Transducers in Audio Electronics

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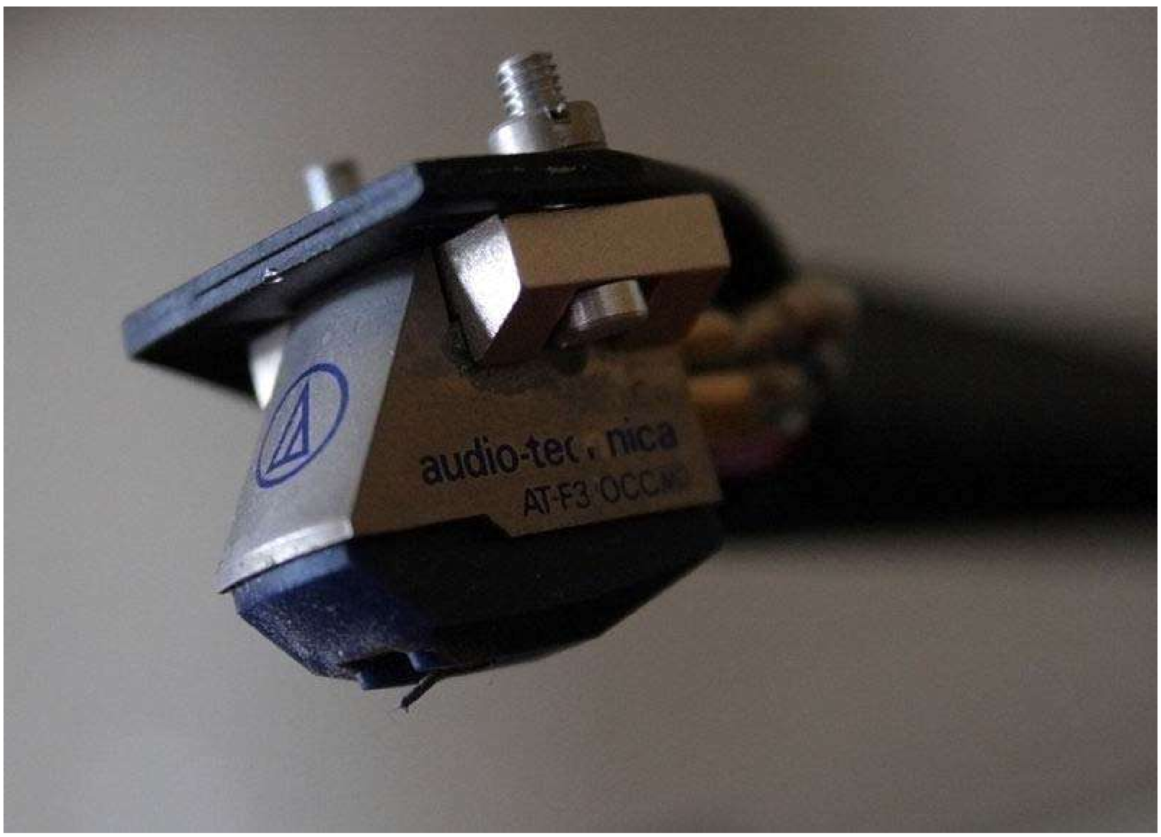
Magnetic Cartridge and Carbon Microphone

Magnetic cartridge



An Audio Technica AT-F3 MC cartridge

A **magnetic cartridge** is a transducer used for the playback of gramophone records on a turntable or phonograph. It converts mechanical vibrational energy from a stylus riding in a spiral record groove into an electrical signal that is subsequently amplified and then converted back to sound by a loudspeaker system.

Types

In high-fidelity systems, crystal and ceramic pickups have been replaced by the magnetic cartridge, using either a **moving magnet** or a **moving coil**.

Compared to the crystal and ceramic pickups, the magnetic cartridge usually gives improved playback fidelity and reduced record wear by tracking the groove with lighter pressure. Magnetic cartridges use lower tracking forces and thus reduce the potential for groove damage. They also have a lower output voltage than a crystal or ceramic pickup, in the range of only a few millivolts, thus requiring greater amplification.

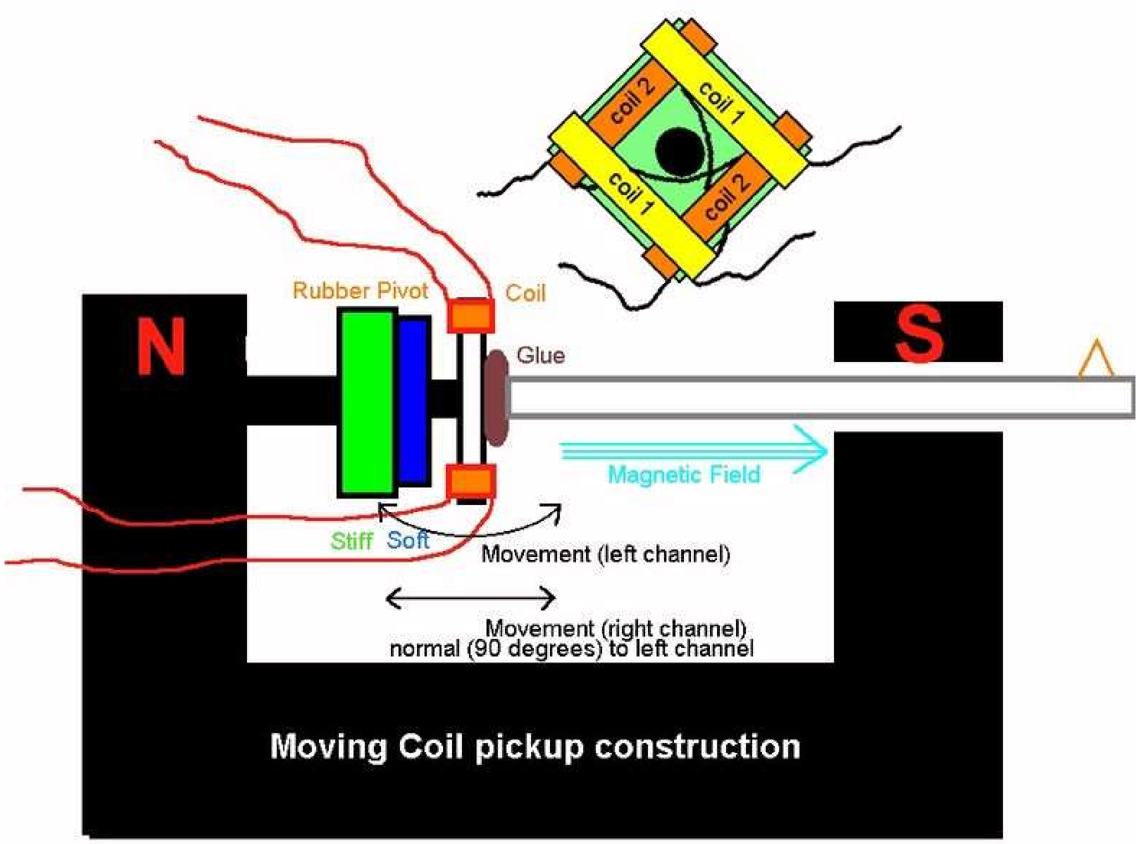
Moving Magnet (MM) and Moving Iron (MI) cartridges

In a moving magnet cartridge, the stylus cantilever carries a tiny permanent magnet, which is positioned between two sets of fixed coils (in a stereophonic cartridge), forming a tiny electromagnetic generator. As the magnet vibrates in response to the stylus following the record groove, it induces a tiny current in the coils.

Because the magnet is small and has little mass, and is not coupled mechanically to the generator (as in a ceramic cartridge), a properly adjusted stylus follows the groove more faithfully while requiring less tracking force (the downward pressure on the stylus).

Moving iron and induced magnet types (ADC being a well known example) have a moving piece of iron or other ferrous alloy is coupled to the cantilever (instead of a magnet), while a permanent, bigger magnet is over the coils, providing the necessary magnetic flux.

Moving Coil (MC) cartridges



The MC design is again a tiny electromagnetic generator, but (unlike an MM design) with the magnet and coils reversed: the coils are attached to the stylus, and move within the field of a permanent magnet. The coils are tiny and made from very fine wire.

Since the amount of windings that can be supported in such armature is small, the output voltage level is corresponsingly small. The resulting signal is only a few hundred microvolts, and thus more easily swamped by noise, induced hum, etc. Thus it is more challenging to design a preamplifier with the extremely low noise inputs needed for moving-coil cartridge, therefore a "step up transformer" is sometimes used instead.

However, there are available many "high output" moving coil cartridges that have output levels similar to MM cartridges.

Moving coil cartridges are extremely small precision instruments and are therefore generally expensive, but are frequently preferred by audiophiles due to a subjectively better performance.

Moving Micro Cross (MMC) cartridges

The MMC design was invented and patented by Bang & Olufsen. The MMC cartridge is variation of the Moving Iron (MI) design. Magnets and coils are stationary while a *micro cross* moves with the stylus, thereby varying the distances between the arms of the cross and the magnets. It is claimed that the MMC design allows for superior channel separation, since each channel's movements appear on a separate axis.

Moving Magnet vs. Moving Coil debate

Moving magnet cartridges are more commonly found at the 'lower-end' of the market, while the 'higher-end' tends to be dominated by moving coil designs. The debate as to whether MM or MC designs can ultimately produce the better sound is often heated and subjective. The distinction between the two is often blurred by cost and design considerations - i.e. can an MC cartridge requiring another step-up amplification outperform well made MM cartridges that need simpler front-end stages? Every now and then a design comes along to re-open this debate.

- MC cartridges offer very low inductance and impedance, which means that the effects of capacitance (in the cable that goes from the cartridge to the preamp) are negligible, unlike MM cartridges, who comparatively sport very high inductance and impedance. In the latter, cable capacitance can negatively affect the flatness of frequency response and linearlity of phase response. This would account for a sonic advantage to MC types.

- It is generally believed that MC cartridges sport lower moving masses. However, quality MM cartridges are able to sport as low as a moving mass than the best MC cartridges. For example, the state-of-the-art Technics EPC-100CMK4 with 0.055mg of effective tip mass, of moving magnet design. Comparatively, the popular Denon DL-301 moving coil cartridge has an effective tip mass of 0.270mg.

- To discriminate cartridges by engine (MC vs MM) overlooks the fact that the stylus tip shape and cantilever have a significant influence in the sound, and this may account for even more sonic differences than the engine type used.

"London Decca" Cartridges

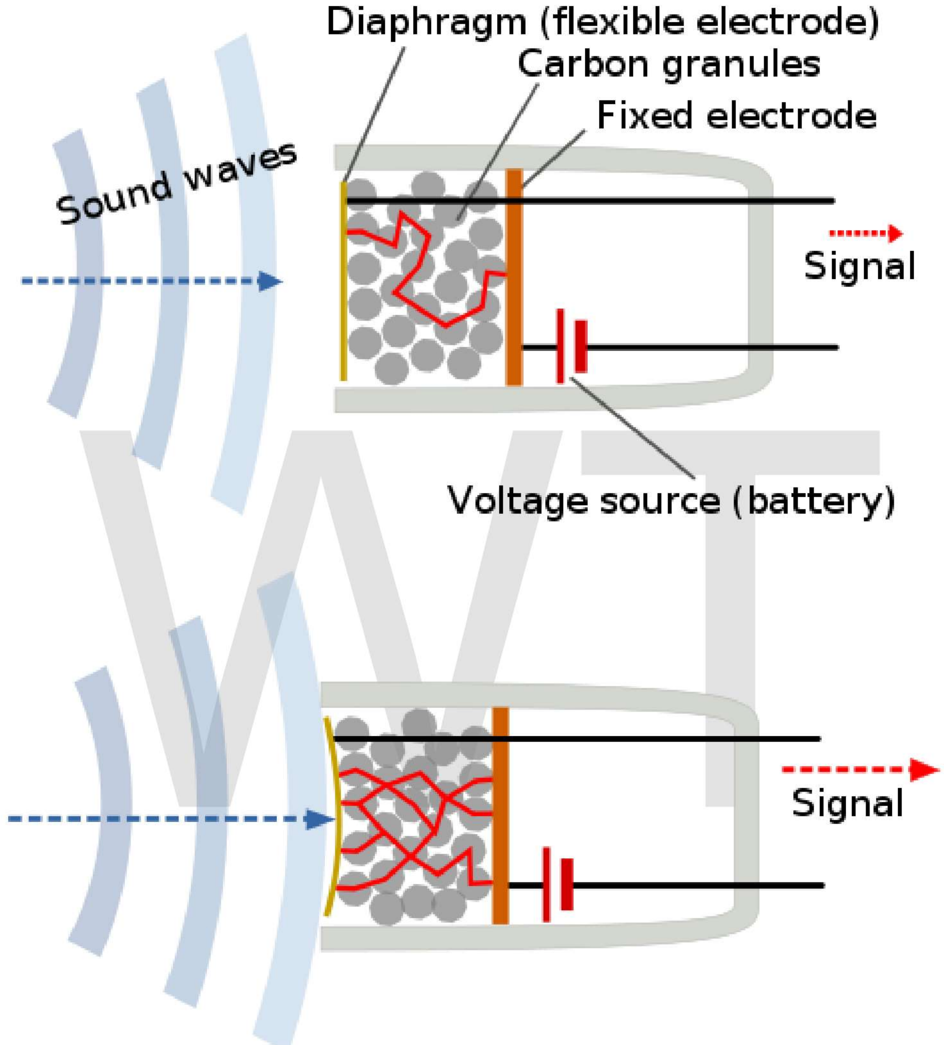
The Decca phono cartridges were a unique design, with fixed magnets and coils. The stylus shaft was composed of the diamond tip, a short piece of soft iron, and an L-shaped cantilever made of non-magnetic steel. Since the iron was placed very close to the tip (within 1 mm), the motions of the tip could be tracked very accurately. Decca engineers called this "positive scanning". Vertical and lateral compliance was controlled by the shape and thickness of the cantilever. Decca cartridges had a reputation for being very musical; however early versions required more tracking force than competitive designs - making record wear a concern.

Carbon microphone

Carbon microphone from Western Electric telephone.

The **carbon microphone**, also known as a **carbon button microphone** (or sometimes just a **button microphone**) or a **carbon transmitter**, is a sound-to-electrical signal transducer consisting of two metal plates separated by granules of carbon. One plate faces outward and acts as a diaphragm. When sound waves strike this plate, the pressure on the granules changes, which in turn changes the electrical resistance between the plates. (Higher pressure lowers the resistance as the granules are pushed closer together.) A direct current is passed from one plate to the other, and the changing resistance results in a changing current, which can be passed through a telephone system, or used in other ways in electronics systems to change the sound into an electrical signal.

Before the proliferation of vacuum tube amplifiers in the 1920s, carbon microphones were the only practical means of obtaining strong audio signals, and were widely used in telephone systems. Their low cost, inherently high output and "peaked" frequency response characteristic were well suited for this application, and their use in new telephone installations continued up to the 1980s, long after they had been replaced by other types of microphones in other applications. Carbon microphones were widely used in early AM radio broadcasting systems (usually modified telephone microphones), but their limited frequency response, as well as a fairly high noise level, led to their abandonment for that use by the late 1920s. They continued to be widely used for low-end public address, and military and amateur radio applications for some decades afterwards.



Operation of carbon microphone. When a sound wave presses on the conducting diagphragm, the granuales of carbon are pressed together and decrease their resistance.

History

The invention of the carbon microphone (then called a "transmitter") was claimed both by Thomas Alva Edison in March 1877 and separately by Emile Berliner who filed related patent applications in June 1877 and August 1879. The two sides fought a long legal battle over the patent rights. Ultimately a federal court awarded Edison full rights to the invention of the carbon microphone, saying "Edison preceded Berliner in the transmission of speech...The use of carbon in a transmitter is, beyond controversy, the invention of Edison" and the Berliner patent was ruled invalid. British courts also ruled in favor of Edison over Berliner. Having settled the Dowd suit (after Peter A. Dowd, agent of Western Union) out of court in 1881, Western Union left the telephone business, and sold Edison's patent rights and related assets to the Bell company in exchange for 20% of telephone rental receipts. Subsequently Bell telephones used the Bell receiver and the Edison transmitter. Later, carbon granules were used between carbon buttons. Carbon microphones were widely used in telephones in the United States from 1890 until the 1980s.

Carbon microphones used as amplifiers

One of the surprising attributes of carbon microphones is that they can actually be used as amplifiers. This capability was used in early telephone repeaters, making long distance phone calls possible in the era before vacuum tube amplifiers. In these repeaters, a magnetic telephone receiver (an electrical-to-mechanical transducer) was mechanically coupled to a carbon microphone. Because a carbon microphone works by varying a current passed through it, instead of generating a signal voltage as with most other microphone types, this arrangement could be used to boost weak signals and send them down the line. These amplifiers were mostly abandoned with the development of vacuum tubes, which offered higher gain and better sound quality. Even after vacuum tubes were in common use, carbon amplifiers continued to be used during the 1930s in portable audio equipment such as hearing aids. The Western Electric 65A carbon amplifier was 1.2" in diameter and 0.4" high and weighed less than 1.4 ounces. Such carbon amplifiers did not require the heavy bulky batteries and power supplies used by vacuum tube amplifiers. By the 1950s, carbon amplifiers for hearing aids had been replaced by miniature vacuum tubes (only to be shortly replaced by transistors). However, carbon amplifiers are still being produced and sold.

One illustration of the amplification provided by carbon microphones was the oscillation caused by feedback, that resulted in an audible squeal from the old "candlestick telephone" if its earphone was placed near the carbon microphone.

Early radio

Early AM radio transmitters relied on carbon microphones for voice modulation of the radio signal. In the first long-distance audio transmissions by Reginald Fessenden in 1906, a continuous wave from an Alexanderson alternator was fed directly to the transmitting antenna through a water-cooled carbon microphone. Later systems using vacuum tube oscillators often used the output from a carbon microphone to modulate the grid bias of the oscillator or output tube to achieve modulation.

Current usage

Apart from legacy telephone installations in Third World countries, carbon microphones are still used today in certain niche applications in the developed world. An example is the Shure 104c, which is still in demand because of its wide compatibility with existing equipment.

The principal advantage carbon microphones have over other microphone types is that they can produce high-level audio signals from very low DC voltages, without needing any form of additional amplification or batteries.

Old-fashioned carbon transmitter telephones are still found in remote locations at the end of very long telephone lines, whose voltage drop would disable an electronic telephone that lacks supplementary power. Most all-electronic telephones need at least 3 volts DC to work, whereas old-fashioned carbon transmitter telephones will continue to work down to a fraction of a volt. Electronic telephones also suffer from the so-called "cliff effect", whereby they abruptly stop working when the line voltage falls below the critical level, while a carbon microphone on the same line would simply have reduced output. In this situation, maintaining the older technology is seen to be more cost-effective than upgrading the line.

Carbon microphones are also widely used in safety-critical applications such as mining and chemical manufacture, where higher line voltages cannot be used, due to the risk of sparking and consequent explosions. Also, such installations often have a large existing communication infrastructure already based around carbon microphones, and again, it is often considerably cheaper to maintain the existing (if antiquated) structure, than to replace it with new technology.

Carbon-based telephone systems are also extremely resistant to damage from high-voltage transients such as those produced by lightning strikes and electromagnetic pulses of the type generated by nuclear explosions, and so are still maintained as backup communication systems in critical military installations.

Chapter-2

Headphones

Sennheiser HD555 headphones, used in audio production environments

Sony MDR-7506 headphones for professional audio

Headphones are a pair of small loudspeakers, or less commonly a single speaker, held close to a user's ears and connected to a signal source such as an audio amplifier, radio, CD player or portable media player. They are also known as **stereophones**, **headsets** or, colloquially **cans**. The in-ear versions are known as **earphones** or **earbuds**. In the context of telecommunication, the term headset is used to describe a combination of headphone and microphone used for two-way communication, for example with a telephone.

Applications

Headphones may be used both with fixed equipment such as CD or DVD players, home theater, personal computers and with portable devices (e.g. digital audio player/mp3 player, mobile phone, etc.). Cordless headphones are not connected via a wire, receiving a radio or infrared signal encoded using a radio or infrared transmission link, like FM, Bluetooth or Wi-Fi. These are actually made of powered receiver systems of which the headphone is only a component, these types of cordless headphones are being used more frequently with events such as a Silent disco or Silent Gig.

In the professional audio sector headphones are used in live situations by disc jockeys with a DJ mixer and sound engineers for monitoring signal sources. In radio studios, DJs use a pair of headphones when talking to the microphone while the speakers are turned off, to eliminate acoustic feedback and monitor their own voice. In studio recordings, musicians and singers use headphones to play along to a backing track. In the military, audio signals of many varieties are monitored using headphones.

Wired headphones are attached to an audio source. The most common connection standards are 6.35mm (¼″) and 3.5mm TRS connectors and sockets. The larger 6.35mm connector tending to be found on fixed location home or professional equipment. Sony introduced the smaller, and now widely used, 3.5mm "minijack" stereo connector in 1979, adapting the older monophonic 3.5mm connector for use with its Walkman portable stereo tape player and the 3.5mm connector remains the common connector for portable application today. Adapters are available for converting between 6.35mm and 3.5mm devices.

Electrical characteristics

Electrical characteristics of dynamic loudspeakers may be readily applied to headphones since most headphones are small dynamic loudspeakers.

Impedance

Headphones are available with low or high impedance measured at 1 kHz. Low-impedance headphones are in the range 75 to 150 ohms and high impedance headphones are about 600 ohms. High impedance headphones have been popular among tube amplifier aficionados, and in classroom or studio situations requiring many headphones connected in parallel to the same source. Low impedance headphones yield a louder sound from a standard headphone jack, and require less voltage to achieve a target sound pressure level—an important consideration for portable electronics.

Sensitivity

Sensitivity is a measure of a transducer's output when driven with a specific reference input. Headphone manufacturers often loosely use the term "efficiency" where sensitivity should be used. Headphone efficiency (power in/power out) *is* a type of sensitivity, but efficiency is usually not an important characteristic to measure for headphones.

Common "units" for headphone sensitivity are "dB/mW" and "dB/mV". This notation is an inappropriate simplification, but what these mean are dB SPL (sound pressure level) measured in a standard ear for a 1 kHz sinusoidal headphone input of either 1 milliwatt or one millivolt. Technical notation would be "dB ref. 20μPa/mW" or "dB ref. 20μPa/mV". One can convert between these two references if the impedance is known.

Types of headphones

The particular needs of the listener determine the choice of headphone. The need for portability indicates smaller, lighter headphones but can mean a compromise in fidelity. Headphones used as part of a home hi-fi do not have the same design constraints and can be larger and heavier. Generally, headphone form factors can be divided into four separate categories: *circumaural*, *supra-aural*, *earbud*, and *in-ear*.

Circumaural

Circumaural headphones have large pads that surround the outer ear.

Circumaural headphones (sometimes called **full size headphones**) have circular or ellipsoid earpads that encompass the ears. Because these headphones completely surround the ear, circumaural headphones can be designed to fully seal against the head to attenuate any intrusive external noise. Because of their size, circumaural headphones can be heavy and there are some sets which weigh over 500 grams (1 lb). Good headband and earpad design is required to reduce discomfort resulting from weight.

Supra-aural

A pair of supra-aural headphones.

Supra-aural headphones have pads that sit on top of the ears, rather than around them. They were commonly bundled with personal stereos during the 1980s. This type of headphone generally tends to be smaller and more lightweight than circumaural headphones, resulting in less attenuation of outside noise.

In-ear headphones

Earbuds

Earbuds / earphones

Earbuds or **earphones** are headphones of a much smaller size that are placed directly outside of the ear canal, but without fully enveloping it. They are generally inexpensive and are favored for their portability and convenience. Due to their inability to provide any isolation they are often used at higher volumes in order to drown out noise from the user's surroundings, which increases the risk of hearing-loss. During the 1990s and 2000s, earbuds became a common type bundled with personal music devices.

In-ear monitors

In-ear monitors extend into the ear canal, providing isolation from outside noise.

In-ear monitors (also known as **IEMs** or **canalphones**) are earphones that are inserted directly into the ear canal. Canalphones offer portability similar to earbuds, and also act as earplugs to block out environmental noise. There are two main types of IEMs: universal and custom. Universal canalphones provide one or more stock sleeve size(s) to fit various ear canals, which are commonly made out of silicone rubber, elastomer, or foam, for noise isolation. Custom canalphones are fitted to the ears of each individual. Castings of the ear canals are made and the manufacturer uses the castings to create custom-molded silicone rubber or elastomer plugs that provide added comfort and noise isolation. Because of the individualized labor involved, custom IEMs are more expensive than universal IEMs and resell value is very low as they are unlikely to fit other people.

Headset

A typical example of a headset used for voice chats.

A headset is a headphone combined with a microphone. Headsets provide the equivalent functionality of a telephone handset with hands-free operation. The most common uses for headsets are in console or PC gaming, Call centres and other telephone-intensive jobs and also for personal use at the computer to facilitate comfortable simultaneous conversation and typing. Headsets are made with either a single-earpiece (mono) or a double-earpiece (mono to both ears or stereo). The microphone arm of headsets is either an external microphone type where the microphone is held in front of the user's mouth, or a voicetube type where the microphone is housed in the earpiece and speech reaches it by means of a hollow tube.

Telephone headsets

Telephone headsets connect to a fixed-line telephone system. A telephone headset functions by replacing the handset of a telephone. All telephone headsets come in a standard 4P4C commonly called an RJ-9 connector.

For older models of telephones, the headset microphone impedance is different from that of the original handset, requiring a telephone amplifier for the telephone headset. A telephone amplifier provides basic pin-alignment similar to a telephone headset adaptor, but it also offers sound amplification for the microphone as well as the loudspeakers. Most models of telephone amplifiers offer volume control for loudspeaker as well as microphone, mute function and headset/handset switching. Telephone amplifiers are powered by batteries or AC adaptors.

Technology

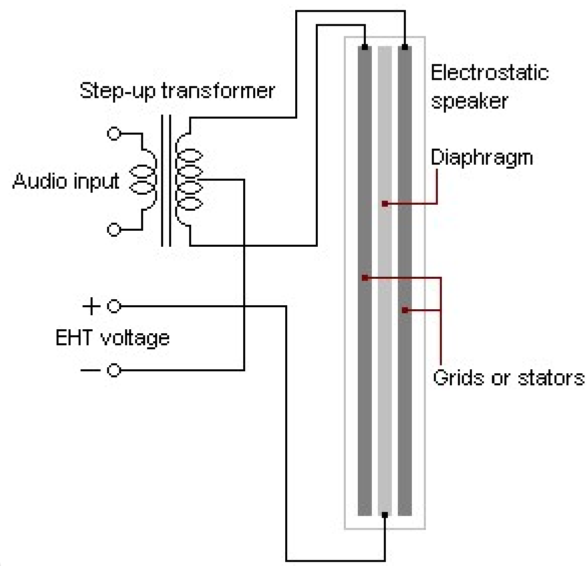
A typical moving-coil headphone transducer

Headphone transducers employ one or more of several methods of sound reproduction.

Moving-coil

The moving coil driver, more commonly referred to as a "dynamic" driver is the most common type used in headphones. The operating principle consists of a stationary magnetic element affixed to the frame of the headphone which sets up a static magnetic field. The magnetic element in headphones is typically composed of ferrite or neodymium. The diaphragm, typically fabricated from lightweight, high stiffness to mass ratio cellulose, polymer, carbon material, or the like, is attached to a coil of wire (voice coil) which is immersed in the static magnetic field of the stationary magnet. The diaphragm is actuated by the attached voice coil, when an audio current is passed through the coil. The alternating magnetic field produced by the current through the coil reacts against the static magnetic field in turn, causing the coil and attached diaphragm to move the air, thus producing sound. Modern moving-coil headphone drivers are derived from microphone **capsule** technology.

Electrostatic



Electrostatic loudspeaker diagram

Electrostatic drivers consist of a thin, electrically charged diaphragm, typically a coated PET film membrane, suspended between two perforated metal plates (electrodes). The electrical sound signal is applied to the electrodes creating an electrical field; depending on the polarity of this field, the diaphragm is drawn towards one of the plates. Air is forced through the perforations; combined with a continuously changing electrical signal driving the membrane, a sound wave is generated. Electrostatic headphones are usually more expensive than moving-coil ones, and are comparatively uncommon. In addition, a special amplifier is required to amplify the signal to deflect the membrane, which often requires electrical potentials in the range of 100 to 1000 volts.

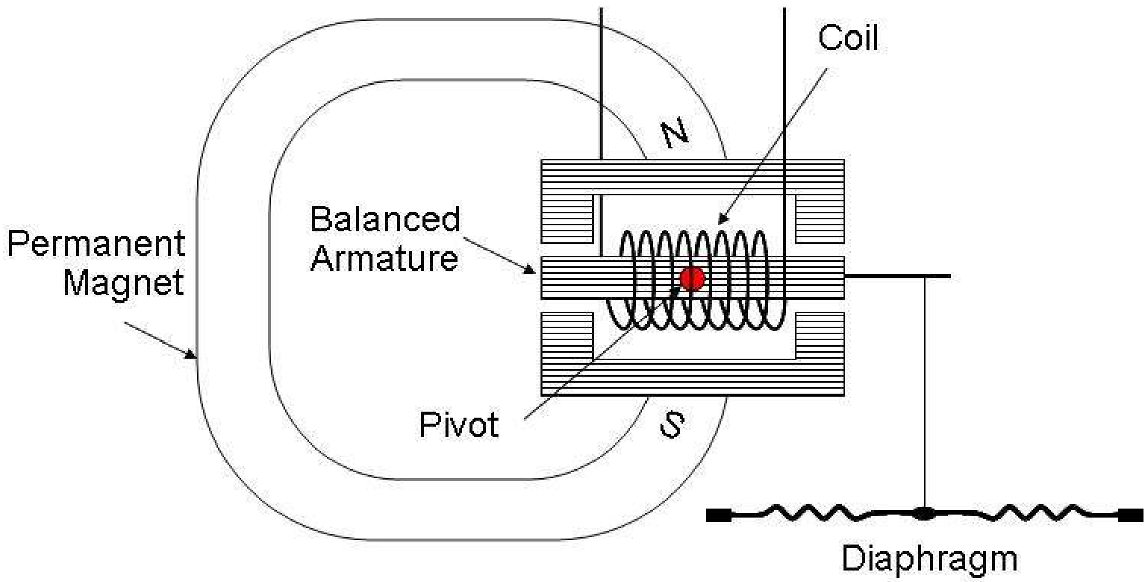
Due to the extremely thin and light diaphragm membrane, often only a few micrometers thick, and the complete absence of moving metalwork, the frequency response of electrostatic headphones usually extends well above the audible limit of approximately 20 kHz. The high frequency response means that the low midband distortion level is maintained to the top of the audible frequency band, which is generally not the case with moving coil drivers. Also, the frequency response peakiness regularly seen in the high frequency region with moving coil drivers is absent. The result is significantly better sound quality, if designed properly.

Electrostatic headphones are powered by anything from 100v to over 1kV, and are in proximity to a user's head. The usual method of making this safe is to limit the possible fault current to a low and safe value with resistors.

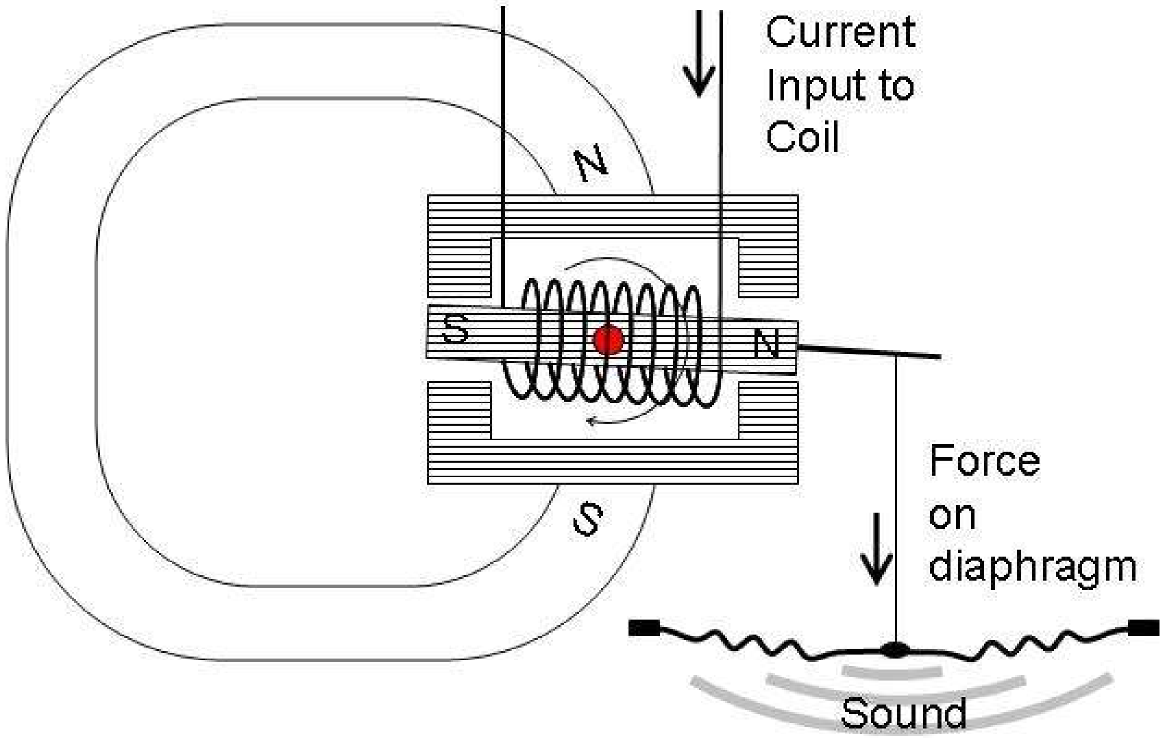
Electret

An electret driver functions along the same electromechanical means as an electrostatic driver. However the electret driver has a permanent charge built into it, whereas electrostatics have the charge applied to the driver by an external generator. Electret headphones, like electrostatics are relatively uncommon. They are also typically cheaper and lower in technical capability and fidelity than electrostatics.

Balanced armature



Balanced armature transducer with armature balanced and exerting no force on diaphragm



Balanced armature transducer with armature torqued and exerting a force on diaphragm

A balanced armature is a sound transducer design primarily intended to increase the electrical efficiency of the element by eliminating the stress on the diaphragm characteristic of many other magnetic transducer systems. As shown schematically in the first diagram, it consists of a moving magnetic armature that is pivoted so it can move in the field of the permanent magnet. When precisely centered in the magnetic field there is no net force on the armature, hence the term 'balanced.' As illustrated in the second diagram, when there is electric current through the coil, it magnetizes the armature one way or the other, causing it to rotate slightly one way or the other about the pivot thus moving the diaphragm to make sound.

