Sound, Vibration, Structural Acoustics, Etc.

What is sound ?

Sound is the quickly varying pressure wave within a medium. We usually mean audible sound, which is the sensation (as detected by the ear) of very small rapid changes in the air pressure above and below a static value. This "static" value is atmospheric pressure (about 100,000 Pascals) which does nevertheless vary slowly, as shown on a barometer. Associated with the sound pressure wave is a flow of energy. Sound is often represented diagrammatically as a sine wave, but physically sound (in air) is a longitudinal wave where the wave motion is in the direction of the movement of energy. The wave crests can be considered as the pressure maxima whilst the troughs represent the pressure minima. How small and rapid are the changes of air pressure which cause sound? When the rapid variations in pressure occur between about 20 and 20,000 times per second (ie at a frequency between 20Hz and 20kHz) sound is potentially audible even though the pressure variation can sometimes be as low as only a few millionths of a Pascal. Movements of the ear drum as small as the diameter of a hydrogen atom can be audible! Louder sounds are caused by greater variation in pressure - 1 Pascal, for example, will sound quite loud, provided that most of the acoustic energy is in the mid- frequencies (1kHz - 4kHz) where the ear is most sensitive. What makes sound? Sound is produced when the air is disturbed in some way, for example by a vibrating object. A speaker cone from a hi- fi system serves as a good illustration. It may be possible to see the movement of a bass speaker cone, providing it is producing very low frequency sound. As the cone moves forward the air immediately in front is compressed causing a slight increase in air pressure, it then moves back past its rest position and causes a reduction in the air pressure (rarefaction). The process continues so that a wave of alternating high and low pressure is radiated away from the speaker cone at the speed of sound.

What is a decibel (dB) ?

The decibel is a logarithmic unit which is used in a number of scientific disciplines. In all cases it is used to compare some quantity with some reference value. Usually the reference value is the smallest likely value of the quantity. Sometimes it can be an approximate average value.

In acoustics the decibel is most often used to compare sound pressure, in air, with a reference pressure. References for sound intensity, sound power and sound pressure in water are amongst others which are also commonly in use.

Reference sound pressure (in air) = 0.00002 = 2E- 5 Pa (rms) " " intensity = 0.000000000001 = 1E- 12 W/ m^ 2 " " power = 0.000000000001 = 1E- 12 W " " pressure (water) = 0.000001 = 1E- 6 Pa

Acousticians use the dB scale for the following reasons:

1) Quantities of interest often exhibit such huge ranges of variation that a dB scale is more convenient than a linear scale. For example, sound pressure radiated by a submarine may vary by eight orders of magnitude depending on direction.

2) The human ear interprets loudness on a scale much closer to a logarithmic scale than a linear scale.

How is sound measured ?

A sound level meter is the principal instrument for general noise measurement. The indication on a sound level meter (aside from weighting considerations) indicates the sound pressure, p, as a level referenced to 0.00002 Pa.

Sound Pressure Level = 20 x lg (p/ 0.00002) dB

Peak levels are occasionally quoted. During any given time interval peak levels will be numerically greater, and often much greater than the (rms) sound pressure level.

What does dB(A) or "A- weighted" mean ?

Noise was not of particular concern at the beginning of the century. The first electrical sound meter was reported by George W Pierce in Proceedings of the American Academy of Arts and Sciences, v 43 (1907- 8) A couple of decades later the switch from horse- drawn vehicles to automobiles in cities led to large changes in the background noise climate. The advent of "talkies" - film sound - was a big stimulus to sound meter patents of the time, but there was still no standard method of sound measurement.

The first tentative standard for sound level meters (Z24.3) was published by the American Standards Association in 1936, sponsored by the Acoustical Society of America. The tentative standard shows two frequency weighting curves "A" and "B" which were modeled on the ear's response to low and high levels of sound respectively.

