Nicholas Hender

Detailed Study, Sound, Study Notes

Sound

Sound is a **form of energy**, transmitted from its source as a **mechanical longitudinal wave**, therefore requiring media through which to travel, consisting of compressions (increased pressure) and rarefactions (decreased pressure).

Sound waves can be represented as transverse waves by graphing distance or time on the horizontal axis against pressure on the vertical axis. Compressions are represented as crests, and rarefactions are represented as troughs.

Speed, Frequency And Wavelength

The speed of sound in air at $20°\text{C}$ is 343ms^{-1}. This increases with the density of the media (slowest in gases, faster in liquids and fastest in solids). In gases the speed of sound increases with temperature and pressure.

The "wave equation", $v=f\lambda$, applies to sound, relating frequency, f, and wavelength, λ, with velocity, v. If v is constant, high values of f have short values of λ, and low values of f have long values of λ.

Loudness: Wm^{-2} (Intensity), Dispersion And Distance

Sound intensity is measured as an **amount of power per unit of area**, and therefore with the unit Watts per metre squared, $\mathbf{Wm^{-2}}$ ("Watt metres to the negative two"). 10^{-12}Wm^{-2} is the lowest intensity considered to be detectable by human ears (at a frequency of 1000Hz). 1Wm^{-2} usually causes pain and damage to human ears.

Sound waves move away from the source in straight lines in all directions, so wavefronts (compressions) are spherical (three dimensional circular wavefronts). Therefore **the area used for the calculation of intensity is the area of a sphere** (of radius equal to the distance between the source of the sound and the point of power measurement). Each wave (as a straight line) moves away from all other waves, spreading out such that **intensity varies inversely with the square of the distance from the source** (sound gets quieter as the listener moves away from the source, or as the source moves away from the listener).

$$I = \frac{P}{4\pi r^2}$$

Where P = power of source, in W
r = radius of separation (distance from source), in m
I = intensity, in Wm^{-2}

$$I \propto \frac{1}{r^2}$$

$$I_1 r_1^2 = I_2 r_2^2$$

Where I_1 = intensity at r_1, in Wm^{-2}
r_1 = first radius of separation (distance from source), in m
I_2 = intensity at r_2, in Wm^{-2}
r_2 = second radius of separation (distance from source), in m

Loudness: Decibels (Level)

Human ears detect variations in sound intensity as ratios rather than differences. Intensity **level**, measured with the unit **decibel**, dB, is a unit for loudness that represents the way in which we hear sound more closely than Wm^{-2}.

- A sound of f=1000Hz at an **intensity of $10^{-12}Wm^{-2}$** is equal to a **level of 0dB**.
- An increase of a factor of ten in Wm^{-2} is equal to an increase of 10dB, so:

	Sound Intensity, Wm^{-2}	Sound Level, dB
Threshold of hearing	10^{-12}	0
	10^{-11}	10
	10^{-10}	20
	10^{-9}	30
	10^{-8}	40
	10^{-7}	50
	10^{-6}	60
	10^{-5}	70
	10^{-4}	80
	10^{-3}	90
	10^{-2}	100
	10^{-1}	110
Threshold of pain	1	120

- An increase of 3dB corresponds with a perceived doubling of loudness.
- An decrease of 3dB corresponds with a perceived halving of loudness.
- Equal changes in sound level are perceived as equal changes in loudness (for example, an increase from 20dB to 25dB sounds like the same difference as an increase from 80dB to 85dB).
- A normal human ear can perceive changes of 1dB or greater.

Conversion From Intensity (Wm^{-2}) To Level (dB)

$$L = 10\log\frac{I}{I_0}$$

Where I = sound intensity, in Wm^{-2}
I_0 = reference intensity, usually the lower threshold of human hearing, $\mathbf{10^{-12}Wm^{-2}}$
L = sound level, in dB

Conversion From Level (dB) To Intensity (Wm^{-2})

$$I = I_0 10^{L/10}$$

Where I_0 = reference intensity, usually the lower threshold of human hearing, $\mathbf{10^{-12}Wm^{-2}}$
L = sound level, in dB
I = sound intensity, in Wm^{-2}

Loudness: Phons

The loudness of sound perceived by human ears differs with frequency. The Phon is unit for loudness that takes this into account. A graph of a typical human ear's sensitivity to different frequencies of sound allows conversion between dB and Phons. Sound level in decibels is on the vertical axis, with frequency on the horizontal axis. A range of curves indicates how the perceived loudness of different frequencies varies with the actual level of sound.

