Handbook of

Microphones & Loudspeakers

(Audio Electronic Devices)

Kale Sheffield

First Edition, 2012

ISBN 978-81-323-1560-5

Published by:
Learning Press
4735/22 Prakashdeep Bldg,
Ansari Road, Darya Ganj,
Delhi - 110002
Email: info@wtbooks.com

Table of Contents

Chapter- 1

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.

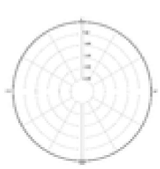
Most microphones today use electromagnetic induction (dynamic microphone), capacitance change (condenser microphone), piezoelectric generation, or light modulation to produce an electrical voltage signal from mechanical vibration.

Capsule design and directivity

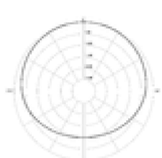
The inner elements of a microphone are the primary source of differences in directivity. A pressure microphone uses a diaphragm between a fixed internal volume of air and the environment, and responds uniformly to pressure from all directions, so it is said to be omnidirectional. A pressure-gradient 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

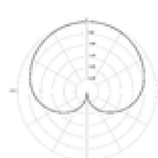
(Microphone facing top of page in diagram, parallel to page):



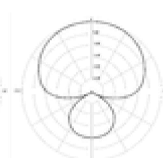
Omnidirectional



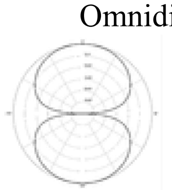
Subcardioid



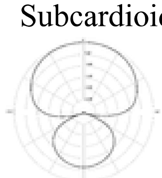
Cardioid



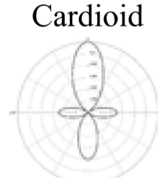
Supercardioid



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 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.)

Cardioids

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 from both the front and back of the element. Most ribbon microphones are of this pattern.

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.

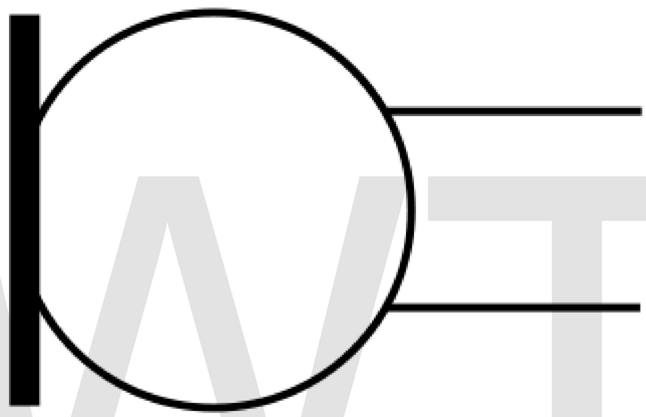
A contact microphone picks up vibrations directly from a solid surface or object, as opposed to sound vibrations carried through air. One use for this is to detect sounds of a very low level, such as those from small objects or insects. The microphone commonly consists of a magnetic (moving coil) transducer, 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 6.5 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. Condenser microphones (after the built-in preamp) typically have an output impedance between 50 and 200 ohms.

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.

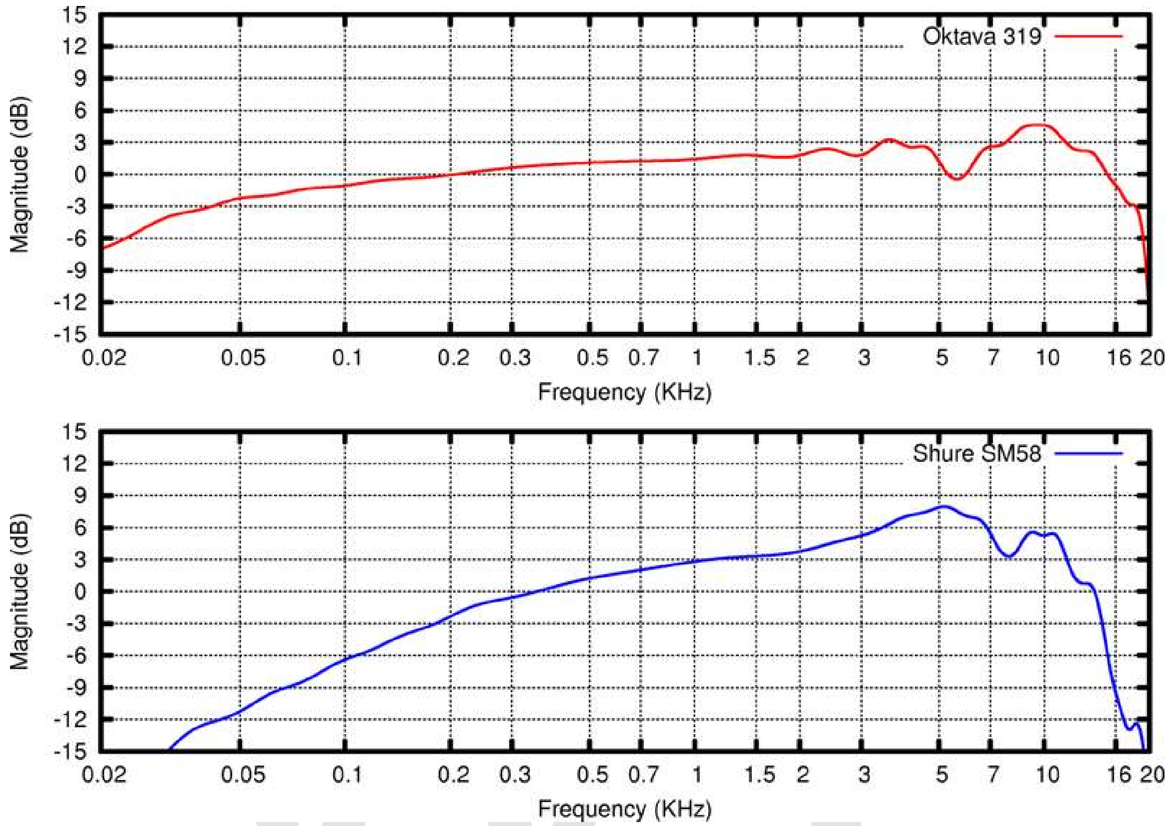
Most microphones are designed *not* to have their impedance matched by the load they are connected to. Doing so can alter their frequency response and cause distortion, especially at high sound pressure levels. Certain ribbon and dynamic microphones are exceptions, due to the designers' assumption of 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 (sound pressure level) 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, usually caused by the diaphragm reaching its absolute displacement limit (or by the preamplifier), 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.

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

A pop filter in use during a recording session

A **pop filter** or **pop shield** is an anti-pop noise protection filter for microphones, typically used in a recording studio. It serves to reduce or eliminate 'popping' sounds caused by the mechanical impact of fast moving air on the microphone during recorded speech and singing. It can also protect against the accumulation of saliva on the microphone element. Salts in human saliva are corrosive and thus use of a pop filter may prolong the life of the microphone.

A typical pop filter is composed of one or more layers of acoustically semi-transparent material such as woven nylon stretched over a circular frame, and often includes a clamp and a flexible mounting bracket. Metal pop filters use a fine mesh metal screen in place of the nylon. An improvised pop shield, functionally identical to the professional units, can be made with material from tights or stockings stretched over a kitchen sieve, embroidery hoop or a loop of wire such as a bent coat hanger. It is important that the pop shield is not attached directly to the microphone as vibrations will be transmitted from the shield to the mic.

Popping sounds occur particularly in the pronunciation of aspirated plosives (such as the first 'p' in the English word "popping"). Pop filters are designed to attenuate the energy of plosive, which otherwise might exceed the design input capacity of the microphone, leading to clipping. Pop filters do not appreciably affect hissing sounds or sibilance, which is why de-essers are used instead.

A pop filter differs from a microphone windscreen. Pop filters are generally used in a studio environment, while windscreens are typically used outdoors. Windscreens are also used by vocalists on stage to reduce plosives and saliva, though they may not be as acoustically transparent as a studio pop filter.

Blimps

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

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- 2

Types of Microphones

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. 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.

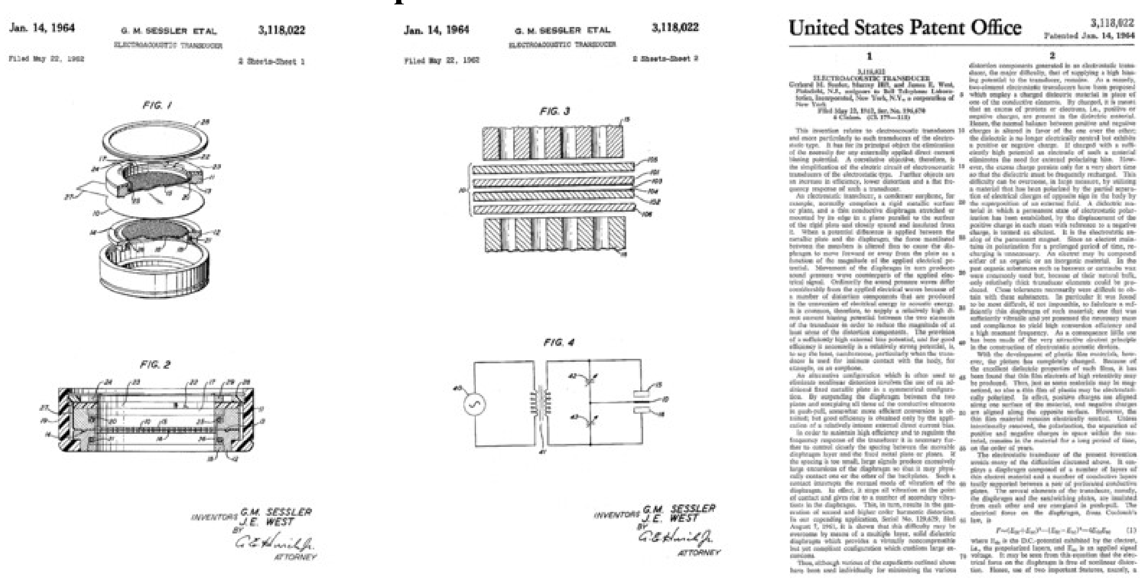
Within the time-frame of the capacitance change (as much as 50 ms at 20 Hz audio signal), the charge is practically constant and the voltage across the capacitor changes instantaneously to reflect the change in capacitance. The voltage across the capacitor varies above and below the bias voltage. The 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.

AKG C451B small-diaphragm condenser microphone

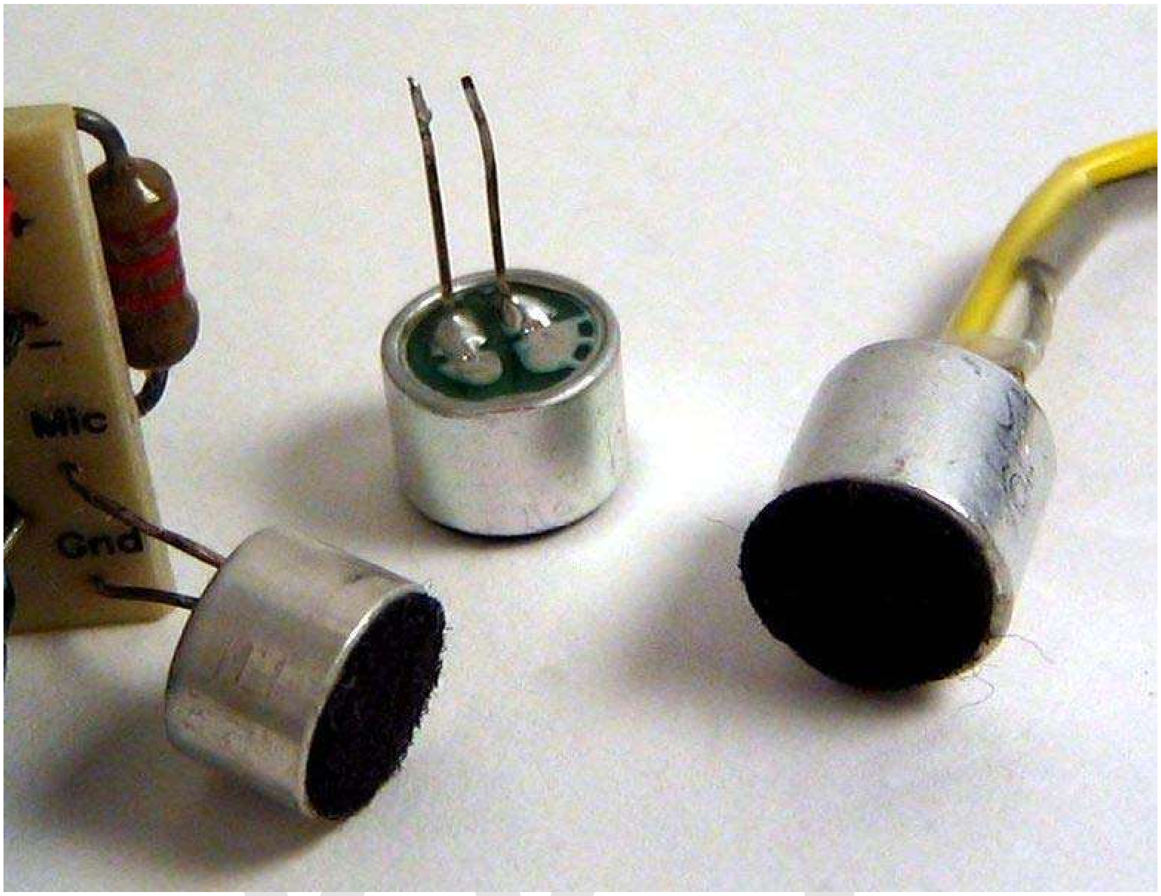
RF condenser microphones use a comparatively low RF voltage, generated by a low-noise oscillator. The oscillator may either be amplitude modulated by the capacitance changes produced by the sound waves moving the capsule diaphragm, or the capsule may be part of a resonant circuit that modulates the frequency 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 outputs 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 (see below), 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.

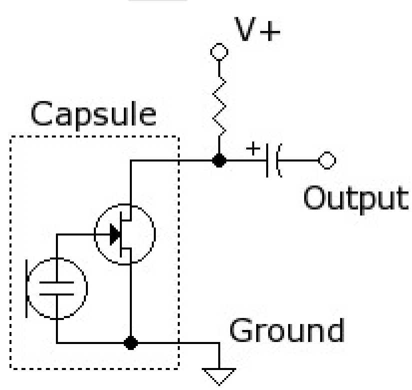
Electret condenser microphone



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



Electret condenser microphone capsules



A typical electret microphone preamp circuit uses an FET in a common source configuration. The two-terminal electret capsule contains an FET which must be externally powered by supply voltage V_+. The resistor sets the gain and output impedance. The audio signal appears at the output, after a DC-blocking capacitor.

An **electret microphone** is a type of condenser microphone, which eliminates the need for a polarizing power supply by using a permanently-charged material.

An *electret* is a stable dielectric material with a permanently-embedded static electric charge (which, due to the high resistance and chemical stability of the material, will not decay for hundreds of years). The name comes from *electro*static and magn*et*; drawing analogy to the formation of a magnet by alignment of magnetic domains in a piece of iron. Electrets are commonly made by first melting a suitable dielectric material such as a plastic or wax that contains polar molecules, and then allowing it to re-solidify in a powerful electrostatic field. The polar molecules of the dielectric align themselves to the direction of the electrostatic field, producing a permanent electrostatic "bias". Modern electret microphones use PTFE plastic, either in film or solute form, to form the electret.

History

Electret materials have been known since the 1920s, and were proposed as condenser microphone elements several times, but were considered impractical until the foil electret type was invented at Bell Laboratories in 1962 by Gerhard Sessler and Jim West, using a thin metallized Teflon foil. This became the most common type, used in many applications from high-quality recording and lavalier use to built-in microphones in small sound recording devices and telephones.

Though electret mics were once considered low cost and low quality, the best ones can now rival capacitor mics in every respect apart from low noise and can even have the long-term stability and ultra-flat response needed for a measuring microphone. Few electret microphones rival the best DC-polarized units in terms of noise level, but this is not due to any inherent limitation of the electret. Rather, mass production techniques needed to produce electrets cheaply do not lend themselves to the precision needed to produce the highest quality microphones.

Types

There are three major types of electret microphone, differing in the way the electret material is used:

Foil-type or diaphragm-type
: A film of electret material is used as the diaphragm itself. This is the most common type, but also the lowest quality, since the electret material does not make a particularly good diaphragm.

Back electret
: An electret film is applied to the back plate of the microphone capsule and the diaphragm is made of an uncharged material which may be mechanically more suitable for the transducer design being realized.

Front electret
: In this newer type, the back plate is eliminated from the design, and the condenser is formed by the diaphragm and the inside surface of the capsule. The electret film is adhered to the inside front cover and the metalized diaphragm is connected to

the input of the FET. It is equivalent to the back electret in that any conductive film may be used for the diaphragm.

Unlike other condenser microphones electret types require no polarizing voltage, but they normally contain an integrated preamplifier which does require a small amount of power (often incorrectly called polarizing power or bias). This preamp is frequently phantom powered in sound reinforcement and studio applications. Other types simply include a 1.5V battery in the microphone housing, which is often left permanently connected as the current drain is usually very small.

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.

