Interesting and inexpensive experiments for high school physics.

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Input and constraints. This set of experiments responds to requests from teachers of physics in the state of New South Wales for experiments to address some areas in the new syllabus. Given that one wants to have several sets of gear so that students can work in small teams, the implicit budget constraint is severe. We have tried to respect that. Some of the components will be in labs (rulers, watches), others may be readily bought or borrowed (eg transistor radio), others are readily and cheaply available from electronics or hardware stores. A few (transformer cores, transformer windings) are more difficult to buy in small quantities, but are available relatively cheaply in bulk.

The NSW syllabus requires students to use electronic data collection. This is also a good idea. Most of these experiments therefore use a computer for data acquisition.

Computers and sound cards as oscilloscopes.

Most schools have several computers (even if they're not in the physics lab.) Further, old computers are very cheap and may be acquired by donation or by bidding at auctions. Nearly all computers built in the last several years have a (stereo) sound card whose input side is a pair of reasonably high quality analog to digital converters (ADCs). Input to the ADCs is via a stereo mini-phono jack into a socket usually marked with a microphone icon, or another marked 'line'. The manufacture of sound cards is not standardised, but:

Typical specifications: Sample at rates including 44.1 kHz, both channels. Range about –1 V to +1 V. Sensitivity to rather better than mV with reasonable linearity. The line input can withstand several volts without damage. The microphone input has a preamplifier that is also fairly robust. The range of your card can be tested by inputting a sine wave and gradually increasing the amplitude until clipping (flattening of the extrema in the oscilloscope display) occurs.

Frequency range 20 Hz to several kHz. The high frequency limit is not a problem unless one would like to observe radio signals directly. The low frequency limit is a nuisance in some applications, but not in those described here. Because of the variability in manufacture, there is no preset voltage calibration, but this can easily be done with an oscillator and an oscilloscope or multimeter.

Free software for displaying voltage as a function of time and frequency V(t) and V(f) display is available from the net. These are quite powerful: you get a storage CRO, plus a spectrum analyser, plus editing and some averaging facilities. Understandably, the commercial versions are more powerful. We encourage fair use of shareware.

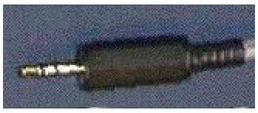
Free oscilloscope software downloads from http://polly.phys.msu.su/~zeld/oscill.html. It requires a 80486 or higher PC running Windows95 or later, or there's an older Windows 3.x version.

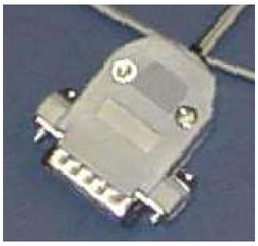
Free recording and editing software, which includes V(t) and V(f) functions among much else, is available from http://www.syntrillium.com/cooledit/ The free download is adequate for the experiments described here. It requires Windows 95/98/ME NT/2000/XP

[1] With many thanks to Gary Keenan, Jason Whittaker, Tamara Reztsova, Pritipal Baweja, Ken Jackson, Attila Stopic and John Tann who built sets of the equipment described here for a high school teachers' workshop held at the University of New South Wales in November 2002 .

Both software packages have **cursors for measuring** parameters and intervals on the display. This is a powerful feature and is included in virtually all modern electronic instrumentation, so it is worthwhile becoming familiar with it.

PC sound card → Oscilloscope conversion kits are not absolutely necessary for the experiments described here. The input of the sound card may be used directly. However, care must be taken not to use large signals (so as not to damage the card) and some very small signals may be improved by amplification. The conversion kits supply these, plus a high input impedance. They are cheap, they increase the range of signals, they make damage to the sound card less likely and they make the input look more like an oscilloscope. The kits are sold in parts and require soldering, but no electronic skill, to assemble. http://www1.jaycar.com.au Stock code KA1811 A$30 (described in Electronics Australia August 1998).





The computer and sound card as a data logger.

Many of the most obvious applications are impossible because the cards do not have DC response. In principle one can short out the series capacitance, but this requires taking the sound card out and may cause damage. However some of the exercises described here use the computer as a data logger in AC mode. One could also amplitude modulate the input signal at an appropriate frequency. This limits the frequency range at the high end and produces an AC signals. This extra complication might have pedagogical disadvantages.

Sets of interface electronics with their own ADCs are available as data loggers. These are robust and in some cases designed for use in school labs. They are not discussed here because manufacturers have their own web sites and publicity information.

If you don't have computers.

Most of these experiments may be done with an oscilloscope instead, and a few are more convenient that way. However, it may be worth trying to find a commercial enterprise about to upgrade its computers, and offer an alternative to land fill.

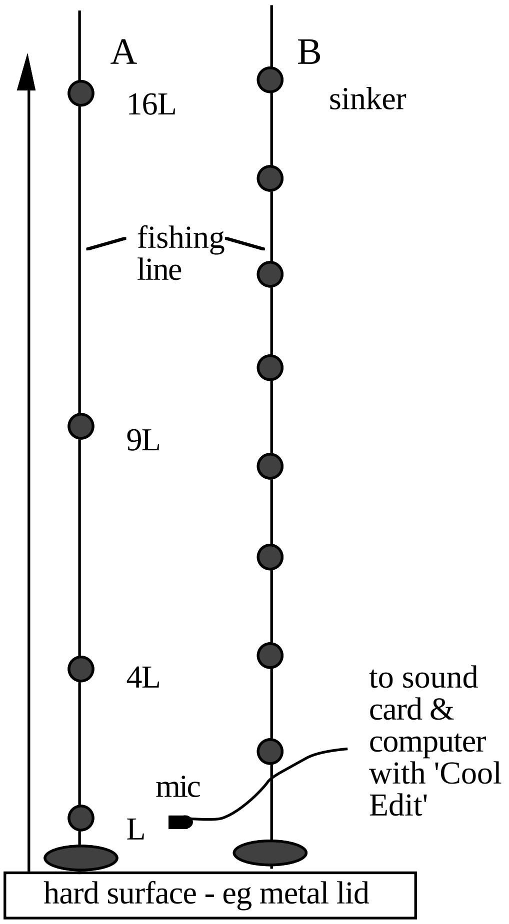
Microphones

are used in several of these experiments (even when sound is not the topic studied). 'Lapel microphones' are electret microphones that come complete with a 1.5 V battery powered FET pre-amplifier. These are high quality, sensitive microphones, and their cost is about A$20.

Assembly and bulk buying.

Some of the experiments described here require assembly (even if it is just making leads with the appropriate connectors). Others require materials that are considerably cheaper if bought in bulk than bought separately (cables, wire, magnets, transformer cores, components). Perhaps schools might wish to coordinate this, or perhaps the Board of Studies might wish to do it. If not, it might be possible for some students in the School of Physics to do it as a vacation enterprise.

Motion under constant acceleration



Aim: to determine whether acceleration under gravity is constant and, if so, its value.

Apparatus:

Essential: fishing line or thread, sinkers or other small weights, hard surface, ruler or tape measure.

Preferable: microphone, computer, soundcard, "Cool edit" software.

Alternative: stopwatch or clock with second hand.

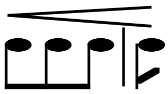
Preparation. Tie sinkers onto line. For the sound card method, they may be tied at arbitrary separations (eg equally spaced: B in diagram). For the alternative method, they should be positioned at distances n^2L from one end, n an integer and L on the order of 10-15 cm (A in diagram). 10 cm is more comfortable and makes the arithmetic easier, 15 cm is easier to count but requires standing on a desk.

Method B with 'Cool Edit'. Hold line with lowest weight just far enough away so that it makes a recordable sound. Start recording sound. Release line. Stop recording and expand the section containing the sounds of the weights hitting the surface fill the screen. Record times t of fall and heights h of fall for each weight.

Plot t *vs* $\sqrt{h}$ OR Plot *ln* t *vs* *ln* h.

Method A with stopwatch. Tie the weights at n^2L from one end and let them fall. Hold line with lowest weight just touching the ground. Let fall and listen carefully to rhythm of weights striking surface. If acceleration is constant –g and initial velocity is zero:

$y = y_o + v_o t - \frac{1}{2} gt^2$ so for y = 0, $gt^2 = 2y_o = 2n^2L$

$\therefore t_n = n\sqrt{\frac{2L}{g}}$ so we hear[2]: 

If the quavers are not even, then the either the weights aren't spaced at n^2L or v_o was not zero, or acceleration is not constant (or other possibilities).

