LOUDSPEAKER

HANDBOOK

AND

LEXICON

by
Winslow Burhoe
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ART OR SCIENCE?

This handbook has been prepared to take just a touch of the mystery out of the subject of loudspeaker design and performance, with an eye out to the characteristics that make a loudspeaker sound good (or bad).

Since this is a personal statement of one designer, who necessarily is in business for profits as well as enjoyment, the handbook makes no claim to being either objective or comprehensive.

For the reader's convenience, the handbook has three related, but different, sections:

1: A general introduction which tersely ties together some fundamental concepts and sets forth the problem;

2: An expansion of some of the key concepts of speaker design and speaker construction; and

3: A lexicon of commonly encountered speaker terms and vocabulary.

Throughout this document, any term defined in the Lexicon is in bold italic type.

Hoping you'll have a bit of fun with this handbook, even if it may seem to raise more questions than it answers. . .

Dedicated to Edgar Villchur,

who invented not only most of

the key innovations of modern

high fidelity speakers, but

also the techniques of

popularizing the scientific

aspect of high fidelity.

PART ONE

SPEAKER DESIGN

At first glimpse, speaker design tends to look relatively simple and uncomplicated: the designer seems to have little more to do than select the right combination of drivers and electrical components, mount them in a cabinet which will enhance the sound they create together, and then send the results down to the production department.

However, technically speaking, the task at hand is converting electrical waveforms from an amplifier into acoustical, three-dimensional waves. The challenge is to accomplish this with the least possible distortion. Ideally there should be none. Usually there's quite a lot. And that's just the first problem.

The designer is challenged by the materials he must work with. The woofer, from which room-filling bass must come, in reality is generally a fragile paper cone with unpredictable minor movements, pushed by a wound wire moving in the gap of a MAGNET whose magnetic field should be uniform but seldom can be. The tweeter, on which the highest audible frequencies depend, is generally a tiny object called upon to perform the virtually impossible task of emitting sound identical to that which comes from large musical instruments with vast radiating surfaces. And, dividing sounds between woofer and tweeter is the job of an electronic or mechanical CROSSOVER which cannot, by definition, do the job perfectly.

Then there's the problem of inefficiency. Almost all loudspeakers are notoriously inefficient, producing far more heat than music. Improving loudspeaker efficiency is certainly part of the goal. Large drivers, large magnets, and large enclosures help. However, improvements in efficiency are seldom free. They are usually paid for by narrowing of frequency response (bandwidth). (Bandwidth is enhanced by small drivers and long voice coils; both are enemies of efficiency.) Then there's the problem of size versus performance. Large boxes give better bass efficiency and power output, but add to the cost of the speaker. Smaller boxes provide better dispersion and frequency response, but without the realism. Designers must therefore constantly compromise between cost and size in commercial designs.

Not only must the designer compromise between size and cost, bandwidth and efficiency. Each time he designs a new speaker, there are scores of factors to be juggled. Theory and math can only be a loose framework. Pure theory has its limits, first, because the rooms in which speakers will eventually be used can only be guessed at, yet will influence frequency response and dispersion, and also because psycho-acoustic factors (how and what the ear hears, how it selects and interprets) have a strong influence on the final sound or "response" of the speaker.

In its painstaking advance towards less and less distortion, successful speaker design requires of its practitioners a thorough grasp of how sound behaves and how the ear hears, but that's only one prerequisite. A second qualification for success may be and attic or basement or perhaps a back room full of speakers of all sizes and shapes, all unique prototypes, each built with care and the hope that this particular combination of design considerations may be the designer's finest speaker yet. Some designers seem to build one fine speaker after another, yet often the real secrets of their successful and famous designs derive from many unfamous and even unnamed prototypes representing years of effort but now practically forgotten.

Hopefully what the designer learned from them is not, so that when you choose a speaker by a good designer, you're getting more than an enclosure and electronics. You're also getting a unique blend of design factors and the designer's experience: a share of the important things learned among the prototypes in the back room.

PART TWO

The Basics

FREQUENCY RESPONSE

The most important thing about any speaker is its frequency response. It is here that designers spend most of their effort, for the wider the frequency response (the higher and lower the musical tones it can reproduce), the more you hear.

Various frequencies of vibration generate various pitches. For example, middle C on a piano creates a vibration or sound wave or 256 cycles per second.

A good example of high frequency sound is the almost inaudible (for many people) hiss or whistle given off by the flyback transformer of a television set, which vibrates at 15,750 cycles per second in order to generate the 15,570 lines per second displayed on the TV screen. If you can't hear this on your own set, try listening in a TV showroom. This is a good test to try out the top end of your hearing, for 15,750 cycles per second is near or past the top limit for most people's hearing.

WOOFERS AND TWEETERS

The search for speakers with the widest possible frequency response led to the specialization of drivers as woofers and tweeters. Bass or low frequency response depends on a speaker moving large quantities of air, and therefore generally requires a speaker cone of large size and area (and also a long voice coil).

A woofer becomes more efficient as frequency rises. This is because higher frequencies have shorter wave-lengths, closer to the size of the woofer cone itself, resulting in more effective coupling of the woofer cone and the air.

Generating high frequency sounds is a totally different problem, requiring very little mass and size . Usually, the smaller the tweeter, the higher its frequency response. A large tweeter will play more loudly, but with less range and relatively poorer accuracy at the higher frequencies.

VOICE COIL

One of the moving parts and hence moving mass of a dynamic speaker or driver. The design and shape of a voice coil is a critical aspect of speaker design. The coil must not only perform perfectly as an electrical component, accepting and converting current from the amplifier into mechanical energy; but through its length and its mass it also influences the physical behavior of the speaker.

Low frequencies require a relatively heavy long voice coil.

Initial design, winding accuracy, quality of material and workmanship and precise quality control are all factors in voice coil production.

DRIVER

A speaker. Since the term "speaker" refers both to an individual component and to a system of several speakers, the word "driver", which refers only to a single element, is often used to avoid ambiguity. Examples of drivers include a woofer, a mid-range, a tweeter, a horn, an electrostatic unit.

CROSSOVER

In speaker design, a mechanical or electrical device to split a speaker's input signal into high-frequency and low-frequency components. In a two-way speaker system, the crossover is essentially two filters: an inductor which allows only low frequencies to reach the woofer, and a capacitor which allows only high frequencies to reach the tweeter.