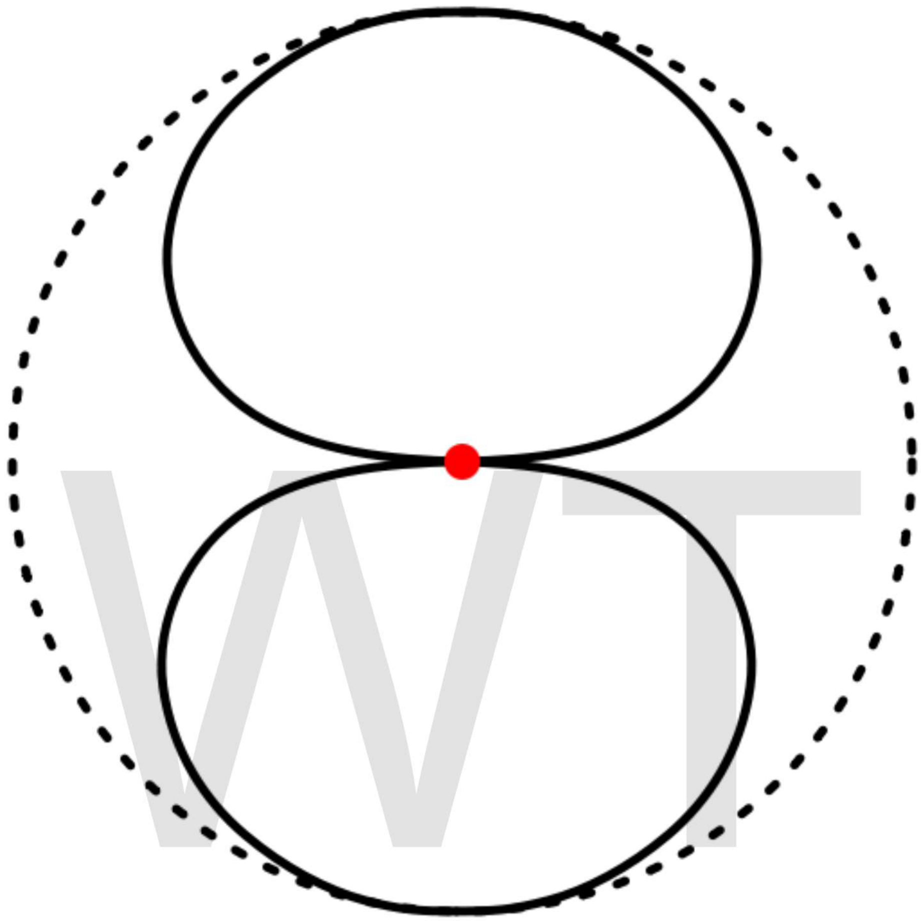
Moving-coil microphones use the same dynamic principle as in a loudspeaker, only reversed. A small movable induction coil, positioned in the magnetic field of a permanent magnet, is attached to the diaphragm. When sound enters through the windscreen of the microphone, the sound wave moves the diaphragm. 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 in 1942

A **ribbon microphone** is a type of dynamic microphone that uses a thin aluminum, duraluminum or nanofilm ribbon placed between the poles of a magnet to generate voltages by electromagnetic induction. Ribbon microphones are typically bidirectional, meaning they pick up sounds equally well from either side of the microphone.

Principle of operation



The sensitivity pattern of a bidirectional microphone (*red dot*) viewed from above

In the moving coil microphone, the diaphragm is attached to a light movable coil that generates a voltage as it moves back and forth between the poles of a permanent magnet. In ribbon microphones, a current is induced at right angles to both the ribbon velocity and magnetic field direction. As the sound wave causes the ribbon to move, the induced current in the ribbon is proportional to the particle velocity in the sound wave. The voltage output of older ribbon microphones is typically quite low compared to a dynamic moving coil microphone and a step-up transformer is used to increase the voltage output and increase the output impedance. Modern ribbon microphones do not suffer from this problem due to improved magnets and more efficient transformers, and have output levels that can exceed typical stage dynamic microphones.

Ribbon microphones were once delicate, and expensive, but modern materials make certain present-day ribbon microphones very durable and may be used for loud rock music and stage use. They are prized for their ability to capture high-frequency detail, comparing very favorably with condenser microphones, which can often sound subjectively "aggressive" or "brittle" in the high end of the frequency spectrum. Due to their bidirectional pick-up pattern, ribbon microphones are often used in pairs to produce the Blumlein Pair recording array. In addition to the standard bidirectional pick-up pattern, ribbon microphones can also be configured to have cardioid, hypercardioid, omnidirectional, and variable polar patterns.

As many mixers are equipped with phantom power in order to enable the use of condenser microphones, care should be taken when using condenser and ribbon microphones at the same time. If the ribbon microphone is improperly wired, which is not unheard of with older microphones, this capacity can damage some ribbon elements, but improvements in designs and materials have made those concerns largely a thing of the past.

History

RCA "44" ribbon microphone

In the early 1920s, Drs. Walter H. Schottky and Erwin Gerlach co-invented the first ribbon microphone. By turning the ribbon circuit in the opposite direction, they also invented the first ribbon loudspeaker.

In the late 1920s, Dr. Harry F. Olson of RCA began development of ribbon microphones, first with field coils and then with permanent magnets. One of the first ribbon microphones was the RCA PB-31. Produced in 1931, it was a breakthrough technology in sound, and revolutionized the audio recording and broadcasting industries, setting a new standard in frequency response. The clarity and realism were unmatched by any of the condenser microphones of its day.

Just a few months later, in 1932, the PB-31 was replaced by the 44A, which was enormously successful and highly regarded for its smooth tone and defined pattern control, which not only reduced the effects of reverberation on soundstages, but also offered higher gain-before-feedback in live sound applications. The 44A was updated with improved magnetic material in the 44B/44BX models. RCA also launched the unidirectional 77A/77B models and the multi-pattern 77C/77D mics. Nearly three-quarters of a century later, many of these RCA ribbon models are still hard-working audio tools prized by engineers worldwide.

Also of note is the ST&C Coles 4038 (or PGS - pressure gradient single) designed by the BBC in 1954 and still used for some applications to this day. Its uses varied from talks to symphony concerts and is regarded as a delicate, fine traditional microphone.

Around 2002, relatively inexpensive ($80 - $200) Chinese-manufactured ribbon microphones inspired by the RCA-44 and older Russian Oktava ribbon microphones became available..

In 2007, microphones employing ribbon elements made of strong nanomaterials became available, offering orders of magnitude improvement in signal purity and output level.

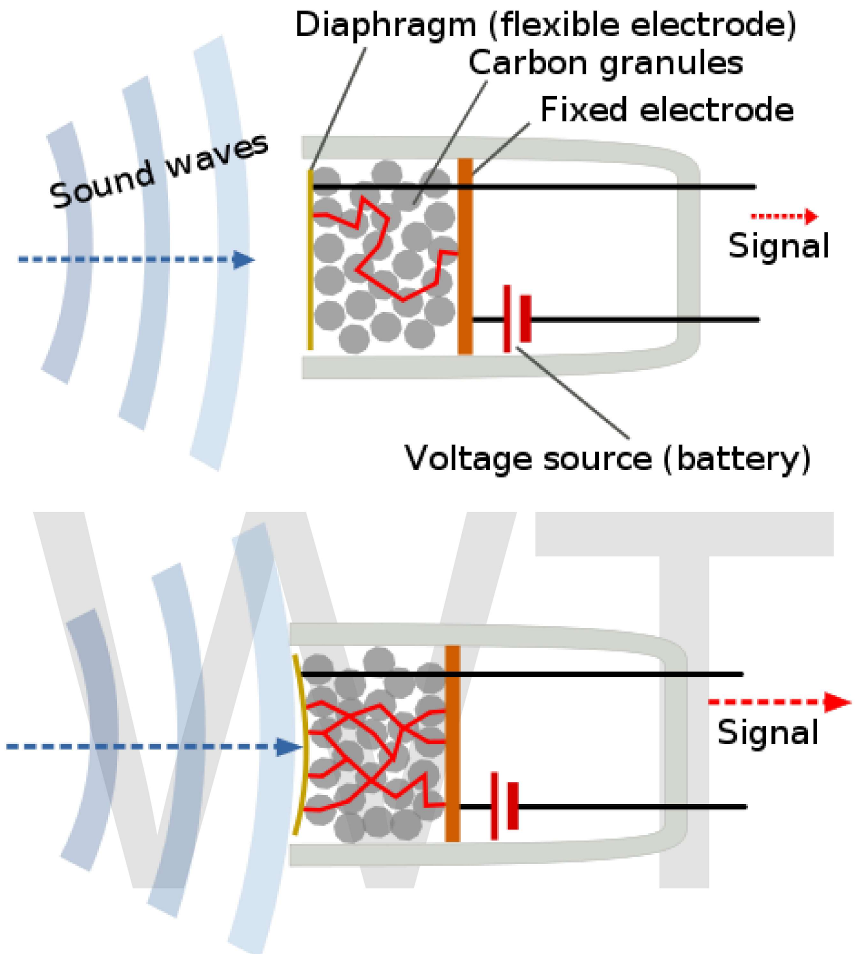
The ribbon microphone is an electrically simple design with no active circuitry; it is possible to build one from a kit, or with basic tools and materials. The acoustic complexity of ribbon microphones is comparable to other types of air coupled transducers.

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 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.

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.

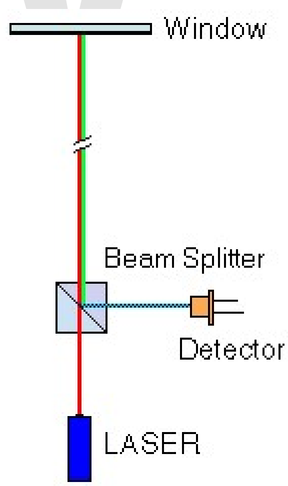
During operation, light from a laser source travels through an optical fiber to illuminate the surface of a tiny, sound-sensitive reflective diaphragm. Sound causes the diaphragm to vibrate, thereby minutely changing the intensity of the light it reflects. The modulated light is then transmitted over 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



A diagram of a simple laser microphone

The main type of **laser microphone** is a surveillance device that uses a laser beam to detect sound vibrations in a distant object. The object is typically inside a room where a conversation is taking place, and can be anything that can vibrate (for example, a picture on a wall) in response to the pressure waves created by noises present in the room. The object preferably has a smooth surface. The laser beam is directed into the room through a window, reflects off the object and returns to a receiver that converts the beam to an audio signal. The beam may also be bounced off the window itself. The minute differences in the distance traveled by the light as it reflects from the vibrating object are detected interferometrically. The interferometer converts the variations to intensity variations, and electronics are used to convert these variations to signals that can be converted back to sound.

This technology can be used to eavesdrop with minimal chance of exposure. However, countermeasures exist in the form of specialized light sensors that can detect the light from the beam. Rippled glass can be used as a defense, as it provides a poor surface for a laser microphone.

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. A video made at AES by Gearwire can be found via the external link, below.

Origins

The technique of using a light beam to remotely record sound probably originated with Léon Theremin in the Soviet Union at or before 1947, when he developed and used the *Buran* eavesdropping system. This worked by using a low power infrared beam (not a laser) from a distance to detect the sound vibrations in the glass windows. Lavrentiy Beria, head of the KGB, had used this Buran device to spy on the U.S., British, and French embassies in Moscow.

It has been reported that the National Security Agency makes use of laser microphones.

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.

Chapter- 3

Special Purpose Microphones and Applications

Hydrophone

A hydrophone

A (Greek "hydro" = "water" and "phone" = "sound") is a microphone designed to be used underwater for recording or listening to underwater sound. Most hydrophones are based on a piezoelectric transducer that generates electricity when subjected to a pressure change. Such piezoelectric materials, or transducers can convert a sound signal into an electrical signal since sound is a pressure wave. Some transducers can also serve as a projector (emitter), but not all have this capability, and may be destroyed if used in such a manner.

A hydrophone can "listen" to sound in air, but will be less sensitive due to its design as having a good acoustic impedance match to water, the denser fluid. Likewise, a microphone can be buried in the ground, or immersed in water if it is put in a waterproof container, but will give similarly poor performance due to the similarly bad acoustic impedance match.

History

The hydrophone was used late in World War I. Convoy escorts used them to detect U-boats, greatly lessening the effectiveness of the submarine. Ernest Rutherford, in England, led pioneer research in hydrophones using piezoelectric devices. His only patent was for a hydrophone device.

From late in World War I until the introduction of active sonar, hydrophones were the sole method for submarines to detect targets while submerged, and remain useful today.

Directional hydrophones

A small single cylindrical ceramic transducer can achieve near perfect omnidirectional reception. Directional hydrophones increase sensitivity from one direction using two basic techniques:

Focused Transducers

This device uses a single transducer element with a dish or conical-shaped sound reflector to focus the signals, in a similar manner to a reflecting telescope. This type of hydrophone can be produced from a low-cost omnidirectional type, but must be used while stationary, as the reflector impedes its movement through water. A new way to direct is to use a spherical body around the hydrophone. The advantage of Directivity Spheres is that you can move the hydrophone within the water and you get rid of the interferences produced by a conical-shaped element.

Arrays

Multiple hydrophones can be arranged in an array so that it will add the signals from the desired direction while subtracting signals from other directions. The array may be steered using a beamformer. Most commonly, hydrophones are arranged in a "line array" but may be in two or three dimensional arrangements.

SOSUS hydrophones, laid on the seabed and connected by underwater cables, were used, beginning in the 1950s, by the U.S. Navy to track movement of Soviet submarines during the Cold War along a line from Greenland, Iceland and the United Kingdom known as the GIUK gap.

Geophone

The term **geophone** derives from the Greek word "geo" meaning "earth" and "phone" meaning "sound".

A **geophone** is a device which converts ground movement (displacement) into voltage, which may be recorded at a recording station. The deviation of this measured voltage from the base line is called the seismic response and is analyzed for structure of the earth.

Geosource Inc. MD-79—8Hz, 335Ω geophone

Geophones have historically been passive analog devices and typically comprise a spring-mounted magnetic mass moving within a wire coil to generate an electrical signal. Recent designs have been based on Microelectromechanical systems technology which generates an electrical response to ground motion through an active feedback circuit to maintain the position of a small piece of silicon.

The response of a coil/magnet geophone is proportional to ground velocity, while microelectromechanical systems devices usually respond proportional to acceleration. Microelectromechanical systems have a much higher noise level (50 dB velocity higher) than geophones and can only be used in strong motion or active seismic applications.

The frequency response of a geophone is that of a harmonic oscillator, fully determined by corner frequency (typically around 10 Hz) and damping (typically 0.707). Since the corner frequency is proportional to the inverse root of the moving mass, geophones with low corner frequencies (< 1Hz) become unpractical. It is possible to lower the corner frequency electronically, at the price of higher noise and cost.

Although waves passing through the earth have a three-dimensional nature, geophones are normally constrained to respond to single dimension - usually the vertical. However, some applications require the full wave to be used and three-component or 3-C geophones are used. In analog devices, three moving coil elements are mounted in an orthogonal arrangement within a single case.

The majority of geophones are used in reflection seismology to record the energy waves reflected by the subsurface geology. In this case the primary interest is in the vertical motion of the Earth's surface. However, not all the waves are upwards travelling. A strong, horizontally transmitted wave known as ground-roll also generates vertical motion that can obliterate the weaker vertical signals. By using large areal arrays tuned to the wavelength of the ground-roll the dominant noise signals can be attenuated and the weaker data signals reinforced.

Analog geophones are very sensitive devices which can respond to very distant tremors. These small signals can be drowned by larger signals from local sources. It is possible though to recover the small signals caused by large but distant events by correlating signals from several geophones deployed in an array. Signals which are registered only at one or few geophones can be attributed to small, local events and thus discarded. It can be assumed that small signals that register uniformly at all geophones in an array can be attributed to a distant and therefore significant event.

The sensitivity of passive geophones is typically 30 Volts/(meter/second), so they are in general not a replacement for broadband seismometers.

Conversely, some applications of geophones are interested only in very local events. A notable example is in the application of Remote Ground Sensors (RGS) incorporated in Unattended Ground Sensor (UGS) Systems. In such an application there is an area of interest which when penetrated a system operator is to be informed, perhaps by an alert which could be accompanied by supporting photographic data.

Ionophone

An **ionophone** is a transducer for converting acoustic vibrations in plasma into an electrical signal, or for converting an electrical signal into acoustic vibrations in plasma. It can be used for "generation of subsonic or ultrasonic waves as well as a microphone or loudspeaker".

Ionophones are to plasma what microphones and loudspeakers are to air. Because the acoustic impedance of air and plasma are similar, an ionophone will work well as a microphone or loudspeaker when appropriately coupled to air.

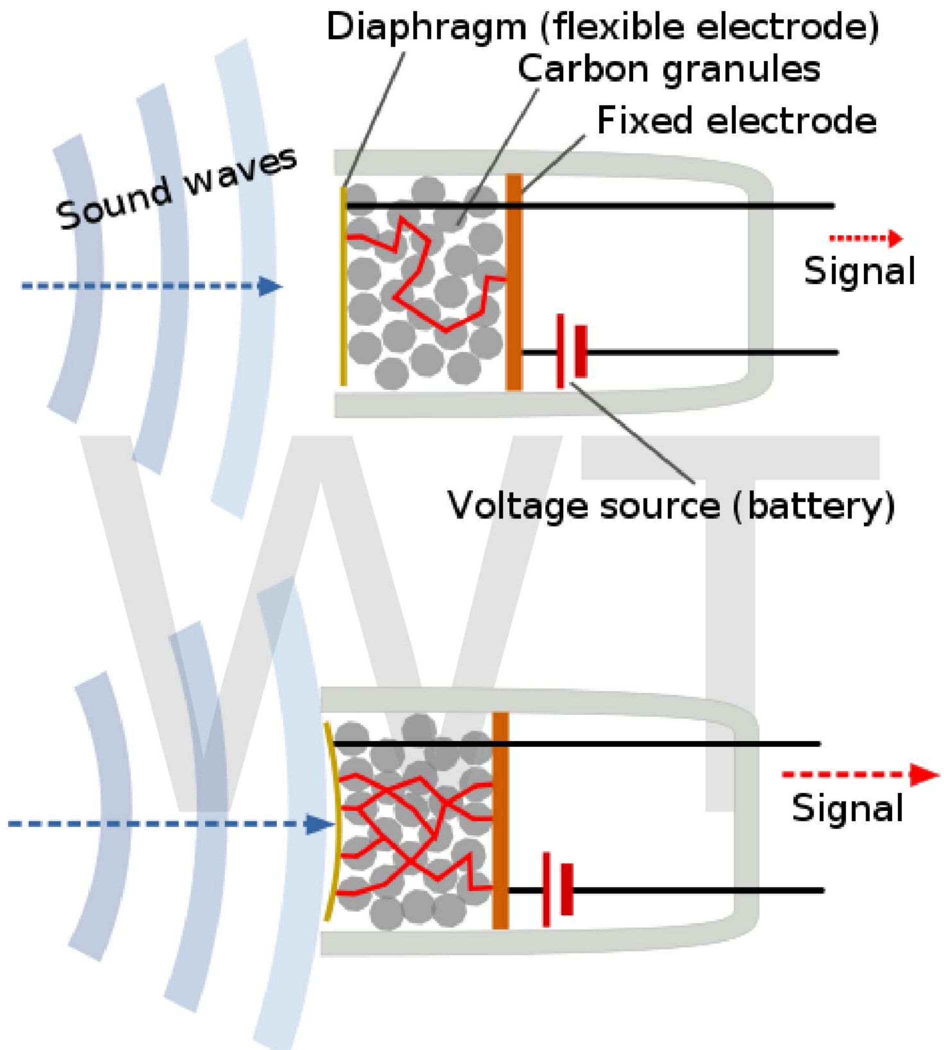
Ionophones are more commonly used as loudspeakers than as microphones. Unlike a loudspeaker having solid matter (cone) that moves to push air (or water in the case of an underwater speaker or hydrophone) sound is produced by plasma (ionized gas) in a spark gap. Typically a high frequency (ultrasonic) electric arc is established, and then modulated by audio frequencies.

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 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.

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.