Listen carefully to the rhythm. Count an integral number m of 2/4 bars at this rhythm (remember to start counting from zero) and record the time T. m may be several tens: most people have a sense of rhythm that is precise to a % or so over tens of bars. The time between successive falls is $(t_n - t_{n-1})$

$$\Delta t = \sqrt{\frac{2L}{g}} = \frac{T}{4m} \qquad \text{so} \qquad g = \frac{32m^2L}{T^2}$$

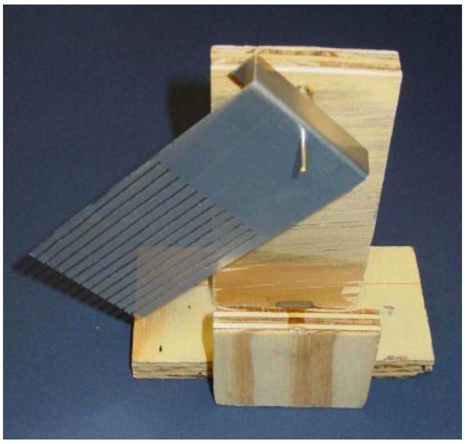
[2] A famous minimalist composer used this riff in almost every bar of a movement of his best known work. It would be an interesting musicological research project to enquire whether he lived near a physics teacher.

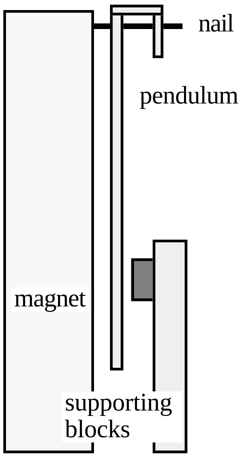
Eddy currents

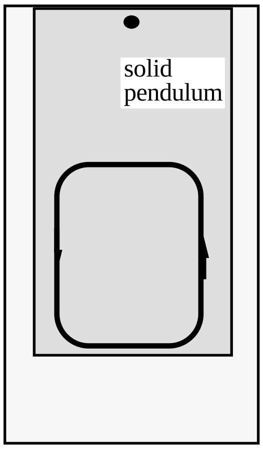
The NSW high school syllabus has quite a bit on eddy currents, and we were often asked for experiments.

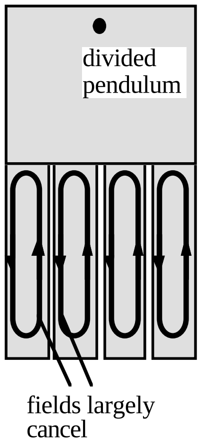
Aims: To demonstrate eddy current braking, to demonstrate the effect of laminations on eddy current losses. Optional: to give an order of magnitude estimate of the latter effect.

Apparatus: Pendulum, divided pendulum, magnets, ruler, stopwatch. The pendula are made from aluminium sheet with two right angle bends and two holes to clear a nail upon which they are suspended. A hole in the wooden block allows the rare earth magnet to be positioned near but not touching the pendulum.









Experiment: Measure the time taken for the amplitude of oscillation of the pendulum to fall from a given initial amplitude (measured with a ruler at the release point) to another given point, eg, the point at which the pendulum is no longer visible from behind the block, sighting along one edge.

Compare the times taken for (i) solid and (ii) divided pendula with no magnet. (Is the mass of the pendulum important? Would you expect air resistance to be different? Important?)

Compare the times taken for (i) solid and (ii) divided pendula with magnet.

Vary the distance between magnet and pendulum.

Optional 1: The Earth's field is weak (~0.1 mT. At Sydney, it points mainly upwards, with a component from South to North.) Can its effect be seen? Use the solid pendulum swinging NS (~ no flux) and EW (~ maximum flux for a vertical plane). Warning: frictional effects are often irreproducible: repetition is required.

Optional 2: The slits give rise to greater turbulent drag. Fill the slits with epoxy and sand it smooth.

Optional 3: In most cases, including here, it is not easy to be quantitative about eddy currents because of complicated geometries. For the purposes of analysis, let's neglect friction (ie forces due to eddy currents are the only dissipative force). For angular velocity ω,

$$I\frac{d\omega}{dt} = \text{ torque} \propto \text{eddy force} \propto \text{eddy current} \propto \text{induced emf} \propto \frac{d\phi}{dt} \propto \omega$$

$\therefore$ ***if*** *frictional losses are neglected*, we expect damped simple harmonic motion,

$\therefore$ the time for the amplitude to decay by a given fraction would be constant.

In the two diagrams shown above, if currents were the same, we should expect comparable forces and torques, because of the cancellation of pairs of currents, as indicated, in the divided pendulum. However, the area and therefore the flux is reduced by ~ number of laminations. Further, the cross sectional area for current flow is reduced (calculation not simple), so resistance is increased.

Further, the field with the magnet present is not uniform. So it would be a tedious calculation to determine the lamination effect here. However, we should expect that the force should be decreased by a factor exceeding the number of laminations.

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Transverse waves, Faraday's law, generators, harmonics etc.

Aims: To measure the relative transverse velocities in standing waves in a string. To measure Faraday emfs in a wire moving in a magnetic field, to examine harmonics in a string.

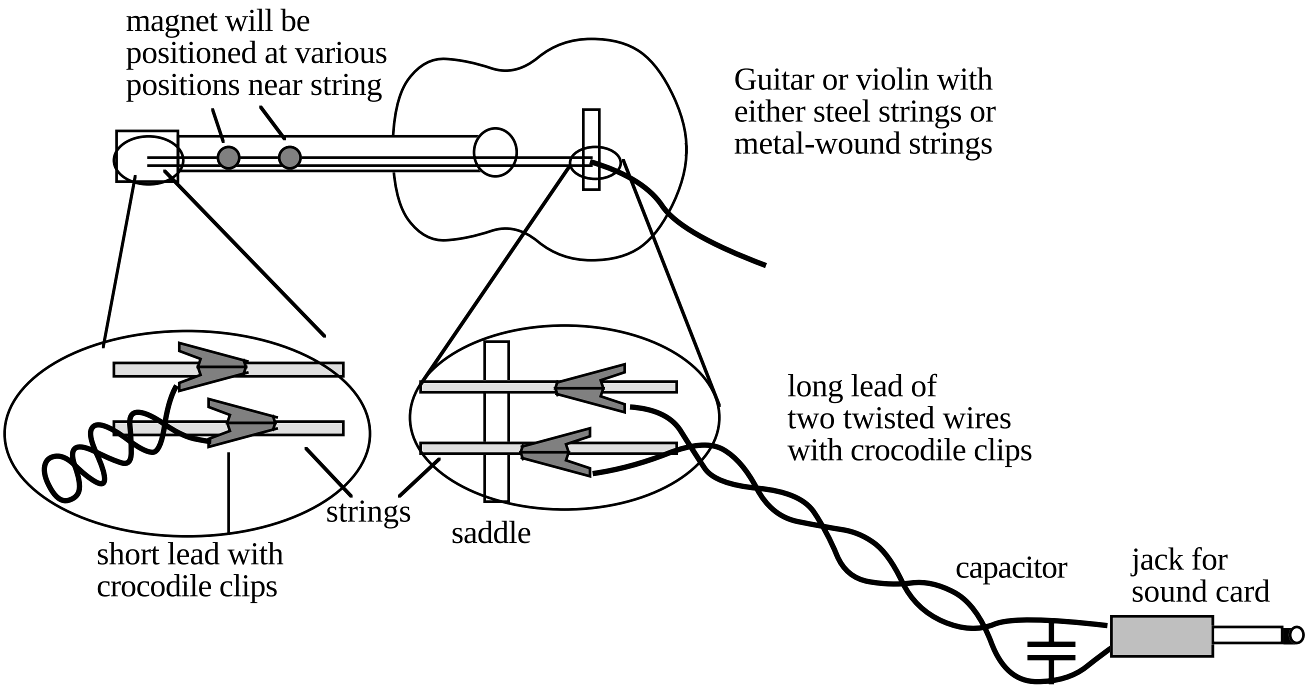
Apparatus: Instrument with either metal strings or two metal wound strings[3] (eg guitar or violin).