More complex crossovers have combinations of coils, capacitors, resistors and other components. The use of more crossover components allows the designer either to make the crossover slopes steeper (so that there is less overlapping of the sound from the two drivers) or to shape the frequency response as desired to improve the frequency response of the drivers.

CROSSOVER POINT

In a two-way system, the low end of the tweeter's frequency response and the high end of the woofer's frequency response. The tweeter's low end therefore should be just about the same as the high end of the woofer. One advantage to a speaker company which manufactures its own drivers, rather than buying them ready made, is that the designer can precisely control frequency response through precise variations in such factors as coil mass and cone shape.

EFFICIENCY

Speaker efficiency is the ratio of sound energy to power input. The range of efficiency in speakers runs from about one third or a percent to about ten percent. Very few speakers have better than ten percent efficiency, and most run about one percent.

The loss or wasted energy is mostly heat. In this sense, a speaker is essentially a heater which produces music as a by-product, which is why speakers occasionally burn out or blow up. The trend towards very high-powered amplifiers dramatically increases the possibility of such mishaps.

A general rule of thumb is that more efficiency in a loudspeaker actually means a narrower frequency response. Consider the speaker as analogous to a filter and the reasons become clear. Any filter allows only a certain bandwidth to pass through. A narrow band has higher amplitude. If the filter is designed to allow a wide-range signal to pass, there will necessarily be less amplitude. In this sense a speaker can be perfectly represented by symbols which are the same as electronic symbols. (Also see THIELE.)

DISPERSION

Uniform dispersion of sound exists when you can sit anywhere around a speaker and hear the same sound. Dispersion is an important challenge in speaker design. Ideally a speaker should produce a spherical sound wave at all frequencies. In reality both woofers and tweeters begin to beam sound as frequency gets higher.

The smaller a speaker, the higher the frequency up to which it gives uniform dispersion (spherical waveform). For example, a one-inch tweeter is able to give spherical dispersion up to about 12,000 or 13,000 Hz. A half-inch tweeter is good to nearly 20,000 Hz. A woofer usually begins to beam at about the range of the female voice (800-900·Hz).

Most of the sound that you hear is reflected off the wall. If there's a frequency range where dispersion is poor, then you don't hear the reflected sound wave and the overall power response is off in that frequency range.

Because of the difficulty in obtaing drivers that provide ideal dispersion, some designers try to take advantage of the limited dispersion to enhance some other aspect of the performance, such as, stereo imaging or absence of microphone feedback.

DISTORTION

When light goes through a lens, a certain amount of fuzziness is always introduced to the original image. When an electrical sound-wave enters a loudspeaker, a certain amount of fuzziness or distortion is always introduced while converting the electrical energy into sound energy.

This distortion comes about because the motion of the speaker is never exactly the same as the input. When the voice coil moves, the force behind the motion may not be uniform because the coil moves in relation to the stationary magnetic field and because opposition to the coil motion is not uniform because of the varying resistance and stiffness of the suspension

Ways of decreasing distortion are:

1. Longer voice coils

2. Uniform suspension elements which allow a lot of play (excursion).

3. Oversized elements. (Large drivers, long voice coils, large suspension elements.)

Distortion will occur: if the force from the motor driving the cone is non-linear, if parts of a cone move independently, if air is escaping in and out of small vents in the enclosure, if there is ringing from parts which should not vibrate, or if components emit their own resonant frequencies when excited by other frequencies.

Distortion also occurs when dispersion of the sound wave is uneven, or if the phase response is not uniform as a function of frequency.

If the magnetic field of the magnet is unequal on two sides, this will also cause distortion.

Since the room in which the speaker functions can also introduce distortion into the speaker's output, placement of speakers in the room, particularly bass reflex designs, can be crucial in improving sound.

Forms of distortion:

HARMONIC: easiest to measure

LIMITED BANDWIDTH: most important

INTER MODULATION OR IM: most unpleasant

TRANSIENT INTERMODULATION OR TIM

UNEVEN DISPERSION: difficult to hear

UNEVEN PHASE RESPONSE: difficult to hear

DOPPLER: difficult to measure

SPURIOUS BUZZES, RESONANCES OR VIBRATIONS: from poor construction

ENVIRONMENTAL EFFECTS: room acoustics.

Q

Q is the formula for what happens electrically and physically at resonant frequency. Q is related to how slowly a natural vibration dies down. Q is a neat term that combines many elements of speaker design, so it's an important term to speaker designers.

Q is important to the woofer because in a speaker system where the Q is too high, you'll get a whomp in the bass. If a speaker system's Q is too low, then the bass output is relatively weak, because frequency response is highly damped in that region.

The best Q for a speaker to have is one that gives flat response down to the resonant frequency.

If you do nothing to a speaker but make the cabinet smaller, you increase stiffness and therefore Q goes up. But Q is not a word for stiffness alone. It's a ration involving mass, stiffness, and resistance (or damping). Designers often juggle these three factors. Good bass is not enough. Good bass with uniform frequency response and spherical wave-form is the goal. Obtaining that goal without too much sacrifice in efficiency is the challenge.

EFFICIENCY AND MAGNETIC FORCE

In a dynamic speaker, the voice coil and the magnet structure interact to force the speaker to move. Send an electric current through a coil in or near a magnet and a force is produced. The force results in an alternating movement or motion which is converted to variations in air pressure by a cone. The pressure is radiated as a sound wave.

The stronger the magnetic field, the higher the force, the higher the efficiency. Bigger magnets often lead to stronger magnetic fields, but not always, because the strength is related, not only to magnet size, but also to the size and shape of the gap that the voice coil moves in.To get a more efficient speaker of the same size, you can use a stronger magnetic field. However, this gives you lower Q. You'll get more output at the middle frequencies, but less at the high and low ends. This can give relatively less bass even though you now have higher efficiency. Therefore, if you increase magnetic field, you have to go back and redesign the tuned port of the cabinet to get back to correct response.

CABINET DESIGN

Speaker cabinets have two jobs: they serve as a baffle for mounting of drivers, and serve to modify, re-phase, improve, radiate, suppress, or redirect the back wave from the woofer and keep it separate from the front wave.