History

Various individuals and organizations claim to be the inventors of the wireless microphone.

Reg Moores developed a radio microphone that was first used in "Aladdin on Ice" in 1949.

John F. Stephens developed an FM wireless microphone for a Navy musical show in 1951 on the Memphis Naval base. Each of the principal players/singers had their own microphone/transmitter. Subsequently, the Secret Service had Stephens modify his invention to be used in government "bugging" operations.

Herbert "Mac" McClelland, founder of McClelland Sound in Wichita, Kansas, fabricated a wireless microphone to be worn by baseball umpires at major league games broadcast by NBC from Lawrence-Dumont Stadium in 1951. The transmitter was strapped to the umpire's back. Mac's brother was Harold M. McClelland, the chief communications architect of the U.S. Air Force.

Shure Incorporated claims that its "Vagabond" system from 1953 was the first.

In 1957 German audio equipment manufacturer Sennheiser, at that time called Lab W, working with the German broadcaster Norddeutscher Rundfunk (NDR) exhibited a wireless microphone system. From 1958 the system was marketed through Telefunken under the name of Mikroport.

The first recorded patent was filed by Raymond A. Litke, an American electrical engineer with Educational Media Resources and San Jose State College, who invented the wireless microphone in 1957 to meet the multimedia needs for television, radio, and classroom instruction. His U.S. patent number 3134074, originally filed January 8, 1960, and granted May 19, 1964, is for the world's first portable and practical wireless microphone. At last, a dependable and wireless microphone with clarity, sound, and range proved to be as good as a microphone which used cords and cables. Two types were made available for purchase in 1959: hand-held and lavalier. Litke coined the term “lavalier microphone”, including the word in his patent application. The main transmitter module was a cigar-sized device which weighed seven ounces. The Federal Communications Commission (FCC) granted Litke 12 frequencies at his approval hearing.

Also called the Vega-Mike after Vega Electronics Corporation which first manufactured it in 1959, the midget device was first used by the broadcast media at the 1960 Democratic and Republican National Conventions. It allowed television reporters to roam the floor of the convention to interview participants where Presidential candidates Kennedy and Nixon were the first celebrities to use the wireless microphone. The American Broadcasting Company (ABC) completed testing in 1959, prior to the conventions. Television anchor John Daly was exuberant with his praises for Litke's invention during a TV news broadcast in July 1960. The wireless microphone was first tested at the Olympic trials held at Stanford University in 1959.

Litke attributes the inspiration of his invention to the winged communication of the bee.

Another German equipment manufacturer, Beyerdynamic, claim that first wireless microphone, was invented by Hung C. Lin. Called the "transistophone", it went into production in 1962. It is claimed that the first time a wireless microphone was used to record sound during filming of a motion picture was on Rex Harrison in the 1964 film My Fair Lady. However, Litke's microphone was the first used for public broadcasting by ABC at the Democratic and Republican National Conventions in 1960 where Presidential candidates Kennedy and Nixon were the first celebrities to use the wireless microphone. Furthermore, Vega Electronics Corporation was manufacturing Litke's hand-held wireless in 1959. That was the beginning of a workable and dependable wireless microphone.

Modern wireless microphone technology, which for the first time offered performance with audio and dynamic range equivalent to a cord, originated with the introduction of the first compander wireless microphone offered by Nady Systems, Inc in 1976 according to company claims. Todd Rundgren and The Rolling Stones were the first popular musicians to use these systems live in concert. Nady systems, Inc was honored with an Emmy award for this breakthrough technical achievement in 1996.

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. Electro-Voice, Shure, Nady Systems, Inc, Audio Ltd, Sennheiser, Lectrosonics, MIPRO, Samson Technologies, AKG Acoustics Audio-Technica 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 (see below).
- **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 demodulating 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, 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.

Microphone practice

There exist a number of well-developed microphone techniques used for miking musical, film, or voice sources. Choice of technique depends on a number of factors, including:

- The collection of extraneous noise. This can be a concern, especially in amplified performances, where audio feedback can be a significant problem. Alternatively, it can be a desired outcome, in situations where ambient noise is useful (hall reverberation, audience reaction).
- Choice of a signal type: Mono, stereo or multi-channel.
- Type of sound-source: Acoustic instruments produce a very different sound than electric instruments, which are again different from the human voice.
- Situational circumstances: Sometimes a microphone should not be visible, or having a microphone nearby is not appropriate. In scenes for a movie the microphone may be held above the pictureframe, just out of sight. In this way there is always a certain distance between the actor and the microphone.
- Processing: If the signal is destined to be heavily processed, or "mixed down", a different type of input may be required.
- The use of a windshield as well as a pop shield, designed to reduce vocal plosives.

Basic techniques

There are several classes of microphone placement for recording and amplification.

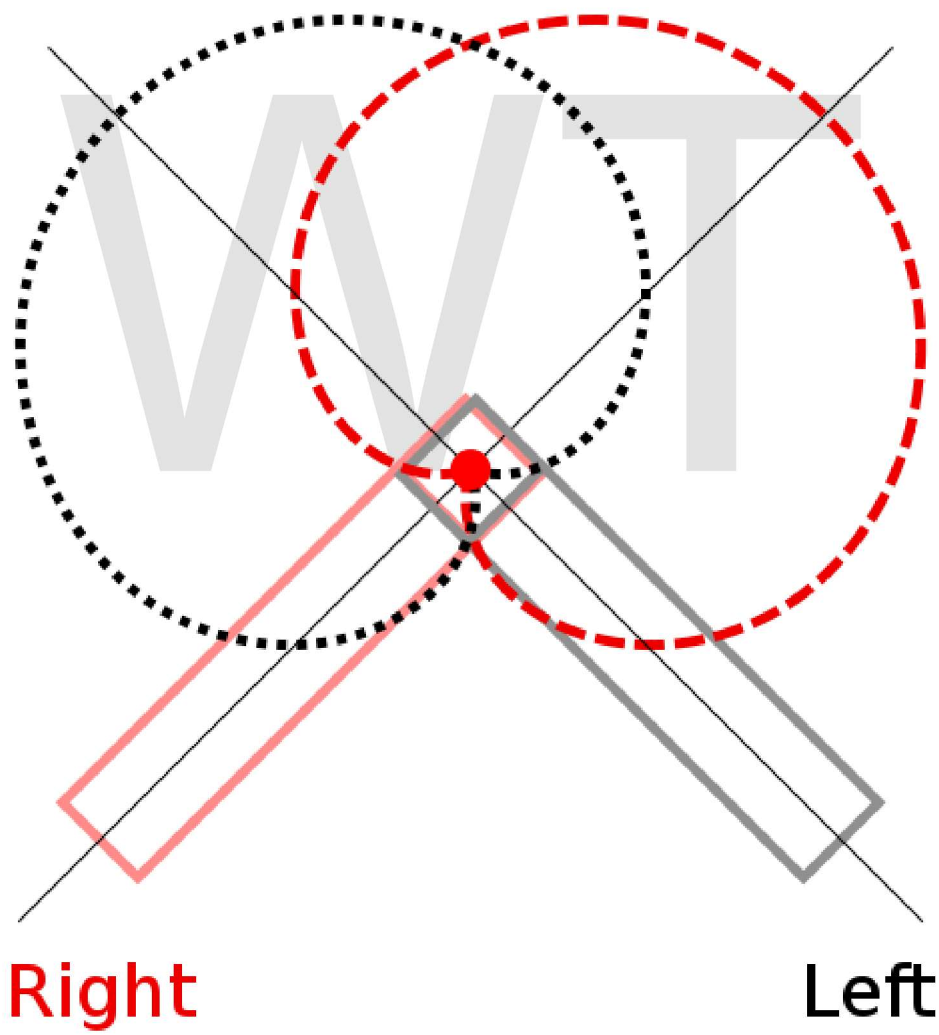
- In **close miking**, a microphone is placed relatively close to an instrument or sound source. This serves to reduce extraneous noise, including room reverberation, and is commonly used when attempting to record a number of separate instruments while keeping the signals separate, or when trying to avoid feedback in an amplified performance.
- In **ambient** or **distant miking**, a microphone — typically a sensitive one — is placed at some distance from the sound source. The goal of this technique is to get a broader, natural mix of the sound source or sources, along with ambient sound, including reverberation from the room or hall.

Stereo recording techniques

There are two features of sound that the human brain uses to place objects in the stereo sound-field between the loudspeakers. These are the relative level (or loudness) difference between the two channels *Δ L*, and the time-delay difference in arrival times for the same sound in each channel *Δ t*. The "interaural" signals (binaural *ILD* and *ITD*) at the ears are not the stereo microphone signals which are coming from the loudspeakers, and are called "interchannel" signals (*Δ L* and *Δ t*). These signals are normally not mixed. Loudspeaker signals are different from the sound arriving at the ear.

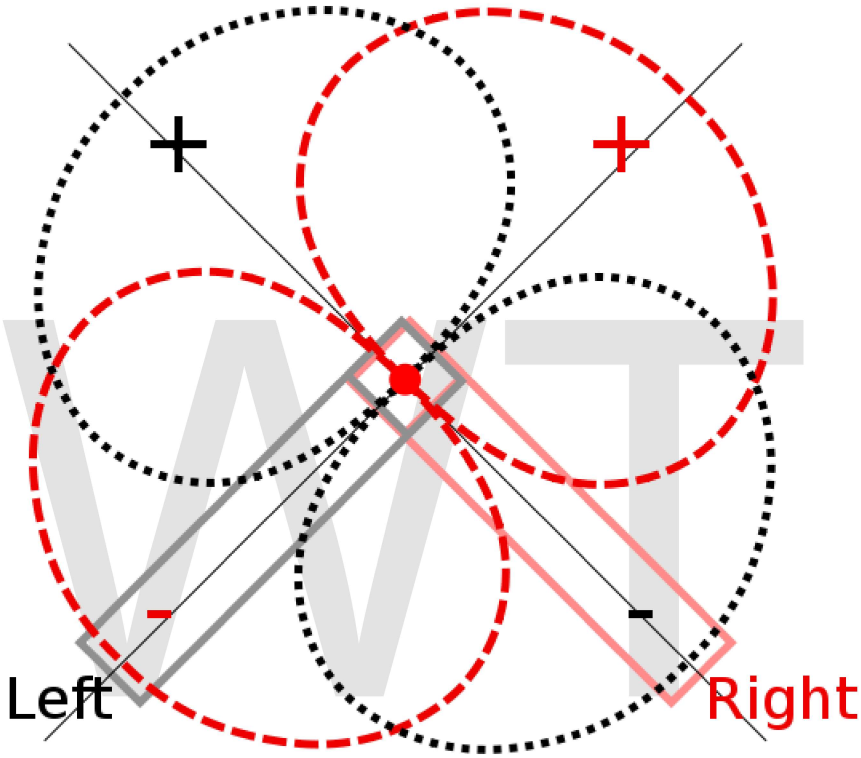
Various methods of stereo recording

X-Y technique: intensity stereophony



XY Stereo

Here there are two directional microphones at the same place, and typically placed at 90° or more to each other. A stereo effect is achieved through differences in sound pressure level between two microphones. Due to the lack of differences in time-of-arrival and phase-ambiguities, the sonic characteristic of X-Y recordings is generally less "spacy" and has less depth compared to recordings employing an AB-setup.



Blumlein Stereo

When the microphones are bidirectional and placed facing +-45° with respect to the sound source, the X-Y-setup is called a Blumlein Pair. The sonic image produced by this configuration is considered by many authorities to create a realistic, almost holographic soundstage.

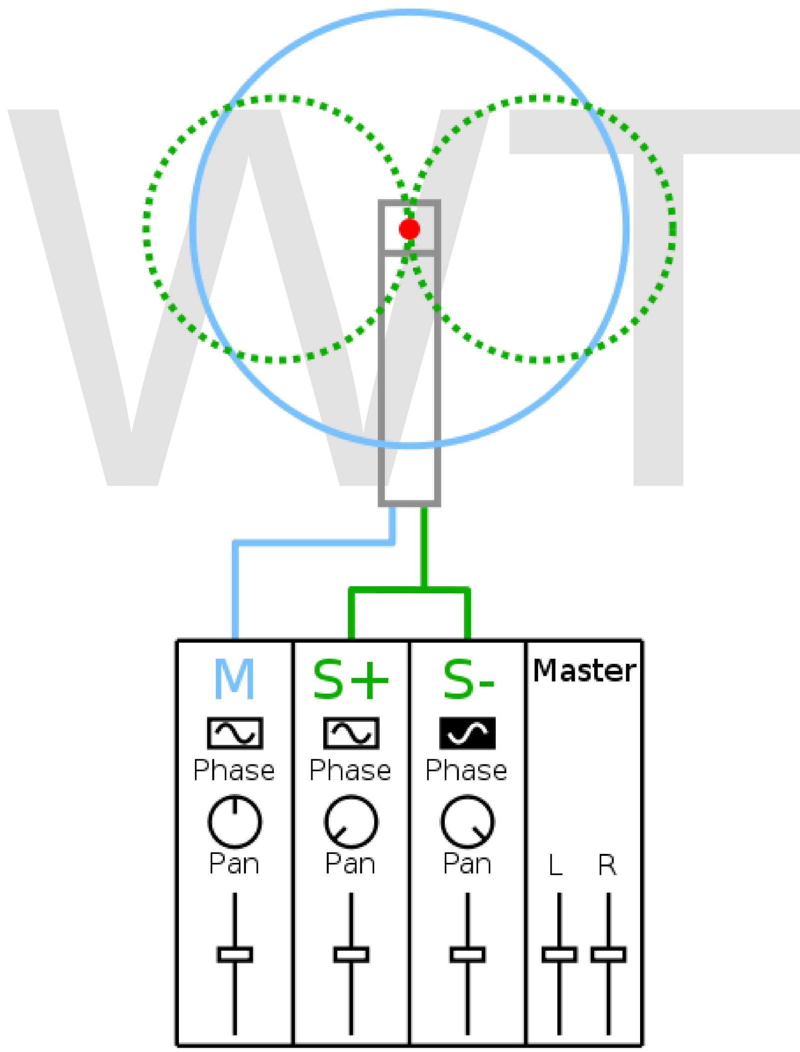
X-Y and Blumlein recordings offer the best compatibility with mono reproduction, barring the M/S technique.

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. Some such microphones have an adjustable angle of coverage between the two channels.

A-B technique: time-of-arrival stereophony

This uses two parallel omnidirectional microphones some distance apart, so capturing time-of-arrival stereo information as well as some level (amplitude) difference information, especially if employed close to the sound source(s). At a distance of about 50 cm (0.5 m) the time delay for a signal reaching first one and then the other microphone from the side is approximately 1.5 msec (1 to 2 msec). If you increase the distance between the microphones you effectively decrease the pickup angle. At 70 cm distance it is about equivalent to the pickup angle of the near-coincident ORTF setup.

M/S technique: Mid/Side stereophony



Mid-Side Stereo

This coincident technique employs a bidirectional microphone facing sideways and a cardioid (generally a variety of cardioid, although Alan Blumlein described the usage of an omnidirectional transducer in his original patent) at an angle of 90° facing the sound source. One mic is physically inverted over the other, so they share the same distance. The left and right channels are produced through a simple matrix: Left = Mid + Side, Right = Mid - Side (the polarity-reversed side-signal). This configuration produces a completely mono-compatible signal and, if the Mid and Side signals are recorded (rather than the matrixed Left and Right), the stereo width can be manipulated after the recording has taken place. This makes it especially useful for film-based projects.

Chapter- 4

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 converts an electrical signal into sound. The speaker moves in accordance with the variations of an electrical signal and causes sound waves to propagate through a medium such as air or water.

After the acoustics of the listening space, loudspeakers (and other electroacoustic transducers) are the most variable elements in a modern audio system and are usually responsible for most distortion and audible differences when comparing sound systems.

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.

History

Johann Philipp Reis installed an electric loudspeaker in his telephone in 1861; it was capable of reproducing pure tones, but also could reproduce speech. Alexander Graham Bell patented his first electric loudspeaker (capable of reproducing intelligible speech) as part of his telephone in 1876, which was followed in 1877 by an improved version from Ernst Siemens. Nikola Tesla reportedly made a similar device in 1881, but he was not issued a patent. During this time, Thomas Edison was issued a British patent for a system using compressed air as an amplifying mechanism for his early cylinder phonographs, but he ultimately settled for the familiar metal horn driven by a membrane attached to the stylus. In 1898, Horace Short patented a design for a loudspeaker driven by compressed air; he then sold the rights to Charles Parsons, who was issued several additional British patents before 1910. A few companies, including the Victor Talking Machine Company and Pathé, produced record players using compressed-air loudspeakers. However, these designs were significantly limited by their poor sound quality and their inability to reproduce sound at low volume. Variants of the system were used for public address applications, and more recently, other variations have been used to test space-equipment resistance to the very loud sound and vibration levels that the launching of rockets produces.

The modern design of moving-coil (also called *dynamic*) drivers was established by Oliver Lodge in 1898. The first practical application of moving-coil loudspeakers was established by Peter L. Jensen and Edwin Pridham, in Napa, California. Jensen was denied patents. Being unsuccessful in selling their product to telephone companies, in 1915 they changed strategy to public address, and named their product Magnavox. Jensen was, for years after the invention of the loudspeaker, a part owner of The Magnavox Company.

The moving-coil principle as commonly used today in direct radiators was patented in 1924 by Chester W. Rice and Edward W. Kellogg. The key difference between previous attempts and the patent by Rice and Kellogg was the adjustment of mechanical parameters so that the fundamental resonance of the moving system took place at a lower frequency than that at which the cone's radiation impedance had become uniform.

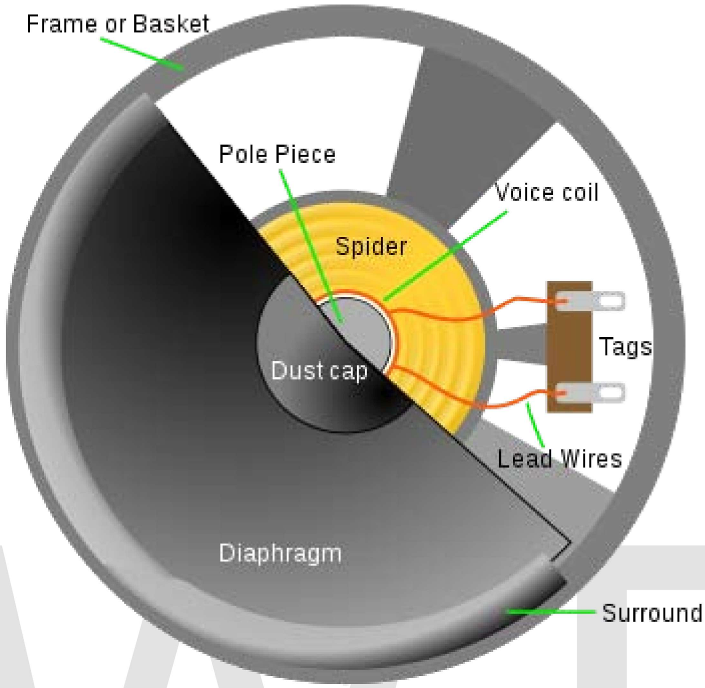
About this same period, Walter H. Schottky invented the first ribbon loudspeaker.