The design is not mechanically stable; a slight imbalance makes the armature stick to one pole of the magnet. A fairly stiff restoring force is required to hold the armature in the 'balance' position. Although this reduces its efficiency, this design can still produce more sound from less power than any other. Popularized in the 1920s as Baldwin Mica Diaphragm radio headphones, balanced armature transducers were refined during World War II for use in military 'sound-powered' telephones. Some of these achieved astonishing electro-acoustic conversion efficiencies in the range 20% to 40% for narrow bandwidth voice signals.

Today they are typically used only in canalphones and hearing aids due to their diminutive size and low impedance. They generally are limited at the extremes of the hearing spectrum (e.g. below 20 Hz and above 16 kHz) and require a seal more than other types of drivers to deliver their full potential. Higher end models may employ multiple armature drivers, dividing the frequency ranges between them using a passive crossover network. A few combine an armature driver with a small moving-coil driver for increased bass output.

Other transducer technologies

Transducer technologies employed much less commonly for headphones include the Heil Air Motion Transformer (AMT); Piezoelectric film; Ribbon planar magnetic; Magnetostriction and Plasma-ionisation. The first Heil AMT headphone was marketed by ESS Laboratories and was essentially an ESS AMT tweeter from one of the company's speakers being driven at full range. Since the turn of the century, only Precide of Switzerland have manufactured an AMT headphone. Piezoelectric film headphones were first developed by Pioneer, their two models both used a flat sheet of film which limited the maximum volume of air that could be moved. Currently TakeT produce a piezoelectric film headphone which is shaped not unlike an AMT transducer but which like the driver Precide uses for their headphones, has a variation in the size of transducer folds over the diaphragm. It additionally incorporates a two way design by its inclusion of a dedicated tweeter/supertweeter panel. The folded shape of a diaphragm allows a transducer with a larger surface area to fit within smaller space constraints. This increases the total volume of air that can be moved on each excursion of the transducer given that radiating area.

Magnetostriction headphones, sometimes sold under the label of "Bonephones" are headphones that work via the transmission of vibrations against the side of head, transmitting the sound via bone conduction. This is particularly helpful in situations where the ears must be left unobstructed or when used by those who are deaf for reasons which do not affect the nervous apparatus of hearing. Magnetostriction headphones though, have greater limitations to their fidelity than conventional headphones which work via the normal workings of the ear. Additionally, there was also one attempt to market a plasma-ionisation headphone in the early 1990s by a French company called Plasmasonics. It is believed that there are no functioning examples left.

Benefits and limitations

Two Sony MDR-V6 headphones; one folded for travel

Headphones may be used to prevent other people from hearing the sound either for privacy or to prevent disturbance, as in listening in a public library. They can also provide a level of sound fidelity greater than loudspeakers of similar cost. Part of their ability to do so comes from the lack of any need to perform room correction treatments with headphones. High quality headphones can have an extremely flat low-frequency response down to 20 Hz within 3dB. However, rated frequency response distortion figures do not provide information on what character the sound reproduced at that frequency will be. Marketed claims such as 'frequency response 4 Hz to 20 kHz' are usually overstatements; the product's response at frequencies lower than 20 Hz is typically very small.

Headphones are also useful for video games that use 3D positional audio processing algorithms, as they allow players to better judge the position of an off-screen sound source (such as the footsteps of an opponent).

Although modern headphones have been particularly widely sold and used for listening to stereo recordings since the release of the Walkman, there is subjective debate regarding the nature of their reproduction of stereo sound. Stereo recordings represent the position of horizontal depth cues (stereo separation) via volume differences of the sound in question between the two channels. When the sounds from two speakers mix, they create the phase difference the brain uses to locate direction. Through most headphones, because the right and left channels do not combine in this manner, the illusion of the phantom center can be perceived as lost. Hard panned sounds will also only be heard only in one ear rather than from one side. This latter point is of particular importance for earlier stereo recordings which were less sophisticated, sometimes playing vocals through one channel and music through the other.

Binaural recordings use a different microphone technique to encode direction directly as phase, with very little amplitude difference (except above 2 kHz) often using a dummy head, and can produce a surprisingly life-like spatial impression through headphones. Commercial recordings almost always use stereo recording, because historically loudspeaker listening has been more popular than headphone listening. It is possible to change the spatial effects of stereo sound on headphones to better approximate the presentation of speaker reproduction by using frequency-dependent cross-feed between the channels, or—better still—a Blumlein shuffler (a custom EQ employed to augment the low-frequency content of the difference information in a stereo signal). While cross-feed can reduce the unpleasantness that some listeners find with hard panned stereo in headphones, the use of a dummy head during recording, with artificial pinnae, can allow on playback through headphones, the experience of hearing the performance as though situated in the position of the dummy head. Optimal sound is achieved when the dummy head matches the listener's head, since pinnae vary greatly in size and shape.

Headsets can have ergonomic benefits over traditional telephone handsets. They allow call center agents to maintain better posture instead of tilting their head sideways to cradle a handset.

Over time, headphone cables fail. The common scenario in which a replacement might need to be purchased is the physical breakdown of copper wiring at junction points on the cord (at the TRS jack, or at the point of connection to the headphone). These are the sites of greatest and most stressful motion on a cord and so they are typically fitted with some kind of strain relief.

Dangers and volume solutions

Using headphones at a sufficiently high volume level may cause temporary or permanent hearing impairment or deafness due to an effect called "masking." The headphone volume has to compete with the background noise, especially in excessively loud places such as subway stations, aircraft, and large crowds. Extended periods of the excessively loud volume may be damaging; however, one hearing expert found that "fewer than 5% of users select volume levels and listen frequently enough to risk hearing loss." Some manufacturers of portable music devices have attempted to introduce safety circuitry that limited output volume or warned the user when dangerous volume was being used, but the concept has been rejected by most of the buying public, which favors the personal choice of high volume. Koss introduced the "Safelite" line of cassette players in 1983 with such a warning light. The line was discontinued two years later for lack of interest.

The government of France has imposed a limit on all music players sold in the country: they must not be capable of producing more than 100dBA (the threshold of hearing damage during extended listening is 80dB, and the threshold of pain, or theoretically of immediate hearing loss, is 130dB). Many users decry this as an infringement on personal choice, and use third-party options to reverse the volume limits placed on such devices. Still, other users welcome the government's pro-health stance.

Other risks arise from the reduced awareness of external sounds—some jurisdictions regulate the use of headphones while driving vehicles, usually limiting the use of earphones to a single ear. The complete isolation from outside noise can be a hazard in itself, as a user could miss the sound of a car horn and walk into traffic with fatal consequences. Losing situational awareness can also lead to theft, particularly in busy environments where bumping into another person would be ignored, e.g., subway stations.

Motorcycle and other power-sport riders benefit by wearing foam earplugs when legal to do so to avoid excessive road, engine and wind noise, but their ability to hear music and intercom speech is actually enhanced when doing so. The ear can normally detect 1-billionth of an atmosphere of sound pressure level, hence it is incredibly sensitive. At very high sound pressure levels, muscles in the ear tighten the tympanic membrane and this leads to a small change in the geometry of the ossicles and stirrup that results in lower transfer of force to the oval window of the inner ear. Since earplugs reduce the noise in the auditory canal, this protective mechanism is less likely to trigger, and full sensitivity of the ear is maintained. This technique allows excellent hearing of speech, music and most external sounds at sustainable levels without hearing damage.

Listening to music through headphones while exercising can be dangerous. Blood may be diverted from the ears to the limbs leaving the inner ear more vulnerable to damage from loud sound. A Finnish study recommended that exercisers should set their headphone volumes to half of their normal loudness and only use them for a half hour.

Chapter-3

Pickup (Music Technology)

Three magnetic pickups on a Peavey Raptor with the pickup configuration of a fat-strat (H-S-S). The bridge (right) pickup is a humbucker and the neck (left) and middle pickups are single coils.

A **pickup** device acts as a transducer that captures mechanical vibrations, usually from suitably-equipped stringed instruments such as the electric guitar, electric bass guitar, Chapman Stick, Kelstone or electric violin, and converts them to an electrical signal which can then be amplified, recorded and broadcast.

Magnetic pickups

A magnetic pickup consists of a permanent magnet such as an AlNiCo, wrapped with a coil of a few thousand turns of fine enameled copper wire. The pickup is most often mounted on the body of the instrument, but can be attached to the bridge, neck and/or pickguard, as on many electro-acoustic archtop jazz guitars and string basses. The vibration of the nearby soft-magnetic strings modulates the magnetic flux linking the coil, thereby inducing an alternating current through the coil of wire. This signal is then carried to amplification or recording equipment via a cable. There may also be an internal preamplifier stage between the pickup and cable. More generally, the pickup operation can be described using the concept of a magnetic circuit, in which the motion of the string varies the magnetic reluctance in the circuit created by the permanent magnet.

Output

The output voltage of pickups varies between 100 mV rms to over 1 V rms for some of the higher output types. Some high-output pickups achieve this by employing very strong magnets, thus creating more flux and thereby more output. This can be detrimental to the final sound because the magnet's pull on the strings can cause problems with intonation as well as damp the strings and reduce sustain. Other high-output pickups have more turns of wire to increase the voltage generated by the string's movement. However, this also increases the pickup's output resistance/impedance, which can affect high frequencies if the pickup is not isolated by a buffer amplifier or a DI unit.

Pickup sound

Single coil pickups, Fender Stratocaster (1963)

The turns of wire in proximity to each other have an equivalent self-capacitance which, when added to any cable capacitance present, resonates with the inductance of the winding. This resonance can accentuate certain frequencies, giving the pickup a characteristic tonal quality. The more turns of wire in the winding, the higher the output voltage but the lower this resonance frequency. The inductive source impedance inherent in this type of transducer makes it less linear than other forms of pickups, such as piezoelectric or optical. The tonal quality produced by this nonlinearity is, however, subject to taste, and may therefore also be considered by some guitarists and luthiers to be aesthetically superior to that of a more linear transducer.

The external load usually consists of resistance (the volume and tone potentiometer in the guitar, and any resistance to ground at the amplifier input) and capacitance between the hot lead and shield in the guitar cable. The electric cable also has a capacitance which can be a significant portion of the overall system capacitance. This arrangement of passive components forms a resistively-damped second-order low-pass filter. Pickups are usually designed to feed a high input impedance, typically a megohm or more, and a low impedance load will reduce the high-frequency response of the pickup because of the filtering effect of the inductance.

Humbuckers

PRS's Dragon humbucker

Single coil pickups also act like an antenna and are prone to pick up mains hum (nuisance electromagnetic interference generated by electrical power cables, power transformers, and fluorescent light ballasts in the area) along with the musical signal. Mains hum consists of a fundamental signal at a nominal 50 or 60 Hz, depending on local alternating current frequency, and usually some harmonic content. The changing magnetic flux caused by the mains current links with the windings of the pickup, inducing a voltage by transformer action. The pickups also are sensitive to the electromagnetic field from nearby cathode ray tubes in video monitors or televisions.

To overcome this effect, the humbucking pickup was invented by Raymond J. Butts, but Seth Lover of Gibson was also working on one himself. Ray Butts initially developed one on his own and later worked with Gretsch. Who developed it first is a matter of some debate, but Ray Butts was awarded the first patent (U.S. Patent 2,892,371)and Seth Lover came next (U.S. Patent 2,896,491). Ultimately, both men developed essentially the same concept, but Ray Butts was never recognized as the true First. Another way to reduce nuisance hum when recording with humbuckers or single coils is to aim the instrument's neck in a different direction to find a location that minimizes the received noise signal.

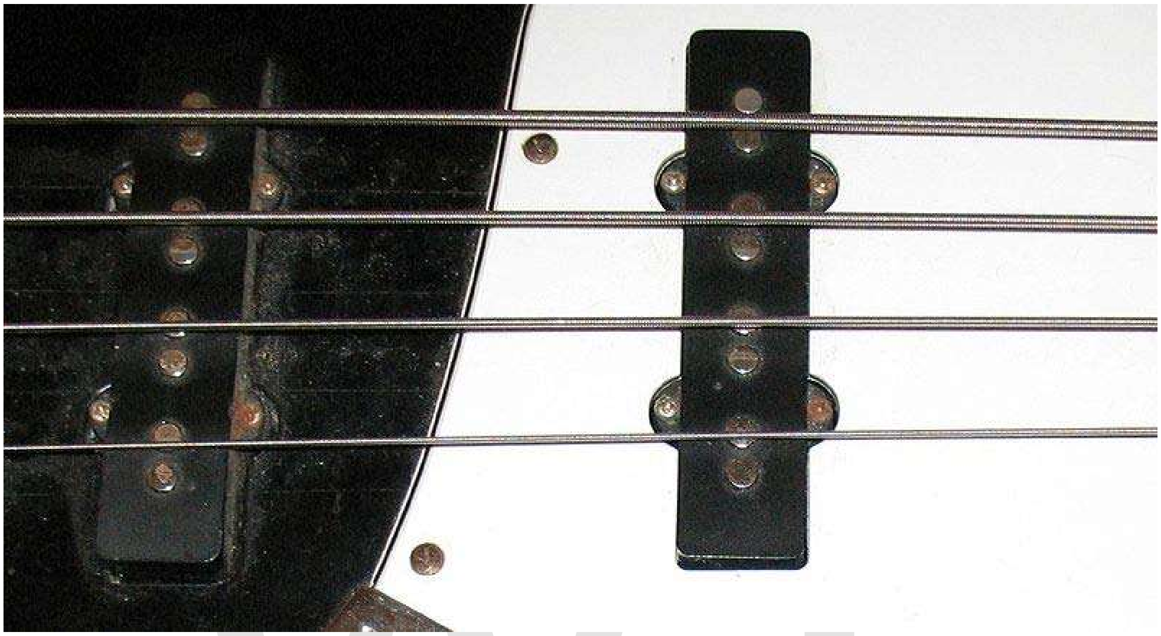
A humbucking pickup, shown in the image on the right, is composed of two coils. Each coil is wound reverse to one another. However, the six magnetic poles are opposite in polarity in each winding. Since ambient hum from power-supply transformers, radio frequencies, or electrical devices reaches the coils as common-mode noise, it induces an electrical current of equal magnitude in each coil. Because the windings are reversed in each pickup coil, the electro-magnetic interference sine wave signals in each pickup are equal and 180 degrees out of phase to one another, resulting in them canceling each other. However, the signal from the guitar string is doubled, due to the phase reversal caused by the out of phase magnets. The magnets being out of phase in conjunction with the coil windings being out of phase put the guitar string signal from each pickup in phase with one another. When the two in-phase guitar string sine wave signals meet, the amplitude of the wave doubles, and as does the signal strength.

When wired in series, as is most common, the overall inductance of the pickup is increased, which lowers its resonance frequency and attenuates the higher frequencies, giving a less trebly tone (i.e., "fatter") than either of the two component single-coil pickups would give alone. Because the two coils are wired in series, the resulting signal that is output by the pickup is larger in amplitude, thus more able to overdrive the early stages of the amplifier. This is the essence of the "humbucker tone."

An alternative wiring places the coils in *buck* parallel. The equal common-mode mains hum interference cancels, while the string variation signal sums. This method has a more neutral effect on resonant frequency: mutual capacitance is doubled (which if inductance were constant would result in a lowering of resonant frequency), and inductance is halved (which would raise the resonant frequency without the capacitance change). The net is no change in resonant frequency. This pickup wiring is rare, as guitarists have come to expect that humbucking 'has a sound', and is not neutral. On fine jazz guitars, the parallel wiring will produce significantly cleaner sound, as the lowered source impedance will drive capacitive cable with lower high frequency attenuation.

A side-by-side humbucking pickup senses a wider section of the string (has a wider aperture) than a single-coil pickup. This affects tone. By picking up a larger portion of the vibrating string more lower harmonics are present in the signal produced by the pickup resulting in a "fatter" tone. Stacked humbuckers have the narrower aperture of a single coil and sound closer to one.

Construction



Split pole pickups, Fender Jazz Bass

Pickups have magnetic polepieces (with the notable exceptions of rail and lipstick tube pickups) — one or two for each string. These polepiece centers should be perfectly aligned with the strings, or else sound will be suboptimal as the pickup would capture only a part of the string's vibrational energy. An exception to this rule are the J- and P-style pickups (found on the Fender Jazz Bass and Precision Bass, respectively) where the two polepieces per string are positioned on either side of each string.

String spacing is not even on most guitars: it starts with minimal spacing at nut and ends with maximal at bridge. Thus, bridge, neck and middle pickups usually have different polepiece spacing on the same guitar.

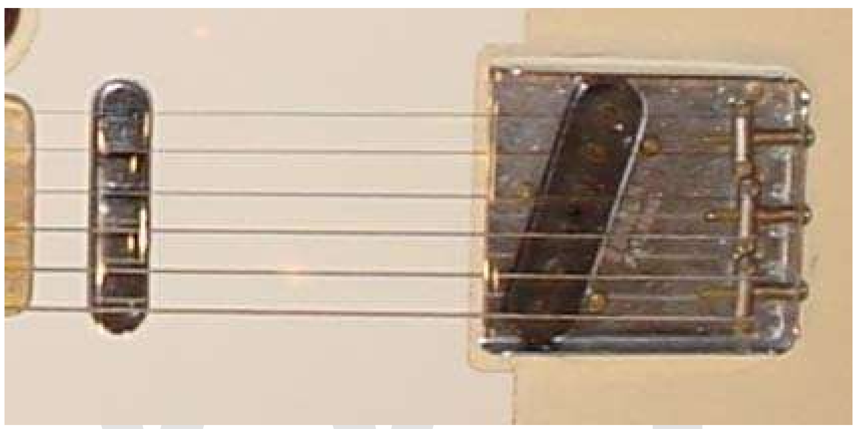
There are several standards on pickup sizes and string spacing between the poles. Spacing is measured either as a distance between 1st to 6th polepieces' centers (this is also called "E-to-E" spacing), or as a distance between adjacent polepieces' centers.

	1st-to-6th	**Adjacent**
Standard spacing (Vintage Gibson guitars)	1.90" 48 mm	0.380" 9.6 mm
F-spacing (Most Fender guitars, modern Gibson, Floyd Rose bridges)	2.01" 51 mm	0.402" 10.2 mm
Very close to bridge, extra pickup	2.060"	0.412"

(Roland guitar synth hex pickups)	52.3 mm	10.5 mm
Telecaster spacing (Fender Telecaster guitars)	2.165" 55 mm	0.433" 11 mm
Steinberger Spirit GT-Pro spacing (may be typical for other Steinberger guitars)	2.362" 60 mm	0.3937" 10 mm

Notation

Usually an electric guitar has more than one magnetic pickup. A combination of pickups is called a *pickup configuration*. It is usually notated by just writing out the pickup types, using "S" for single-coil and "H" for humbucker, in order from bridge pickup to neck pickup. Popular pickup configurations include:



S-S (Fender Telecaster)

S-S-S (Fender Stratocaster)

H-H (Gibson Les Paul, Fender Double Fat Stratocaster)

H-S-S (Peavey Raptor EXP)

H-S-H (Superstrats)

Less frequently found configurations are:

- **S** (Fender Esquire, early Gibson Les Paul Juniors, Gibson Melody Maker, some Telecasters)
- **H** (some hollow body guitars like Gibson ES-165 Herb Ellis; minimalistic rock/metal guitars like Kramer Baretta; later Les Paul Juniors)
- **H-S** (minimalistic guitars like Hamer Californian Deluxe and Les Paul BFG, entry-level guitars like Squier '51)

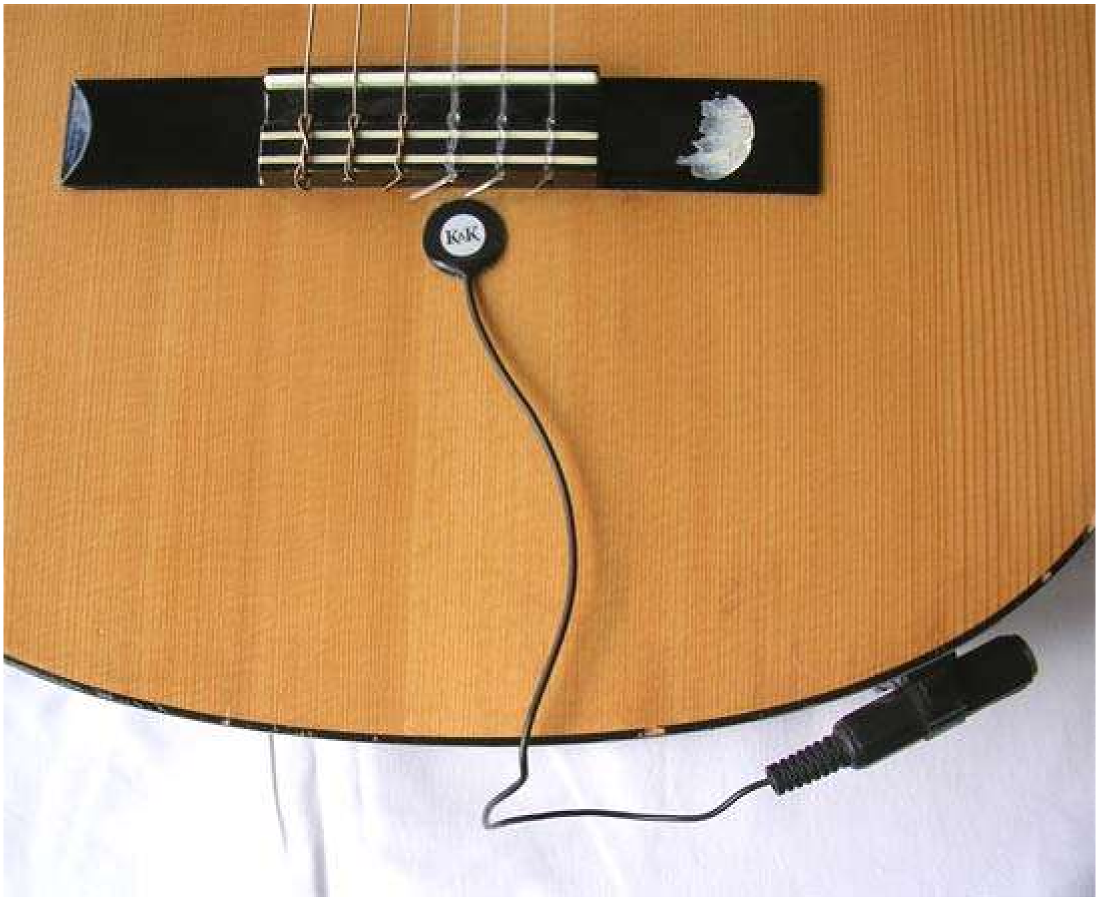
Examples of rare configurations that only a few particular models use include:

- **H-H-H** (some Gibson Les Paul Gold Tops and Customs, Gibson SG-3, Gibson ES-5 Switchmaster (after 1957), Kramer Jersey Star, Ibanez Destroyer, Ibanez PGM200)
- **H-S-S-H** (Music Man Steve Morse Signature)
- **H-H-S** (some ESP Stephen Carpenter Models and Alembic Jerry Garcia Models)
- **H-H** (some early seven-string ESP horizon models. But unlike the regular H-H setup, the humbuckers are in the Middle and bridge position)
- **S-H-S** (Fender Wayne Kramer Signature)

- **S-H** (some Telecasters have this configuration)

Piezoelectric pickups

Sensors



A piezoelectric pickup on an acoustic guitar.

A piezoelectric Dual pickup in Australia.

Many semi-acoustic and acoustic guitars, and some electric guitars and basses, have been fitted with piezoelectric pickups instead of, or in addition to, magnetic pickups. These have a very different sound, and also have the advantage of not picking up any other magnetic fields, such as mains hum and feedback from monitoring loops. In hybrid guitars, this system allows switching between magnetic pickup and piezo sounds, or simultaneously blending the output. Solid bodied guitars with only a piezo pickup are known as silent guitars, which are usually used for practicing by acoustic guitarists. Piezo pickups can be also built into electric guitar bridges for conversion of existing instruments.

Most pickups for traditionally acoustic instruments, such as cello, violin, and double bass, are piezoelectric pickups. These usually fit onto the bridge, where the strongest vibration is present.

Preamps

Piezoelectric pickups have a very high output impedance and appear as a capacitance in series with a voltage source. They therefore often have an instrument-mounted buffer amplifier fitted to maximize full frequency response. Piezo pickups are usually mounted under the bridge and sometimes form part of the bridge assembly itself.

The piezo pickup gives a very wide frequency range output compared to the magnetic types and can give large amplitude signals from the strings. For this reason, the buffer amplifier is often powered from relatively high voltage rails (about ±9 V) to avoid distortion due to clipping. Some musicians prefer a preamp that isn't as linear (like a single-FET amplifier) in which the clipping is "softer". Such an amplifier starts to distort sooner, which makes the distortion less *"buzzy"* and less audible than a more linear, but less forgiving op-amp. However, at least one study indicates that most people can not tell the difference between FET and op-amp circuits in blind listening comparisons of electric instrument preamps, a finding which correlates with results of formal studies done in other types of audio devices. Sometimes, piezoelectric pickups are used in conjunction with magnetic types to give a wider range of available sounds.

Other transducers

Some pickup products are installed and used similarly to piezoelectric pickups, but use different underlying technology, for instance electret or condenser microphone technology.

Multi-transducer pickups

Hexaphonic pickups (also called *divided pickups* and *polyphonic pickups*) have a separate output for each string (*Hexaphonic* assumes six strings, as on a guitar). This allows for separate processing and amplification for each string. It also allows a converter to sense the pitch coming from individual string signals for producing note commands, typically according to the MIDI (musical instrument digital interface) protocol. A hexaphonic pickup and a converter are usually components of a guitar/synthesizer.

Such pickups are uncommon (compared to normal ones), and only a few notable models exist. Hexaphonic pickups can be either magnetic or piezoelectric.

Optical

Optical pickups are a fairly recent development that work by sensing the interruption of a light beam by the string. The light source is usually an LED, and the detector is a photodiode or phototransistor. These pickups have complete insensitivity to magnetic or electric interference and also have a very wide and flat frequency response, unlike magnetic pickups.

Optical pickup guitars were first shown at the 1969 NAMM in Chicago, by Ron Hoag.

Active and passive pickups

Pickups can be either active or passive. Pickups, apart from optical types, are inherently passive transducers. "Active" pickups incorporate electronic circuitry to modify the signal. "Passive" pickups are usually wire wound around a magnet, and are the most common type used. They can generate electric potential without need for external power, though their output is relatively low, and the harmonic content of output depends greatly on the winding.

EMG 81 and EMG 85 — pair of popular active pickups

Active pickups require an electrical source of energy (usually one or two 9V batteries) to operate and include an electronic preamp, active filters, active EQ and other sound-shaping features. They can sometimes give much higher possible output. They also are less affected in tone by varying lengths of the electric cable connecting the guitar to the amplifier, and amplifier input characteristics. Magnetic pickups used with 'active' circuitry usually feature a lower inductance (and initially lower output) winding that tends to give a flatter frequency response curve.

The disadvantages of active pickup systems (pickups->preamp) are the power source (usually either a battery or phantom power) and higher cost. They are more popular on electric bass, because of their solid tone and improved clarity; most high-end basses feature active pickup systems. Most piezoelectric and all optical pickups are active and include some sort of preamp.

The main advantages of active bass pickup systems is their cleaner, clearer more "Hi Fi" sound. Many players, notably Stanley Clarke, Flea, Victor Wooten, Abraham Laboriel and Doug Wimbish have used active bass pickups to produce their characteristic bass tones.

They also allow more "headroom" and dynamic range. Good quality active systems produce less noise and hum compared to their passive counterparts.

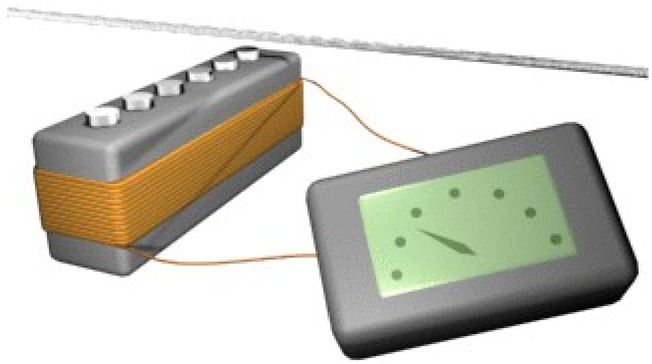
Stereo and multiple pickups with individual outputs

Rickenbacker was the first manufacturer who began producing stereo bass guitars with a stereo output for each pickup section. The neck pickup had one output and the bridge pickup had one. Also Teisco produced a guitar with a stereo option. Teisco divided the two sections in the upper three strings and the lower three strings for each individual output. The Gittler guitar was an experimental guitar with six pickups, one for each string, with six outputs. The Go! Team has modified a Fender Telecaster with an additional rotated pickup for the upper string, causing a simulation of a one string bass sound.

Chapter-4

Single Coil

This image shows three single coil pickups on a Stratocaster guitar. Left to right: bridge, middle and neck pickups.

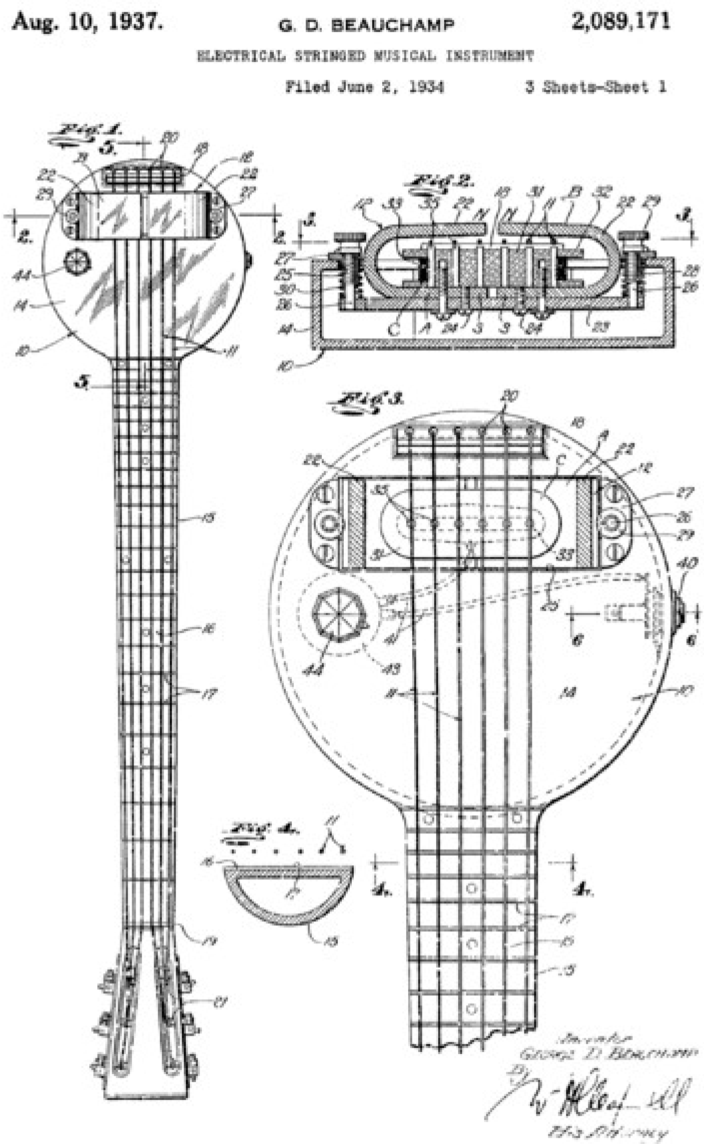


String effect on a single coil (electric guitar). The coil is connected to a multimeter that indicates the voltage changes when the string moves. This signal is normally sent to the amplifier.

A **single coil pickup** is a type of magnetic transducer, or pickup, for the electric guitar and the electric bass. It electromagnetically converts the vibration of the strings to an electric signal. Single coil pickups are one of the two most popular designs, along with dual-coil or "humbucking" pickups.

History

Beauchamp



Sketch of Rickenbacker "frying pan" lap steel guitar from 1934 patent application.

In the mid 1920s George Beauchamp, a Los Angeles, California guitarist, began experimentation with electric amplification of the guitar. Originally using a phonograph pickup assembly, Beauchamp began testing many different combinations of coils and magnets hoping to create the first electromagnetic guitar pickup. He wound his earliest coils using a motor out of a washing machine, later on switching to a sewing machine motor, and eventually using single coiled magnets.

Beauchamp was backed in his efforts by Adolph Rickenbacker, an engineer and wealthy owner of a successful tool and die business. Beauchamp eventually produced the first successful single coil pickup. The pickup consisted of two massive "U" shaped magnets and one coil and was known as the "horseshoe pickup". The two horseshoe-shaped magnets surrounded the strings that passed over a single core plate (or blade) in the center of the coil.

Beauchamp outfitted the pickup in a custom built lap slide guitar. The production model based on this prototype became the Hawaiian Electro lap steel guitar, nicknamed the "Frying Pan" for its round, flat body.

In 1931 Beauchamp founded the Ro-Pat-In Company with Rickenbacker and his associates. Ro-Pat-In eventually became The Electro String Instrument Corporation and subsequently the Rickenbacker International Corporation. The company introduced its first "Electro-String Instruments" to the public in 1932.

Gibson

The Gibson Guitar Corporation introduced the "bar pickup" in 1935 for its new line of Hawaiian lap steel guitars. The pickup's basic construction is that of a metal blade inserted through the coil as a shared pole piece for all the strings. A pair of large flat magnets were fastened below the coil assembly.

In 1936 Gibson introduced the ES-150, its first electric Spanish styled guitar. The ES-150 was outfitted with the bar pickup. Jazz guitar innovator, Charlie Christian, began playing an ES-150 in the late 1930s with the Benny Goodman Orchestra. This caused the popularity of the electrified guitar to soar. Due to Christian's close association with the ES-150 it began being referred to as the "Charlie Christian Model" and Gibson's now famous bar pickup as the "Charlie Christian pickup" or "CC unit".

Sound

The sound of a single coil pickup can range from the strong, fat midrange sound of the Gibson P-90 to the bright and clear Fender Telecaster single-coil tone.

Common designs

Gibson P-90

The P-90 is a single-coil pickup designed by the Gibson Guitar Corporation. These pickups have a large, flat coil with adjustable steel screws as pole pieces, and a pair of flat alnico bar magnets lying under the coil bobbin. The adjustable pole pieces pick up the magnetism from the magnets. Moving the screw closer or further away from the magnet determines signal strength, thus tone as well. There are two variations of P-90 pickup that differ mainly by mounting options:

Gibson P-90 soap bar

- **Soap bar** casing has true rectangular shape and the mounting screws are contained within the coil perimeter, positioned between the pole pieces, between strings 2-3 and 4-5, thus creating irregular and somewhat unusual pattern. Occasionally, they are mistaken for pole pieces; thus, the P-90 is sometimes erroneously said to have eight pole pieces. The "soap bar" nickname most probably comes from its predominantly rectangular shape and proportions resembling a bar of soap, and the fact that the first P-90s on the original Gibson Les Paul Model of 1952 were white.