To find the perceived loudness of a frequency in Phons, the level/frequency graph can be interpreted in this way:

- For a sound with frequency of **1000Hz**, the Phon level is always equal to the dB level (**at 1000Hz, Phon=dB**).
- A curve passing trough the co-ordinates (1000Hz, X dB) is the "X Phon curve".
- The dB values of all the other frequencies along this curve are considered to have a Phon level of X, because **these frequencies are all perceived as the same loudness** (equal to that of 1000Hz at X dB).

For example:

- At 1000Hz, 60Phon=60dB.
- A curve passing trough the co-ordinates (1000Hz, **60**dB) is the "**60** Phon curve".
- The dB values of all the other frequencies along this **60** Phon curve are considered to have a Phon level of **60**, because **these frequencies are all perceived as the same loudness** (equal to that of 1000Hz at **60**dB).

Standing (or "Stationary") Waves And Resonance

When waves reflect, interference occurs between:

- the **incident** wave (**before** reflecting, travelling in a direction **towards** the point of reflection), and
- the **reflected** wave (**after** reflecting, travelling in a direction opposite to that of the incident wave, **away** from the point of reflection).

This interference results in a wave of maximum displacement (or "amplitude" or "intensity", perceived as "loudness" in the case of sound) **equal to the sum of the displacements of the incident and reflected waves**. As well being considered as interference, the occurrence of more than one wave in the same place at the same time is often referred to as **superposition**.

If the reflected wave (travelling in the **opposite direction** to that of the original incident wave) is reflected back again, causing a wave travelling in the **same direction** to that of the original incident wave, **and** the distance between the two points of reflection is equal to a multiple of either **λ/4** or **λ/2** (depending on the particular circumstances of reflection), the wave resulting from the interference has yet greater displacement, as it is the sum of waves of already increased displacement. This wave is called a **standing, or stationary, wave**, because the rather than appearing to move along as its energy is transmitted, the **nodes remain fixed in their locations**, and **antinodes alternate** between being compressions (represented as crests) and rarefactions (represented as troughs). The occurrence of a standing sound wave is called **resonance**.

Every object that can vibrate has a set of frequencies (corresponding to wavelengths) at which waves travelling through it result in standing waves. These are an object's **resonant frequencies**. When caused to vibrate by other vibrating objects, objects with a wide range of resonant frequencies can be used to amplify the intensity of the sound from the other vibrating objects. This is how the body of a guitar or violin, or the soundboard of piano, amplifies the intensity of the sound made by the strings attached to it.

Resonance In Strings And Open Ended Pipes, And The Harmonic Series

A string fixed at both ends and in tension (such as that of a violin) produces sound when plucked or bowed. An open ended pipe (such as that of a pan-flute) produces sound when air of a high enough velocity is blown across its top. The frequencies of the sound such a string or pipe can produce (without changing the length or diameter [or tension, in the case of a string]) are the string or tube's resonant frequencies, and are known as the **harmonic series**.

Each frequency of the harmonic series is called an **harmonic**. The lowest harmonic is often called the **fundamental frequency**, or the first harmonic, f_1. Each subsequent harmonic is called the second, f_2, third, f_3, fourth, f_4, and so on. Each harmonic is a whole number multiple of the first, such that:

$$f_n = nf_1$$

Where f_n = frequency of the nth harmonic, in Hz
f_1 = frequency of the 1st harmonic (the fundamental frequency), in Hz
n = harmonic number, a whole number

The wavelength of a sound at which a string or open ended pipe will resonate is given by:

$$\lambda_n = \frac{2L}{n}$$

Where L = length of string or pipe, in m
n = harmonic number, a whole number
λ_n = wavelength of the nth harmonic, in m

The frequency of the harmonic associated with this wavelength (f_n) can then be found by the wave equation:

$$v = f\lambda$$
$$\rightarrow f_n = \frac{v}{\lambda_n}$$

Where f_n = frequency of the nth harmonic, in Hz
λ_n = wavelength of the nth harmonic, in m
v = speed of sound, in ms^{-1} ($343ms^{-1}$ in air at 20°C)

Harmonics are sometimes referred to as **overtones**, however the first harmonic, the fundamental, is not considered as an overtone, so **the first overtone is the second harmonic**; overtone 1 = f_2, overtone 2 = f_3, overtone 3 = f_4, and so on.