Computer with sound card and either 'Cool Edit' software or 'Oscilloscope' software. (CRO could be used instead, but unless it is a storage CRO with DFT function, the software version is preferable.)

Magnet: You must be able to arrange to have the field at right angles to string and to string motion. A strong one[4]. Total flux (not just flux density) should be large: a big magnet is better than a small one with the same field, provided the width of the magnet is much less than the length of the string. One short lead with crocodile clips. (If not short, twist it to reduce stray magnetic flux.) One long lead made from two (insulated) wires twisted together, with crocodile clips on one end and a jack for the sound card on the other. A small capacitor (ceramic, ~10 to 100 nF, the value is not critical) is connected across this cable, preferably at the jack end.

Optional: a bow and rosin. (Do not over tighten bows. Relax tension after use.)

Optional: an amplifier and speaker, or an external input to a ghetto blaster etc.



[3] Note: if metal wound strings are very worn, the winding may not make a complete circuit. This experiment may cause minor damage to the strings, depending on the strength and sharpness of the crocodile clips, but only in places where they rarely break or wear. It is unlikely to break a string or to reduce its performance.

[4] Depending on the strength of your magnet and the noise in your soundcard and elsewhere, this experiment might require the extra gain of a CRO adaptor or a real CRO. But try to get a big, strong magnet.

Preparation: The two strings form part of the circuit. We shall only vibrate one of them. Adjacent strings are better to reduce electrical interference. The short lead shorts the two strings. It is attached to the non-vibrating sections of the strings between the nut and the tuning pegs. The long lead is attached to the non-vibrating section of the string between the saddle and the attachment point of the strings (tailpiece on violin, hole in bridge on guitar). Careful not to short here: insulated crocodiles or use tape. The capacitor is to short out high frequency noise, mainly radio. **Tip**: if there is too much 50 Hz or 100 Hz noise in the absence of a vibrating string, then try moving away from motors or high power appliances and try different orientations of the instrument.

Method. Pluck or bow ***one*** string so as to induce respectively transient or sustained oscillations. (It is usually more convenient to bow near the nut, which is taking *sul tasto* rather literally.) Pluck or bow so that the vibration is at right angles to the magnetic field. The field is only large in the vicinity of the magnet. If the transverse speed as a function of the position on the string is v(x), then the emf is

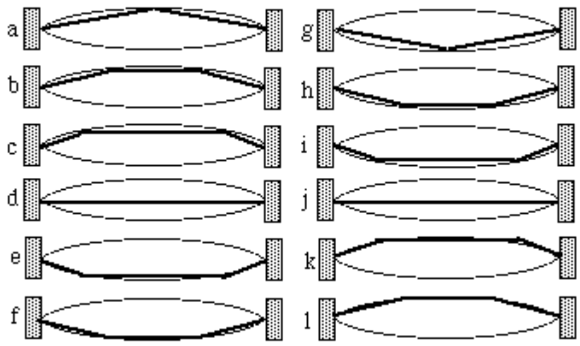
$$V = -\frac{d\phi}{dt} = \int |\mathbf{B}\times\mathbf{v}|\, dx \cong v \int_{\text{near magnet}} B\, dx \equiv v(x)B_{eff}L$$

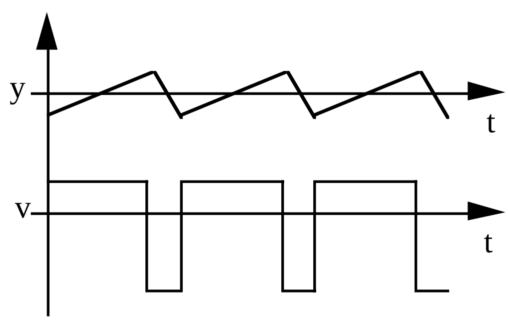
where B_{eff} is the effective field over a characteristic length L which defines the region over which it is large, probably 2-4 cm for a rare earth magnet, larger for others.

To the extent that we can control the amplitude induced by plucking or bowing, we can now measure v(x) by moving the magnet and repeating the experiment. (One can only measure at one position at a time, of course.) A way of standardising plucking is to pull the string a given displacement or tension with a cotton thread which is then cut. It is relatively easy to bow a string consistently, especially if a string player can show how. (To get from v(x) to displacement, one can simply integrate using an RC filter on the input of an oscilloscope. See http://www.phys.unsw.edu.au/~jw/RCfilters.html)

Some discussion.

In a plucked string, the high frequencies die away quickly, leaving only the fundamental, which is sinusoidal in both position and time. However, using the 'stop' or 'store' facility in the software, you may observe the initial, harmonic rich signal. (The signal from the low harmonics will be weak when the magnet is near the nut or bridge. that from the n^{th} harmonic will be weak or absent when the magnet is at a rational fraction m/n along the length.) In a bowed string, steady motion is possible.





The figure at left shows successive positions of string plucked at the centre. The fine line is the envelope of the motion. What one 'sees' resembles more closely the envelope than the real shape, because the string is instantaneously stationary at extrema of its motion, and so reflects more light. The figure at right shows the idealised displacement and velocity at a point approximately one quarter of the way along a bowed string. Because of the capacitor, sharp corners will not be observed in this experiment. Further, there will be small waves present due to the finite thickness of the strings. In both cases, a strobe light would show the 'instantaneous' shape of the string if the strobe is tuned close to the fundamental frequency.

Harmonics.

Touch the string lightly at a point 1/n of its length from the end (where n is 1, 2, 3 etc), then bow the string close to the end. Alternatively, touch the string very lightly at a point 1/n of its length from the end, pluck the string close to the end and release the first finger as soon as you have plucked. Touching the string produces a node where you touch, and so you excite (mainly) the mode which has a node there. You will find that you can play bugle tunes using harmonics two to six of a string. Here it is useful to use the **frequency analysis** option in the software to look at the **frequency spectrum**.

Optional.

1. Use a stereo-double mono adaptor to plug a microphone into one channel and this circuit into the other. One now has string motion on one channel and sound on the other. Somewhat crudely, this measures the acoustic efficiency of the guitar/violin as a function of frequency.

Connect the jack to an amplifier and speaker (or the external/mic input on a ghetto blaster/karioke box) and you have an electric guitar/violin. Because of the differential dependence of the harmonic amplitude on x (discussed above), putting the magnet near the bridge or nut produces a brighter sound, compared to the mellow sound obtained with the magnet near the centre. (Standard guitar pickups also work on string speed, but they use the perturbation by the ferromagnetic string of the inhomogeneous field of the pickup.)

More information:

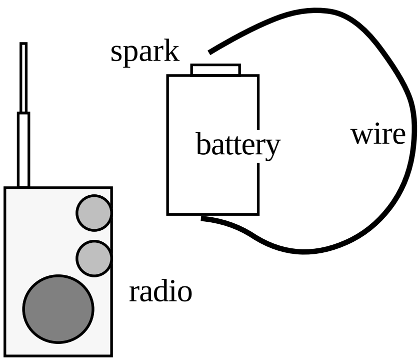
For more information about plucked and bowed strings, including animated versions of the above diagrams, see www.phys.unsw.edu.au/~jw/strings.html and www.phys.unsw.edu.au/~jw/Bows.html.

Hertzian waves

Background: Many early experiments with radio used sparks as detectors and as sources of electromagnetic radiation.

Aims: To demonstrate the electromagnetic nature of radio waves, to demonstrate the decrease of intensity with distance from the transmitter, to investigate electromagnetic shielding.

Apparatus: Portable radio (preferably one with a visible antenna), fresh battery, piece of wire. ***Do not do this with a high current battery such as a car battery, or any other acid battery***. A 1.5 V "A" battery is fine.



Method. Tune the radio either to a station or between stations. It is worth doing both. The radio almost certainly has an automatic gain control and so will be more sensitive when tuned between stations. However the background noise—also broad band noise—will be stronger too.

Hold one end of the wire to one end of the battery. With the other end of the wire ***briefly*** scrape the surface of the other battery terminal, making sparks that will be visible in dim light. (***Do not maintain a short circuit for more than a few seconds. If the battery begins to feel hot, cease the experiment.***)

Listen for the broad band ('static') noise of the signal radiated by the plasma formed between the wire and the battery terminal. Vary the distance between the battery (the transmitter in this instance) and the receiver. Report any changes in the intensity of the broad band signal.