As a baffle, the speaker cabinet is usually less than ideal. Since the cabinet sticks out from the wall, and has sharp edges and corners, acoustic problems develop as sound waves pass over the edges of the cabinet. One relatively effective response to this problem is a cabinet whose shape flow smoothly "into the wall" because of beveled or curved panels.

Cabinets should ideally be designed to avoid:

1. Motion of the walls of the cabinet.

2. Transmission of sound through the cabinet walls.

Therefore, the ideal cabinet is heavy and rigid. These considerations are particularly important in acoustic suspension designs, where the back wave the woofer must be completely "contained".

Today, almost all speaker cabinets are made from particle board. This material is particularly good because it has no spaces into which air can flow (unlike plywood), which often does). However, particle board is relative flexible, therefore failing to avoid condition 1 above, and often requires astute bracing and very solid construction techniques (screws, glue, reinforcement, etc.) in order to approach required rigidity.

There are many types of speaker designs where the cabinet design is integral to the speaker design. Some that come to mind are:

TRANSMISSION LINE

Isobaric

Omni-directional (Bi-polar)

BAND-PASS

Horns

Limited dispersion

PART THREE

The Lexicon

A

ABSOLUTE PHASE

In the recording and reproduction chain, there are many opportunities to reverse polarity, especially when hooking up speaker wire betwen the amplifier and the speaker. For most people there is no difference in the sound when the polarity is reversed. However, some audiophiles argue that there would be a difference in the sound if the polarity were to be changed for something like an explosion and if the reproduction could reproduce all the low frequencies in the sound. The sound of an explosion would have positive polarity, you see.

(See POLARITY, PHASE.)

ACOUSTIC FEEDBACK

If a portion of the sound output from loudspeakers can somehow excite the turntable or stylus of the system, a new, unwanted signal is created, which is in turn amplified by the electronics and reproduced by the speakers. If this signal is in phase with the original signal at a particular frequency (usually bass) a tremendous resonance (oscillation, a form of positive feedback) can occur in the room. Even small amounts of acoustic feedback can cause a considerable amount of distortion.

The careful positioning of speakers and turntable is the usual remedy to the problem. Placing the turntable anywhere near the speakers is usually a mistake.

An effect similar to acoustic feedback occurs when vibrations from walking in the room are transmitted up to the turntable from the floor. Generally if a turntable is positioned so that it is free from this kind of vibration, it will also be safe from acoustic feedback.

CD players and amplifiers are a lot less susceptible to acoustic feedback, but it does exist, and precautions sometimes help.

Another example of acoustic feedback (this time at high frequencies) is the squeal that occurs in microphone hookups where output from the speaker re-enters the system through the microphone.

No matter what the frequency, even if the gain of the amplifier is not high enough to cause oscillation, distortion in the form of a boost in response at certain frequencies will result from acoustic feedback. This can cause even the best speakers to sound "boomy". Ironically, the general recent improvement in low frequency response of quality systems has simply intensified the problem, so that acoustic feedback has become a major problem in quality sound installations.

"ACOUSTIC RESEARCH", or "AR, Inc."

This is the company founded by Ed Villchur (see Dedication) and Henry Kloss (later of KLH and Advent). Here began the invention of acoustic suspension and many other principles and techniques of modern speaker design.

ACOUSTIC SUSPENSION

Mounting a speaker in a hermetic or air-tight enclosure traps a fixed quantity of air behind it. The springiness of this air can effectively replace the suspension spring used in speakers to return the speaker cone to its "normal" position, hence the term "acoustic suspension" (either "air suspension" or "pneumatic suspension" would be more accurate). The ADVENT, the AR 3, and the EPI 100 are (old) examples of acoustic suspension speakers.

Acoustic suspension speakers are traditionally much less efficient than bass reflex speakers and therefore need more powerful amplifiers.

The major advantage of the design is simplicity and therefore simpler manufacturing and lower initial cost. However, any savings may be eaten up by the need for a more powerful amplifier. This is the hidden but unavoidable cost of inefficient speakers.

ANALOG

A method of reproducing sound waves that is analogous to how they are in air. Analog sound waves can be converted into digital information bits in an A to D converter.

(See DIGITAL)

B

BAFFLE

A baffle is a mounting for a speaker which assists sound radiation. A baffle is important because many sound wavelengths are longer than the speaker. The ideal baffle is flat and extends forever, creating a hemispherical sound wave.

The baffle is usually part of the cabinet. Sometimes special baffles are built for testing purposes.

When a small speaker is placed near the wall, the wall becomes part of its baffle.

The term infinite baffle is another way of describing the ideal baffle mentioned above. Actually, infinite baffle is often used to describe any situation where the back wave from the speaker is completely isolated from the front, as when a speaker is mounted in a hole in a wall.

BAND-PASS

A variation of the bass reflex design. A speaker enclosure is divided into two chambers with the woofer mounted between them. One of the chambers is then vented with a tuned port. The tuned vent acts like a filter which limits the frequency responce to a desired bandwidth, thus the term band-pass. This type of design is used primarily for sub-woofers and bass units of satellite systems.

(Also see FILTER)

BANDWIDTH

The range of frequencies in which amplitude remains constant. Bandwidth is generally the subject of much controversy and confusion, since there is no standard method of measurement. Should speaker bandwidth be measured in a living room or in an anechoic chamber? How far should the microphone be from the speaker: Should the mike record reflected sound? Etc.

bandwidth is another way of saying frequency response.

(See SPEAKER SPECIFICATIONS)

BASS REFLEX

A speaker in an infinite baffle or a sealed enclosure uses only its front wave to produce sound. In a bass reflex or tuned port design, part of the back wave of the speaker is used to reinforce the front wave usually in the bass.

The back wave excites a resonant system which then radiates sound. This resonant system has two components: mass (actually air in this case), and the stiffness of the air suspension in the box.

The resonant system using the backwave is far more efficient than the primary system. Very little power excites great sound from bass reflex systems in the frequency range to which the port is tuned (generally below 100 Hz).

The increased output obtained in a bass reflex design can be used by the designer either to improve efficiency or to extend low frequency response.

