Special Focus on Speakers

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Innovations in Speaker Design

BY GEORGE TLAMSA

IN THEIR never-ending quest to produce the *perfect* loudspeaker, engineers are continuously developing new materials, enclosures, transducers, circuits and systems. The result? Literally hundreds of loudspeaker systems in a great variety of shapes and sizes, employing dramatically different operating principles, all vie for consumer acceptance. This often bewildering array of loudspeaker designs can easily create confusion and make a meaningful buying decision difficult at best.

In this article, we will examine the basic types of loudspeaker components and systems, placing special emphasis on the latest advances in loudspeaker technology. First, we'll look at the various transducers (drivers), and then loudspeaker enclosures. Finally, we will see how loudspeaker engineers are putting all the components together to produce contemporary speaker *systems*, taking both the effects of the listening room and psychoacoustics into account.

The Drivers.

As its name implies, the function of a driver (or more properly an electroacoustic transducer) is to impart motion to the air surrounding it. This motion will in the ideal case correspond exactly to the time-varying nature of the electrical signal applied to it. Our ears, in turn, will perceive the motion as sound.

There are many ways to convert a time-varying electrical signal into sound waves, as witnessed by the great variety of drivers found in today's speaker systems. Let's examine the principle types of drivers presently in use, and see how each measures up to the ideal transducer.

The Dynamic Speaker. The oldest of all the driver types employed today is the dynamic speaker. Its design goes back to the mid-Twenties when Chester Rice and E.W. Kellogg produced the first functional prototype. Although the dynamic driver has over the years been improved in a number of ways, the principle is the same.

The typical dynamic speaker employs a conical diaphragm driven by a magnetic motor. Motion is imparted to the diaphragm via the interaction of a time-varying magnetic field generated by an electromagnet and a static field set up by a permanent magnet. In the

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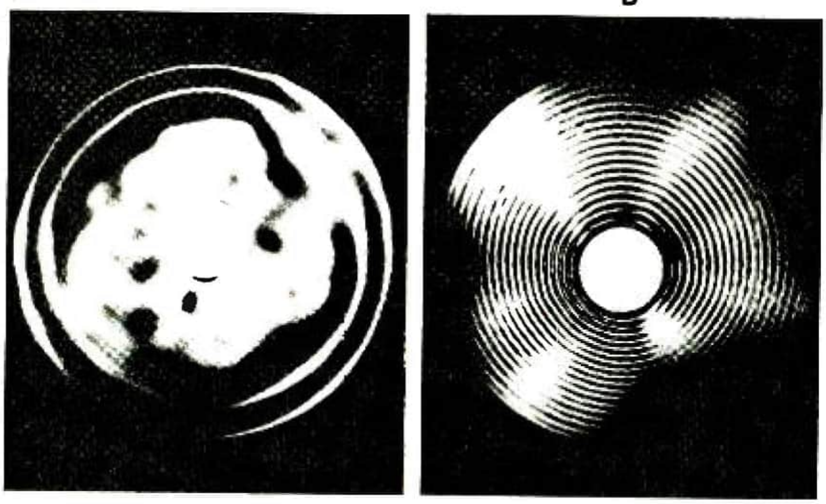


Fig. 1. Holographic analysis shows cone breakup in dynamic driver. In 8-inch woofer driven by 2000-Hz signal (A), standing waves appear at edge. At 9000 Hz (B), they occupy entire cone surface.

early days of dynamic speakers, sufficiently strong permanent magnets were not available, so the static field was supplied by an electromagnet called a *field coil*.

The time-varying field is set up by a *voice coil*, an electromagnet driven by the output of the power amplifier. The magnetic field set up by the voice coil varies in step both in amplitude and polarity with the audio-frequency current supplied by the amplifier. Alternate repulsion and attraction between the two magnets cause the cone (diaphragm), which is attached to the voice coil and a supporting structure, to compress or rarefy the nearby air, depending on its motion relative to the air mass.

When the cone is moving forward, it compresses the air in front of it and rarefies the air behind it. Similarly, when the cone is moving backward it compresses the air behind and rarefies the air in front. It can thus be seen that the driver is a dipole radiator, simultaneously generating two out-of-phase acoustic signals. At low frequencies, these two signals will meet while still in an out-of-phase condition and cancel each other out. To prevent this destructive interference, either the "front wave" or the "back wave" will have to be phase-shifted before it meets up with its counterpart. Alternatively, one component of the speaker output must be attenuated or otherwise prevented from reaching the other. Whichever of these design alternatives is chosen is usually performed by the loudspeaker *enclosure*. More on this later.

Ideally, the dynamic driver behaves like a rigid piston over its entire operating (frequency) range. A practical driver, however, cannot provide this ideal over the entire audible spectrum. Its useful bandwidth will always be limited to those frequencies where it can approximate a *point source*. That is, to those frequencies whose wavelengths are large compared to the diaphragm's physical dimensions.

Above these frequencies, the driver begins to "beam" (become directional) due to diffraction effects. The cone of the driver ceases to act as a rigid piston and undergoes a series of flexing motions that are structural resonances. This can be seen in Figs. 1A and 1B.

These images were generated in the Netherlands at the Philips Research Labs by holographically observing the motion of an 8-inch (20.3-cm) woofer cone. At low frequencies, the cone vibrates as a rigid surface. Above a certain frequency, standing waves begin to appear on the cone. For example, Fig. 1A reveals loops and nodes just starting to appear at the periphery of the cone when the woofer is driven by a 2000-Hz sine wave. When the frequency of the drive signal is increased to 9000 Hz, severe cone breakup occurs. Its entire surface is covered with loops and nodes (Fig. 1B), and only a little sound is radiated.

Not only do these effects make the driver directional as the operating frequency is increased, but they also cause fluctuations in its frequency response. If the dimensions of the driver are kept small to enhance its midrange and high-frequency response, it will not be able to move enough air to provide the substantial acoustic output at low frequencies required for high-fidelity reproduction.

It is obvious that no conventional dynamic driver can single-handedly cover the full range of audible frequencies. This has led to the develop-

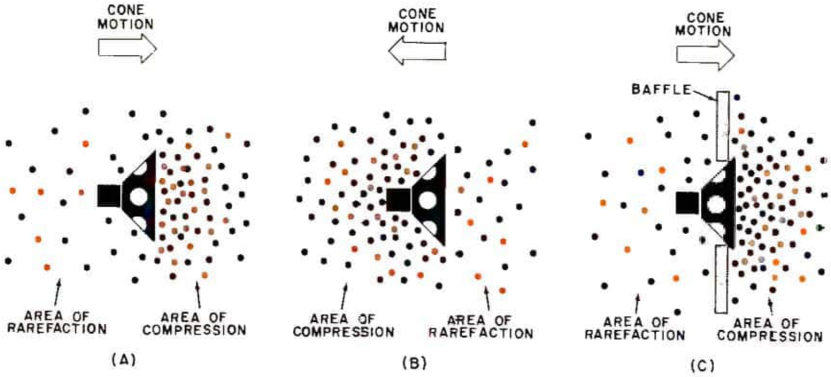
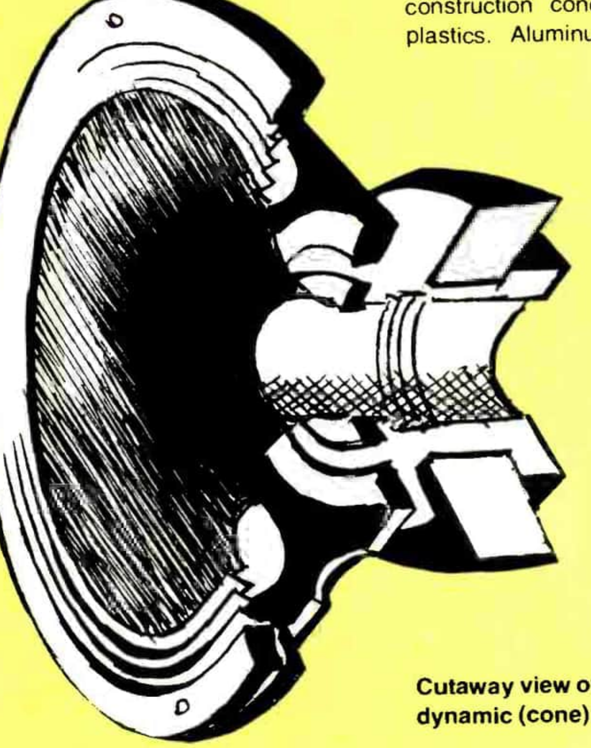


Fig. 2. Cone compresses and rarefies air as shown at (A) and (B). Baffle (C) separates two air masses to prevent cancellation.



Cutaway view of dynamic (cone) driver.

ment of specialized dynamic speakers, each designed to handle a given portion of the audio spectrum. A typical speaker system employing all dynamic drivers will contain a woofer, a midrange driver, and a tweeter. Some systems employ four or even more drivers, with a *supertweeter* handling the extreme highs and/or a *subwoofer* reproducing the deepest bass notes. From the standpoint of system engineering, however, it is often desirable to employ drivers with useful frequency ranges as large as possible.

In the case of a woofer, useful frequency range can be increased by making the cone more rigid. One early approach to increasing the diaphragm's rigidity was to change its shape from that of a simple (true) cone into a "cone" with rounded sides. Today's really large woofers like Electro-Voice's 30-inch (76.2-cm) unit are constructed in this manner.

Most of the engineering effort dedicated to improving the dynamic woofer has been channeled into the development of new materials for the cone to replace the paper traditionally used. The ideal cone material would be lightweight (for efficiency and good transient response) and *very* stiff (for extended frequency response). Recently, some woofers have been made using carbon-fiber pulp as the cone material. During the Sixties, the BBC experimented with sandwich-construction cones utilizing special plastics. Aluminum has also been used in the fabrication of woofer cones. Finally, some manufacturers have tried doping paper cones with special coatings for added stiffness. The Bextrene woofer is a representative product of this type of experimentation, and is commonly employed in British speaker systems.

Tweeters can be made using paper cones, but today the dome tweeter is an increasingly popular alternative. The dome has the advantage of permitting the use of a large voice coil (for power handling), but must be made of very lightweight material if it is to be an efficient radiator. Among the materials used to fabricate tweeter domes are Mylar-type plastics, polystyrenes, treated fabrics, and, notably, beryllium alloys. Beryllium is one of the hardest metals known and is extremely lightweight. It is thus ideal for dome diaphragms. Dynamic tweeters, by the way, needn't have the familiar voice-coil construction, as evidenced by the ribbon tweeter. This driver utilizes a thin corrugated metal "ribbon" placed in a static magnetic field. Audio-frequency current is passed directly through the ribbon, which acts as both a voice coil and a diaphragm.

One problem that the designer of any dynamic driver must confront is the dissipation of heat produced in the voice coil, which generally has a very low impedance (4 to 8 ohms). If the coil draws too much current, the heat generated in its windings can melt the insulation on the wire (and perhaps the wire itself) resulting in a burn-out of the driver. Even if this doesn't happen, excessive heat will deform the coil and weaken the binder that holds it to the coil form. A discussion of these problems is included in another article in this section.