These first loudspeakers used electromagnets, because large, powerful permanent magnets were generally not available at a reasonable price. The coil of an electromagnet, called a field coil, was energized by current through a second pair of connections to the driver. This winding usually served a dual role, acting also as a choke coil, filtering the power supply of the amplifier to which the loudspeaker was connected. AC ripple in the current was attenuated by the action of passing through the choke coil; however, AC line frequencies tended to modulate the audio signal being sent to the voice coil and added to the audible hum of a powered-up sound reproduction device.

In the 1930s, loudspeaker manufacturers began to combine two and three bandpasses' worth of drivers in order to increase frequency response and sound pressure level. In 1937, the first film industry-standard loudspeaker system, "The Shearer Horn System for Theatres" (a two-way system), was introduced by Metro-Goldwyn-Mayer. It used four 15″ low-frequency drivers, a crossover network set for 375 Hz, and a single multi-cellular horn with two compression drivers providing the high frequencies. John Kenneth Hilliard, James Bullough Lansing, and Douglas Shearer all played roles in creating the system. At the 1939 New York World's Fair, a very large two-way public address system was mounted on a tower at Flushing Meadows. The eight 27″ low-frequency drivers were designed by Rudy Bozak in his role as chief engineer for Cinaudagraph. High-frequency drivers were likely made by Western Electric.

Altec Lansing introduced the '604', which was to become their most famous coaxial Duplex driver, in 1943, incorporating a high-frequency horn sending sound through the middle of a 15-inch woofer for near-point-source performance. Altec's "Voice of the Theatre" loudspeaker system arrived in the marketplace in 1945, offering better coherence and clarity at the high output levels necessary in movie theaters. The Academy of Motion Picture Arts and Sciences immediately began testing its sonic characteristics; they made it the film house industry standard in 1955. Subsequently, continuous developments in enclosure design and materials led to significant audible improvements. The most notable improvements in modern speakers are improvements in cone materials, the introduction of higher-temperature adhesives, improved permanent magnet materials, improved measurement techniques, computer-aided design, and finite element analysis.

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, fiberglass, 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. 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.

Mid-range driver

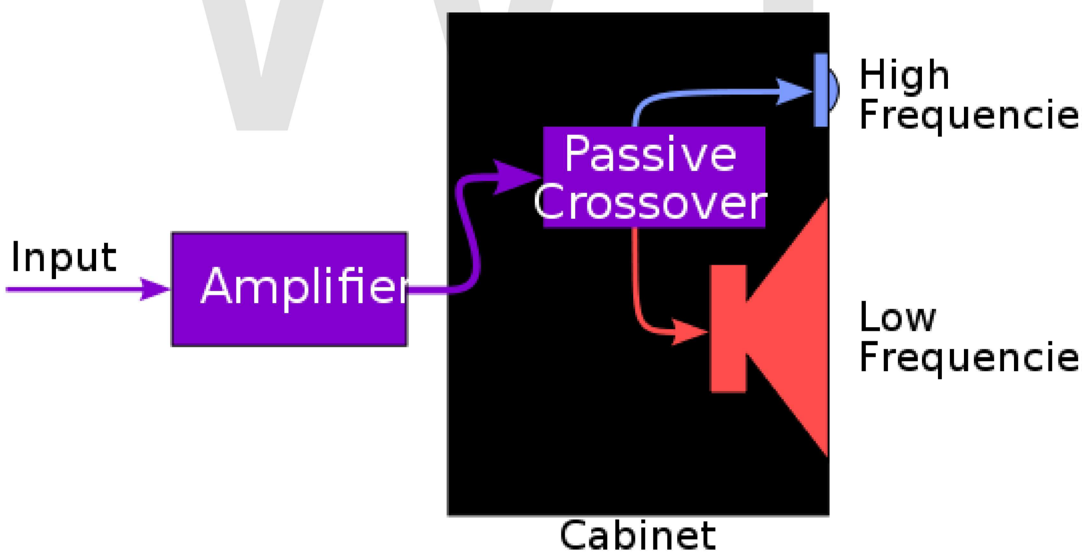
A mid-range speaker is a loudspeaker driver that reproduces middle frequencies. Mid-range driver diaphragms can be made of paper or composite materials, and can be direct radiation drivers (rather like smaller woofers) or they can be compression drivers (rather like some tweeter designs). If the mid-range 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.

Coaxial drivers

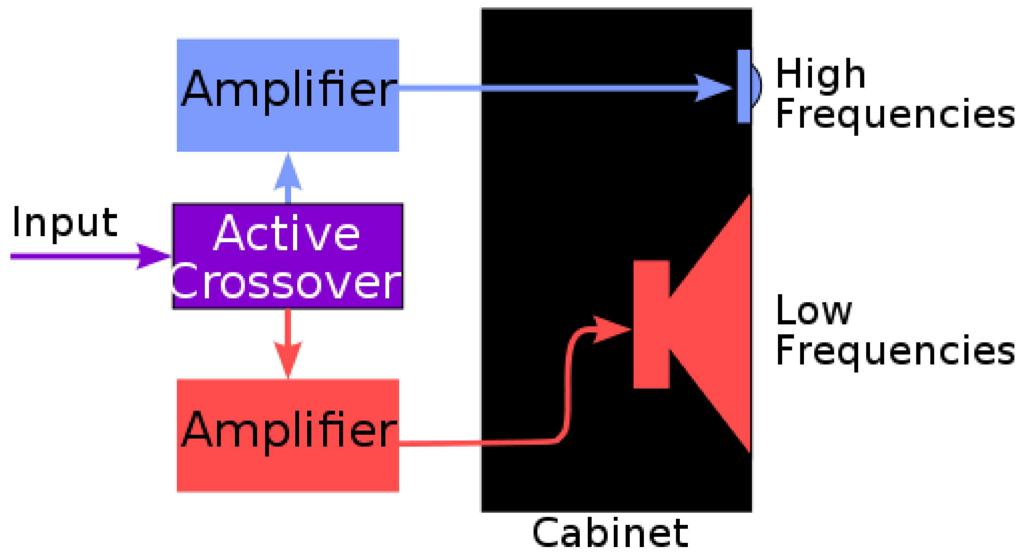
A coaxial driver is a loudspeaker driver with two or several combined concentric drivers. Coaxial drivers have been produced by many companies, such as Altec, Tannoy, Pioneer, KEF, BMS, Cabasse and Genelec.

Loudspeaker system design

Crossover



A passive crossover



Bi-amped

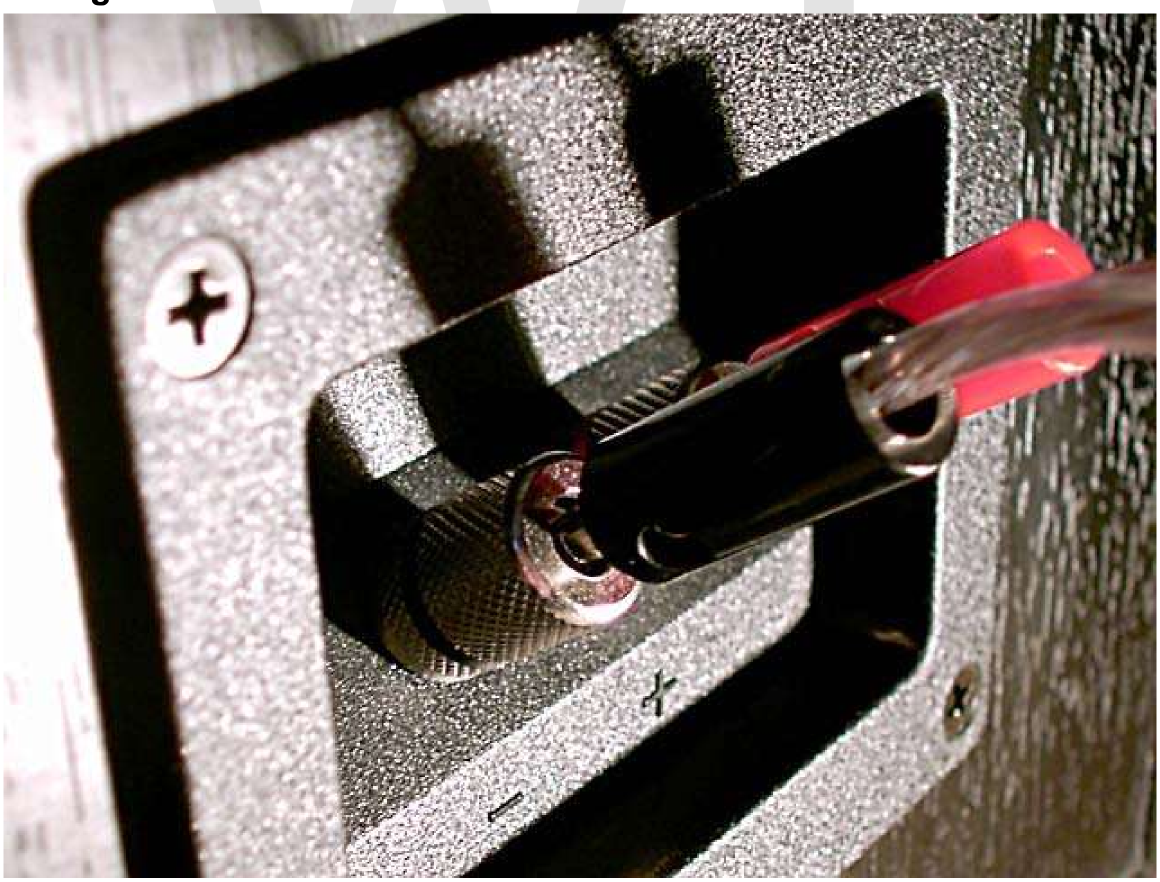
Used in multi-driver speaker systems, the crossover is a subsystem that separates the input signal into different frequency ranges suited to each driver. The drivers receive only the power in their usable frequency range (the range they were designed for), thereby reducing distortion in the drivers and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

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 diaphraghm, 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.

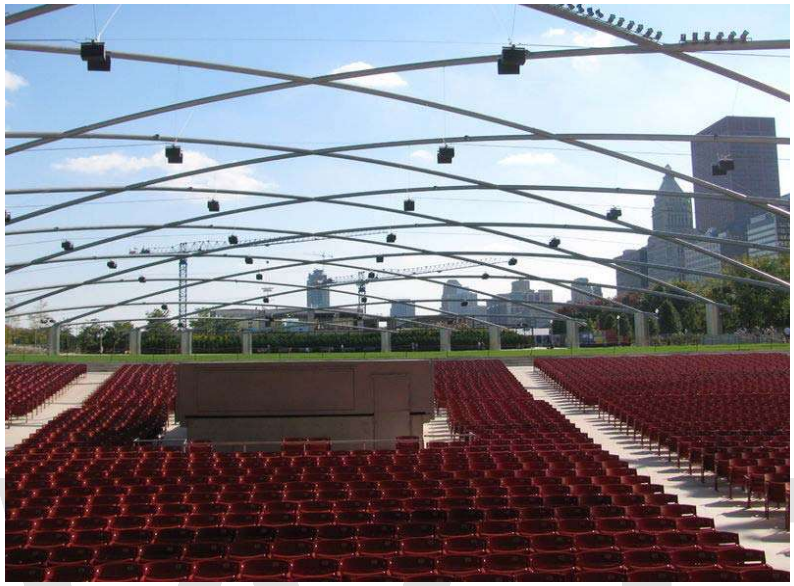
- Typical home loudspeakers have sensitivities of about 85 to 95 dB for 1 W @ 1 m—an efficiency of 0.5–4%.
- Sound reinforcement and public address loudspeakers have sensitivities of perhaps 95 to 102 dB for 1 W @ 1 m—an efficiency of 4–10%.
- Rock concert, stadium PA, marine hailing, etc. speakers generally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

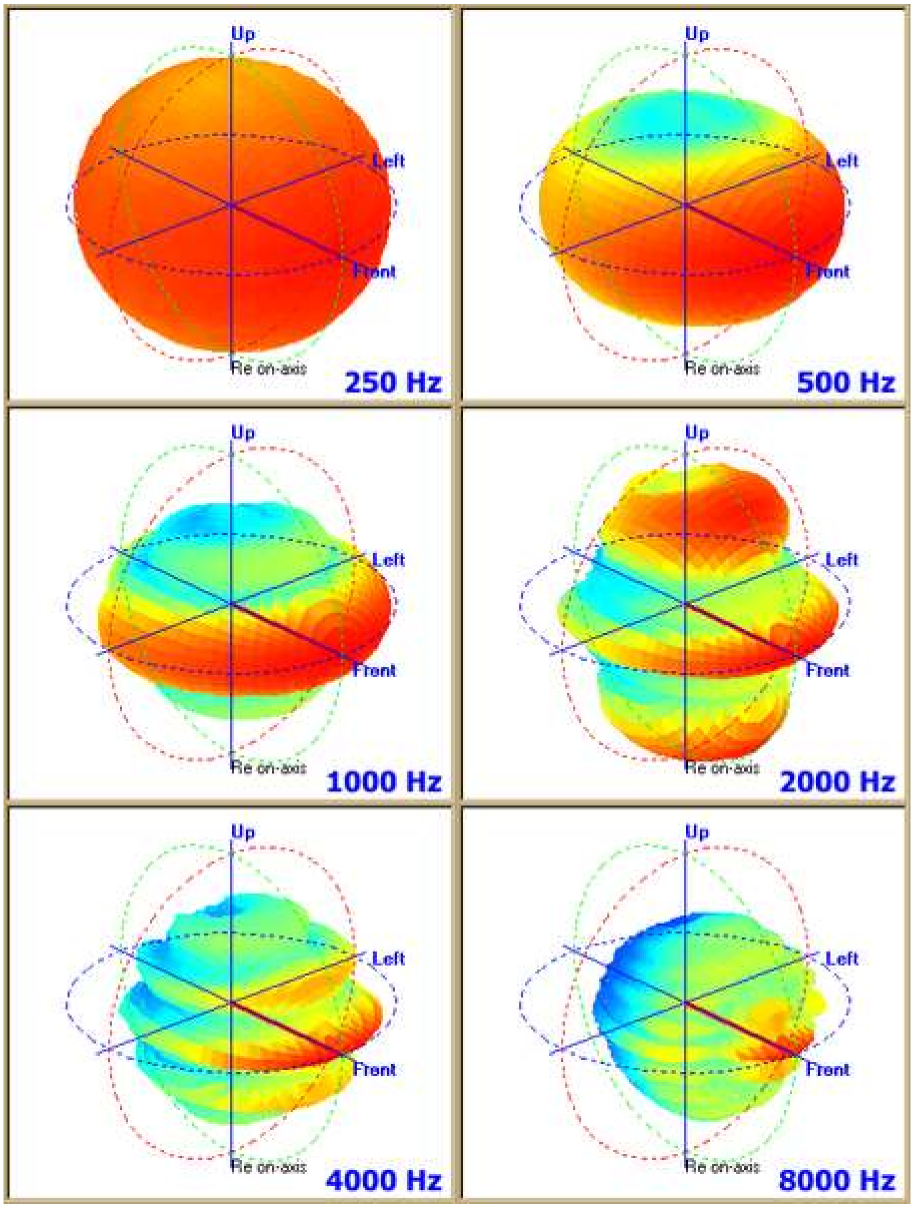
In a typical rectangular listening room, the hard, parallel surfaces of the walls, floor and ceiling cause primary acoustic resonance nodes in each of the three dimensions: left-right, up-down and forward-backward. Furthermore, there are more complex resonance modes involving three, four, five and even 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.

Chapter- 5

Subwoofer

A 12" subwoofer driver without an enclosure

A **subwoofer** (or simply "sub") is a woofer, or a complete loudspeaker typically between 8" and 21" in diameter, 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. Because of their limited frequency range, most subwoofers are used to augment the output of loudspeakers covering higher frequency bands.

Subwoofers are made up of one or more woofers in a well-braced wood or plastic loudspeaker enclosure, in one of a variety of designs, including bass reflex (with a port or tube in the enclosure), infinite baffle, horn-loaded, and bandpass designs, each of which has advantages and disadvantages in efficiency, size, distortion, cost, and power handling. 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.

History

The very first subwoofer was developed during the 1960s by Ken Kreisel, the former president of the Miller & Kreisel Sound Corporation in Los Angeles. When Kreisel's business partner, Jonas Miller, who owned a high-end audio store in Los Angeles, told Kreisel that some purchasers of the store's high-end electrostatic speakers had complained about a lack of bass response in the electrostatics, Kreisel designed a powered woofer that would reproduce only those frequencies that were too low for the electrostatic speakers to convey. Infinity's full range electrostatic speaker system that was developed during the 1960s also used a woofer to cover the lower frequency range that its electrostatic arrays did not handle adequately.

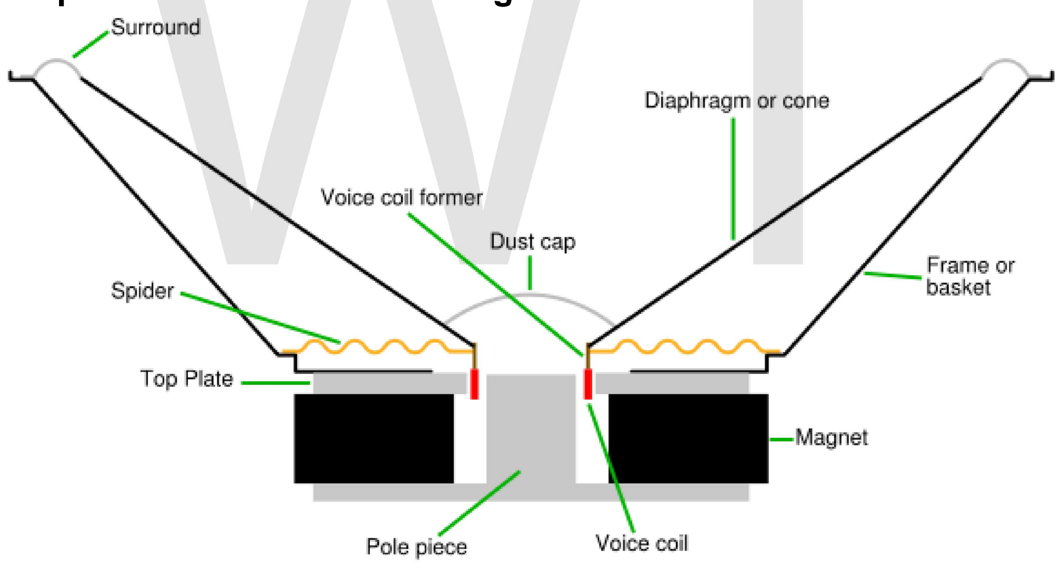
The first use of a subwoofer in a recording session was in 1973 for mixing the Steely Dan album *Pretzel Logic* when recording engineer Roger Nichols arranged for Kreisel to bring a prototype of his subwoofer to Village Recorders. Further design modifications were made by Kreisel over the next ten years, and in the 1970s and 1980s by engineer John P. D'Arcy; record producer Daniel Levitin served as a consultant and "golden ears" for the design of the crossover network (used to partition the frequency spectrum so that the subwoofer would not attempt to reproduce frequencies too high for its effective range, and so that the main speakers would not need to handle frequencies too low for their effective range).