The most common weighting today is "A- weighting" dB(A), which is very similar to that originally defined as Curve "A" in the 1936 standard. "C- weighting" dB(C), which is used occasionally, has a relatively flat response. "U- weighting" is a recent weighting which is used for measuring audible sound in the presence of ultrasound, and can be combined with A- weighting to give AU- weighting. The A- weighting formula is given in section 8 of the FAQ.

In addition to frequency weighting, sound pressure can be weighted in time with fast, slow or impulse response. Measurements of sound pressure level with A- weighting and fast response are also known as the "sound level".

Some sound level meters can measure the average sound level of a noise over a given time. It is called the equivalent continuous sound level (L sub eq) and is A- weighted but not time weighted.

How do sound levels add ?

If there are two sound sources in a room - for example a radio producing an average sound level of 62.0 dB, and a television producing a sound level of 73.0 dB - then the total sound level is a logarithmic sum ie

Combined sound level = 10 x lg (10^(62/ 10) + 10^(73/ 10))
= 73.3 dB

Note: for two different sounds, the combined level cannot be more than 3 dB above the higher of the two sound levels. However, if the sounds are phase related there can be up to a 6dB increase in SPL.

How does the ear work ?

The eardrum is connected by three small jointed bones in the air- filled middle ear to the oval window of the inner ear or cochlea, a fluid-filled spiral coil about one and a half inches in length. Over 10,000 hair cells on the basilar membrane along the cochlea convert minuscule movements to nerve impulses, which are transmitted by the auditory nerve to the hearing center of the brain.

The basilar membrane is wider at its apex than at its base, near the oval window, whereas the cochlea tapers towards its apex. Different groups of the delicate hair sensors on the membrane, which varies in stiffness along its length, respond to different frequencies transmitted down the coil. The hair sensors are one of the few cell types in the body which do not regenerate. They may therefore become irreparably damaged by large noise doses. Refer to the Tinnitus FAQ for more information on hearing disorders.

At what level does sound become unsafe ?

It is best, where possible, to avoid any unprotected exposure to sound pressure levels above 100dB(A). Use hearing protection when exposed to levels above 85dB(A), especially if prolonged exposure is expected. Damage to hearing from loud noise is cumulative and is irreversible. Exposure to high noise levels is also one of the main causes of tinnitus. The safety aspects of ultrasound scans are the subject of ongoing investigation.

There are other health hazards from extended exposure to vibration. An example is "white finger", which is found amongst workers who use hand-held machinery such as chain saws.

What is sound intensity ?

This may be defined as the rate of sound energy transmitted in a specified direction per unit area normal to the direction. With good hearing the range is from about 0.000000000001 Watt per square metre to about 1 Watt per square metre (12 orders of magnitude greater). The sound intensity level is found from intensity I (W/ m^ 2) by:

Sound Intensity Level = 10 x lg (I/ 1.0E- 12) dB

Note: 1.0E- 12 W/ m^ 2 normally corresponds to a sound pressure of about 2.0E- 5 Pascals which is used as the datum acoustic pressure in air.

Sound intensity meters are becoming increasingly popular for determining the quantity and location of sound energy emission.

How does sound decay with distance ?

The way sound changes with distance from the source is dependent on the size and shape of the source and also the surrounding environment and prevailing air currents. It is relatively simple to calculate provided the source is small and outdoors, but indoor calculations (in a reverberant field) are rather more complex.

If the noise source is outdoors and its dimensions are small compared with the distance to the monitoring position (ideally a point source), then as the sound energy is radiated it will spread over an area which is proportional to the square of the distance. This is an 'inverse square law' where the sound level will decline by 6dB for each doubling of distance.

Line noise sources such as a long line of moving traffic will radiate noise in cylindrical pattern, so that the area covered by the sound energy spread is directly proportional to the distance and the sound will decline by 3dB per doubling of distance.

Close to a source (the near field) the change in SPL will not follow the above laws because the spread of energy is less, and smaller changes of sound level with distance should be expected.

In addition it is always necessary to take into account attenuation due to the absorption of sound by the air, which may be substantial at higher frequencies. For ultrasound, air absorption may well be the dominant factor in the reduction.