An interesting variation on the standard dynamic driver is the Watkins dual-impedance woofer used by Infinity Systems in its top-of-the-line speakers. This otherwise conventional woofer has two voice coils, one with a standard impedance and a secondary coil with a low impedance. The two are electrically coupled together by series and parallel LC networks. Ordinarily, the tuned circuit applies the amplifier output to the standard voice coil. However, at frequencies near the fundamental resonance of the speaker system, where the impedance presented to the amplifier reaches its maximum value, the tuned circuit routes signals to the secondary, low-impedance voice coil. This provides the amplifier with a virtually constant load impedance throughout the bass region.

The Electrostatic Driver. As its name implies, this type of driver generates a motive force for its diaphragm by the interaction of electric rather than magnetic fields. In a typical electrostatic design, a large diaphragm of lightweight material is placed between two perforated (acoustically transparent) electrodes. The diaphragm is electrically polarized relative to the electrodes, which maintain a large electrostatic field. The audio signal is applied to the two electrodes in push-pull fashion. Under these conditions, the diaphragm will vibrate in step with the audio drive signal and produce sound.

In an electrostatic driver, the driving force is uniform over the entire diaphragm surface. (Compare that to the dynamic driver, in which the diaphragm is driven over a small portion of its overall surface.) The result is that the electrostatic doesn't suffer as

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drastically from "breakup" (the excitation of resonant modes on the diaphragm) as the dynamic driver. Transient response is excellent due to the diaphragm's low mass. The electrostatic driver is basically hung on a frame. It doesn't have a box enclosure, so a great deal of "coloration" (or frequency-response fluctuations caused by the enclosure) is avoided.

Of course, the electrostatic driver is not completely free from problems. Bass cancellation caused by destructive interference between the front and back waves does occur. Also, the diaphragm does become directional at high frequencies, where breakup modes occur. It is difficult to get a large acoustic output from an electrostatic. The diaphragm cannot make large excursions, and placing it in an electric field strong enough to produce high sound levels will result in dielectric breakdown of the air and arcing, which almost always causes permanent damage to the speaker. Without special design features, the acoustic output of an electrostatic driver is limited. Nonetheless, these designs, when executed properly, are remarkably good speakers. Some of the most highly regarded "purist" speakers are electrostatics. The famous Quad electrostatic speaker is a good example.

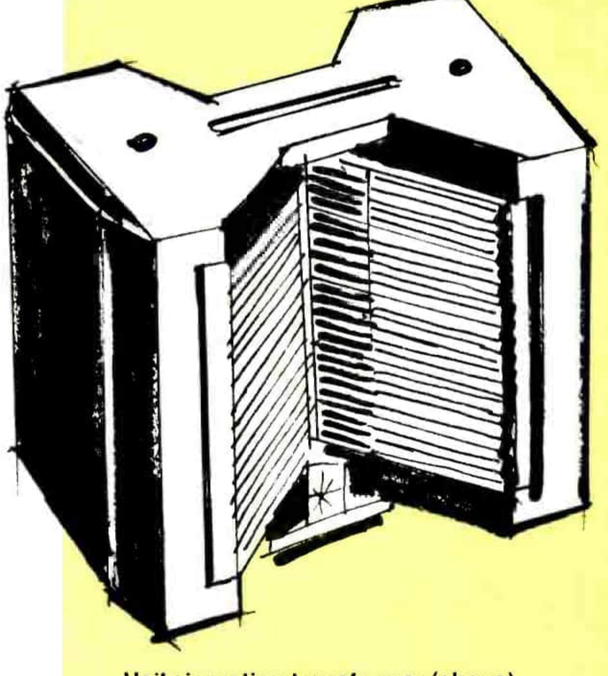
A solution to the limited-output problem of electrostatics has been developed by Dayton Wright Associates, Ltd. The driver is sealed in an airtight plastic bag. Actually, this is standard practice, because the electrodes and diaphragm must be kept free of contamination. The trick is that the bag is filled with SF_6 (sulphur hexafluoride) rather than air. This gas is a better dielectric than air in that it can support a much larger electric field before it breaks down and arcing takes place. Hence, in the Dayton Wright electrostatic driver, the voltage on the electrodes can be stepped up very high, making the speaker capable of generating increased acoustic levels.

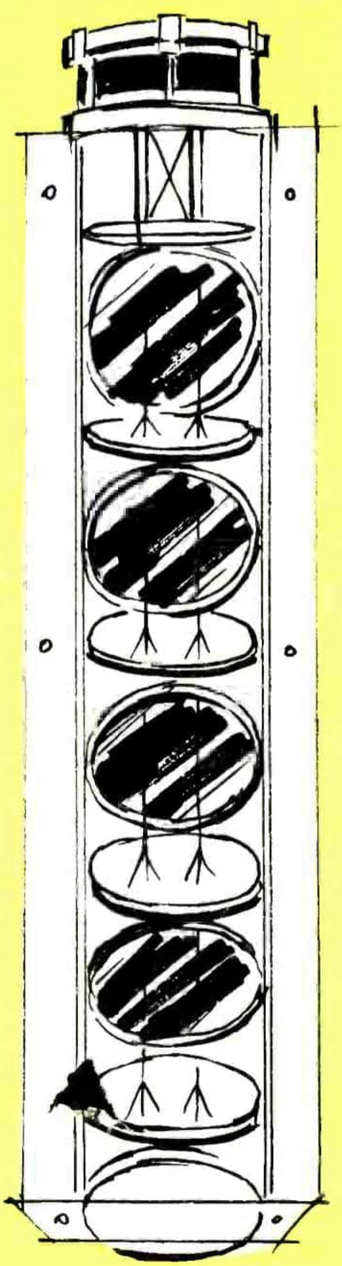
One last word about electrostatic drivers—the impedance they present to the power amplifier is high, and is largely made up of capacitive reactance. This is in strong contrast to the low impedance of the typical dynamic driver, which has a large resistive component. Very few amplifiers are able to drive an electrostatic driver directly. (Most of those that can are made by the speaker manufacturer and are sold with the drivers as integrated systems.) A step-up transformer can be used between an amplifier and the driver, but the amplifier will still be subjected to reactive loading. Some power amplifiers can tolerate this, but others cannot.

The Walsh Driver. Both the dynamic and electrostatic drivers can be considered "conventional" transducers in that they have been with us for many years. A relatively recent trend among loudspeaker engineers, however, has been to dispense with conventional drivers entirely and develope new types of transducers. One product typifying this approach is the Walsh driver, which is used by Ohm Acoustics in its Model F loudspeaker system.

This driver is a tall, upright cone constructed of three successive bands of material—titanium on top, then aluminum, and finally paper at the bottom of the cone. It is driven at the top by a specially-designed voice coil, and the bottom of the cone is held in place by a surround. This unique driver is a full-range, omnidirectional, and coherent radiator. There is no phase cancellation of waves leaving different parts of the diaphragm as the sound is radiated as a uniform wavefront. The Walsh driver's only major drawback is its relative inefficiency.

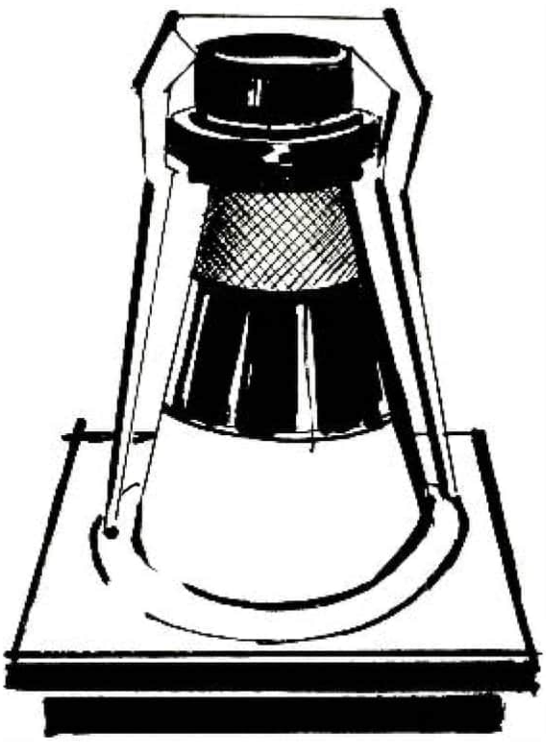
The Planar Magnetic. This is an electrodynamic driver that outwardly looks like an electrostatic speaker. As we have already mentioned, the electrostatic driver utilizes a large planar diaphragm of low mass to achieve excellent transient response, and the





Heil air motion transformer (above) and ATD woofer (right).

planar magnetic is its electromagnetic analog. This driver has a lightweight planar diaphragm with a pattern of conducting material bonded to its surface. The diaphragm is placed between two perforated magnets, thus forming a voice-coil/magnet "motor." Audio current passes through the diaphragm's conductor, generating a magnetic field that interacts with the static magnetic field to produce the force which generates the sound waves. The driver has the good transient response and other advantages of electrostatics. It is hung on a frame support, so there is no box to introduce coloration, but bass cancellation is a problem. In a full-range planar magnetic system, two or more drivers are often used because individual drivers do become directional at wavelengths small compared to their dimensions. This driver was pioneered by Magnepan, and that company's speaker systems use only planar magnetic drivers. The driver's main disadvantage is inefficiency. The planar magnetic driver need not be a full-range transducer. For example, Infinity Systems makes a planar magnetic tweeter (they call it the EMIT tweeter) which it includes in most of its speaker systems.



Walsh driver used in the Ohm F speaker system

Heil Drivers. A distant relative of the planar magnetic driver is the Heil Air Motion Transformer or AMT, a midrange and high-frequency driver found in speakers manufactured by ESS. Its diaphragm utilizes a lightweight Mylar material with an imprinted conductor pattern, but there the similarity with the planar magnetic just about ends. In the Heil driver, the diaphragm is folded up like an accordion and is placed in a uniform magnetic field. Application of audio-frequency current generates a magnetic field, which causes the many folds to squeeze air out from between them, producing sound. The AMT is a high-efficiency device capable of uniform response from the midrange to very high frequencies.

The high efficiency of the Heil AMT tweeter prompted experimentation with a low-frequency driver of the same design. This turned out to be problematic—the Heil driver has a cavity resonance above its nominal operating range. In the case of the tweeter, the resonance falls in the ultrasonic region, but a Heil woofer would resonate in the midrange. Undaunted, Dr. Oskar Heil designed another equally unusual driver, the ATD woofer. This unit has five small (4-inch or 10.2-cm) lightweight horizontal diaphragms mounted one above the other, all connected by a set of four lightweight carbon-fiber rods. The rods are driven by a conventional voice-coil assembly, and they in turn excite the diaphragms. The diaphragms are mounted in acoustic reflectors that isolate the "top wave" radiation from the "bottom wave." The entire columnar driver is mounted on a large, flat baffle intended to isolate front and back radiation as much as possible, but bass cancellation is a problem with this design.