P90 dog ear

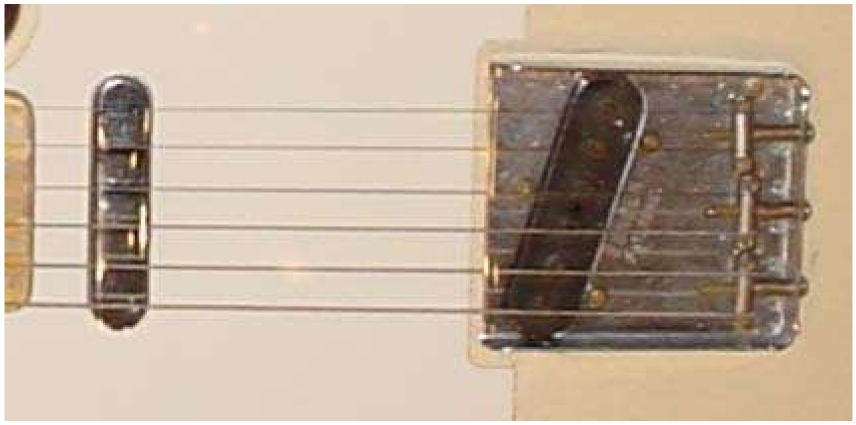
- **Dog ear** is a casing type with extensions at both sides of pickup that somewhat resemble dog's ears. These are extensions of the predominantly rectangular cover that encompass the outlying mounting screws. Dog-ear P-90 pickups were commonly mounted on Gibson's hollowbody guitars like the ES-330 and occasionally on solid body models like the Les Paul Junior. The same pickups were also available on Epiphone models (since Gibson was building Epiphone guitars in the 1950s) and the design is best remembered for its appearance on the hollow body Epiphone Casino of the mid to late 1960s.

The sound of a P-90 is somewhat brighter and more transparent than Gibson's later humbucker pickup, and every bit as crisp and snappy as Fender's single-coil pickups despite its high output and big sound.

Despite its tonal qualities the P-90 fell out of favor with Gibson in the early 1950s as a consequence of guitar players complaining about the amount of hum (noise) it put out. Gibson employee Seth Lover solved the hum problem by designing a hum-canceling pickup known as a humbucker, it was supposed to sound like a P-90 but in fact has quite a different sound. It nevertheless became Gibson's mainstay pickup from that point on. The P-90 likely did not become as popular for that reason, although many guitarists still prefer the tone of the P-90.

The hum problem proved extremely difficult to solve and despite numerous attempts by Gibson with their P-100, and the larger aftermarket pickup manufacturers with their stacked and sidewinders noiseless designs, hum-canceling P-90 pickups lost most of their favored tonal characteristics and generally did not gain acceptance among guitar players.

Telecaster design



Two pickups on a Telecaster

The Fender Telecaster features two single coils. The neck pickup produces a mellower sound, while the bridge pickup produces an extremely twangy, sharp tone with exaggerated treble response, because the bridge pickup is mounted on a steel plate. These design elements allow musicians to emulate steel guitar sounds, making it particularly appropriate for country music.

Pickups are selected with a three-position switch, and two wiring schemes exist:

- **Vintage**: 1) neck pickup with treble cutoff for a bassier sound; 2) neck pickup only; 3) bridge pickup only.
- **Modern**: 1) neck pickup only, with no treble cutoff; 2) neck and bridge; 3) bridge pickup only.

The Fender Esquire has a variation to the Vintage wiring scheme by using the scheme on a single pickup. This gives a treble cutoff in the first position, normal in the middle position, and a tone control cutoff in the third position.

Stratocaster design

Stratocaster pickups, viewed along the neck profile. Note that the poles are of different heights.

The traditional Stratocaster design guitar features three single coils. The guitarist can control which pickup or combination of pickups are selected with a lever switch. The pickup positions are usually referred to as the bridge, middle and neck pickups based on their proximity to those parts of the instrument.

Pickup position, number of coil winds, type of magnet wire, magnets and other factors shape the sound. A given pickup in the neck position will give louder, mellower and warmer sound, while an identical pickup in the bridge position will have lower output and produce a brighter, sharper sound. The reason the neck pickup has the most output is that the string's vibration has a higher amplitude at the neck position, being near the middle of the string length. Some manufacturers overwind the bridge pickup for more output to compensate for this difference.

The magnet poles have different heights. This is called a **magnet stagger** and is done to compensate for the different outputs of the string for two reasons. The first reason is that the fretboard has a radius (also called *camber*) of between 7 and 12 inches usually. Naturally the strings will follow the radius of the fretboard and so must the top surface of the magnets, generally speaking. The second reason is that some strings have naturally higher output, the plain or non-wound G-string being the most significant and this calls for the corresponding magnet to be further compensated, resulting in an apparent odd looking stagger. Fender Strat pickups generally follow the traditional design and have the G string's magnet pole piece as tall as the D-string's, but this causes the G-string of modern string sets to be excessively loud and dominate all the other strings. This comes about because Stratocaster pickups were designed in the 1950s when string sets came with a wound G-string, but modern rock and blues players found it difficult to stretch or bend wound G-strings across the fretboard because of their inherently higher tension. In the 1970s, string manufacturers responded and introduced the now standard non-wound G-string which has lower tension and can be stretched more easily, but which produces much higher output. In order for the G-string to have the same output the corresponding magnet pole should have the greatest gap between the string and the magnet pole piece, thus different strings have magnets with differently compensated heights.

The first Stratocasters had a three-way pickup selector switch, selecting either the neck, middle or bridge pickup. Innovative guitarists found they could get an interesting sound by carefully positioning the selector switch lever between detented positions, where any two adjacent pickups would be on simultaneously. Some players wedged a plectrum between the pickguard and the selector switch to lock it in these positions. Later on, Fender introduced the now standard five-way selector switch, which uses additional detents between the original three positions to allow the combinations of any two adjacent pickups.

Modern Stratocasters have five-position pickup selector switch. Positions 1, 3 and 5 activate only one pickup (bridge, middle or neck respectively), while positions 2 and 4 activate a combination of two pickups (bridge and middle, or middle and neck, respectively). Some pickup sets have a reverse wound and reverse polarity middle pickup that when in combination with the normal bridge or neck pickups will cancel electromagnetic interference (noise/hum) which single coil pickups suffer badly from. The sonic effect of positions 2 and 4 is sometimes referred to as a "quack" or "notch positions", and some guitar notation includes directions to use these pickup combinations. One example is "Sultans of Swing" by Dire Straits which is played in position 2 (bridge and middle).

Noise problems

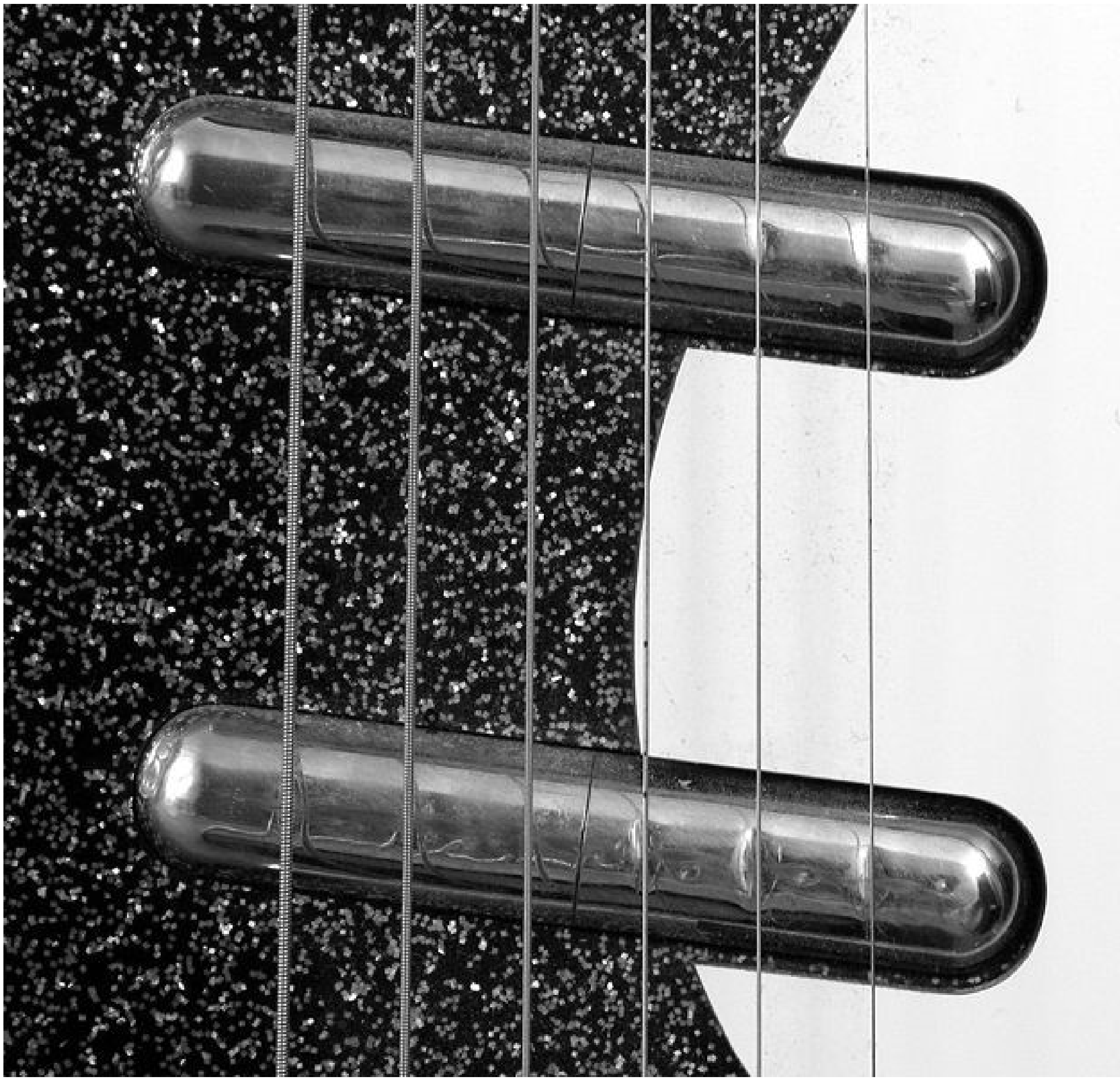
Fender Stratocaster and Telecaster pickups, being of the single-coil type, output a type of noise known as mains hum or 50Hz / 60Hz hum. Mains hum has its origin in wiring of a building and electrical apparatus/appliances such as transformers, electric motors and lighting. Hum is undesirable because it pollutes the musical notes being played on the instrument with its own sound of fixed unchanging frequency or pitch (usually 50 or 60 hertz) which is discordant with the musical sounds. To address this undesirable situation various attempts to eliminate mains hum signal from Fender single-coil pickups were made dating back to the early 1970s. DiMarzio, Seymour Duncan and EMG manufactured what are commonly known as stacked single coils which canceled mains hum. Unfortunately these stacks also canceled string signal and had a detrimental effect on sound quality. EMG used active circuitry within the pickup to compensate for the losses caused by stacked coils by boosting and reshaping the damaged sound but this required an on-board battery with its attendant problems. The resultant sound was not authentic Fender trade mark sound but EMG pickups became popular for their own sound.

Actodyne General manufactured a low-noise design of single-coil pickup known as Lace Sensors, Don Lace being the inventor. The Lace Sensor pickup had a rubberized particle magnet and used ferrous shielding to reduce hum. Being the best at the time, Fender installed Lace Sensors on the Strat Plus model for many years as a solution to the mains hum problem. However the Lace Sensor was a stopgap solution because the sound was not authentic Fender trademark sound. Fender purists wanted the genuine sound of the original Fender pickups with Alnico rod magnets and Fender eventually discontinued Lace Sensors as their mainstay solution to mains hum circa 1998. Lace Sensors continue to be used by many guitar players regardless.

The search for an acceptable solution to mains hum gained new impetus around 1995 as guitar players became increasingly intolerant of degraded stacked single-coil sound. Fender was researching new techniques to solve the loss of tone around that time and eventually came out with their Vintage Noiseless design circa 1998.

Notable single-coil pickups

There are several well-known single-coil pickups that have a distinctive sound:



"Lipstick"-style single coil pickups on a Danelectro guitar

- Rickenbacker pickups (including the original 1930s "horseshoe" pickup as used in lap steel and solidbody upright basses, and later 6 string electric guitars, pedal steels, and electric bass guitars; also the "Toaster" and "Hi-Gain")
- Gibson bar pickup (1935) — later called the Charlie Christian pickup (1938)
- Gibson P-90 (1946)
- Fender Telecaster, Stratocaster, Jazzmaster, Jaguar, and other pickups
- Danelectro Lipstick
- Gretsch pickups (including the "HiLoTron")

- DeArmond pickups (found on various 50s and 60s guitars by various manufacturers including Gretsch, Guild, Epiphone, Martin, Kustom, Harmony, Regal, Premier, and others, but produced by the Rowe - DeArmond company of Toledo, Ohio; the trade name is now owned by Fender; single coil models including the 200 aka Dynasonic, 2K, and 2000, "mustache", various "gold foil" types, and many clip on, rail, or screw mount pickups designed for acoustic guitars and other instruments). The Fender "Tele-Sonic" featured large DeArmond single coils.
- Valco single coil pickups by Ralph Keller (1954) can be found in Airline, Supro, National, English Electronics, and a few Gretsch models of guitar from the 50's, 60's, and 70's. The majority of these pickups maintain the physical appearance of a larger, humbucking pickup. Early variations on the over-strings horseshoe pickup can be found on a number of similarly branded lapsteel guitars.
- Epiphone "New York" pickups
- Lace Sensor pickups (1987)

Chapter-5

Loudspeaker

An inexpensive, low fidelity 3½-inch **speaker**, typically found in small radios.

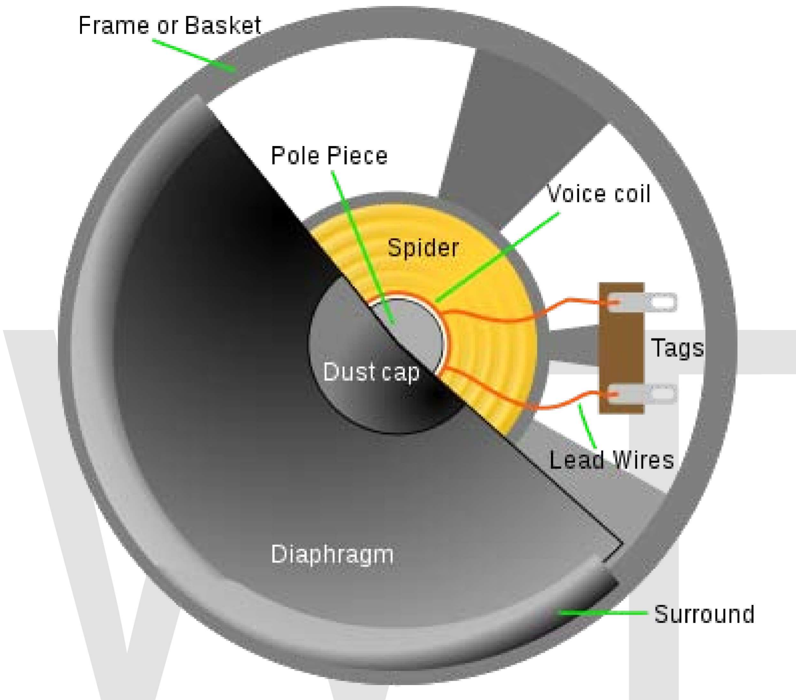
A four-way, high fidelity **loudspeaker system**.

A **loudspeaker** (or "speaker") is an electroacoustic transducer that produces sound in response to an electrical audio signal input.

Terminology

The term "loudspeaker" may refer to individual transducers (known as "drivers") or to complete speaker systems consisting of an enclosure including one or more drivers. To adequately reproduce a wide range of frequencies, most loudspeaker systems employ more than one driver, particularly for higher sound pressure level or maximum accuracy. Individual drivers are used to reproduce different frequency ranges. The drivers are named subwoofers (for very low frequencies); woofers (low frequencies); mid-range speakers (middle frequencies); tweeters (high frequencies); and sometimes supertweeters, optimized for the highest audible frequencies. The terms for different speaker drivers differ, depending on the application. In two-way systems there is no mid-range driver, so the task of reproducing the mid-range sounds falls upon the woofer and tweeter. Home stereos use the designation "tweeter" for the high frequency driver, while professional concert systems may designate them as "HF" or "highs". When multiple drivers are used in a system, a "filter network", called a crossover, separates the incoming signal into different frequency ranges and routes them to the appropriate driver. A loudspeaker system with *n* separate frequency bands is described as "*n*-way speakers": a two-way system will have a woofer and a tweeter; a three-way system employs a woofer, a mid-range, and a tweeter.

Driver design



Cutaway view of a dynamic loudspeaker.

A stamped steel loudspeaker basket frame is clearly visible (here, blue-grey).

The most common type of driver uses a lightweight diaphragm, or *cone*, connected to a rigid *basket*, or *frame*, via a flexible suspension that constrains a coil of fine wire to move axially through a cylindrical magnetic gap. When an electrical signal is applied to the voice coil, a magnetic field is created by the electric current in the voice coil, making it a variable electromagnet. The coil and the driver's magnetic system interact, generating a mechanical force that causes the coil (and thus, the attached cone) to move back and forth, thereby reproducing sound under the control of the applied electrical signal coming from the amplifier. The following is a description of the individual components of this type of loudspeaker.

The diaphragm is usually manufactured with a cone- or dome-shaped profile. A variety of different materials may be used, but the most common are paper, plastic, and metal. The ideal material would be 1) rigid, to prevent uncontrolled cone motions; 2) have low mass, to minimize starting force requirements and energy storage issues; 3) be well damped, to reduce vibrations continuing after the signal has stopped with little or no audible ringing due to its resonance frequency as determined by its usage. In practice, all three of these criteria cannot be met simultaneously using existing materials; thus, driver design involves trade-offs. For example, paper is light and typically well damped, but is not stiff; metal may be stiff and light, but it usually has poor damping; plastic can be light, but typically, the stiffer it is made, the poorer the damping. As a result, many cones are made of some sort of composite material. For example, a cone might be made of cellulose paper, into which some carbon fiber, Kevlar, glass, hemp or bamboo fibers have been added; or it might use a honeycomb sandwich construction; or a coating might be applied to it so as to provide additional stiffening or damping.

The chassis, frame, or basket, is designed to be rigid, avoiding deformation which would change critical alignments with the magnet gap, perhaps causing the voice coil to rub against the sides of the gap. Chassis are typically cast from aluminum alloy, or stamped from thin steel sheet, although molded plastic and damped plastic compound baskets are becoming common, especially for inexpensive, low-mass drivers. Metallic chassis can play an important role in conducting heat away from the voice coil; heating during operation changes resistance, causing physical dimensional changes, and if extreme, may even demagnetize permanent magnets.

The suspension system keeps the coil centered in the gap and provides a restoring (centering) force that returns the cone to a neutral position after moving. A typical suspension system consists of two parts: the "spider", which connects the diaphragm or voice coil to the frame and provides the majority of the restoring force, and the "surround", which helps center the coil/cone assembly and allows free pistonic motion aligned with the magnetic gap. The spider is usually made of a corrugated fabric disk, impregnated with a stiffening resin. The name comes from the shape of early suspensions, which were two concentric rings of Bakelite material, joined by six or eight curved "legs". Variations of this topology included the addition of a felt disc to provide a barrier to particles that might otherwise cause the voice coil to rub. The German firm Rulik still offers drivers with uncommon spiders made of wood.

The cone surround can be rubber or polyester foam, or a ring of corrugated, resin coated fabric; it is attached to both the outer diaphragm circumference and to the frame. These different surround materials, their shape and treatment can dramatically affect the acoustic output of a driver; each class and implementation having advantages and disadvantages. Polyester foam, for example, is lightweight and economical, but is degraded by exposure to ozone, UV light, humidity and elevated temperatures, limiting its useful life to about 15 years.

The wire in a voice coil is usually made of copper, though aluminum—and, rarely, silver—may be used. The advantage of aluminum is its light weight, which raises the resonant frequency of the voice coil and allows it to respond more easily to higher frequencies. A disadvantage of aluminum is that it is not easily soldered, and so connections are instead often crimped together and sealed. These connections can corrode and fail in time. Voice-coil wire cross sections can be circular, rectangular, or hexagonal, giving varying amounts of wire volume coverage in the magnetic gap space. The coil is oriented co-axially inside the gap; it moves back and forth within a small circular volume (a hole, slot, or groove) in the magnetic structure. The gap establishes a concentrated magnetic field between the two poles of a permanent magnet; the outside of the gap being one pole, and the center post (called the pole piece) being the other. The pole piece and backplate are often a single piece, called the poleplate or yoke.

Modern driver magnets are almost always permanent and made of ceramic, ferrite, Alnico, or, more recently, rare earth such as neodymium and Samarium cobalt. A trend in design—due to increases in transportation costs and a desire for smaller, lighter devices (as in many home theater multi-speaker installations)—is the use of the last instead of heavier ferrite types. Very few manufacturers still use electrically powered field coils, as was common in the earliest designs (one such is French). When high field-strength permanent magnets became available, Alnico, an alloy of aluminum, nickel, and cobalt became popular, since it dispensed with the power supply issues of field-coil drivers. Alnico was used for almost exclusively until about 1980. Alnico magnets can be partially degaussed (i.e., demagnetized) by accidental 'pops' or 'clicks' caused by loose connections, especially if used with a high power amplifier. This damage can be reversed by "recharging" the magnet.

After 1980, most (but not quite all) driver manufacturers switched from Alnico to ferrite magnets, which are made from a mix of ceramic clay and fine particles of barium or strontium ferrite. Although the energy per kilogram of these ceramic magnets is lower than Alnico, it is substantially less expensive, allowing designers to use larger yet more economical magnets to achieve a given performance.

The size and type of magnet and details of the magnetic circuit differ, depending on design goals. For instance, the shape of the pole piece affects the magnetic interaction between the voice coil and the magnetic field, and is sometimes used to modify a driver's behavior. A "shorting ring", or Faraday loop, may be included as a thin copper cap fitted over the pole tip or as a heavy ring situated within the magnet-pole cavity. The benefits of this complication is reduced impedance at high frequencies, providing extended treble output, reduced harmonic distortion, and a reduction in the inductance modulation that typically accompanies large voice coil excursions. On the other hand, the copper cap requires a wider voice-coil gap, with increased magnetic reluctance; this reduces available flux, requiring a larger magnet for equivalent performance.

Driver design—including the particular way two or more drivers are combined in an enclosure to make a speaker system—is both an art and science. Adjusting a design to improve performance is done using some combination of magnetic, acoustic, mechanical, electrical, and material science theory; high precision measurements, generally with the observations of experienced listeners. A few of the issues speaker and driver designers must confront are distortion, lobing, phase effects, off-axis response, and crossover complications. Designers can use an anechoic chamber to ensure the speaker can be measured independently of room effects, or any of several electronic techniques which can, to some extent, replace such chambers. Some developers eschew anechoic chambers in favor of specific standardized room setups intended to simulate real-life listening conditions.

The fabrication of finished loudspeaker systems has become segmented, depending largely on price, shipping costs, and weight limitations. High-end speaker systems, which are typically heavier (and often larger) than economic shipping allows outside local regions, are usually made in their target market region and can cost $140,000 or more per pair. The lowest-priced speaker systems and most drivers are manufactured in China or other low-cost manufacturing locations.

Driver types

Individual electrodynamic drivers provide optimal performance within a limited pitch range. Multiple drivers (e.g., subwoofers, woofers, mid-range drivers, and tweeters) are generally combined into a complete loudspeaker system to provide performance beyond that constraint.

Full-range drivers

A full-range driver is designed to have the widest frequency response possible. These drivers are small, typically 3 to 8 inches (7.6 to 20 cm) in diameter to permit reasonable high frequency response, and carefully designed to give low-distortion output at low frequencies, though with reduced maximum output level. Full-range (or more accurately, wide-range) drivers are most commonly heard in public address systems, in televisions (although some models are suitable for hi-fi listening), small radios, intercoms, some computer speakers, etc. In hi-fi speaker systems, the use of wide-range drive units can avoid undesirable interactions between multiple drivers caused by non-coincident driver location or crossover network issues. Fans of wide-range driver hi-fi speaker systems claim a coherence of sound, said to be due to the single source and a resulting lack of interference, and likely also to the lack of crossover components. Detractors typically cite wide-range drivers' limited frequency response and modest output abilities (most especially at low frequencies), together with their requirement for large, elaborate, expensive enclosures—such as transmission lines, or horns—to approach optimum performance.

Full-range drivers often employ an additional cone called a *whizzer*: a small, light cone attached to the joint between the voice coil and the primary cone. The whizzer cone extends the high-frequency response of the driver and broadens its high frequency directivity, which would otherwise be greatly narrowed due to the outer diameter cone material failing to keep up with the central voice coil at higher frequencies. The main cone in a whizzer design is manufactured so as to flex more in the outer diameter than in the center. The result is that the main cone delivers low frequencies and the whizzer cone contributes most of the higher frequencies. Since the whizzer cone is smaller than the main diaphragm, output dispersion at high frequencies is improved relative to an equivalent single larger diaphragm.

Limited-range drivers, also used alone, are typically found in computers, toys, and clock radios. These drivers are less elaborate and less expensive than wide-range drivers, and they may be severely compromised to fit into very small mounting locations. In these applications, sound quality is a low priority. The human ear is remarkably tolerant of poor sound quality, and the distortion inherent in limited-range drivers may enhance their output at high frequencies, increasing clarity when listening to spoken word material.

Subwoofer

A subwoofer is a woofer driver used only for the lowest part of the audio spectrum: typically below 200 Hz for consumer systems, below 100 Hz for professional live sound, and below 80 Hz in THX-approved systems. Because the intended range of frequencies is limited, subwoofer system design is usually simpler in many respects than for conventional loudspeakers, often consisting of a single driver enclosed in a suitable box or enclosure.

To accurately reproduce very low bass notes without unwanted resonances (typically from cabinet panels), subwoofer systems must be solidly constructed and properly braced; good speakers are typically quite heavy. Many subwoofer systems include power amplifiers and electronic sub-filters, with additional controls relevant to low-frequency reproduction. These variants are known as "active subwoofers". In contrast, "passive" subwoofers require external amplification.

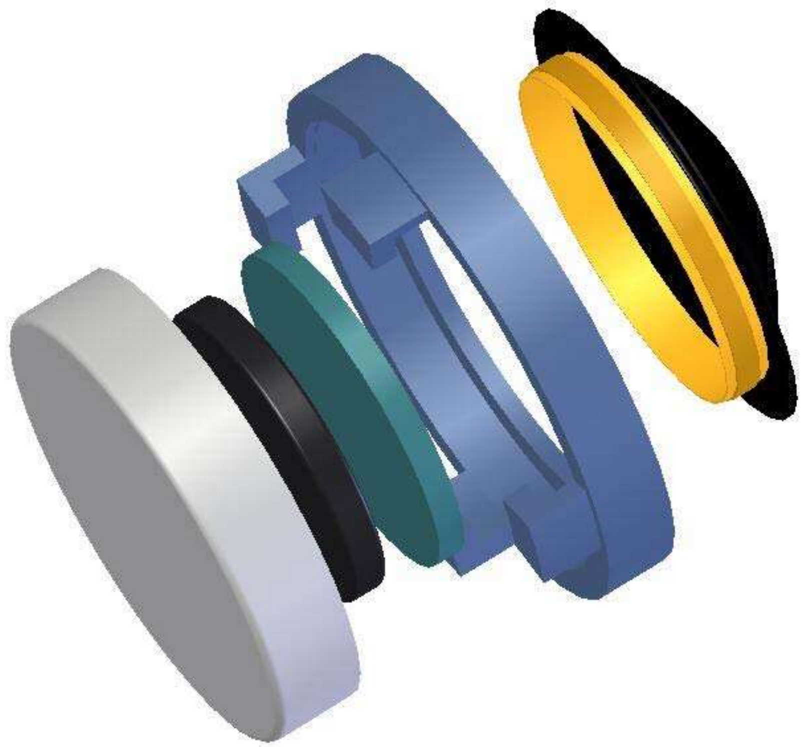
Woofer

A woofer is a driver that reproduces low frequencies. The driver combines with the enclosure design to produce suitable low frequencies. Some loudspeaker systems use a woofer for the lowest frequencies, sometimes well enough that a subwoofer is not needed. Additionally, some loudspeakers use the woofer to handle middle frequencies, eliminating the mid-range driver. This can be accomplished with the selection of a tweeter that can work low enough that, combined with a woofer that responds high enough, the two drivers add coherently in the middle frequencies.

Mid-range driver

A mid-range speaker is a loudspeaker driver that reproduces middle frequencies. Mid-range driver diaphragms can be made of paper or composite materials, and can be direct radiation drivers (rather like smaller woofers) or they can be compression drivers (rather like some tweeter designs). If the mid-range driver is a direct radiator, it can be mounted on the front baffle of a loudspeaker enclosure, or, if a compression driver, mounted at the throat of a horn for added output level and control of radiation pattern.

Tweeter



Exploded view of a dome tweeter.

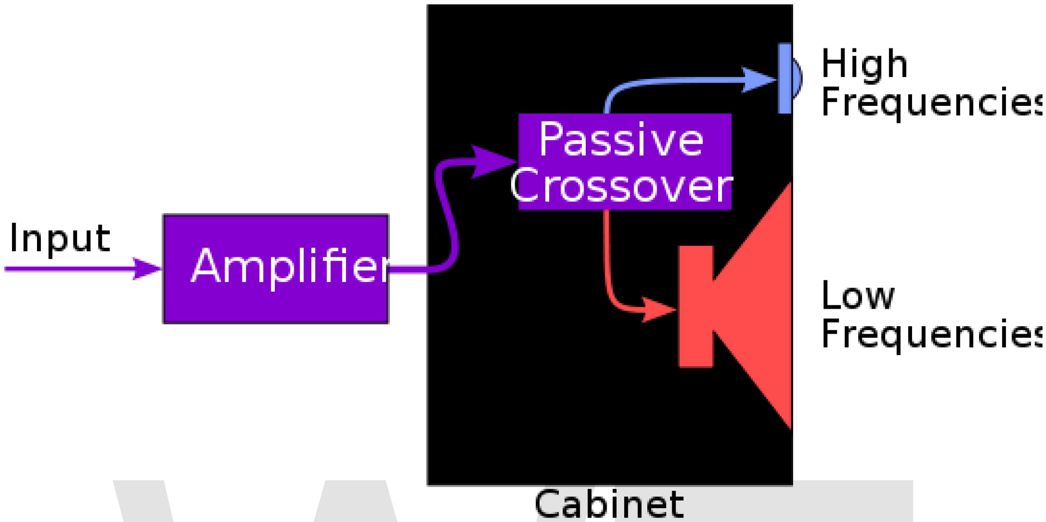
A tweeter is a high-frequency driver that reproduces the highest frequencies in a speaker system. Many varieties of tweeter design exist, each with differing abilities with regard to frequency response, output fidelity, power handling, maximum output level, etc. Soft-dome tweeters are widely found in home stereo systems, and horn-loaded compression drivers are common in professional sound reinforcement. Ribbon tweeters have gained popularity in recent years, as their output power has been increased to levels useful for professional sound reinforcement, and their output pattern is wide in the horizontal plane, a pattern that has convenient applications in concert sound.

Coaxial drivers

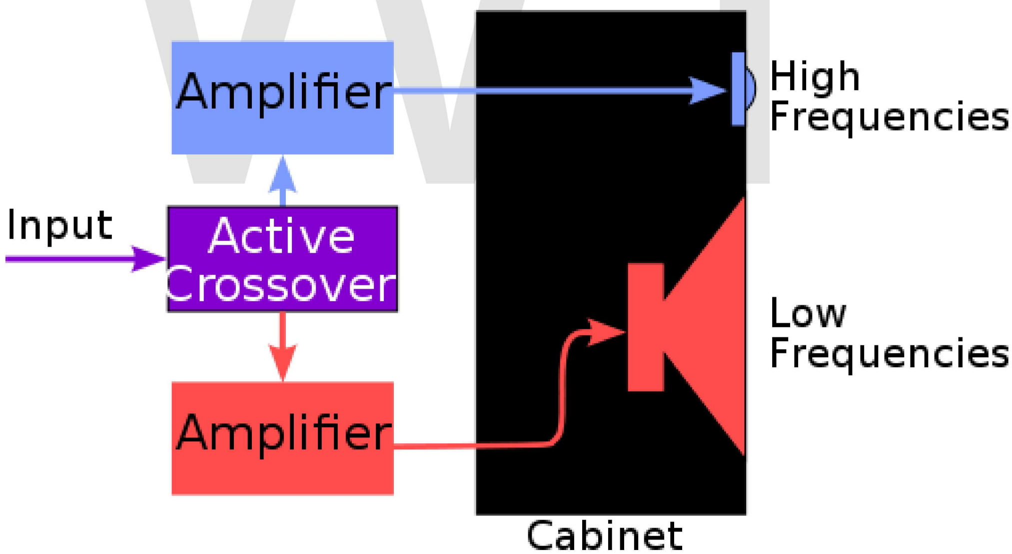
A coaxial driver is a loudspeaker driver with two or several combined concentric drivers. Coaxial drivers have been produced by many companies, such as Altec, Tannoy, Pioneer, KEF, BMS, Cabasse and Genelec.

Loudspeaker system design

Crossover



A passive crossover.



Bi-amped.

Used in multi-driver speaker systems, the crossover is a subsystem that separates the input signal into different frequency ranges suited to each driver. The drivers receive only the power in their usable frequency range (the range they were designed for), thereby reducing distortion in the drivers and interference between them.

Crossovers can be *passive* or *active*. A passive crossover is an electronic circuit that uses a combination of one or more resistors, inductors, or non-polar capacitors. These parts are formed into carefully designed networks and are most often placed between the power amplifier and the loudspeaker drivers to divide the amplifier's signal into the necessary frequency bands before being delivered to the individual drivers. Passive crossover circuits need no external power beyond the audio signal itself, but do cause overall signal loss and a significant reduction in damping factor between the voice coil and the crossover. An active crossover is an electronic filter circuit that divides the signal into individual frequency bands *before* power amplification, thus requiring at least one power amplifier for each bandpass. Passive filtering may also be used in this way before power amplification, but it is an uncommon solution, due to inflexibility compared to active filtering. Any technique that uses crossover filtering followed by amplification is commonly known as bi-amping, tri-amping, quad-amping, and so on, depending on the minimum number of amplifier channels. Some loudspeaker designs use a combination of passive and active crossover filtering, such as a passive crossover between the mid- and high-frequency drivers and an active crossover between the low-frequency driver and the combined mid- and high frequencies.

