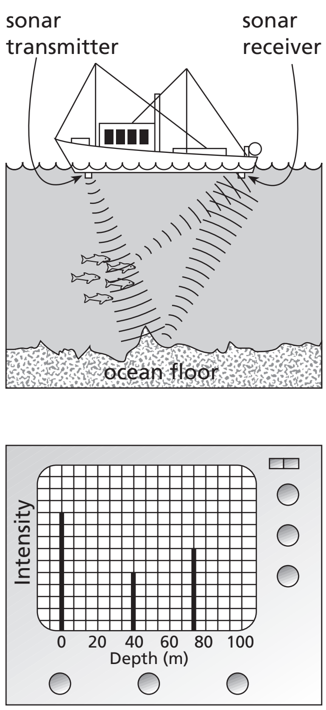
A bat emits a frequency of 120 kHz. What size insect could it best detect? You may need to read the section on radar in Chapter 15 (Section 15.8) to help you answer this.

Ultrasound is sound at frequencies above that of human hearing range, that is, above 20 000 Hz. In Latin *ultra* means 'beyond'. Artificial ultrasounds are produced by vibrating quartz crystals, which are induced to vibrate by high-frequency alternating currents.

The uses of ultrasound waves or **ultrasonics** are increasing very rapidly especially in the medical profession. Reflection and refraction of ultrasound waves are used to see unborn babies, tumours and body organs. High-frequency ultrasound is used to make particles of objects vibrate at such a rate as to make them shatter. This is used by dentists to remove plaque from teeth and by doctors to break up kidney stones. (Refer to Chapter 33, Medical Physics.) Ultrasound is also used to heal muscular injuries; high-frequency ultrasound causes a muscle fibre to vibrate, thus generating heat and increasing the blood flow to the area, improving the rate of repair to damaged muscle.

In industry, ultrasound is used for cleaning small parts, for welding plastics and metals, for driving piles, and for drilling holes in glass. In most of these high-power applications the action is caused not by the direct agitation of the sound wave but by heating and bubble formation.

Figure 16.7
Sonar equipment makes use of ultrasound waves. Reflected waves are recorded on a ship's sonar equipment as echoes.



Ultrasound has been used for a number of years in sonar equipment developed during the Second World War to detect enemy submarines. Sonar comes from the term *so*und *n*avigation *a*nd *r*anging. Because ultrasound waves have shorter wavelengths, they are less diffracted by water than sound waves are and they are not absorbed by sea water as much as microwaves are. They can therefore penetrate to great depths in water. Objects in water, the ocean floor, a school of fish, submerged ships, or enemy submarines, can be detected by the reflection of ultrasound waves.

The use of ultrasound waves to find objects is thousands of years old. Can you think of an animal that uses ultrasound to find its way around?

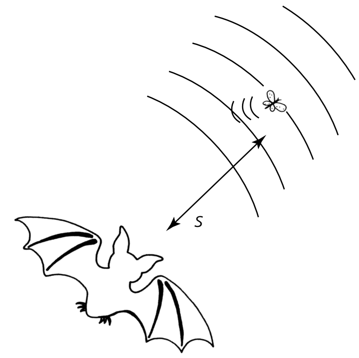
This principle is also used in guidance systems to help the blind. Ultrasound transmitters emit waves that are reflected from objects and the reflected waves picked up by a detector are heard through an earpiece. Blind people are trained to make sense of the reflected sounds, thus enabling them to identify obstacles.

Bats

Bats use ultrasound to assist their poor eyesight. They are able to produce ultrasound pulses 0.010 s apart and of approximately 100 000 Hz with approximately 3.0 mm wavelength. Because the wavelength is so small these waves can reflect from small objects. Bats can also determine the nature of the reflecting surface — a hard surface gives a hard reflection; a soft powdery surface gives softer sounds. Some bats emit a short constant-frequency signal and can analyse the return signal for frequency changes (see Doppler effect later in this chapter). Bats' ears are also concave to concentrate the reflected sound waves.

Most insects' hearing extends into the ultrasonic region. Because insects are common prey for bats their ears have become sensitive to bat frequencies. On hearing these frequencies the insects fly the other way. Ingenious scientists make use of this in developing insect repelling devices that emit ultrasound waves at bat frequencies. Some species of moths have evolved very clever methods of evading bats. They are able to produce sounds at the bat ultrasound frequencies. When they detect bat ultrasonic waves they emit bursts of ultrasonic waves at bat frequencies but of a lower intensity to confuse the bats. This causes the bats to swerve or stop and listen. While this is going on the moth folds its wings and falls to the ground undetected. Other animals such as whales and dolphins use ultrasound but the method is not fully understood.

Figure 16.8
Bats use the reflection of ultrasound waves to find their way around as well as to find their food.



Questions

12 Why do bats use ultrasonic waves rather than audible sound waves?

13 A fishing vessel looking for schools of fish sends out pulses of ultrasonic waves and finds the echo returns 1 s and 3 s later.
(a) What is the explanation for this?
(b) At what depth would you expect to find fish?
(The velocity of sound waves in sea water is 1450 m s^{-1}.)

14 The minimum wavelength of ultrasound waves bats can emit is 3.3 mm. What is the highest frequency of sound that bats can emit?

15 Explain the technique of submarine captains resting their vessels on the bottom to prevent them being detected by a surface vessel's sonar.

16.6 THE PRODUCTION OF MUSICAL SOUNDS

Musical instruments produce sounds due to the standing waves set up in three different media:

- Strings; for example, the guitar, piano, violin, viola, cello.
- Air columns; for example, the flute, clarinet, recorder, organ, trumpet.
- Membranes; for example, drums, bongos, cymbals.

Strings

When a guitar string is plucked a standing wave is formed between the ends of the string. The frequency of vibration of the string produces the resulting sound. A number of different standing wave patterns can form in a string. (Recall from Chapter 13 that there are nodes formed at the fixed ends.)

The simplest standing wave pattern established in a guitar string is shown in Figure 16.9**(a)**.

The length of the string is equal to $\frac{1}{2}\lambda$. This string length produces the **fundamental frequency** (f_0).

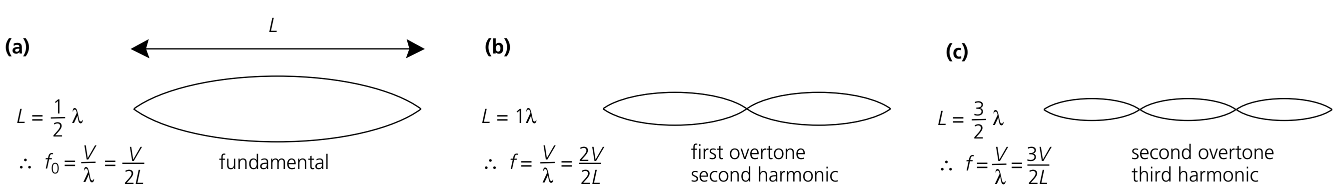
The second possible standing wave established is shown in Figure 16.9**(b)**. Here the length of the string is equal to 1λ. This is the first **overtone** and because it is twice the fundamental it is also the second **harmonic**. Musicians often refer to overtones as **partials**.

The third possibility is shown in Figure 16.9**(c)**. This time the length is equal to $1\frac{1}{2}\lambda$. This is the second overtone and as it is equal to three times f_0 it is the third harmonic. Stringed instruments — guitars, banjos, violins, and pianos — as well as percussion instruments such as drums and bongos, rely on the above principle.

NOVEL CHALLENGE

If you told a violinist that you are a physicist and she should play the strings about one-seventh of their length from the end what would she say? Measure where she plays — is it one-seventh?

Figure 16.9
The types of standing waves set up in guitar strings.



The frequency of sounds produced by string instruments has been shown to depend on the tension of the string, the length of the string and the mass per unit length of the string. For example, the six strings of a guitar are all of the same length but they produce different notes because they have different tensions and masses per unit length. The relationship between the tension, length, and mass per unit length of the string is expressed by the mathematical formula:

$$f = \frac{1}{2L}\sqrt{\frac{T}{M}}$$

where f is the frequency of the note produced in Hz; L is the length of the string in metres; T is the tension in the string in newtons; M is the mass per unit length in kg m^{-1}.

Example

Find the fundamental frequency produced by a 48 cm wire of mass 1.0 g under a tension of 85 N.

Solution

- $L = 0.48$ m.
- $T = 85$ N.
- $M = 1.0 \times 10^{-3}/0.48$ m $= 2.1 \times 10^{-3}$ kg m^{-1}.

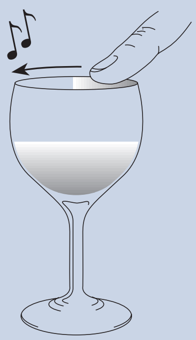
$$f = \frac{1}{2L}\sqrt{\frac{T}{M}}$$
$$= \frac{1}{2 \times 0.48}\sqrt{\frac{85}{0.00\,21}}$$
$$= 209.6 \text{ Hz}$$

NOVEL CHALLENGE

What is the squeaky sound when you are washing your hair with shampoo and what is the similarity with the shrill sound beginners produce on a violin?

NOVEL CHALLENGE

You have dipped your finger in some wine (or metho) and run it around the rim of a wineglass. A loud sound is produced. *Why doesn't it work if your finger is not degreased by the wine?*



EI Activity 16.7 RUBBER BAND GUITAR

1. Place a rubber band between the index fingers of each hand. Stretch it and pluck it with your thumb.
2. Listen to the frequency of the sound produced. Stretch it more and listen to the sound again. Let it get less taut and listen to the sound again.
3. You may be surprised with the result. Try to explain it using the law above. *Hint:* as you increase the tension what happens to the mass per unit length?

Wind instruments

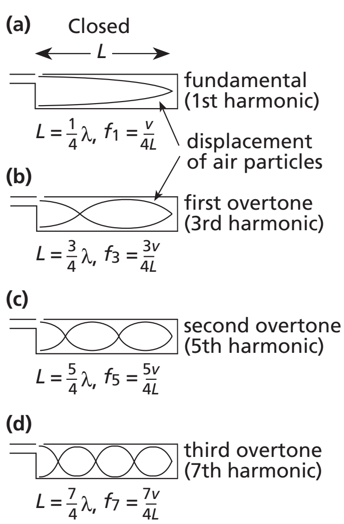
Wind instruments such as flutes also produce standing waves but in an air column. Again, as with stringed instruments, the air column can vibrate in a number of ways. The way it vibrates also depends on the nature of the instrument, whether it is closed-ended or open-ended.