An advantage of bass reflex design over

acoustic suspension design is that the relatively higher efficiency of the former permits the same amount of bass to be produced with far less woofer travel or excursion. Since this generally increases linearity of the response, it produces less distortion, particularly in the low bass.

(Also see BANDWIDTH, THIELE, TUNED PORT)

BASS UNIT

A speaker designed for low frequencies only, usually for use in conjunction with two or more satellite speakers.

A bass unit is different from a sub-woofer in that it does not claim full bass coverage and it is designed for use with specific satellites.

BEAMING

(See DIRECTIONALITY.)

C

CAPACITOR

An electronic element in which two metallic plates are close to each other but not in contact, with voltage on one plate affecting the voltage on the other. A capacitor will not conduct a direct current, but will allow alternating current to pass.

Capacitors have an impedance with respect to frequency which declines at a regular rate of 6 dB per octave. At low frequencies a capacitor therefore has high L, making it an excellent filter for keeping low frequencies away from a tweeter. Therefore, capacitors are common elements in crossovers.

CAPACITANCE

The magnitude or size of a capacitor in microfarads.

(See IMPEDANCE.)

D

DAMPING

Any form of resistance or friction, electrical, mechanical, or acoustic.

Most of the damping of woofers is caused by the amplifier (electrical damping). Fiberglass in speaker enclosures is another form of damping (acoustical damping). In tweeters damping may be accomplished by means of a magnetic fluid bathing the voice coil (See FERROFLUID).

In general, damping reduces the amplitude of resonance

The careful control of damping is important in speaker design because resistance or damping is a factor in the calculation of Q, which greatly modifies speaker response across a wide audible frequency range.

DAMPING FACTOR

The ratio of a speaker's impedance to the output impedance of the amplifier which drives it. Since speaker impedance is usually around 8 ohms, and amplifier output impedance is generally less than .1 ohms, damping factors may reach or surpass 100.

Most amplifiers have comparable damping factors. An amplifier can be designed to create more damping effect, but it is not certain that a higher damping factor is better. Amplifier damping is often related to negative feedback, which is not necessarily a good thing.

(See NEGATIVE FEEDBACK.)

DAMPING FLUID

Damping fluid is not necessarily ferrofluid, but ferrofluid is so useful and versatile that it has completely replaced other types in loudspeakers.

(See FERROFLUID)

dB

dB refers to the ratio between two power levels. 10 dB is equivalent to a power ratio of 10; 3 dB is equivalent to doubling; -3 dB is equivqalent to halving. 20 dB is multiplying by 100. 100 dB means multiplying by 10,000,000,000.

(see SPL, DYNAMIC RANGE.)

DIGITAL

A way of storing and processing bits of information in a format that is suitable for computers to manipulate. Before digital information can be heard from a loudspeaker it has to be converted to analog in a D to A converter.

(See ANALOG.)

DIRECTIONALITY

At its highest frequencies, a speaker directs almost all of its sound straight ahead . At low frequencies, sound radiates equally in all directions. The former effect, referred to as "beaming", can be a problem in speaker design. *(See DISPERSION.)*

Beamed sound has a psychological effect which can be pleasant: it causes voices to sound more "up front". It can also be unpleasant.

STEREO EFFECT is exaggerated with beamed sound. If you get good dispersion, there's a lot of sound arriving at your ear in complex patterns: the first reflection, the second, etc. making it confusing to judge when the sound actually happened. The phase information used by the mind to derive the stereo image is emphasized when beamed. This is one of the appeals of earphones, where virtually all sound is beamed with no dispersion.

Some designers choose to use drivers in their directional range (or beaming range) in order to obtain the desirable effects of beaming. If the designer is trying to achieve a more linear speaker (with respect to dispersion) he will wish to eliminate beaming. This can be done by crossover design, using crossovers to deprive drivers of frequencies where they beam, or through driver design, since drivers can be designed so that they don't produce any sound at frequencies where they would beam.

Though directionality is generally thought of as a high frequency problem, it also occurs at the midrange frequencies at the "high-end" of the woofer.

DOLBY

Dolby Labs is a company prominent in the design and manufacture of hardware and systems to modify the reproduction of sound, most notably for the "Dolby Prologic" and "Dolby Digital" systems.

(See HOME THEATER.)

DOME TWEETER

The original appeal of the domed tweeter is that it physically resembles the spherical waveform which an ideal loudspeaker should produce. Actually, the real advantage is that the dome shape gives the tweeter a very rigid structure so that it does not deform at high frequencies as much as a cone tweeter, e.g. the dome shape resists deformation and therefore has a more uniform response.

The conventional domed tweeter is convex, with a voice coil of the same diameter.

Also see SOFT DOME TWEETER, INVERTED DOME TWEETER)

DOUBLING

A speaker generally has some low frequency at which there is a dramatic increase in distortion; reviewers generally refer to this phenomenon as "doubling." The origin of the term may be the fact that most speakers have a tendency to produce the harmonics of any tone they are required to reproduce, and one such harmonic is double the original frequency. Also the loudness seems to double. In reality, it is the third harmonic which sounds less pleasant than the second, so that what is labeled doubling may in effect be trebling.

Another type of distortion often called doubling occurs when a speaker causes a low-frequency vibration in the grill-cloth assembly. Another occurs when a speaker goes beyond its normal limits of travel and hits up against its suspension or frame (bottoms out).

DYNAMIC RANGE

The difference between the loudest and the softest sounds. This is normally expressed in dB. The dynamic range of human hearing is about 130 dB. The dynamic range of symphonic music is about 100 dB. For a typical CD it is about 80 dB; for a typical LP, about 40 dB. The lower limit of dynamic range is determined by background noise; the more noise, the less dynamic range. The upper limit of dynamic range is determined by physical restraint: amplifier power, for example; for human hearing it is physical pain; for absolute sound, it is the elastic limits of air.

(see dB, SPL,NOISE.)

E

ELECTROSTATIC RADIATOR

The simplest form of electrostatic radiator is a capacitor in which one of the plates is allowed to move, causing sound waves. The force is created by the alternating electric field which comes from the amplifier. Adding a polarizing voltage to one plate of the radiator will usually improve efficiency and lower distortion, so many electrostatic designs use power supplies of their own. Very often high voltages are involved (up to 3000 volts) because efficiency increases with voltage. This voltage can be obtained from house current, in which case the speaker will plug into the wall, or from the audio signal. This latter technique causes some leakage and loss of signal, leading to clipping and a bit of distortion.