Passive crossovers are commonly installed inside speaker boxes and are by far the most usual type of crossover for home and low-power use. In car audio systems, passive crossovers may be in a separate box, necessary to accommodate the size of the components used. Passive crossovers may be simple for low-order filtering, or complex to allow steep slopes such as 18 or 24 dB per octave. Passive crossovers can also be designed to compensate for undesired characteristics of driver, horn, or enclosure resonances, and can be tricky to implement, due to component interaction. Passive crossovers, like the driver units that they feed, have power handling limits, have insertion losses (10% is often claimed), and change the load seen by the amplifier. The changes are matters of concern for many in the hi-fi world. When high output levels are required, active crossovers may be preferable. Active crossovers may be simple circuits that emulate the response of a passive network, or may be more complex, allowing extensive audio adjustments. Some active crossovers, usually digital loudspeaker management systems, may include facilities for precise alignment of phase and time between frequency bands, equalization, and dynamics (compression and limiting) control.

Some hi-fi and professional loudspeaker systems now include an active crossover circuit as part of an onboard amplifier system. These speaker designs are identifiable by their need for AC power in addition to a signal cable from a pre-amplifier. This active topology may include driver protection circuits and other features of a digital loudspeaker management system. Powered speaker systems are common in computer sound (for a single listener) and, at the other end of the size spectrum, in modern concert sound systems, where their presence is significant and steadily increasing.

Enclosures

An unusual three-way speaker system. The cabinet is narrow in order to raise the frequency at which a diffraction effect called the "baffle step" occurs.

Most loudspeaker systems consist of drivers mounted in an enclosure, or cabinet. The role of the enclosure is to provide a place to physically mount the drivers, and to prevent sound waves emanating from the back of a driver from interfering destructively with those from the front; these typically cause cancellations (e.g., comb filtering) and significantly alter the level and quality of sound at low frequencies.

The simplest driver mount is a flat panel (i.e., baffle) with the drivers mounted in holes in it. However, in this approach, sound frequencies with a wavelength longer than the baffle dimensions are canceled out, because the antiphase radiation from the rear of the cone interferes with the radiation from the front. With an infinitely large panel, this interference could be entirely prevented. A sufficiently large sealed box can approach this behavior.

Since panels of infinite dimensions are impractical, most enclosures function by containing the rear radiation from the moving diaphragm. A sealed enclosure prevents transmission of the sound emitted from the rear of the loudspeaker by confining the sound in a rigid and airtight box. Techniques used to reduce transmission of sound through the walls of the cabinet include thicker cabinet walls, lossy wall material, internal bracing, curved cabinet walls—or more rarely, visco-elastic materials (e.g., mineral-loaded bitumen) or thin lead sheeting applied to the interior enclosure walls.

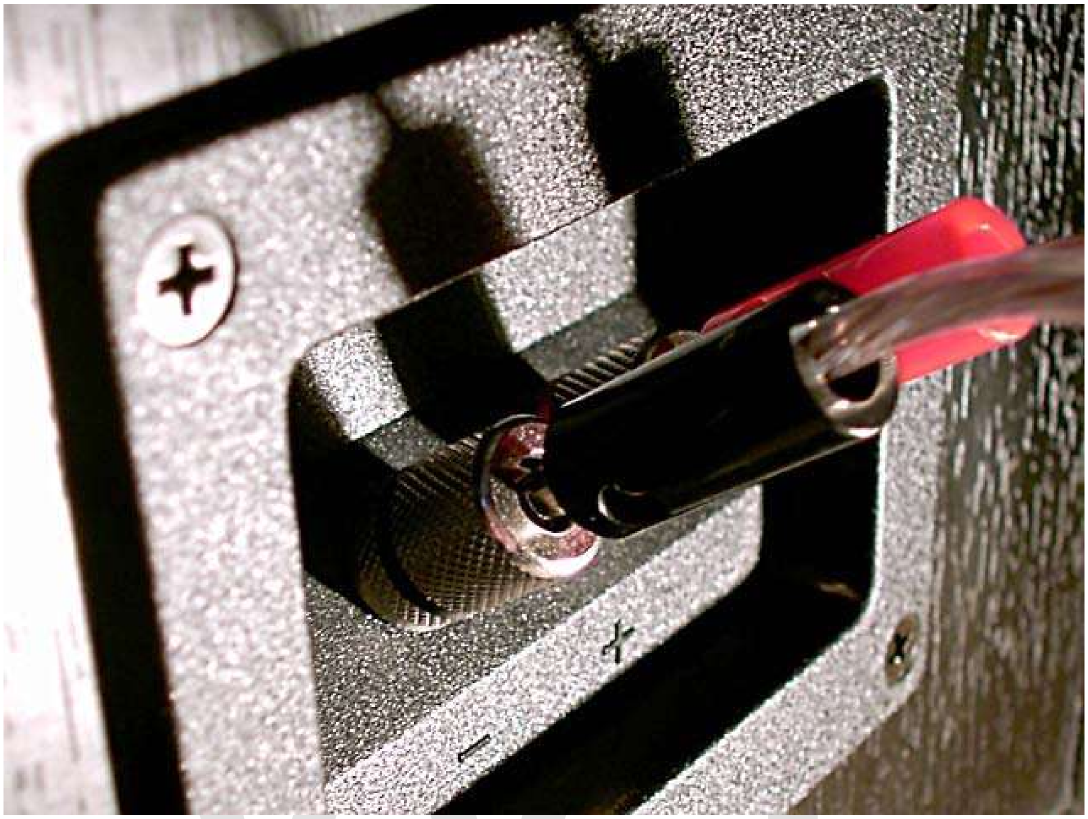
However, a rigid enclosure reflects sound internally, which can then be transmitted back through the loudspeaker diaphraghm—again resulting in degradation of sound quality. This can be reduced by internal absorption using absorptive materials (often called "damping"), such as glass wool, wool, or synthetic fiber batting, within the enclosure. The internal shape of the enclosure can also be designed to reduce this by reflecting sounds away from the loudspeaker diaphragm, where they may then be absorbed.

Other enclosure types alter the rear sound radiation so it can add constructively to the output from the front of the cone. Designs that do this (including *bass reflex*, *passive radiator*, *transmission line*, etc.) are often used to extend the effective low-frequency response and increase low-frequency output of the driver.

To make the transition between drivers as seamless as possible, system designers have attempted to time-align (or phase adjust) the drivers by moving one or more driver mounting locations forward or back so that the acoustic center of each driver is in the same vertical plane. This may also involve tilting the face speaker back, providing a separate enclosure mounting for each driver, or (less commonly) using electronic techniques to achieve the same effect. These attempts have resulted in some unusual cabinet designs.

The speaker mounting scheme (including cabinets) can also cause diffraction, resulting in peaks and dips in the frequency response. The problem is usually greatest at higher frequencies, where wavelengths are similar to, or smaller than, cabinet dimensions. The effect can be minimized by rounding the front edges of the cabinet, curving the cabinet itself, using a smaller or narrower enclosure, choosing a strategic driver arrangement, using absorptive material around a driver, or some combination of these and other schemes.

Wiring connections



Two-way binding posts on a loudspeaker, connected using banana plugs.

A 4-ohm loudspeaker with two pairs of binding posts capable of accepting bi-wiring after the removal of two metal straps.

Most loudspeakers use two wiring points to connect to the source of the signal (for example, to the audio amplifier or receiver). This is usually done using binding posts or spring clips on the back of the enclosure. If the wires for the left and right speakers (in a stereo setup) are not connected "in phase" with each other (the + and − connections on the speaker and amplifier should be connected + to + and − to −), the loudspeakers will be out of polarity. Given identical signals, motion in one cone will be in the opposite direction of the other. This will typically cause monophonic material within a stereo recording to be canceled out, reduced in level, and made more difficult to localize, all due to destructive interference of the sound waves. The cancellation effect is most noticeable at frequencies where the speakers are separated by a quarter wavelength or less; low frequencies are affected the most. This type of wiring error doesn't damage speakers, but isn't optimal.

Specifications

Specifications label on a loudspeaker.

Speaker specifications generally include:

- **Speaker or driver type** (individual units only) – Full-range, woofer, tweeter, or mid-range.
- **Size** of individual drivers. For cone drivers, the quoted size is generally the outside diameter of the basket. However, it may less commonly also be the diameter of the cone surround, measured apex to apex, or the distance from the center of one mounting hole to its opposite. Voice-coil diameter may also be specified. If the loudspeaker has a compression horn driver, the diameter of the horn throat may be given.
- **Rated Power** – Nominal (or even continuous) power, and peak (or maximum short-term) power a loudspeaker can handle (i.e., maximum input power before destroying the loudspeaker; it is never the sound output the loudspeaker produces). A driver may be damaged at much less than its rated power if driven past its mechanical limits at lower frequencies. Tweeters can also be damaged by amplifier clipping (amplifier circuits produce large amounts of energy at high frequencies in such cases) or by music or sine wave input at high frequencies. Each of these situations passes more energy to a tweeter than it can survive without damage. In some jurisdictions, power handling has a legal meaning allowing comparisons between loudspeakers under consideration. Elsewhere, the variety of meanings for power handling capacity can be quite confusing.

- **Impedance** – typically 4 Ω (ohms), 8 Ω, etc.
- **Baffle or enclosure type** (enclosed systems only) – Sealed, bass reflex, etc.
- **Number of drivers** (complete speaker systems only) – two-way, three-way, etc.

and optionally:

- **Crossover frequency(ies)** (multi-driver systems only) – The nominal frequency boundaries of the division between drivers.
- **Frequency response** – The measured, or specified, output over a specified range of frequencies for a constant input level varied across those frequencies. It sometimes includes a variance limit, such as within "± 2.5 dB".
- **Thiele/Small parameters** (individual drivers only) – these include the driver's F_s (resonance frequency), Q_{ts} (a driver's Q; more or less, its damping factor at resonant frequency), V_{as} (the equivalent air compliance volume of the driver), etc.
- **Sensitivity** – The sound pressure level produced by a loudspeaker in a non-reverberant environment, often specified in dB and measured at 1 meter with an input of 1 watt (2.83 rms volts into 8 Ω), typically at one or more specified frequencies. This rating is often specified by manufacturers to be impressive.
- **Maximum SPL** – The highest output the loudspeaker can manage, short of damage or not exceeding a particular distortion level. This rating is often specified by manufacturers to be impressive, and is commonly given without reference to frequency range or distortion level.

Electrical characteristics of a dynamic loudspeaker

The load that a driver presents to an amplifier consists of a complex electrical impedance—a combination of resistance and both capacitive and inductive reactance, which combines properties of the driver, its mechanical motion, the effects of crossover components (if any are in the signal path between amplifier and driver), and the effects of air loading on the driver as modified by the enclosure and its environment. Most amplifiers' output specifications are given at a specific power into an ideal resistive load; however, a loudspeaker does not have a constant resistance across its frequency range. Instead, the voice coil is inductive, the driver has mechanical resonances, the enclosure changes the driver's electrical and mechanical characteristics, and a passive crossover between the drivers and the amplifier contributes its own variations. The result is a load resistance that varies fairly widely with frequency, and usually a varying phase relationship between voltage and current as well, also changing with frequency. Some amplifiers can cope with the variation better than others can.

To make sound, a loudspeaker is driven by modulated electrical current (produced by an amplifier) that pass through a "speaker coil" (a coil of copper wire), which then (through resistance and other forces) magnetizes the coil, creating a magnetic field. The electrical current variations that pass through the speaker are thus converted to varying magnetic forces, which move the speaker diaphragm, which thus forces the driver to produce air motion that is similar to the original signal from the amplifier.

Electromechanical measurements

Fully characterizing the sound output quality of a loudspeaker driver or system in words is essentially impossible. Objective measurements provide information about several aspects of performance so that informed comparisons and improvements can be made, but no combination of measurements summarizes the performance of a loudspeaker system in use, if only because the test signals used are neither music nor speech. Examples of typical measurements are: amplitude and phase characteristics vs. frequency; impulse response under one or more conditions (e.g., square waves, sine wave bursts, etc.); directivity vs. frequency (e.g., horizontally, vertically, spherically, etc.); harmonic and intermodulation distortion vs. SPL output, using any of several test signals; stored energy (i.e., ringing) at various frequencies; impedance vs. frequency; and small-signal vs. large-signal performance. Most of these measurements require sophisticated and often expensive equipment to perform, and also good judgment by the operator, but the raw sound pressure level output is rather easier to report and so is often the only specified value—sometimes in misleadingly exact terms. The sound pressure level (SPL) a loudspeaker produces is measured in decibels (dB_{spl}).

Efficiency vs. sensitivity

Loudspeaker efficiency is defined as the sound power output divided by the electrical power input. Most loudspeakers are actually very inefficient transducers; only about 1% of the electrical energy sent by an amplifier to a typical home loudspeaker is converted to acoustic energy. The remainder is converted to heat, mostly in the voice coil and magnet assembly. The main reason for this is the difficulty of achieving proper impedance matching between the acoustic impedance of the drive unit and that of the air into which it is radiating (at low frequencies improving this match is the main purpose of speaker enclosure designs). The efficiency of loudspeaker drivers varies with frequency as well. For instance, the output of a woofer driver decreases as the input frequency decreases because of the increasingly poor match between air and the driver.

Driver ratings based on the SPL for a given input are called sensitivity ratings and are notionally similar to efficiency. Sensitivity is usually defined as so many decibels at 1 W electrical input, measured at 1 meter (except for headphones), often at a single frequency. The voltage used is often 2.83 V_{RMS}, which is 1 watt into an 8 Ω (nominal) speaker impedance (approximately true for many speaker systems). Measurements taken with this reference are quoted as dB with 2.83 V @ 1 m.

The sound pressure output is measured at (or mathematically scaled to be equivalent to a measurement taken at) one meter from the loudspeaker and on-axis (directly in front of it), under the condition that the loudspeaker is radiating into an infinitely large space and mounted on an infinite baffle. Clearly then, sensitivity does not correlate precisely with efficiency, as it also depends on the directivity of the driver being tested and the acoustic environment in front of the actual loudspeaker. For example, a cheerleader's horn produces more sound output in the direction it is pointed by concentrating sound waves from the cheerleader in one direction, thus "focusing" them. The horn also improves impedance matching between the voice and the air, which produces more acoustic power for a given speaker power. In some cases, improved impedance matching (via careful enclosure design) will allow the speaker to produce more acoustic power.

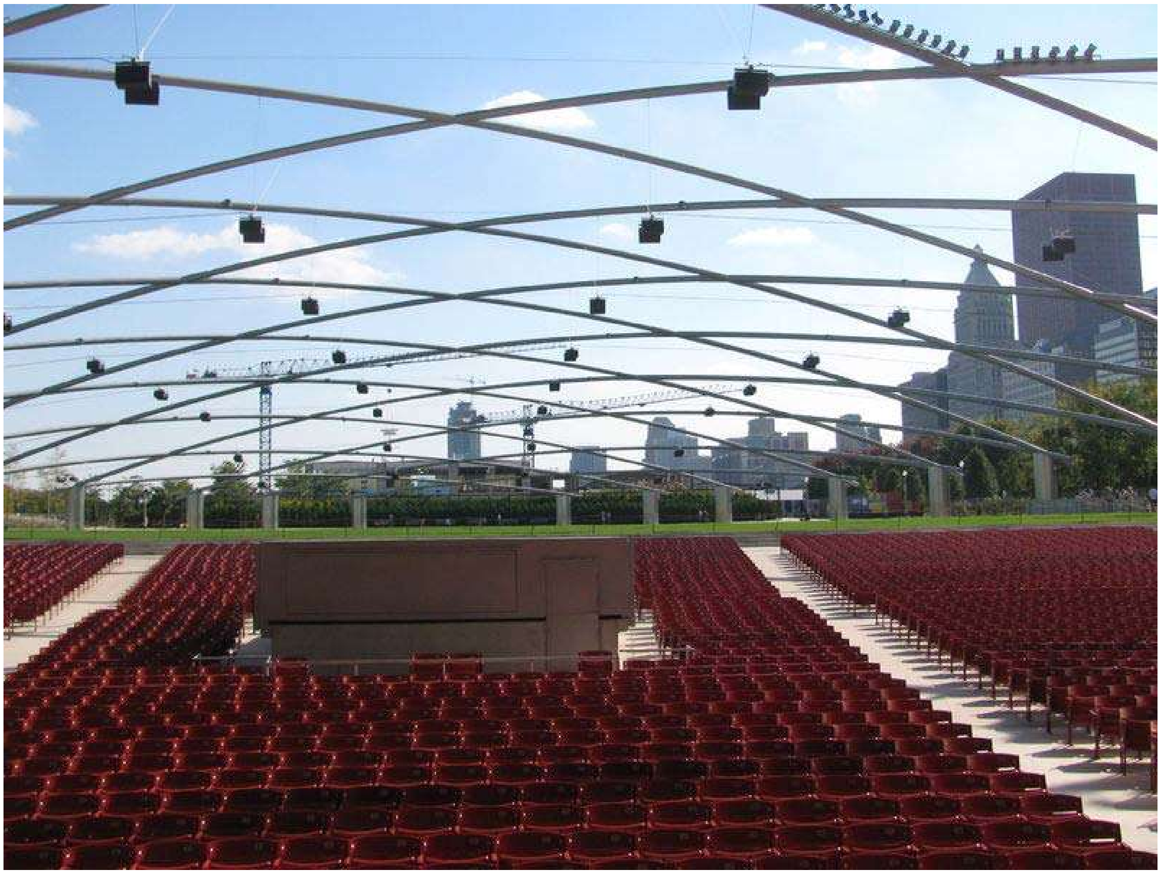
- Typical home loudspeakers have sensitivities of about 85 to 95 dB for 1 W @ 1 m—an efficiency of 0.5–4%.
- Sound reinforcement and public address loudspeakers have sensitivities of perhaps 95 to 102 dB for 1 W @ 1 m—an efficiency of 4–10%.
- Rock concert, stadium PA, marine hailing, etc. speakers generally have higher sensitivities of 103 to 110 dB for 1 W @ 1 m—an efficiency of 10–20%.

A driver with a higher maximum power rating cannot necessarily be driven to louder levels than a lower-rated one, since sensitivity and power handling are largely independent properties. In the examples that follow, assume (for simplicity) that the drivers being compared have the same electrical impedance; are operated at the same frequency within both driver's respective pass bands; and that power compression and distortion are low. For the first example, a speaker 3 dB more sensitive than another will produce double the sound power (or be 3 dB louder) for the same power input; thus, a 100 W driver ("A") rated at 92 dB for 1 W @ 1 m sensitivity will put out twice as much acoustic power as a 200 W driver ("B") rated at 89 dB for 1 W @ 1 m when both are driven with 100 W of input power. In this particular example, when driven at 100 W, speaker A will produce the same SPL, or loudness, as speaker B would produce with 200 W input. Thus, a 3 dB increase in sensitivity of the speaker means that it will need half the amplifier power to achieve a given SPL. This translates into a smaller, less complex power amplifier—and often, to reduced overall system cost.

It is not possible to combine high efficiency (especially at low frequencies) with compact enclosure size and adequate low frequency response. One can, more or less, choose only two of the three parameters when designing a speaker system. So, for example, if extended low-frequency performance and small box size are important, one must accept low efficiency. This rule of thumb is sometimes called Hoffman's Iron Law (after J.A. Hoffman, the "H" in KLH).

Listening environment

Jay Pritzker Pavilion



At Jay Pritzker Pavilion, a LARES system is combined with a zoned sound reinforcement system, both suspended on an overhead steel trellis, to synthesize an indoor acoustic environment outdoors.

The interaction of a loudspeaker system with its environment is complex and is largely out of the loudspeaker designer's control. Most listening rooms present a more or less reflective environment, depending on size, shape, volume, and furnishings. This means the sound reaching a listener's ears consists not only of sound directly from the speaker system, but also the same sound delayed by traveling to and from (and being modified by) one or more surfaces. These reflected sound waves, when added to the direct sound, cause cancellation and addition at assorted frequencies (e.g., from resonant room modes), thus changing the timbre and character of the sound at the listener's ears. The human brain is very sensitive to small variations, including some of these, and this is part of the reason why a loudspeaker system sounds different at different listening positions or in different rooms.

A significant factor in the sound of a loudspeaker system is the amount of absorption and diffusion present in the environment. Clapping one's hands in a typical empty room, without draperies or carpet, will produce a zippy, fluttery echo which is due both to a lack of absorption and to reverberation (that is, repeated echoes) from flat reflective walls, floor, and ceiling. The addition of hard surfaced furniture, wall hangings, shelving and even baroque plaster ceiling decoration, will change the echoes, due primarily to the diffusion caused by somewhat reflective objects with shapes and surfaces having sizes on the order of the sound wavelengths being diffused. This somewhat breaks up the simple reflections otherwise caused by bare flat surfaces, and spreads the reflected energy of an incident wave over a larger angle on reflection.

Placement

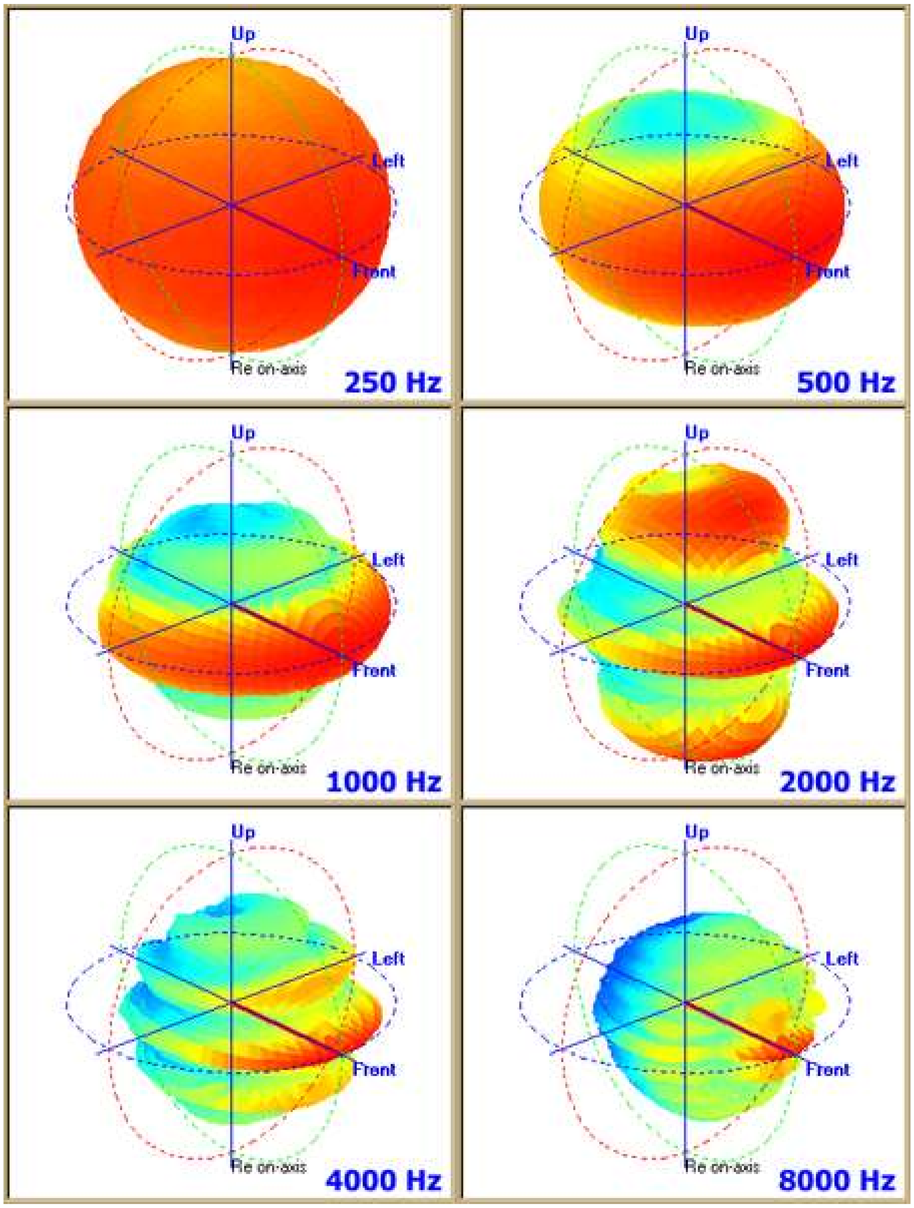
In a typical rectangular listening room, the hard, parallel surfaces of the walls, floor and ceiling cause primary acoustic resonance nodes in each of the three dimensions: left-right, up-down and forward-backward. Furthermore, there are more complex resonance modes involving three, four, five and even all six boundary surfaces combining to create standing waves. Low frequencies excite these modes the most, since long wavelengths are not much affected by furniture compositions or placement. The mode spacing is critical, especially in small and medium size rooms like recording studios, home theaters and broadcast studios. The proximity of the loudspeakers to room boundaries affects how strongly the resonances are excited as well as affecting the relative strength at each frequency. The location of the listener is critical, too, as a position near a boundary can have a great effect on the perceived balance of frequencies. This is because standing wave patterns are most easily heard in these locations and at lower frequencies, below the Schroeder frequency – typically around 200–300 Hz, depending on room size.

Directivity

Acousticians, in studying the radiation of sound sources have developed some concepts important to understanding how loudspeakers are perceived. The simplest possible radiating source is a point source, sometimes called a simple source. An ideal point source is an infinitesimally small point radiating sound. It may be easier to imagine a tiny pulsating sphere, uniformly increasing and decreasing in diameter, sending out sound waves in all directions equally, independent of frequency.

Any object radiating sound, including a loudspeaker system, can be thought of as being composed of combinations of such simple point sources. The radiation pattern of a combination of point sources will not be the same as for a single source, but rather will depend on the distance and orientation between the sources, the position relative to them from which the listener hears the combination, and the frequency of the sound involved. Using geometry and calculus, some simple combinations of sources are easily solved; others are not.

One simple combination is two simple sources separated by a distance and vibrating out of phase, one miniature sphere expanding while the other is contracting. The pair is known as a doublet, or dipole, and the radiation of this combination is similar to that of a very small dynamic loudspeaker operating without a baffle. The directivity of a dipole is a figure 8 shape with maximum output along a vector which connects the two sources and minimums to the sides when the observing point is equidistant from the two sources, where the sum of the positive and negative waves cancel each other. While most drivers are dipoles, depending on the enclosure to which they are attached, they may radiate as monopoles, dipoles (or bipoles). If mounted on a finite baffle, and these out of phase waves allowed to interact, dipole peaks and nulls in the frequency response result. When the rear radiation is absorbed or trapped in a box, the diaphragm becomes a monopole radiator. Bipolar speakers, made by mounting in-phase monopoles (both moving out of or into the box in unison) on opposite sides of a box, are a method of approaching omnidirectional radiation patterns.



Polar plots of a four-driver industrial columnar public address loudspeaker taken at six frequencies. Note how the pattern is nearly omnidirectional at low frequencies, converging to a wide fan-shaped pattern at 1 kHz, then separating into lobes and getting weaker at higher frequencies

In real life, individual drivers are actually complex 3D shapes such as cones and domes, and they are placed on a baffle for various reasons. A mathematical expression for the directivity of a complex shape, based on modeling combinations of point sources, is usually not possible, but in the farfield, the directivity of a loudspeaker with a circular diaphragm will be close to that of a flat circular piston, so it can be used as an illustrative simplification for discussion. As a simple example of the mathematical physics involved, consider the following: the formula for farfield directivity of a flat circular piston in an infinite baffle is $p(\theta) = \frac{p_0 J_1(k_a \sin\theta)}{k_a \sin\theta}$ where $k_a = \frac{2\pi a}{\lambda}$, p_0 is the pressure on axis, a is the piston radius, λ is the wavelength (i.e. $\lambda = \frac{c}{f} = \frac{\text{speed of sound}}{\text{frequency}}$) θ is the angle off axis and J_1 is the Bessel function of the first kind.

A planar source will radiate sound uniformly for low frequencies whose wavelength is longer than the dimensions of the planar source, and as frequency increases, the sound from such a source will be focused into an increasingly narrower angle. The smaller the driver, the higher the frequency where this narrowing of directivity occurs. Even if the diaphragm is not perfectly circular, this effect occurs such that larger sources are more directive. Several loudspeaker designs have been built which have approximately this behavior. Most are electrostatic or planar magnetic designs.

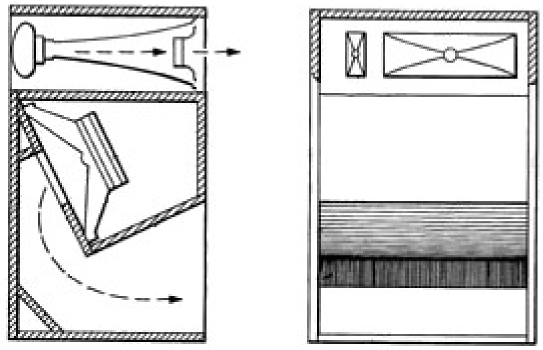
Various manufacturers use different driver mounting arrangements to create a specific type of sound field in the space for which they are designed. The resulting radiation patterns may be intended to more closely simulate the way sound is produced by real instruments, or simply create a controlled energy distribution from the input signal (some using this approach are called monitors, as they are useful in checking the signal just recorded in a studio). An example of the first is a room corner system with many small drivers on the surface of a 1/8 sphere. A system design of this type was patented by, and actually produced commercially, by Professor Amar Bose—the 2201. Later Bose models have deliberately emphasized production of both direct and reflected sound by the loudspeaker itself, regardless of its environment. The designs are controversial in high fidelity circles, but have proven commercially successful. Several other manufacturers' designs follow similar principles.

Directivity is an important issue because it affects the frequency balance of sound a listener hears, and also the interaction of the speaker system with the room and its contents. A speaker which is very directive (i.e., on an axis perpendicular to the speaker face) may result in a reverberant field lacking in high frequencies, giving the impression the speaker is deficient in treble even though it measures well on axis (e.g., "flat" across the entire frequency range). Speakers with very wide, or rapidly increasing directivity at high frequencies, can give the impression that there is too much treble (if the listener is on axis) or too little (if the listener is off axis). This is part of the reason why on-axis frequency response measurement is not a complete characterization of the sound of a given loudspeaker.

Other driver designs

Other types of drivers which depart from the most commonly used direct radiating electro-dynamic driver mounted in an enclosure include:

Horn loudspeakers



A three-way loudspeaker that uses horns in front of each of the three drivers: a shallow horn for the tweeter, a long, straight horn for mid frequencies and a folded horn for the woofer

Horn loudspeakers are the oldest form of loudspeaker system. The use of horns as voice-amplifying megaphones dates at least to the 17th century, and horns were used in mechanical gramophones as early as 1857. Horn loudspeakers use a shaped waveguide in front of or behind the driver to increase the directivity of the loudspeaker and to transform a small diameter, high pressure condition at the driver cone surface to a large diameter, low pressure condition at the mouth of the horn. This increases the sensitivity of the loudspeaker and focuses the sound over a narrower area. The size of the throat, mouth, the length of the horn, as well as the area expansion rate along it must be carefully chosen to match the drive to properly provide this transforming function over a range of frequencies (every horn performs poorly outside its acoustic limits, at both high and low frequencies). The length and cross-sectional mouth area required to create a bass or sub-bass horn require a horn many feet long. 'Folded' horns can reduce the total size, but compel designers to make compromises and accept increased complication such as cost and construction. Some horn designs not only fold the low frequency horn, but use the walls in a room corner as an extension of the horn mouth. In the late 1940s, horns whose mouths took up much of a room wall were not unknown amongst hi-fi fans. Room sized installations became much less acceptable when two or more were required.

A horn loaded speaker can have a sensitivity as high as 110 dB at 2.83 volts (1 watt at 8 ohms) at 1 meter. This is a hundredfold increase in output compared to a speaker rated at 90 dB sensitivity, and is invaluable in applications where high sound levels are required or amplifier power is limited.

Piezoelectric speakers

Piezoelectric speakers are frequently used as beepers in watches and other electronic devices, and are sometimes used as tweeters in less-expensive speaker systems, such as computer speakers and portable radios. Piezoelectric speakers have several advantages over conventional loudspeakers: they are resistant to overloads which would normally destroy most high frequency drivers, and they can be used without a crossover due to their electrical properties. There are also disadvantages: some amplifiers can oscillate when driving capacitive loads like most piezoelectrics, which results in distortion or damage to the amplifier. Additionally, their frequency response, in most cases, is inferior to that of other technologies. This is why they are generally used in single frequency (beeper) or non-critical applications.

Piezoelectric speakers can have extended high frequency output, and this is useful in some specialized circumstances; for instance, sonar applications in which piezoelectric variants are used as both output devices (generating underwater sound) and as input devices (acting as the sensing components of underwater microphones). They have advantages in these applications, not the least of which is simple and solid state construction which resists the effects of seawater better than, say, a ribbon based device would.

Magnetostrictive speakers

Magnetostrictive transducers, based on magnetostriction, have been predominantly used as sonar ultrasonic sound wave radiators, but their usage has spread also to audio speaker systems. Magnetostrictive speaker drivers have some special advantages: they can provide greater force (with smaller excursions) than other technologies; low excursion can avoid distortions from large excursion as in other designs; the magnetizing coil is stationary and therefore more easily cooled; they are robust because delicate suspensions and voice coils are not required. Magnetostrictive speaker modules have been produced by Fostex and FeONIC and subwoofer drivers have also been produced.

Electrostatic loudspeakers

Electrostatic loudspeakers use a high voltage electric field (rather than a magnetic field) to drive a thin statically charged membrane. Because they are driven over the entire membrane surface rather than from a small voice coil, they ordinarily provide a more linear and lower distortion motion than dynamic drivers. They have the disadvantage that the diaphragm excursion is severely limited because of practical construction limitations; the further apart the stators are positioned, the higher the voltage must be to achieve acceptable efficiency, which increases the tendency for electrical arcs as well as the increasing the speaker's attraction of dust particles. For many years electrostatic loudspeakers had a reputation as a generally unreliable and occasionally dangerous product. Arcing remains a potential problem with current technologies, especially when the panels are allowed to collect dust or dirt, or when driven with high signal levels.

Electrostatics are inherently dipole radiators and due to the thin flexible membrane are less suited for use in enclosures to reduce low frequency cancellation as with common cone drivers. Due to this and the low excursion capability, full range electrostatic loudspeakers are large by nature, and the bass will roll off at a frequency corresponding to a quarter wavelength of the narrowest panel dimension. To reduce the size of commercial products, they are sometimes used as a high frequency driver in combination with a conventional dynamic driver which handles the bass frequencies.

Ribbon and planar magnetic loudspeakers

A **ribbon speaker** consists of a thin metal-film ribbon suspended in a magnetic field. The electrical signal is applied to the ribbon which moves with it, thus creating the sound. The advantage of a ribbon driver is that the ribbon has very little mass; thus, it can accelerate very quickly, yielding very good high-frequency response. Ribbon loudspeakers are often very fragile—some can be torn by a strong gust of air. Most ribbon tweeters emit sound in a dipole pattern; a very few have backings which limit the dipole radiation pattern. Above and below the ends of the more or less rectangular ribbon, there is less audible output due to phase cancellation, but the precise amount of directivity depends on ribbon length. Ribbon designs generally require exceptionally powerful magnets which make them costly to manufacture. Ribbons have a very low resistance that most amplifiers cannot drive directly. As a result, a step down transformer is typically used to increase the current through the ribbon. The amplifier "sees" a load that is the ribbon's resistance times the transformer turns ratio squared. The transformer must be carefully designed so that its frequency response and parasitic losses do not degrade the sound, further increasing cost and complication relative to conventional designs.