Closed-ended pipes

The most common examples of closed pipes are the clarinet family and stopped flue pipes in the organ. In such pipes, the simplest standing wave pattern set up is shown in Figure 16.10**(a)**.

Notice that a node occurs at the fixed or closed end and an antinode occurs at the open or free end. For this standing wave the length of the pipe is equal to one-quarter of a wavelength, that is, $L = \frac{1}{4}\lambda$.

Figure 16.10
The first three harmonics produced by a closed-ended pipe. Notice nodes form at fixed ends and antinodes form at open ends. The different standing wave patterns produce the harmonics. (The lines show the displacement of air particles with time.)



$$\therefore \lambda = 4L$$

Since
$$v = f\lambda$$
$$f = \frac{v}{\lambda}$$
$$f_0 = \frac{v}{4L}$$

This is the fundamental frequency (f_0), also called the first harmonic.

The second possible standing wave pattern produced is shown in Figure 16.10**(b)**. Here the length $L = \frac{3}{4}\lambda$.

$$\therefore 3\lambda = 4L$$
$$\lambda = \frac{4L}{3}$$
$$f = \frac{3v}{4L}$$

This is the first overtone but since its frequency is equal to 3 times f_0 it is the third harmonic.

Figure 16.10**(c)** shows the third possible standing wave. This is the second overtone or the fifth harmonic.

NEI Activity 16.8 OPEN-ENDED PIPES

Using the information in Figure 16.11, draw a diagram showing the next two waveforms for the 5th and 6th harmonics of the open-ended pipe. Also label these with correct overtone names.

You may also wish to research a quite famous mathematical relationship which is used in the physics of vibrating strings. It is called Mersenne's law, and relates the vibration frequency of the string or wire to its linear mass, tension and length. Present your findings in a research report including a discussion of the formula.
Find out Mersenne's full name, his nationality, and how Galileo fits into the story.

Open-ended pipes

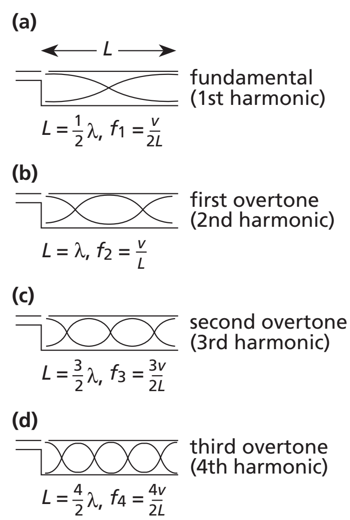
The major difference between open-ended and closed-ended wind pipes is that the standing waves formed in open-ended pipes have antinodes at both ends. Apart from the clarinet family and stopped flue pipes in the organ, most other wind instruments are of the open-ended variety.

The simplest possible standing wave formed in an open-ended pipe is shown in Figure 16.11**(a)**.

Here
$$L = \frac{1\lambda}{2}$$
$$\therefore \lambda = 2L$$
$$\therefore f = \frac{v}{2L}$$

This is the fundamental or the first harmonic.

Figure 16.11
In open-ended pipes antinodes form at the open ends.



The second possibility in shown in Figure 16.11**(b)**.

Here $$L = 1\lambda = \frac{2\lambda}{2}$$

$$\therefore f = \frac{2v}{2L}$$

Therefore this is the first overtone or the second harmonic.

The third possible standing wave is shown in Figure 16.11**(c)**. This is the second overtone or the third harmonic.

Changes in length

In pipes and strings, you have seen that changes in length of the string or air column alter the frequency of the standing waves. Here are some examples with musical instruments:

Guitars When a string is pressed down on to a 'fret' (the wood or metal bars on the fingerboard), the length of the string is shortened and produces the next semitone higher. When the bass E string is pressed down at the fifth fret its frequency corresponds to that of the second string (A). A combination of finger placements on the different strings produces harmonic sounds called 'chords'.

Flutes and reeds (recorder, clarinet, organ, bassoon) Fixed-length pipes have been around since the beginning of human history but they can only produce set harmonics as shown in the discussion above. In about the tenth century, to fill in the missing tones of the musical scale, finger holes were cut into musical pipes. This effectively shortens the length of the air column, making an antinode at the position of the uncovered hole. In the eighteenth and nineteenth centuries, makers covered the holes of trumpets and horns with keys to enable them to play the complete scale.

Trombone Another way of providing missing tones was to increase the sounding length of the tube using a telescoping slide. Hand-stopping the 'bell' (the open end) was discovered in the eighteenth century to fill in more gaps in the instrument's harmonic series.

NOVEL CHALLENGE

An orchestra tunes up at the start of a concert but as the theatre warms up the musicians have to retune their instruments.

Do they find that the pitch of their instruments rises or falls as the theatre warms up? String musicians can change the tension of the strings. What do wind musicians do?

End correction

The wavelength formulas of closed-ended and open-ended pipes discussed above are not strictly correct. The particles of air at the open end of the tube do not strictly vibrate in one dimension, and a small correction, which depends on the diameter of the pipe and takes account of the motion of the particles in other dimensions, is needed.

To be strictly correct, the fundamental wavelength produced in a closed-ended pipe is:

$$\lambda = 4(L + 0.4d)$$

where λ is the fundamental wavelength; L is the length of the pipe; d is the diameter of the pipe.

For an open-ended pipe this correction factor is required for both ends. Thus the fundamental wavelength produced in an open-ended pipe is:

$$\lambda = 2(L + 0.4d + 0.4d)$$

or $$\lambda = 2(L + 0.8d)$$

Only apply end correction in problems when you are told to do so.

Activity 16.9 OPEN-ENDED PIPES

Part A

1 Draw the next two possible standing wave patterns. (Refer to Figure 16.11.)

2 Determine the overtones and harmonics of these two.

Part B

1 Blow into a recorder gently and listen to the sound. If you don't have a recorder, use a test tube half filled with water.

2 Blow harder and see if you can produce the first overtone. Musicians call this 'overblowing'.

3 Blow harder still and if you are lucky you may create the next overtone. It sounds awful but this is physics, not culture.

TEST YOUR UNDERSTANDING

Have you ever heard someone with a very loud and expensive car stereo driving down the road? Why do you just hear a thumping bass sound and no high frequencies if it is supposed to be such a good stereo?

Activity 16.10 A USEFUL SUMMARY

Prepare a data table as shown (Table 16.3) and fill it out for the first four standing wave patterns for **(a)** strings; **(b)** closed pipes; **(c)** open pipes. Keep this for revision.

Table 16.3

DIAGRAM	L	λ	f	OVERTONE NUMBER	HARMONIC NUMBER

Questions

16 **(a)** Is a higher frequency note produced in a long guitar string or a short string of the same tension?
(b) Find the frequency of the note produced by a 51 cm guitar string of mass 0.50 g under a tension of 90 N.

17 A student blows across the mouth of a closed-ended plastic pipe 0.25 m long. Calculate the fundamental frequency and the frequency of the third harmonic.

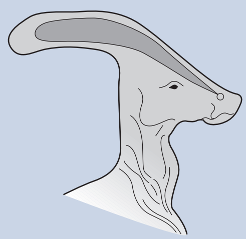
18 The distance between the reed and a hole in a recorder is 15 cm. Calculate the fundamental frequency and the next three harmonics that are heard when this instrument is played.

19 Draw the standing wave patterns set up in an open-ended pipe whose length is 20 cm and where the pipe's length equals **(a)** $1\frac{1}{2}\lambda$; **(b)** 3λ; **(c)** $\frac{1}{2}\lambda$. **(d)** Calculate the resultant frequencies emitted by the pipe in the above situations.

RESONANCE AND FORCED VIBRATIONS 16.7

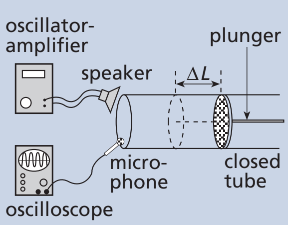
PHYSICS UPDATE

The 'trombone duckbill' was a 10 m high dinosaur known as Parasaurolophus. It had a 1.5 m long skull (see figure), which had a large cavity consisting of tubes and chambers. Scientists at the New Mexico Museum of Natural History deduced that this was a resonance chamber used for mating and warning calls. The sound produced would have been about the same frequency as the lowest note on the piano. You can sample this sound on the Internet — just search for the museum (nmmnh + trombone).



TUBE RESONANCE

A closed tube with a variable length can be made to resonate to particular sound frequencies. The change in length of the tube between successive resonant positions is equal to one-half a wavelength. Thus $v = \Delta L \,.\, 2f$ This fact can be used to measure the speed of sound in the laboratory using the apparatus below.



Forced vibrations

When a tuning fork is struck with a mallet it vibrates at its natural fundamental frequency as well as emitting a few less intense lower order harmonics. This fundamental frequency depends on the length, thickness and composition of the fork. The intensity of the sound produced can be increased by placing the end of the fork on a table top. The table top is forced to vibrate at the same frequency as the fork thus intensifying the sound produced. The same occurs for a guitar string. When it is held between two clamps and plucked it does not produce a very intense sound, but when attached across the bridge of a guitar and plucked the sound is more intense because the wood of the guitar (the 'soundboard') is forced to vibrate in response to the vibrating string. Violins, basses and other musical instruments use the same principle of forced vibrations to intensify the sound produced.

EI Activity 16.11 FORCED VIBRATIONS

1 Excite a tuning fork and time its 'life' — that is, how long you can hear it.
2 Repeat the above but this time hold its stem on a desk.
3 What do you hear now? Why is this?