The area of the plate reached by electric current gets smaller as frequency increases. (It is, after all, a capacitor.) Designers can control what part of the plate current goes to by use of electrodes in the plate, by use of multiple plates, and by controlling the conductivity of the material itself.

Polyvinylidene (of Saran-wrap fame is often used for plates because it has the right combination of conductivity and low weight. Mylar can also be used.

EQUALIZATION

Years ago, recording studios used speakers which were chosen primarily for efficiency rather that bandwidth. People in recording studios had no idea what kind of high and low frequency response was going onto their records. The introduction of the equalizer allowed them to change the power to the speaker to make up for its deficiencies, and a lot of studios can now hear stuff they couldn't hear before. Therefore the quality of sound on records, particularly in the last five years, has improved a great deal. This has in turn caused speaker manufacturers to become more and more interested in the high and low frequency response of their products.

The improvement in records has also had beneficial effects on cartridge design. In turn, the improvement in cartridges allowed the recording industry to make yet another round of improvement through better hearing. This is, perhaps, the best example of symbiotic effect in audio. Also true, of course, with speakers and other components.

The introduction of CD's and digital technology has accelerated this process even more.

EQUALIZER

A component with lots of knobs, each one representing a certain bandwidth, or frequency division (an octave or fraction of an octave) which allows you to boost or decrease energy going to a speaker in that range. The result is that you can compensate for speaker limitations or acoustic distortion in the room itself. Now common in recording studios and becoming more common in living rooms.

F

FERROFLUID

Ferrofluid is a trademark of Ferrofluidics, Inc. It is truly a space age material, a magnetic fluid, originally created for lubricating ball bearings in space. The liquid has tiny magnetic particles attached to the fluid base the way soap attaches particles of dirt. This fluid flows toward stronger magnetic fields.

In speakers, ferrofluid can be used in the gap of a tweeter magnet. Because of the fluid's magnetic properties, it is held in the gap and cannot run out. The presence of this fluid helps cool and therefore protect the tweeter by improving conductivity between the coil and the frame. ferrofluid also helps lower distortion because it prevents air movement in the gap. (This eliminates the possibility of spurious whine or whistle.)

Once sold at more than $100 an ounce, now a bit cheaper, this fluid is more in the class of optimization than luxury. The net result is that the loudspeaker can take greater power input and produce greater amounts of sound without burning out tweeters.

FILTER

Something which limits the bandwidth of transmission. A filter can be electrical, mechanical or acoustic. Typically, filters are hi-pass, low-pass or band-pass, referring of course to frequencies. For example, the suspension of an automobile is a low-pass filter, hopefully blocking all but the very slowest bumps from the tender sensibilities of the passengers.

(See CROSSOVER, BAND-PASS.)

FLETCHER-MUNSON EFFECT

Fletcher and Munson proved conclusively that at low volume levels, the responsiveness of the ear to both high and low frequencies declines. In other words, as you reduce volume, both bass and high treble disappear before the midrange sounds do. Many amplifiers have a loudness control to boost low and high-end response at low volume settings and thus compensate for this particularity of the human ear.

(See LOUDNESS, HUMAN HEARUNG.)

FREQUENCY

An example of frequency is a single musical note. Frequency was originally described as cycles per second, now simplified to Hz. (See HERTZ). The frequency of a second hand on a clock is one cycle per minute 1/60 Hz.A single frequency is a series of identical waves. The numerical value of the frequency is the number of complete waves that pass a single point in one second.

(See OCTAVE, HARMONICS, WAVELENGTH.)

K

KLIPSCHORN

Book shelf speakers seldom have efficiency above one percent. For the Klipschorn (an invention of Paul Klipsch), it's something about 30%. The horn-loading makes it far more efficient, but the speakers really have to be big. The Klipschorn principle uses the walls of the room as effective extensions of the speaker itself. This effect applies only to sub-woofer frequencies.

L

LINEAR, LINEARITY

As opposed to non-linear. When dealing with cause and effect, an effect is linear when it is exactly proportional to the input. When you pull on a spring with a specific force, twice the force should produce exactly twice the deflection. In a speaker, twice as much power from the amplifier should produce exactly twice as much sound.

All devices have some degree on non-linearity, hence distortion.

LOUDNESS

In audio, a term which usually refers to the principle (documented by Fletcher and Munson) that the response of the human ear to sound is not regular, but varies enormously with the intensity or loudness of the music.

At very low levels both bass and treble frequencies seem to decline more than the midrange frequencies. Many audio amplifiers therefore have some form of loudness control which causes the amplifier to boost certain frequencies at low-volume playback.

LOUDNESS CONTROL

Ideally a loudness control should affect both high and low frequencies. Actually, many so-called loudness controls affect only bass response, therefore doing only half the job. This is worth investigating when buying an amplifier.

Ideally, at concert-hall levels of sound, the frequency response of the amplifier should be flat whether the loudness control is engaged or not. Accurate compensation requires two controls, a gain control in addition to the loudness control. The gain control must be used to compensate for gain variations in all elements of the system (speaker efficiency, room acoustics, room size, source level, amplifier gain, etc.) to set the amp to the one gain setting at which loudness contour will provide a realistic balance.

Loudness systems which consist only of a button to convert the volume control into a loudness control and back again can't usually provide exact compensation. As a result, many audiophiles have never heard music with proper loudness compensation.

Understanding loudness compensation is important because properly used it can provide the same sonic balance from a system at low listening level as at full volume. Loudness controls, though, are generally misunderstood and often are used simply to increase bass response (sometimes because people feel nervous about any deviation from "flat" response and fell vaguely that the loudness button is "legal" while the bass control is not.)

M

MAGNET

The source of the magnetic field which interacts with electrical current flowing through the voice coil wire to produce the force which moves the speaker cone.

Alnico, which used to be the material of choice, has been mostly replaced by ceramic ferrites. In cases where the magnet has to be very small or very strong, a neodymium alloy is now used.

MOTOR

The combination of magnet, its steel housing and the voice coil. These parts provide the power and the force to move the loudspeaker.