Piezoelectric Drivers. An alternative driver for the high frequencies is offered by the piezoelectric driver. Pioneer uses a novel tweeter, dubbed the HPM, which is made of a piezoelectric polymer it developed. This is a plastic-like material that, unlike a crystal, can be made into thin sheets and fashioned into a variety of shapes. The HPM driver utilizes the film in a cylindrical configuration, and is loaded with an acoustic "lens" to control dispersion. Transient and high-frequency response of the driver is very good because the moving mass of the diaphragm is negligible. The driver operates by the structural expansion and contraction of the film. It is, however, inappropriate for low-frequency applications as it is incapable of large air displacements.

The "Massless" Driver. The plasma driver, found in the Hill Type-1 speaker system by Plasmatronics, is a realization of the loudspeaker engineer's dream—a transducer with essentially no mass. Sound radiation is achieved by the expansion and contraction of ionized air through which an audio current is passed. In the plasma driver, an air/helium mixture in a small cavity is subject to a high-voltage discharge which completely ionizes it (strips valence electrons from the molecules to produce positive and negative ions). The gas then becomes an electrically conductive plasma. Passage of audio-frequency current through the plasma causes local temperature and pressure fluctuations, the pressure fluctuations being sound waves. Because the plasma driver has no diaphragm and effectively no mass, it has excellent high-frequency and transient response. The idea is not new, but the Plasmatronics driver is said to eliminate many of the drawbacks of older designs, such as noisy operation and carbonizing of the electrodes. In the Hill Type-1 system, the plasma driver handles frequencies from 700 Hz up. It's major disadvantage is high operating cost. Bottles of pressurized helium are built into the loudspeaker systems and must be replaced an average of two or three times annually, for an operating cost of about 30¢ per hour.

The Enclosures.

The area of perhaps the most controversy among loudspeaker engineers and audiophiles has been the design of the enclosure. Predictably, there has been a proliferation of enclosure types, each characterized by certain advantages and disadvantages. What follows is a discussion of the role of the enclosure in sound reproduction and an examination of those types found in today's systems.

The Infinite Baffle. The simplest speaker enclosure, the infinite baffle, neatly illustrates the role the enclosure plays. We have already mentioned that such transducers as dynamic and electrostatic drivers radiate two out-of-phase components. The process is shown in Fig. 2. When the cone of the dynamic driver moves forward, the air in front is pressurized and the air behind is rarefied (A). The reverse occurs when the diaphragm is moving backward (B). A substantial pressure gradient is therefore set up by the motion of the loudspeaker's diaphragm.

When driven by a low-frequency signal, the diaphragm moves relatively slowly and radiates long acoustic wavelengths. In an attempt to equalize pressures, the compressed air rushes to the other side of the cone. This results in the mutual cancellation of the two out-of-phase sound waves, and little acoustic energy is radiated. At higher frequencies, however, the diaphragm is moving much more quickly and wavelengths are short. The pressurized air does not have time to reach the rarefied region and destructive interference does not occur. Cancellation, therefore, is a problem encountered principally in the bass region.

To prevent bass cancellation, the driver can be mounted on an infinite baffle (Fig. 2C), in theory an infinitely large flat surface. Such a baffle will prevent any destructive interference by totally shielding the rarefied region. Obviously, no baffle can be infinitely large, and practical "infinite" baffles have finite dimensions.

The baffle must be large if satisfactory bass reproduction is to be achieved. This is the drawback of the infinite baffle principal. Good design practice dictates that, with the driver mounted at the center of the baffle, the dimension of any one side should not be less than one wavelength at the lowest frequency to be reproduced. For a cutoff frequency of 40 Hz, the baffle size on each side of center should therefore be 14 feet (4.3 m). Few listening rooms can accommodate such a structure!

Sealed Enclosures. To attempt to provide the infinite baffle's prevention of bass cancellation in a design of more manageable dimensions, sealed enclosures are used. If the driver is mounted in a sealed box, the back wave has no place to go and stays inside the cabinet. The enclosure must be filled with sound-absorbing material to damp the rear radiation of the diaphragm. Otherwise, standing waves will form at any frequency where one dimension of the box equals an integral multiple of one-half wavelength of the radiated sound wave. As a result, not only would the system's frequency response suffer, but also the mechanical resonances of the box itself would be excited by the sound pressures created by the standing waves.

The sealed box has its problems. It is inherently inefficient because all of the rear radiation is damped out and converted into heat. Sealed enclosures housing standard woofers must be large if significant bass output is to be obtained, because the acoustic impedance seen by the rear of the driver will increase as the box volume decreases. Furthermore, if the air in the box is too stiff for the driver, the fundamental resonant frequency of the system will be too high for good bass performance. This design, however, enjoys the significant advantage of having a relatively gradual roll-off in low-frequency response below the fundamental resonance (about 12 dB per octave). This means, among other things, that the system won't ring excessively at resonance.

One special type of sealed box is the *acoustic suspension* enclosure. In this design, a woofer with a very compliant suspension is placed in a small sealed box. The small volume of air in the box has a high acoustic reactance that compensates for the high mechanical compliance of the driver. The box is completely sealed, with the back radiation of the woofer absorbed by damping material. This scheme enables a suitable driver mounted in an enclosure of small dimensions to reproduce very low frequencies. The acoustic suspension enclosure was introduced to audiophiles in the 1950's by Edgar Villichur of Acoustic Research, and was largely responsible for the bookshelf speaker coming into its own. The design is found in many contemporary speaker systems and can be viewed as a sealed-box enclosure with a special matching of the woofer compliance to the volume of the box.

An interesting variation on the acoustic suspension theme has been developed by Cerwin-Vega which the company calls Thermo-Vapor Suspension. It involves the use of an acoustic suspension enclosure filled with a gas that for a given volume is more compressible (compliant) than air. The Cerwin-Vega design results in high-level deep bass from a small enclosure with relatively high efficiency. (Most acoustic suspension designs are notoriously inefficient.)

The Bass Reflex. To make use of the rear radiation that goes to waste in a sealed box, the bass relex was introduced. An enclosure of this type is not completely sealed. Rather, it has an opening (a "port" or "vent") of carefully selected proportions that acts as a sound radiator at very low frequencies to reinforce the frontwave output of the driver. The port is "tuned" (its area and length predetermined) to radiate in phase with the active driver. The bass reflex enclosure is really a Helmholtz resonator—of the same sort as a wine jug or an ocarina—because when excited by sound waves of the right wavelength it resonates and produces sound from the port.

If a ported system is designed properly, bass output is augmented and the overall efficiency of the system compared to that of a sealed enclosure is increased. The box volume can be made quite small and still generate significant bass output. The enclosure's disadvantages are an unevenness in bass output and a notable tendency to ring near the frequency of resonance. (Vented-box systems roll off at 18 dB per octave below the system resonance.) Still, the design is fundamentally successful, and this is verified by the fact that a vast number of speaker systems on the market today employ it.

The Acoustic Labyrinth. This is another distinct enclosure type, although its most common incarnation, the *transmission line* (see Fig. 3), is rudimentarily similar to the bass reflex design. The basic concept behind the acoustic labyrinth is to provide a long acoustic path behind the woofer for attenuation and phase shifting of different components of the woofer's rear radiation. One way to achieve this result is to set up a complex series of internal baffles within the enclosure, these being filled with sound-absorbing material. This is often impractical as the cabinet must be quite large.

The transmission line is basically a long duct, one end of which is coupled to the rear of the woofer. The other end terminates in a port. The duct accomplishes two things. First, it provides a long path for attenuation of the rear radiation of the woofer at certain frequencies. Secondly, it results in reinforcement of the woofer's front-wave output near the system's fundamental resonance. The duct is filled with sound-absorbing material, and the frequency-dependent characteristic of the attenuation is such that very long wavelengths are relatively unaffected, but the shorter wavelengths (which comprise most of the woofer's output) are strongly suppressed. The unattenuated sound, which is very low in frequency, reaches the port in phase with the front wave of the woofer and augments it. The interesting twist to this design is that the sound-absorbing material in the duct not only attenuates acoustic energy, but also maintains a slower speed of sound propagation than that in air. This means that for a given frequency, the wavelength of a sound wave will be shorter in the duct than it would be in air. The duct can therefore be made to reinforce very low audio frequencies without being enormously long—transmission lines of 8 feet in length can provide useful response down to around 30 Hz.

The actual transmission line need not be a straight tube. To the contrary, it is almost always a folded duct. Reflections from the folds in the duct are minimized by using densely-packed absorbing material at these spots. The ideal absorbing material is not the fiberglass insulation universally used in other enclosures but long-fiber wool. Transmission line speakers with awesome bass output can be made surprisingly small—perhaps the best example of this being the Obelisk produced by Shahinian Acoustics. The speaker system, which measures only 26¾ × 14 × 12 inches (67.9 × 35.6 × 30.5 cm), can really hold its own on pipe organ music. What you gain in clean bass from a transmission line you pay for in terms of efficiency. These systems are generally less efficient than bass reflex designs.

The Passive Radiator. A more direct spin-off of the bass reflex enclosure is one that replaces the port with a passive diaphragm—the passive radiator. Although the acoustic impedance seen at the diaphragm "vent" is different from that in a ported system, the passive radiator works according to the same basic principles. The box resonates at a frequency near the fundamental resonance of the active driver, at which point the passive radiator augments the output of the driver.

Fig. 3. The acoustic labyrinth loudspeaker enclosure.

It is worth mentioning that ported designs in general have reached a high level of sophistication in recent years due to the application of a unified theory of vented-box speakers developed by A. N. Thiele. Thiele's theory has enabled designers to optimize bass reflex designs in a relatively straightforward manner. The result has been an increasing number of ported systems without many of the uneven response problems of the older bass reflex systems.

Horns. No discussion of speaker enclosures would be complete without some mention of horns. A number of speaker systems available today use horns on some of the individual drivers, almost always the midrange and high-frequency units. A horn is basically an acoustic transformer that couples the air at the surface of the driver's diaphragm with the air in the listening room. The "throat" of the horn is small and fits over the diaphragm. Its "mouth" is much larger and conducts the radiated sound waves into the room.

The horn permits driver displacement to be small without sacrificing acoustic output. This means lower distortion and higher power-handling, as well as increased efficiency. The dimensions and shape of a horn must be carefully determined for the device to function without acting as a resonant tube. In addition, the interface between the driver and horn must be set up to avoid phase cancellation due to diffraction. For this reason, "phasing plugs" are used to couple the driver with the horn.

You'll find horns used mainly as midrange and high-frequency drivers in most hi-fi systems because they are of manageable size. Predictably, the effective horn length required for bass frequencies is very large. (You can sometimes see bass speaker bins with straight horns at rock concerts. They are very long, as much as 6 to 8 feet or 1.8 to 2.4 meters.) One solution to the size problem is the "folded" horn in which the horn is folded back on itself to decrease the overall size of the cabinet. The famous Klipschorn speaker uses this principle, as shown in Fig. 4. However, the horns are difficult and expensive to construct.

Most horn designs seen today are not new. Folded horns, multicell horns, and sectoral horns have all been around for a while. One interesting design used by JBL, among others, involves the use of an acoustic lens at the horn mouth. These lenses look somewhat like the louvers on a ventillation duct. Their purpose is to alter the directional characteristics of the horn—an important task, because many horns tend to beam severely in parts of their frequency ranges, resulting in a shrill sound. Other developments in horns include improved structural designs for making the horn rigid and lightweight (and less expensive), and the use of novel phasing plugs such as the "Tangerine" developed by Altec.