Try to 'shield' the receiver by putting it inside a container (try whatever materials come to hand: filing cabinet, carton[5], waste paper bin).

Discussion. One might expect an electrical conductor to shield, but the magnetic fields can produce rapidly varying currents in a conductor, and these radiate electromagnetic radiation. 'Mu metal' is a metal with a high value of magnetic permeability (and it is a conductor). Soft iron also has a high value of magnetic permeability.

Not optional. If an electron with an energy of 1.5 eV enters aluminium (work function 4.2 eV), what is the wavelength of the photon it might produce? Should you worry about X ray production from the sparks?

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Transformers

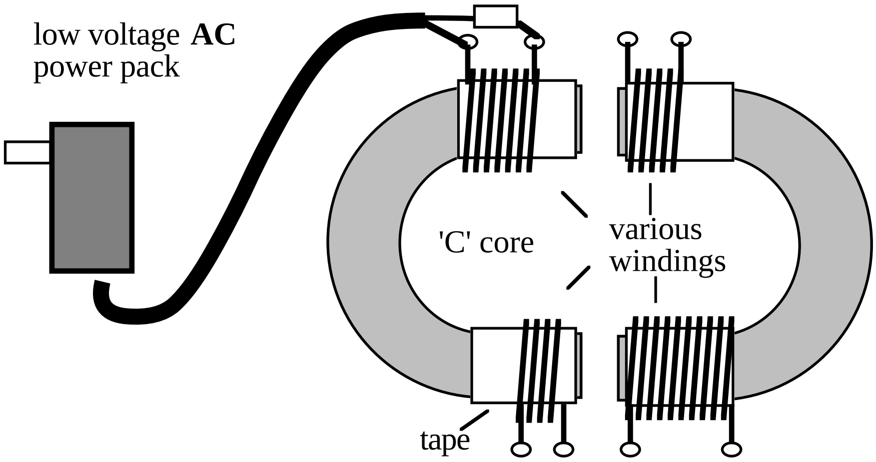
Warning: **It is possible, though not easy, to cause injury. Make sure that this experiment is done under close supervision.**

Aims: To demonstrate how transformers work.

Apparatus: Two transformer 'C' cores. Varnished (ie insulated) copper wire. Tape. AC multimeter. 10 V ac plug pack (). 10 Ω resistor rated at several watts (it should be physically large: at least several mm diameter[6]). Pack of wires with crocodile clips. Optional: oscilloscope or PC with Oscilloscope converter. (*Do not use sound card without the oscilloscope adaptor for this experiment: the risk of damaging the sound card is high. Attenuate with a resistive divider if necessary*).

Optional: several resistors in the range 10 Ω to 1 kΩ. The low value resistors should be high power ratings so they don't get too hot. Connecting wires.

Preparation: Tape around the C cores to protect the windings. Wind coils to choice, but caution: do not make any of the ratios large! eg, one might use 20 turns, 40 turns 40 turns and 60 turns. Put a 4k7 resistor in series with the primary for added safety.



Safety: We use a 10 Vac plug pack. The inductance from 40 coils on the core is sufficiently low that a 10 Ω resistor sufficed to reduce the voltage on our primary to 3 V. Note that this gives about 10 W dissipation in the resistor, so it needs to be rated at this power or greater so that it doesn't get hot. Even if students put all secondaries in series they are unlikely to produce an output of more than 30 V. This is unlikely to cause currents that can be felt unless applied to bodily orifices or via conducting electrodes applied to the skin.

[5] The penetration by EM rays of cardboard and many other materials depends on frequency. Cardboard was used in collimators for N rays (http://skepdic.com/blondlot.html).

[6] A 10W 10Ω resistor is readily made from, for example, 5 2W 47Ω resistors in parallel.

Sufficiently inventive students might manage to hurt themselves or others. For instance, if one takes two ac power packs and connects their outputs together, and plugs one into the wall socket, then the pins on the other are now near 240 V. Alternatively, pieces of wire plugged straight into the wall socket could be dangerous. And a 'C' core dropped on the head from a sufficient height could be fatal too.

Method: Using only one 'C' core, measure the terminal voltages on the two coils. Use either the AC range of the multimeter[7] or preferably the two channels of the oscilloscope.

Now get the second 'C' core and bring them into contact (careful: do not get your fingers pinched between the two cores). Now measure the unloaded voltages of all of the coils. Here we have a primary and three secondaries.

Optional: The secondary coils may be connected in series[8]. ***Warning:*** this is where the largest voltages may be produced.

Compare the voltage ratios with the turns ratios.

Measure the voltage between a point in the primary circuit and one in the secondary. (An important use of transformers is for isolation: there is no electrical connection between them.)

Optional: Leaving the primary connected to the power, take a simple piece of wire, wrap it around the core once, then connect it to an oscilloscope. Then disconnect it, put several turns on and reconnect it to the oscilloscope. I cannot speak for others, but I remember doing this for the first time as a schoolkid and being delighted by being able to produce a substantial voltage in a circuit that contained just a piece of wire.

Connect various resistances across one of the secondaries. It may be possible to measure the current with the AC meter, or it may be too small, in which case one has to calculate it from the voltage and Ohm's law. How ideal is the transformer[9]?

The 10 Ω resistor also makes current calculation easy: multiply the volts by 100 to get the current in mA.

Separate the two 'C' cores with layers of paper and comment on the ideality of the transformer.

The primary core makes an electromagnet. It is moderately strong, so be careful not to pinch fingers. You may hear the 100 Hz vibrations as the field turns on and off 100 times per second, especially if you try to pull the cores apart.

Short one of the secondaries. What happens to the voltages on the other secondaries? Why?

Calculate input and output currents by measuring the voltage across the series resistor on the primary or across a load resistor on the secondary. Partly because of the small number of turns and therefore relatively small inductance, this is far from an ideal transformer: the output power is rather less than the input power.

Discussion: Without the second core, the transformer is very far from ideal. Quite small gaps break the flux continuity and make it a non-ideal transformer.

See http://www.phys.unsw.edu.au/~jw/transformers.html

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[7] Note that the AC range of some multimeters, especially large, display models is inaccurate at voltages of order 1 V and less.

[8] In the transformers used in electronic devices, the secondary coil is 'tapped' at several points to provide, effectively, several secondary coils in series.

[9] Because the number of coils and therefore the inductance in this transformer is rather low, it is far from ideal, even when the flux is complete.

Standing waves. Reflection and transmission. Speed of sound.

Aims: To investigate sound radiation, reflection and transmission

Apparatus: Oscillator[10]. Speaker. No amplifier?[11]. Microphone. Metre rule. CRO or PC. Pipe with cap (Postpack tube or PVC pipe). The diameter of the pipe should be rather larger than the width of the ruler. Materials for sound absorption testing (felt, foam rubber, acoustic wool).

Note: For these experiments, the spectrum analysis option in the oscilloscope software may be very useful. Its difficult to estimate the amplitudes of two sine wave components in the time domain, but easy in the frequency domain. If several different sound sources are being used at once in the same room, users can be encouraged to use different frequencies. The spectrum allows each team to identify their signal and the neighbours' separately.

The NSW syllabus asks students to 'model the effect of different materials on the reflection and absorption of sound'. This can be done using a standing wave in a pipe.

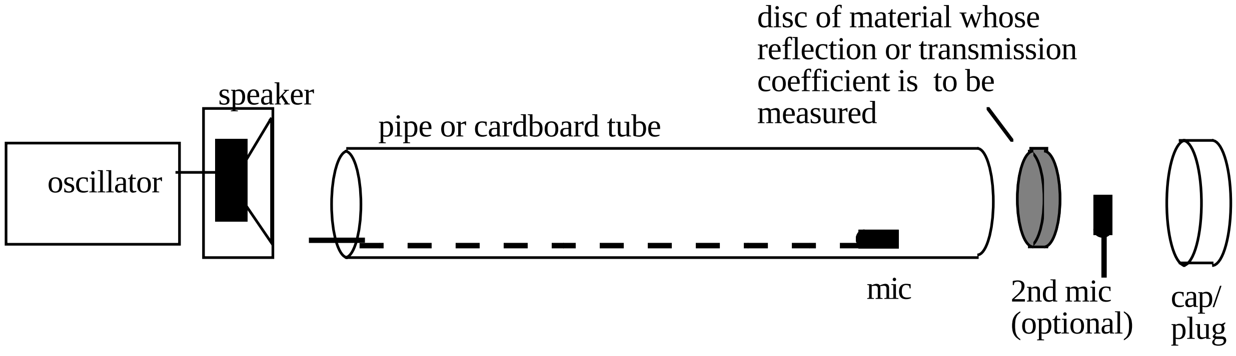
For background on standing waves in pipes, see www.phys.unsw.edu.au/~jw/pipes.html There are sound files at www.phys.unsw.edu.au/~jw/flutes.v.clarinets.html

Method. The speaker is mounted near but not touching the open end of the pipe. The microphone, taped or blu-tacked to a metre rule, is to be slid along the pipe to measure at various different positions. Keep the wavelength larger than the diameter of the pipe (so that it's a plane wave in the pipe) but somewhat less than 4 times the length of the pipe (so that one can find both a node and an antinode).