Resonance

Resonance is the effect that occurs when a body vibrates at its natural frequency. All bodies possess a natural resonance frequency. They can be made to start to resonate by another body touching them or being in close proximity to another vibrating object at the correct frequency. For example, a vibrating tuning fork can cause another one of the same frequency to resonate if it is close by. Here are some practical examples of resonance:

- Resonance vibration is why soldiers do not march in step when crossing older bridges as they might cause the bridge to resonate and possibly collapse.
- Mechanical resonance caused the collapse of the Tacoma Narrows Bridge in the USA in 1940. On a windy day four months after the completion of this suspension bridge it began to vibrate at its resonant frequency, causing it to collapse.
- Opera singers can shatter glasses because they can sing notes at frequencies that hit the resonance frequency of the glass.
- Singing in the shower sounds so good because the column of air in the shower enclosure is about the right dimension to resonate as a closed pipe to amplify many singing notes in your voice.
- If you run your moist finger around the lip of a crystal glass you hear a high-pitched squeal. This is the resonant frequency of the glass. You may first have to dip your finger in wine or metho to remove traces of oil.
- You may also have noticed that your hair gets squeaky clean after you wash it. This too is a resonance effect of longitudinal vibrations in the hair strands.
- Much louder sounds occur in musical instruments when standing waves in the tubes resonate in harmony with the vibration in the mouthpiece or reed, etc. This can be demonstrated if a tuning fork is attached to a resonance box. The sound of the tuning fork becomes much louder.
- Good loudspeakers are designed such that no one (or more) of their resonance frequencies (of which there are many) is dominant. Otherwise you would hear some frequencies louder than others.
- A seashell acts like a closed-ended pipe or resonator. The surrounding soft background noise provides sounds containing all frequencies the ear can hear. However, the shell increases the intensity of the frequency that is the same as its resonance frequency, thus creating a louder, almost pure, frequency or tone.

Activity 16.12 RESONANCE

Activities to demonstrate resonance:

Part A

1 Set up apparatus as shown in Figure 16.12.
2 Set one pendulum swinging and observe what happens to the other.
3 Shorten one to 15 cm and repeat the experiment.
4 What do you notice?

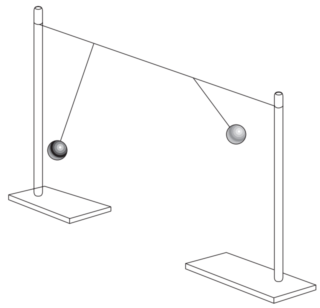


Figure 16.12 The apparatus used to demonstrate resonance. If one weight is made to oscillate the other will begin to oscillate.

Part B

1 Excite a tuning fork and hold it near your open mouth.
2 If resonance does not occur explain why and try another fork.
3 What distinction can you find between boys' and girls' mouth cavities?

Part C

1 Suspend a ping-pong ball on a piece of thread so that it hangs just touching the prong of a tuning fork.
2 Excite another matching fork nearby and see if the ball moves.
3 Explain this in terms of resonance.

NOVEL CHALLENGE

If you hold the right tuning fork up to your mouth cavity you can cause the cavity to resonate. *Would you expect boys or girls to have the lower resonant frequency? Why?*

16.8 BEATS

Beats are heard when sound waves of slightly different frequencies occur together. For example, when frequencies of 320 Hz and 322 Hz occur together, the constructive and destructive interference of these sound waves causes sounds that get louder and softer at regular intervals. The above example has a **beat frequency** of 2 beats per second, which is the difference in the two frequencies: $f_B = f_2 - f_1$.

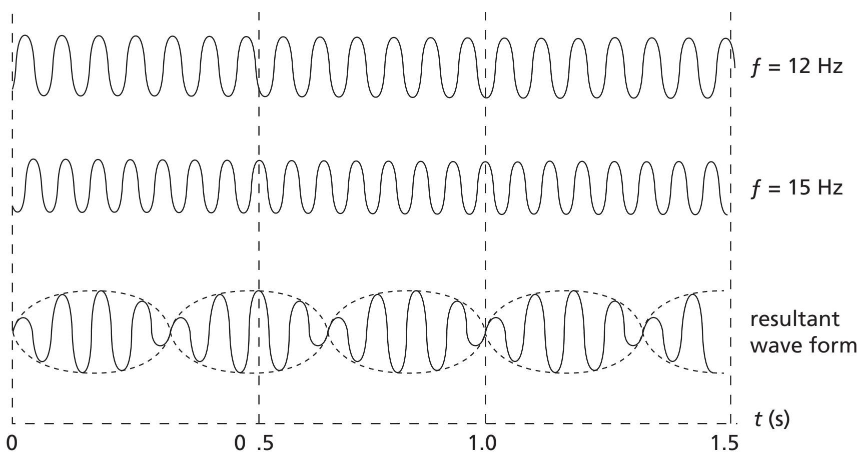


Figure 16.13 Beats are formed by the constructive and destructive interference of waves that are produced together but have slightly different frequencies.

Beats are used to tune pianos and other musical instruments. If a note on the piano and a tuning fork are sounded at the same time beats may be heard. The tension in the piano string can be adjusted until no beats are heard. At this time the piano is producing the same frequency as the tuning fork. Do you wonder why it is called a tuning fork?

EI Activity 16.13 BEATS

1. Sound a pair of matched tuning forks together and listen for beats. There should be none.
2. Add a small lump of Blu-tack or a rubber band on one prong of one fork and excite both again.
3. What do you hear?
4. Does the beat frequency increase or decrease as the Blu-tack 'load' is moved down the prong? Explain why.
5. Take a look at Figure 16.5 again and redraw the diagram to illustrate how a similar set-up could be used in the laboratory to produce 'sound wave beats'. You will need two signal generators. Try to set up the system with your teacher.

Questions

20 **(a)** Explain what resonance is and how it is produced.
(b) Explain why some older cars and buses start to vibrate when their engines reach certain revs.

21 State the beat frequency when the following pairs of tuning forks are sounded together: **(a)** 220 Hz and 217 Hz; **(b)** 340 Hz and 336 Hz; **(c)** 682 Hz and 688 Hz.

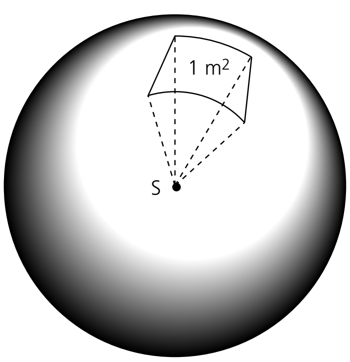
INTENSITY OF SOUND 16.9

One of the commonest properties of sound discussed in newspapers or on TV is sound intensity. There are ongoing protests about noise levels by people who live near airports or near outdoor concert venues. People living near an airport have to cope with noise levels of about 100 dB. The level of noise produced by rock bands may be as high as 120 dB. To safeguard the health and safety of workers, governments pass legislation that places restrictions on the noise levels in working environments. However, what do these noise levels 100 dB, 120 dB, etc. mean?

Sound waves, like all waves, carry energy, which spreads out from the source like a pebble being dropped into a pond. Suppose we take a 1 metre square area at some distance from a source, as shown in Figure 16.14.

The intensity of sound at this distance from the source is defined as the amount of sound energy that passes through this area per second.

Figure 16.14
The intensity of sound is measured by the sound energy passing through a square metre square area per second. It is measured in W m^{-2}. The intensity level is measured in dB.



The absolute sound intensity (I) is the energy carried by the waves per second through an area of 1.0 m^2

Since power is energy per unit time, then absolute sound intensity is the power passing through a unit area perpendicular to the propagation of the wave.

$$I = \text{power/area}$$

Therefore the units of absolute sound intensity are W m^{-2}.

A range of intensities of various sound sources is given in Table 16.4. Notice the large range of intensities the human ear can detect.

Table 16.4 SOUND INTENSITIES OF VARIOUS COMMON SOURCES

NOISE	ABSOLUTE INTENSITY, ($W\ m^{-2}$)	RELATIVE INTENSITY LEVEL (dB)
Jet plane taking off	10^3	150
Pain-producing	1	120
Rock concert	1	120
Chain-saw	$10^{0.5}$	115
Power mower	10^{-2}	100
Jackhammer	10^{-2}	100
Noisy restaurant	10^{-4}	80
Vacuum cleaner	$10^{-4.5}$	75
Ordinary conversation	10^{-6}	60
Average home	10^{-7}	50
Purring cat	$10^{-9.5}$	25
Whisper	10^{-10}	20
Rustling leaves	$10^{-10.5}$	15
Faintest sound that can be heard	10^{-12}	0

The terms sound intensity and sound loudness do not mean the same thing, although they are related. The absolute intensity of any sound wave of a given frequency depends on its amplitude. The greater the wave amplitude, the greater the sound intensity. This is reasonable since the greater the amplitude the greater the energy initially expended in setting up the sound wave. The sensation of loudness is related to the measurable quantity, absolute intensity. In general, sound waves of higher intensity are louder to the ear, but the ear is not equally sensitive to all frequencies. Consequently a high-frequency sound may not seem as loud as a low-frequency sound of the same intensity. The relationship between intensity and loudness is not linear. For example, for a sound of a given frequency to be twice as loud to the ear it must have ten times the absolute intensity. The relationship between the two is therefore logarithmic.

Because of the great range it is often convenient to use a comparative scale to express relative sound intensity levels — the **decibel scale**. This is a logarithmic scale.

$$\beta = 10 \log (I/I_0)$$

where I is the intensity of the sound in W m^{-2}; I_0 is the reference level, taken as the least audible sound (10^{-12} W m^{-2}); β is the relative intensity level in dB.
Therefore the relative intensity level of a whisper in dB is:

$$\begin{aligned}\beta &= 10 \log \frac{10^{-10}}{10^{-12}} \\ &= 10 \log 10^2 \\ &= 20 \text{ dB}\end{aligned}$$

The conversion of absolute sound intensities to relative intensity levels in dB for many common sounds is also shown in Table 16.4.

The decibel scale was named in honour of Alexander Graham Bell (1847–1922), the inventor of the telephone. One bel (1 B) is equal to 10 decibels (10 dB). The bel is too big for general use.

NOVEL CHALLENGE

When the physics laboratory is quiet, a dropped pin can be heard clearly at the back of the room. Calculate the energy arriving to the ear of a person at the back. If you want second-hand data, assume the pin has a mass of 0.2 g and it is dropped from a height of 1.0 m (use GPE = ***mg**h*). Assume all GPE is transformed into sound energy that radiates outwards as a large spherical surface ($A_{sphere} = 4\pi r^2$). *Calculate the amount of energy per square centimetre at the back of the room (say r = 5 m). Is it more or less than 10^{-9} J/cm^2? Not much huh?*

NOVEL CHALLENGE

The Tacoma Narrows suspension bridge in the USA collapsed in a mild windstorm (67 km/h) on 7 November 1940. It started oscillating up and down 30 times per minute with an initial amplitude of 1 m, which later increased to 8 m prior to collapse. You can simulate the effect by directing an air dryer jet sideways onto a strip of paper held at both ends. *What different motions can you achieve? Did you get twisting and bending? How does tautness affect it?*

NOVEL CHALLENGE

A Think of sound waves radiating outwards in a spherical shell. Calculate the intensity of a rock band's sound at 100 m if it has an intensity of 120 dB close up. Note: the surface area of a sphere = $4\pi r^2$.