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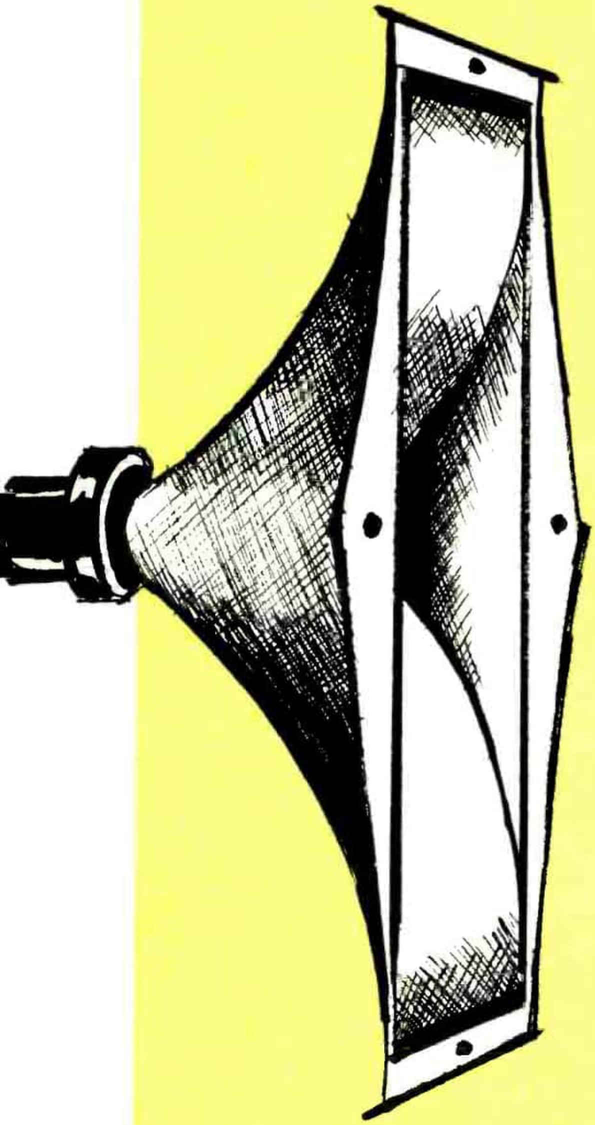


Typical horn loudspeaker.

The System.

If a speaker design is to be successful, it must be engineered as a *system*, with each component designed to work harmoniously with all the others. Accordingly, there have been design innovations that involve the overall system rather than one particular component such as the driver, crossover network or enclosure. Some of these innovations flow from considering the listening room as part of the audio system. Others are due to increased attention to phase response, low-bass response, etc. Let's look at how this systems approach is affecting loudspeaker design.

The Room/Speaker Interface. From the loudspeaker engineer's point of view, the least controllable element in any audio system is the room/speaker interface and a number of speaker designers have directed their attention to trying to cope with it. An excellent example of this heightened sensitivity to room effects is the well-known Bose 901 series of loudspeakers, whose design is based on what Bose calls Direct/Reflected Sound. The 901 system employs eight rear-firing drivers and one front-firing driver. Its rear baffle is angled to direct the two sets of four drivers each to different room areas, and the front baffle has a mild curvature. This system, when used in a reasonably reverberant room, will reflect large amounts of sound off the walls, creating a reflected sound field that works with the direct sound of the front driver to produce a feeling of spaciousness and a "concert hall" effect. Another interesting feature of the design is the fact that its drivers are all 4-inch cone radiators. Low-frequency reproduction is achieved by constructive interference of the many drivers, whose individual outputs combine in phase at low frequencies to give the desired bass output. These drivers are somewhat limited in high-frequency response, so an equalizer is provided with the system to adjust the characteristic of the drive signal accordingly.

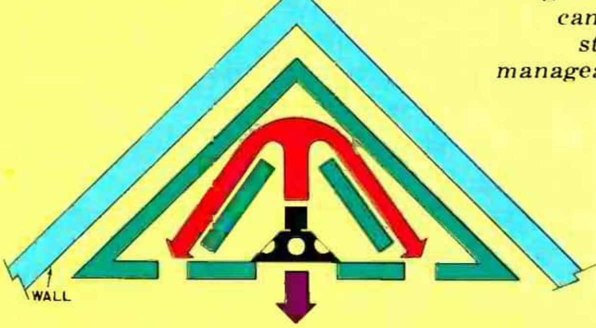


Fig. 4. Large bass horn can be folded into a structure of more manageable dimensions. Walls of room act as extensions of the horn.

Allison Acoustics also designs its speakers with the listening room in mind. The Allison Model 1, for example, has a prism-shaped columnar enclosure with side-firing, acoustic-suspension woofers placed to minimize frequency-response aberrations due to room reflections. Allison's Model 4 is a true bookshelf system designed to use the shelf to its advantage. The woofer is placed at the top of the cabinet and effectively radiates into the shelf. Again, standing-wave formation is minimized with this scheme. The top-of-line AR speaker, the AR-9, also employs side-firing woofers.

It should be mentioned that *all* speakers are to some extent sensitive to room placement. As a rule, those designed to take the listening room directly into account are, not surprisingly, more sensitive to room placement than others. To get smooth frequency response and optimum directional characteristics, speaker placement must be experimented with until the "right" location for a room is found.

A number of speakers on the market today permit you to alter their directional characteristics to suit a particular room. One representative system is Infinity's massive Quantum Reference Standard, which has both forward- and rear-firing tweeters mounted on a cabinet with "flaps" that swivel in the horizontal plane. Moving the flaps enables you to change the dispersion characteristics of the speaker. The Leak Model 3090 speaker system has its mid-range and high-frequency drivers in a separate enclosure from the woofer. The two enclosures are connected by a swiveling mount. This enables the listener to change the firing position of the woofer relative to the higher-frequency drivers. For example, the woofer can be aimed at a nearby wall, and the midrange and tweeter pointed directly at the listener.

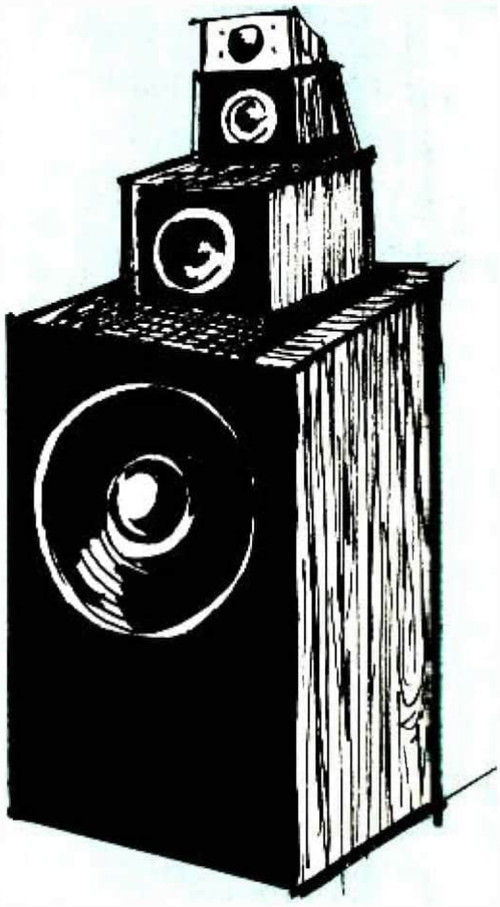
"Linear Phase" Speakers. The philosophy of the new linear phase speaker is this: if phase effects are important in program listening, the hi-fi system should have flat phase-vs.-frequency response as well as flat amplitude-vs.-frequency response. For a speaker to have flat phase response, it should introduce no phase shift that can be detected at the listener's position. Now, the typical multi-way speaker system does introduce phase shift, bcause of the effects of its crossover circuits, and because of the fact that the different drivers, mounted on a flat baffle, have effective centers of radiation that are spatially displaced, resulting in a path-length difference between each driver and the listener's ears. In fact, the phase shift introduced in the crossover is the more serious of the two because there is a large discontinuity in the phase response of the network at the crossover frequency. The sharper the crossover's filters (say, having rolloffs of 12 or 18 dB/octave as opposed to 6 dB/octave), the greater the discontinuity. On the other hand, the path length difference resulting from the mounting of the drivers creates a time delay on the order of 1 millisecond, which translates to about 6 degrees of phase shift at 1000 Hz.

Bang and Olufsen was a major force in the early work on phase-compensated speakers. The most interesting work in this area has involved the design of crossover networks that eliminate phase anomalies. The ideal crossover for flat phase response uses active circuits (that is, multi-amping is necessary) and a "filler" driver that is active over a narrow band of frequencies centered at the crossover frequency. The filler driver is not intended to radiate significant acoustic power over a wide bandwidth. Rather, it's there only to smooth out the rift in the phase-response curve without serious disruption of the amplitude response.

In Bang and Olufsen's phase-compensated speakers, the filler-driver/crossover combination is employed and the drivers are mounted on an angled baffle. The angle of the baffle is set so that, at the hypothetical listening position, there is no path-length difference between the two drivers. Even though B&O systems use filler drivers, they don't require multi-amping because the filler drivers are carefully designed to work with passive crossover networks (they are more efficient than the main drivers, for one thing).

A number of other manufacturers, such as Technics and B&W, have produced systems intended to have flat phase response. All of these systems utilize offset mounting of the drivers, with the tweeter generally set several inches in back of the woofer to eliminate path-length differences. Often, for example, the main cabinet houses the woofer and the midrange and tweeter are mounted on small baffles set back on top of the woofer enclosure. It's worth noting that the Walsh driver mentioned earlier (it's employed in the Ohm F system) is, besides its other qualities, a true linear-phase speaker.

It must be said that there is no agreement as to whether a speaker with flat phase response is necessary. In fact, a growing body of evidence indicates that, when music or speech program material is heard in a reverberant environment, phase effects are of little importance to the listener. It is only with specialized test signals and very specialized listening environments that phase effects other than basic phase cancellation (destructive interference) can be heard. After all, the typical listening room, introducing sound reflections as it does, totally randomizes the phase information in the acoustic signal by the time it reaches the listener's ears. It has been said that "it can't hurt" to have a linear-phase speaker, which is true, ideally. But a number of designs currently on the market introduce amplitude response aberrations due to diffraction effects caused by the offset mounting of the drivers. These response anomalies *are* audible. This is worthy of consideration, because a number of manufacturers go to great pains to make their baffles smooth and free of structural discontinuities simply to minimize diffraction of high-frequency sound (consider the latest designs by Avid, for example).