Resonances. It is of course interesting to find the resonances of the tube. With the cap on (**closed pipe**), put the mic near the cap (a pressure antinode[12]). Scan through frequencies until you get maxima at the cap (you'll also hear the maxima). You may not find the fundamental because its frequency could be too low for your speaker. If you're keen, get the amplifier and woofer, but we warned you. **Open pipe.** It's also interesting to compare these resonances with those of the tube without the cap. Here there are pressure nodes just outside the open end.

Transmission and reflection. For these, the measurements will be more reproducible if you choose a frequency that is ***not*** a resonance, as the amplitudes will then depend less critically on losses associated with the ruler and microphone.

Measuring the distance between nodes gives $\lambda/2$ and thus, with the frequency, the speed of sound. Now to measure reflection and transmission coefficients.



[10] If you don't have an oscillator and speaker, a recorder and a player would do, at the cost of the boredom of the latter. For this option, use the frequency analyser to distinguish the fundamental from the overtones.

[11] You probably won't need an amplifier. Even if the output impedance of the oscillator is a lot higher than that of the speaker, there'll still be enough power transfer for these experiments. And a school issue oscillator ought to be short-circuit proof.

[12] Textbooks often draw displacement or velocity nodes and antinodes, even though these are extraordinarily difficult to measure. In a pipe, an open end is (very near) a pressure node and an amplitude or velocity antinode. At the closed end it is a pressure antinode.

Let the incident wave have a pressure amplitude p_i and the reflected p_r. The maximum sound pressure that we can measure (at the antinodes of the standing wave) will be $p_i + p_r$. The minimum will be $p_i - p_r$. The microphone gives a signal proportional to the pressure. So, sliding the microphone inside the pipe, we shall find maxima and minima with amplitudes $V_{max} = V_i + V_r$ and $V_{min} = V_i - V_r$. The reflection coefficient R is = $p_r/p_i = V_r/V_i$.

$$R = p_r/p_i = V_r/V_i = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$

Various measurements are possible: Measurements with and without the cap and with no absorbing material should both give reflection coefficients close to one. Usually the measurement that is of commercial interest is the factor by which the material reduces the R below that of a rigid reflecting surface. (If I cover my wall with this material, how much less sound will be reflected, at a given frequency?) So the measurement with the absorbing material inside the cap on the end of the pipe is often made. It is also possible to estimate transmission coefficient by placing the material at the open end, then placing a microphone just inside and just outside the material. For this measurement, one should try to minimise external radiation by putting the speaker very near the tube (inside if small). One can also tune to a resonance for this part.

One might at first be disappointed by the relatively high reflection coefficient of say acoustic wool over a rigid wall. However, consider the reverberation. If 99% were reflected, the signal would be reduced after 10 reflections by 0.99^{10} = 90%. If 80% were reflected, ten reflections reduces it to 0.80^{10} = 11%. A few curtains, carpets and bits of furniture make a large acoustic difference to a large empty room with hard walls.

Experiments with ray sonics. $1/r^2$, dishes, directional microphone.

Ray sonics are difficult indoors because of all the reflections. If you can do it out the window or outdoors it's easier. To avoid interference effects (interesting though they be) from near surfaces, use experimental distances << λ.

Note: For these experiments, the spectrum analysis option in the oscilloscope software may be very useful. Its difficult to estimate the amplitudes of two sine wave components in the time domain, but easy in the frequency domain. If several different sound sources are being used at once in the same room, users can be encouraged to use different frequencies. The spectrum allows each team to identify their signal and the neighbours' separately.

Apparatus: Oscillator[13]. Speaker. No amplifier?[14]. Microphone. Metre rule. CRO or PC with sound card.

Warning: This series of experiments could be seriously annoying if several teams are operating simultaneously and all attempting . Cheap ear plugs (about $1 at pharmacists) provide about 20 dB or more reduction in sound level.

What frequency? Low frequencies benefit from the mercifully lower sensitivity of our ears below a few hundred Hz. Too low and there is too much competition from noise, added to the inefficiency of loudspeakers. Too high and interference effects become more likely (see below). We use a few hundred Hz.

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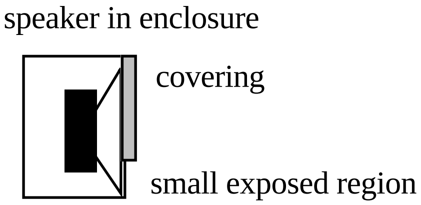
[13] If you don't have an oscillator and speaker, a recorder and a player would do at a pinch, at the cost of the boredom of the latter. For this option, use the frequency analyser to distinguish the fundamental from the overtones.

[14] You probably won't need an amplifier. Even if the output impedance of the oscillator is a lot higher than that of the speaker, there'll still be enough power transfer for these experiments. And a school issue oscillator ought to be short-circuit proof. Sound experiments with amplifiers are not only bad for your ears, but also for speakers. Because our ears are very insensitive at frequencies of say 100 Hz and below, it is easy to blow speakers in the low frequency range.



How to make a 'point source' for use at low frequencies, eg for $1/r^2$ measurements. It's important that these be enclosed (approximately monopoles), unless you wish to investigate the field of a dipole[15] (the latter is however useful as analogy of EM transmission which is often from a dipole antenna). Either use a small speaker (though these are very inefficient at low f) or cover a speaker leaving a small (cm^2) hole. Very heavy cardboard is rigid enough.

$1/r^2$ dependence. If the source can be isolated in space (away from reflecting surfaces), one would expect the intensity to be $P/4\pi r^2$, where P is the acoustic power, which is usually tens or hundreds of times smaller than the electric power supplied to the speaker. A small speaker suspended well away from floor, ceiling, walls etc should approximate an isolated source over a distance range from a few cm to tens of cm. Another possibility, is to put the source very close to a hard floor (which we assume to be a perfect reflector). Now the sound radiates not into a sphere, but a hemisphere, so $I = P/2\pi r^2$.

Note that the sound intensity is proportional to the square of the acoustic pressure. The voltage output from a microphone is approximately proportional to the acoustic pressure. And the electrical power is proportional to the square of the voltage. So what is a $1/r^2$ law for intensity becomes a 1/r dependence for sound pressure and voltage. So plot 1/V *vs* r. Or plot *ln* V *vs li* r.

Small, inexpensive loudspeakers are very inefficient at low frequency, but this may make them more pleasant to work with. Some are sold at prices that makes them almost disposable.

Interference effects.

The problem (or rather the wealth of interesting phenomena) that arises from using small wavelengths in the presence of reflecting surfaces is the effect of interference. Interference is not explicitly in the NSW syllabus, but the syllabus does include several topics that cannot be understood without it. To see how important such effects can be, see 'Young's experiment with sound' and 'comb filters', which are below.

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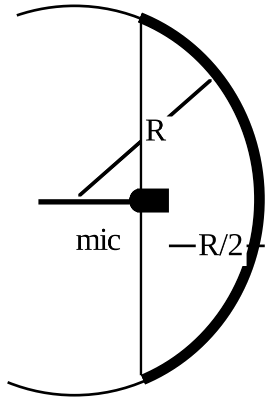
[15] A naked speaker is approximately a dipole: when the cone moves, it compresses the air on one side but rarefies it on the other.

Modelling communication dishes: the directional microphone.



How to build a directional microphone.