What is the sound power level ?

Sound power level, Lw, is often quoted on machinery to indicate the total sound energy radiated per second. The reference power is taken as 1pW.

For example, a lawn mower with sound power level 88dB(A) will produce a sound level of about 60dB(A) at a distance of 10 metres. If the sound power level was 78dB(A) then the lawn mower sound level would be only 50dB(A) at the same distance.

What is the speed of sound in air, water .. ?

The speed of sound in air at a temperature of 0 degC and 50% relative humidity is 331.6 m/ s. The speed is proportional to the square root of absolute temperature and it is therefore about 12 m/ s greater at 20 degC. The speed is nearly independent of frequency and atmospheric pressure but the resultant sound velocity may be substantially altered by wind velocity.

A good approximation for the speed of sound in other gases at standard temperature and pressure can be obtained from

c = sqrt (gamma x P / rho)

where gamma is the ratio of specific heats, P is 1.013E5 Pa and rho is the density.

The speed of sound in water is approximately 1500 m/ s. It is possible to measure changes in ocean temperature by observing the resultant change in speed of sound over long distances. The speed of sound in an ocean is approximately:

c = 1449.2 + 4.6T - 0.055T^ 2 + 0.00029T^ 3 + (1.34- 0.01T)(S- 35) + 0.016z

T temp in degrees Celsius, S salinity in parts per thousand z is depth in meters

What is meant by loudness?

Loudness is the human impression of the strength of a sound. The loudness of a noise does not necessarily correlate with its sound level. Loudness level of any sound, in phons, is the decibel level of an equally loud 1kHz tone, heard binaurally by an otologically normal listener. Historically, it was with a little reluctance that a simple frequency weighting "sound level meter" was accepted as giving a satisfactory approximation to loudness. The ear senses noise on a different basis than simple energy summation, and this can lead to discrepancy between the loudness of certain repetitive sounds and their sound level.

A 10dB sound level increase is considered to be about twice as loud in many cases. The sone is a unit of comparative loudness with 0.5 sone= 30 phons, 1 sone= 40 phons, 2 sones= 50 phons, 4 sones = 60 phons etc. The sone is inappropriate at very low and high sound levels where subjective perception does not follow the 10dB rule.

Loudness level calculations take account of "masking" - the process by which the audibility of one sound is reduced due to the presence of another at a close frequency. The redundancy principles of masking are applied in digital audio broadcasting (DAB), leading to a considerable saving in bandwidth with no perceptible loss in quality.

What is vibration ?

When something oscillates about a static position it can be said to vibrate. The vibration of a speaker diaphragm produces sound, but usually vibration is undesirable. Common examples of unwanted vibration are the movement of a building near a railway line when a train passes, or the vibration of the floor caused by a washing machine or spin dryer. Floor vibration can be reduced with vibration isolators; however there is often a penalty to pay in the form of a slight increase in the machinery vibration and its consequent deterioration.

How is vibration measured ?

Vibration is monitored with an accelerometer. This is a device that is securely attached by some means to the surface under investigation. The accelerometer produces a tiny electrical charge output, proportional to the surface acceleration, which is then amplified by a charge amplifier and recorded or observed with a meter. The frequencies of interest are generally lower than sound, and range from below 1 Hz to about 1 kHz.

It is sometimes more useful to know the velocity or displacement rather than the acceleration. In the case of velocity, it is necessary to integrate the acceleration signal. A second integration will provide a displacement output. If the vibration is sinusoidal at a known frequency, f, then an integration is easily calculated by dividing the original by 2 x pi x f (noting that there is a phase change)

Example: A machine is vibrating sinusoidally at 79.6 Hz with an rms acceleration of 10 m/ s^ 2. Its rms velocity is therefore 10/(2 x pi x 79.6) = 20 mm/ s Its rms displacement is 10/(4 x pi^ 2 x 79.6^ 2) = 0.04 mm

How is vibration isolated and controlled ?