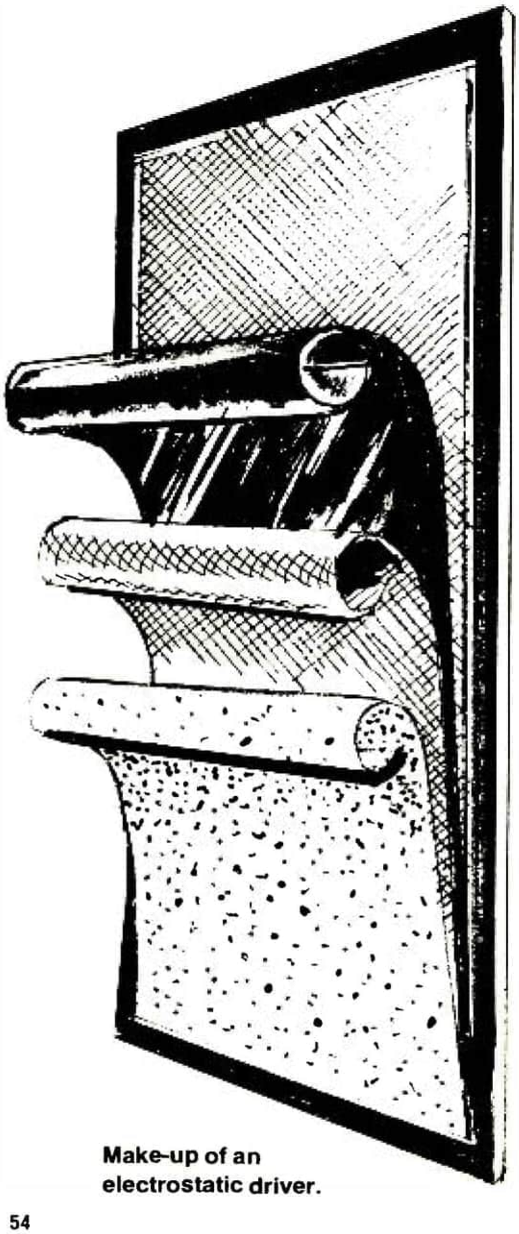


"Linear phase" speaker system.

Electrostatic Systems. As mentioned earlier, a solution to the limited-output problem of electrostatics is exemplified by the formidable Dayton Wright electrostatic speaker. The electrodes and diaphragm are immersed in an atmosphere of SF_6, which can support very high polarizing and driving voltages without breaking down and arcing. Hence, the Dayton Wright system is capable of generating more acoustic output than standard electrostatics. The system is provided with a special power amplifier that is capable of driving the almost totally reactive load of the speakers (which would be a horror for many conventional amplifiers). The Dayton Wright is a two-way system. Frequencies above 10,000 Hz are handled by a piezoelectric tweeter.

Another novel (and equally massive) electrostatic system is the Beveridge 2SW. This speaker employs a single 6-foot electrostatic panel, but the driver is loaded with a unique acoustic lens that gives the system wide dispersion in the horizontal plane. Ordinarily, a single electrostatic panel would become highly directional at high frequencies, especially if it were large enough to provide significant bass output. In the Beveridge speaker, the effective size of the radiator is quite small, because the mouth of the lens is a narrow slit rather than a large, wide panel. The Beveridge system is provided with a specially-designed vacuum tube power amplifier as well as two subwoofers to augment the electrostatic driver's bass output in the lowest octave.

The Beveridge 2SW represents an intresting means of obtaining controlled high-frequency output from an electrostatic speaker. Other designers take different approaches, such as multi-way systems with two or more electrostatic drivers of different panel sizes. Some manufacturers, such as RTR, utilize electrostatic tweeters composed of several small panels placed at angles to one another so they radiate into a wide solid angle.



Make-up of an electrostatic driver.

Equalized Systems. Some designers utilize electronic equalizers to compensate for frequency response deficiencies of the speaker system, as is done in the Bose 901 series mentioned earlier. These equalizers are generally placed between the preamp and power amp, and have either fixed equalization or a minimum of user controls so their performance isn't hindered by user adjustments. Another example of equalized systems is Electro-Voice's Interface line of speakers, some of which include equalizers that smooth out bass response and control high-frequency roll-off.

Powered Speakers. Speaker designers who want absolute control over the electronic circuits to be interfaced with their systems use powered speakers. These speakers contain power amplifiers that are built right into the speaker cabinet. Systems of this kind are almost always multi-amped. That is, the system will contain a separate amplifier for each driver, as well as an electronic crossover at the input, so you drive the "speaker" with the output of a preamp. There are some obvious advantages to this scheme. For one thing, it is probably the most cost-effective way to drive a speaker system. Multi-amping enables you to match each transducer with its appropriate drive level, and therefore gives you a maximally efficient system. Additionally, there are none of the power losses that occur in higher-order passive networks, so you can provide each driver directly with only the power it needs and no more. (This isn't really critical in hi-fi systems, but if you're setting up a stage system for the Grateful Dead or designing a monitor system for a big studio, it is.) Other advantages of multi-amping are potentially lower distortion and smoother frequency response. Regarding the latter, the active crossover can be designed to introduce no phase shift of one band relative to another. Accordingly, near the crossover point, where more than one driver is radiating the same signal, destructive interference won't occur. In any event, if multi-amping is important in a particular application, the powered loudspeaker is an efficient way to do it. Another feature, exemplified by the Powered Advent speaker system, is compactness. The bulky power amps are built right into the cabinet.

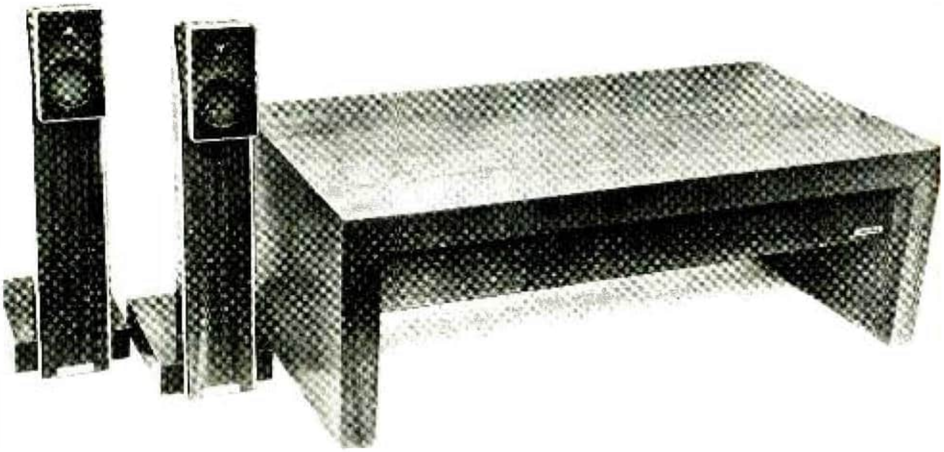
The ultimate powered speaker system is one that utilizes *motional feedback* to correct for driver nonlinearities such as in the Philips RH-545. This is a three-way tri-amplified system with a small transducer mounted on the woofer cone. The transducer generates a signal proportional to the woofer output, which is fed back and subtracted from the input signal (just like the negative feedback used in every audio amplifier). The signal that drives the speaker is thus "compensated" for the woofer's nonlinearities. Feedback systems of this sort are primarily useful for reducing distortion in a speaker's bass output. They aren't generally used for the higher-frequency drivers for practical reasons.

Subwoofers. A relatively recent trend in the design of loudspeaker systems involves the use of large dynamic drivers for reproduction of the deepest bass and separate "satellite" speakers for the upper bass, midrange, and treble frequencies. Because audio frequencies below 70 Hz or so are nondirectional to the human ear (it can't locate their source), a system can be set up to operate with a single subwoofer covering the range of, say, 20 through 70 Hz, and a pair of satellite speakers for the higher frequencies. (Alternatively, a separate woofer can be used for each channel and its operating range increased accordingly.) As evidence of this trend you need only note the increasing number of "mini" speakers on the market, made by manufacturers like ADS, Visonik, Braun, etc. If the satellites are not intended to reproduce any low bass (anything significantly below 100 Hz) they can be made very small. The ADS Model 200 II, for example, measures only 6¾ × 4¼ × 4⅝ inches (17.1 × 11.7 × 10.8 cm) and has a rated frequency response of 85 to 20,000 Hz ±3 dB. (Response goes down to 55 Hz ±6 dB.) The advantage of using a single subwoofer in a low-profile cabinet and two small, lightweight satellites is clear from the standpoint of aesthetics and placement in the average room. If desired, the woofer can be hidden in an out-of-the-way place because its output does not affect stereo imaging. The compact satellites are easy to find room for.

Acoustique 3A International has

taken the idea of the hidden subwoofer to its extreme with its Triphonic system. Here the woofer is a self-powered, motional-feedback system concealed in a coffee table. The higher frequencies are handled by two small (17 × 10 × 7 inches) satellite speakers. A single stereo amplifier is required to drive the system because the woofer is self-powered.

One disadvantage of the subwoofer/satellite approach in terms of cost is the fact that these systems generally have to be bi-amplified. An electronic crossover and separate amp for the woofer is needed. If the woofer is reproducing frequencies down to 20 Hz, and is of the acoustic suspension type (as is not uncommon), you'll need plenty of power for it, too!



Mesa's subwoofer housed in an enclosure disguised as a coffee table is meant to be used with a pair of small satellite speakers (left).

In Conclusion.

Although there aren't all that many generic types of speakers and enclosures, the number of variations on the basic designs that have been tried at one time or another is huge. Each design attempts to solve the problem(s) its creator sees as most pernicious (whether they are real or imagined). However, one thing is certain—as time has passed, the fundamental designs have been improved and refined to the point that well-executed examples of each design type will *all* be good-sounding speakers. ◇

Interpreting Speaker Test Results

BY JULIAN HIRSCH
Hirsch-Houck Laboratories

FEW IF ANY difficulties are encountered in testing electronic audio components such as amplifiers and tuners in a hi-fi system. The reason for this is that we are dealing with generally well-understood effects (power, sensitivity, distortion, noise, etc.), and the measuring procedure is defined by standards issued by the Institute of High Fidelity (IHF) and other organizations. Hence, we can reasonably well define the performance of such products under test. Not so for speaker systems, which have yet to come under a testing standard (although an IHF technical committee has been activated to generate such a standard).

Unlike electronic components, the electromechanical loudspeaker is far from a simple device with a uniquely defined output that can be directly related to an input signal. Until a testing standard is formulated, those of us who test speaker systems must establish our own laboratory procedures based on what we consider to be important characteristics. This is the rub, because as things now stand, there is no agreement as to just which characteristic of a speaker system's output is related to its sound and which, therefore, should be measured.

The Problems. Although the input to a speaker system is electrical and can be specified quite well (assuming one considers it as a voltage only because the complex input makes it very difficult to measure the true input power), its output is *acoustic* energy. This energy is commonly expressed as a sound pressure level, or SPL, in decibels relative to 0.0002 dynes/cm^2. The SPL measurement per se is not difficult to make with a suitable calibrated microphone. However, there remains the question of the testing environment and the physical relationship between the speaker system and microphone during the test.

The acoustical SPL output of a speaker system is a function of the direction and distance between the microphone and speaker system and the driving frequency. If the frequency response is measured in an anechoic chamber—the most commonly used test environment, with the microphone on the central axis of the system—one may obtain a reasonably "flat" response curve. Bear in mind, though, that "flat" for a speaker system is not the same as "flat" when applied to the response of an amplifier or even of a phono cartridge. Every driver in a speaker system is subject to countless unwanted response variations from its cone and voice-coil suspensions, the cone itself, the supporting basket, the cabinet edges, and many other factors. As a result, what is obtained in an actual test is a very rough, irregular response curve. This response curve is usually so irregular that its useful content is obscured. Hence, it is common practice to use filtering and pen damping in the graphic level recorder to smooth out the rough edges, so to speak, leaving a more general—and often more informative—contour of the speaker system's response.