Cut a section of a large spherical[16] ball or buoy or some moderately rigid sphere. Cut a bit less than half[17] and mount a microphone at the focus, halfway between the centre and the ball, using (for example) coat hanger wire.

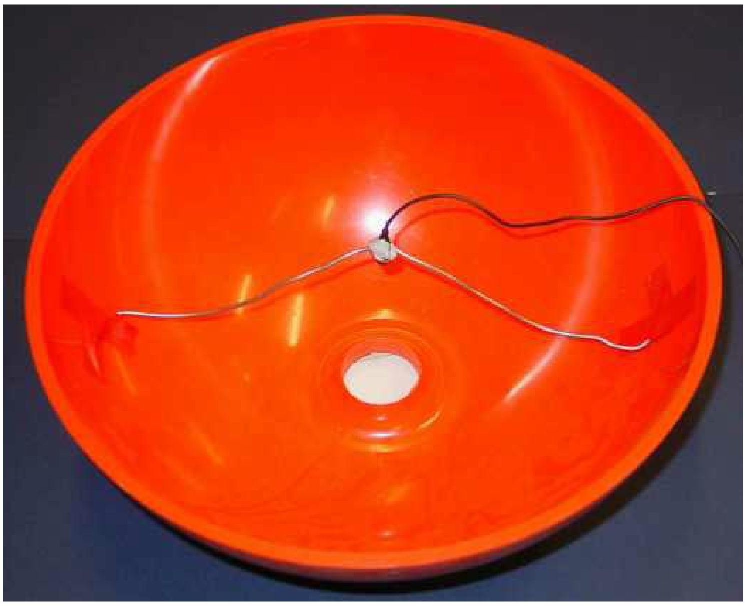
Use a large[18] ball if possible: it only works for $\lambda \ll R$. Your wave must be at least a couple of cm if its audible, and rather less if you are using the PC oscilloscope. Of course, if your speaker, oscillator and CRO go supersonic, that has an advantage for your ears!

Ray sonics are difficult indoors because of all the reflections. If you can do it out the window or outdoors it's easier.

Apart from eavesdropping, this experiment is interesting because one section of the NSW syllabus ('space') concerns the use of dishes in microwave communication with satellites and spacecraft. The wavelength dependence and dish diameter dependence here is completely analogous: focussing dishes produce a much more tightly focussed beam when the dish is much larger than a wavelength.

Method. Use a small sound source. Find an environment where reflections are minimised (see previous and following experiments). Point microphone at source. Measure the amplitude as a function of angle between the axis of the microphone and the direction of the source. Do the same for several frequencies.

Overhearing. Most of the information in speech is carried in the frequency range 300-8000 kHz. (The telephone carries 300-3000 Hz.) If one loses the band 300-1000 Hz, speech is still quite intelligible, although isolated vowels would be hard to identify. So a directional microphone of modest size will give some improvement in eavesdropping. Probably not enough to cause worries about privacy.



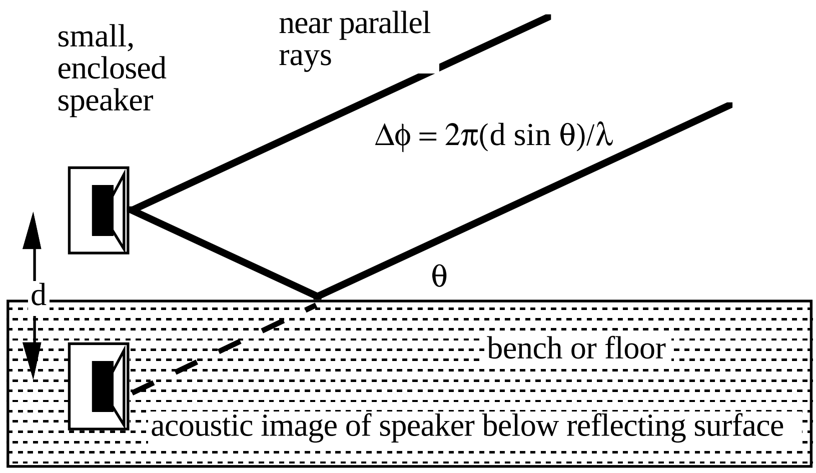
A directional mic made from half a fishing buoy, a piece of wire and a lapel microphone.

[16] Of course a parabola would give a better focus than a sphere, but that distinction won't worry us here. If one can find a spare satellite dish, on the other hand, that makes for an excellent though large directional microphone.

[17] What to do with the other half? One could make two directional mics. Or one could mount a *little* speaker at the focus and have a directional source.

[18] We thought about using an umbrella, but unless the fabric is treated its reflection coefficient is not good enough. For spies, a foldable spy mic could be useful.

Young's experiment with sound



In electromagnetism, this arrangement is called Lloyd's mirror. In the EM case, the electric field vector has a π phase change at the conducting boundary. Here, the pressure has approximately zero phase change at the boundary. (There is a pressure antinode at the rigid surface.)

Even indoors one can do a reasonable Young's experiment with this apparatus. The wavelength should be several times less than d, and both should be small compared to the distance to the microphone. This in turn should be small compared to distances to walls and other reflecting surfaces. So the speaker should be small, or else prepared as discussed above.

This gives an antinode at the floor, and then a series of nodes and antinodes as $\Delta\phi$ goes through $m\pi$ and $(m+\frac{1}{2})\pi$ respectively. These will be clearly seen with a microphone and an oscilloscope or soundcard, provided that other reflections (see below) do not complicate matters excessively.

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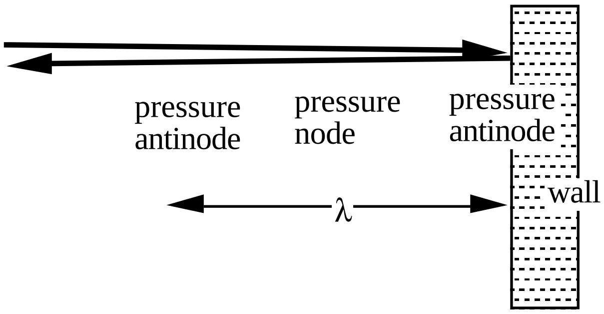
Comb filters and reflections from a wall



Reflections at near right angles are of course just the reflections that we discussed in standing waves in tubes. Unless the signal is very far from the wall, the wave will be diverging, so the reflected wave will be weaker than the incident. So the node will not have zero signal.

This is an interesting effect in itself, and is responsible for what acousticians call comb filtering (production of nodes at discrete and regularly spaced frequencies). It can be quite a nuisance for sound recording, and also quite a nuisance for sound experiments indoors unless one performs small scale experiments

Speed of sound by time of flight. (i) waveguide method.

Background: Waveguides are used to guide microwaves (eg in telecommunications and microwave ovens), light (eg in optical fibres and insect eyes) and quantum mechanical matter waves (in nanowires). Here we use an acoustic waveguide.

Apparatus: Microphone PC with sound card and 'Cool Edit' (or storage oscilloscope and slow trace). Hose or pipe at least 10 m long, and at least 1 cm diameter (we use a coil of lightweight plastic irrigation hose, 19 mm internal diameter, 25 m long (one roll).

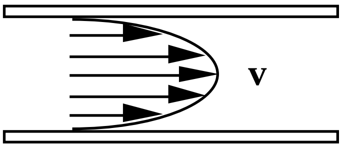
Method: Strike the end of the pipe a quick but light slap with the palm of your open hand. Your hand should temporarily close the pipe and should then 'bounce' off it. You will hear the initial 'click' as your hand closes the pipe and then, a fraction of a second later, a softer and less bright 'thump' as the echo returns. (The reason for the changed timbre of the sound is high frequency loss in the viscous layer near the pipe wall.) If the background noise is low, you may hear the echo of the echo.

Tape a microphone on the outside of and close to one end of the pipe. The other end may be open or closed: either will give an excellent reflection.

Start recording and record the slap and echo(s). Stop the recording, select and magnify this section. Measure the time between slap and echo(s): this is the time for a round trip along your pipe.

Optional 1 Here is a low-tech version: think of the slap and echo as the first two beats in a rhythm. Listen and count them (one, two) to yourself a few times. Now look at the second hand of your watch and time while you count eleven beats in the same rhythm as the two that you have heard. This gives you ten times the duration of the round trip. Record the time at the eleventh imagined beat and divide by ten. For a more precise measurement, listen very carefully to the rhythm of the slap and echo, and count more beats. It should be possible to measure time to better than a few % this way.