N

NEGATIVE FEEDBACK

In general, feedback is the application of a portion of the output signal to the input signal of any system which has gain. The most common usage of feedback is in audio amplifiers, since distortion in the output signal of an amplifier can generally be reduced by feeding a percentage of the output signal back into the input of the circuit with polarity reversed (hence the term "negative feedback").

In fact since all design is a matter of trade-offs, any decrease in distortion obtained by this technique must be paid for by a decrease in gain in the amplifier. Since negative feedback is essentially a corrective technique with an inevitable trade-off cost, a better solution is to create an amplifier with so little DISTORTION that negative feedback is not required to correct it.

Since negative feedback changes the damping factor of an amplifier (and therefore changes Q, therefore changing the frequency response of the speaker) some attention should be paid to this question when choosing an amplifier since speakers may sound quite different when driven by amplifiers with different amounts of negative feedback.

NOISE

Random sound at all frequencies. Produced everywhere, even by the collision of molecules in the air around us. (This noise is just below the level of sound audible to humans, but exists nonetheless.) In audio, noise is also caused by the flow of electric current through transistors, vacuum tubes, and resistors.

Since there is always noise in the listening environment, sound-producing systems must be turned up loud enough to mask the noise. This fact may become a problem with components, like amps, which produce their own noise, since internally-generated noise may increase faster than the musical signal as volume is increased. Though modern high fidelity equipment usually generates so little noise that this not much of a problem, some internally-generated noise always shows up nevertheless in almost all component-generated sound. (Also see DISTORTION in PART TWO.)

Systems which are to be used in a noisy environment should include relatively efficient loudspeakers. In such environments, low efficiency loudspeakers may require too much power in order to come up to levels which effectively mask ambient noise.

(Also see WHITE NOISE AND PINK NOISE.)

O

OCTAVE

A frequency term: doubling or halving a frequency. Called an octave because it is eight notes of a musical scale.

Also see HARMONICS.

OVERTONES

Same as harmonics

P

PASSIVE RADIATOR

A loudspeaker component that looks like a woofer but isn't, for it has neither voice coil nor magnet. Like a tuned port, a passive radiator is part of a secondary sound producing system used either to improve efficiency or to widen bandwidth by utilizing the real woofer's back wave.

A passive radiator may be loaded with more mass than can be obtained in a tuned port of standard size, so it may be the solution in a particularly small cabinet. A sales advantage of the passive radiator is that it often makes the buyer think he is getting a large woofer when he isn't.

PHASE or PHASING

Sometimes called polarity, when it refers to the plus/minus of interconnections. In a more general sense, it refers to the timing of a waveform. Particularly important in stereo. Two speakers connected to the amplifier out of phase so that one is pushing (waveform is increasing) while the other is pulling (waveform is decreasing) will usually sound flat. Mis-phasing can cancel out most of the bass, particularly if the speakers are close together. Speaker hook-up wire is generally color-coded or otherwise coded so that both speakers can be grounded to the negative output connection of the amp, putting them "in phase".

PHASE is also important in each individual speaker system. Suppose you have a woofer and tweeter producing the same sound simultaneously. The timing or phasing of the sounds will be influenced by cross-over design, mass, , impedance etc.

In the more general meaning of phase, one complete cycle of a sound wave can be considered to be 360 degrees of phase. 180 degrees is one half way through the sound cycle, etc. As frequency rises, inductance alone can cause phase shifts of as much as 90 degrees. At the cross-over point phase shifts as great as 120 degrees are not uncommon. For this reason, crossovers are sometimes wired 180 degrees out of phase (by reversing wires) for a net reduction in mis-phasing or mis-timing.

Another interesting area of phase change is around woofer resonance. Right at resonance there is generally minimum phase shift, but phase does shift dramatically just above and just below resonance.

In a speaker with high Q, phase shift around resonance will be greater. There is some dispute and discussion as to whether one can actually hear this phase shift. Since variations in frequency response are often associated with variations in phase response, it's difficult to know which one is being heard. It's generally accepted that drastic shifts in phase are audible as distortion. To put it differently, it's desirable to have a system in which phase shifts only gradually with change in frequency response

POLARITY

A special case of phasing (either 0 or 180 degrees, plus or minus).If you touch a battery's terminals to those of a woofer, the woofer will move in one direction, either in or out. The convention is that if the woofer cone moves out, the positive terminal of the battery is touching the positive terminal of the woofer. In this case the polarity is positive. If the battery's terminals were reversed, the woofer cone would move in, representing negative polarity , or a reversal of polarity.

PRO LOGIC

Produced by DOLBY Labs for home theater systems, using four amplifier channels for Left, Right, Center and Rear (Ambient) speakers.

S

SATELLITES

Very small speaker enclosures with only a tweeter or a mid-range and a tweeter. These are designed for high frequencies only and are usually combined with a bass unit, sub-woofer or other main speakers.

Satellites can be used as main speakers, when combined with a sub-woofer or bass unit or as rear channel speakers.

(See HOME THEATER)

SOFT DOME TWEETER

A dome tweeter made out of relatively soft material, which allows radiation from only a portion of the dome (mostly adjacent to the bond to the voice coil)

(See DOME TWEETER.)

SPEAKER CONTROLS

Some loudspeakers have controls which allow the owner to modify their sound. Usually there's a knob to change tweeter response, and in three-way or four-way systems, a knob or knobs to change the intensity of midrange response.

The purpose of these controls is to permit the speakers to be "tuned" to the room in which they will perform and/or to the ear of the listener. Some individuals spend a great deal on speakers then fail to get maximum performance from them by neglecting to find the optimum settings for controls (a process which may take quite a bit of time and experimentation.)

There's no system so good that it can't be improved. Therefore minor adjustments to speaker and electronic component controls may easily create an improvement in overall sound. The listener should feel free to experiment until he arrives at the sound quality which suits both the environment and his own hearing. The ear, not some arbitrary ideal setting (e.g. flat controls) should be the ultimate test.

SPEAKER SPECIFICATIONS

Something to be used with extreme skepticism.

Most specifications which are used to promote and sell loudspeaker are simply those which are easily made. The equipment used to produce them is easily obtained and easy to use. So, most specifications you may read about a piece of equipment are not generally very important to the performance of the unit. Every time someone invents a new piece of test equipment, one finds new specs getting into speaker literature.