No matter how the measurement is made, we cannot avoid the fact that the response will be different for every different microphone position relative to the speaker system. An on-axis response is essentially worthless as an indicator of how the speaker system will sound or even of its intrinsic merit. If the response is measured at a number of different microphone locations while sampling the sound field over a wide angle in front of, or even through a full sphere surrounding, the speaker system, it is possible to process the data with a computer to obtain a plot of the total power response as a function of frequency. This is a measurement of the total acoustic energy the speaker system radiates in *all* directions, into the hemisphere or spherical volume that loads the system.

There is good reason to think that the power response of a speaker system is more closely related to the way it sounds in a real listening room than in any anechoic measurement along a single axis or several axes. This is not to imply that such a measurement—indeed, *any* possible measurements—will ever be able to define or describe the performance of a speaker system with the accuracy that such measurements can describe the nature of amplifier or tuner performance. There are many orders of magnitude of difference between the subtleties detectable by the human ear and processed by the brain and anything that can be picked up by a microphone and processed by a computer. Nevertheless, in a home environment a speaker system *does* radiate sound in all directions although not necessarily uniformly. Most of this output, after reflection and some absorption, eventually reaches the listener's ears. For this and other reasons, we have long felt that such a measurement (power response rather than any single-axis pressure response) is the most meaningful way to measure the general, octave-by-octave, frequency response of a speaker system.

Our Procedure. Fortunately, a computer or a large number of microphones are not really needed for a power-response measurement. Another and often simpler technique is to use a reverberant chamber for a test environment. This is an odd-shaped room with no parallel surfaces, its walls, floor, and ceiling made of hard, nonabsorbing material. All the sound emitted by a speaker system in such a room, after many reflections, will produce a uniform, homogenized sound field at any point in the room. Like an anechoic chamber, a reverberant room can be used only for middle- and high-frequency measurements. A reasonable-sized anechoic room may be useful only down to 100 Hz or so, and the cut-off for a reverberant room can go up to 500 Hz.

At Hirsch-Houck Laboratories, we have neither an anechoic nor a reverberant chamber. We have found that a normally furnished listening room behaves much like a reverberant chamber. Beyond a distance of 10 ft (3 m) or so from the speaker system, the sound field is semireverberant. That is, the measured SPL changes little as the microphone is moved about. The room's response is not "flat" with frequency, of course, due to the normal absorption by the boundary surfaces and furnishings. These surfaces cause the high-frequency response to roll off in any measurement. However, we have been able to compensate for this rolloff with gratifying success. To do this, we measure the response of two calibrated speaker systems in a normal stereo listening setup at the front of the room, spaced about 10 feet apart. We place the microphone on-axis with and about 12 ft (3.7 m) from the left speaker system. At this point, the microphone is angled about 40° off-axis from the right speaker system. Then we run response curves for both speaker systems separately but on the same chart paper, using a "warble tone" swept oscillator and heavy pen damping in the recorder to obtain the smoothest possible curve. The two curves, from 100 to 20,000 Hz, usually have "ripples" from standing wave effects in the room. We find, however, that these standing waves tend to cancel out between the speaker systems. This yields a relatively smooth line average for a reasonable curve. (Fig. 1).

Knowing the actual reverberant-room response of our speaker systems from data supplied by the manufacturer and knowing that our microphone is flat within ±1 dB up to 20,000 Hz, we can draw a correction curve. When this curve is added to the response curve of any other speaker system measured in the same room, we obtain a response curve roughly equivalent to the speaker system's total power response. Although acousticians will probably flinch at the approximate nature of this measurement and the various assumptions that have been made in its performance, we find that we usually come

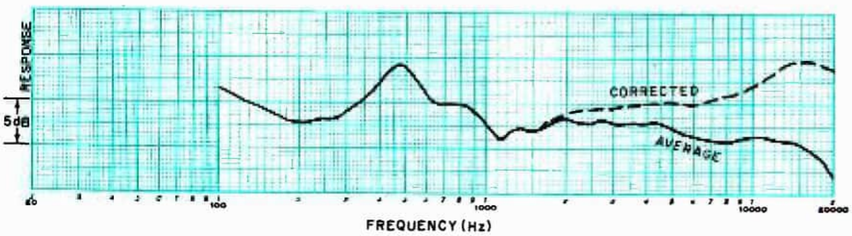


Fig. 1. Average response curve is for left and right speakers and corrected curve takes room characteristics into account.

within 2 or 3 dB of a true reverberant response measurement when we have access to that data from a speaker manufacturer. This is accurate enough for our purposes.

Although the curve obtained extends down to 100 Hz, it is not valid below about 300 Hz because of the unavoidable resonance effects of our room. At lower frequencies, we place the microphone as close as possible to the woofer's cone and run a curve from 20 to 1000 Hz (Fig. 2). (No warble tone is needed since close-microphone measurement is not affected by the room and gives essentially a true anechoic response.) This measurement is not valid much above 300 Hz, where the dimensions of the woofer cone become comparable to the wavelength of sound, but we carry it up to 1000 Hz for the same reason that our quasireverberant measurement is carried down to 100 Hz.

Having obtained two sets of curves, we splice them together to form a single composite frequency response (Fig. 3). By extending each curve beyond its most accurate range, we find it easier to overlap them for the best splice. On occasion, there is little overlap range and we must make an educated guess about the splice point. In many cases, the two curves have considerable overlap and there is no doubt about the accuracy of the composite frequency-response curve.

One might fairly ask how we can call this composite curve a "frequency response" plot of a speaker system. Our reasoning is that the midrange and high-frequency portion of the curve is a fair representation of the speaker system's total power output over that range and consequently represents how much energy it will radiate into almost any room. Regardless of the room's characteristics, a peak or a drop in output at the high frequencies will almost always be heard as a brightness or dullness of the sound, and irregularities in the midrange generally correlate with difficult-to-define colorations one often hears from a speaker system. While this portion of the curve is fairly applicable to any listening room, the bass response is drastically affected by the room size and placement of the speaker systems. In most cases, it would be quite impossible to produce any curve that really reflects how the speaker system will perform in any arbitrary room.

Given the above, we content ourselves with showing the low-frequency (anechoic) response obtained with close microphone spacing. This is a "worst-case" condition; the actual bass performance will always be better in any real room, where the boundaries will reinforce the low-frequency output of the system. The curve we give is useful for comparing speaker systems. The magnitude and width of the low-frequency output rise at resonance also give a good indication of the Q of the woofer.

In spite of its unorthodox derivation, the composite response curve relates well to how the speaker system sounds in our test room and how it is likely to sound in most "real" listening rooms. It is worth noting that the two separate response curves at high frequencies reveal very clearly how good the dispersion of a tweeter is in the upper registers. An omnidirectional speaker system, or a system with very good dispersion, will reveal little or no difference between the two curves, one on-axis and the other 40° off-axis, at any frequency, but most speaker systems will reveal at least several decibels difference at frequencies beyond 10,000 Hz.

Other Tests. We also measure the harmonic distortion of the woofer between 100 Hz and its lower frequency limit (Fig. 2). We use the same close microphone technique here employed for the bass response measurement, but the microphone's output goes to a spectrum analyzer instead of a chart recorder. The output levels of second and third harmonics (higher order components are almost never significant) are combined to obtain a THD reading at each 10-Hz frequency increment. The woofer is driven with a constant 2.83 volts and then at a constant 8.94 volts, which correspond to power levels of 1 watt and 10 watts into an 8-ohm load. We do not measure distortion at higher frequencies because the irregularity of the system's response makes the test impossibly complicated unless very specialized automatic plotting equipment is used.

A bass distortion measurement tells

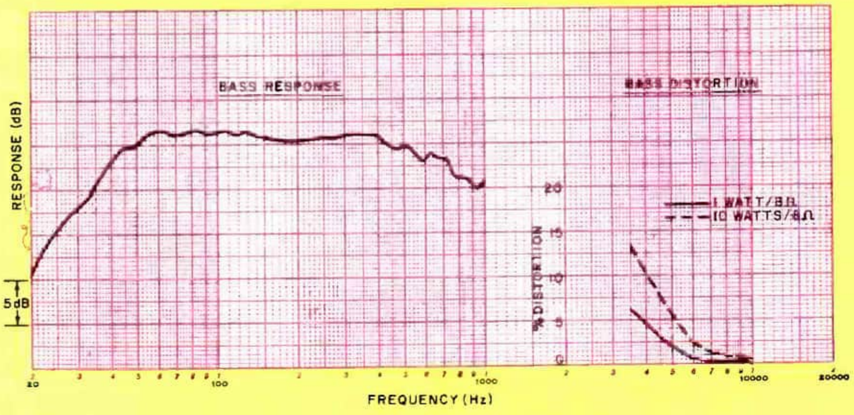


Fig. 2. At left is a typical bass response from 20 to 1000 Hz. At right are typical distortion curves below 100 Hz at two power levels.

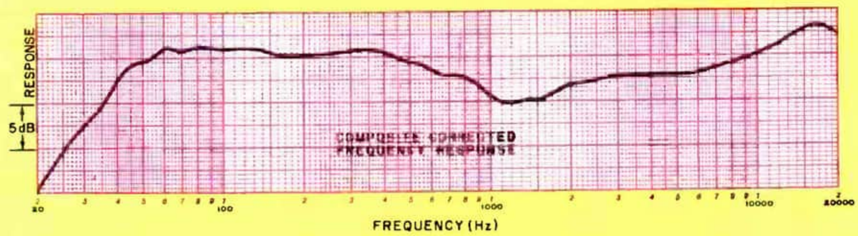


Fig. 3. A composite corrected frequency response. It is formed by splicing the curves from 100 to 20,000 and from 20 to 1000 Hz.

us how low the speaker system will go before the woofer ceases to be coupled to the air load of the room. At this frequency, distortion begins to rise rapidly. For a typical bookshelf speaker system, the distortion may measure in the vicinity of 1 percent or so down to about 60 Hz, rising somewhat at 50 Hz and very abruptly at lower frequencies. Regardless of its measured frequency response (measured at a low drive level), such a system will not produce usable, undistorted bass at frequencies much below 50 Hz because its cone is no longer coupled to the room, causing excessive distortion.

One thing everyone should know about his speaker system is how much drive is required to produce a given SPL output. (This is related to how much amplifier power capability is needed.) This information is given in our *sensitivity* measurement. (This is not *efficiency*, although the two are related, since we do not know how much electrical power we are actually delivering to the speaker system, or how much acoustic power it is producing.) We drive the speaker system with 2.83 volts of band-limited pink noise (an octave wide and centered at 1000 Hz) and measure the SPL at 1 meter in front of the grille. We prefer this signal to full-bandwidth pink noise because it measures the sensitivity in the important midrange, where most of the subjective volume is produced.