Subwoofers received a great deal of publicity in 1974 with the movie *Earthquake* which was released in Sensurround. Initially installed in 17 U.S. theaters, the Sensurround system used large subwoofers which were driven by racks of 500 watt amplifiers which were triggered by control tones printed on one of the audio tracks on the film. Four of the subwoofers were positioned in front of the audience under (or behind) the film screen and two more were placed together at the rear of the audience on a platform. Powerful noise energy in the range of 17 Hz to 120 Hz was generated at the level of 110–120 decibels of sound pressure level, abbreviated dB(SPL). The new low frequency entertainment method helped the film become a box office success. More Sensurround systems were assembled and installed. By 1976 there were almost 300 Sensurround systems leapfrogging through select theaters. Other films to use the effect include the WW II naval battle epic *Midway* in 1976 and *Rollercoaster* in 1977.

For owners of 33 rpm LPs and 45 singles, loud *and* deep bass was limited by the ability of the phonograph record stylus to track the groove. Some hi-fi aficionados solved the problem by using reel-to-reel tape players which were capable of delivering accurate, naturally deep bass from acoustic sources, or synthetic bass not found in nature. With the popular introduction of the compact cassette and the CD, it became possible to add more low frequency content to recordings, and satisfy a larger number of consumers. Home subwoofers grew in popularity, as they were easy to add to existing multimedia speaker setups and they were easy to position or hide.

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-inch 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:

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

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, plastic or other dense materials. Because of the high power they use, subwoofers often require strong internal crossbracing to add strength and reduce box resonances.

There is a great deal of variety in the size of enclosures and, in the case of bass reflex systems, vent designs. When two or more subwoofers are placed in the same enclosure, they work together to move a greater mass of air, resulting in lower frequency extension. For example, if a single 12" subwoofer enclosure can go down to 40 Hz, a larger enclosure with four of these 12" drivers may be able to go to 30 Hz.

The smallest subwoofers are typically those designed for home theater users with limited space. The largest common subwoofer enclosures are those used for concert sound reinforcement systems or nightclub 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 in movie theaters 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.

Back panel of a Polk subwoofer. Notice consumer line-level and speaker-level inputs, the polarity switch and the crossover frequency control.

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.

A homemade car audio subwoofer speaker box with a 15 inch Boss Audio subwoofer and an empty space for a second driver

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

$$V_d = x_{max} \times S_d$$

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 low-frequency effects (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.

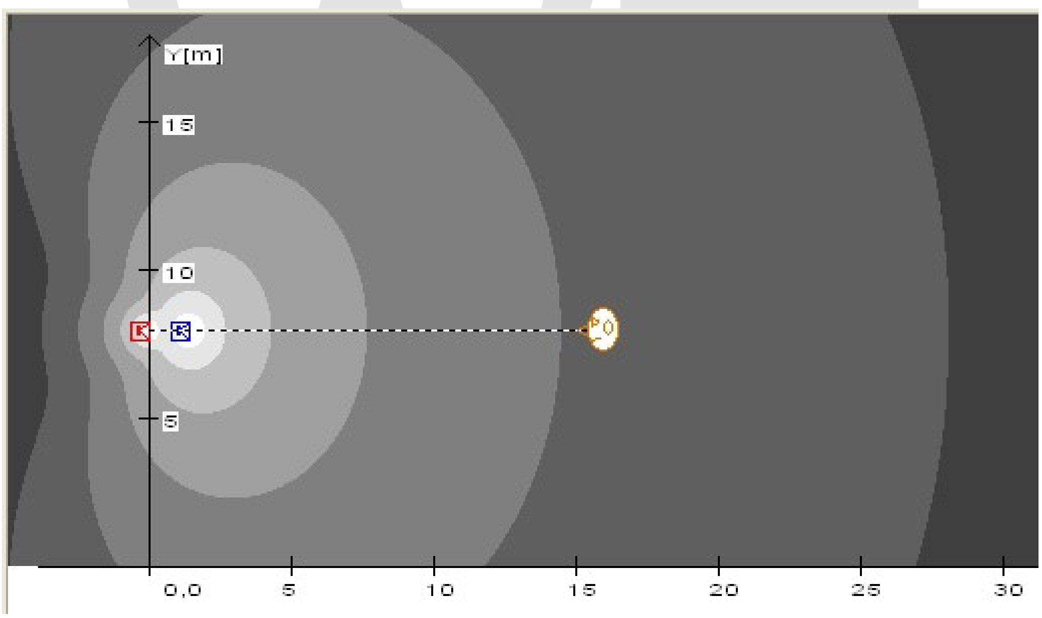
Full-range system

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



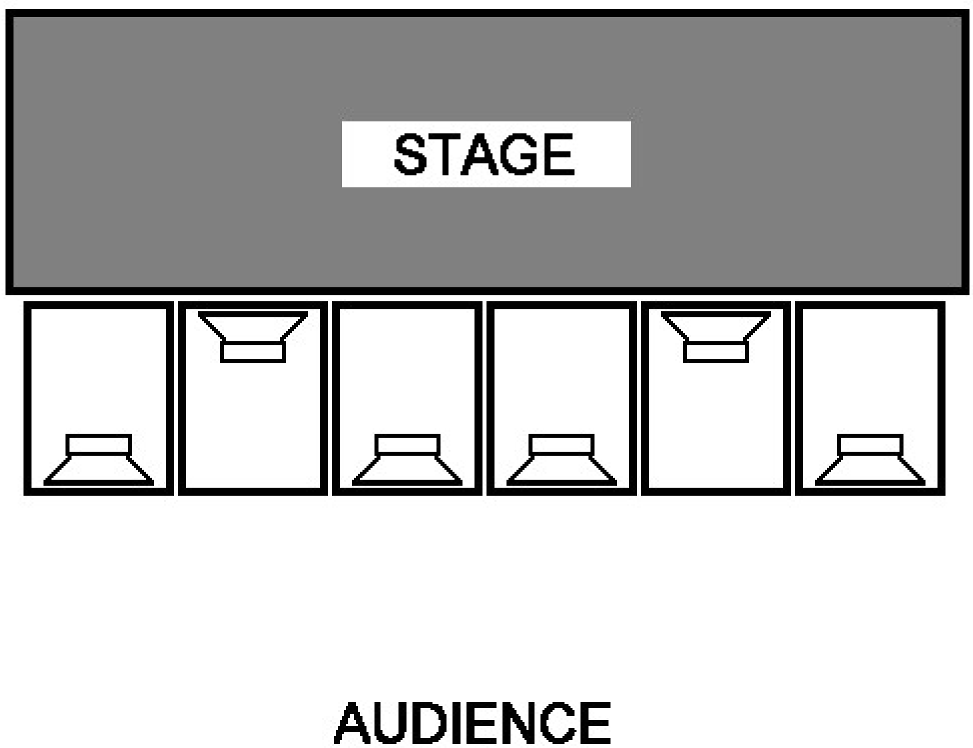
Cardioid dispersion pattern of two end-fire subwoofers placed one in front of the other. The enclosure nearest the listener is delayed by a few milliseconds.

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

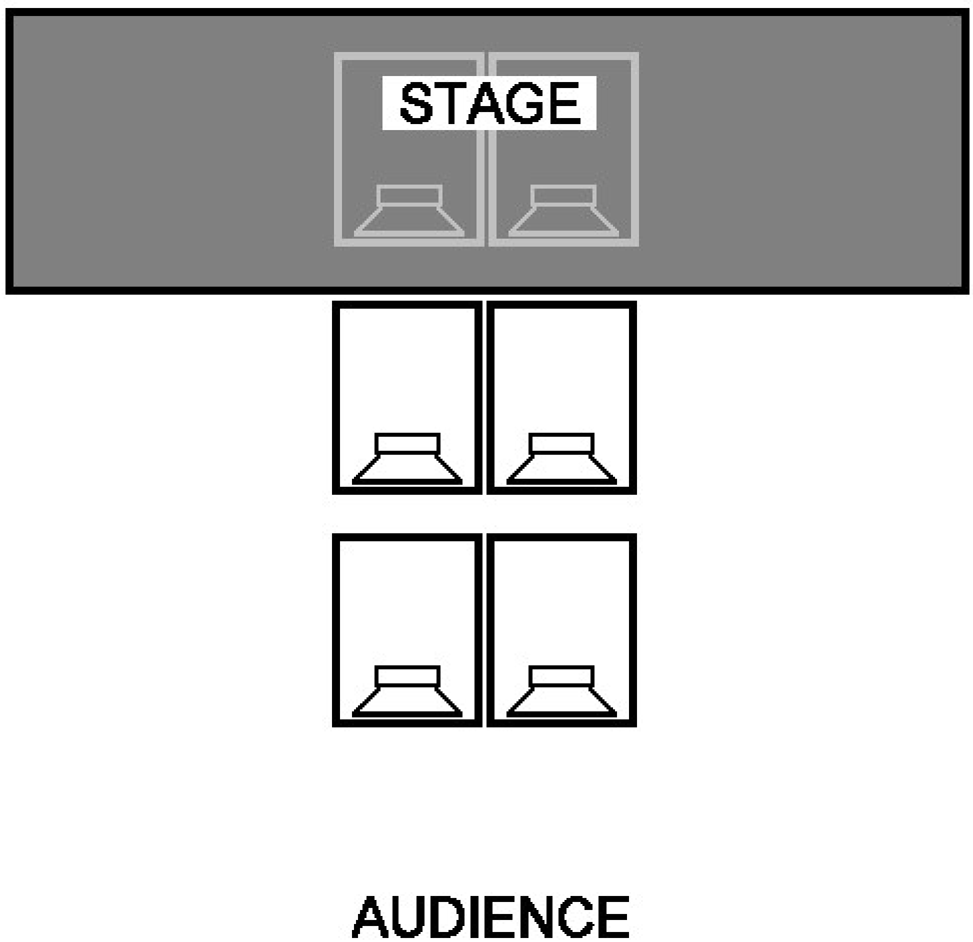


CSA: Six subwoofers arranged for less bass energy on stage. Signal going to the reversed enclosures is delayed a few milliseconds.

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- 6

Woofer and Tweeter

Woofer

A pair of woofers from Peerless

Woofer is the term commonly used for a loudspeaker driver designed to produce low frequency sounds, typically from around 40 hertz up to about a kilohertz or higher. The name is from the onomatopoeic English word for a dog's bark, "woof" (in contrast to the name used for speakers designed to reproduce high-frequency sounds, *tweeter*). The most common design for a woofer is the electrodynamic driver, which typically uses a stiff paper cone, driven by a voice coil which is surrounded by a magnetic field. The voice coil is attached by adhesives to the back of the speaker cone. The voice coil and magnet form a linear electric motor. When current flows through the voice coil, the coil moves in relation to the frame according to Fleming's left hand rule, causing the coil to push or pull on the driver cone in a piston-like way. The resulting motion of the cone creates sound waves as it moves in and out.

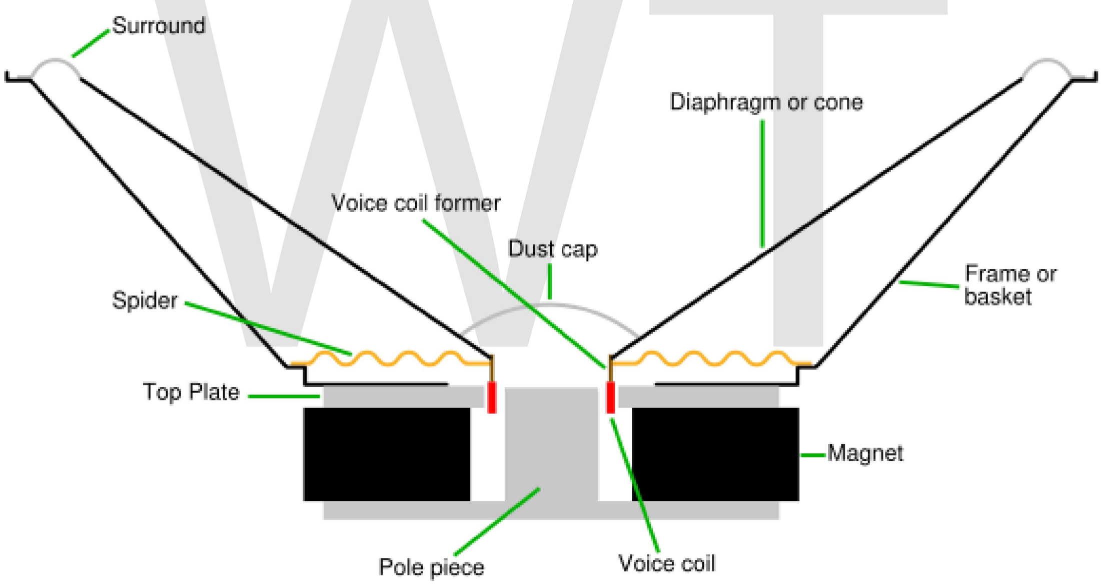
The frame, or basket, is the structure holding the cone, voice coil and magnet in the proper alignment, and must be as rigid as possible. Cast metal baskets are the most expensive and the most rigid in all directions. Stamped steel baskets are cheaper but less rigid, and cast plastic is cheaper still and has come into common use.

An important woofer specification is its power rating, the amount of power the woofer can handle without damage. The power rating is not easily characterized (see below) and many manufacturers cite momentary peak power ratings which would damage the speaker if maintained.

At ordinary sound pressure levels (SPL), most humans can hear down to about 20 Hz. Woofers are generally used to cover the lowest octaves of the system's frequency range. In two-way loudspeaker systems, the drivers handling the lower frequencies are also obliged to cover a substantial part of the midrange, often as high as 1000 or 2000Hz; such drivers are commonly termed *mid woofers*. Since the 1990s, a sub-type of woofer (termed subwoofer), which is designed for very low frequencies, has come to be commonly used in home theater systems and PA systems to augment the bass response; they usually handle the very lowest two or three octaves (i.e., from as low as 20 to perhaps 80 or 120 Hz).

Woofer design

Good woofer design requires effectively converting a low frequency amplifier signal to mechanical air movement with high fidelity and acceptable efficiency, and is both assisted and complicated by the necessity of using a loudspeaker enclosure to couple the cone motion to the air. If done well, many of the other problems of woofer design (for instance, linear excursion requirements) are reduced.



Cross-section of a standard loudspeaker, not to scale

In most cases the woofer and its enclosure must be designed to work together. Usually the enclosure is designed to suit the characteristics of the speaker or speakers used. The size of the enclosure is a function of the longest wavelengths (lowest frequencies) to be reproduced, and the woofer enclosure is much larger than required for mid-range and high frequencies.

A crossover network, either passive or active, filters the band of frequencies to be handled by the woofer and other speakers. Normally the crossover and speaker system, including the woofer, is expected to convert the electrical signal supplied by the amplifier to an acoustic signal of identical waveform without other interaction between the amplifier and speakers, although sometimes the amplifier and speakers are designed together with the speakers supplying distortion-correcting negative feedback to the amplifier.

There are many challenges in woofer design and manufacture. Most have to do with controlling the motion of the cone so the electrical signal to the woofer's voice coil is faithfully reproduced by the sound waves produced by the cone's motion. Problems include damping the cone cleanly without audible distortion so that it does not continue to move, causing ringing, when the instantaneous input signal falls to zero each cycle, and managing high excursions (usually required to reproduce loud sounds) with low distortion. There are also challenges in presenting to the amplifier an electrical impedance which is approximately constant at all frequencies.

An early version of the now widely-used bass-reflex cabinet design was patented by Albert L. Thuras of Bell Laboratories in 1932. Earlier speaker system designs paid little attention to the mounting and enclosure of the driver, and the longest wavelength (lowest frequency) handled was limited by the shortness of the distance the unwanted out-of-phase signal from the back of the driver had to travel before reaching and interfering with the wanted signal from the front. A. Neville Thiele in Australia, and later Richard H. Small (who taught and did design work in Australia, of England and the United States), first comprehensively adapted electronic filter theory to the design of loudspeaker enclosures, particularly at the low frequencies handled by woofers. This was a very considerable advance in the practice of woofer subsystem design, and is now almost universally used by system designers when applicable, although the T/S approach does not fully apply to some types of enclosure.

To use what are known as Thiele/Small (sometimes called T/S) design techniques, a woofer must first be carefully measured to characterize its electrical, magnetic, and mechanical properties; these are collectively known as the Thiele/Small parameters. They are now commonly included in the specification sheets for most higher-quality woofer drivers; not all, of course, have been carefully measured, and in any case, specific drivers may vary from the average run produced. In addition, some of these parameters can change during a driver's lifetime (especially during its first few hours or days of use) and so these parameters should really be measured after a suitable burn-in period to best match the enclosure design to the driver actually being used. This complicates manufacturing.

Resonant frequency is an important parameter, and is determined by a combination of the compliance (i.e., flexibility) of the cone suspension and by the mass of the moving parts of the speaker (the cone, voice coil, dust cap and some of the suspension). When the resonant frequency of the driver is combined with the magnetic motor characteristics, the electrical characteristics of the driver, and the acoustic environment provided by the enclosure, there will be a related, but different resonance characteristic, that of the loudspeaker system as a whole. As a rule of thumb the lower the system's resonant frequency, the lower the frequency reproducible by the speaker system for a given level of distortion. The resonant frequency of the driver is usually listed in its specification sheet as F_s.

All woofers have electrical and mechanical properties that very strongly influence the correct size of enclosure of a given type (e.g., bass reflex, sealed enclosure, "infinite baffle", etc.) for a given performance and efficiency. There is a tradeoff between sealed enclosure size, system resonance, and power efficiency; for a given driver specifying two of these quantities determines the third. This is known as Hoffman's Iron Law (Hoffman is the H in KLH). Similar relationships apply to other types of enclosure. As high-power solid-state amplifiers are available, a common tradeoff is to sacrifice efficiency and produce a relatively small enclosure capable of reproducing low frequencies, but requiring a lot of power; in the days of vacuum tube amplifiers a 35W amplifier was large and expensive, and large, efficient, enclosures were more often used.