B Humans' hearing is pretty sensitive. If a pin is dropped from 1 m high on to a desktop, you can hear it from the back of the classroom (try it!). Make some estimations to calculate the sound energy received by your ear. Recall that the sound radiates out as a concentric spherical surface and this has an area of $4\pi r^2$. How far is it to the back of a room? How big is your ear hole? What does a pin weigh?

Questions

22 Convert 10^{-7} W m^{-2} to decibels.
23 Convert 5×10^{-8} W m^{-2} to decibels.
24 Convert 85 dB to W m^{-2}.

Noise pollution

Sounds from cars, aircraft, lawn mowers, and rock bands can be more than just annoying. **Noises** — sounds that are not periodic and lack order — can cause damage to the ear, and may cause temporary or permanent deafness. They can cause other physical ailments such as tiredness and lack of concentration as well as stress. Some controversies concerning noise:

- A great deal of debate occurs when changes to runways of airports result in houses being in flight paths.
- Councils in residential areas place bans on the operation of noisy machinery between certain hours, and especially on Sundays. See if you can find the local regulation with regard to this.
- Many complaints are received by police and councils about barking dogs and even squawking birds.
- Workplace safety regulations require operators of noisy machinery — tractors, pneumatic drills, jackhammers, etc. — to wear ear protection.
- Police also have a duty to control the noise emissions from cars.

All the above, combined with the emission of smoke, odours, etc., affect people's environment. People have a right to not have to suffer from activities that pollute their environment.

NEI Activity 16.14 NOISE POLLUTION

Use one of the five bullet points above to form the basis of a research report. Select one of the controversies listed and discuss fully your research into the laws of physics that are relevant to their solutions. You should present your own ideas to help solve the problem topic you choose.

MODERN SOUND TECHNOLOGY 16.10

Basic components

Modern audio engineering involves the recording, amplifying and playback of a range of sounds for music recording, public address systems and special effects such as in movie soundtracks. This process utilises both analog and digital signals and the aim is to represent and reproduce original sounds without distortion. Audio engineers need to be able to use musical instrument transducers, microphones, amplifiers, digital signal processors, audio and video cassette decks, CD–DVD players and speaker systems. These can be found in professional theatres and movie complexes, home theatre sound systems or even in modern car audio systems. Photo 16.3 shows a typical home theatre set-up with multiple components and speakers. We will briefly discuss some of these components here, but they are also further discussed in Chapters 24, 25, 26 and 31.

Photo 16.3
A typical home theatre system.

Microphones A microphone detects sound and converts the waves into electrical signals which can be further amplified and/or recorded. Dynamic microphones use a thin diaphragm attached to a voice coil which vibrates inside a magnetic field when struck by sound waves, producing a continuous AC micro-voltage. Crystal or piezo-microphones rely on a movable diaphragm that distorts a small piezo-crystal when sound waves strike and again produce AC micro-voltages. Capacitive type or Electret microphones rely on the capacitive effect of two small thin metallic plates separated by a dielectric gap, across which is a larger DC voltage. Sound waves striking the capacitor's movable plate cause a small AC micro-voltage variation in the DC voltage applied. (Refer to Figures 16.15 and 16.16.)

Especially in sound stage work, microphones have been developed that are based on these principles but are further enhanced. Shot-gun mikes provide a very narrow response beam; pressure-zone PZM mikes produce an omni-directional wide field response to sound-wave pressure variations. The microphones that most rock band singers use are dynamic types because they need to be used close to the mouth; they are fairly insensitive and do not pick up sounds from a wide direction pattern.

Amplifiers These devices electronically increase the voltage amplitude of the output signals from microphones, instrument transducers such as guitar pick-ups, or other auxiliary devices such as CD players or digital effects processors (synthesisers), in order to make the voltage and electrical power levels high enough to drive loudspeaker systems directly. Many electronic audio devices will also have many stages of internal pre-amplification before the output signal is sent to a separate multi-channel instrument or system power amplifier.

Many amplifiers rely on integrated circuit chip modules to carry out the voltage and power amplification, with output power measured in watts for a typical radio receiver amplifier, up to a few hundred watts per channel for a home theatre amplifier, and finally up to several thousand watts for large amplifiers used by the major rock bands. Amplifiers dissipate a large amount of heat and need to be protected from overheating with appropriate installation housings and cooling fans. Often amplifiers can sense an appropriate input signal level and switch into standby mode when not in use, to assist in heat control.

Loudspeaker systems These electrical devices are at the end of the audio chain and convert electrical AC signals from the amplifier outputs back into sound waves. There are many designs for speakers, and it is very complex to produce a design involving multiple drivers because the behaviour of sound waves inside closed or open boxes, as well as the nature of the listening environment, needs to be taken into account. The term **speaker** usually refers to the electromagnetic driver as well as its box housing, while the term **driver** refers to the actual device reproducing the sound. All drivers will eventually move air backwards and forwards by the use of a paper or polymer diaphragm or cone, producing rarefactions and compressions which are the sound pressure waves that our ears respond to. In many speaker driver designs the cone is attached via an assembly called a 'spider' to the voice coil, which carries the AC current waves from the amplifier. The voice coil moves as a result of being within the strong magnetic field of the speaker magnet assembly. When AC current flows through the voice coil the magnet induces a force on the coil which causes it to move laterally. This magnetic induction effect is discussed in more detail in Chapter 25. (Refer to Figure 16.17.)

Figure 16.15
A crystal microphone uses piezoelectric crystals, which are able to produce currents when subjected to pressure.

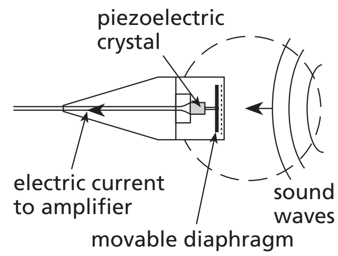
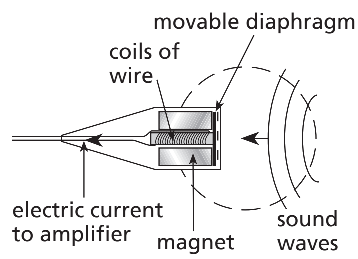


Figure 16.16
A moving coil microphone produces currents when the pressure of sound waves causes coils to move in magnetic fields.



NOVEL CHALLENGE

It is common to state amplifier output power in a variety of ways. Sometimes manufacturers use misleading terminology when describing output power in order to make their product seem better than it actually is. Find out the meaning of the measurement units called Peak, Peak to Peak, RMS or even PMPO; and find out which is the industry standard.
Why can the use of these terms be misleading to the consumer?

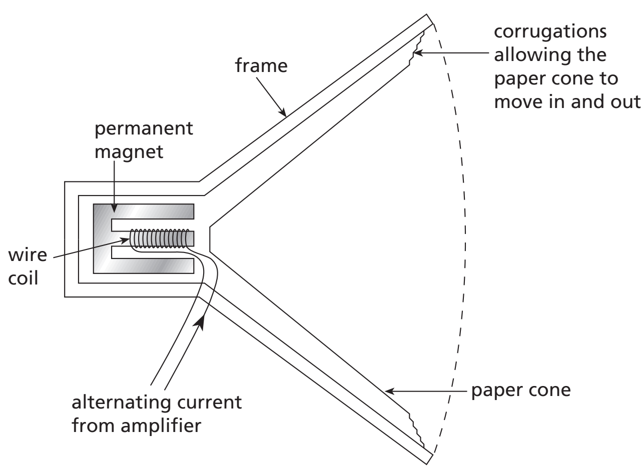


Figure 16.17
A schematic diagram of a speaker. Variations in the current in the coils cause variations in the movement of the paper cone, which results in the production of different sound waves.

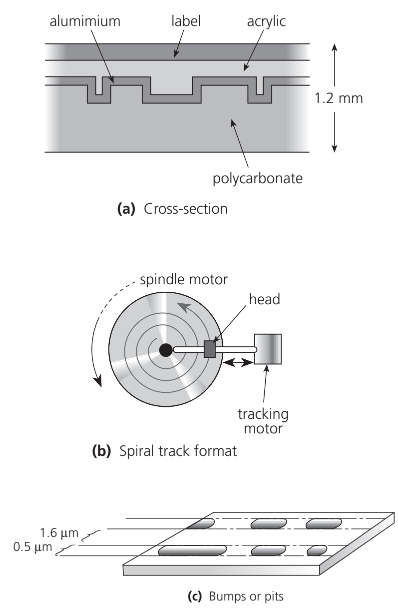
Modern speaker driver research is concentrating on the materials that make the magnet assembly. Australian CSIRO researchers are among the world's leaders, producing exotic magnetic alloys that provide very strong magnetic field densities. This enables speakers of high power-handling capability to be made smaller and smaller. A visit to the local hi-fi component retailer will confirm the incredibly small size of some modern speakers for the sound output they provide.

Speaker drivers are often designed to reproduce only a small part of the audio spectrum efficiently (from 20 Hz to 20 kHz); so when used in combination they need to be separated with electronic cross-over networks involving capacitor–inductor frequency filters. Some interesting names are given to various frequency component speaker drivers. For example, the **Subwoofer** is the high-power driver used to reproduce low-frequency effects, LFE signals such as explosions and deep bass music notes, usually below about 150 Hz. These are usually housed in large boxes. **Woofers** are used to reproduce low to mid-range frequencies up to about 1.0 kHz. **Squawkers or mid-range** drivers can handle a wide range of frequencies in the audio spectrum and are often used as the centre speech-reproducing driver, typically from several hundred hertz to about 50 kHz. **Tweeters** are the specialist high-frequency drivers that reproduce top-end sounds such as hisses and chirps from several thousand hertz right up to 20 kHz. Often they only need to be quite low in output power as the large amount of air movement is not necessary.

A well-designed speaker system may contain multiple drivers, with woofer–squawker–tweeter in the one box cabinet, and be able to handle simply one output channel from the amplifier. Speaker resistance or impedance to the flow of AC electric current is measured in ohms. Typical home theatre speaker systems are rated at 8 ohms, while typical car audio speaker systems may be rated at only 4 ohms.

The home theatre digital sound revolution

Figure 16.18
A compact disc.