Optional 2 Try talking to a friend at the other end of this waveguide. It helps to have a horn at both input and output end, and you can easily make a primitive horn by cupping your hands around the end of the pipe. Whisper very quietly and see whether you can still hear each other. Why is the sound muffled?



The layer of air immediately next to the wall of the tube does not move backwards and forwards with the sound wave. The next layer moves a little and so on, with the air in the middle travelling fastest, as shown in the diagram. This affects high frequencies more than low.

Speaking tubes—cylindrical pipes several cm in diameter—were used in ships for communication: you can see some examples in the retired warship *Vampire* at the Maritime Museum in Sydney.

Speed of sound by time of flight. (ii) outdoors method.

Apparatus: A vertical wall, extending to the ground, facing an open space with no other walls for several tens of metres. A watch and a tape measure. Ideally, the background noise should be reasonably low. If you have trouble hearing the echoes, try clapping with blocks of wood instead of hands.

Method: Stand at least a few tens of metres from the wall. Clap and listen for the echo. Clap again. Adjust the timing of your claps so that the echo falls midway between them. Your claps are now the time for two round trips. Have a colleague measure the time for a series of claps. Measure the shortest distance to the wall. Repeat for other distances.

For someone with a reasonable sense of rhythm, this very low-tech method will give surprisingly consistent results.

The loudspeaker as reciprocating motor, generator and microphone.

A loudspeaker is a linear motor with a small range. It has a single moving coil that is permanently but flexibly wired to the voltage source, so there are no brushes. It moves in the field of a permanent magnet, which is usually shaped to produce maximum force on the coil. The moving coil has no core, so its mass is small and it may be accelerated quickly, allowing for high frequency motion. In a loudspeaker, the coil is attached to a light weight paper cone, which is sealed (and to some extent supported) at the edge by a circular pleated paper 'spring' whose cross-section is shown here. For low frequency, large wavelength sound, one needs large cones. Such speakers are called woofers. They have large mass and are therefore difficult to accelerate rapidly for high frequency sounds. (Note: tweeters (loudspeakers designed for high frequencies) may be just speakers of similar design, but with small, low mass cones and coils. Alternatively, they may use piezoelectric crystals to move the cone.)

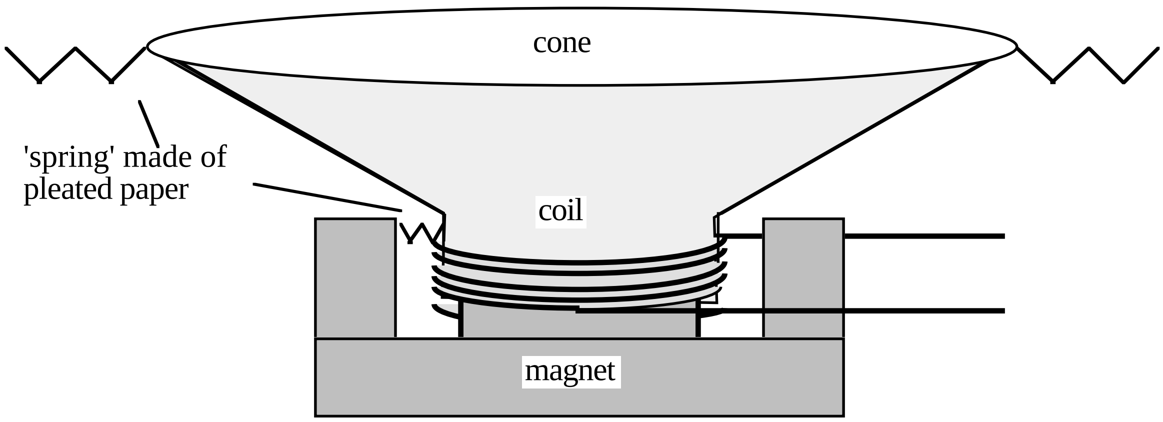
Aims: To investigate a reciprocating motor and generator.

Apparatus: Two loudspeakers. CRO or PC and soundcard. Oscillator and (possibly) amplifier.

Method: The loudspeaker as generator/microphone. Connect loudspeaker to CRO input or directly to sound card. Sing or whistle at the loudspeaker and observe the waveform. ('eee' will give something that looks approximately like a sine wave.) The sound source should be close to the speaker, especially if the speaker is not in an enclosure.

The use of a loudspeaker as a motor/sound source is familiar. However, if a high power speaker is available, it is interesting to drive it at low frequencies and to feel or even to see the motion of the cone. Because our ears are very insensitive at frequencies of say 100 Hz and below, it is easy to blow speakers in the low frequency range.

Dissect one of the loudspeakers to show components. Do not attempt to cut the magnet, but hacksaw through the supporting frame, cone and supports to reveal the cone and its orientation with respect to the internal geometry of the magnet.



Comments: The loudspeaker is a type of motor or actuator. Because it is reciprocating, no brushes or commutator are required. Similar motors are used to move read/write heads on computer discs and CDs.

Motors and generators not only look similar, they often are the same thing. Electric trains use regenerative braking: the motors are used as generators to convert the train's kinetic energy back to electrical energy that is fed back to the power grid. In some early motor cars, the starter motor and generator were the same component.

Monopole vs dipole: note that a naked speaker (~dipole) has sensitivity that falls more rapidly with distance than does that of a speaker in a sealed enclosure (~monopole).

For more on motors, see http://www.phys.unsw.edu.au/~jw/HSCmotors.html

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The solar constant, photovoltaic cell efficiency and calorimetry.

Aims: To estimate the intensity of solar radiation at ground level and the efficiency of a solar cell

Apparatus: A sunny day (preferable) or a high temperature lamp[19] such as those used for photography. (**Warning:** Keep water away from lamps.) Solar panel. Resistor with a power rating at least as great as that of the panel (preferably greater so that it doesn't get hot). Voltmeter. Small, light, disposable, aluminium baking dish. It should have nearly vertical sides if possible (ours were 'home brand', 10 x 18 x 5 cm). Black paint (eg in a spray can). A ruler and a stick. Clock. A balance or scale.

Preparation: Paint all of the inside of the dish black and leave to dry.

Method: Pour in a small, known mass m of water (of known temperature—optional). The depth should be greater than the variation in depth, so the bottom of the dish should be reasonable flat and smooth. Place the dish in direct sunlight, but protected from the wind. Put it on an insulator (piece of cardboard) and keep it flat.

Measure the angle θ of the sun to the horizontal by measuring the length of shadow cast by a stick held vertically.

If the water of mass m is all evaporated in time t, then the energy given to the water[20] is ~Lm where L is the latent heat of water (L_{vap} = 2.3 MJ.kg^{-1}: It is not a strong function of temperature). The heat capacity of the dish is negligible, as are conduction losses through the walls, provided there is no wind. The intensity of solar radiation is then estimated as

$$I_{solar} \equiv \frac{\text{power}}{\text{perpendicular area}} \sim \frac{Lm}{A_{dish}.t.\sin\theta}$$

The power delivered by the solar panel to the resistor R is V^2/R. So its efficiency[21] is

$$e \equiv \frac{\text{electrical power out}}{\text{solar power in}} = \frac{P_{panel}}{A_{panel}} \cdot \frac{1}{I_{solar}}$$

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[19] If using a lamp, the distance from lamp to absorber and their orientation become important. Keep them constant.

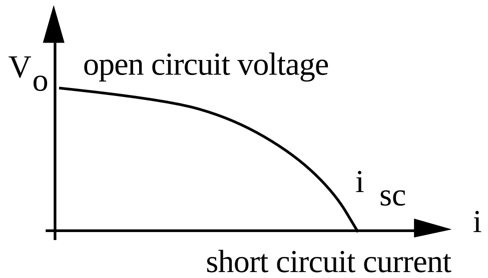
[20] The heat to evaporate the water is much larger than that required to warm it and this experiment makes several crude approximations. It is worthwhile doing this calculation: L/c = 540 K >> temp rise during the experiment.

[21] For the world's most efficient silicon solar cells, and some introductory material, see http://www.pv.unsw.edu.au/

$$= \frac{V^2}{R.A_{panel}} \cdot \frac{A_{dish}.t.\sin\theta}{Lm}$$

The power delivered by the solar cell into a load resistor is determined from the voltage across the resistor.