Vibration problems are solved by considering the system as a number of springs and masses with damping. It is sometimes possible to reduce the problem to a single mass supported by a spring and a damper.

If the vibration is produced by a motor inside a machine, it is usually desirable to ensure that the frequency of motor oscillations (the forcing frequency) is well above the frequency of the natural resonance of the machine on its support. This is achieved by altering the mass or stiffness of the system as appropriate.

The method of vibration isolation is very easy to demonstrate with a weight held from a rubber band. As the band is moved up and down very slowly the suspended weight will move by the same amount. At resonance the weight will move much more, but as the frequency is increased still further the weight will become almost stationary. In practical circumstances springs are more likely to be used in compression than tension, but the principles are exactly the same.

A further method of vibration control is to attempt to cancel the forces involved using a Dynamic Vibration Absorber. Here an additional "tuned" mass- spring combination is added so that it exerts a force equal and opposite to the unwanted vibration. They are only appropriate when the vibration is of a fixed frequency.

Active vibration control, using techniques akin to active noise control, is now coming into use.

Important: -Intuitive attempts to reduce vibration from machinery can sometimes instead aggravate the problem. This is especially true when care was originally taken to minimize vibration at the time of design, manufacture and installation.

Architectural & Building Acoustics -----------

What is reverberation time ?

Work on room acoustics was pioneered by Wallace Clement Sabine 1868- 1919 (see his Collected Papers on Acoustics, 1922). The reverberation time, T, is defined as the time taken for sound energy to decay in a room by a factor of one million (ie by 60 dB). It is dependent on the room volume and its total absorption.

What is the sound absorption coefficient ?

The absorption coefficient of a material is ideally the fraction of the randomly incident sound power which is absorbed, or otherwise not reflected. It can be determined in two main ways, and there are often variations in the results depending upon the method of measurement chosen. It is standard practice to measure the coefficient at the preferred octave frequencies over the range of at least 125Hz - 4kHz.

For the purposes of architectural design, the Sabine coefficient (calculated from reverberation chamber measurements) is preferred. Interestingly some absorbent materials are found to have a

Sabine coefficient in excess of unity at higher frequencies. This is due to edge effects and when this occurs the value can be taken as 1.0

What is the difference between insulation & absorption ?

There is often confusion between sound insulation and sound absorption.

Sound insulation is required in order to eliminate the sound path from a source to a receiver such as between apartments in a building, or to reduce unwanted external noise inside a concert hall. Heavy materials like concrete tend to be the best materials for sound insulation -doubling the mass per unit area of a wall will improve its insulation by about 6dB. It is possible to achieve good insulation with much less mass by instead using a double leaf partition (two separated independent walls).

Sound absorption occurs when some or all of the incident sound energy is either converted into heat or passes through the absorber. For this reason good sound absorbers do not of themselves make good sound insulators. Although insulation and absorption are different concepts, there are many instances where the use of sound absorbers will improve insulation. However absorption should not be the primary means of achieving good sound insulation.

How is sound insulation measured ?

The measurement method depends on the particular situation. There are standards for the measurement of the insulation of materials in the laboratory, and for a number of different field circumstances. Usually the procedures involve generating a loud sound of a specified type and monitoring the transmitted noise.

It is very useful to have a single number to characterize the insulation of a partition. Measurements are often conducted in third-octaves, and the reduction plotted on a graph. A reference curve is then fitted to the measurements using a specified procedure, and the value of this curve at 500 Hz is taken as the figure. There is a slight difference in procedure between the U. S. and ISO standards, but the methods are basically similar. The same is also true for impact noise transmission assessment, where a standard tapping machine is in use to hammer floors. Sound pressure levels in the room below are monitored.

How do I improve the noise insulation of my house/ dwelling?