Resonance In Pipes Closed At One End

Pipes closed at one end only resonate at **odd numbered** harmonic frequencies (f_1, f_3, f_5, f_7, f_9, and so on), such that:

$$f_n = nf_1$$

Where f_n = frequency of the **odd** nth harmonic, in Hz
f_1 = frequency of the 1st harmonic (the fundamental frequency), in Hz
n = harmonic number, a whole number

The wavelength of a sound at which a pipe closed at one end will resonate is given by:

$$\lambda_n = \frac{4L}{n}$$

Where L = length of string or pipe, in m
n = harmonic number, an **odd** whole number
λ_n = wavelength of the **odd** nth harmonic, in m

The frequency of the harmonic associated with this wavelength (f_n) can then be found by the wave equation, as described in the previous section.

Diffraction Of Sound Waves

Sound waves diffract in the same way as all other types of waves. The direction of their motion changes as their wavefronts pass by the edge of barriers. This causes sound to "spread out" and reach an area greater than that which would otherwise be reached by the length of the wavefront passing the edge of a barrier.

Greater wavelengths, and hence lower frequencies, demonstrate diffraction to a greater extent than shorter wavelengths, and hence higher frequencies (which demonstrate diffraction to a lesser extent):

- **Long λ, low f, more diffraction.**
- **Short λ, high f, less diffraction.**

This is why a "dull roar" or "low rumble" can be heard on the outside side of freeway noise barriers. The high frequencies (short wavelengths) produced by the cars on the freeway diffract very little, and continue over the barrier across the tops of houses without reaching the ground until their energy is dispersed. The low frequencies (long wavelengths) produced by the cars on the freeway diffract downwards as they cross the top of the barrier, and as a consequence are audible on the ground on the other side to the cars.

Diffraction is also why sound from one room can be heard in the next room, even when not standing directly in front of the doorway. The wall is a barrier, and the doorway acts as a gap in that barrier, diffracting the sound to "spread out" as it enters the next room.

If a wavefront passes through a gap between two barriers (a "slit"), it can diffract to be a circular wavefront, but only if the width of the gap (the "slit-width", w) is equal to or less than the wave's wavelength. The extent of diffraction is therefore inversely proportional to w.

- **Wide w, less diffraction.**
- **Narrow w, more diffraction.**
- **$w \leq \lambda$ for circular wavefront (maximum "spreading out").**

Loudspeakers: How They Work

Loudspeakers are "transducers"; devices that convert one form of energy into another. They produce sound energy by converting electrical energy. A loudspeaker consists of these main parts:
- The "cone", usually made from stiff paper or light cardboard.
- The "coil", consisting of many loops of fine wire, attached to the back of the cone.
- A permanent magnet, situated inside and/or around the coil.

These parts work together to convert electrical energy into sound in this way:
1. A sound wave is represented as an **alternating current** from a magnetic tape (such as a cassette), record (such as a vinyl LP), digital data file (such as a CD or MP3) or microphone.
2. The alternating current is connected to the two ends of the wire that makes the coil.
3. This causes the coil to have a magnetic field, alternating with the electrical representation of the sound.
4. The magnetic field of the coil causes it to be attracted to, or repelled from, the permanent magnet, alternating with the frequency of the current (which represents the frequency of the sound).
5. This causes the coil to move inwards and outwards in relation to the permanent magnet, at the frequency of the alternating current.
6. The coil is attached to the back of the cone, so the cone moves inwards and outwards as well, also at the frequency of the alternating current.
7. When the cone moves **outward**, it causes a **compression** in the air, corresponding with a **crest** in the alternating current.
8. When the cone moves **inward**, it causes a **rarefaction** in the air, corresponding with a **trough** in the alternating current.

At this point the loudspeaker has converted electrical energy, representing the frequency and displacement of sound as an alternating current, into actual sound waves in the air.

Tweeters are speakers designed specifically for the reproduction of **high frequency** sounds. They are usually small (between 2.5 and 5cm in diameter) and made of light materials, such as stiffened and tightly woven silk, so they can oscillate fast enough to accurately reproduce high frequencies. The cones of tweeters are also often dome shaped (rather than cone shaped) to assist in dispersing high frequency sounds across a larger area, as the short wavelengths associated with the high frequencies are quite direct. This is because the diameter of the cone (or dome, as is often the case with a tweeter) acts as a gap in a barrier through which the waves diffract, and short wavelengths don't diffract much. Tweeters are sometimes covered by "horns" (like small trumpet bells) or relatively narrow diffraction slits, corresponding to the short wavelengths of high frequencies, for the purpose of dispersing the otherwise more direct high frequency sounds.

Woofers are speakers designed specifically for the reproduction of **low frequency** sounds. They are usually large (between 15 and 35cm in diameter), because we (humans) hear low frequencies with less loudness. Larger cones create larger compressions in the air, which we hear as louder sounds.

Mid-range speakers can be used to help "fill the gap" between the highest frequencies a woofer can produce and the lowest frequencies a tweeter can produce.