These woofer characteristics also strongly affect the crossover components needed for a given performance in a particular loudspeaker system.

A given woofer may work well in one enclosure type, but not in another due to its T/S measurements. For instance, a woofer with a small maximum excursion (often those with critically-hung voice coils) will not be suited to acoustic suspension designs (which require generous excursions), nor for use in bass-reflex designs without an electrical filter preventing signals much below the system resonance from reaching the woofer. In this last case, the enclosure no longer seriously "loads" the woofer (i.e., controls excursion via impedance matching to the atmosphere) below that resonance frequency; cone excursions increase greatly, and for some drivers beyond safe limits. It is essential to know and understand at least the Thiele/Small parameters of a driver in order to design a satisfactory loudspeaker system using it.

There are several computer programs, both open source and proprietary, that perform the complicated T/S calculations.

Thiele/Small analysis is not applicable to horn or transmission-line enclosures. Only recently has a mathematical model of transmission-line enclosures suitable for design use been produced.

Active loudspeakers

In 1965, Sennheiser Electronics introduced the Philharmonic sound system, which used electronics to overcome some of the problems ordinary woofer subsystems confront. They added a motion sensor to the woofer, and used the signal corresponding to its actual motion to feedback as a control input to a specially designed amplifier. If carefully done, this can improve performance (both in 'tightness', and extension of low frequency performance) considerably at the expense of flexibility (the amplifier and the speaker are tied together permanently) and cost. In the US, L W Erath, an oil industry engineer, introduced a line of high end speakers along very much the same lines.

As electronics costs have decreased, it has become common to have sensor-equipped woofers in inexpensive 'music systems', boom boxes, or even car audio systems. This is usually done in an attempt to get better performance from inexpensive or undersized drivers in lightweight or poorly designed enclosures. This approach presents difficulties as not all distortion can be eliminated using servo techniques, and a poorly designed enclosure can swamp the benefits from any attempt at electronic correction.

Equalized loudspeakers

Because the characteristics of a loudspeaker can be measured, and to a considerable extent predicted, it is possible to design special circuitry that somewhat compensates for the deficiencies of a speaker system.

Equalization techniques are used in most public address and sound reinforcement applications. Here, the problem is not primarily hi-fi reproduction, but managing the acoustic environment. In this case, the equalization must be individually adjusted to match the particular characteristics of the loudspeaker systems used and the room in which they are used.

Digital filtering crossover and equalization

Computer techniques, in particular DSP techniques make possible a higher precision crossover technique. By using FIR and other digital techniques, the crossovers for a bi-amped or tri-amped system can be accomplished with a precision not possible with analog filters, whether passive or active. Furthermore, many driver peculiarities (down to and including individual variances) can be remedied at the same time, using the same techniques. One of Klein and Hummel's recent designs is implemented using these techniques. Because of the complex and advanced techniques involved, this approach is unlikely to be used in lower cost equipment for some time to come.

Cone materials

Two P-Audio Woofers. Note the cast frame, vented pole piece and reinforced paper cone.

All cone materials have advantages and disadvantages. The three chief properties designers look for in cones are light weight, stiffness, and lack of coloration (due to absence of ringing). Exotic materials like Kevlar and magnesium are light and stiff, but can have ringing problems, depending on their fabrication and design. Materials like paper (including coated paper cones) and various polymers will generally ring less than metal diaphragms, but can be heavier and not as stiff. There have been good and bad woofers made with every type of cone material. Almost every kind of material has been used for cones, from fibreglass and bamboo fibers to expanded aluminum honeycomb sandwich panel material and mica-loaded plastic cones.

Frame design

The frame, or basket, is the structure holding the cone, voice coil and magnet in the proper alignment. Since the voice coil gap is quite narrow (clearances are typically in the low thousandths of an inch), rigidity is important to prevent rubbing of the voice coil against the magnet structure in the gap and also avoid extraneous motions. There are two main metal frame types, stamped and cast. Stamped baskets (usually of steel) is a lower-cost approach. The disadvantage of this type of frame is that the basket may flex if the speaker is driven at high volumes, there being resistance to bending only in certain directions. Cast baskets are more expensive, but are usually more rigid in all directions, have better damping (reducing their own resonance), can have more intricate shapes, and are therefore usually the preferred for higher quality drivers.

Power handling

An important woofer specification is its power rating, the amount of power the woofer can handle without damage. The electrical power rating is not easily characterized and many manufacturers cite peak ratings attainable only for very brief moments without damage. Woofer power ratings become important when the speaker is pushed to extremes: applications requiring high output, amplifier overload conditions, unusual signals (i.e., non-musical ones), very low frequencies at which the enclosure provides little or no acoustic loading (and so there will be maximum cone excursion), or amplifier failure. In high-volume situations, a woofer's voice coil will heat up, increase its resistance, causing "power compression", a condition where output sound power level decreases after extended high power activity. Further heating can physically distort the voice coil, causing scuffing, shorting due to wire insulation deterioration, or other electrical or mechanical damage. Sudden impulse energy can melt a section of voice coil wire, causing an open circuit and a dead woofer; the necessary level will vary with driver characteristics. In normal listening level music applications, the electrical power rating of woofers is generally unimportant; it remains important for higher frequency drivers.

There are four types of power handling in loudspeaker drivers, including woofers: thermal (heat), electrical (both covered above), mechanical, and acoustic. The mechanical power handling limit is reached when cone excursion extends to its maximum limit. Thermal power handling limits may be reached when fairly high power levels are fed to a woofer for too long, even if not exceeding mechanical limits at any time. Most of the energy applied to the voice coil is converted to heat, not sound; all of the heat is ultimately passed to the pole piece, the rest of the magnet structure, and the frame. From the woofer structure, the heat is eventually dissipated into the surrounding air. Some drivers include provisions for better cooling (e.g., vented magnet pole pieces, dedicated heat conduction structures) to reduce increased coil/magnet/frame temperatures during operation, especially high power level conditions. If too much power is applied to the voice coil as compared to its ability to shed heat, it will eventually exceed a maximum safe temperature. Adhesives can melt, the voice coil former can melt or distort, or the insulation separating the voice coil windings can fail. Each of these events will damage the woofer, perhaps beyond usability.

Acoustic power handling is directly related to driver and enclosure efficiency. Some combinations are much more efficient so they can handle much more applied power than less efficient combinations. Energy input which is emitted as sound does not contribute to voice coil heating. So, as a rule of thumb, a voice coil in a magnet structure will be able to safely handle more applied power if the system components working together are more efficient at creating sound output.

Public address (PA) and instrument applications

Woofers designed for public address system(PA) and instrument amplifier applications are similar in makeup to home audio woofers, except that they are more heavy duty.

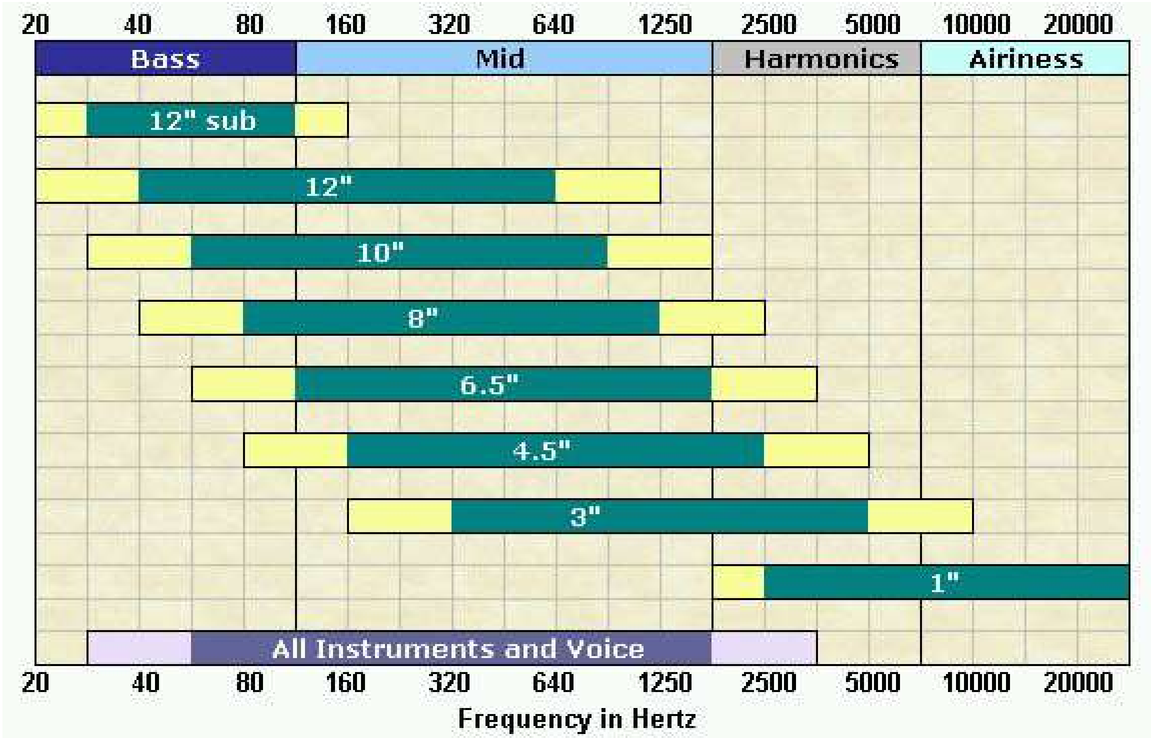
Typically, design variances include: cabinets built for repeated shipping and handling, larger woofer cones to allow for higher sound pressure levels, more robust voice coils to withstand higher power, and higher suspension stiffness. Generally, a home woofer used in a PA/instrument application can be expected to fail more quickly than a PA/instrument woofer. On the other hand, if you use a PA/instrument woofer in a home audio application it will not likely have the same quality of performance, particularly at low volumes. A PA woofer will not produce the same audible high fidelity which is the goal of high quality home audio due those differences.

PA system woofers typically have high efficiency, and high power handling capacity. The trade off for high efficiency at reasonable cost is usually relatively low excursion capability (i.e., inability to move "in and out" as far as many home woofers can) as they are intended for horn or large reflex enclosures. They are also usually ill-suited to extended low bass response since the last octave of low frequency response increases size and expense considerably, and is increasingly uneconomic to attempt at high levels as in a PA application. A home stereo woofer, because it is used at relatively low volumes, may be able to handle very low frequencies. Because of this, most PA woofers are not well suited to use in high quality high fidelity home applications and vice versa.

Frequency ranges

At ordinary sound pressure levels (SPL), most humans can hear down to about 20 Hz. To accurately reproduce the lowest tones, a woofer, or group of woofers, must move an appropriately large volume of air; a task that becomes more difficult at lower frequencies. The larger the venue, the more air the woofer movement will have to displace in order to produce the required sound power at low frequencies.

The chart below gives an approximate account of the frequency ranges of several sizes of woofers; it is suggestive only. For instance, in special cases like full-sized horns, small drivers can reproduce surprisingly low frequency material at useful levels with low distortion. The green area represents the range a single woofer of that size can commonly manage with reasonable distortion and useful output levels, while the yellow shows uncommon extended frequency capability. The purple area at the bottom represents the fundamental musical frequency range of common instruments. The lighter purple areas note instrument ranges with rarely played notes, for instance, the first and last 10 keys on a standard piano; the frequency range of the notes on a standard 88-key piano is 27 to 4,096 Hz. It's important to understand that pianos, like all instruments, produce harmonic overtones along with every fundamental note which are important in properly reproducing their sound; these are marked at the top of the chart. The "airiness" frequency block is an attempt to distinguish a frequency range with less harmonic content, but clearly present in live instrument performance, as reproduction through loudspeakers with and without filters blocking these frequencies are distinguishable; no term for the effect is really satisfactory. By comparing the instrument ranges (and harmonics) versus typical driver ranges, an indication of the difficulties faced by speaker designers can be seen. No woofer can do everything well.



Note that this chart does not include bigger woofers such as 15", 18", 21" and the rare larger sizes. It also does not show the effects of two or more woofers working together; they can move a greater mass of air, resulting in lower frequency extension or perhaps merely greater output at a given distortion level. Furthermore, it does not include the narrowing of a woofer's polar pattern at the higher end of its frequency range, which is often a significant effect.

Tweeter

A shielded Peerless v-line dome tweeter

A **tweeter** is a loudspeaker designed to produce high 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 an amplifier to a coil. The electrified voice coil produces a varying magnetic field, which works against the fixed magnetic field, forcing the voice coil—and the diaphragm attached to it—to move. Since the coil is attached to a diaphragm, its motions become those of the diaphragm creating air motions, 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.

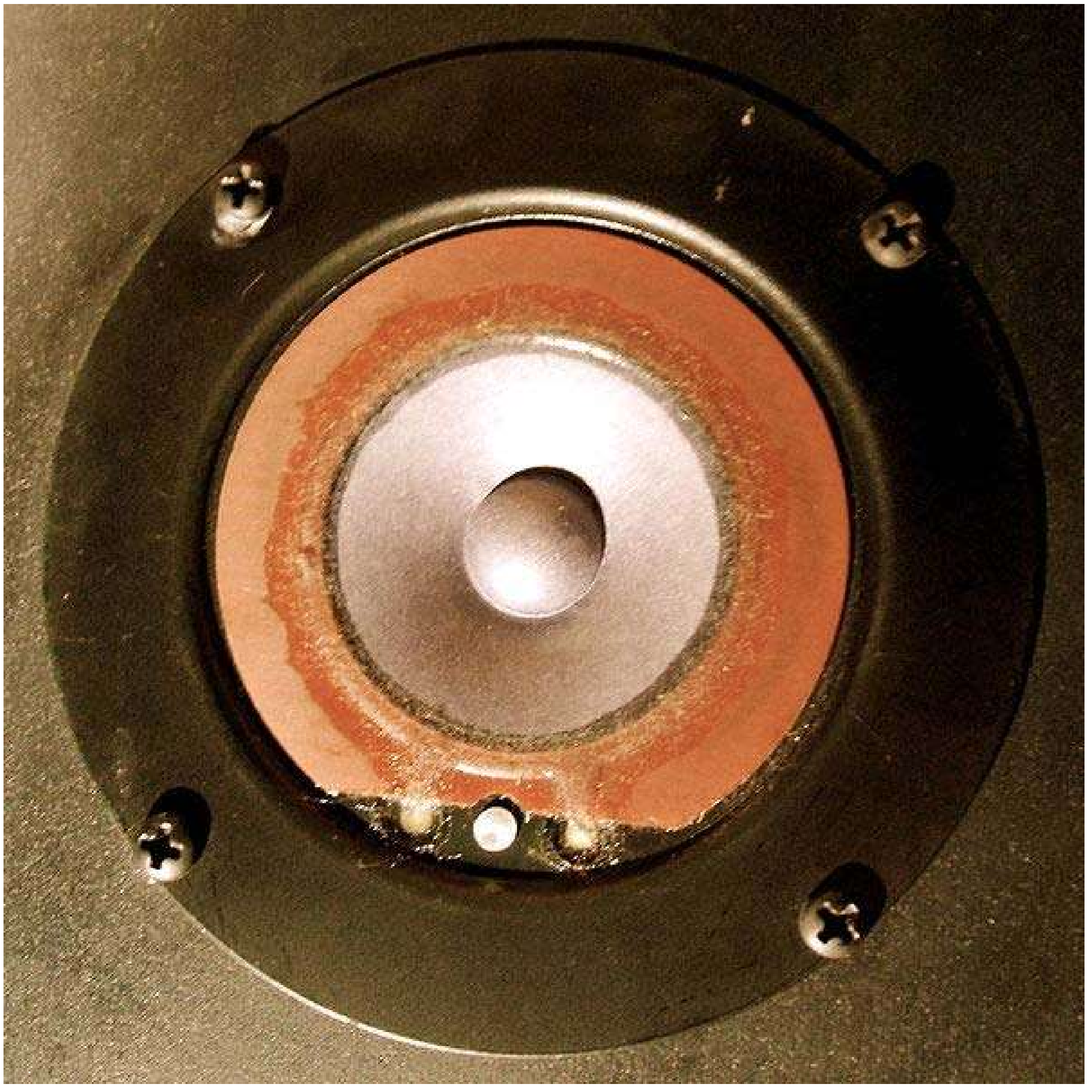
Professional sound applications

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 aluminum 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 further 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 (see above). 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.

Tweeter voice coils are not often rewound, as tweeters are usually not high price items.

Chapter- 7

Loudspeaker Enclosure

MTX Audio loudspeaker enclosures which can mount 15 inch woofers

A **loudspeaker enclosure** is a purpose-engineered cabinet in which speaker drivers and associated electronic hardware, such as crossover circuits and amplifiers, are mounted. Enclosures may range in design from simple, rectangular particle-board boxes to very complex cabinets that incorporate composite materials, internal baffles, ports and acoustic insulation.

The primary role of the enclosure is to prevent sound waves generated by the rearward-facing surface of the diaphragm of an open driver interacting with sound waves generated at the front of the driver. Because the forward- and rearward-generated sound is out of phase with each other, any interaction between the two in the listening space creates a distortion of the original signal as it was intended to be reproduced. Additionally, because they would travel different paths through the listening space, the sound waves would arrive at the listener's position at slightly different times, introducing echo and reverberation effects not part of the original sound.

The enclosure also plays a role in managing vibration induced by the driver frame and moving airmass within the enclosure, as well as heat generated by driver voice coils and amplifiers (especially where woofers and subwoofers are concerned). Sometimes considered part of the enclosure, the base may include specially designed "feet" to decouple the speaker from the floor.

History

Before the 1950s many manufacturers did not fully enclose their loudspeaker cabinets; the back of the cabinet was typically left open. This was done for several reasons, not least because electronics (at that time tube equipment) could be placed inside and cooled by convection in the open enclosure. Early on, it was observed that the enclosure had a strong effect on the bass response of the speaker. Since the rear of the loudspeaker radiates sound out of phase from the front, there will be constructive and destructive interference for loudspeakers without enclosures, and below frequencies related to the baffle dimensions in open-baffled loudspeakers. This causes loss of bass and comb filtering (i.e. response peaks and dips in power regardless of the signal meant to be reproduced). Most of the enclosure types discussed here were invented either to wall off the out of phase sound from one side of the driver, or to modify it so that it could be used to enhance the sound produced from the other side.

Background

Medium density fiberboard is a common material out of which loudspeaker enclosures are built.