Impedance is measured by driving the speaker system's voice coil directly from the swept oscillator of our frequency response plotter, with the graphic level recorder in parallel with the output of our oscillator and the speaker system. The 600-ohm output of the oscillator looks like a constant-current source to the speaker system. Hence, the voltage across the speaker system and recorder is directly proportional to the impedance as the frequency sweeps from 20 to 20,000 Hz. Since the amplitude scale of the recorder chart is logarithmic, we calibrate it before each measurement by substituting a precision resistance decade box for the speaker system and calibrating over a range of 1 to 100 ohms (Fig. 4).

Although we still examine the tone-burst responses of a speaker system as part of our test sequence, these do not lend themselves to objective interpretation unless the speaker system is unusually good or bad. We usually drive the speaker with 4-cycle bursts while varying the frequency and observing the behavior of the acoustic output in the operating range of each driver. It is generally necessary to locate the microphone fairly close to the speaker system (within a foot or so) and on the axis of the driver being studied to avoid interference from the other drivers and room reflections. This is one of the tests that really should be made in an anechoic chamber, since it is almost impossible to eliminate room effects from delayed reflections off the walls unless the microphone is so close to the speaker that it may affect the actual output. (One cannot assess total system performance under this condition.)

Interpreting Results. From the general shape of the composite response curve, one can determine by inference whether the system will sound bright or heavy, have midrange coloration, or perhaps be one of the few really smooth and uncolored reproducers available. A very flat curve usually means that the speaker system is very good, but moderate irregularities do not necessarily mean that the system is a poor performer. Our comments on the sound of a speaker in the "User Comment" section of our reports should help in interpreting the curves.

Very low bass distortion usually means that a speaker system will sound clean in the low bass range and that it can probably be equalized for a flatter, more extended bass output with a suitable graphic equalizer, without risking excessive distortion or damage. Higher bass distortion, such as 2% or 3% in the 50-to-100-Hz range does not necessarily sound bad since the ear is very tolerant of low-order harmonic distortion, especially at low frequencies. If the speaker system is capable of delivering a healthy output in the low-bass range, a moderate amount of distortion will probably never be noticed.

The impedance curve is important to anyone who plans to parallel two pairs of speaker systems on a single amplifier. We give in our reports the lowest impedance observed.

The sensitivity rating is a rough guide to how much amplifier power will be needed to drive the speaker system to a given SPL. Most acoustic-suspension systems produce an 85-dB SPL at 1 meter for a nominal 1-watt input (the range is typically 82 to 88 dB). Ported or vented systems may range from 88 to 92 dB, while a few will reach 94 or 95 dB. These figures do not tell how loud a speaker system will play; they tell you how much power is required to play at a given level. Their usefulness is mainly to permit you to compare competing systems. For example, a speaker system rated at 92 dB will require only one-tenth as much amplifier power as one with an 82-dB sensitivity rating, for the same listening level.

Finally, we should mention that our test figures will rarely, if ever, agree with any supplied by a speaker system manufacturer. This is because totally different test methods were used, which is not a reflection on either the manufacturer or on our test results. ◇

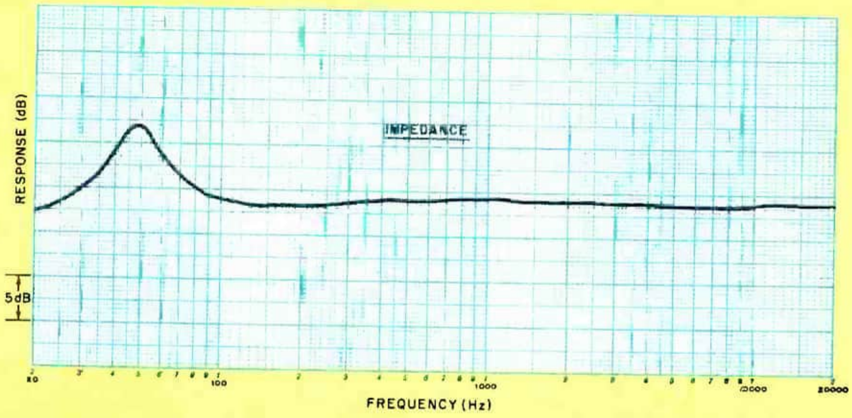


Fig. 4. A typical impedance curve obtained by driving the speaker system. from the swept oscillator of frequency response plotter.

Special Focus on Speakers

The Importance of Power-handling Capacity

BY TIM HOLL
Teledyne Acoustic Research

THE AMOUNT of power a speaker can handle without incurring damage is little understood by most audio enthusiasts. For example, why can a loudspeaker with a high power rating be damaged in a particular situation when one with a lower rating is undamaged? Why do some speaker systems that are provided with fuses still suffer overload damage without the fuse being blown? To answer these and other questions, we must first recognize the power limitations of a loudspeaker system in two failure areas—thermal and mechanical.

Thermal Failure. To understand why thermal failure occurs, we first determine what a loudspeaker does with the input power it receives. We are familiar with audio amplifiers that deliver 50, 100, 200, or even as much as 700 watts per channel. However, few of us know how many watts—that is, actual acoustic watts—a loudspeaker delivers. An indication can be obtained, however, by considering the output of a large pipe organ. It will typically deliver 12 to 14 acoustic watts in a spacious environment. A conventional saxophone, on the other hand, delivers about 0.3 watts, a piano 0.4 watts, a bass singer 0.03 watts and speech at a normal level about 0.000024 watts.

Obviously, we are now looking at much smaller power levels than those considered by the average hi-fi enthusiast because he is thinking of electrical input to a speaker, not its acoustic output. The enormity of the difference is seen if we consider the 12 to 14 watts delivered by a large pipe organ. In the average house, this amount of acoustic power would literally shake the house.

Thus, we can see that, for even extremely loud listening levels in the home, we are only considering acoustic power outputs of no more than a few watts. To supply this power, however, it is often necessary to have an amplifier with a large power output because of speaker inefficiences. This brings up the subject of just how efficient high-efficiency speakers are when compared to low-efficiency units—how inefficient even the high-efficiency systems are is not generally appreciated. Acoustic-suspension speakers can have efficiencies as low as 0.2%, ported systems about 1% and horns up to approximately 15% to 20%. For most current speakers, for every 100 W of electrical power delivered to a speaker, only about 0.2 to 1 watt of acoustical power is delivered as actual sound! The rest of the power (over 99 watts, in this case) goes almost entirely into heating the voice coils on the speaker drive units. (A very small amount of power is used to overcome mechanical resistances in the drive units while another small amount heats the leads from amplifier to speaker).

Let us now look at what happens when we apply this power to a speaker's input. Figure 1 shows typical heating and cooling of a conventional midrange unit with a 1"-diameter voice coil for a constant sine-wave input. A steady state is rapidly reached around 105°C (221°F) above ambient (about 20°C or 68°F). Usually, thermal breakdown occurs when adhesives used in the construction melt or fail. This is

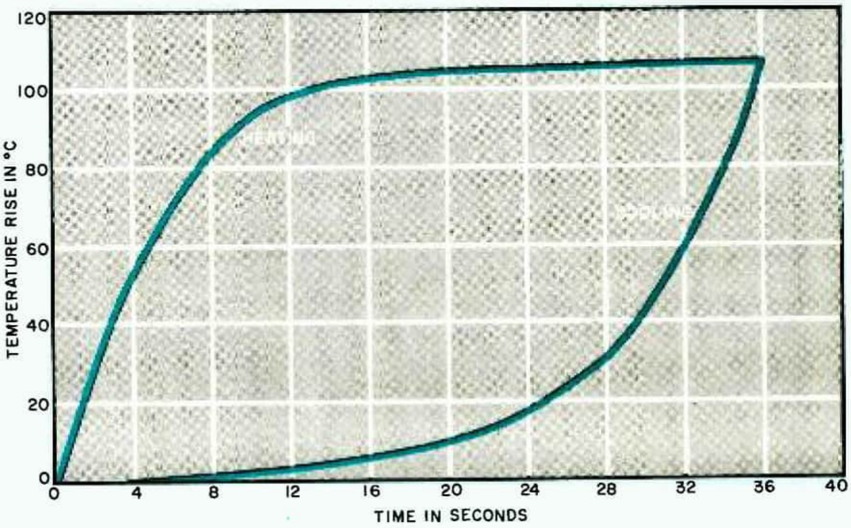


Fig. 1. Heating rate of a midrange coil when a nominal 10 watts of 1-kHz sine wave is applied, and the cooling rate when the signal is removed.

Special Focus on Speakers

"...a loudspeaker cannot [always] be played as loud after several hours of use as it can when first turned on."

usually at temperatures of about 170°C (338°F) to 180°C (356°F). Consequently, it would appear to be safe to use 10 watts continuously applied to our hypothetical unit. Unfortunately, it is not!

To understand why, let's first examine what happens when the temperature rise apparently levels out. At this point, all the power that's producing heat is not merely raising the temperature. It is travelling across the air gap in which the coil sits to the other metal parts and magnet (Fig. 2) causing them to heat up. Since the metal parts have a larger mass than the voice coil, however, the former is slower in reaching its maximum heat level. As shown in Fig. 3, it takes about two hours for the temperature to stabilize to a point where all the heat received by the structure is, in turn, transmitted to the air around the unit. During this time, the metal rises to about 68°C (154°F) above ambient. But, more significiantly, the voice coil rises with it to about 155°C (311°F) above ambient--close to the failure level of adhesives.

What does all this mean to the loudspeaker user and loudspeaker designer? We need to look even further into the subject to see how the designer can optimize the situation, but we already see one danger point for the user. A loudspeaker voice coil will rapidly heat and cool with variations in level applied with a typical music signal, as indicated by Fig. 1. However, this rapid variation in temperature will have upper levels determined by how long the system has been playing, as seen in Fig. 3. Consequently, a loudspeaker cannot be played as loud after several hours of use as it can when first turned on. This may explain why simple fusing so often fails to protect a system. A low-current fuse which would provide adequate protection for all signals after any period of playing is simply not practical, as it would mean limiting the system to unrealistically low levels for most normal listening conditions.

What can the designer do? One obvious answer is to transfer more heat away from the unit, possibly with heat-sink fins on the metalwork. Unfortunately, this will work only if he can efficiently transfer the heat from voice coil to metalwork. This is where we have the major temperature differential in the order of 90°C (194°F). If we could reduce this differential to, say, 20°C (68°F), then for the same input power of 10 watts, we would only get a long-term voice coil rise of about 90°C (194°F) instead of 155°C (311°F). This would occur because, under steady-state conditions, about 9.9 watts is generated as heat in the voice coil, crosses the air gap to the metalwork, and is then transmitted by the metalwork to the surrounding air. The temperature difference necessary between coil and metalwork for this 9.9 watts is about 90°C. How can it be reduced to, say, 20°C (68°F) and still transfer the 9.9 watts? To answer this takes a complex study of heat transfer mechanisms. The results of one such study are described below.