Audiophiles of the past usually raved over their quadraphonic or stereophonic systems, which normally consisted of analog components such as amplifier, record player, AM–FM radio receiver, and one or more magnetic tape decks. This was usually coupled with an inefficient stereo left–right speaker system which, if the system was a good one, reproduced original music with minimal distortion and noise levels. With the coming of the digital revolution, audio and videophiles now have components such as digital AM–FM stereo PLL (phase locked loop) tuners, CD and DVD players, hi-fi stereo video recorders and DAT (digital audiotape) decks coupled with HD (high-definition) digital televisions, multi-channel and decoder amplifiers, which reproduce audio and video with incredible fidelity and without discernible electronic noise. The computer revolution has allowed this equipment to be used with ease to produce audio and video source material in the home without serious sound studios.

Let's now take a look some of the technology behind this digital audio–video revolution and find out how it works. The process of analog to digital conversion, which underlies this technology, is discussed in Chapter 24 (Section 24.2).

AM–FM radio tuners A radio tuner will receive transmitted amplitude or frequency modulated waves from a local radio station, usually in stereo (tuning principles are discussed in Chapter 31) and convert small microvolt signals into a form ready to be amplified.

Cassette or digital audio tape decks These contain electromagnetic induction heads that decode magnetic patterns stored on magnetic tape and convert them into an electronic signal stream, again allowing amplification.

Audio compact discs (CDs) An audio music (CD-DA format) or computer data (CD-ROM format) device is a simple injection-moulded circular disc of polycarbonate plastic about 12 cm in diameter and about 1.2 mm thick, onto which is evaporated a thin layer of reflective aluminium. This is then covered with a layer of acrylic for protection and marketing labels are attached to the top of the acrylic layer. When the CD is recorded or written to by a laser beam, small indentations that represent the 1s and 0s encoded digital bitstream are made into the reflective aluminium–plastic layer. (Refer to Figure 16.18, which of course is not to scale.) These indentations become raised bumps when read from the scanning pickup laser head side of the disc.

The track written onto a CD is one long continuous spiral 0.5 µm wide (1 µm = 1×10^{-6} m), laid down with separations of only about 1.5 µm. Typical data bumps on the track are at least 0.83 µm long and 126 nm high (1 nm = 1×10^{-9} m). For a typical 700 MB CD the recording track may be about 5.0 km in total length. The CD player has the difficult job of reading the data track and has four major functions to perform:

1 *The drive or spindle motor* must spin the disc at different speeds depending on what part of the data track is being read. Speeds of between 200 and 500 rpm are common, but CD-ROM discs in a computer drive may spin even more quickly.

2 *The tracking motor* moves the laser head linearly over the disc surface from inside to outside. The tracking motor position also determines the spindle motor speed because the reflective bumps towards the outer rim of the CD will be travelling more rapidly past the pickup head, and the spindle motor has to be slowed down so that data comes off the disc at a constant rate.

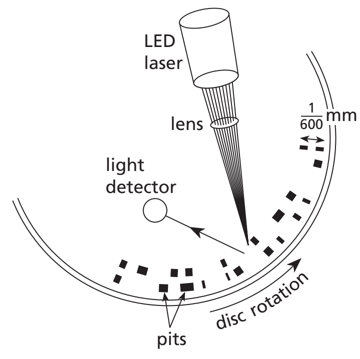
3 *The lens and laser assembly*. This is why all drives have the LASER CLASS device warning symbol attached to them, but in reality they are not dangerous at all when used normally. The lens system must focus the laser beam onto the lands and bumps of reflective aluminium. The laser beam passes through the polycarbonate layer and reflects off the bumps and lands, producing a variable light signal to the opto-electronic sensor. Further electronic digital gates in the drive provide the digital bitstream to the DAC and finally to the internal signal amplifier. Figures 16.19 and 16.20 show these processes.

4 *The error-correcting and subcode-data-reading electronics.* This system monitors any laser head misreads due to dust or scratches on the disc surface (called burst errors) and allows the bitstream to be recovered. The subcode data encodes position information on the disc so that operations like finding and skipping to a particular music track can be accomplished. These systems are most important on a computer data CD-ROM. Can you see why?

It is interesting to further consider the differences between a normal CD and the recordable or writable **CD-R** format, or even the rewritable **CD-RW** format. Engineers are constantly improving the capabilities of these digital formats and player-readers. Figure 16.18 shows the layers of a normal CD. In the CD-R format there are no bumps or lands, just a smooth reflective aluminium layer that rests on top of a layer of photosensitive dye. On a blank CD-R disc this dye layer is translucent, but when the burner laser heats it up the dye becomes opaque. The data track thus becomes a series of dark and reflective spots. One disadvantage of this is that, once burnt, the CD-R layers are permanent.

To produce the re-recordable CD-RW format disc even more layers are required. Sitting below the aluminium layer is a special chemical layer containing a crystalline phase-changing compound of metallic alloy. Under normal conditions this compound's form is crystalline and translucent, but again if the burner write laser heats it above 600°C for an instant it melts and becomes amorphous and opaque even after it is cooled. This process allows the digital bitstream to be encoded as before. The CD-RW drive also contains an erase laser setting that can hold the phase-change layer long enough at its melting temperature to allow the compound to revert to its translucent state, thus allowing data to be rewritten over and over again.

Figure 16.19
CDs contain pits that result in the interference of light to produce electrical currents.



tracks on a digital disc

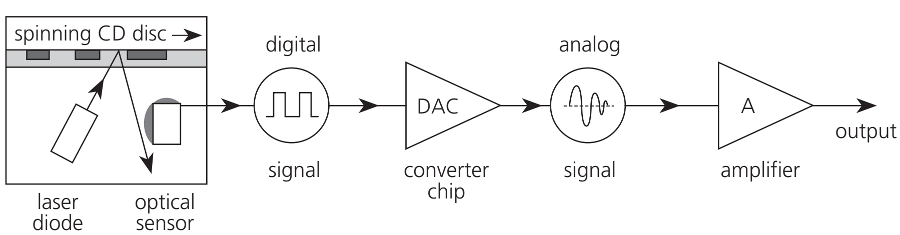


Figure 16.20
Schematic diagram of CD player electronics.

Digital versatile discs (DVDs) An early product of analog technology called the laser-disc provided better quality than magnetic videotape and had many advantages over that format, but did not last long in the industry. It has now been superseded by the digital versatile disc (DVD). A DVD is similar to a CD but can hold up to seven times as much data. Even the drive, tracking and read electronics is much the same as a standard CD. DVDs, because of their increased data-holding ability, are used for storing huge amounts of digital data such as complete multi-volume encyclopaedias or full-length MPEG-2 encoded (a data compression technique) movies and associated extras.

Compared to the 0.5 μm wide tracks of CDs, the DVD tracks are just 320 nm wide, separated by only 740 nm. The bumps encoded on the metal layers are just 120 nm high and have a minimum length of 400 nm. Not only is an aluminium layer used in DVDs but also semi-reflective gold layers are used to allow the laser beam to access multiple layers. Most DVD movies are encoded using 96 kHz 24-bit digital information, so this means that the DVD player must have at least a 96 kHz 24-bit DAC for replay. Table 16.5 lists the DVD layering process formats available.

Table 16.5 COMPARISON OF THE DVD LAYERING FORMATS

DVD LAYER FORMAT	DIGITAL CAPACITY	PLAYING TIME AVAILABLE
Single-sided/single-layer	4.38 GB	Approx 2 hours
Single-sided/double-layer	7.95 GB	4 hours
Double-sided/single-layer	8.75 GB	4.5 hours
Double-sided/double-layer	15.9 GB	Over 8 hours

To really enjoy recorded music at its brilliant digital best the latest DVD format is DVD-audio. This format requires the use of a 192 kHz 24-bit digital-to-analog converter DAC, which is higher than a typical DVD player. The following table compares the older CD-DA audio compact disc with the latest DVD-audio standard.

Table 16.6 COMPARISON OF CD-DA AND DVD-AUDIO COMPACT DISCS

AUDIO SPECIFICATION	CD-DA STANDARD	DVD-AUDIO STANDARD
Sampling rate	44.1 kHz	192 kHz
Sampling resolution	16-bit	24-bit
DAC Output levels	65 536	16 777 216

Home theatre digital sound formats As we said at the beginning of this section, one of the greatest advances in sound technology has been in the use of multi-channel sound systems and the improvements in video output. A typical DVD player and digital home theatre amplifier (receiver-amplifier if it also includes an AM-FM stereo radio-tuner) provides signal outputs for up to eight independent sound channels, commonly referred to as 'surround sound'. **Dolby Digital 5.1** or DD surround sound is one of the best known and is developed by Dolby Laboratories, famous for noise reduction technology in previous generations of sound equipment.

In general, to produce surround sound modes, electronic equipment needs to be able either to synthesise new channel information from the originally recorded simple stereo left–right (L–R) pair of signals, or to handle the already encoded multi-channel information coming from the recorded medium such as a DVD. The amplifier is said to contain sound system '**decoders**' if it is able to perform this function. Let's take a look at some of the more common digitally enhanced surround sound modes.

1 Three-channel stereo — an analog electronic simulation of an extra front centre channel: L–C–R.

2 Dolby Pro-Logic — an analog electronic simulation of a front centre channel, as well as the simulation of a mono rear channel, sent most often to two separate surround speakers: L–C–R–[SR–SL]. A most recent version called Dolby Pro-Logic 2 is available that increases the signal level amplitude of both the C and rear S channels.

3 Dolby AC-3 or Dolby Digital 5.1 — Full processing of a digital bitstream at either 384 or 448 kilobits per second of up to full discrete full bandwidth channels (20 Hz–20kHz) plus a low frequency effects LFE channel at a bandwidth of 120 Hz (this is the so-called 0.1 channel or subwoofer output): 6 speakers L–C–R–LS–RS–LFE.

Digital compression of the bitstream is used in all digital sound formats. DD 5.1 uses a compression ratio of about 12:1. Most amplifiers will provide only a low level signal output for the LFE channel, so it will often require extra amplification. Figure 16.21 illustrates the home theatre speaker set-up for this type of system compared to the audience position. It is important to remember that all speakers except for the subwoofer should ideally be at typical ear height, or just above, for the audience.