In some respects, the ideal mounting for a low-frequency loudspeaker driver would be a rigid flat panel of infinite size with infinite space behind it. This would entirely prevent the rear sound waves from interfering (i.e., comb filter cancellations) with the sound waves from the front. An "open baffle" loudspeaker is an approximation of this, since the driver is mounted on a panel, with dimensions comparable to the lowest wavelength to be reproduced. In either case, the driver would need a relatively stiff suspension to provide the restoring force which might have been provided at low frequencies by a smaller sealed or ported enclosure, so few drivers are suitable for this kind of mounting.

The forward- and rearward-generated sounds of a speaker driver appear out-of-phase from each other because they are generated through opposite motion of the diaphragm and because they travel different paths before converging at the listener's position. A speaker driver mounted on a finite baffle will display a physical phenomena known as interference which can result in a perceivable frequency-dependent sound attenuation. This phenomenon is particularly noticeable at low frequencies where the wavelengths are large enough that interference will affect the entire listening area.

Since infinite baffles are impractical and finite baffles tend to suffer poor response as wavelengths approach the dimensions of the baffle (i.e. at lower frequencies), most loudspeaker cabinets use some sort of structure (usually a box) to contain the out of phase sound energy. The box is typically made of wood, wood composite, or more recently plastic, for reasons of ease of construction and appearance. Stone, concrete, plaster, and even building structures have also been used.

Enclosures can have a significant effect beyond what was intended, with panel resonances, diffraction from cabinet edges and standing wave energy from internal reflection/reinforcement modes being among the possible problems. Bothersome resonances can be reduced by increasing enclosure mass or rigidity, by increasing the damping of enclosure walls or wall/surface treatment combinations, by adding stiff cross bracing, or by adding internal absorption. Wharfedale, in some designs, reduced panel resonance by using two wooden cabinets (one inside the other) with the space between filled with sand. Home experimenters have even designed speakers built from concrete, granite and other exotic materials for similar reasons.

Many diffraction problems, above the lower frequencies, can be alleviated by the shape of the enclosure, such as by avoiding sharp corners on the front of the enclosure. Research experiments from the 1930s by Dr. Harry F. Olson showed that curved loudspeaker baffles reduce some response deviations due to sound wave diffraction, although his research did not show that careful placement of a speaker on even a sharp-edged baffle can reduce diffraction-caused response problems; this was discovered later. Sometimes the differences in phase response at frequencies shared by different drivers can be addressed by adjusting the vertical location of the smaller drivers (usually backwards), or by leaning or 'stepping' the front baffle, so that the wavefront from all drivers is coherent at and around the crossover frequencies in the speaker's normal sound field. The acoustic center of the driver dictates the amount of rearward offset needed to "time-align" the drivers.

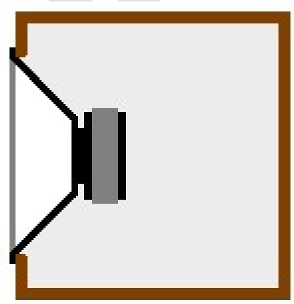
Woofer and subwoofer enclosures

A car audio system in a vehicle; Pyle Audio 18 inch sub-woofers

Enclosures used for woofers and subwoofers can be adequately modeled in the low-frequency region (approximately 100–200 Hz and below) using acoustics and the lumped component models. Electrical filter theory has been used with considerable success for some enclosure types. For the purposes of this type of analysis, each enclosure must be classified according to a specific topology. The designer must balance low bass extension, linear frequency response, efficiency, distortion, loudness and enclosure size, while simultaneously addressing issues higher in the audible frequency range such as diffraction from enclosure edges, the baffle step effect when wavelengths approach enclosure dimensions, crossovers, and driver blending.

Sealed (or closed) enclosures

A box stuffed with fiberglass insulation to increase the perceived volume of the box



Sealed enclosure

The loudspeaker driver's moving mass and compliance (stiffness of the suspension) determines the driver's resonant frequency. In combination with the damping properties of the system (both mechanical and electrical) all these factors affect the low-frequency response of sealed-box systems. Output falls below the system's resonant frequency (F_s), defined as the frequency of peak impedance. In a closed-box, the air inside the box acts as a spring, returning the cone to the 'zero' position in the absence of a signal. There may be an increase in output at the resonant frequency. The enclosure may be entirely empty, lined, filled or packed tightly with damping material, generally polyester foam, bonded acetate fibre (BAF) also known as pillow stuffing, fibreglass, or long fibre wool, which absorbs internal reflections, and changes the thermodynamic properties of the enclosed air mass from adiabatic to isothermal, making the enclosure behave as though it were slightly larger. The enclosure or driver must have a small leak so internal and external pressures can equalise over time, to compensate for barometric pressure or altitude. The porous nature of paper cones is sufficient to provide this slow pressure equalisation.

Infinite baffle

A variation on the 'open baffle' approach is to mount the loudspeaker driver in a very large sealed enclosure, providing minimal 'air spring' restoring force to the cone. This minimizes the change in the driver's resonant frequency caused by the enclosure. Some infinite baffle 'enclosures' have used an adjoining room, basement, or a closet or attic. This is often the case with exotic rotary woofer installations as they are intended to go to frequencies lower than 20 Hertz and displace large volumes of air.

Acoustic suspension

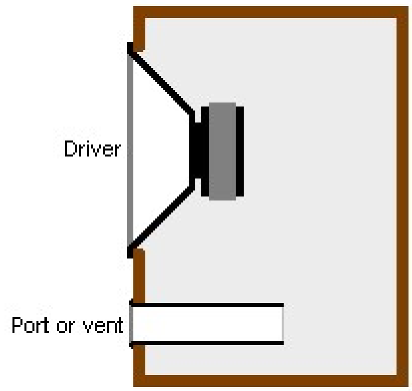
Small sealed box with Boss Audio subwoofer

A variation of the closed-box enclosure, using a smaller box to exploit the almost linear air spring which results. The "spring" suspension that restores the cone to a neutral position is a combination of an exceptionally compliant (soft) woofer suspension, and the air inside the enclosure. At frequencies below system resonance, the air pressure caused by the cone motion is the dominant force. Although no longer popular in commercial designs, the acoustic suspension principal takes advantage of this relatively linear spring.

The enhanced suspension linearity of this type of system is off-set by rather low efficiency. Drivers for these designs rely more upon the enclosure characteristics than typical drivers, and most modern woofers are not well suited to acoustic suspension use.

Ported (or reflex) enclosures

Bass-reflex



Bass reflex enclosure schematic (cross-section)

RCA Bass Reflex shelf stereo speakers

A **Bass reflex** system (also known as a **ported**, **vented box** or **reflex port**) is a type of loudspeaker enclosure that uses the sound from the rear side of the diaphragm to increase the efficiency of the system at low frequencies as compared to a typical closed box loudspeaker or an infinite baffle mounting.

A *reflex port* is the distinctive feature of a very popular enclosure variety. The design approach enhances the reproduction of the lowest frequencies generated by the woofer. The port generally consists of one or more tubes mounted in the front (baffle) or rear face of the enclosure. Depending on the exact relationship between driver parameters, the enclosure volume (and filling if any), and the tube cross-section and length, the low frequency limit or efficiency can be substantially improved over the performance of a similarly sized sealed box enclosure.

Though helpful with extending bass performance, bass reflex cabinets can have poor transient response compared to sealed enclosures at frequencies near the lower limit of performance. Proper adjustment of the cabinet and port size, and matching with driver characteristics can reduce much of this problem.

Achieving a balanced bass reproduction from a sealed box is simpler than properly aligning the components of a bass reflex system, and requires less effort expended in corrections to quality control variations of the components.

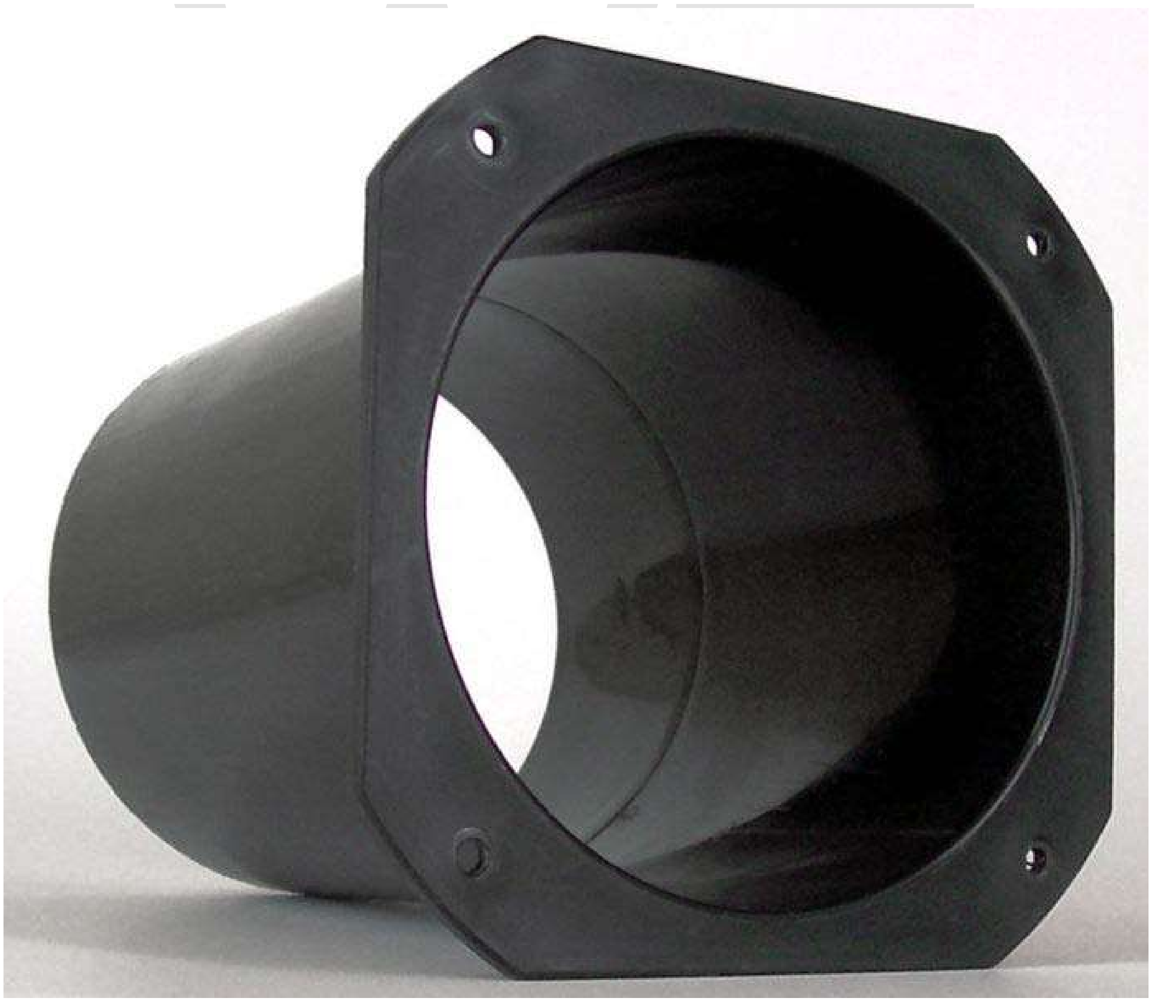
Explanation

Unlike closed box loudspeakers, which are substantially airtight, a bass reflex system has an opening called a *port* or *vent*, generally consisting of a pipe or duct (typically circular or rectangular cross section). The air mass in this opening resonates with the "springiness" of the air inside the enclosure in exactly the same fashion as the air in a bottle resonates when a current of air is directed across the opening. The frequency at which the box/port system resonates, known as the Helmholtz resonance, depends upon the effective length and cross sectional area of the duct, the internal volume of the enclosure, and the speed of sound in air. However, if the driver's optimal frequency range is above the bass reflex tuning frequency the driver will act as if loaded by a closed box.

When speakers are designed for home use, manufacturers often consider the advantages of porting to outweigh disadvantages. The design is popular among consumers and manufacturers (speakers cabinets can be smaller and lighter, for more or less equivalent performance) but the increase in bass output requires close matching of driver, the enclosure, and port. Poorly matched reflex designs can have unfortunate characteristics, making them unsuitable for settings requiring high accuracy and neutrality of sound, e.g. in monitoring facilities, recording studios etc.

History

The effect of the various speaker parameters, enclosure sizes and port (and duct) dimensions on the performance of bass reflex systems was not well understood until the early 1960s. At that time, pioneering analyses by A.N. Thiele and Richard H. Small related these factors in a series of "alignments" (sets of the relevant speaker parameters) that produced useful, predictable responses. These made it possible for speaker manufacturers to design speakers to match various sizes of enclosures, and to match enclosures to given speakers with great predictability. All of this is constrained by the laws of physics, which is covered in detail in Thiele and Small's work. It is not possible to have a small speaker in a small enclosure producing extended bass response at high efficiencies (i.e., requiring only a low-powered amplifier). It's possible to have two of these attributes, but not all; this has been termed Hoffman's Iron Law after J. Anton Hoffman of KLH. The sound pressure produced depends upon the efficiency of the speaker, the mechanical or thermal power handling of the driver, the power input and the size of the driver.



Bass reflex tube

Advantages

Such a resonant system augments the bass response of the driver and, if designed properly, can extend the frequency response of the driver/enclosure combination to below the range the driver would reproduce in a similarly sized sealed box. The enclosure resonance has a secondary benefit in that it limits cone movement in a band of frequencies centered around the tuning frequency, reducing distortion in that frequency range.

Limitations

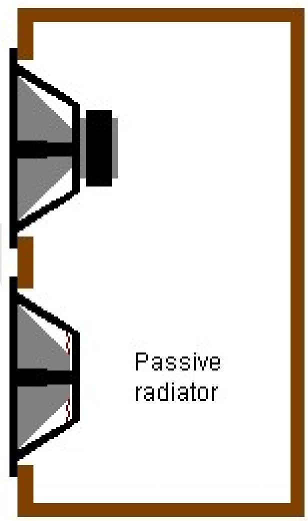
By their own nature, resonant systems cannot start and stop instantly. Ported speakers stagger *two* resonances, one from the driver and boxed air and another from the boxed air and port, in order to achieve their bass output, a more complex case than an equivalent sealed box. This causes increased time delay (increased group delay imposed by the twin resonances), both in the commencement of bass output and in its cessation. Therefore a flat steady-state bass response does not occur at the same time as the rest of the sonic output; rather, it starts later (lags) and accumulates over time as a longish resonant "tail". Because of this complex, frequency-dependent loading, ported enclosures generally result in poorer transient response at low frequencies than in well-designed sealed box systems.

Another trade-off for this augmentation is that, at frequencies below 'tuning', the port unloads the cone and allows it to move much as if the speaker were not in an enclosure at all. This means the speaker can be driven past safe mechanical limits at frequencies below the tuning frequency with much less power than in an equivalently sized sealed enclosure. For this reason, high-powered systems using a bass reflex design are often protected by a filter that removes signals below a certain frequency. One such filter is the rumble filter often built in to receivers or amplifiers designed to be used with LP records because of undesired LF rumble from the mechanical parts of the turntable, or from the strong subsonic excitation caused by warped vinyl discs. Unfortunately, electrical filtering adds further frequency-dependent group delay. Even if such filtering can be adjusted not to remove musical content, it may interfere with sonic information connected with the size and ambiance of the recording venue, information which often exists in the low bass spectrum.

Whether or not the effects of these in a properly designed system are audible remains a matter of debate. Many people prefer (or are used to) this kind of overindulgent bass, especially if they live in buildings that absorb LF energy (such as typical dry-wall construction housing). A poorly designed bass reflex system, generally one that is tuned too high or too loosely, can ring at the tuning frequency and create a 'booming' one-note quality to the bass frequencies. In effect, this is due to the port resonance imposing its characteristics to the note being played, and is grossly exacerbated if the port resonance coincides with one of the resonant modes of the room, a not unusual occurrence. In general, the lower in frequency a port is tuned, the less objectionable these problems are likely to be.

Ports often are placed in the front baffle, and may thus transmit unwanted midrange frequencies reflected from within the box. If poorly designed, a port may also generate "wind noise" or "chuffing", due to turbulence around the port openings at high air speeds. Enclosures with a rear facing port mask these effects to some extent, but they cannot be placed directly against a wall without causing audible problems. They require some free space around the port so they can perform as intended. Poor room placement can significantly reduce the performance of this type of loudspeaker. Some manufacturers incorporate a floor-facing port within the speaker stand or base, offering predictable and repeatable port performance within the design constraints.

Passive radiator

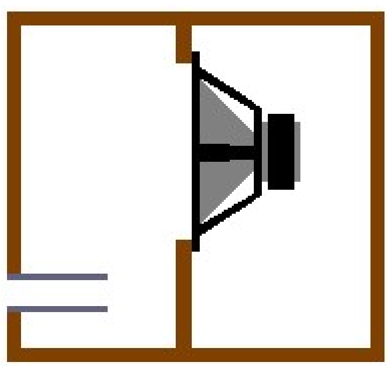


Passive radiator enclosure

A passive radiator speaker uses a second "passive" driver, or drone, to produce similar low-frequency extension, or efficiency increase, or enclosure size reduction, similar to ported enclosures. The passive driver is not wired to an amplifier; instead, it moves in response to changing enclosure pressures. In theory, such designs are variations of the bass reflex type, but with the advantage of avoiding a relatively small port or tube through which air moves, sometimes noisily. Tuning adjustments for a passive radiator are usually accomplished more quickly than with a bass reflex design since such corrections can be as simple as mass adjustments to the drone. The disadvantages are that a passive radiator requires precision construction quite like a driver, thus increasing costs, and has excursion limitations.

Compound or band-pass

Aiwa 4th order bandpass subwoofer



Compound or 4th order band-pass enclosure

A 4th order electrical bandpass filter can be simulated by a vented box in which the contribution from the rear face of the driver cone is trapped in a sealed box, and the radiation from the front surface of the cone is into a ported chamber. This modifies the resonance of the driver. In its simplest form a compound enclosure has two chambers. The dividing wall between the chambers holds the driver; typically only one chamber is ported.

If the enclosure on each side of the woofer has a port in it then the enclosure yields a 6th order band-pass response. These are considerably harder to design and tend to be very sensitive to driver characteristics. As in other reflex enclosures, the ports may generally be replaced by passive radiators if desired. An eighth order bandpass box is another variation which also has a narrow frequency range. They are often used to achieve sound pressure levels in which case a bass tone of a specific frequency would be used versus anything musical. They are complicated to build and must be done quite precisely in order to perform nearly as intended.