This is one of the most commonly asked questions of noise consultants. Firstly you should consider whether better insulation is really essential. The method of noise insulation will depend on the exact situation, so the advice of a competent person should be sought at an early stage. Sound insulation is most often asked for in order to keep out unwanted noise, but is occasionally requested for the purpose of minimizing disturbance to others. The following ideas may serve as guidelines.
When the noise is from an external source such as a main road it may be possible, if planning authorities permit, to screen with a noise barrier. These can be effective providing that the direct line of sight between traffic and house is concealed by the barrier.

The weak point for sound transmission to and from a building is most often via the windows. Double glazing will usually afford noticeably better protection than single glazing, but in areas of high external noise it might be preferable to have double windows with a large air gap and acoustic absorbent material in the reveals. A drawback of improving external insulation is that, for some people, the resultant lower background level can itself be disturbing; it can also make noise transmission through party walls more apparent. The fitting of new windows may reduce the level of air ventilation, and it will be vital to compensate for this, if necessary with a noise attenuating system.

You may also need to consider noise penetration through the roof, floors, ceilings and walls.

Noise through party walls can be reduced by the addition of a false wall. This is constructed from a layer of sound insulating material, commonly plasterboard, separated from the party wall by a large void containing acoustic quilting. The false wall must not be connected to the party wall because that would allow sound transmission paths. The quality of construction is an important consideration if optimal levels of attenuation are desired. It is advisable to contact an independent noise consultant before allowing any building works to commence.

What is active noise control ?

ANC is an electronic method of reducing or removing unwanted sound by the production of a pressure wave of equal amplitude but opposite sign to the unwanted sound. When the electronically produced inverse wave is added to original unwanted sound the result is sound cancellation.

This method of noise control is becoming increasingly popular for a variety of uses. It is sometimes considered a miracle "cure- all" for noise problems which, at the present time, is not the case. For example noise cancellation in 3D spaces, such as living areas, is very difficult to achieve. However it can be more successful locally, eg for a passenger sitting in an aircraft or car. There are many institutions and companies around the world working on the technology to increase the circumstances where ANC can be used effectively. The award winning Active Noise Control FAQ is maintained by Chris Ruckman and available at a number of sites worldwide.

What causes a sonic boom ?

When the speed of an aircraft is supersonic, the pressure waves cannot get away ahead of the aircraft as their natural speed is slower than that of the aircraft. Slower, in this context, means just over 1200 km/ hr at sea level and about 10% less at normal cruising altitude. Because they cannot get away, the pressure disturbances coalesce and lag behind the aeroplane, which is in effect travelling at the apex of a conical shock wave. The main shock wave is generated by the extreme nose of the aeroplane, but ancillary shocks are generated by all the major fuselage discontinuities.

A body moving through the air pushes the air aside. Small disturbances move away at the speed of sound. Disturbances from a slowly moving body go out in circles, like ripples from a pebble in a pond. If the body moves faster, the circles are closer in the direction of travel. If the body is supersonic, then the circles overlap. The envelope of circles forms a cone. The angle of the cone is determined by its vertex moving in the body's travel direction at the body's speed, while the circles grow at the sound speed. The existence of the "Mach cone", "Mach waves" and the corresponding angle, was discovered by Ernst Mach in the nineteenth century.

Can you focus sound ?

Sound can be focused like light, but in the case of sound the "optics" must be much larger because you are dealing with longer wavelengths. The effect is heard in some domed buildings such as the Capitol in Washington, and St Paul's Cathedral in London providing noise background conditions permit. Large parabolic reflectors can be used very effectively to send and receive sound over significant distances. Check out your local science museum or exploratorium - there may be a demonstration. It is also possible to refract sound and focus it using a lens. The lens is constructed from a large thin bubble, say 2 metres across, filled with carbon dioxide. The effect is not very pronounced.

Sound can be directed by making use of constructive and destructive interference. This idea is used in column speakers, and commercial systems for reducing noise levels outside the dance floor area of discos.

What is sonoluminescence ?