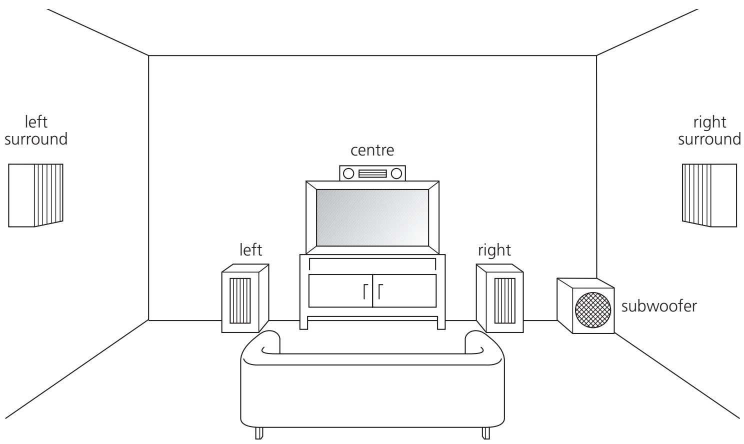


Figure 16.21
Dolby Digital 5.1 speaker set-up.

4 Digital Theatre System DTS — very similar to DD 5.1, with the advantage that digital compression of only 4:1 is used for supposedly greater fidelity. Typical DTS soundtracks are encoded with a 1.4 megabits per second bitstream, but of course require a separate decoder. The first DTS encoded DVD available in Australia was the hugely successful *Gladiator*, starring Russell Crowe as Maximus.

5 Dolby Digital Surround EX — again basically the same as DD 5.1 except that it includes an extra discrete sixth channel designed to be placed immediately behind the audience. It is like DD 5.1 enhanced with Dolby Pro-Logic: L–C–R–LS–RS–LFE–SC. The first DD-EX encoded movie was George Lucas's *Star Wars: Episode 1 — the Phantom Menace*.

6 Remaining systems worth mentioning in this context are the Sony Dynamic Digital Sound (SDDS) decoding, which is used primarily for large movie cinema sound reproduction, and the Motion Picture Experts Group (MPEG-2) encoding, which also can be used for DVDs.

You may also notice on recently released movies and DVDs the term **THX certification**. What this means is not another surround sound encoding system, but rather a set of performance standards established by the LucasFilm company and called the Tomlinson-Holman Experiment! This calls for particular functional and performance requirements from the audio equipment, such as decoders, equalisers, DVD players, amplifiers and speaker systems, both in professional cinemas (THX-Ultra for over 85 m^3 spaces) and home theatres (THX-Select for typical spaces of around 57 m^3). You know you have the ultimate in a home theatre system if you are able to achieve 'Home THX certification'. Now that would be something to brag about, but it would be very, very expensive.

The rear panel of a modern DVD player or home theatre amplifier contains a large number of connector jacks to allow for signals into and out of the device. Taking a typical DVD player, we might find at least the following:

- *Audio outputs:* $2 \times$ analog stereo L–R, RCA connectors colour coded as red for right and white for left. $6 \times$ analog 5.1 channel outputs, a single coaxial RCA digital output as well as a single optical digital output for the fibre-optic cable that is capable of carrying all six DD channels.
- *Video outputs:* single or double C-V (composite video) connectors, which provide lowest quality video signals directly to a television monitor; these are usually colour-coded yellow. One or two S-video outputs which provide good quality video signals to an S-video equipped monitor or data projector. The highest-quality signals are provided by the three separate RCA component video connectors, colour-coded red, green and blue. These provide RGB signals directly to the electron guns of the monitors or high-definition display devices.

Digital television, data projectors and plasma screens As well as surround sound systems, any good home theatre set-up will also provide the best in video displays, whether that be by normal large-screen television monitors, high-definition digital TV, wide-screen plasma panels or digital data projectors.

In Australia as from 2001 most TV stations began broadcasting some programs in digital format (wide-screen digital) as well as the normal analog PAL format. A fully digital TV monitor, or at least an electronic set-top box (STB decoder, similar to the currently available analog cable TV), is required to view these transmissions. The Australian government will phase out all PAL analog television by about 2008, so by then we will all be watching it. The main advantages of this digital technology will be greatly improved picture quality, which will be about the same as current PAL DVDs for the standard form (**SDTV**) and even better for high definition (**HDTV**). Television reception will be improved, as will the availability of surround-encoded sound formats such as DD 5.1 and the ability to include caption, subtitles and multi-angle viewing for sporting events etc.

The video or picture information is encoded using the MPEG-2 digital compression format at two resolution settings, being either SDTV = 576i = interlaced scanning at 50Hz, 576 active scanning lines and 720 pixels per line (720×570 pixels screens) or HDTV = 1080i = interlaced scanning at 50 Hz, 1080 lines at 1280 pixels per line (1080×1280 pixel screens). SDTV can be broadcast in either normal 4×3 ratio screen dimensions (letterbox format) or the 16×9 ratio (wide-screen); HDTV will be available only in wide-screen. Consumers will start to notice SDTV, HDTV and STB devices appearing in retail stores over the next few years. Good websites from which to discover more about the Australian digital TV scene are either Robert Simons at <http://www.digitaltv.com.au/index.html>, or the Digital Broadcasting Authority at <http://www.dba.org.au/>, or even the newsgroup aus.tv.digital. Take a look.

Another method of displaying video information is with **video or data projectors**. These are typically used where larger image sizes are required. Two main techniques are used, called LCD (liquid crystal display technology, better for smaller screens) and DLP (digital light processing technology, more suitable for larger cinema screens). George Lucas, in premiering his digitally produced *Star Wars: Episode 1 — The Phantom Menace*, called for DLP projectors to enhance the visual experience.

LCD projectors work by splitting the light from the projector lamp into three primary colour beams — red, green and blue (R-G-B). Each beam then passes through a small LCD panel which acts like an electronic slide. Each LCD panel typically has 800×600 pixel elements, and can be switched on and off according to the video recorded digital signal, to provide a correct full-colour image when the beams are recombined and passed through the main projector lens onto a screen. LCD panels and the transistors used to switch them on and off can produce quite a lot of heat, and the final image can look very pixellated on the screen due to the spacing between pixel elements on the LCD panels.

DLP projectors developed by the Texas Instrument semiconductor company are based on TI's digital micro-mirror devices (DMDs invented by Dr Larry Hornbeck in 1987), which are basically large-scale integrated circuit chips that contain a huge array of micro-miniature aluminium-coated mirror elements that pivot back and forth under the control of a separate digital signal processor (DSP). One mirror exists for each pixel element, which has an area of only 16 μm^2 at gaps of 1.0 μm; hence a 1024 $\times$ 758 (XGA) resolution DMD will contain 786 432 mirrors. Also, micro-torsion bar tilting of the mirror pivots is done with electric fields so that each mirror element is pointing (or not pointing) at the screen. DLP projectors can use one, two or three DMD devices. The three-chip approach (each separately handling R-G-B information) is the best, and the large expensive HDTV DLP cinema projectors contain DMDs that provide 1 310 720 mirror elements, giving S-XGA resolutions of 1280 $\times$ 1024 pixels. Texas Instrument specifications suggest that the DMD mirror elements switch at a rate of 5000 times per second. Not bad? Figure 16.22 indicates the DMD device structure as well as the two-chip DLP set-up.

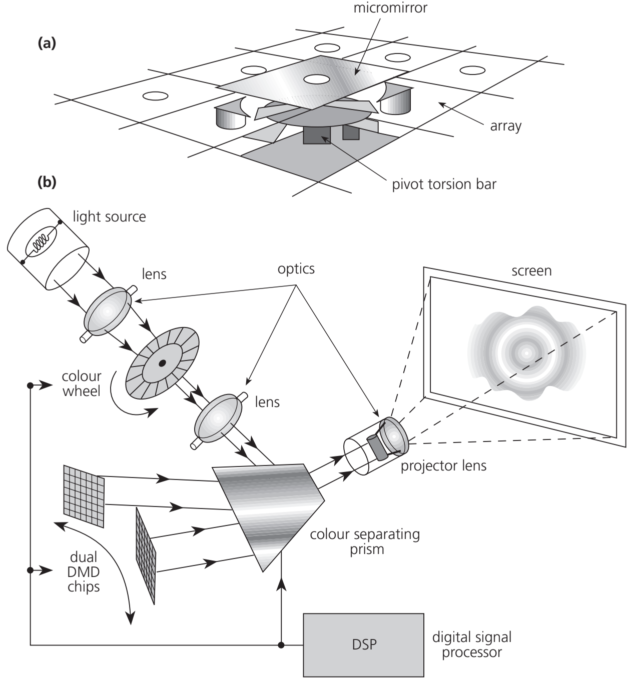


Figure 16.22
(a) DMD device.
(b) Two-chip DLP projection system.

A recent addition to the armoury of possible display devices is the plasma screen displays. These devices can be very thin in design and can easily accommodate large wide-screen video formats. Plasma screens do not use cathode ray tubes (CRTs) or LCD-DLP projectors, instead creating images by using an array of cells that receive a constant flow of low-pressure neon and xenon gas (hence the use of the word plasma). The cells are arranged in a rectangular matrix between sheets of thin glass and are covered with electrodes. When the electrodes are fired, the voltage stimulates the gas to emit UV light in a similar way to a fluorescent light tube. This UV light is then converted to visible coloured light by hitting phosphor coatings on another layer. Each cell is restricted to one particular R-G-B and each pixel that makes up the display image has three different cells, one for each colour. Plasma screen displays are still quite expensive, just like DLP projectors, but as with normal consumer market forces their prices will come down as manufactured numbers increase and more consumers start using the technology.

One final point worth noting concerning the home theatre sound technology improvements is that your lounge or living-room design might not be up to scratch for the best viewing and audio experience. You may very well research this point, making use of acoustics ideas from the next section. We're sure, though, that your parents will be quite willing to pay for an upgrade to your own study room so that you may have the best conditions when studying for your next physics paper. Right?

Activity 16.15 RESEARCH ON HOME THEATRE TECHNIQUES

1 The electrical signal sent from the amplifier to the speakers is sent through a filter to select the correct range of frequencies for each of the different speaker types. Discuss the construction of a filter and explain the physics principles involved. Compare and contrast a low-pass and a high-pass filter.

2 The circuit for a Dolby Digital decoder is not available, because the exact decoding process is strictly an industrial secret and needs to be added by manufacturers without alteration. Try to report on circuits for other types of surround sound decoders that are more freely available.

3 Three characteristics of a sub-woofer are: **(a)** you only need one for a stereo system, whereas other speakers handle left- and right-channels individually; **(b)** it doesn't matter where you put it; **(c)** it usually has its own power supply. Explain the physics behind each of these design features.