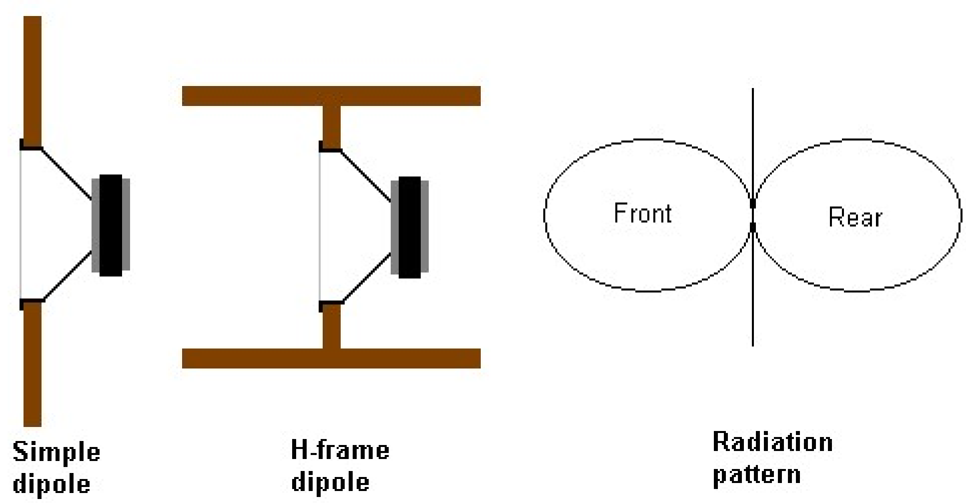
Aperiodic enclosures

Marantz HD77 aperiodic enclosure; note removable foam cylinder allowing either ported/bass reflex or aperiodic properties

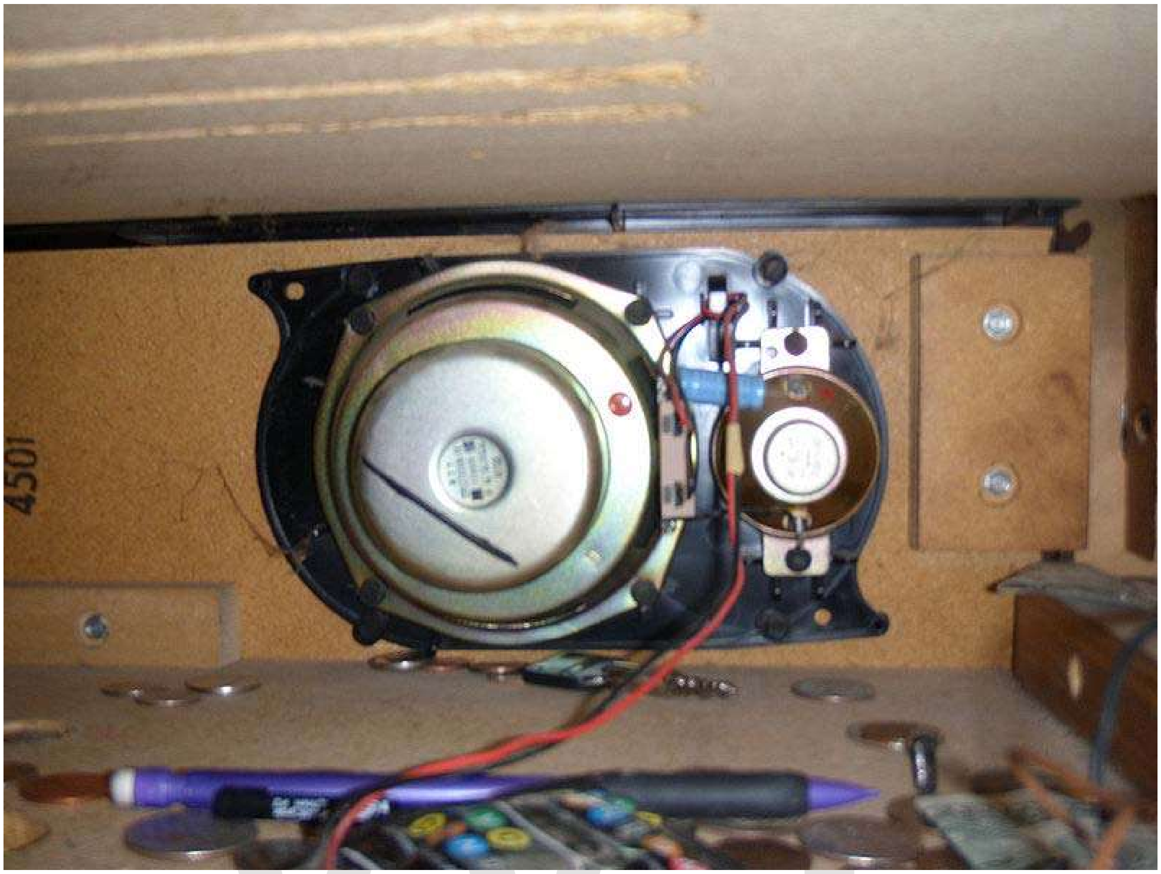
This design falls between acoustic suspension and bass reflex enclosures. It can be thought of as either a leaky sealed box or a ported box with large amounts of port damping. By setting up a port, and then blocking it precisely with sufficiently tightly packed fiber filling, it's possible to adjust the damping in the port as desired. The result is control of the resonance behavior of the system which improves low-frequency reproduction, according to some designers (and driver manufacturers). Dynaco was a primary producer of these enclosures for many years, using designs developed by a Danish driver maker. The design remains uncommon among commercial designs currently available. A reason for this may be that adding damping material reduces the efficiency of the system; the same alignment can be achieved by simply choosing a loudspeaker driver with the appropriate parameters and precisely tuning the enclosure and port for the desired response.

A similar technique has been used in aftermarket car audio; it is called "aperiodic membrane" (AP). A resistive mat is placed in front of or directly behind the loudspeaker driver (usually mounted on the rear deck of the car in order to use the trunk as an enclosure). The loudspeaker driver is sealed to the mat so that all acoustic output in one direction must pass through the mat. This increases mechanical damping, and the resulting decrease in the impedance magnitude at resonance is generally the desired effect, though there is no perceived or objective benefit to this. Again, this technique reduces efficiency and the same result can be achieved through selection of a driver with a lower Q factor, or even via electronic equalization. This is reinforced by the purveyors of AP membranes; they are often sold with an electronic processor which, via equalization, restores the bass output lost through the mechanical damping. The effect of the equalization is opposite to that of the AP membrane, resulting in a loss of damping and an effective response similar to that of the loudspeaker without the aperiodic membrane and electronic processor.

Dipole enclosures



Dipole speakers and their radiation pattern



RCA Dimensia speaker mounted on a baffle with no back

A **dipole speaker** enclosure in its simplest form is constructed by mounting a loudspeaker driver on a flat panel. The panel may be folded to conserve space.

The term dipole derives from the fact that the polar response consists of two lobes, with equal radiation forwards and backwards, and none perpendicular to the axis. This can be useful in reducing the stimulation of resonant room modes at low frequencies. It also results in high frequencies being reflected from any rear wall, which can enhance the naturalness of the sound in typical listening rooms by creating more diffuse reverberation, though in theory it could detract from stereo localization. For this reason dipole speakers are often used as surround channel speakers, where a diffuse sound is desired to create ambience.

A dipole speaker works by creating air movement (as sound pressure waves) directly from the front and back surfaces of the driver, rather than by impedance matching one or both outputs to the air. As a result, diaphragm motion is constrained primarily by the driver's restoring force (eg, diaphragm suspension) and not by acoustic loading from an enclosure. This implies that cone motion will be larger at the same output level than in a more usual enclosure, and that power handling will be accordingly limited. Especially at lower frequencies, dipole drivers tend to be large and flat, and necessarily open at both front and back. Common examples include electrostatic or ribbon drivers, though a conventional cone driver mounted in an open baffle also works as a dipole loudspeaker. All of these variations are characterised by a "figure-of-eight" radiation pattern in which the loudness falls towards the sides of the enclosure where interference between front and back waves is maximized. Sometimes the enclosure is modified into an "H-frame" with the driver located on a wall dividing two open compartments. Such enclosures require some control over the radiated sound from the rear of the enclosure to achieve the desired response. This is usually done by mounting two drivers one over the other in a push-pull configuration.

Some of the speaker cabinets referred to as dipolar arrange the two poles at an angle of about 90 degrees rather than 180 degrees, especially for wall-mounted rear speakers (e.g. Jamo C-80-SUR, Castle dipoles, and Theophany S4).

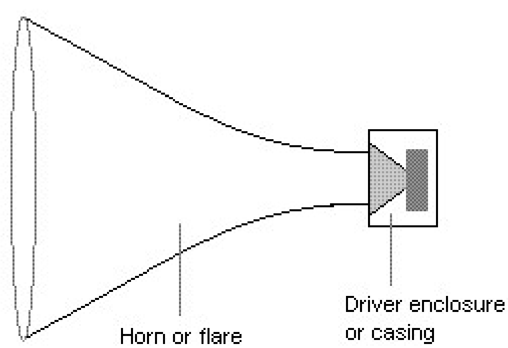
Advantages

An advantage of dipoles is that the sound is concentrated in the listening area due to the figure-of-eight polar response. This means that for a given output loudness, locations falling within the "dead-zone" of the speaker do not perceive as loud a sound as they would with more traditional enclosures. Dipole speakers are said to be better for rear speakers in surround-sound systems, although this is disputed (or at least success depends on factors such as room acoustics, type of music, and so on).

Disadvantages

However, these enclosures are less efficient because, for the same driver, a dipole results in less sound pressure level than a closed or ported enclosure, and certainly far less than a properly designed horn. This means that the drivers mounted on a dipole enclosure must have large maximum excursions, large square areas, or both.

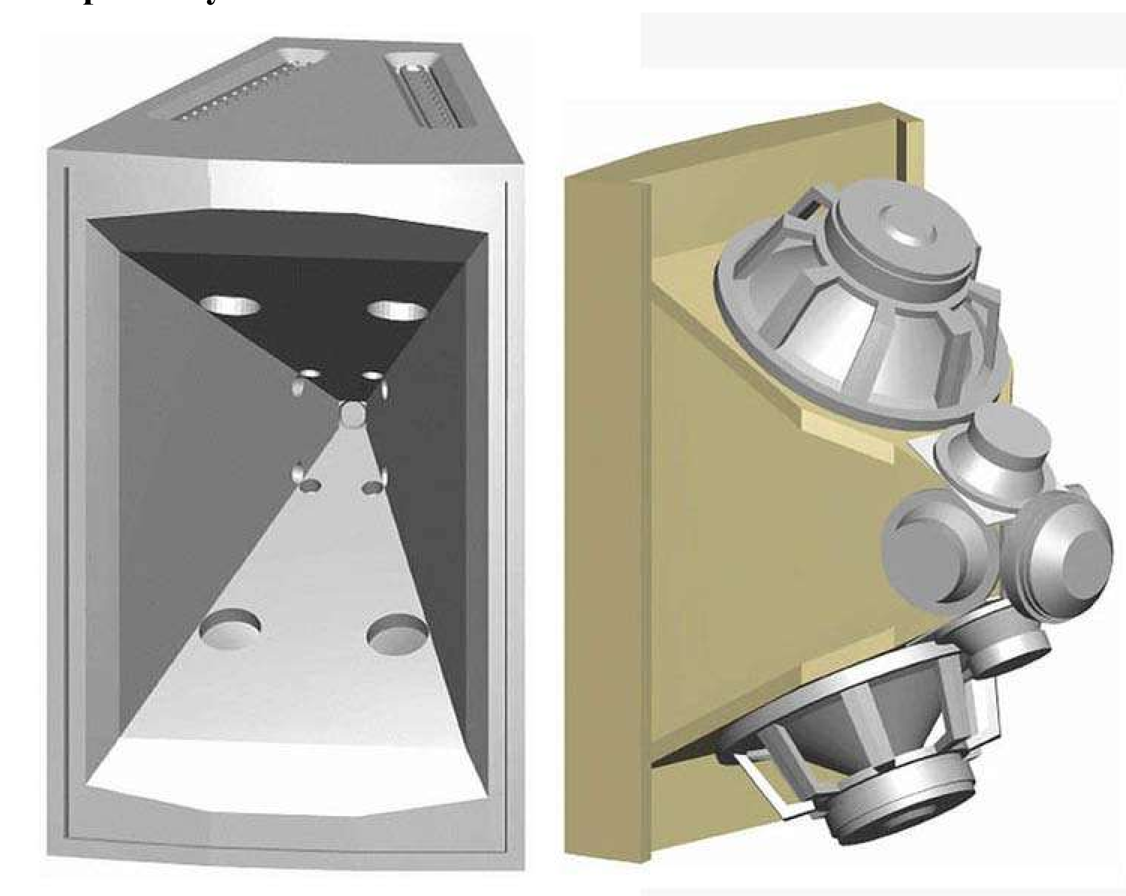
Horn enclosures



Horn loudspeaker schematic

A horn loudspeaker is a speaker system using a horn to match the driver cone to the air. The horn structure itself does not amplify, but rather improves the coupling between the speaker driver and the air. Properly designed horns have the effect of making the speaker cone transfer more of the electrical energy in the voice coil into the air; in effect the driver appears to have a larger cone surface area than it actually does. Horns can help control dispersion at higher frequencies which is useful in some applications such as sound reinforcement. The mathematical theory of horn coupling is well developed and understood, though implementation is sometimes difficult. Properly designed horns for high frequencies are small (above say 3 kHz or so, a few centimetres (a few inches)), those for mid-range frequencies (perhaps 300 Hz to 2 kHz) much larger, perhaps 30 to 60 cm (1 or 2 feet), and for low frequencies (under 300 Hz) very large, a few metres (dozens of feet). In the 1950s, a few high fidelity enthusiasts actually built full sized horns whose structures were built into a house wall or basement. With the coming of stereo (two speakers) and surround sound (four or more), plain horns became even more impractical. Various speaker manufacturers have produced folded low-frequency horns which are much smaller (e.g., Altec Lansing, JBL, Klipsch, Lowther, Tannoy) and actually fit in practical rooms. These are necessarily compromises, and because they are physically complex, they are expensive.

Multiple entry horn



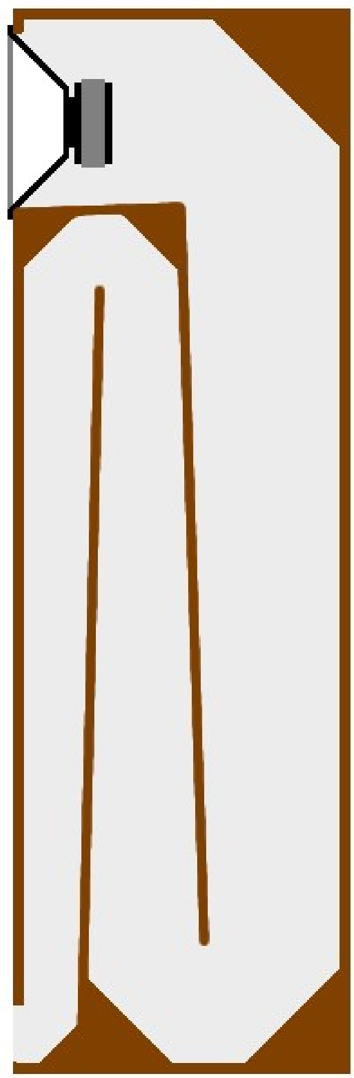
Multiple entry horn

The multiple entry horn (also known as a *coentrant horn*, *unity horn* or *synergy horn*) uses several drivers mounted on the horn at stepped distances from the horn's apex, where the high frequency driver is placed. Depending on implementation, this design offers an improvement in transient response as each of the drivers is aligned in phase and time and exits the same horn mouth. A more uniform radiation pattern throughout the frequency range is also possible. A uniform pattern allows smooth arraying of multiple enclosures.

Tapped horn

Both sides of a long-excursion high-power driver in a tapped horn enclosure are ported into the horn itself, with one path length long and the other short. These two paths combine in phase at the horn's mouth within the frequency range of interest. This new design is especially effective at subwoofer frequencies and offers reductions in enclosure size along with more output.

Transmission line



Transmission line enclosure

A perfect transmission line enclosure has an infinitely long line, stuffed with absorbent material such that all the rear radiation of the driver is fully absorbed, down to the lowest frequencies. Theoretically, the vent at the far end could be closed or open with no difference in performance. The density of and material used for the stuffing is critical, as too much stuffing will cause reflections due to back-pressure, whilst insufficient stuffing will allow sound to pass through to the vent. Stuffing often is of different materials and densities close to the cone, and changes as one gets further from the cone.

Consequent to the above, practical Transmission Line loudspeakers are not true Transmission Lines, as there is generally output from the vent at the lowest frequencies. They can be thought of as a waveguide in which the structure shifts the phase of the driver's rear output by at least 90°, thereby reinforcing the frequencies near the driver's F_s. Transmission lines tend to be larger than ported enclosures of approximately comparable performance, due to the size and length of the guide required (typically 1/4th the longest wavelength of interest).

The design is often described as non-resonant, and some designs are sufficiently stuffed with absorbent material that there is indeed not much output from the line's port. But it is the inherent resonance (typically at 1/4 wavelength) that can enhance the bass response in this type of enclosure, albeit with less absorbent stuffing. Among the first examples of this enclosure design approach were the projects published in *Wireless World* by Bailey in the early 1970s, and the commercial designs of the now defunct IMF Electronics which received critical acclaim at about the same time.

A variation on the transmission line enclosure uses a tapered tube, with the terminus (opening/port) having a smaller area than the throat. The tapering tube can be coiled for lower frequency driver enclosures to reduce the dimensions of the speaker system, resulting in a seashell like appearance. Bose uses similar patented technology on their Wave and Acoustic Waveguide music systems. Bowers & Wilkins have used this approach in their flagship Nautilus speaker as well as smaller straight tapering tubes in many of their other lines.

Numerical simulations by George L. Augspurger and Martin J. King have helped refine the theory and practical design of these systems.

Tapered quarter-wave pipe

The *tapered quarter-wave pipe* (TQWP) is an example of a combination of transmission line and horn effects. It is highly regarded by some speaker designers. The concept is that the sound emitted from the rear of the loudspeaker driver is progressively reflected and absorbed along the length of the tapering tube, almost completely preventing internally reflected sound being retransmitted through the cone of the loudspeaker. The lower part of the pipe acts as a horn while the top can be visualised as an extended compression chamber. The entire pipe can also be seen as a tapered transmission line in inverted form. (A traditional tapered transmission line, confusingly also sometimes referred to as a TQWP, has a smaller mouth area than throat area.) Its relatively low adoption in commercial speakers can mostly be attributed to the large resulting dimensions of the speaker produced and the expense of manufacturing a rigid tapering tube. The TQWP is also known as a Voigt pipe and was introduced in 1934 by Paul G. A. H. Voigt, Lowther's original driver designer.