Planar magnetic speakers (having printed or embedded conductors on a flat diaphragm) are sometimes described as "ribbons", but are not truly ribbon speakers. The term planar is generally reserved for speakers which have roughly rectangular shaped flat surfaces that radiate in a bipolar (i.e., front and back) manner. Planar magnetic speakers consist of a flexible membrane with a voice coil printed or mounted on it. The current flowing through the coil interacts with the magnetic field of carefully placed magnets on either side of the diaphragm, causing the membrane to vibrate more or less uniformly and without much bending or wrinkling. The driving force covers a large percentage of the membrane surface and reduces resonance problems inherent in coil-driven flat diaphragms.

Bending wave loudspeakers

Bending wave transducers use a diaphragm that is intentionally flexible. The rigidity of the material increases from the center to the outside. Short wavelengths radiate primarily from the inner area, while longer waves reach the edge of the speaker. To prevent reflections from the outside back into the center, long waves are absorbed by a surrounding damper. Such transducers can cover a wide frequency range (80 Hz to 35,000 Hz) and have been promoted as being close to an ideal point sound source. This uncommon approach is being taken by only a very few manufacturers, in very different arrangements. The line of Ohm Walsh speakers use a unique driver designed by Lincoln Walsh. Lincoln Walsh was a brilliant engineer who was part of the engineering team that developed radar during World War II. He later designed audio amplifiers, and his final project was a unique, one-way speaker with one driver. It was a large cone that faced down into a sealed, airtight enclosure. Rather than move back-and-forth as conventional speakers do, the cone rippled and created sound using a principle known as “transmission line”. The new speaker created a single, perfectly rendered sound wave of remarkable clarity. A new company, Ohm Acoustics, was formed to develop and market Walsh’s new speaker design. Lincoln Walsh died before his speaker was released to the public. After developing the Ohm A prototype, in 1973 Ohm introduced the Ohm F speaker to critical acclaim.

Flat panel loudspeakers

There have been many attempts to reduce the size of speaker systems, or alternatively to make them less obvious. One such attempt was the development of voice coils mounted to flat panels to act as sound sources. These can then be made in a neutral color and hung on walls where they will be less noticeable than many speakers, or can be deliberately painted with patterns in which case they can function decoratively. There are two related problems with flat panel techniques: first, a flat panel is necessarily more flexible than a cone shape in the same material, and therefore will move as a single unit even less, and second, resonances in the panel are difficult to control, leading to considerable distortions. Some progress has been made using such lightweight, rigid, materials such as Styrofoam, and there have been several flat panel systems commercially produced in recent years.

Distributed mode loudspeakers

A newer implementation of the flat panel speaker system involves an intentionally flexible panel and an "exciter", mounted off-center and located so as to excite the panel to vibrate with minimal resonances. Speakers using such techniques can reproduce sound with a wide directivity pattern (paradoxically somewhat like a point source) and have been used in some computer speaker designs and bookshelf loudspeakers.

Heil air motion transducers

Oskar Heil invented the air motion transducer in the 1960s. In this approach, a pleated diaphragm is mounted in a magnetic field and forced to close and open under control of a music signal. Air is forced from between the pleats in accordance with the imposed signal, generating sound. The drivers are less fragile than ribbons and considerably more efficient (and able to produce higher absolute output levels) than ribbon, electrostatic, or planar magnetic tweeter designs.

ESS, a California manufacturer, licensed the design, employed Heil, and produced a range of speaker systems using his tweeters during the 1970s and 1980s. Radio Shack, a large US retail store chain, also sold speaker systems using such tweeters for a time. At present, there are two manufacturers of these drivers in Germany, one of which produces a range of high end professional speakers using tweeters and mid-range drivers based on the technology. Martin Logan produces several AMT speakers in the US.

Plasma arc speakers

Plasma arc loudspeakers use electrical plasma as a radiating element. Since plasma has minimal mass, but is charged and therefore can be manipulated by an electric field, the result is a very linear output at frequencies far higher than the audible range. Problems of maintenance and reliability for this approach tend to make it unsuitable for mass market use. In 1978 Alan E. Hill of the Air Force Weapons Laboratory in Albuquerque, NM, designed the Plasmatronics Hill Type I, a tweeter whose plasma was generated from helium gas. This avoided the ozone and nitrous oxide produced by RF decomposition of air in an earlier generation of plasma tweeters made by the pioneering DuKane Corporation, who produced the Ionovac (marketed as the Ionofane in the UK) during the 1950s. Currently, there remain a few manufacturers in Germany who use this design, and a do-it-yourself design has been published and has been available on the Internet.

A less expensive variation on this theme is the use of a flame for the driver, as flames contain ionized (electrically charged) gases.

Digital speakers

Digital speakers have been the subject of experiments performed by Bell Labs as far back as the 1920s. The design is simple; each bit controls a driver, which is either fully 'on' or 'off'.

There are problems with this design which have led to it being abandoned as impractical for the present. First, for a reasonable number of bits (required for adequate sound reproduction quality), the physical size of a speaker system becomes very large. Secondly, due to inherent analog digital conversion problems, the effect of aliasing is unavoidable, so that the audio output is "reflected" at equal amplitude in the frequency domain, on the other side of the sampling frequency, causing an unacceptably high level of ultrasonics to accompany the desired output. No workable scheme has been found to adequately deal with this.

The term "digital" or "digital-ready" is often used for marketing purposes on speakers or headphones, but these systems are not digital in the sense described above. Rather, they are conventional speakers which can be used with digital sound sources (e.g., optical media, MP3 players, etc.), as can any conventional speaker.

Chapter-6

Tweeter

A shielded Peerless v-line dome tweeter

A **tweeter** is a loudspeaker designed to produce high audio frequencies, typically from around 2,000 Hz to 20,000 Hz (generally considered to be the upper limit of human hearing). Some tweeters can manage response up to 45 kHz. The name is derived from the high pitched sounds made by some birds, especially in contrast to the low woofs made by many dogs, after which low-frequency drivers are named (woofers).

Operation

Nearly all tweeters are electrodynamic drivers, using a voice coil suspended within a fixed magnetic field. These designs operate by applying current from the output of an amplifier circuit to a coil or voice coil. The electrified voice coil produces a varying magnetic field, which works against the fixed magnetic field of a magnet around which the cylindrical voice coil is suspended, forcing the voice coil—and the diaphragm attached to it—to move. This mechanical movement exactly resembles the waveform of the electronic signal supplied from the amplifier's output to the voice coil. Since the coil is attached to a diaphragm, the vibratory motion of the voice coil transmits to the diaphragm; the diaphragm in turn vibrates the air — thus creating air motions or audio waves, which we hear as high sounds.

Modern tweeters are typically different from older tweeters, which were usually small versions of woofers. As tweeter technology has advanced, different design applications have become popular. Many soft dome tweeter diaphragms are thermoformed from polyester film, or silk or polyester fabric that has been impregnated with a polymer resin. Hard dome tweeters commonly employ aluminium, aluminium-magnesium alloys, or titanium.

Tweeters are intended to convert an electrical signal into mechanical air movement with nothing added or subtracted, but the process is imperfect, and real-world tweeters involve trade-offs. Among the challenges in tweeter design and manufacture are: providing adequate damping, to stop the dome's motion rapidly when the signal ends; ensuring suspension linearity, to allow high output at the low end of its frequency range; ensuring freedom from contact with the magnet assembly, keeping the dome centered as it moves; and providing adequate power handling without adding excessive mass.

Dome materials

All dome materials have advantages and disadvantages. Three properties designers look for in domes are low mass, high stiffness and good damping. Celestion were the first manufacturers to fabricate dome tweeters out of a metal, copper. Nowadays other metals such as aluminium, titanium, magnesium, and beryllium, as well as various alloys thereof, are used, being both light and stiff but having low damping; their resonant modes occur above 20 kHz. More exotic materials, such as synthetic diamond, are also being used for their extreme stiffness. Polyethylene terephthalate film and woven silk suffer less ringing, but are not nearly as stiff, which can limit their very high frequency output.

In general, smaller dome tweeters provide wider dispersion of sound at the highest frequencies. However, smaller dome tweeters have less radiating area, which limits their output at the lower end of their range; and they have smaller voice coils, which limit their overall power output.

Ferrofluid

Ferrofluid is a suspension of very small (typically 10 nm) iron oxide magnetic particles in a very low volatility liquid, typically a synthetic oil. A wide range of viscosity and magnetic density variants allow designers to add damping, cooling, or both. Ferrofluid also aids in centering the voice coil in the magnetic gap, reducing distortion. The fluid is typically injected into the magnetic gap and is held in place by the strong magnetic field. If a tweeter has been subjected to elevated power levels, some thickening of the ferrofluid occurs, as a portion of the carrier liquid evaporates. In extreme cases, this can degrade the sound quality and output level of a tweeter, and the fluid must be removed and new fluid installed.

Professional sound applications

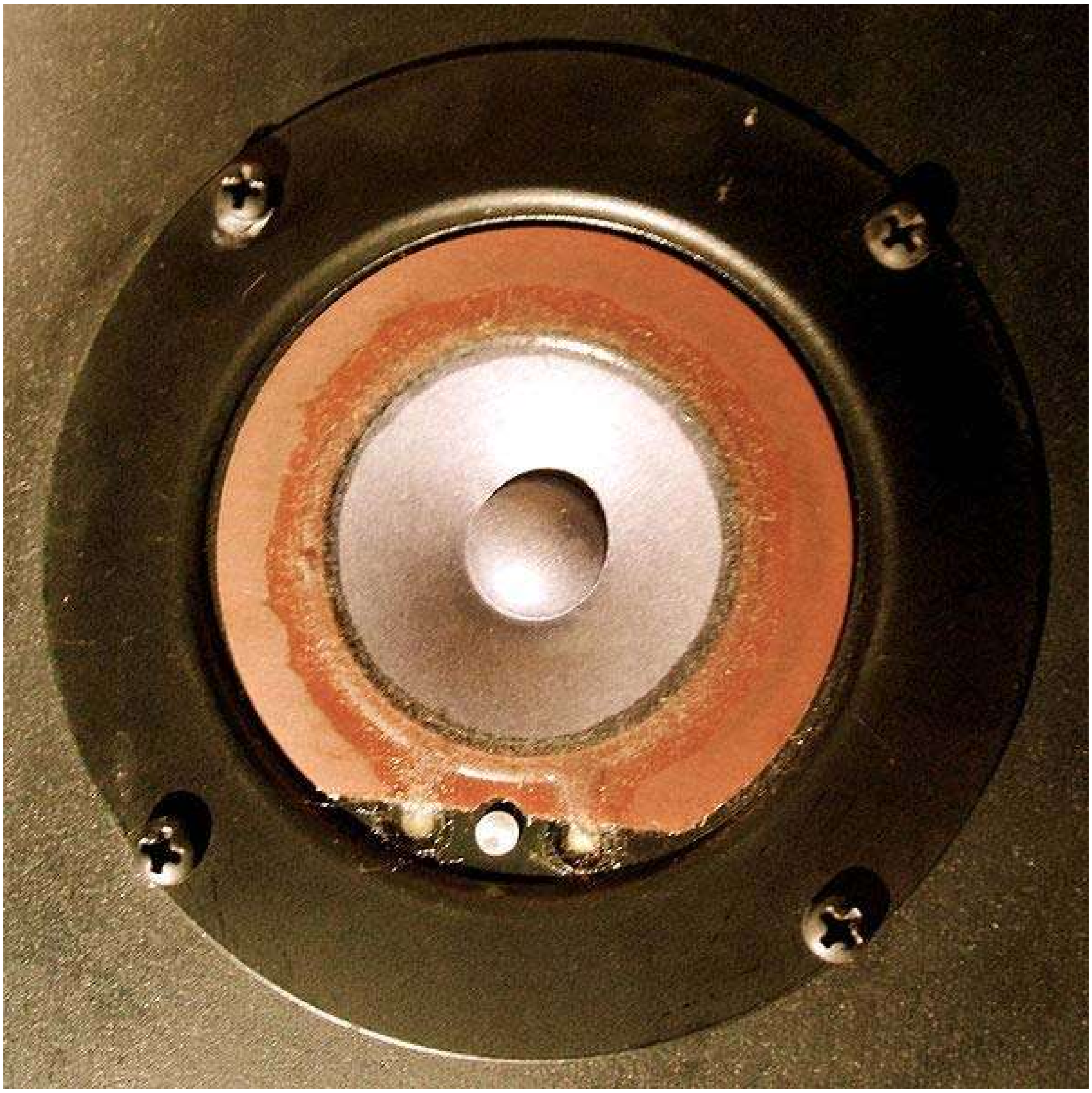
Tweeters designed for sound reinforcement and musical instrument applications are broadly similar to high fidelity tweeters, though they're usually not referred to as tweeters, but as "high frequency drivers". Key design requirement differences are:

mountings built for repeated shipping and handling, drivers often mounted to horn structures to provide for higher sound levels and greater control of sound dispersion, and more robust voice coils to withstand the higher power levels typically encountered. High frequency drivers in PA horns are often referred to as "compression drivers" from the mode of acoustic coupling between the driver diaphragm and the horn throat.

Various materials are used in the construction of compression driver diaphragms including titanium, aluminium, phenolic impregnated fabric, polyimide and PET film, each having its own characteristics. The diaphragm is glued to a voice coil former, typically made from a different material from the dome, since it must cope with heat without tearing or significant dimensional change. Polyimide film, Nomex, and glassfibre are popular for this application. The suspension may be a continuation of the diaphragm and is glued to a mounting ring, which may fit into a groove, over locating pins, or be fastened with machine screws. The diaphragm is generally shaped like an inverted dome and loads into a series of tapered channels in a central structure called a 'phase plug', which equalizes the path length between various areas of the diaphragm and the horn throat, preventing acoustic cancellations between different points on the diaphragm surface. The phase plug exits into a tapered tube, which forms the start of the horn itself. This slowly expanding throat within the driver is continued in the horn flare. The horn flare controls the coverage pattern, or directivity, and as an acoustic transformer, adds gain. A professional horn and compression driver combination has an output sensitivity of between 105 and 112dB/watt/meter. This is substantially more efficient (and less thermally dangerous to a small voice coil and former) than other tweeter construction.

Types of tweeters

Cone tweeter



The cone tweeter from a Marantz 5G loudspeaker

Cone tweeters have the same basic design and form as a woofer with optimizations to operate at higher frequencies. The optimizations usually are:

- a very small and light cone so it can move rapidly;
- cone materials chosen for stiffness (e.g., ceramic cones in one manufacturer's line), or good damping properties (e.g., silk or coated fabric) or both;
- the suspension (or spider) is stiffer than for other drivers—less flexibility is needed for high frequency reproduction;

- small voice coils (3/4 inch is typical) and light (thin) wire, which also helps the tweeter cone move rapidly.

Cone tweeters are relatively cheap, but do not have the dispersion characteristics of domes. Thus they are routinely seen in low cost applications such as factory car speakers, shelf stereo systems, and boom boxes. Cone tweeters can also be found in older stereo hi-fi system speakers designed and manufactured before the advent of the dome tweeter. They are now a rare sight in modern hi-fi usage.

Dome tweeter

A dome tweeter is constructed by attaching a voice coil to a dome (made of woven fabric, thin metal or other suitable material), which is attached to the magnet or the top plate via a low compliance suspension. These tweeters typically do not have a frame or basket, but a simple front plate attached to the magnet assembly. Dome tweeters are categorized by their voice coil diameter, and range from 19 mm (0.75 in), through 38 mm (1.5 in). The overwhelming majority of dome tweeters presently used in hi-fi speakers are 25 mm (1 in) in diameter.

A variation is the ring radiator in which the 'suspension' of the cone or dome becomes the major radiating element. These tweeters have different directivity characteristics when compared to standard dome tweeters.

Piezo tweeter

A piezo (or piezo-electric) tweeter contains a piezoelectric crystal coupled to a mechanical diaphragm. An audio signal is applied to the crystal, which responds by flexing in proportion to the voltage applied across the crystal's surfaces, thus converting electrical energy into mechanical. The conversion of electrical pulses to mechanical vibrations and the conversion of returned mechanical vibrations back into electrical energy is the basis for ultrasonic testing. The active element is the heart of the transducer as it converts the electrical energy to acoustic energy, and vice versa. The active element is basically a piece of polarized material (i.e. some parts of the molecule are positively charged, while other parts of the molecule are negatively charged) with electrodes attached to two of its opposite faces. When an electric field is applied across the material, the polarized molecules will align themselves with the electric field, resulting in induced dipoles within the molecular or crystal structure of the material. This alignment of molecules will cause the material to change dimensions. This phenomenon is known as electrostriction. In addition, a permanently-polarized material such as quartz (SiO_2) or barium titanate ($BaTiO_3$) will produce an electric field when the material changes dimensions as a result of an imposed mechanical force. This phenomenon is known as the piezoelectric effect.

Ribbon tweeter

A Philips ribbon tweeter.

A ribbon tweeter uses a very thin diaphragm (often of aluminum, or perhaps metalized plastic film) which supports a planar coil frequently made by deposition of aluminium vapor, suspended in a powerful magnetic field (typically provided by neodymium magnets) to reproduce high frequencies. The development of ribbon tweeters has more or less followed the development of ribbon microphones. The ribbon is of very lightweight material and so capable of very high acceleration and extended high frequency response. Ribbons have traditionally been incapable of high output (large magnet gaps leading to poor magnetic coupling is the main reason). But higher power versions of ribbon tweeters are becoming common in large scale sound reinforcement line array systems, which can serve audiences of thousands. They are attractive in these applications since nearly all ribbon tweeters inherently exhibit useful directional properties, with very wide horizontal dispersion (coverage) and very tight vertical dispersion. These drivers can easily be stacked vertically, building a high frequency line array that produces high sound pressure levels much farther away from the speaker locations than do conventional tweeters.

Planar-magnetic tweeter

Some loudspeaker designers use a planar-magnetic tweeter, sometimes called a quasi-ribbon. Planar magnetic tweeters are generally less expensive than true ribbon tweeters, but are not precisely equivalent as a metal foil ribbon is lighter than the diaphragm in a planar magnetic tweeter and the magnetic structures are different. Usually a thin piece of PET film or plastic with a voice coil wire running numerous times vertically on the material is used. The magnet structure is less expensive than for ribbon tweeters. The concept is most similar to that of electrostatic tweeters, with the advantage that there is no DC voltage field needed as in electrostatics, nor arcing, nor dust attraction.

Electrostatic tweeter

A Shackman MHT85 Electrostatic Tweeter.

An electrostatic tweeter operates on the same principles as a full-range electrostatic speaker or a pair of electrostatic headphones. This type of speaker employs a thin diaphragm (generally plastic and typically PET film), with a thin conductive coating, suspended between two screens or perforated metal sheets, referred to as stators.

The output of the driving amplifier is applied to the primary of a step-up transformer with a center-tapped secondary, and a very high voltage—several hundred to several thousand volts—is applied between the center tap of the transformer and the diaphragm. Electrostatics of this type necessarily include a high voltage power supply to provide the high voltage used. The stators are connected to the remaining terminals of the transformer. When an audio signal is applied to the primary of the transformer, the stators are electrically driven 180 degrees out of phase, alternately attracting and repelling the diaphragm.

An uncommon way of driving an electrostatic speaker without a transformer is to connect the plates of a push-pull vacuum tube amplifier directly to the stators, and the high voltage supply between the diaphragm and ground.

Electrostatics have reduced even-order harmonic distortion because of their push-pull design. They also have minimal phase distortion. The design is quite old (the original patents date to the 1930s), but occupies a very small segment of the market because of high costs, low efficiency, large size for full range designs, and fragility.

AMT tweeter

The Air Motion Transformer tweeter works by pushing air out perpendicularly from the pleated diaphragm. Its diaphragm is the folded pleats of film (typically PET film) around aluminium struts held in a strong magnetic field. In past decades, ESS of California produced a series of hybrid loudspeakers using such tweeters, along with conventional woofers, referring to them as Heil transducers after their inventor, Oskar Heil. They are capable of considerable output levels and are rather more sturdy than electrostatics or ribbons, but have similar low-mass moving elements.

Most of the current AMT drivers in use today are similar in efficiency and frequency response to the original Oskar Heil designs of the 1970s.

Horn tweeter

A horn tweeter is any of the above tweeters coupled to a flared or horn structure. Horns are used for two purposes — to control dispersion, and to couple the tweeter diaphragm to the air for higher efficiency. The tweeter in either case is usually termed a compression driver and is quite different from more common types of tweeters. Properly used, a horn improves the off-axis response of the tweeter by controlling (i.e., reducing directivity) of the tweeter. It can also improve the efficiency of the tweeter by coupling the relatively high acoustic impedance of the driver to the lower impedance of the air. The larger the horn, the lower the frequencies at which it can work, since large horns provide coupling to the air at lower frequencies. There are different types of horns, including radial and constant directivity (CD). Horn tweeters may have a somewhat 'different' sonic signature than simple dome tweeters. Poorly designed horns, or improperly crossed-over horns, have predictable problems in the accuracy of their output, and the load that they present to the amplifier. Perhaps concerned about the image of poorly designed horns, some manufacturers use horn loaded tweeters, but avoid using the term. Their euphemisms include "elliptical aperture" "Semi-horn" and "Directivity controlled". These are, nonetheless, a form of horn loading.

Plasma or Ion tweeter

Because ionized gas is electrically charged and so can be manipulated by a variable electrical field, it is possible to use a small sphere of plasma as a tweeter. Such tweeters are called a "plasma" tweeter or "ion" tweeter. They are more complex than other tweeters (plasma generation is not required in other types), but offer the advantage that the moving 'diaphragm' is optimally low mass, and so very responsive to the signal input. These types of tweeters are not capable of high output, nor of other than very high frequency reproduction, and so are usually used at the throat of a horn structure to manage usable output levels. One disadvantage is that the plasma arc typically produces ozone, a poison gas, in small quantities as a by-product. Because of this, German-made Magnat "magnasphere" speakers were banned from import to the USA in the 1980s.

In the past, the dominant supplier was DuKane near St Louis in the US, who made the Ionovac; also sold in a UK variant as the Ionophane. Electro-Voice made a model for a short time under license from DuKane. These early models were finicky and required regular replacement of the cell in which the plasma was generated (the DuKane unit used a precision machined quartz cell). As a result, they were expensive units in comparison to other designs. Those who have heard the Ionovacs report that, in a sensibly designed loudspeaker system, the highs were 'airy' and very detailed, though high output wasn't possible.

In the 1980s, the Plasmatronics speaker also used a plasma tweeter, though the manufacturer did not stay in business very long and very few of these complex units were sold.

Repair

Some tweeters are prone to damage, and their repair is part of the work of repair shops and maintenance crews.

Dome tweeters are often little protected in domestic speaker cabinets, and are vulnerable to dome denting. Whether a dented dome works acceptably or not depends on whether the distortion makes the voice coil out of round. Domes are undented by various methods, including:

- vacuum cleaner nozzle
- sticky tape
- bent pin
- removal & refit of the dome assembly, enabling access to the rear of the dome

Paper cone tweeters are sometimes prone to tearing of the paper cone. However these are usually old tweeters with acceptable but uninspired performance, and low value, and repair is usually considered not worthwhile. Cones are sometimes repaired with a small piece of plasticised paper (e.g., vinyl record lining paper) and a flexible glue, though this adds weight and thus affects high frequency performance. Glue alone adds less weight but is more prone to failure.

Electrostatic tweeters can suffer holing of the membrane due to arcing. Whole membranes are replaced if in poor condition, but the membrane resistance requires matching for proper performance. Either OEM film is used, or charcoal is applied to bare plastic film and polished off to reach the required resistance.

Horn tweeters occasionally need debris removed. It is either fished out with a hook or the horn is removed.

Chapter-7

Subwoofer

a 12" subwoofer driver without an enclosure

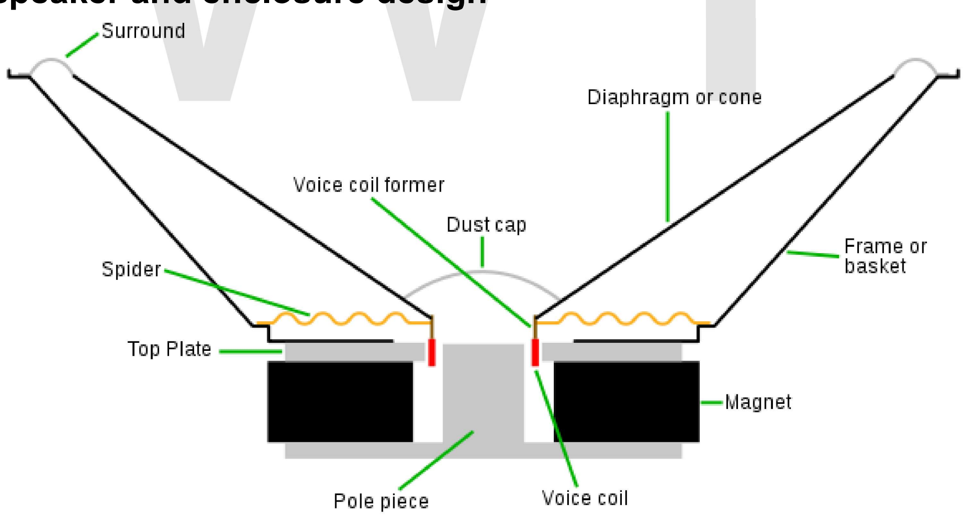
A **subwoofer** (or simply "sub") is a woofer, or a complete loudspeaker, which is dedicated to the reproduction of low-pitched audio frequencies (the "bass"). The typical frequency range for a subwoofer is about 20–200 Hz for consumer products, below 100 Hz for professional live sound, and below 80 Hz in THX-approved systems. Subwoofers are intended to augment the low frequency range of loudspeakers covering higher frequency bands.

Subwoofers are made up of one or more woofers in a loudspeaker enclosure capable of withstanding air pressure while resisting deformation. Subwoofer enclosures come in a variety of designs, including bass reflex (with a port or passive radiator in the enclosure), infinite baffle, horn-loaded, and bandpass designs, representing unique tradeoffs with respect to efficiency, bandwidth, size and cost. Passive subwoofers have a subwoofer driver and enclosure and they are powered by an external amplifier. Active subwoofers include a built-in amplifier.

The first subwoofers were developed in the 1960s to add bass response to home stereo systems. Subwoofers came into greater popular consciousness in the 1970s with the introduction of Sensurround in movies such as *Earthquake*, which produced loud low-frequency sounds through large subwoofers. With the advent of the compact cassette and the compact disc in the 1980s, the easy reproduction of deep *and* loud bass was no longer limited by the ability of a phonograph record stylus to track a groove, and producers could add more low frequency content to recordings. As well, during the 1990s, DVDs were increasingly recorded with "surround sound" processes that included a Low-frequency effects (LFE) channel, which could be heard using the subwoofer in home theater systems. During the 1990s, subwoofers also became increasingly popular in home stereo systems, custom car audio installations, and in PA systems. By the 2000s, subwoofers became almost universal in sound reinforcement systems in nightclubs and concert venues.

Construction and features

Loudspeaker and enclosure design



Cross-section of a subwoofer drive unit. *Image not to scale.*

Subwoofers use speaker drivers (woofers) typically between 8" and 21" in diameter. Some uncommon subwoofers use larger drivers, and single prototype subwoofers as large as 60" have been fabricated. On the smaller end of the spectrum, subwoofer drivers as small as 4" may be used, depending on the design of the loudspeaker enclosure, the desired sound pressure level, the lowest frequency targeted and the level of permitted distortion. The most common subwoofer driver sizes used for sound reinforcement are 10", 12", 15" and 18" models. The largest available sound reinforcement subwoofers, 21" drivers, are less commonly seen.

The efficiency of a speaker driver is given by:

$$\eta_0 = \left(\frac{4\pi^2 F_s^3 V_{as}}{c^3 Q_{es}} \right) \times 100\ \%$$

Where the variables are Thiele/Small parameters. Deep low frequency extension is a common goal for a subwoofer and small box volumes are also considered desirable. Hoffman's Iron Laws therefore mandate low efficiency under those constraints, and indeed most subwoofers require considerable power, much more than other individual drivers.

So for the example of a sealed speaker box, the box volume to achieve a given Q_{ts} is proportional to Vas:

$$V_b = \frac{V_{as}}{\alpha} \text{ Where: } \alpha = \frac{Q_{tc}^2}{Q_{ts}^2} - 1$$

Therefore a decrease in box volume and the same F_3 will decrease the efficiency of the sub woofer. Similarly the F_3 of a speaker is proportional to Fs:

$$F_c = \frac{(Q_{tc} F_s)}{Q_{ts}}$$

As the efficiency is proportional to Fs^3, small improvements in low frequency extension with the same driver and box volume will result in very significant reductions in efficiency. For these reasons, subwoofers are typically very inefficient at converting electrical energy into sound energy. This combination of factors accounts for the higher power output of subwoofer amplifiers, and the requirement for greater power handling for subwoofer drivers. Enclosure variations (e.g., bass reflex designs) are sometimes used for subwoofers to increase the efficiency of the driver/enclosure system, helping to reduce the amplifier power requirement.

Subwoofer mounted in a sealed enclosure

Subwoofers have been designed using a number of enclosure approaches: bass reflex, acoustic suspension, infinite baffle, horn loaded, tapped horn, transmission line and bandpass. Each enclosure type has advantages and disadvantages in efficiency increase, bass extension, cabinet size, distortion, and cost. Subwoofers are typically constructed by mounting one or more woofers in a cabinet of medium-density fibreboard (MDF), oriented strand board (OSB), plywood, fiberglass, aluminum or other stiff materials. Because of the high air pressure they produce in the cabinet, subwoofer enclosures often require internal bracing to distribute the resulting forces.

The smallest subwoofers are typically those designed for desktop multimedia systems. The largest common subwoofer enclosures are those used for concert sound reinforcement systems or dance club sound systems. An example of a large concert subwoofer enclosure is the 1980s-era ElectroVoice MT-4 "Bass Cube" system, which used four 18" drivers. An example of a subwoofer that uses a bass horn is the Bassmaxx B-Two, which loads an 18" driver onto an 11-foot (3.4 m) long folded horn. Folded horn-type subwoofers can typically produce a deeper range with greater efficiency than the same driver in an enclosure that lacks a horn. Some experimental fixed-installation subwoofer horns have been constructed using brick and concrete to produce a very long horn that allows a very deep sub-bass extension.

Subwoofer output level can be increased by increasing cone surface area or by increasing cone excursion. Since large drivers require undesirably large cabinets, most subwoofer drivers have large excursions. Unfortunately, high excursion, at high power levels, tends to produce more distortion from inherent mechanical and magnetic effects in electro-dynamic drivers (the most common sort). The conflict between assorted goals can never be fully resolved; subwoofer designs are necessarily compromises. Hoffman's Iron Law (the efficiency of a woofer system is directly proportional to its cabinet volume and to the cube of its cutoff frequency) applies to subwoofers just as to all loudspeakers.

Frequency range and frequency response

The typical frequency range for a subwoofer is between 20–200 Hz. Professional concert sound system subwoofers typically operate below 100 Hz, and THX-approved systems operate below 80 Hz. The frequency response specification of a speaker "attempts to describe the range of frequencies or musical tones a speaker can reproduce, measured in Hertz" Subwoofers vary in terms of the range of pitches that they can reproduce, depending on a number of factors such as the size of the cabinet and the construction and design of the enclosure and driver(s). Specifications of frequency response depend wholly for relevance on an accompanying amplitude value—measurements taken within a wider amplitude range will give any loudspeaker a wider frequency response. For example, the JBL 4688 TCB Subwoofer System, a now-discontinued system which was designed for movie theaters, had a frequency response of 23–350 Hz when measured within a 10-decibel boundary (0 dB to -10 dB) and a narrower frequency response of 28–120 Hz when measured within a six-decibel boundary (±3 dB).

As well, subwoofers vary in regards to the sound pressure levels achievable and the distortion levels they can produce over their range. The Abyss subwoofer, for example can reproduce pitches from 18 Hz (which is about the pitch of the lowest rumbling notes on a huge pipe organ with 32-foot (9.8 m) bass pipes) to 120 Hz (±3 dB). Nevertheless, even though the Abyss subwoofer can go down to 18 Hz, its lowest frequency and maximum SPL with a limit of 10% distortion at 2 meters in a large room is 35.5 Hz at 79.8 dB. This means that a person choosing a subwoofer needs to consider more than just the lowest pitch that that sub can reproduce.

Amplification

'Active subwoofers' include their own dedicated amplifiers within the cabinet. Some also include user-adjustable equalization that allows boosted or reduced output at particular frequencies; these vary from a simple "boost" switch, to fully parametric equalizers meant for detailed speaker and room correction. Some such systems are even supplied with a calibrated microphone to measure the subwoofer's in-room response, so the automatic equalizer can correct the combination of subwoofer, subwoofer location, and room response to minimize effects of room modes and improve low frequency performance.

'Passive subwoofers' have a subwoofer driver and enclosure, but they do not include an amplifier. They sometimes incorporate internal passive crossovers, with the filter frequency determined at the factory. These are generally used with third-party power amplifiers, taking their inputs from active crossovers earlier in the signal chain. While few high-end home-theater systems use passive subwoofers, this format is still popular in the professional sound industry. Using a passive subwoofer adds flexibility for the user, because the user can select which type of amplifier (Class AB or Class D, for example); brand of amplifier; or features (e.g., limiting to prevent distortion) that they want to use with their speaker or speakers.

Equalization

Equalization can be used to adjust the in-room response of a subwoofer system. Designers of active subwoofers sometimes include a degree of corrective equalization to compensate for known performance issues (e.g., a steeper than desired low end roll-off rate). In addition, many amplifiers include an adjustable low-pass filter, which prevents undesired higher frequencies from reaching the subwoofer driver. For example, if a listener's main speakers are usable down to 80 Hz, then the subwoofer filter can be set so the subwoofer only works below 80. Realizable filter behavior does not permit such sharp cutoffs, so some overlap is to be expected and must be compensated for. Digital crossover filters can produce sharper and more precise cutoff characteristics than analog filters. The crossover section may also include a high-pass "infrasonic" filter which prevents the subwoofer driver from attempting to reproduce frequencies below its safe capabilities.

Some systems use parametric equalization in an attempt to correct for room frequency response irregularities. Equalization is often unable to achieve flat frequency response at all listening locations in part because of the resonance (i.e., standing wave) patterns at low frequencies in nearly all rooms. Careful positioning of the subwoofer within the room can also help flatten the frequency response. Multiple subwoofers can manage a flatter general response since they can often be arranged to excite room modes more evenly than a single subwoofer, allowing equalisation to be more effective.

Phase control

Changing the relative phase of the subwoofer with respect to the woofers in other speakers may or may not help to minimize unwanted destructive acoustic interference in the frequency region covered by both subwoofer and main speakers. It may not help at all frequencies, and may create further problems with frequency response, but is even so generally provided as an adjustment for subwoofer amplifiers. Phase control circuits may be a simple polarity reversal switch or a more complex continuously variable circuits.