Woofer, tweeter and sometimes mid-range speakers are usually used in combination to reproduce the entire spectrum of frequencies audible to us (humans), at equal levels.

Loudspeakers: Baffles, Enclosures And Ports

When the front of a woofer cone moves outwards to produce a compression, a rarefaction is produced by the back of the cone behind the speaker. Two sound waves are produced when the cone of a loudspeaker oscillates; one in front of the cone and one behind the cone. The two waves are **out of phase** with each other, such that a compression of one occurs at the same time as a rarefaction of the other. Because the two waves are of equal frequency and intensity, but out of phase, they can interfere destructively where they meet and cancel each other out. This significantly reduces the effectiveness of the speaker.

Mounting the woofer in a **baffle** assists in decreasing the intensity of the wave coming from the rear of the cone, so where it meets the wave coming from the front of the cone there's less destructive interference. A "baffle" is simply a barrier, usually made from timber, with a correctly sized hole in which the speaker can be properly fitted and fixed in place. The larger and heavier the baffle, the more effective it is at reducing interference between waves from each side of the speaker. An "infinite baffle" is ideal, but impractical.

Mounting the baffle to the front of an **enclosure** (a box, with the front being the baffle) isolates the wave coming from the back of the speaker yet more, containing it almost completely, almost eliminating interference between waves from each side of the speaker. As with baffles, the larger and heavier the enclosure, the more effective it is. The inside of speaker enclosures is often filled with soft absorbent material (like cushion stuffing) to reduce reflections of waves and resonances inside the enclosure.

A **port** is a hole in either the front or back of an enclosure. In the port, extending inwards from the surface of the enclosure, is usually a pipe. The length of this pipe is carefully calculated such that:

- by the time the wave coming from the back of the speaker exits the pipe, it is **in phase** with the wave coming from the front, and therefore interferes with it **constructively**, reinforcing the intensity of the sound rather than cancelling it out, and
- the resonant frequencies of the enclosure (acting as a pipe closed at on end) match those of the woofer itself, thereby further reinforcing the intensity of sound it produces.

Properly designed baffles, enclosures and ports (especially) result in woofers being as effective as those more than twice in diameter.

Microphones

Microphones are “transducers”; devices that convert one form of energy into another. Microphones convert sound energy into electrical energy in the form of alternating current. This alternating current can then be amplified, recorded, processed, converted to radio waves and transmitted and received, and ultimately converted back to sound by loudspeakers.

Dynamic microphones work in the same way as loudspeakers, but in reverse. Sound waves cause a diaphragm to oscillate. Attached to the back of the diaphragm is a coil of wire, situated in a permanent magnetic field. When the coil oscillates relative to the fixed magnetic field, alternating current is induced in the coil. This alternating current represents the sound.

Ribbon or “velocity” microphones consist of a thin metal foil (a “ribbon”) situated between the poles of a magnet. Sound waves cause the ribbon to oscillate. As a result of oscillating within a fixed magnetic field, a varying voltage is generated in the ribbon, which causes an alternating current through it. This alternating current represents the sound.

Condenser or “electret” microphones operate as variable capacitors (“condenser” is another word for “capacitor”). Voltage is applied across the two capacitor plates. One of the plates is mechanically connected to a diaphragm. Sound waves cause the diaphragm, and hence the capacitor plate to which it’s connected, to oscillate. As one plate moves back-and-forth in relation to the other (which is fixed in its position), the potential difference (voltage) between the plates varies, causing an alternating current. This alternating current represents the sound.

Crystal microphones contain a piezoelectric crystal. Piezoelectric crystals generate voltage when deformed. Sound waves cause a diaphragm to oscillate. As it oscillates with the sound, the diaphragm applies varying pressure to the piezoelectric crystal which in turn generates a varying voltage, which causes an alternating current. This alternating current represents the sound.

Frequency response curves.

These are graphs. They show decibel level on the vertical axis and frequency on the horizontal. The frequency scale is normally logarithmic (**not linear**), so care is required when interpreting these graphs. The purpose of frequency response curves is to show how particular devices, usually speakers or microphones, respond to the range of frequencies audible by human ears (20 to 20000Hz).

Ideally, the frequency response curve for any device intended for the accurate reproduction of sound as we (humans) hear it will be “flat”, indicating that the device responds to all frequencies equally. Practically however, devices can only be designed and constructed to respond equally to a limited range of frequencies (hence the requirement for woofers and tweeters, as no single speaker responds as effectively to all frequencies as the combination of the two). The frequency response curve for a particular microphone will indicate its usefulness for recording high or low pitched musical instruments.