ROOM ACOUSTICS 16.11

Acoustics, the study of sound, sound technology and its effect on humans, plays an important part in the design of rooms, auditoriums and theatres, especially those used for high-quality performances. Walls, floors, ceilings all cause reflections of sound. These reflections may cause deterioration of the performance. However, some reflection is required — a listener would have trouble hearing if sounds were received from the source alone. These reflections affect the reverberation time and thus the acoustical quality. The **reverberation time** is the time it takes for the sound intensity to fall to one-millionth of its original intensity; that is, to fall by 60 dB. In lecture theatres, concert halls, etc. it is an important consideration. Multiple reflections are undesirable in lecture theatres because they obscure the spoken word. It is therefore desirable that the reverberation time be less than 1.0 s. Reflections are more desirable in concert halls as we want the listener to be totally immersed in the sound. It is therefore desirable that the reverberation time be of the order of 2.0 s. The reflections and reverberations lift the intensity of the sounds, but no single reflection should arrive at an ear later than one-twentieth of a second after the original sound or it will be heard as an echo.

The reverberations in a room depend on the size and shape of the room and the way the sound-absorbing linings of the room reflect or absorb sound. The absorbing quality of materials varies. Table 16.7 indicates the absorbing quality of several common materials. However, these qualities also depend on the frequency of the sound.

Table 16.7 SOUND ABSORPTION QUALITIES OF SOME COMMON MATERIALS

MATERIAL	SOUND ABSORPTION QUALITIES (PERCENTAGE OF INCIDENT ENERGY ABSORBED)
Glass window	4
Plasterboard	10
Carpet	25
Thick wool over brick	70

THE DOPPLER EFFECT 16.12

Everyone has observed the variation in frequency of sound from a police car, ambulance or fire engine as it rushes past. As the vehicle is approaching, the frequency of the sound of the siren is higher and at the moment it passes the frequency drops. This apparent change in frequency due to the object's motion is called the **Doppler effect**, and is attributed to an Austrian physicist and mathematician, Christian Doppler (1803–1853), who first investigated this phenomenon. This frequency change can be noticed if either the source of the waves is moving toward or away from a stationary observer, or the observer is moving toward or away from the stationary source of the waves.

Figure 16.23 will help to explain this effect.

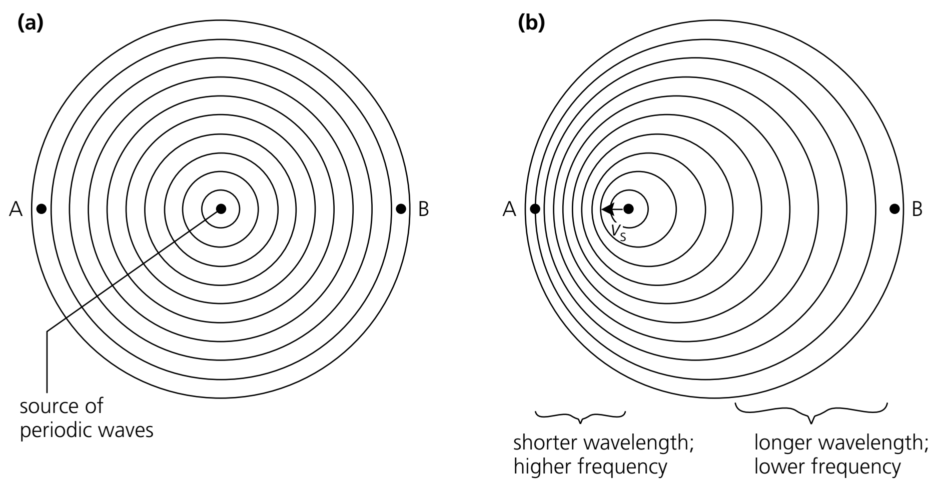


Figure 16.23
The Doppler effect. When the source of sound is moving toward the observer the waves are compressed.

Figure 16.23**(a)** shows the wave pattern produced by a stationary source. The waves are equally spaced and will arrive at points A and B at regular intervals. Now if the source is moving toward A (Figure 16.23**(b)**), the waves will be closer together in the direction of motion than if the source was not moving.

For example: Let the velocity of the waves be 10 m s^{-1}, the velocity of the source be 5.0 m s^{-1}, and the frequency of generation of the waves be 10 per second (10 Hz). If the source was not moving, after 1 second 10 waves would be produced and the furthest one would be 10 m from the source. The wavelength would be 1 m. However, if the source was moving at 5.0 m s^{-1}, the source would have moved 5.0 m and the 10 waves would exist in the 5.0 m between the source and A. The wavelength would be 0.50 m. The first wave would still have moved 10 m to point A. The waves would still have the same velocity as they would still be in the same medium, but the wavelength would be shorter and therefore the frequency increased.

For an observer at point B: After 1 s the distance between the source and B would now be 15 m with 10 waves between the source and B. The wavelength would be 1.5 m. This is a greater wavelength therefore the frequency would be lower.

For Figure 16.23**(a)** the wavelength is given by the equation: $v = f\lambda$ or $\lambda = v/f$, where λ is the wavelength; f is the frequency; v is the velocity of the wave.

Now for Figure 16.23**(b)**: the period of the wave $T = 1/f$ and the velocity of the source is v_s. After a certain time (t), after the production of n waves, $t = nT$: the distance between the source and point A is the difference between the distance the waves have travelled and the distance the source has moved.

This distance is:

$$\begin{aligned} d - d_s &= vt - v_s t \\ &= (v - v_s)t \\ &= (v - v_s)nT \\ &= (v - v_s)n/f \end{aligned}$$

NOVEL CHALLENGE

The frequency shift effect was first proposed by Doppler in 1842 but the first experiment was not done until French scientist Buys-Ballot had a go in 1845. He arranged for a carriage full of brass musicians to go past him in a train as they blew a steady note. *To study this effect quantitatively, what sort of measuring devices would be needed?*

NOVEL CHALLENGE

A motorcycle horn emits a note of 400 Hz when stationary. If a motorcyclist approached a wall emitting a 400 Hz sound and this was reflected back, what pitch would a stationary observer hear (higher, lower, the same)? What would the cyclist hear? If the motorcycle was moving at 20 m s^{-1}, calculate both of these frequencies.

The new wavelength is this distance divided by the number of waves:

$$\frac{d - d_s}{n} = \frac{(v - v_s)n}{fn}$$
$$\lambda' = \frac{v - v_s}{f}$$

Since $v = f\lambda$
Then:

$$\lambda' = \frac{v - v_s}{f}$$
$$\frac{v}{f'} = \frac{v - v_s}{f}$$
$$f' = f\frac{v}{(v - v_s)}$$

For our previous example:

$$\begin{aligned} f' &= f\frac{v}{(v - v_s)} \\ &= 10 \times \frac{10}{10 - 5} \\ &= \frac{100}{5} \\ &= 20 \text{ Hz} \end{aligned}$$

This is a higher frequency.

Similarly for point B, as the source is moving away the new distance becomes $d + d_s$. Therefore:

$$f' = f\frac{v}{(v + v_s)}$$

A similar analysis can be carried out with the observer moving instead of the source, producing an apparent frequency of:

$$f' = f\frac{(v + v_o)}{v}$$

if the observer is moving towards the source, and:

$$f' = f\frac{(v - v_o)}{v}$$

if the observer is moving away from the source.

One equation can be used for all situations, using a positive or negative to take account of the relative motion of the source or observer. This equation is:

$$f' = f\,\frac{(v \pm v_0)}{(v \pm v_0)}$$

where f' is the apparent frequency; f is the frequency of waves produced by the stationary source; v is the velocity of the waves; v_s is the velocity of the source; v_0 is the velocity of the object.

Example

A super-train moving past a station at a speed of 180 km h^{-1} (50 m s^{-1}) sounds its whistle as it comes into the station. If the frequency of the whistle on a stationary train is 320 Hz, what would be the frequency heard by the station-master standing on the platform if: **(a)** the train was approaching the platform; **(b)** the train was moving away from the platform? (The velocity of sound in still air is 341 m s^{-1}.)

Solution

(a) The observer is stationary ($v_0 = 0$). The source is moving toward the observer, therefore:

$$f' = f\,\frac{(v \pm v_0)}{(v \pm v_s)}$$

$$f' = f\,\frac{v}{(v - v_s)}$$

$$= 320\;\frac{341}{341 - 50}$$

$$= 375\text{ Hz}$$

(b) The observer is stationary ($v_0 = 0$). The source is moving away from the observer, therefore:

$$f' = f\,\frac{v}{(v + v_s)}$$

$$= 320\;\frac{341}{341 + 50}$$

$$= 279\text{ Hz}$$

Rule:

Source moving, observer stationary When the source moves toward the observer, the frequency is greater (toward equals greater) which requires a negative (–) sign in the denominator. If the source moves away from the observer, a plus (+) is used in the denominator.

Observer moving, source stationary When the observer moves toward the source, the frequency is greater (toward equals greater) which requires a plus (+) sign in the numerator. If the observer moves away from the source, a negative (–) is used in the numerator.

Some practical examples of the Doppler effect

- Astronomers use the Doppler shift of light frequencies to measure speeds of distant galaxies.
- Physicians can detect heartbeats of a foetus by means of a Doppler shift of ultrasound.
- Police radar units use the Doppler effect to measure the speed of cars, baseballs and cricket balls.
- Soldiers can tell if a rocket is coming toward them or going away by listening to the Doppler shift. The loudness enables them to also estimate this distance.

NOVEL CHALLENGE

Subsonic bullets don't emit a loud 'crack' when fired. What advantage would this be? Do any hand guns have subsonic bullets?

Questions

25 State what a listener observes about the apparent frequency of a sound source when the following occurs. Complete a copy of Table 16.8 but do not write in this book.

Table 16.8

LISTENER	SOURCE	APPARENT CHANGE IN FREQUENCY
Still	Approaching	
Still	Receding	
Moving away	Stationary	
Moving toward	Stationary	

26 Students celebrating the finish of Year 12 drive along the street at a speed of 60 km h^{-1} while sounding a whistle that has a frequency of 1200 Hz. Other students standing on the side of the road hear the noise as the car approaches and goes away. What is the apparent frequency of the whistle: **(a)** as the car approaches; **(b)** as the car goes away? (The speed of sound is 330 m s^{-1}.)