Continuously variable phase control circuits are common in subwoofer amplifiers, and may be found in crossovers and as do-it-yourself electronics projects. Phase controls allow the listener to change the arrival time of the subwoofer sound waves relative to the same frequencies from the main speakers (i.e., at and around the crossover point to the subwoofer). A similar effect can be achieved with the delay control on many home theater receivers. The subwoofer phase control found on many subwoofer amplifiers is actually a polarity inversion switch. It allows users to reverse the polarity of the subwoofer relative to the audio signal it is being given. This type of control allows the subwoofer to either be in phase with the source signal, or 180 degrees out of phase.

Servo subwoofers

Some active subwoofers use a servo feedback mechanism based on cone movement which modifies the signal sent to the voice coil. The servo feedback signal is derived from a comparison of the input signal to the amplifier versus the actual motion of the cone. The usual source of the feedback signal is a few turns of voice coil attached to the cone or a microchip-based accelerometer placed on the cone itself. An advantage of a well-implemented servo subwoofer design is reduced distortion making smaller enclosure sizes possible. The primary disadvantages are cost and complexity.

Servo controlled subwoofers are not the same as Servodrive subwoofers whose primary mechanism of sound reproduction avoids the normal voice coil and magnet combination in favor of a high-speed belt-driven servomotor. The Servodrive design increases output power, reduces harmonic distortion and virtually eliminates the loss of loudspeaker output that results from an increase in voice coil impedance due to overheating of the voice coil (called *power compression*.) This feature allows high power operation for extended periods of time. Intersonics was nominated for a TEC Award for its Servo Drive Loudspeaker (SDL) design in 1986 and for the Bass Tech 7 model in 1990.

Applications

Home audio

The use of a subwoofer augments the bass capability of the main speakers, and allows them to be smaller without sacrificing low frequency capability. A subwoofer does not necessarily provide superior bass performance in comparison to large conventional loudspeakers on ordinary music recordings due to the typical lack of very low frequency content on such sources. However, there are recordings with substantial low frequency content that most conventional loudspeakers are ill-equipped to handle without the help of a subwoofer, especially at high playback levels, such as music for pipe organs with 32' bass pipes (16 Hz), very large bass drums on symphony orchestra recordings and electronic music with extremely low synth bass parts.

Low frequencies are not easily localized; hence many stereo and multichannel audio systems feature only one subwoofer channel and a single subwoofer can be placed off-center without affecting the perceived sound stage, since the sound produced is difficult to localize. The intention in a system with a subwoofer is often to use small main ("satellite") speakers (of which there are two for stereo and five or more for surround sound or movie tracks) and to hide the subwoofer elsewhere (e.g. behind furniture or under a table), or to augment an existing speaker to save it from having to handle woofer-destroying low frequencies at high levels.

Some users add a subwoofer because high levels of low bass are desired, even beyond what is in the original recording, as in the case of house music enthusiasts. Thus, subwoofers may be part of a package that includes satellite speakers, may be purchased separately, or may be built into the same cabinet as a conventional speaker system. For instance, some floor standing tower speakers include a subwoofer driver in the lower portion of the same cabinet. Physical separation of subwoofer and "satellite" speakers not only allows placement in an inconspicuous location, but since sub-bass frequencies are particularly sensitive to room location (due to room resonances and reverberation 'modes'), the best position for the subwoofer is not likely to be where the "satellite" speakers are located.

The 1987 Bose Acoustimass 5 stereo bass driver contained one six-inch (152 mm) driver per channel and provided crossover filtering for its two satellites

For greatest efficiency and best coupling to the room's air volume, subwoofers can be placed in a corner of the room, far from large room openings, and closer to the listener. This is possible since low bass frequencies have a long wavelength; hence there is little difference between the information reaching a listener's left and right ears, and so they cannot be readily localized. All low frequency information is sent to the subwoofer. However, unless the sound tracks have been carefully mixed for a single subwoofer channel, it's possible to have some cancellation of low frequencies if bass information in one channel is out of phase with another.

The physically separate subwoofer/satellite arrangement has been popularized by multimedia speaker systems such as Bose Acoustimass Home Entertainment Systems, Polk Audio RM2008 Series and Klipsch Audio Technologies ProMedia. Low-cost "home theater in a box" systems advertise their integration and simplicity.

Particularly among low cost "Home Theater in a Box" systems and with "boom boxes", however, inclusion of a subwoofer may be little more than a marketing device. It is unlikely that a small woofer in an inexpensively-built compact plastic cabinet will have better bass performance than well-designed conventional (and typically larger) speakers in a plywood or MDF cabinet. Mere use of the term "subwoofer" is no guarantee of good or extended bass performance. Many multimedia "subwoofers" might better be termed "bass drivers" as they are too small to produce deep bass.

Further, poorly designed systems often leave everything below about 120 Hz (or even higher) to the subwoofer, meaning that the subwoofer handles frequencies which the ear can use for sound source localization, thus introducing an undesirable subwoofer "localization effect". This is usually due to poor crossover designs or choices (too high crossover point or insufficient crossover slope) used in many computer and home theater systems; localization also comes from port noise and from typically large amounts of harmonic distortion in the subwoofer design. Home subwoofers sold individually usually include crossover circuitry to assist integration into an existing system.

Car audio

A number of subwoofers in a car hatchback

Automobiles are well suited to the "hidden" subwoofer approach due to space limitations in the passenger compartments. It is not possible, in most circumstances, to fit such large drivers and enclosures into doors or dashboards, so subwoofers are installed in the trunk or back seat space. Some car audio enthusiasts compete to produce very high sound pressure levels in the confines of their vehicle's cabin; sometimes dangerously high. The "SPL wars" have drawn much attention to subwoofers in general, but subjective competitions in sound quality ("SQ") have not gained equivalent popularity. Top SPL cars are not able to play normal music, or perhaps even to drive normally as they are designed solely for competition. Many subwoofers are capable of generating high levels in cars due to the small volume of a typical car interior. High sound levels can cause hearing loss and tinnitus if one is exposed to them for an extended period of time.

In the 2000s, several car audio manufacturers have produced subwoofers using non-circular shapes from manufacturers, including Kicker, Sony, Bazooka, and X-Tant. These shapes typically carry some sort of distortion penalties. In situations of limited mounting space they provide a greater cone area and assuming all other variables are constant, greater maximum output. An important factor in the "square sub vs round sub" argument is the effects of the enclosure used. In a sealed enclosure, the maximum displacement is determined by

$$V_{\mathrm{d}} = x_{\mathrm{max}} \times S_{\mathrm{d}}$$

where

- V_d stands for volume of displacement (in m^3)
- x_{max} to the amount of linear excursion the speaker is mechanically capable of (in m)
- S_d to the cone area of the sub woofer (in m^2).

These are some of the Thiele/Small parameters which can either be measured or found with the driver specifications.

Cinema sound

After the introduction of Sensurround, movie theater owners began installing permanent subwoofer systems. Dolby Stereo 70 mm Six Track was a six channel film sound format introduced in 1976 that used two subwoofer channels for stereo reproduction of low frequencies. In 1981, Altec introduced a dedicated cinema subwoofer model tuned to 20 Hz: the 8182. Starting in 1983, THX certification of the cinema sound experience quantified the parameters of good audio for watching films, including requirements for subwoofer performance levels and enough isolation from outside sounds so that noise did not interfere with the listening experience. This helped provide guidelines for multiplex cinema owners who wanted to isolate each individual cinema from its neighbors, even as louder subwoofers were making isolation more difficult. Specific cinema subwoofer models appeared from JBL, Electro-Voice, Eastern Acoustic Works, Kintek, Meyer Sound Laboratories and BGW Systems in the early 1990s. In 1992, Dolby Digital's six-channel film sound format incorporated a single LFE channel, the "point one" in 5.1 surround sound.

Tom Horral, a Boston-based acoustician, blames subwoofers for louder cinema sound in general. He says that before subwoofers made it possible to have loud, relatively undistorted bass, movie sound levels were limited by the distortion in less capable systems at low frequency and high levels.

Sound reinforcement

Each stack of speakers in this sound reinforcement setup consists of two EAW SB1000 direct radiating subwoofers (each contains two 18" drivers) and two EAW KF850 full range cabinets for the mid and high frequencies.

Professional audio subwoofers must be capable of very high output levels. This is reflected in the design attention given in recent years to the subwoofer applications for sound reinforcement, public address systems, dance club systems and concert systems. Consumer applications (as in home use) are considerably less demanding due to much smaller listening space and lower playback levels. Subwoofers are now almost universal in professional sound applications such as live concert sound, churches, nightclubs, and theme parks. Movie theatres certified to the THX standard for playback always include high capability subwoofers. Some professional applications require subwoofers designed for very high sound levels, using multiple 12", 15", 18" or 21" drivers. Drivers as small as 10" are occasionally used, generally in horn loaded enclosures.

The number of subwoofer enclosures used in a concert depends on a number of factors, including the size of the venue, whether it is indoors or outdoors, the amount of low-frequency content in the band's sound, the desired volume of the concert, and the design and construction of the enclosures (e.g., direct-radiating versus horn-loaded. A small bar may use a single direct-radiating 15-inch sub cabinet. A large dance club may have a row of four or five twin 18-inch subwoofer cabinets, or more). In the largest stadium venues, there may be a very large number of subwoofer enclosures. For example, the 2009–2010 U2 360° Tour uses 24 Clair Brothers BT-218 subwoofers (a double 18" box) around the perimeter of the central circular stage, and 72 proprietary Clair Brothers cardioid S4 subwoofers placed underneath the ring-shaped "B" stage which encircles the central main stage.

The main speakers may be 'flown' from the ceiling of a venue on chain hoists, and 'flying points' (i.e., attachment points) are built into many professional loudspeaker enclosures. Subwoofers can be flown or stacked on the ground near the stage. There can be more than 50 double-18-inch cabinets in a typical concert system. Just as consumer subwoofer enclosures can be made of Medium-density fibreboard (MDF), Oriented strand board (OSB), plywood, plastic or other dense material, professional subwoofer enclosures can be built from the same materials. MDF is commonly used to construct subwoofers for permanent installations as its density is relatively high and weatherproofing is not a concern. Other permanent installation subwoofers have used very thick plywood: the Altec 8182 (1981) used 7-ply 28 mm birch-faced oak plywood. Touring subwoofers are typically built from 18–20 mm thick void-free Baltic birch (Betula pendula or Betula pubescens) plywood from Finland, Estonia or Russia; such plywood affords greater strength for frequently transported enclosures. Not naturally weatherproof, Baltic birch is coated with carpet, thick paint or spray-on truck bedliner to give the subwoofer enclosures greater durability.

Touring subwoofer cabinets are typically designed with features that facilitate moving the enclosure (e.g., wheels, a "towel bar" handle and recessed handles), a protective grill for the speaker (in direct radiating-style cabinets), metal or plastic protection for the cabinets to protect the finish as the cabinets are being slid one on top of another, and hardware to facilitate stacking the cabinets (e.g., interlocking corners) and for "flying" the cabinets from stage rigging.

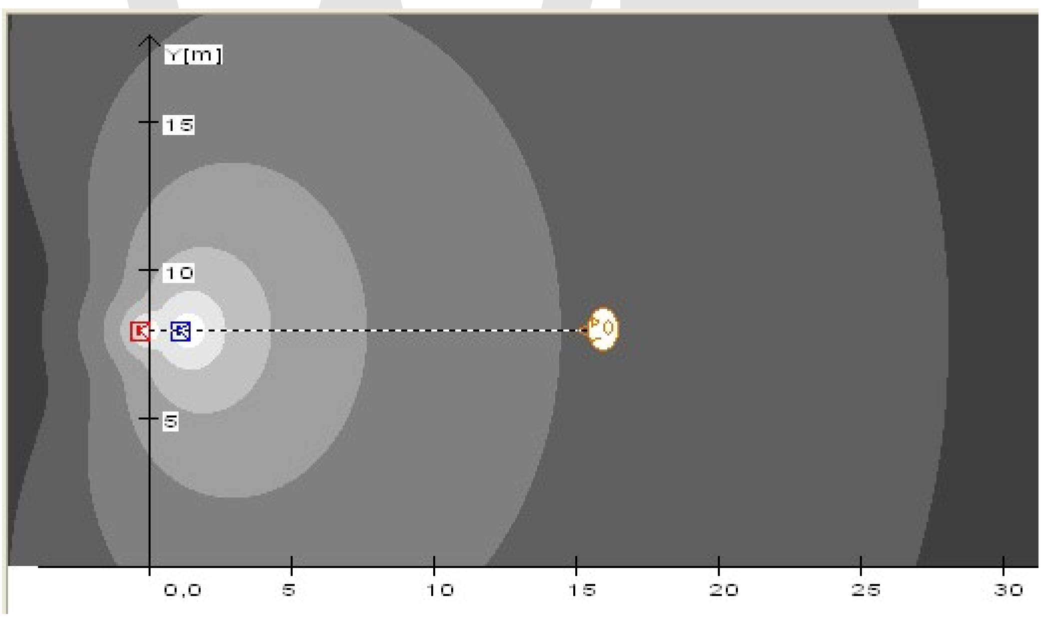
Full-range system

In professional concert sound system design, subwoofers can be incorporated seamlessly with the main speakers into a stereo or mono full-range system by using an active crossover. Such a system receives its signal from the main mono or stereo mixing console mix bus and amplifies all frequencies together in the desired balance. If the main sound system is stereo, the subwoofers can also be in stereo. Otherwise, a mono subwoofer channel can be derived within the crossover from a stereo mix, depending on the crossover make and model.

Aux-fed subwoofers

Instead of being incorporated into a full-range system, concert subwoofers can be supplied with their own signal from a separate mix bus on the mixing console; often one of the auxiliary sends ("aux" or "auxes") is used. This configuration is called "aux-fed subwoofers", and has been observed to significantly reduce low frequency "muddiness" that can build up in a concert sound system which has on stage a number of microphones each picking up low frequencies and each having different phase relationships of those low frequencies. The aux-fed subs method greatly reduces the number of sources feeding the subwoofers to include only those instruments that have desired low frequency information; sources such as kick drum, bass guitar, samplers and keys. This simplifies the signal sent to the subwoofers and makes for greater clarity and low punch. Aux-fed subs can even be stereo, if desired, using two auxiliary mix buses.

Directional bass



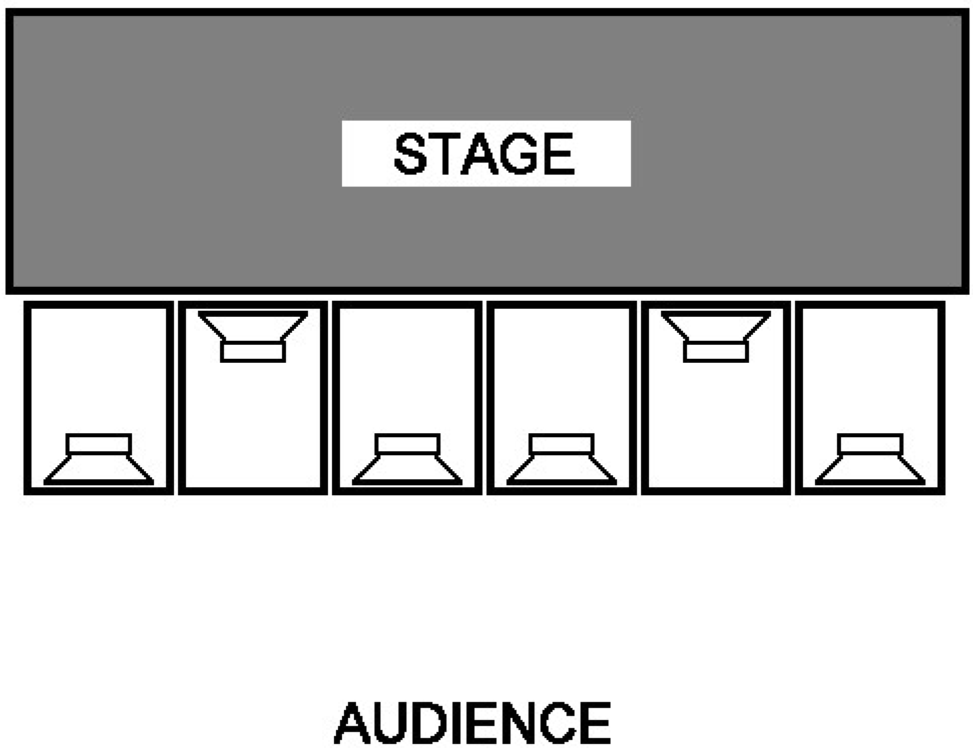
Cardioid dispersion pattern of two end-fire subwoofers placed one in front of the other. The enclosure nearest the listener is delayed by a few milliseconds.

In order to keep low frequency energy focused on the audience area and not on the stage, and to keep low frequencies from bothering people outside of the event space, a variety of techniques have been developed in concert sound to turn the naturally omnidirectional radiation of subwoofers into a more directional pattern. These techniques include setting up subwoofers in a vertical array; using combinations of delay and polarity inversion; and setting up a delay-shaded system.

Vertical array

Stacking or rigging the subwoofers in a vertical array focuses the low frequencies forward to a greater or lesser extent depending on the physical length of the array. Longer arrays have a more directional effect at lower frequencies. The directionality is more pronounced in the vertical dimension, yielding a radiation pattern that is wide but not tall. This helps reduce the amount of low frequency sound bouncing off the ceiling indoors and assists in mitigating external noise complaints outdoors.

Rear delay array

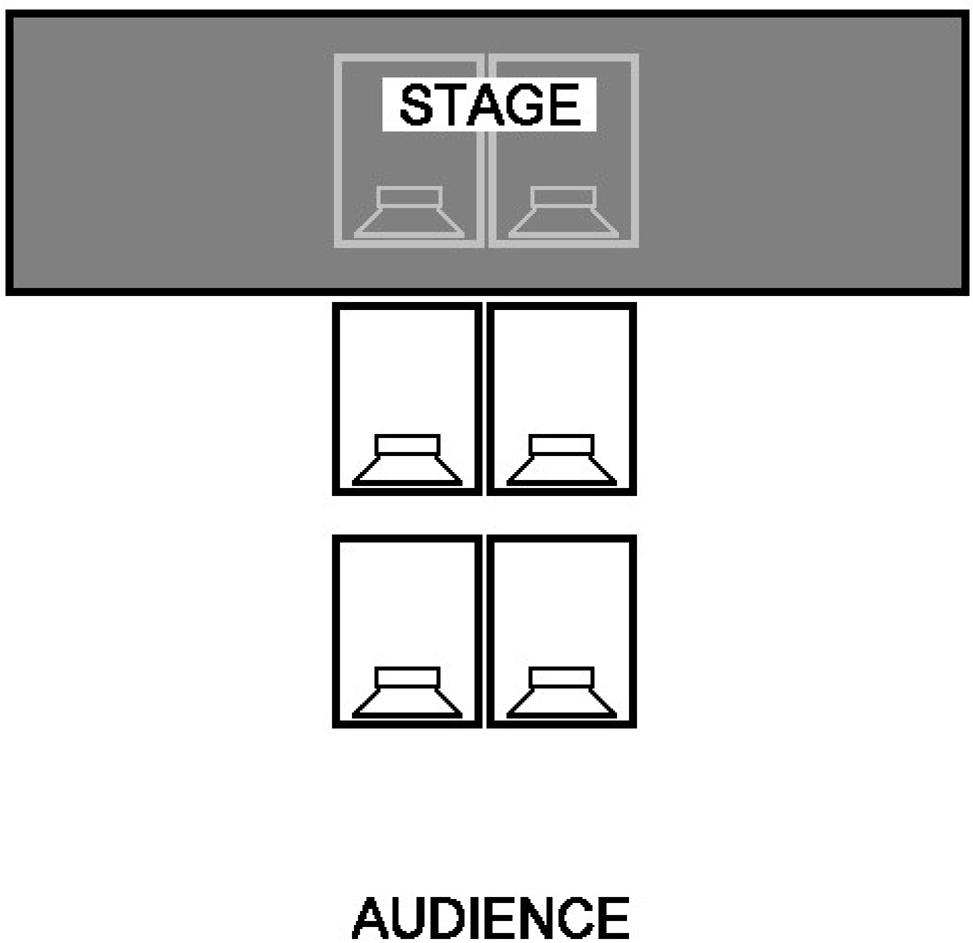


CSA: Six subwoofers arranged for less bass energy on stage. Signal going to the reversed enclosures is delayed a few milliseconds.

Another cardioid subwoofer array pattern can be used horizontally, one which takes few channels of processing and no change in required physical space. Often called "cardioid subwoofer array" or "CSA", though the pattern of all directional subwoofer methods is cardioid, the method inverts the polarity of one out of every three subwoofers across the front of the stage, and delays those enclosures for maximum cancellation of the target frequency on stage. Polarity inversion can be implemented electronically, or by physically placing the enclosure to face rearward. This method reduces forward output relative to a tight-packed, flat-fronted array of subwoofers, but can solve problems of unwanted low frequency energy coming into microphones on stage. Compared to the end-fire array, this method has less on-axis energy but more even pattern control throughout the audience, and more predictable cancellation rearward. The effect spans a range of slightly more than one octave.

A second method of rear delay array combines end-fire topology with polarity reversal, using two subwoofers positioned front to back, the drivers spaced one-quarter wavelength apart, the rear enclosure inverted in polarity and delayed by a few milliseconds for maximum cancellation on stage of the target frequency. This method has the least output power directed toward the audience, compared to other directional methods.

End-fire array



End-fire array using three rows of subwoofers. Each row is delayed a few milliseconds more than the previous row.

The end-fire subwoofer method, also called "forward steered arrays", places subwoofer drivers co-axially in one or more rows, using destructive interference to reduce emissions to the sides and rear. This can be done with separate subwoofer enclosures positioned front to back with a spacing between them of one-quarter wavelength of the target frequency, the frequency that is least wanted on stage or most desired in the audience. Each row is delayed beyond the first row by an amount related to the speed of sound in air; typically a few milliseconds. The arrival time of sound energy from all the subwoofers is near-simultaneous from the audience's perspective, but is canceled out to a large degree behind the subwoofers because of offset sound wave arrival times. Directionality of the target frequency can achieve as much as 25 dB rear attenuation, and the forward sound is coherently summed in line with the subwoofers. The positional technique of end-fire subwoofers came into widespread use in European live concert sound in 2006.

The end-fire array trades a few decibels of output power for directionality, so it requires more enclosures for the same output power as a tight-packed, flat-fronted array of enclosures. Sixteen enclosures in four rows were used in 2007 at one of the stages of the Ultra Music Festival, to reduce low frequency interference to neighboring stages. Because of the physical size of the end-fire array, few concert venues are able to implement it. The output pattern suffers from comb-filtering off-axis, but can be further shaped by adjusting the frequency response of each row of subwoofers.

Delay-shaded array

A long line of subwoofers placed horizontally along the front edge of the stage can be delayed such that the center subs fire several milliseconds prior to the ones flanking them, which fire several milliseconds prior to *their* neighbors, continuing in this fashion until the last subwoofers are reached at the outside ends of the subwoofer row. This method helps to counteract the extreme narrowing of horizontal dispersion pattern seen with a horizontal subwoofer array. Such delay shading can be used to virtually reshape a loudspeaker array.

Directional enclosure

Some subwoofer enclosure designs rely on drivers facing to the sides or to the rear in order to achieve a degree of directionality. End-fire drivers can be positioned within a single enclosure that houses more than one driver.

Bass instrument amplification

In rare cases, sound reinforcement subwoofer enclosures are also used for bass instrument amplification by electric bass players and synth bass players. For most bands and most small- to mid-size venues (e.g., nightclubs and bars), standard bass guitar speaker enclosures or keyboard amplifiers will provide sufficient sound pressure levels for onstage monitoring.Since a regular electric bass has a low "E" (41 Hz) as its lowest note, most standard bass guitar cabinets are only designed with a range that goes down to about 40 Hz. However, in some cases, performers wish to have extended sub-bass response that is not available from standard instrument speaker enclosures, so they use subwoofer cabinets. Just as some electric guitarists add huge stacks of guitar cabinets mainly for show, some bassists will add immense subwoofer cabinets with 18" woofers mainly for show, and the extension sub cabinets will be operated at a lower volume than the main bass cabinets.

Bass guitar players who may use subwoofer cabinets include performers who play with extended range basses that include a low "B" string (about 31 Hz); bassists who play in styles where a very powerful sub-bass response is an important part of the sound (e.g., funk, Latin, gospel, R & B, etc.); and/or bass players who perform in stadium-size venues or large outdoor venues. Keyboard players who use subwoofers for on-stage monitoring include electric organ players who use bass pedal keyboards (which go down to a low "C" which is about 33 Hz) and synth bass players who play rumbling sub-bass parts that go as low as 18 Hz. Of all of the keyboard instruments that are amplified onstage, synthesizers can produce some of the lowest pitches, because unlike a traditional electric piano or electric organ, which have as their lowest notes a low "A" and a low "C", respectively, a synth does not have a fixed lowest octave. A synth player can add lower octaves to a patch by pressing an "octave down" button, which can produce pitches that are at the limits of human hearing.

Several concert sound subwoofer manufacturers suggest that their subs can be used for bass instrument amplification. Meyer Sound suggests that its 650-R2 Concert Series Subwoofer, a 14-square-foot (1.3 m^2) enclosure with two 18-inch drivers, can be used for bass instrument amplification. While performers who use concert sound subwoofers for onstage monitoring may like the powerful sub-bass sound that they get onstage, sound engineers may find the use of large subwoofers (e.g., two 18" drivers) for onstage instrument monitoring to be problematic, because it may interfere with the "Front of House" sub-bass sound.

Bass shakers

Since much very low bass is felt, sub-bass can be 'augmented' using tactile transducers. Unlike a typical subwoofer driver, which produces audible vibrations, tactile transducers produce low-frequency vibrations that are designed to be felt by individuals who are touching the transducer or indirectly through a piece of furniture or a wooden floor. Tactile transducers have recently emerged as a device class, called variously "bass shakers", "butt shakers" and "throne shakers". They are attached to a seat, for instance a drummer's stool ("throne") or gamer's chair, car seat or home theater seating, and the vibrations of the driver are transmitted to the body then to the ear in a manner similar to bone conduction. They connect to an amplifier like a normal subwoofer. They can be attached to a large flat surface (for instance a floor or platform) to create a large low frequency conduction area, although the transmission of low frequencies through the feet is not as efficient as the seat.

The advantage of tactile transducers used for low frequencies is that they allow a listening environment that is not filled with loud low frequency waves. This helps the concert drummer to monitor his or her kick drum performance without "polluting" the stage with powerful low frequency waves from a 15" subwoofer monitor. By not having a subwoofer monitor, a bass shaker also enables a drummer to lower the sound pressure levels that he is exposed to during a performance. For home cinema or videogame use, bass shakers help the user avoid disturbing others in nearby apartments or rooms, because even powerful sound effects such as explosion sounds in a war videogame or the simulated rumbling of an earthquake in an adventure film will not be heard by others. However, some critics argue that the felt vibrations are disconnected from the auditory experience, and they claim that that music is less satisfying with the "butt shaker" than sound effects. As well, critics have claimed that the bass shaker itself can rattle during loud sound effects, which can distract the listener.

World record claims

With varying measures upon which to base claims, several subwoofers have been said to be the world's largest, loudest or lowest.

Matterhorn

The Matterhorn is a subwoofer model completed in March 2007 by Danley Sound Labs in Gainesville, Georgia after a U.S. military request for a loudspeaker that could project infrasonic waves over a distance. The Matterhorn was designed to reproduce a continuous sine wave from 15 to 20 Hz, and generate 94 dB at a distance of 250 meters (820 ft), and more than 140 dB for music playback measured at the horn mouth. It can generate a constant 15 Hz sine wave tone at 140 dB for 24 hours a day, seven days a week with extremely low harmonic distortion. The subwoofer has a flat frequency response from 15 to 80 Hz, and is down 3 dB at 12 Hz. It was built within an intermodal container 20 feet (6.1 m) long and 8 by 8 feet (2.4 × 2.4 m) square. The container doors swing open to reveal a tapped horn driven by 40 long-throw 15-inch speaker drivers each powered by its own 1000-watt amplifier. The manufacturer claims that 53 13-ply 18 mm 4-by-8-foot (1.2 × 2.4 m) sheets of plywood were used in its construction, though one of the fabricators wrote that double-thickness 26-ply sheets were used for convenience. A diesel generator is housed within the enclosure to supply electricity when external power is unavailable. Of the constant tone output capability, designer Tom Danley wrote that the "target 94 dB at 250 meters is not the essentially fictional 'burst' or 'peak SPL' nonsense in pro sound, or like the 'death burp' signal used in car sound contests." At the annual National Systems Contractors Association (NSCA) convention in March 2007, the Matterhorn was barred from making any loud demonstrations of its power because of concerns about damaging the building of the Orange County Convention Center. Instead, using only a single 20 amp electrical circuit for safety, visitors were allowed to step inside the horn of the subwoofer for an "acoustic massage" as the fractionally powered Matterhorn reproduced low level 10–15 Hz waves.

Royal Device custom installation

Another subwoofer claimed to be the world's biggest is a custom installation in Italy made by Royal Device primarily of bricks, concrete and sound-deadening material consisting of two subwoofers embedded in the foundation of a listening room. The horn-loaded subwoofers each have a floor mouth that is 2.2 square meters (24 sq ft), and a horn length that is 9.5 meters (31 ft), in a cavity 1 meter (3 ft 3 in) under the floor of the listening room. Each subwoofer is driven by eight 18-inch subwoofer drivers with 100 millimeters (3.9 in) voice coils. The designers assert that the floor mouths of the horns are additionally loaded acoustically by a vertical wooden horn expansion and the room's ceiling to create a 10 Hz "full power" wave at the listening position.

Concept Design 60-inch

A single 60-inch (1,500 mm) diameter subwoofer driver was designed by Richard Clark and David Navone with the help of Dr. Eugene Patronis of Georgia Institute of Technology. The driver was intended to break sound pressure level records when mounted in a road vehicle, calculated to be able to achieve more than 180 dBSPL. It was built in 1997, driven by DC motors connected to a rotary crankshaft somewhat like in a piston engine. The cone diameter was 54 inches (1,400 mm) and was held in place with a 3-inch (76 mm) surround. With a 6-inch (150 mm) peak-to-peak stroke, it created a one-way air displacement of 6,871 cubic inches (112,600 cm^3). It was capable of generating 5–20 Hz sine waves at various DC motor speeds—not as a response to audio signal—it could not play music. The driver was mounted in a stepvan owned by Tim Maynor but was too powerful for the amount of applied reinforcement and damaged the vehicle. MTX's Loyd Ivey helped underwrite the project and the driver was then called the MTX "Thunder 1000000" (one million). Still unfinished, the vehicle was entered in an SPL competition in 1997 at which a complaint was lodged against the computer control of the DC motor. Instead of using the controller, two leads were touched together in the hope that the motor speed was set correctly. The drive shaft broke after one positive stroke which created an interior pressure wave of 162 dB. The Concept Design 60-inch was not shown in public after 1998.

MTX Jackhammer

The largest production subwoofer intended for use in automobiles is the MTX Jackhammer by MTX Audio which has a 22-inch (560 mm) diameter cone. The Jackhammer can take a total of 6000 watts sent to dual voice coils moving within a 900-ounce (26 kg) strontium ferrite magnet. The Jackhammer weighs 369 pounds (167 kg) and has an aluminum heat sink. The Jackhammer has been featured on the television show Pimp My Ride.

Chapter-8

Microphone

A Neumann U87 condenser microphone with shock mount

A **microphone** (colloquially called a **mic** or **mike**) is an acoustic-to-electric transducer or sensor that converts sound into an electrical signal. In 1876, Emile Berliner invented the first microphone used as a telephone voice transmitter. Microphones are used in many applications such as telephones, tape recorders, karaoke systems, hearing aids, motion picture production, live and recorded audio engineering, FRS radios, megaphones, in radio and television broadcasting and in computers for recording voice, speech recognition, VoIP, and for non-acoustic purposes such as ultrasonic checking or knock sensors.

Most microphones today use electromagnetic induction (dynamic microphone), capacitance change (condenser microphone), piezoelectric generation, or light modulation to produce an electrical voltage signal from mechanical vibration.

Components

The sensitive transducer element of a microphone is called its *element* or *capsule.* A complete microphone also includes a housing, some means of bringing the signal from the element to other equipment, and often an electronic circuit to adapt the output of the capsule to the equipment being driven. A wireless microphone contains a radio transmitter.

Varieties

Microphones are referred to by their transducer principle, such as condenser, dynamic, etc., and by their directional characteristics. Sometimes other characteristics such as diaphragm size, intended use or orientation of the principal sound input to the principal axis (end- or side-address) of the microphone are used to describe the microphone.

Condenser microphone

Inside the Oktava 319 condenser microphone

The **condenser microphone**, invented at Bell Labs in 1916 by E. C. Wente is also called a **capacitor microphone** or **electrostatic microphone**. Here, the diaphragm acts as one plate of a capacitor, and the vibrations produce changes in the distance between the plates. There are two types, depending on the method of extracting the audio signal from the transducer: DC-biased and radio frequency (RF) or high frequency (HF) condenser microphones. With a DC-biased microphone, the plates are biased with a fixed charge (*Q*). The voltage maintained across the capacitor plates changes with the vibrations in the air, according to the capacitance equation (C = Q / V), where Q = charge in coulombs, C = capacitance in farads and V = potential difference in volts. The capacitance of the plates is inversely proportional to the distance between them for a parallel-plate capacitor. The assembly of fixed and movable plates is called an "element" or "capsule."

A nearly constant charge is maintained on the capacitor. As the capacitance changes, the charge across the capacitor does change very slightly, but at audible frequencies it is sensibly constant. The capacitance of the capsule (around 5 to 100 pF) and the value of the bias resistor (100 megohms to tens of gigohms) form a filter that is high-pass for the audio signal, and low-pass for the bias voltage. Note that the time constant of an RC circuit equals the product of the resistance and capacitance.