Maximum power ouput



The maximum power output of a solar cell is often approximated by the expression $F_f.V_o.i_{sc}$ where V_o is the open circuit voltage and i_{sc} the short circuit current. F_f is called the "fill factor". It is an empirical factor to describe the non linearity of the V(i) curve and has a value of typically 0.7. The short circuit current from a small solar cell will not damage ammeters. To determine the V(i) curve for your particular solar cell, get a rheostat or a large size variable resistor or a set of resistors (they're very cheap) and determine the fill factor. Check that the power rating of the resistor considerably exceeds that of the PV cell. (Typically resistors are rated at 1/4 W, but physically larger ones with higher power ratings are also cheap.)

The world communicates

Background: This is the name of one section of the UNSW high school syllabus.

The most commonly used radiation for communication is sound and most information is carried in the 0.2—8 kHz band. In Western languages, much of the linguistic information is encoded in broad frequency bands called formants that fall in this band. Several of them correspond to resonances of the vocal tract.

In normal speech, the vocal folds are closed part of the time and nearly closed for the rest. So it acts approximately as a closed tube. For the vowel [ ] as in "heard", the vocal tract behaves approximately as a uniform pipe, so the usual equations for standing waves apply approximately. For other vowels, the two resonances can be moved up or down substantially, and independently.

There is more background material (including experimental measurements of the resonances for Australian English) on www.phys.unsw.edu.au/speech The nature of speech (as well as speech itself) is widely misunderstood, so it is worth reading up on this topic before starting experiments.

The spectral response of the vocal tract is often difficult to see in a spectrum of a spoken or sung voice, especially women and children's voices, where the frequency range is more sparsely sampled by harmonics of the voice. Whispered speech has a broad spectrum input from the turbulent noise.

Aims: To demonstrate resonances in the human vocal tract and how they vary for different phonemes.

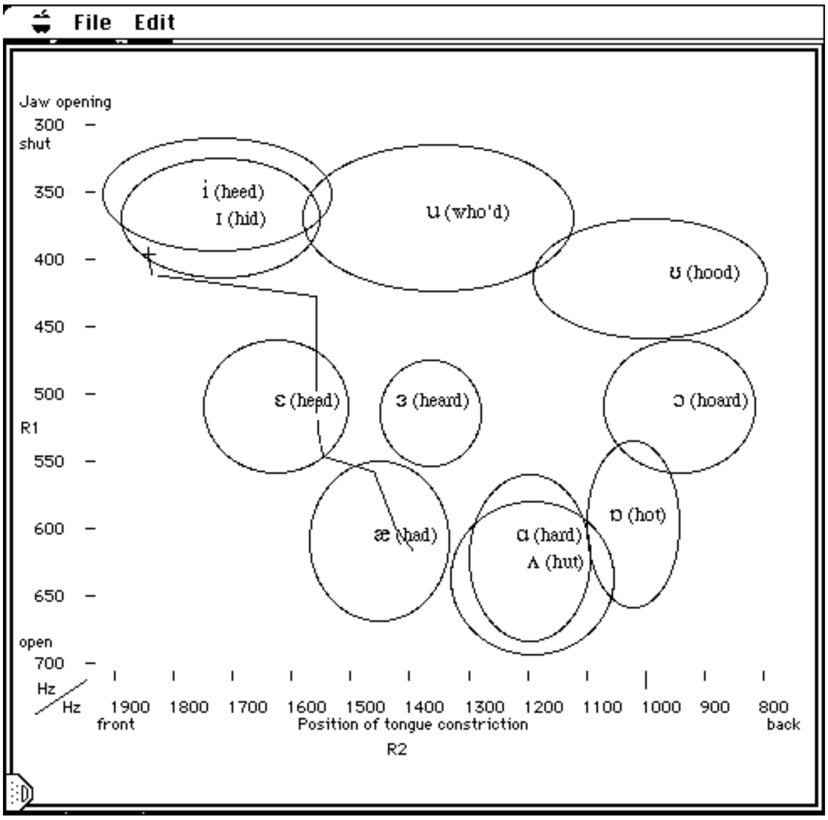
Apparatus: Vocal tract. Microphone or speaker. Sound card and PC with both oscilloscope and Cool Edit software.

Method: Estimate or measure the length of the vocal tract from vocal folds ('Adam's apple') to lips. For most humans it is between 130 and 200 mm. Determine in advance the resonances you would expect for the vowel [ ].

The software is mainly used in spectrum analysis mode.

Sample a sustained, whispered vowel [ ] as in "heard". Look for resonances principally in the range 200 Hz to 1 kHz and 1 kHz to 2 kHz. Using the oscilloscope function, the irregularity of turbulent noise may make regular shapes hard to see. To overcome this, one can sample a sustained vowel using Cool Edit, then select the vowel and perform a spectral analysis, which will automatically average over the whole sample, which by tens of seconds. Estimate the frequencies (f_1,f_2) resonances in or near the two regions mentioned above.

Repeat for other vowels. The eleven vowels of Australian English are placed in a common phonemic context in the following words: "heed", "hid", "head", "had", "hard", "hod", "hoard", "hood", "who'd", "hut" and "heard"[22]. Make a plot of the (f_1, f_2) and compare with the results in www.phys.unsw.edu.au/speech. (Because phoneticians used similar plots before the acoustics of speech was understood, plots of (f_1,f_2) are traditionally done with f_2 as the abscissa, and with both axes reversed.)



The resonances of the vocal tract for Australian English. The subjects were 33 Australian physics students. The centre of each ellipse is the mean frequency of the first resonance (vertical axis) and second resonance. The semi-axes give the standard deviations. The trajetory is a real time plot of the resonance changes in the classic Australian mulit-vowel in 'G'day'. More details at www.phys.unsw.edu.au/speech

[22] We're waiting for wide use of the acronym for Head Up Display to complete the set of words with these consonants.

References and further info from UNSW:

Teachers physics resource forum : www.phys.unsw.edu.au/forums/list.php?f=19/

The establishment of this site was one of the recommendations from our meeting with HSC teachers' earlier this year. The site is not linked from the public HSC sites because we would rather that high school students don't find out about it. (Or rather we would assume that high school teachers might prefer that students don't find out about it.) As yet, it is not much used, but it can be as useful as you want to make it. You will be asked to log on: we do this only so that you do not get spammed.

The UNSW HSC Physics resource site: www.phys.unsw.edu.au/hsc/

This has resource material and references on much of the syllabus.

The HSC FAQ in physics: www.phys.unsw.edu.au/~jw/FAQ.html

Answers the questions most frequently put to us by high school teachers and students. It has moderately detailed explanations, quite a few diagrams and often has links to more detailed material.

The HSC bulletin board:

Used by teachers and students. If your question is not answered in this or in the FAQ, write to us and we'll answer it. http://bat.phys.unsw.edu.au/hsc/

The UNSW Acoustics lab's sites on music, speech and acoustics

www.phys.unsw.edu.au/speech
www.phys.unsw.edu.au/music

A list of educational web pages written by the author of this document:

www.phys.unsw.edu.au/~jw/education.html

Notes about this document.

These notes are written for high school teachers, not for high school students. That is why some of the complicating details are discussed, especially in footnotes. Teachers will want to modify these experiments because different apparatus and circumstances will require different parameter values and conditions.

This document was written for a workshop for high school teachers in the state of New South Wales, Australia. The syllabus for high school physics in this state is unusual, in that it places a relatively large emphasis on history, social studies, contextual discussions and superficial descriptions of technological applications of physics and relatively little emphasis on analytical and quantitative understanding of the physics involved. It also has some unexpected emphases. The syllabus has several advantages, especially to those students (and perhaps even some teachers) who are not fond of doing physics and the mathematics that it usually requires, but who like discussing historical, social and other issues in a qualitative way.

These notes have been influenced by the subject matter in that syllabus and so in many of the experiments the extent of quantitative measurement and analysis is limited. Nevertheless, I have tried to choose experiments that would be suitable for a physics course rather than a physics appreciation course. It is important that students know about the history, social context and applications of physics. But it is also important that our future engineers, technologists and physicists learn to understand the world in an analytical and quantitative way.