FREQUENCY RESPONSE isa specification where there has been confusion for years. That's because there is no simple or agreed-upon way of measuring the frequency response of a speaker. Microphone placement, room size and shape, reflecting surfaces, etc. may vary and give immensely different specifications for the same speaker. For example, it's quite easy for frequency response specifications to go from +/- 2 dB to +/- 10 dB simply by repositioning the microphone, creating perfectly "accurate" yet completely different specifications for the same speaker.

Historically the meaning for frequency response has been "any audio output in that frequency". In these terms, a response curve reading 15 Hz to 25 KHz is completely meaningless because both these frequencies are beyond the range of human hearing.

Furthermore, they are both outside the range of the usual test equipment used by speaker manufacturers to test their products. This means that such specifications are either being made up by copywriters and engineers, or that they are judging by any kind of audio output. (For example, if you put 15 Hz into many speakers, you'll hear something. It won't be 15 Hz, but one of its harmonics: 30, 45, Hz etc.)

On-axis measurement of frequency response is also tricky. Lots of speakers have response up to about 16 KHz +/- 3 to 5 dB if the microphone is placed dead center in front of the speaker. That kind of measurement gives good specifications. However, anywhere but straight ahead the actual power output at the highest frequencies may be negligible compared to other frequencies. (See DIRECTIONALITY.)

CROSSOVER frequency specifications? Who cares? The usual reasons for including them: they're easy to specify and fill out some of the white space on the page.

A basic tenet of science is that measuring techniques inevitably affect the phenomenon being measured. A good example of this occurs in the field of amplifiers. For years amplifiers have been judged by an inexpensive piece of equipment to measure it. There are amplifiers which sound better with their high harmonic distortion than others with better specifications, but it's the ones with the best specs which sell best. In speakers, a new piece of equipment has just become available to measure phase distortion, and now all kinds of specifications relating to phase are appearing in the industry's literature; yet it's a relatively unimportant consideration. In this sense, the importance given to specs by reviewers may not be helping the public.

SPEAKER WIRE

Audiophiles with extremely sensitive hearing claim that they can hear differences between various forms of speaker wire. Scientific evidence for this is very slight. Even straight wire has some inductance, and there is always at least some capacitance between two wires in proximity to each other. It turns out that there are some wires which have detectable reactance at audible frequencies and therefore to infer that there may be some audible effects.

There are many conjectures about how wire can cause a difference in sound, such as oxidation in the wire, magnetic eddy currents within thick wires, rectification between wire strands, and interference between high and low frequencies.

What is certainly true is that the thinner and longer the wire, the more resistance. Speakers usually have impedance of several ohms. In order that the resistance of the wire be small compared to that of the speaker, one half ohm would be too much.

SPL

Sound Pressure Level is expressed in dB and is defined in terms of the practical hearing range of human hearing. For the human ear, silence is defined as 0 dB SPL. The threshold of pain for the human ear is at about 130 dB SPL.

(See dB.)

STANDING WAVE

An acoustic resonance of high Q, generally an annoying form of distortion, usually in the bass, caused by the shape and dimensions of the room in which music is being reproduced. When the distance between two parallel room surfaces is one fourth of the wave length of a given frequency, or an odd multiple of the wave length (three times, five times, etc.) the pressure of the reflected wave is in phase with the wave itself, creating a room resonance at that frequency.

On the other hand, if the distance between the two room surfaces is one half the wave length, or an integral multiple thereof, the pressure of the reflected wave is out of phase with the wave itself, creating a canceling at the frequency which is another form of distortion but a bit less noticeable.

Of the two types of distortion, the former is the greatest problem, because it tends to augment or complicate acoustic feedback.

Careful repositioning of speakers, careful repositioning of objects in the room (like furniture) and the use of sound-absorbing materials on room surfaces are the classic solutions to standing waves. The fewer parallel planes there are in the room's surfaces, the less likely that there will be standing waves, so that room acoustics are generally improved by angling a wall or two, and/or the ceiling, when the room is being planned.

Organ pipes are based on standing waves.

(See Resonance, Q.)

STEREO and STEREO EFFECT

Also see HUMAN HEARING)

If a click is heard by both ears at once, it will appear to be straight ahead or behind the listener. If the sound reaches one ear slightly sooner, the sound will seem to emanate from that side.

This is the principal cause of stereo effect, a shift in timing or phase.

However, a change in volume in one ear will have a similar stereo effect. Because of studio mixing techniques, many stereo recordings use only this volume differential to convey stereo information, and overlook or ignore the primary information, timing. This stereo is not as complete, but is particularly common in pop recordings, where one track is recorded at a time and then added to the final tape.

At very low frequencies, however, phase information in two channels is almost nil because of the wavelength of the sounds. Conversely, at frequencies where wavelength comes close to the size of the space between the ears, (250-10,000Hz), the perception of directionality increases.

SUB-WOOFER

A separate speaker enclosure (often with its own power amplifier) dedicated only to bass frequecies, e.g., from 15 - 150 Hz. The theory is that most stereo speaker systems lack one or two octaves of the deepest bass, and that this can be added with a separate speaker. Unfortunately many so-called sub-woofers are still missing one or two octaves of the deepest bass.

Sub-woofers are often configured as one monaural channel, on the assumption that low frequencies have very little stereo information.

(See also BAND-PASS)

T

THIELE

An Australian electrical engineer whose name has become quite famous in audio circles. Thiele's work was based on the fact that any loudspeaker system can be expressed in classic electronic symbols. He then applied filter theory (a special branch of electronic analysis) to speaker design.

The advantage of Thiele's work is that it makes it possible for the designer to avail himself of the immense body of statistics and research that has gone into electrical filter theory in order to obtain the desired frequency and phase response in a new design with minimum trial and error.

Thiele's theories have been taken up and popularized by Richard Small, who did a Ph.D. on Thiele's work.

Today's speaker designers often use filter theory, analyzing loudspeakers as analogous to electronic circuits. This approach is particularly useful for designs using either a port or a passive radiator, for Thiele's work is particularly applicable to the selective tuning of the port or radiator to achieve desired frequency response. Still, the Thiele approach is only one of several methods which may be used to arrive at the desired result.