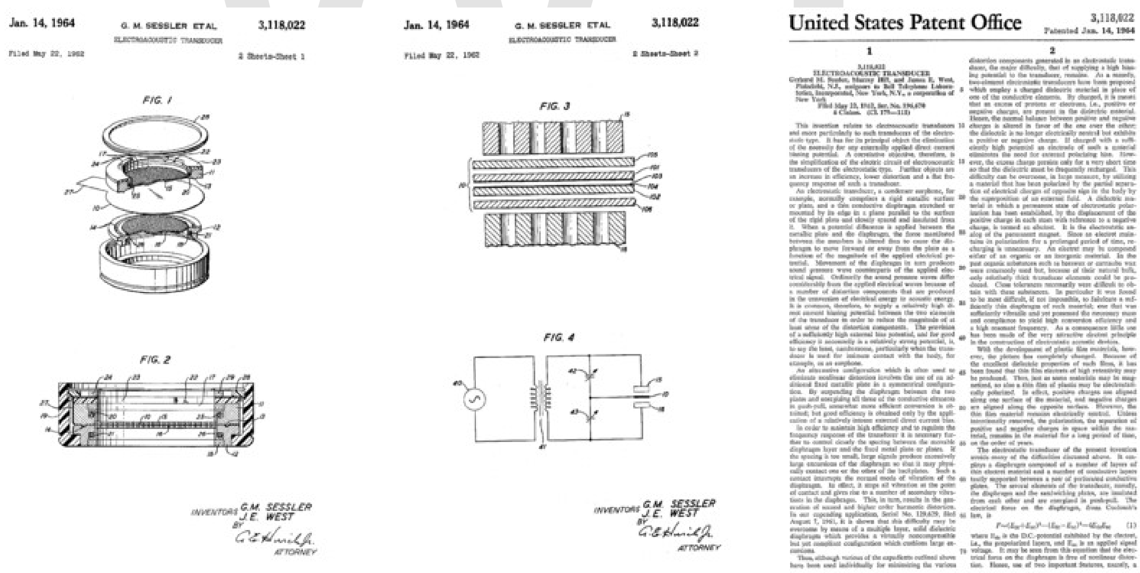
Within the time-frame of the capacitance change (as much as 50 ms at 20 Hz audio signal), the charge is practically constant and the voltage across the capacitor changes instantaneously to reflect the change in capacitance. The voltage across the capacitor varies above and below the bias voltage. The voltage difference between the bias and the capacitor is seen across the series resistor. The voltage across the resistor is amplified for performance or recording. In most cases, the electronics in the microphone itself contribute no voltage gain as the voltage differential is quite significant, up to several volts for high sound levels. Since this is a very high impedance circuit, current gain only is usually needed with the voltage remaining constant. The circuit is therefore often called an "impedance converter" or "follower" because no voltage gain is provided.

AKG C451B small-diaphragm condenser microphone

RF condenser microphones use a comparatively low RF voltage, generated by a low-noise oscillator. The signal from the oscillator may either be amplitude modulated by the capacitance changes produced by the sound waves moving the capsule diaphragm, or the capsule may be part of a resonant circuit that modulates the frequency of the oscillator signal. Demodulation yields a low-noise audio frequency signal with a very low source impedance. The absence of a high bias voltage permits the use of a diaphragm with looser tension, which may be used to achieve wider frequency response due to higher compliance. The RF biasing process results in a lower electrical impedance capsule, a useful by-product of which is that RF condenser microphones can be operated in damp weather conditions that could create problems in DC-biased microphones with contaminated insulating surfaces. The Sennheiser "MKH" series of microphones use the RF biasing technique.

Condenser microphones span the range from telephone transmitters through inexpensive karaoke microphones to high-fidelity recording microphones. They generally produce a high-quality audio signal and are now the popular choice in laboratory and studio recording applications. The inherent suitability of this technology is due to the very small mass that must be moved by the incident sound wave, unlike other microphone types that require the sound wave to do more work. They require a power source, provided either via microphone inputs on equipment as phantom power or from a small battery. Power is necessary for establishing the capacitor plate voltage, and is also needed to power the microphone electronics (impedance conversion in the case of electret and DC-polarized microphones, demodulation or detection in the case of RF/HF microphones). Condenser microphones are also available with two diaphragms that can be electrically connected to provide a range of polar patterns, such as cardioid, omnidirectional, and figure-eight. It is also possible to vary the pattern continuously with some microphones, for example the Røde NT2000 or CAD M179.

Electret condenser microphone



First patent on foil electret microphone by G. M. Sessler et al. (pages 1 to 3)

An electret microphone is a relatively new type of capacitor microphone invented at Bell laboratories in 1962 by Gerhard Sessler and Jim West. The externally applied charge described above under condenser microphones is replaced by a permanent charge in an electret material. An electret is a ferroelectric material that has been permanently electrically charged or *polarized*. The name comes from *electr*ostatic and magn*et*; a static charge is embedded in an electret by alignment of the static charges in the material, much the way a magnet is made by aligning the magnetic domains in a piece of iron.

Due to their good performance and ease of manufacture, hence low cost, the vast majority of microphones made today are electret microphones; a semiconductor manufacturer estimates annual production at over one billion units. Nearly all cell-phone, computer, PDA and headset microphones are electret types. They are used in many applications, from high-quality recording and lavalier use to built-in microphones in small sound recording devices and telephones. Though electret microphones were once considered low quality, the best ones can now rival traditional condenser microphones in every respect and can even offer the long-term stability and ultra-flat response needed for a measurement microphone. Unlike other capacitor microphones, they require no polarizing voltage, but often contain an integrated preamplifier that does require power (often incorrectly called polarizing power or bias). This preamplifier is frequently phantom powered in sound reinforcement and studio applications. Microphones designed for personal computer (PC) use, sometimes called multimedia microphones, use a stereo 3.5 mm plug (though a mono source) with the ring receiving power via a resistor from (normally) a 5 V supply in the computer; unfortunately, a number of incompatible dynamic microphones are fitted with 3.5 mm plugs too. While few electret microphones rival the best DC-polarized units in terms of noise level, this is not due to any inherent limitation of the electret. Rather, mass production techniques needed to produce microphones cheaply don't lend themselves to the precision needed to produce the highest quality microphones, due to the tight tolerances required in internal dimensions. These tolerances are the same for all condenser microphones, whether the DC, RF or electret technology is used.

Dynamic microphone

Patti Smith singing into a Shure SM58 (dynamic cardioid type) microphone

Dynamic microphones work via electromagnetic induction. They are robust, relatively inexpensive and resistant to moisture. This, coupled with their potentially high gain before feedback makes them ideal for on-stage use.

Moving-coil microphones use the same dynamic principle as in a loudspeaker, only reversed. A small movable induction coil, positioned in the magnetic field of a permanent magnet, is attached to the diaphragm. When sound enters through the windscreen of the microphone, the sound wave moves the diaphragm. When the diaphragm vibrates, the coil moves in the magnetic field, producing a varying current in the coil through electromagnetic induction. A single dynamic membrane does not respond linearly to all audio frequencies. Some microphones for this reason utilize multiple membranes for the different parts of the audio spectrum and then combine the resulting signals. Combining the multiple signals correctly is difficult and designs that do this are rare and tend to be expensive. There are on the other hand several designs that are more specifically aimed towards isolated parts of the audio spectrum. The AKG D 112, for example, is designed for bass response rather than treble. In audio engineering several kinds of microphones are often used at the same time to get the best result.

Ribbon microphone

Edmund Lowe using a ribbon microphone

Ribbon microphones use a thin, usually corrugated metal ribbon suspended in a magnetic field. The ribbon is electrically connected to the microphone's output, and its vibration within the magnetic field generates the electrical signal. Ribbon microphones are similar to moving coil microphones in the sense that both produce sound by means of magnetic induction. Basic ribbon microphones detect sound in a bi-directional (also called figure-eight) pattern because the ribbon, which is open to sound both front and back, responds to the pressure gradient rather than the sound pressure. Though the symmetrical front and rear pickup can be a nuisance in normal stereo recording, the high side rejection can be used to advantage by positioning a ribbon microphone horizontally, for example above cymbals, so that the rear lobe picks up only sound from the cymbals. Crossed figure 8, or Blumlein pair, stereo recording is gaining in popularity, and the figure 8 response of a ribbon microphone is ideal for that application.

Other directional patterns are produced by enclosing one side of the ribbon in an acoustic trap or baffle, allowing sound to reach only one side. The classic RCA Type 77-DX microphone has several externally adjustable positions of the internal baffle, allowing the selection of several response patterns ranging from "Figure-8" to "Unidirectional". Such older ribbon microphones, some of which still provide high quality sound reproduction, were once valued for this reason, but a good low-frequency response could only be obtained when the ribbon was suspended very loosely, which made them relatively fragile. Modern ribbon materials, including new nanomaterials have now been introduced that eliminate those concerns, and even improve the effective dynamic range of ribbon microphones at low frequencies. Protective wind screens can reduce the danger of damaging a vintage ribbon, and also reduce plosive artifacts in the recording. Properly designed wind screens produce negligible treble attenuation. In common with other classes of dynamic microphone, ribbon microphones don't require phantom power; in fact, this voltage can damage some older ribbon microphones. Some new modern ribbon microphone designs incorporate a preamplifier and, therefore, do require phantom power, and circuits of modern passive ribbon microphones, *i.e.*, those without the aforementioned preamplifier, are specifically designed to resist damage to the ribbon and transformer by phantom power. Also there are new ribbon materials available that are immune to wind blasts and phantom power.

Carbon microphone

A carbon microphone, also known as a carbon button microphone (or sometimes just a button microphone), use a capsule or button containing carbon granules pressed between two metal plates like the Berliner and Edison microphones. A voltage is applied across the metal plates, causing a small current to flow through the carbon. One of the plates, the diaphragm, vibrates in sympathy with incident sound waves, applying a varying pressure to the carbon. The changing pressure deforms the granules, causing the contact area between each pair of adjacent granules to change, and this causes the electrical resistance of the mass of granules to change. The changes in resistance cause a corresponding change in the current flowing through the microphone, producing the electrical signal. Carbon microphones were once commonly used in telephones; they have extremely low-quality sound reproduction and a very limited frequency response range, but are very robust devices. The Boudet microphone, which used relatively large carbon balls, was similar to the granule carbon button microphones.

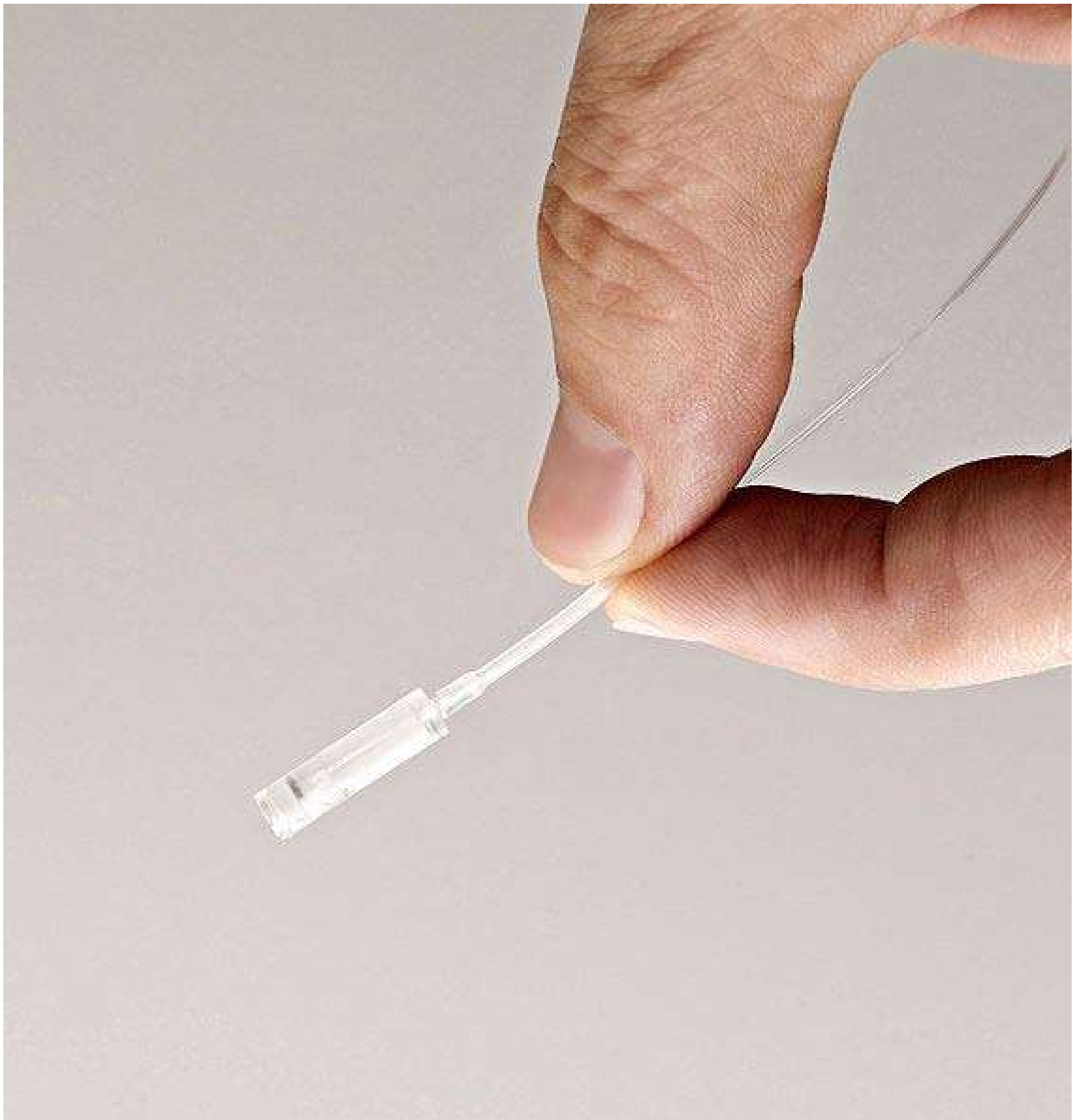
Unlike other microphone types, the carbon microphone can also be used as a type of amplifier, using a small amount of sound energy to control a larger amount of electrical energy. Carbon microphones found use as early telephone repeaters, making long distance phone calls possible in the era before vacuum tubes. These repeaters worked by mechanically coupling a magnetic telephone receiver to a carbon microphone: the faint signal from the receiver was transferred to the microphone, with a resulting stronger electrical signal to send down the line. One illustration of this amplifier effect was the oscillation caused by feedback, resulting in an audible squeal from the old "candlestick" telephone if its earphone was placed near the carbon microphone.

Piezoelectric microphone

A **crystal microphone** or **piezo microphone** uses the phenomenon of piezoelectricity — the ability of some materials to produce a voltage when subjected to pressure — to convert vibrations into an electrical signal. An example of this is potassium sodium tartrate, which is a piezoelectric crystal that works as a transducer, both as a microphone and as a slimline loudspeaker component. Crystal microphones were once commonly supplied with vacuum tube (valve) equipment, such as domestic tape recorders. Their high output impedance matched the high input impedance (typically about 10 megohms) of the vacuum tube input stage well. They were difficult to match to early transistor equipment, and were quickly supplanted by dynamic microphones for a time, and later small electret condenser devices. The high impedance of the crystal microphone made it very susceptible to handling noise, both from the microphone itself and from the connecting cable.

Piezoelectric transducers are often used as contact microphones to amplify sound from acoustic musical instruments, to sense drum hits, for triggering electronic samples, and to record sound in challenging environments, such as underwater under high pressure. Saddle-mounted pickups on acoustic guitars are generally piezoelectric devices that contact the strings passing over the saddle. This type of microphone is different from magnetic coil pickups commonly visible on typical electric guitars, which use magnetic induction, rather than mechanical coupling, to pick up vibration.

Fiber optic microphone



The Optoacoustics 1140 fiber optic microphone

A fiber optic microphone converts acoustic waves into electrical signals by sensing changes in light intensity, instead of sensing changes in capacitance or magnetic fields as with conventional microphones.

During operation, light from a laser source travels through an optical fiber to illuminate the surface of a tiny, sound-sensitive reflective diaphragm. Sound causes the diaphragm to vibrate, thereby minutely changing the intensity of the light it reflects. The modulated light is then transmitted over a second optical fiber to a photo detector, which transforms the intensity-modulated light into analog or digital audio for transmission or recording.

Fiber optic microphones possess high dynamic and frequency range, similar to the best high fidelity conventional microphones.

Fiber optic microphones do not react to or influence any electrical, magnetic, electrostatic or radioactive fields (this is called EMI/RFI immunity). The fiber optic microphone design is therefore ideal for use in areas where conventional microphones are ineffective or dangerous, such as inside industrial turbines or in magnetic resonance imaging (MRI) equipment environments.

Fiber optic microphones are robust, resistant to environmental changes in heat and moisture, and can be produced for any directionality or impedance matching. The distance between the microphone's light source and its photo detector may be up to several kilometers without need for any preamplifier and/or other electrical device, making fiber optic microphones suitable for industrial and surveillance acoustic monitoring.

Fiber optic microphones are used in very specific application areas such as for infrasound monitoring and noise-canceling. They have proven especially useful in medical applications, such as allowing radiologists, staff and patients within the powerful and noisy magnetic field to converse normally, inside the MRI suites as well as in remote control rooms.) Other uses include industrial equipment monitoring and sensing, audio calibration and measurement, high-fidelity recording and law enforcement.

Laser microphone

Laser microphones are often portrayed in movies as spy gadgets. A laser beam is aimed at the surface of a window or other plane surface that is affected by sound. The slight vibrations of this surface displace the returned beam, causing it to trace the sound wave. The vibrating laser spot is then converted back to sound. In a more robust and expensive implementation, the returned light is split and fed to an interferometer, which detects movement of the surface. The former implementation is a tabletop experiment; the latter requires an extremely stable laser and precise optics.

A new type of laser microphone is a device that uses a laser beam and smoke or vapor to detect sound vibrations in free air. On 25 August 2009, U.S. patent 7,580,533 issued for a Particulate Flow Detection Microphone based on a laser-photocell pair with a moving stream of smoke or vapor in the laser beam's path. Sound pressure waves cause disturbances in the smoke that in turn cause variations in the amount of laser light reaching the photo detector. A prototype of the device was demonstrated at the 127th Audio Engineering Society convention in New York City from 9 through 12 October 2009.

Liquid microphone

Early microphones did not produce intelligible speech, until Alexander Graham Bell made improvements including a variable resistance microphone/transmitter. Bell's liquid transmitter consisted of a metal cup filled with water with a small amount of sulfuric acid added. A sound wave caused the diaphragm to move, forcing a needle to move up and down in the water. The electrical resistance between the wire and the cup was then inversely proportional to the size of the water meniscus around the submerged needle. Elisha Gray filed a caveat for a version using a brass rod instead of the needle. Other minor variations and improvements were made to the liquid microphone by Majoranna, Chambers, Vanni, Sykes, and Elisha Gray, and one version was patented by Reginald Fessenden in 1903. These were the first working microphones, but they were not practical for commercial application. The famous first phone conversation between Bell and Watson took place using a liquid microphone.

MEMS microphone

The MEMS (MicroElectrical-Mechanical System) microphone is also called a microphone chip or silicon microphone. The pressure-sensitive diaphragm is etched directly into a silicon chip by MEMS techniques, and is usually accompanied with integrated preamplifier. Most MEMS microphones are variants of the condenser microphone design. Often MEMS microphones have built in analog-to-digital converter (ADC) circuits on the same CMOS chip making the chip a digital microphone and so more readily integrated with modern digital products. Major manufacturers producing MEMS silicon microphones are Wolfson Microelectronics (WM7xxx), Analog Devices, Akustica (AKU200x), Infineon (SMM310 product), Knowles Electronics, Memstech (MSMx), NXP Semiconductors, Sonion MEMS, AAC Acoustic Technologies, and Omron.

Speakers as microphones

A loudspeaker, a transducer that turns an electrical signal into sound waves, is the functional opposite of a microphone. Since a conventional speaker is constructed much like a dynamic microphone (with a diaphragm, coil and magnet), speakers can actually work "in reverse" as microphones. The result, though, is a microphone with poor quality, limited frequency response (particularly at the high end), and poor sensitivity. In practical use, speakers are sometimes used as microphones in applications where high quality and sensitivity are not needed such as intercoms, walkie-talkies or Video game voice chat peripherals, or when conventional microphones are in short supply.

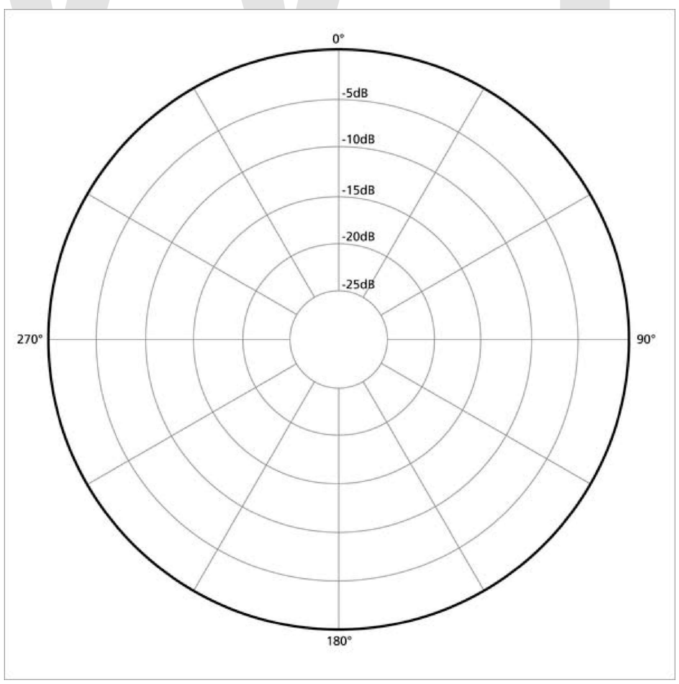
However, there is at least one other practical application of this principle: Using a medium-size woofer placed closely in front of a "kick" (bass drum) in a drum set to act as a microphone. The use of relatively large speakers to transduce low frequency sound sources, especially in music production, is becoming fairly common. A product example of this type of device is the Yamaha Subkick, a 6.5-inch (170 mm) woofer shock-mounted it into a 10" drum shell used in front of kick drums. Since a relatively massive membrane is unable to transduce high frequencies, placing a speaker in front of a kick drum is often ideal for reducing cymbal and snare bleed into the kick drum sound. Less commonly, microphones themselves can be used as speakers, almost always as tweeters. Microphones, however, are not designed to handle the power that speaker components are routinely required to cope with. One instance of such an application was the STC microphone-derived 4001 super-tweeter, which was successfully used in a number of high quality loudspeaker systems from the late 1960s to the mid-70s.

Capsule design and directivity

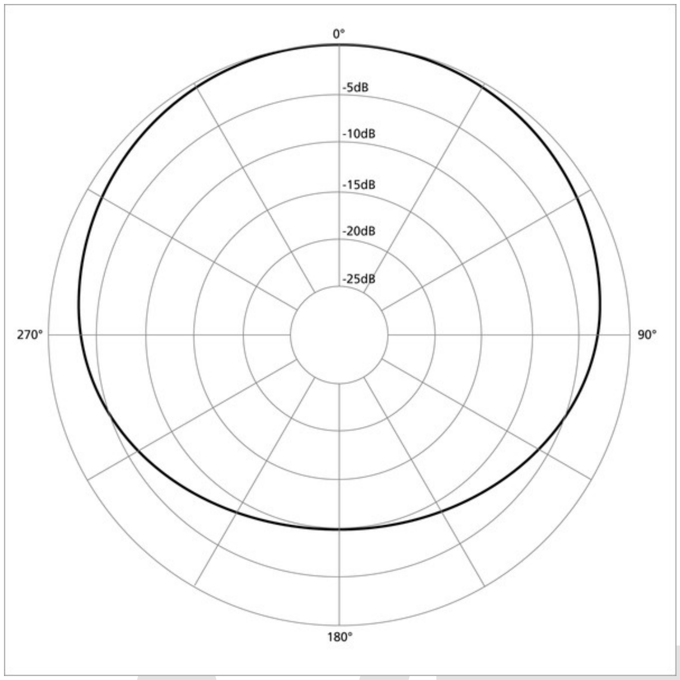
The inner elements of a microphone are the primary source of differences in directivity. A pressure microphone uses a diaphragm between a fixed internal volume of air and the environment, and responds uniformly to pressure from all directions, so it is said to be omnidirectional. A pressure-gradient microphone uses a diaphragm that is at least partially open on both sides. The pressure difference between the two sides produces its directional characteristics. Other elements such as the external shape of the microphone and external devices such as interference tubes can also alter a microphone's directional response. A pure pressure-gradient microphone is equally sensitive to sounds arriving from front or back, but insensitive to sounds arriving from the side because sound arriving at the front and back at the same time creates no gradient between the two. The characteristic directional pattern of a pure pressure-gradient microphone is like a figure-8. Other polar patterns are derived by creating a capsule that combines these two effects in different ways. The cardioid, for instance, features a partially closed backside, so its response is a combination of pressure and pressure-gradient characteristics.

Microphone polar patterns

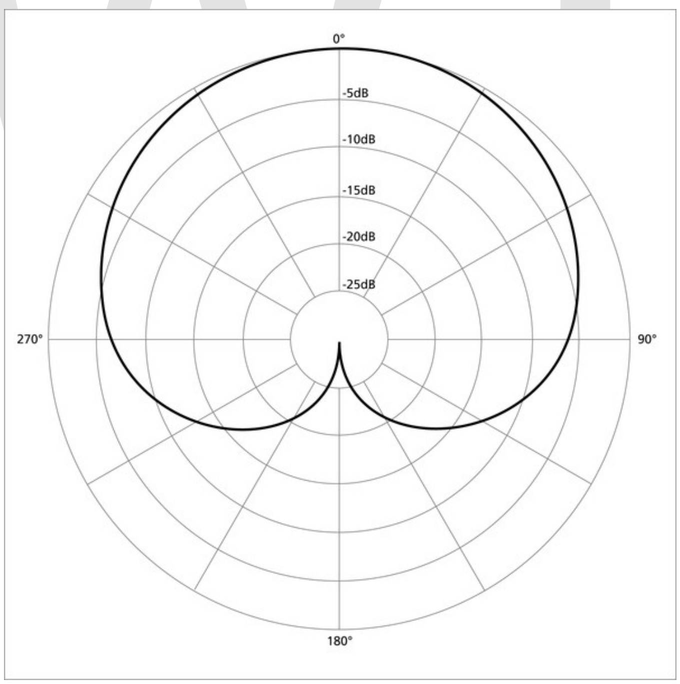
(Microphone facing top of page in diagram, parallel to page):



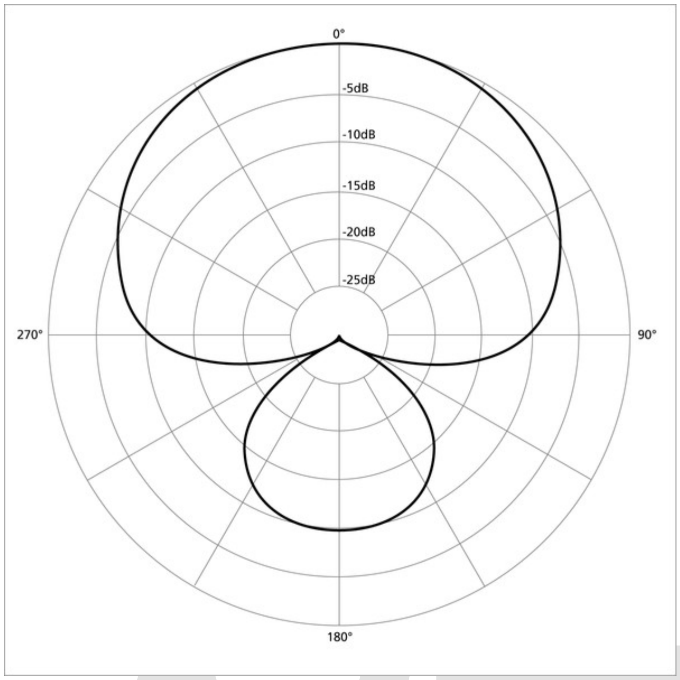
Omnidirectional



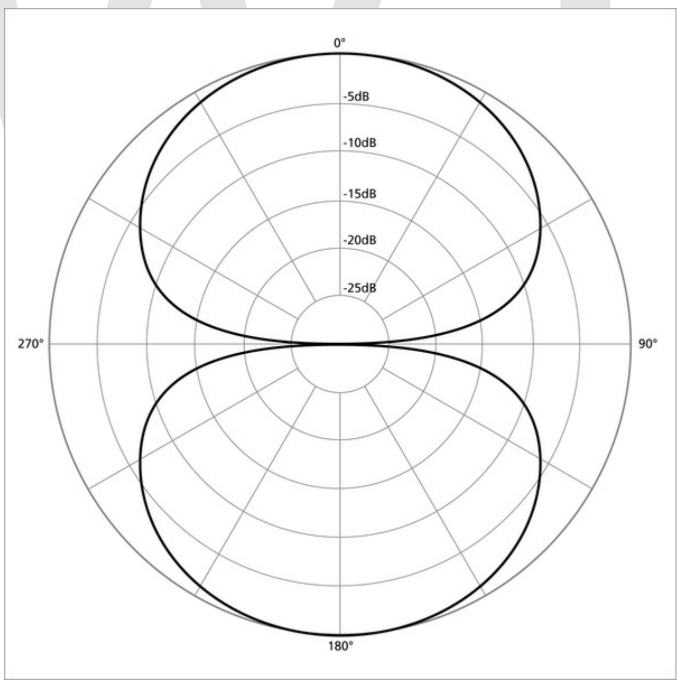
Subcardioid



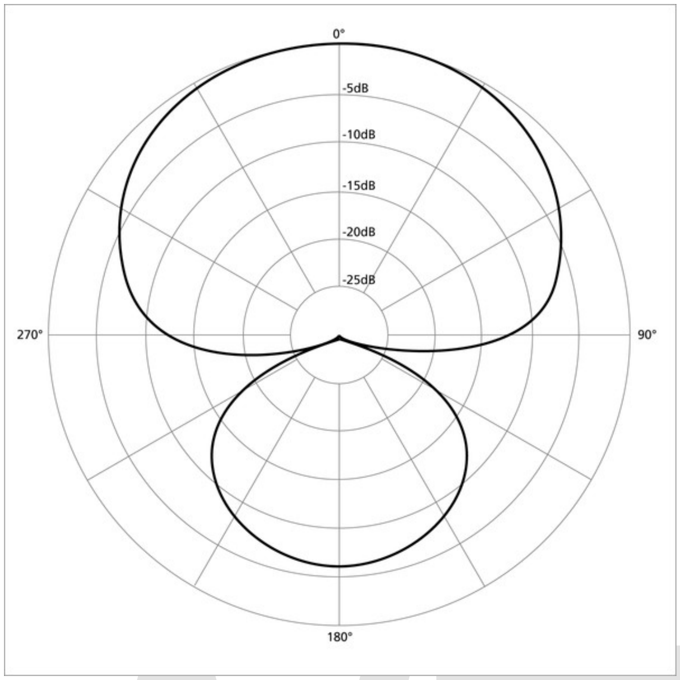
Cardioid



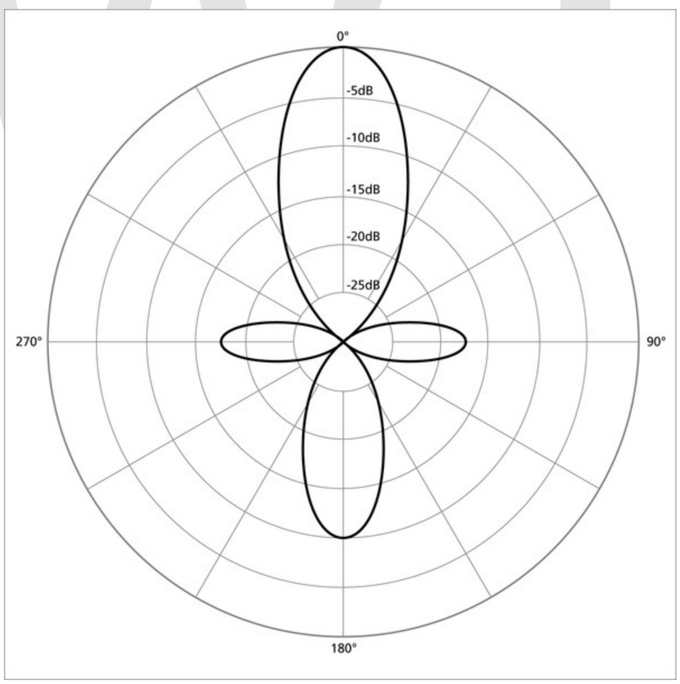
Supercardioid



Bi-directional or Figure of 8



Hypercardioid



Shotgun

A microphone's directionality or polar pattern indicates how sensitive it is to sounds arriving at different angles about its central axis. The polar patterns illustrated above represent the locus of points that produce the same signal level output in the microphone if a given sound pressure level (SPL) is generated from that point. How the physical body of the microphone is oriented relative to the diagrams depends on the microphone design. For large-membrane microphones such as in the Oktava (pictured above), the upward direction in the polar diagram is usually perpendicular to the microphone body, commonly known as "side fire" or "side address". For small diaphragm microphones such as the Shure (also pictured above), it usually extends from the axis of the microphone commonly known as "end fire" or "top/end address".

Some microphone designs combine several principles in creating the desired polar pattern. This ranges from shielding (meaning diffraction/dissipation/absorption) by the housing itself to electronically combining dual membranes.

Omnidirectional

An omnidirectional (or nondirectional) microphone's response is generally considered to be a perfect sphere in three dimensions. In the real world, this is not the case. As with directional microphones, the polar pattern for an "omnidirectional" microphone is a function of frequency. The body of the microphone is not infinitely small and, as a consequence, it tends to get in its own way with respect to sounds arriving from the rear, causing a slight flattening of the polar response. This flattening increases as the diameter of the microphone (assuming it's cylindrical) reaches the wavelength of the frequency in question. Therefore, the smallest diameter microphone gives the best omnidirectional characteristics at high frequencies.

The wavelength of sound at 10 kHz is little over an inch (3.4 cm) so the smallest measuring microphones are often 1/4" (6 mm) in diameter, which practically eliminates directionality even up to the highest frequencies. Omnidirectional microphones, unlike cardioids, do not employ resonant cavities as delays, and so can be considered the "purest" microphones in terms of low coloration; they add very little to the original sound. Being pressure-sensitive they can also have a very flat low-frequency response down to 20 Hz or below. Pressure-sensitive microphones also respond much less to wind noise and plosives than directional (velocity sensitive) microphones.

An example of a nondirectional microphone is the round black *eight ball*.

Unidirectional

A unidirectional microphone is sensitive to sounds from only one direction. The diagram above illustrates a number of these patterns. The microphone faces upwards in each diagram. The sound intensity for a particular frequency is plotted for angles radially from 0 to 360°. (Professional diagrams show these scales and include multiple plots at different frequencies. The diagrams given here provide only an overview of typical pattern shapes, and their names.)

Cardioid

US664A University Sound Dynamic Supercardioid Microphone

The most common unidirectional microphone is a cardioid microphone, so named because the sensitivity pattern is heart-shaped. A hyper-cardioid microphone is similar but with a tighter area of front sensitivity and a smaller lobe of rear sensitivity. A super-cardioid microphone is similar to a hyper-cardioid, except there is more front pickup and less rear pickup. These three patterns are commonly used as vocal or speech microphones, since they are good at rejecting sounds from other directions.

A cardioid microphone is effectively a superposition of an omnidirectional and a figure-8 microphone; for sound waves coming from the back, the negative signal from the figure-8 cancels the positive signal from the omnidirectional element, whereas for sound waves coming from the front, the two add to each other. A hypercardioid microphone is similar, but with a slightly larger figure-8 contribution. Since pressure gradient transducer microphones are directional, putting them very close to the sound source (at distances of a few centimeters) results in a bass boost. This is known as the proximity effect.