In the early 1930s Frenzel and Schultes discovered that photographic plates became "fogged" when submerged in water exposed to high frequency sound. More recent experiments have succeeded in suspending a single luminous pulsating bubble in a standing wave acoustic field, visible in an undarkened room. Generally sonoluminescence is light emission from small cavitating bubbles of air or other gas in water or other fluids, produced when the fluid is acted upon by intense high frequency sound waves. The mechanism is not completely understood, but very high pressures and temperatures are thought to be produced at the centre of the collapsing bubbles.

“ .. I have been sufficiently interested to reconstruct the apparatus for producing this effect -- using a pair of piezoelectric transducers, an old oscilloscope and a signal wave generator -- materials costing only a few hundred dollars.

I am proud to say that tonight I managed to reproduce this effect -- the tiny bubble has the appearance of a tiny blue star trapped in the middle of the flask. It is distinctly visible to the unadapted eye in a dark room, and it is a very startling thing to see.”

Why does blowing over a bottle make a note ?

Resonance in acoustics occurs when some mass- spring combination is supplied with energy. Many musical instruments rely on air resonance to improve their sonority. If you blow across the mouth of a bottle you can often get a note. The bottle behaves as a Helmholtz resonator. The main volume of air inside the bottle is analogous to a spring, whilst the "plug" of air in the neck acts as an attached mass. The resonant frequency is roughly given by:

f = { c sqrt (S/ LV) } / 2pi

c is velocity of sound S is the surface area of the neck opening V is bottle volume L is the effective length of the neck ie the actual length plus ends correction. Ends correction ~ 1.5 times radius of neck opening

Example: A 75 cl (7.5E- 4 m^ 3) wine bottle with neck diameter 19 mm, bottle neck length 8 cm, air temp = 20 degC calculated resonance = 109Hz (actual resonance was 105Hz)

Helmholtz resonators are sometimes employed as a means of passive noise control in air conditioning ducts. They may also be hidden in the wall design of auditoria and offices in order to improve the acoustics.

What is pitch ?

The term "pitch" has both a subjective and an objective sense. Concert pitch is an objective term corresponding to the frequency of a musical note A (at present 440Hz). Using such a standard will define the pitch of every other note on a particular musical scale. For example, with Equal Temperament each semitone is higher or lower in frequency than the previous semitone by a factor of 2^(1/ 12). An octave is a pitch interval of 2: 1. Many sounds with no obvious tonal prominence are considered by musicians to be of indeterminate pitch; for example, the side drum, cymbals, triangle, castanets, tambourine, and likewise the spoken word.

Pitch is also a subjective frequency ordering of sounds. Perceived pitch is dependent on frequency, waveform and amplitude or changing amplitude. Numbers can be assigned to perceived pitch relative to a pure frontal tone of 1000Hz at 40dB (1000 mels) thereby establishing a pitch scale.

What are musical intervals ?

An interval is the ratio in frequency between musical notes. These intervals are sometimes called a second, third, fourth, fifth etc. which refers to the position on the scale that the note is to be found. In the scale of C major: C D E F G A B C, the note 'E' is the third note of the scale and the interval from C to E is therefore called a third. For the scale D major: D E F# G A B C# D, the third will be F#. The term 'interval' can also be used to indicate that the notes are sounded together, in which case there are consonant intervals and dissonant intervals.
The ratio of frequency intervals for Just Intonation is demonstrated below in the scale of C major, though the same ratios apply to all the major keys:

C (9: 8) D (10: 9) E (16: 15) F (9: 8) G (10: 9) A (9: 8) B (16: 15) C <- Octave

The interval between E & F and between B & C is a semitone, whilst the other intervals are tones. The interval between any two notes above can be found by multiplying the intervening ratios; thus if all the above ratios are multiplied together the resultant is 2 because an octave is twice the original frequency.

The notes of minor scales differ from their major counterparts; one important difference being the flattened third. E flat is a minor third above the note C.

The use of Just Temperament causes serious problems of intonation when music modulates between keys. Equal Temperament is nearly always used as a compromise to the problem of tuning.

What causes "helium voice" ?