27 A police car's siren emits sound waves of 1000 Hz. If this car is involved in a car chase and is travelling at 120 km h^{-1} what frequency will a person on the side of the road hear: **(a)** as the car is approaching; **(b)** as the car is going away? **(c)** What frequency will the person driving the pursued car hear? (Assume this car is travelling at the same speed as the police car.) (The speed of sound is 340 m s^{-1}.)

The sound barrier

The speed of sound — Mach 1 (named after Austrian physicist and philosopher Ernst Mach) — is approximately 1200 km h^{-1} at sea-level and decreases as altitude increases. It is about 1050 km h^{-1} at a height of 11 000 m.

When planes fly faster than the speed of sound they are said to break the sound barrier. As they approach the speed of sound, sound waves compress in front of the plane (Doppler effect), which results in the formation of a shock wave.

As the plane breaks through the sound barrier the shock wave is left behind within a 'noise cone'. Within this noise cone the waves emitted in a forward direction accumulate and constructively interfere to make a very large amplitude disturbance. When this noise cone reaches an observer a loud 'bang' is heard. This is known as a **sonic boom** and results in large acoustic pressures.

In 1947, the first experimental piloted aircraft to break the sound barrier was the Bell X-1 powered by a four-chambered liquid-rocket engine and launched in the stratosphere from the underbelly of a flying bomber.

As planes began to break the sound barrier in the 1940s their design changed. As planes approach the sound barrier the air surrounding the plane becomes 'harder' to fly through and the resistance increases greatly, making planes unstable. This resulted in the deaths of many test pilots in the 1940s when the ambition to break the sound barrier was paramount. Supersonic aircraft need to be much sleeker with pointed noses; for example, the Concorde, which could fly at 2000 km/h. Although the Concorde was supersonic, it was also expensive to operate and passenger confidence never recovered after its crash in France in 2000. The last flight of the 12 remaining Concordes was in 2003. New commercial airlines of today focus on fuel economy, quietness and automation, instead of speed. Greater safety, increased reliability, less noise and pollution, better passenger comfort, more navigational aids and less room for pilot error are all guidelines for the commercial airplanes of tomorrow. Supersonic planes are no longer on the drawing boards of any major manufacturer.

Photo 16.4
The Concorde was the only supersonic commercial aircraft in operation. Notice its sleek design, which allows it to safely break the sound barrier.

Practice questions

The relative difficulty of these questions is indicated by the number of stars beside each question number: * = low; ** = medium; *** = high.

Review — applying principles and problem solving

(For all questions unless specified use v_{sound} = 340 m s^{-1}.)

***28** When sound waves travel through a medium, in which direction do the particles of the medium vibrate?

***29** A tuning fork produces 2.4×10^4 compressions and rarefactions in the air particles around it in 10 s. The distance between each compression is 0.14 m.

(a) Find the frequency of the tuning fork.

(b) What is the velocity of sound in air?

***30** In a thunder storm the lightning is seen before the thunder is heard, as the velocity of light is much greater than the velocity of sound. If the thunder is heard 10 s after the lightning is seen how far away is the storm? (The velocity of sound is 340 m s^{-1} at the current temperature.)

***31** Explain how sounds can have the same pitch but different qualities.

****32** Two speakers are placed 2 m apart and produce sound of the same frequency and in phase. A person walks across the front of the speakers as shown in Figure 16.24. In doing so she notes that the intensity of the sound goes down, then up again, then down again at point B 0.5 m from the centre point A. What is the wavelength of the sound emitted by the speakers?

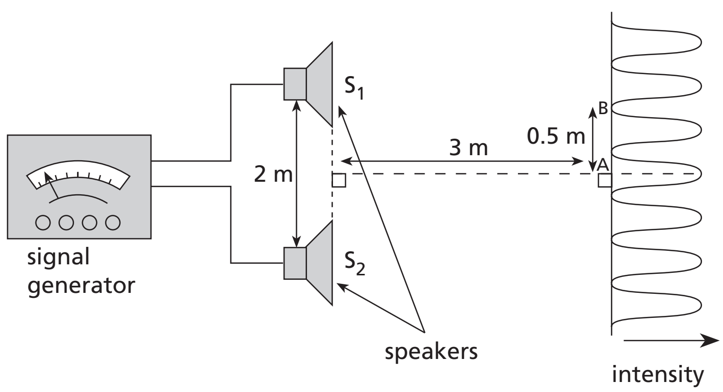


Figure 16.24
For question 32.

***33** A marine survey vessel plotting the contours of the ocean floor sends an ultrasonic wave and receives an echo back 1.2 s later. Calculate the depth of the ocean at this point. (The velocity of sound in sea water is 1400 m s^{-1}.)

***34** A student produces a note by blowing across the mouth of an open-ended piece of plastic pipe 0.2 m long. Calculate the frequency of the third harmonic.

***35** A 40 cm organ pipe is open at both ends. If air is blown over one end what is the fundamental frequency emitted from this pipe?

****36** Open-ended and closed-ended pipes can produce the same fundamental frequency.

(a) Calculate the fundamental frequency produced by a closed-ended pipe of length 25 cm.

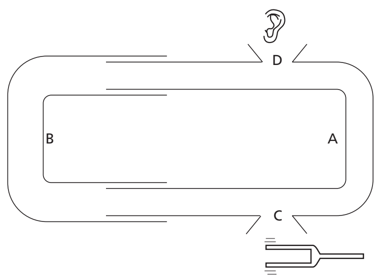
(b) Calculate the length of the open-ended pipe that would produce the same fundamental frequency.

(c) Even though they both produce the same fundamental frequency they would sound different. Explain with calculations why this occurs.

**37 A pipe is open at both ends and is 0.58 m long.
(a) Determine the wavelength of the sound that would produce the fundamental frequency in this pipe.
(b) Calculate the frequency of the second and third overtones.
(c) What harmonic is the fourth overtone?
(The speed of sound is 342 m s^{-1}.)

**38 One method of measuring the speed of sound is to stand a distance of, say, 300 m from a wall in an open area and bang two pieces of wood together while timing and listening for the echo. When the echo is heard bang the wood together again. Continue to do this, say, 10 times, and then stop the timing. Knowing the distance to the wall and the time, the velocity of sound can be calculated. If the distance to the wall is 300 m and the time measured from the first noise to the last echo was 17.5 s, calculate the velocity of sound.

Figure 16.25
For question 39.



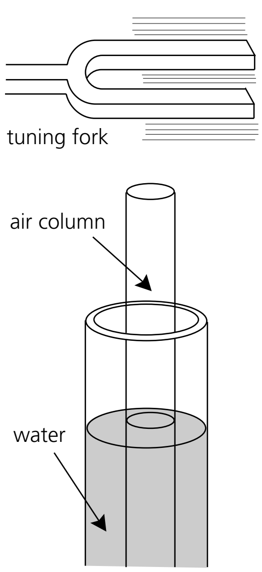
**39 The apparatus in Figure 16.25 can be used to find the speed of sound. Section A is fixed while section B is movable. A tuning fork of known frequency is sounded over one opening, C, and you listen over the other opening, D. If you make the lengths of A and B equal, a maximum of intensity is heard at D, but if B is slowly moved out a minimum of intensity, then a maximum, then another minimum will be heard.
(a) Explain why this happens.
(b) When B is moved out a distance of 21 cm the first minimum of intensity is heard. Calculate the speed of sound. (The tuning fork used had a frequency of 400 Hz.)

40 Draw diagrams to show the standing waves set up in a harp string of length *L* when the length of the string corresponds to **(a) two wavelengths; **(b)** three and a half wavelengths; **(c)** four wavelengths. **(d)** Calculate the frequency of the sound emitted from this string in each of part **(a)**, **(b)**, and **(c)**, when the length of the harp string is 0.60 m, the mass of the string is 20 g, and the string is under a tension of 120 N.

**41 Students experimenting with musical notes set up a row of test-tubes in a test-tube rack. They then fill them to various levels with water. By blowing across the top of the tubes they are able to create different notes. If the distance between the top of the water and the top of the test-tube for the first one is 8.0 cm:
(a) calculate the fundamental frequency emitted by this tube;
(b) describe what the students will need to do to create different frequencies, and calculate the distances from the top of the tubes to the water if they wish to create fundamental frequencies of **(i)** half that in part **(a)**; **(ii)** twice that in part **(a)**; **(iii)** three times that in part **(a)**.

**42 Discuss the possibility of open-ended organ pipes producing different frequency notes on hot or cold days.

Figure 16.26
For question 43.



Extension — complex, challenging and novel

***43 An open tube is placed into a container of water and a vibrating tuning fork placed over the mouth of the tube. (See Figure 16.26.) As the tube is raised so a greater length of the tube is out of the water, resonance is heard. This occurs when the distance from the top of the tube to the water level is 12 cm, and again at 50 cm. Determine the frequency of the tuning fork.

***44 To find the frequency of an unknown tuning fork (Z), two tuning forks (X and Y) of known frequency are used. X has a frequency of 245 Hz and Y has a frequency of 247 Hz. When Z is sounded with X, 30 beats are heard in 10 s. When Z is sounded with Y 10 beats are heard in 10 s. What is the frequency of Z?

*****45** A 442 Hz tuning fork is sounded at the same time as the A string of a guitar. A beat frequency of 50 beats per 10 s is heard. If a rubber band is wrapped tightly around one prong of the tuning fork and this is then sounded at the same time as the guitar string, a beat frequency of 30 beats in 10 s is heard. What is the frequency of the guitar string?

*****46** A ferry crossing the river at 10 km h^{-1} sounds its whistle as it approaches the jetty. Passengers on board the ferry hear two whistles — the whistle itself and its echo from the rock face behind the jetty. They appear to be different. What is the frequency of the reflected sound if the frequency of the whistle is 480 Hz and the speed of sound is 330 m s^{-1}?

*****47** In 1845 the Dutch meteorologist Christoph H. D. Buys Ballot first tested the Doppler effect by having two trumpet players play a musical note of 440 Hz, one on a moving flatcar of a train and the other on the station platform. While they were playing the note, a beat frequency of 3 beats per second was heard by a person at the station. How fast was the train moving?

*****48** To detect the location of cannons on a battlefield, the army used a method called triangulation. Several microphones would be placed in a straight line at 1 km intervals along the 'front line'. When the cannon noise was detected by the closest microphone, a timer would start and the delay for the second and subsequent microphones would be recorded on a paper roll. If three microphones 1 km apart were used and there was a time delay of 1.2 s to the second microphone and 3.7s to the third microphone, determine the position of the cannon (there may be more than one answer). A similar process is used to detect the epicentre of earthquakes.