Bi-directional

"Figure 8" or bi-directional microphones receive sound equally from both the front and back of the element. Most ribbon microphones are of this pattern. In principle they do not respond to sound pressure at all, only to the gradient between front and back; since sound arriving from the side reaches front and back equally there is no gradient and therefore no sensitivity to sound from that direction. While omnidirectional microphones are scalar transducers responding to pressure from any direction, bi-directional microphones are vector transducers responding to the gradient along an axis normal to the plane of the diaphragm. As a result, output polarity is inverted for sounds arriving from the back side.

Shotgun

An Audio-Technica shotgun microphone

Shotgun microphones are the most highly directional. They have small lobes of sensitivity to the left, right, and rear but are significantly less sensitive to the side and rear than other directional microphones. This results from placing the element at the end of a tube with slots cut along the side; wave cancellation eliminates much of the off-axis sound. Due to the narrowness of their sensitivity area, shotgun microphones are commonly used on television and film sets, in stadiums, and for field recording of wildlife.

Boundary or "PZM"

Several approaches have been developed for effectively using a microphone in less-than-ideal acoustic spaces, which often suffer from excessive reflections from one or more of the surfaces (boundaries) that make up the space. If the microphone is placed in, or very close to, one of these boundaries, the reflections from that surface are not sensed by the microphone. Initially this was done by placing an ordinary microphone adjacent to the surface, sometimes in a block of acoustically transparent foam. Sound engineers Ed Long and Ron Wickersham developed the concept of placing the diaphgram parallel to and facing the boundary. While the patent has expired, "Pressure Zone Microphone" and "PZM" are still active trademarks of Crown International, and the generic term "boundary microphone" is preferred. While a boundary microphone was initially implemented using an omnidirectional element, it is also possible to mount a directional microphone close enough to the surface to gain some of the benefits of this technique while retaining the directional properties of the element. Crown's trademark on this approach is "Phase Coherent Cardioid" or "PCC," but there are other makers who employ this technique as well.

Application-specific designs

A lavalier microphone is made for hands-free operation. These small microphones are worn on the body. Originally, they were held in place with a lanyard worn around the neck, but more often they are fastened to clothing with a clip, pin, tape or magnet. The lavalier cord may be hidden by clothes and either run to an RF transmitter in a pocket or clipped to a belt (for mobile use), or run directly to the mixer (for stationary applications).

A wireless microphone transmits the audio as a radio or optical signal rather than via a cable. It usually sends its signal using a small FM radio transmitter to a nearby receiver connected to the sound system, but it can also use infrared waves if the transmitter and receiver are within sight of each other.

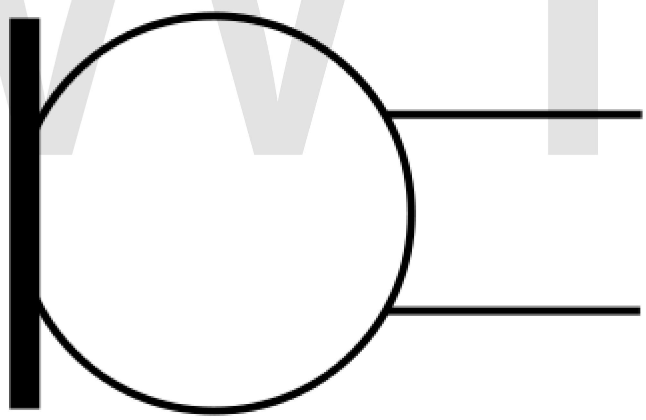
A contact microphone picks up vibrations directly from a solid surface or object, as opposed to sound vibrations carried through air. One use for this is to detect sounds of a very low level, such as those from small objects or insects. The microphone commonly consists of a magnetic (moving coil) transducer, contact plate and contact pin. The contact plate is placed directly on the vibrating part of a musical instrument or other surface, and the contact pin transfers vibrations to the coil. Contact microphones have been used to pick up the sound of a snail's heartbeat and the footsteps of ants. A portable version of this microphone has recently been developed. A throat microphone is a variant of the contact microphone that picks up speech directly from a person's throat, which it is strapped to. This lets the device be used in areas with ambient sounds that would otherwise make the speaker inaudible.

A parabolic microphone uses a parabolic reflector to collect and focus sound waves onto a microphone receiver, in much the same way that a parabolic antenna (e.g. satellite dish) does with radio waves. Typical uses of this microphone, which has unusually focused front sensitivity and can pick up sounds from many meters away, include nature recording, outdoor sporting events, eavesdropping, law enforcement, and even espionage. Parabolic microphones are not typically used for standard recording applications, because they tend to have poor low-frequency response as a side effect of their design.

A stereo microphone integrates two microphones in one unit to produce a stereophonic signal. A stereo microphone is often used for broadcast applications or field recording where it would be impractical to configure two separate condenser microphones in a classic X-Y configuration for stereophonic recording. Some such microphones have an adjustable angle of coverage between the two channels.

A noise-canceling microphone is a highly directional design intended for noisy environments. One such use is in aircraft cockpits where they are normally installed as boom microphones on headsets. Another use is on loud concert stages for vocalists. Many noise-canceling microphones combine signals received from two diaphragms that are in opposite electrical polarity or are processed electronically. In dual diaphragm designs, the main diaphragm is mounted closest to the intended source and the second is positioned farther away from the source so that it can pick up environmental sounds to be subtracted from the main diaphragm's signal. After the two signals have been combined, sounds other than the intended source are greatly reduced, substantially increasing intelligibility. Other noise-canceling designs use one diaphragm that is affected by ports open to the sides and rear of the microphone, with the sum being a 16 dB rejection of sounds that are farther away. One noise-canceling headset design using a single diaphragm has been used prominently by vocal artists such as Garth Brooks and Janet Jackson. A few noise-canceling microphones are throat microphones.

Connectors



Electronic symbol for a microphone

The most common connectors used by microphones are:

- Male XLR connector on professional microphones
- ¼ inch (sometimes referred to as 6.3 mm) jack plug also known as 1/4 inch TRS connector on less expensive consumer microphones. Many consumer microphones use an unbalanced 1/4 inch phone jack. Harmonica microphones

commonly use a high impedance 1/4 inch TS connection to be run through guitar amplifiers.
- 3.5 mm (sometimes referred to as 1/8 inch mini) stereo (wired as mono) mini phone plug on very inexpensive and computer microphones

Some microphones use other connectors, such as a 5-pin XLR, or mini XLR for connection to portable equipment. Some lavalier (or 'lapel', from the days of attaching the microphone to the news reporters suit lapel) microphones use a proprietary connector for connection to a wireless transmitter. Since 2005, professional-quality microphones with USB connections have begun to appear, designed for direct recording into computer-based software.

Impedance-matching

Microphones have an electrical characteristic called impedance, measured in ohms (Ω), that depends on the design. Typically, the *rated impedance* is stated. Low impedance is considered under 600 Ω. Medium impedance is considered between 600 Ω and 10 kΩ. High impedance is above 10 kΩ. Owing to their built-in amplifier, condenser microphones typically have an output impedance between 50 and 200 Ω.

The output of a given microphone delivers the same power whether it is low or high impedance. If a microphone is made in high and low impedance versions, the high impedance version has a higher output voltage for a given sound pressure input, and is suitable for use with vacuum-tube guitar amplifiers, for instance, which have a high input impedance and require a relatively high signal input voltage to overcome the tubes' inherent noise. Most professional microphones are low impedance, about 200 Ω or lower. Professional vacuum-tube sound equipment incorporates a transformer that steps up the impedance of the microphone circuit to the high impedance and voltage needed to drive the input tube; the impedance conversion inherently creates voltage gain as well. External matching transformers are also available that can be used in-line between a low impedance microphone and a high impedance input.

Low-impedance microphones are preferred over high impedance for two reasons: one is that using a high-impedance microphone with a long cable results in high frequency signal loss due to cable capacitance, which forms a low-pass filter with the microphone output impedance. The other is that long high-impedance cables tend to pick up more hum (and possibly radio-frequency interference (RFI) as well). Nothing is damaged if the impedance between microphone and other equipment is mismatched; the worst that happens is a reduction in signal or change in frequency response.

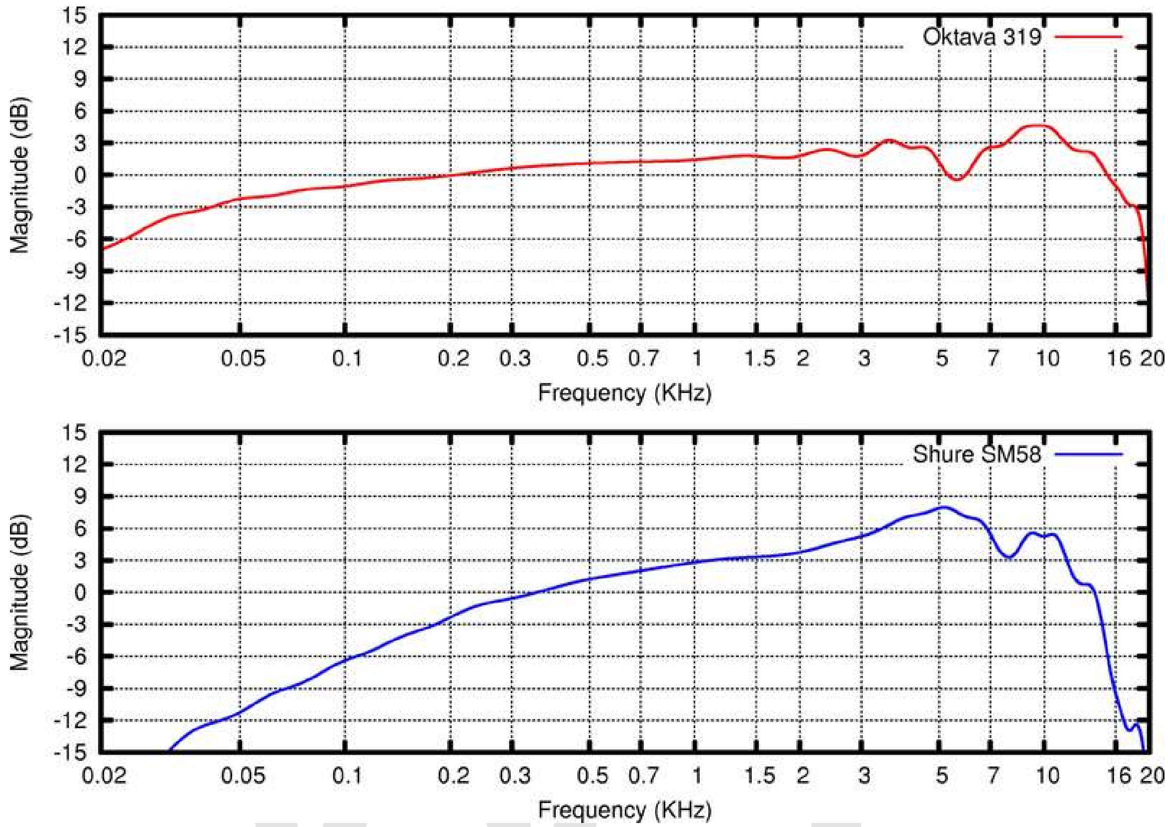
Most microphones are designed *not* to have their impedance matched by the load they are connected to. Doing so can alter their frequency response and cause distortion, especially at high sound pressure levels. Certain ribbon and dynamic microphones are exceptions, due to the designers' assumption of a certain load impedance being part of the internal electro-acoustical damping circuit of the microphone.

Digital microphone interface

Neumann D-01 digital microphone and Neumann DMI-8 8-channel USB Digital Microphone Interface

The AES 42 standard, published by the Audio Engineering Society, defines a digital interface for microphones. Microphones conforming to this standard directly output a digital audio stream through an XLR male connector, rather than producing an analog output. Digital microphones may be used either with new equipment with appropriate input connections that conform to the AES 42 standard, or else via a suitable interface box. Studio-quality microphones that operate in accordance with the AES 42 standard are now available from a number of microphone manufacturers.

Measurements and specifications



A comparison of the far field on-axis frequency response of the Oktava 319 and the Shure SM58

Because of differences in their construction, microphones have their own characteristic responses to sound. This difference in response produces non-uniform phase and frequency responses. In addition, microphones are not uniformly sensitive to sound pressure, and can accept differing levels without distorting. Although for scientific applications microphones with a more uniform response are desirable, this is often not the case for music recording, as the non-uniform response of a microphone can produce a desirable coloration of the sound. There is an international standard for microphone specifications, but few manufacturers adhere to it. As a result, comparison of published data from different manufacturers is difficult because different measurement techniques are used. The Microphone Data Website has collated the technical specifications complete with pictures, response curves and technical data from the microphone manufacturers for every currently listed microphone, and even a few obsolete models, and shows the data for them all in one common format for ease of comparison.. Caution should be used in drawing any solid conclusions from this or any other published data, however, unless it is known that the manufacturer has supplied specifications in accordance with IEC 60268-4.

A frequency response diagram plots the microphone sensitivity in decibels over a range of frequencies (typically at least 0–20 kHz), generally for perfectly on-axis sound (sound arriving at 0° to the capsule). Frequency response may be less informatively stated textually like so: "30 Hz–16 kHz ±3 dB". This is interpreted as meaning a nearly flat, linear, plot between the stated frequencies, with variations in amplitude of no more than plus or minus 3 dB. However, one cannot determine from this information how *smooth* the variations are, nor in what parts of the spectrum they occur. Note that commonly made statements such as "20 Hz–20 kHz" are meaningless without a decibel measure of tolerance. Directional microphones' frequency response varies greatly with distance from the sound source, and with the geometry of the sound source. IEC 60268-4 specifies that frequency response should be measured in *plane progressive wave* conditions (very far away from the source) but this is seldom practical. *Close talking* microphones may be measured with different sound sources and distances, but there is no standard and therefore no way to compare data from different models unless the measurement technique is described.

The self-noise or equivalent noise level is the sound level that creates the same output voltage as the microphone does in the absence of sound. This represents the lowest point of the microphone's dynamic range, and is particularly important should you wish to record sounds that are quiet. The measure is often stated in dB(A), which is the equivalent loudness of the noise on a decibel scale frequency-weighted for how the ear hears, for example: "15 dBA SPL" (SPL means sound pressure level relative to 20 micropascals). The lower the number the better. Some microphone manufacturers state the noise level using ITU-R 468 noise weighting, which more accurately represents the way we hear noise, but gives a figure some 11–14 dB higher. A quiet microphone typically measures 20 dBA SPL or 32 dB SPL 468-weighted. Very quiet microphones have existed for years for special applications, such the Brüel & Kjaer 4179, with a noise level around 0 dB SPL. Recently some microphones with low noise specifications have been introduced in the studio/entertainment market, such as models from Neumann and Røde that advertise noise levels between 5–7 dBA. Typically this is achieved by altering the frequency response of the capsule and electronics to result in lower noise within the A-weighting curve while broadband noise may be increased.

The maximum SPL the microphone can accept is measured for particular values of total harmonic distortion (THD), typically 0.5%. This amount of distortion is generally inaudible, so one can safely use the microphone at this SPL without harming the recording. Example: "142 dB SPL peak (at 0.5% THD)". The higher the value, the better, although microphones with a very high maximum SPL also have a higher self-noise.

The clipping level is perhaps a better indicator of maximum usable level, as the 1% THD figure usually quoted under max SPL is really a very mild level of distortion, quite inaudible especially on brief high peaks. Harmonic distortion from microphones is usually of low-order (mostly third harmonic) type, and hence not very audible even at 3–5%. Clipping, on the other hand produces a harsh sound on peaks, and should be avoided if at all possible. For some microphones the clipping level may be much higher than the max SPL.

The dynamic range of a microphone is the difference in SPL between the noise floor and the maximum SPL. If stated on its own, for example "120 dB", it conveys significantly less information than having the self-noise and maximum SPL figures individually.

Sensitivity indicates how well the microphone converts acoustic pressure to output voltage. A high sensitivity microphone creates more voltage and so needs less amplification at the mixer or recording device. This is a practical concern but is not directly an indication of the mic's quality, and in fact the term sensitivity is something of a misnomer, 'transduction gain' being perhaps more meaningful, (or just "output level") because true sensitivity is generally set by the noise floor, and too much "sensitivity" in terms of output level compromises the clipping level. There are two common measures. The (preferred) international standard is made in millivolts per pascal at 1 kHz. A higher value indicates greater sensitivity. The older American method is referred to a 1 V/Pa standard and measured in plain decibels, resulting in a negative value. Again, a higher value indicates greater sensitivity, so −60 dB is more sensitive than −70 dB.

Measurement microphones

Some microphones are intended for testing speakers, measuring noise levels and otherwise quantifying an acoustic experience. These are calibrated transducers and are usually supplied with a calibration certificate that states absolute sensitivity against frequency. The quality of measurement microphones is often referred to using the designations "Class 1," "Type 2" etc., which are references not to microphone specifications but to sound level meters. A more comprehensive standard for the description of measurement microphone performance was recently adopted.

Measurement microphones are generally scalar sensors of pressure; they exhibit an omnidirectional response, limited only by the scattering profile of their physical dimensions. Sound intensity or sound power measurements require pressure-gradient measurements, which are typically made using arrays of at least two microphones, or with hot-wire anemometers.

Microphone calibration

To take a scientific measurement with a microphone, its precise sensitivity must be known (in volts per pascal). Since this may change over the lifetime of the device, it is necessary to regularly calibrate measurement microphones. This service is offered by some microphone manufacturers and by independent certified testing labs. All microphone calibration is ultimately traceable to primary standards at a national measurement institute such as NPL in the UK, PTB in Germany and NIST in the USA, which most commonly calibrate using the reciprocity primary standard. Measurement microphones calibrated using this method can then be used to calibrate other microphones using comparison calibration techniques.

Depending on the application, measurement microphones must be tested periodically (every year or several months, typically) and after any potentially damaging event, such as being dropped (most such mikes come in foam-padded cases to reduce this risk) or exposed to sounds beyond the acceptable level.

Microphone array and array microphones

A microphone array is any number of microphones operating in tandem. There are many applications:

- Systems for extracting voice input from ambient noise (notably telephones, speech recognition systems, hearing aids)
- Surround sound and related technologies
- Locating objects by sound: acoustic source localization, e.g. military use to locate the source(s) of artillery fire. Aircraft location and tracking.
- High fidelity original recordings
- 3D spatial beamforming for localized acoustic detection of subcutaneous sounds

Typically, an array is made up of omnidirectional microphones distributed about the perimeter of a space, linked to a computer that records and interprets the results into a coherent form.

Microphone windscreens

Windscreens are used to protect microphones that would otherwise be buffeted by wind or vocal plosives from consonants such as "P", "B", etc. Most microphones have an integral windscreen built around the microphone diaphragm. A screen of plastic, wire mesh or a metal cage is held at a distance from the microphone diaphragm, to shield it. This cage provides a first line of defense against the mechanical impact of objects or wind. Some microphones, such as the Shure SM58, may have an additional layer of foam inside the cage to further enhance the protective properties of the shield. One disadvantage of all windscreen types is that the microphone's high frequency response is attenuated by a small amount, depending on the density of the protective layer.

Beyond integral microphone windscreens, there are three broad classes of additional wind protection.

Microphone covers

Various microphone covers

Microphone covers are often made of soft open-cell polyester or polyurethane foam because of the inexpensive, disposable nature of the foam. Optional windscreens are often available from the manufacturer and third parties. A visible example of an optional accessory windscreen is the A2WS from Shure, one of which is fitted over each of the two Shure SM57 microphones used on the United States president's lectern. One disadvantage of polyurethane foam microphone covers is that they can deteriorate over time. Windscreens also tend to collect dirt and moisture in their open cells and must be cleaned to prevent high frequency loss, bad odor and unhealthy conditions for the person using the microphone. On the other hand, a major advantage of concert vocalist windscreens is that one can quickly change to a clean windscreen between users, reducing the chance of transferring germs. Windscreens of various colors can be used to distinguish one microphone from another on a busy, active stage.

Pop filters

Pop filters or pop screens are used in controlled studio environments to minimize plosives when recording. A typical pop filter is composed of one or more layers of acoustically transparent gauze-like material, such as woven nylon (e.g. pantyhose) stretched over a circular frame and a clamp and a flexible mounting bracket to attach to the microphone stand. The pop shield is placed between the vocalist and the microphone. The closer a vocalist brings his or her lips to the microphone, the greater the requirement for a Pop filter. Singers can be trained either to soften their plosives or direct the air blast away from the microphone, in which cases they don't need a pop filter.

Pop filters also keep spittle off the microphone. Most condenser microphones can be damaged by spittle.

Blimps

Two recordings being made—A *blimp* is being used on the left. An open-cell foam windscreen is being used on the right.

a 'dead cat' and a 'dead kitten' windscreens. The dead kitten covers a stereo mic for a DSLR camera. The difference in name is due to the size of the fur.

Blimps (also known as Zeppelins) are large, hollow windscreens used to surround microphones for outdoor location audio, such as nature recording, electronic news gathering, and for film and video shoots. They can cut wind noise by as much as 25 dB, especially low-frequency noise. The blimp is essentially a hollow cage or basket with acoustically transparent material stretched over the outer frame. The blimp works by creating a volume of still air around the microphone. The microphone is often further isolated from the blimp by an elastic suspension inside the basket. This reduces wind vibrations and handling noise transmitted from the cage. To extend the range of wind speed conditions in which the blimp remains effective, many have the option of a secondary cover over the outer shell. This is usually an acoustically transparent, synthetic fur material with long, soft hairs. Common and slang names for this include "dead cat" or "windmuff". The hairs deaden the noise caused by the shock of wind hitting the blimp. A synthetic fur cover can reduce wind noise by an additional 10 dB.

Chapter-9

Wireless Microphone

A **wireless microphone**, as the name implies, is a microphone without a physical cable connecting it directly to the sound recording or amplifying equipment with which it is associated.

More commonly known as a Radio Microphone, there are many different standards, frequencies and transmission technologies used to replace the microphone's cable connection and make it into a wireless microphone. They can transmit, for example, in radio waves using UHF or VHF frequencies, FM, AM, or various digital modulation schemes. Some low cost models use infrared light. Infrared microphones require a direct line of sight between the microphone and the receiver, while costlier radio frequency models do not.

Some models operate on a single fixed frequency, but the more advanced models operate on a user selectable frequency to avoid interference, and allow the use of several microphones at the same time.

Advantages and disadvantages

Wireless microphones awaiting pickup by performers in a musical.

The **advantages** are:

- Greater freedom of movement for the artist or speaker.
- Avoidance of cabling problems common with wired microphones, caused by constant moving and stressing the cables.
- Reduction of cable "trip hazards" in the performance space

The **disadvantages** are:

- Sometimes limited range (a wired balanced XLR microphone can run up to 300 ft or 100 meters). Some wireless systems have a shorter range, while more expensive models can exceed that distance.
- Possible interference with or, more often, from other radio equipment or other radio microphones, though models with many frequency-synthesized switch-selectable channels are now plentiful and cost effective.
- Operation time is limited relative to battery life; it is shorter than a normal condenser microphone due to greater drain on batteries from transmitting circuitry, and from circuitry giving extra features, if present.
- Noise or dead spots (places where it doesn't work, especially in non-diversity systems)
- Limited number of operating microphones at the same time and place, due to the limited number of radio channels (frequencies).

Techniques

The professional models transmit in VHF or UHF radio frequency and have 'true' diversity reception (two separate receiver modules each with its own antenna), which eliminates dead spots (caused by phase cancellation) and the effects caused by the reflection of the radio waves on walls and surfaces in general.

Another technique used to improve the sound quality (actually, to improve the dynamic range), is companding. Nady Systems, Inc was the first to offer this technology in wireless microphones in 1976, which was based on the patent obtained by company founder John Nady.

Some models have adjustable gain on the microphone itself, to be able to accommodate different level sources, such as loud instruments or quiet voices. Adjustable gain helps to avoid clipping.

Some models have adjustable squelch, which silences the output when the receiver does not get a strong or quality signal from the microphone, instead of reproducing noise. When squelch is adjusted, the threshold of the signal quality or level is adjusted.

Products

The original manufacturer of the wireless microphone was Vega Electronics Corporation. AKG Acoustics, Audio Ltd, Audio-Technica, Electro-Voice, Lectrosonics, MIPRO, Nady Systems, Inc, Samson Technologies, Sennheiser, Shure, Sony and Zaxcom are all major manufacturers of wireless microphone systems. They have made significant advances in dealing with many of the disadvantages listed above. For example, while there is a limited band in which the microphones may operate, several high-end systems can consist of over 100 different microphones operating simultaneously. However, the ability to have more microphones operating at the same time increases the cost due to component specifications, design and construction. That is one reason for such large price differences between different series of wireless systems.

Generally there are three wireless microphone types: **handheld**, **plug-in** and **bodypack**:

- **Handheld** looks like a 'normal' wired microphone, but may have a bigger body to accommodate the transmitter and battery pack.
- **Plug-in**, **plug-on**, **slot-in**, or cube-style transmitters attach to the bottom of a standard microphone, thus converting it to wireless operation.
- **Bodypack** is a small box housing the transmitter and battery pack, but not the microphone itself. It is attachable to belt or elsewhere and has a wire going to headset, lavalier microphone or a guitar.

Several manufacturers including Sennheiser, AKG, Nady Systems, Lectrosonics and Zaxcom offer a plug-in transmitter for existing wired microphones, which plugs into the XLR output of the microphone and transmits to the manufacturer's standard receiver. This offers many of the benefits of an integrated system, and also allows microphone types (of which there may be no wireless equivalent) to be used without a cable. For example a television, or film, sound production engineer may use a plug-in transmitter to enable wireless transmission of a highly directional rifle (or "shotgun") microphone, removing the safety hazard of a cable connection and permitting the production engineer greater freedom to follow the action. Plug-in transmitters also allow the conversion of vintage microphone types to cordless operation. This is useful where a vintage microphone is needed for visual or other artistic reasons, and the absence of cables allows for rapid scene changes and reducing trip hazards. In some cases these plug-in transmitters can also provide 48 volt phantom power allowing the use of condenser microphone types. DC-DC converter circuitry within the transmitter is used to multiply the battery supply, which may be three volts or less, up to the required 48 volts.

Receivers

Wireless microphone receiver racks backstage at a large televised music awards event

There are many types of receiver. True Diversity receivers have two radio modules and two antennas. Diversity receivers have one radio module and two antennas, although some times the second antenna may not be obviously visible. Non-diversity receivers have only one antenna.

Receivers are commonly housed in a half-rack configuration, so that two can be mounted together in a rack system. For large complex multi channel radio microphone systems, as used in broadcast television studios and musical theatre productions, modular receiver systems with several (commonly six or eight) true diversity receivers slotting into a rack mounted mainframe housing are available. Several mainframes may be used together in a rack to supply the number of receivers required. In some musical theatre productions, systems with forty or more radio microphones are not unusual.

Receivers specifically for use with video cameras are often mounted in a bodypack configuration, typically with a hotshoe mount to be fitted onto the hotshoe of the camcorder. Small true diversity receivers which slot in to a special housing on many professional broadcast standard video cameras are produced by manufacturers including Sennheiser, Lectrosonics and Sony. For less demanding or more budget conscious video applications small non-diversity receivers are common. When used at relatively short operating distances from the transmitter this arrangement gives adequate and reliable performance.

Bandwidth and Spectrum

Almost all wireless microphone systems use wideband FM modulation, requiring approximately 200 kHz of bandwidth. Because of the relatively large bandwidth requirements, wireless microphone use is effectively restricted to VHF and above.

Many older wireless microphone systems operate in the VHF part of the electromagnetic spectrum. Systems operating in this range are often crystal-controlled, and therefore operate on a single frequency. However, if this frequency is chosen properly, the system will be able to operate for years without any problems.

Most modern wireless microphone products operate in the UHF television band, however. In the United States, this band extends from 470 MHz to 698 MHz. In 2010 the Federal Communication Commission issued new regulations on the operations of TV-band devices. Other countries have similar band limits; for example, Great Britain's UHF TV band extends from 470 MHz to 854 MHz. Typically, wireless microphones operate on unused TV channels, with room for one to two microphones per megahertz of spectrum available.

Intermodulation (IM) is a major problem when operating multiple systems in one location. IM occurs when two or more RF signals mix in a non-linear circuit, such as an oscillator or mixer. When this occurs, predictable combinations of these frequencies can occur. For example, the combinations 2A-B, 2B-A, and A+B-C might occur, where A, B, and C are the frequencies in operation. If one of these combinations is close to the operating frequency of another system (or one of the original frequencies A, B, or C), then interference will result on that channel. The solution to this problem is to manually calculate all of the possible products, or use a computer program that does this calculation automatically.

Digital Hybrid Wireless

Digital Hybrid systems use an analog FM transmission scheme in combination with digital signal processing (DSP) to enhance the system's audio. Using DSP allows the use of digital techniques impossible in the analog domain such as predictive algorithms, thus achieving a flatter frequency response in the audio spectrum and also further reduce noise and other undesirable artifacts when compared to pure analog systems.

Another approach is to use DSP in order to emulate analog companding schemes in order to maintain compatibility between older analog systems and newer systems.

Digital

A number of pure digital wireless microphone systems do exist and there are many different digital modulation schemes possible. Digital audio compression is normally used in order to reduce the occupied RF bandwidth. The RF bandwidth that would be required to transmit un-compressed digital audio with sufficient resolution, dynamic range and audio bandwidth for professional audio applications is generally regarded as prohibitive otherwise.

The Zaxcom, Lectrosonics 700, AKG and MIPRO systems, for example, use 200 kHz narrowband UHF broadcast frequencies, the same UHF frequencies used by ananlogue FM systems, for transmission of a digital signal at a fixed bit rate. These systems encode an RF carrier with one or two channels of digital audio. Advantages of such system include low noise, low distortion, the opportunity for encryption, and enhanced transmission reliability.

Pure digital systems take various forms. Some systems use frequency-hopping spread spectrum technology, similar to that used for cordless phones. As this can require more bandwidth than a wideband FM signal, these microphones typically operate in the 900 MHz, 2.4 GHz or 6 GHz unlicensed (also known as de-regulated or licence exempt) bands. The absence of any requirement for a licence in these frequency bands is an added attraction for many users, regardless of the technology used. The 900 MHz band is not an option outside of the USA and Canada as it is used by GSM cellular mobile phone networks in most other parts of the world. The 2.4 GHz band is increasingly congested with various systems including WiFi (also referred to as Wireless LAN, wireless networks, 802.11b/g/n), Bluetooth and 'leakage' from microwave ovens. The 6 GHz band has problems of range (requires line of sight) due to the extremely short transmission carrier wavelengths.

Digital radio microphones are inherently more difficult for the casual 'scanner' listener to intercept because conventional "scanning receivers" are generally only capable of de-modulating conventional analogue modulation schemes such as FM and AM. However, some digital wireless microphone systems additionally use encryption technology in an attempt to prevent more serious 'eavesdropping' which may be of concern for corporate users and those using radio microphones in security sensitive situations.

Manufacturers currently offering digital wireless microphone systems include AKG-Acoustics, Audio-Technica, Lectrosonics, Line 6, MIPRO, Shure, Sony, and Zaxcom.

Licensing

UK

In the UK, use of wireless microphone systems requires a license, except for the license free bands of 173.8–175.0 MHz and 863–865 MHz (N.B. This is emphatically **NOT** TV Channel 69. Channel 69 is from 854–862 MHz. In the UK Channel 69 frequencies **do** require a license from JFMG Ltd.:).

The UK communications regulator, Ofcom, had said that it would auction that part of the UHF spectrum currently reserved for wireless microphones, to which objections have been raised by Andrew Lloyd Webber and many others. Following public consultations this decision was changed and the UHF frequencies used for radio microphones are to be licensed to a commercially based Band Manager with special obligations to "Programme Making and Special Events" (PMSE) users following a "beauty contest" selection process. The future of the UHF spectrum above 798 MHz, including 'Channel 69' has been further thrown in to doubt by moves across Europe to create a new 800 MHz band for mobile broadband applications.

USA

Licenses are required to use wireless microphones on vacant TV channels in the United States as they are a part of the Broadcast Auxiliary Service (BAS). However, this requirement is often overlooked and rarely enforced by the FCC. Licenses are available only to broadcasters, cable networks, television and film producers. However, the FCC has issued a Report and Order stating that they will no longer allow Broadcast Auxiliary Service (BAS) devices to operate in the 698–806 MHz portion of the spectrum due to their auction of the 700 MHz band. This change is unrelated to, but commonly confused with, the White Space device debate that is currently taking place in the U.S.

The same Report and Order, issued January 15, 2010, also permits most wireless microphones in the "core TV band" (TV channels 2 through 51, except 37) to operate with transmit power up to 50 mW without a license, under a special waiver of Part 15 rules. A rule change to make this permanent is proposed.

There are currently some wireless microphone manufacturers that are marketing wireless microphones for use in the United States that operate within the 944–952 MHz band reserved for studio-transmitter link communications. These microphones have the potential to interfere with studio-transmitter links, and their use must be coordinated by the SBE. Licenses in this band are only available to licensees of radio and TV stations, and broadcasters are likely to report unauthorized use in this band due to the high potential for interference.

Australia

In Australia, operation of wireless microphones of up to 100 mW between 520 MHz and 820 MHz is covered by a class license, allowing any user to operate the devices without obtaining an individual license. The onus, however, is on the user of the wireless microphone to resolve any interference that the use of the microphone may cause to licensed radio communications services.

Other countries

In many other countries wireless microphone use does require a license. Some governments regard all radio frequencies as military assets and the use of unlicensed radio transmitters, even wireless microphones, may be severely punished.

White Space Devices (United States)

There is currently a movement to allow the operation of personal unlicensed, wideband digital devices in the UHF television spectrum in the United States. These devices are backed by firms which seek to develop and deploy these devices as quickly as possible. These 'white space' devices (WSDs) will be required to have GPS and access to a location database to avoid interfering with other users of the band. Initial tests performed by the FCC have shown that in some cases, prototypes of these devices are unable to correctly identify frequencies that are in use, and may therefore accidentally transmit on top of these users. Broadcasters, theaters, and wireless microphone manufacturers are firmly against these types of devices ostensibly for this reason.

Later tests by the FCC indicate that the devices can safely be used. This has not reduced the opposition by broadcasters who may also be concerned by the possibility of entertainment delivery competition from high-speed mobile Internet access delivered in the white spaces. A decision on whether and under what rules to allow these devices is on the docket for the November 4, 2008 meeting of the FCC.

Cognitive Access (UK)

A similar class of device to those known in the US as White Space Devices (WSD) is being researched in the UK and probably many other countries. Whilst the WSD situation in the USA is being closely watched by interested parties in the UK and elsewhere, early in 2009 Ofcom launched research and a public consultation on Cognitive Access to the UHF interleaved spectrum . The outcome of this consultation and the related WSD activities in the USA could have far reaching implications for users of UHF radio microphones in the UK and around the world.