THIELE-SMALL PARAMETERS

These are the values of the basic parts of the electronic circuit analogue proposed by THIELE and popularized by Small. They are related to the physical properties of a woofer, e.g. mass compliance, resonance, Q, Bl, etc. They are commonly circulated between manufacturers and design engineers because there are a number of cookbook recipes which use the Thiele -Small parameters for determining box size and port tuning.Computer programs are now common both for measuring the parameters and for designing the enclosures based on them.

TRANSDUCER

Any device which converts mechanical energy to electrical energy, or vice versa. Therefore, speakers, microphones, generators, alternators, motors, cartridges, etc. In speakers there are generally three types of transducers: dynamic, or conventional magnet and coil and paper cone; electrostatic, and piezo-electric effect.

In the dynamic type, sound comes from the movement of a coil in a magnetic field when current passes through it. In electrostatic, variations in voltage on two plates causes one plate to move in respect to the other (one plate effectively replaces the magnet of the dynamic speaker) transmitting a sound wave to the air in the room. In the piezo-electric effect, current causes a change in the shape of the material, a solid, causing it to vibrate and transmit sound waves to the air. (All these effects also work in the opposite direction.)

Transducers are particularly important in high fidelity because they are used in speakers, microphones, and cartridges, and they all have to be good for the sound to be good.

TRANSIENT RESPONSE

A term often used ambiguously. Specifically, transient response is the ability of a system to reproduce sudden changes, such as clicks or pops, especially at the beginnings of continuous sounds,rather than a steady waveform (sine wave).

In reality transient response is a reflection of frequency response. Pass a pulse through any system with a certain frequency response, and the accuracy of reproduction of a transient waveform will correspond to the limitations of the system's frequency response. A square wave can be used to indicate transient response. If there's a deficiency in high end response there will be a rounding of the square wave and a change in the vertical edge.

Though transient response is usually thought of as a high frequency function, there is also such thing as low frequency transient response. Failure to handle this results in a tilt of the horizontal section of the wave.

Some people confuse high Q with poor low frequency transient response, because drums seem to be muffled in a high Q system, and this can be erroneously labeled "poor low frequency transient response". Actually, poor low frequency transient response is of limited importance since the human ear does not usually recognize the absence of low frequency harmonics (or sub-harmonics).

TRANSMISSION LINE

One type of transmission line is a type of bass reflex system with the added characteristic of providing a long path for the rear wave to follow so that by the time it gets out of the enclosure it is in phase (in a certain frequency range) with the front wave. Not particularly useful, since this design offers neither the frequency range nor the efficiency of a tuned port. It is however appropriate for those who prefer a softer, well-damped bass.

A second type of transmission line, less common, uses a cabinet design which absorbs all the rear energy from the woofer, making it behave as if it were mounted in an infinite baffle.

TUNED PORT/BASS REFLEX

A tuned port is simply a tube or opening allowing air from inside the speaker box to escape or enter. Note that the air in a tuned port has the property of acoustic mass. It moves in the port without being compressed, has inertia, and therefore has its own resonant frequency. The port is generally tuned (dimensioned) in the vicinity of the woofer's resonant frequency in order to augment bass output. A tuned port can be used either to increase efficiency or to increase frequency range, depending on the goals of the designer.

(See PASSIVE RADIATOR, BASS REFLEX.)

W

WAVELENGTH

Inverse to frequency, waves are short at high frequencies and long at low frequncies. The corresponding range for 20 to 20,000 Hz is 50 feet to 1/2 inch. Wavelength is the distance from crest to crest, similar to ocean waves.

(See FREQUENCY.)

WHITE NOISE and PINK NOISE

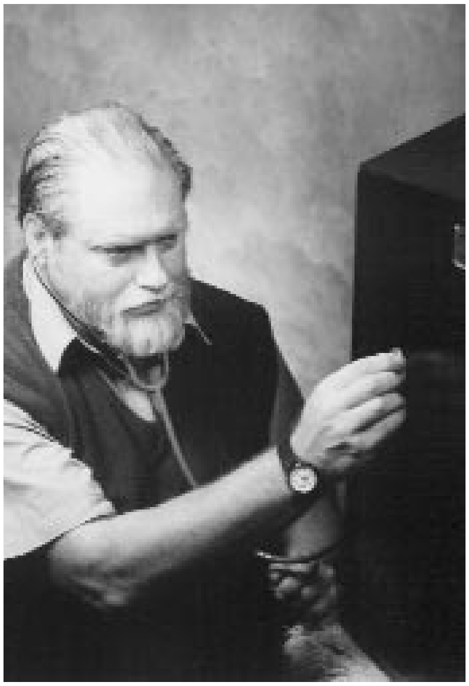
NOISE is often used in testing speakers because it's easy to obtain. Also, a lot of companies like to use noise as a signal source when measuring specifications because it does not show up peaks in the frequency response as a pure tone would.

WHITE NOISE: noise in which there is equal energy per frequency division. (For example, the same amount of energy between 100 and 200 Hz as between 1100 and 1200).

PINK NOISE has equal energy distributed logarithmically, for example, the same energy between 100 and 200 Hz as between 1000 and 2000. Pink noise corresponds fairly closely, on an average basis, to music. (More energy at low frequencies.) In listening tests with speakers, Pink noise is a better listening tool than White noise. In white noise there is a great exaggeration of treble. Surf is an excellent example of white noise.

High frequency sound is attenuated in air (particularly humid air) so that the closer you are to surf (white noise) the shriller it sounds.

Even the humidity in your living room will to some degree influence the treble response of your speakers. On cold winter mornings, sound travels well, not just because the air is cold, thus dense, but also because of its dryness. (Very cold air rejects humidity.) Probably no-one will seriously suggest getting a dehumidifier for your living room in order to improve the high end response of your sound system. Still, acoustically speaking, it would help.



WINSLOW BURHOE

A loudspeaker designer since the early 1960's, starting at Acoustic Research, becoming a production engineer at KLH and founder of Epicure Products with a number of breakthrough speaker designs. Since then he has designed hundreds of speakers for companies that include Boston Acoustics, Inkel, Genesis Physics, Energy, Nuance, and Synergy.