Many people, on hearing the voice of someone who has breathed helium, believe that the person's speech pitch has increased.

WARNING **-** Breathing helium can be very dangerous. ^^^^^^^ A cavity will have certain resonant frequencies. These frequencies depend on the shape and size of the cavity and on the velocity of sound within the cavity. Human vocal cords vibrate non- sinusoidally in the vocal tract, giving rise to a range of frequencies above the fundamental. The vocal tract mainly enhances lower frequency components imparting the recognizable voice spectrum.

The velocity of sound in helium is much greater than in air, so breathing helium will raise the vocal tract's resonant frequencies. Although the vocal cords' vibrational frequencies are little affected by helium, the effect of higher cavity resonances is to alter substantially the relative amplitudes of the voice spectrum components thus leading to apparent pitch change.

What is structural acoustics ?

Structural acoustics is concerned with the coupled dynamic response of elastic structures in contact with non- flowing fluids. (The fluid, although non- flowing, undergoes small- amplitude vibration relative to some equilibrium position.) For heavy fluids like water, the coupling is two-way, since the structural response is influenced by the fluid response, and vice versa. For lighter fluids like air, the coupling may be either one- way (where the structural vibration affects the fluid response, but not vice versa) or two- way (as occurs, for example, in the violin).

Structural acoustics problems of interest involving water include the vibration of submerged structures, acoustic radiation from mechanically- excited, submerged, elastic structures; acoustic scattering from submerged, elastic structures (e. g., sonar echoes); acoustic cavity analysis; and dynamics of fluid- filled elastic piping systems. These problems are of interest for both time-harmonic (sinusoidal) and general time- dependent (transient) excitations. Water hammer in pipes can be thought of as a transient structural acoustics problem.

Structural acoustics problems of interest involving air include determining and reducing noise levels in automobile and airplane cabins.

What is the doppler effect ?

When a sound source is moving, a stationary observer will detect a different frequency to that which is produced by the source. The speed of sound in air is approximately 340 m/ s (see 2.11). The wavelength of the sound emitted will be foreshortened in the direction of motion by an amount proportional to the velocity of the source. Conversely the wavelength of a receding sound source will increase. The doppler effect may be noticed as a marked drop in pitch when a vehicle passes at high speed.

Example 1: A sound source, S, emits 1000 waves per second (1 kHz) and is moving directly towards an observer, O, at a speed of 100 metres per second (equivalent to approx 225 miles per hour).

After 1 second the wave front, which is travelling at the speed of sound, will have travelled 340 metres from the original source position. Also after that second the sound source will have moved 100 metres towards the observer.

0 m 340 m S | | | | | | | | | O
<-------------- 1000 waves ------------------>

100 m 340 m S | | | | | | | | | O <------- 1000 waves --------->

Therefore the same number of waves will occupy a space of 340- 100 = 240 metres and the wavelength will be 240/ 1000 = 0.24 metres. To the observer the frequency heard will be the speed of sound divided by its wavelength = 340/ 0.24 = 1416.7 Hz.

Example 2: An observer moving at 100 metres per second directly approaches a stationary sound source, S, which is emitting 1000 waves per second (1 kHz). In this example there is no change in wavelength. In one second, the observer will hear the number of waves emitted per second plus the number of waves which s/ he has passed in the time (1000+ 100/ 0.34) = 1294.1 Hz.

Note the interesting result - a stationary observer with moving source will not hear the same frequency as a would a moving observer with stationary source.

What is white noise, pink noise ?

The power spectral density of white noise is independent of frequency. Since there is essentially the same energy between any two identical frequency intervals (for example 84- 86Hz and 543- 545Hz), white noise narrow band FFT analysis will show as flat. However octave band analysis will show the level to rise by 3dB per octave because each band has twice the frequency range of the preceding octave.

Pink noise is often produced by filtering white noise and has the same power within each octave. Narrow band analysis will show a fall in level with increasing frequency, but third- octave band or octave band analysis will be flat.