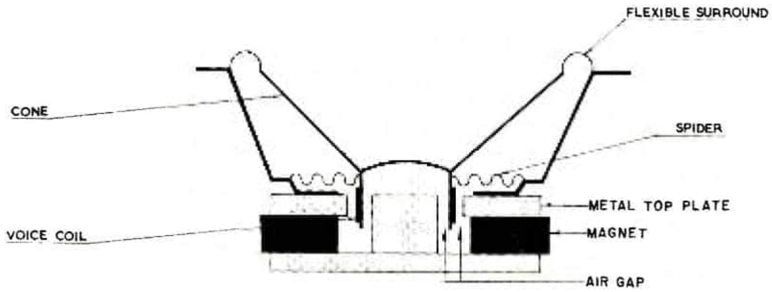


Fig. 2. Cross section of a typical loudspeaker drive unit with voice coil sitting in air gap.

In this study it was found that in the type of unit used as an example, about 3% of the heat was transferred from the coil to the metalwork by radiation, none was transferred by convection, and 97% was conducted through the air in the gap. This explains the high temperature differential between coil and metalwork, as air is a fairly poor conductor of heat (the air in our homes is heated by convection, a mechanism that does not occur in the voice coil gap). Attempts to improve radiation by having blackened coil formers and blackened metalwork had little effect, giving an increase in power handling of only 12%. To make inroads, we need improvements of several hundred percent, as each doubling in power handling means that we can safely play the system 3 dB louder (12% is an improvement of 0.5 dB in output).

The obvious answers to improvement in power handling lie in two areas—increasing the maximum temperature the unit can withstand and improving heat conduction away from the voice coil. Higher maximum temperatures require the use of adhesives that will withstand higher temperatures without softening or breaking down. As stated earlier, these adhesives normally have an upper limit of about 189°C. Adhesives that can withstand higher temperatures are now becoming available and are being used on a few drive units. Generally speaking however, most voice coils still have the sort of temperature limit we have been dealing with.

We are thus limited to the upper temperature limit of about 180°C (356°F). Is it possible to handle more power before reaching this limit? One obvious way is to increase the surface area of the voice coil. High-power speakers for electric guitar amplification, for example, often achieve their power-handling ability by using tweeters with large voice coils—sometimes 4 or 5 inches in diameter. Unfortunately, large voice coils cause two serious degradations in tweeter or midrange performance. First, for accurate reproduction of transients and high frequencies, a tweeter voice coil has to be lightweight, so it can't be very large. Secondly, a large voice coil automatically means a large acoustic radiating surface which, in turn, means a tweeter that will become seriously directional even at relatively low frequencies.

Another way to handle more power *for a given sound pressure level* before reaching this limit is to use a much more efficient tweeter and pad it down in the crossover network to the level of the rest of the system. Thus, much less power is applied to the tweeter (the attenuator network used should, of course, be able to handle the rest of the power). Such a tweeter of midrange element would have to be horn loaded for sufficient increase in efficiency. This approach is quite valid, but does produce systems with treble directionality problems since the radiating area of a horn is that of its mouth. This is really only fully satisfactory when used in systems specifically designed to make use of these directionality effects (a very complex subject in its own right). It also explains why power-handling failures are less likely to occur in fully horn-loaded systems; they don't handle more power, they just play significantly louder. Hence, the amplifier level does not get turned up as much and the loudspeakers do not get as much power fed to them.

Magnetic Fluids. Recently, another method of improving the ability to handle power before failure temperatures are reached has become available to the loudspeaker designer. This method significantly improves the transfer of heat from the voice coil and across the voice coil air gaps by replacing the air in the gaps with a special oil that has an excellent thermal conductivity.

The oil is the base of a remarkable new material called *magnetic fluid*, a molecular suspension of ferrite particles in an oil carrier. This fluid is attracted by magnetic fields and thus is firmly held in the air gap of a loudspeaker, as shown in Fig. 4. Now, for the first time, a new generation of high-power-handling tweeters and midrange units of the small size necessary if a design with good dispersion characteristics is desired can be built. Such units are now incorporated in an increasing number of American manufacturers' models and will, it is believed, soon be seen in systems from both Europe and the Far East.

Using a combination of high-temperature adhesives and magnetic fluid, loudspeakers having much better power handling capabilities than before can be designed. Why, then, is it still possible to damage loudspeakers thermally? This is where we look at the last part of the story, the signal applied to the speaker.

Loudspeaker Signals. In most systems the woofers are large and have large voice coils and lots of metalwork; tweeters have small coils and a much smaller mass of metal; and midrange units lie somewhere in between. Obviously, a tweeter cannot handle power as well as can a woofer. It is fortunate for the designer, therefore, that loudspeakers are designed to play music and not constant-amplitude sine-wave signals. This means that he can take advantage of music spectra, such as those given in Fig. 5, in the design of individual drive units for a system. It also means, unfortunately, that it is relatively easy to misuse the system, often without realizing it.

Misuse of a system, in a thermal damage context, means changing the spectrum of the signal applied so that too much power is applied to the most vulnerable units, the tweeters. The spectra shown in Fig. 5 are for a variety of rock-music records, which generally present the worst thermal problem for two reasons. Firstly, and most obviously, rock music is simply played louder. Secondly, its spectrum generally puts more power into the system at higher frequencies, reaching a broad maximum around 800 Hz. In contrast, classical music reaches a maximum power level somewhere around 500 Hz. Consequently, classi-

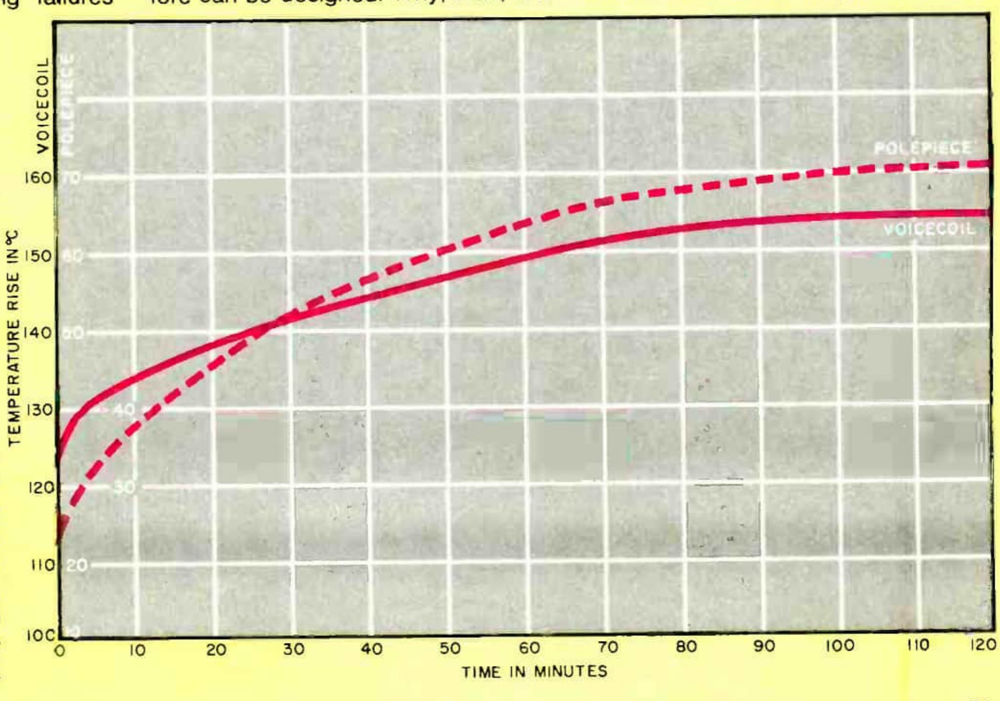


Fig. 3. Temperature rise of the voice coil and pole piece of a mid-range speaker with a 1-kHz, 10-watt (nominal) sine-wave signal applied.

"... many speakers have been damaged by amplifiers that are apparently lower-powered than the rating of the speakers."

cal music does not generally impose such a high power level into the tweeters and midrange units of a system.

In a typical piece of rock music on one system, it was found that, when 35 watts was applied to the system the tweeter received only 1 watt. If we assume that this system was rated at 100 watts on a music signal, then a "safe" test signal of 20 watts could easily destroy the tweeter since it could handle only about 1/35 of 100 watts. This is the first point—*Never* apply sine-wave test signals of more than one or two watts to a loudspeaker system, and never maintain such test signals on a tweeter for more than a few minutes!

Another way of changing the energy content of a music signal is to drive an amplifier into clipping, which immediately produces a disproportionately high average power by reducing the peak-to-average ratio. This is why it is frequently found that loudspeakers have been damaged by amplifiers that are apparently lower-powered than the rating of the loudspeaker. In this context, if high sound levels are frequently desired, and the amplifier power rating is within the limits of the loudspeakers being used, it is a good idea to place a mark on the volume control to represent the maximum "safe" setting. This position is easy to find if an oscilloscope is available (or can be borrowed) because it is the volume level just before clipping sets in on the loudest record in one's record or tape collection. If an oscilloscope is not available, the setting is more uncertain and should thus be a position of the volume control above which distortion audibly appears on loud piano music.

Other areas of such damage can be an amplifier's high-frequency instability above the audio range; careless interconnection of components such as tape-deck monitor circuits; and fast wind on certain tape recorders where the tape is near the playback head, producing an unwanted high-frequency signal. The latter problem is easily prevented by *always* remembering to turn the volume to zero when fast winding. If one wishes to listen while in fast wind to find some section of the tape, the treble control should be turned down to minimum and the volume kept as low as possible.

Finally, one last area of thermal damage to loudspeakers should be explored: continuous, high-level, discotheque music. This type of music should not be played for many continuous hours owing to the severe reduction in thermal overload capacity it causes. Only loudspeakers specifically designed for this type of application are relatively safe to use under these trying circumstances.

Fusing. Fusing is of very limited value in protecting systems against thermal overload. The best type of fuse is one having its own thermal link. To some extent, this allows for both instantaneous high-current overload and a longer-term lower-level current overload. One such series of fuses is the "Fusetron" FNM series, the best value of which should be found by consultation with the loudspeaker manufacturer, if possible.

The problems with fusing a system are multifold. The fuse cannot match the long-term thermal constants of the loudspeaker drive units and, what is more, it can be chosen only for maximum input levels. These occur in the lower midrange where loudspeakers are built to handle the power. However, the vulnerable tweeters remain largely unprotected, so all the other precautions discussed still have to be observed. It is possible to fuse the tweeter individually, of course, to provide much better protection. This, however, will change the frequency response of the system somewhat, as will now be shown.

Let us assume that a typical three-way system is rated as being safe on music program material with amplifiers rated at up to 100 watts rms per channel. This means that the tweeter itself is probably safe with about 3 or 4 watts rms applied. If the system is rated for 8 ohms, the tweeter probably falls to about 6 ohms; for a maximum input of 4 watts, this means a maximum safe current of 0.8 ampere. Such a fuse would have a resistance of about 0.5 ohm, attenuating the tweeter by about 1 dB. This attenuation is worthwhile if useful fusing protection is deemed necessary.

Finally, a word on a somewhat expensive type of thermal protection which could be (but is not yet, to my knowledge) incorporated at the design stage, especially in powered loudspeakers. The idea of any thermal protection is to prevent the voice-coil from rising above a certain temperature (which can be determined in the design stages). This is best achieved by actually detecting that temperature by continuously monitoring the dc resistance of the voice coil, since this resistance is directly related to temperature. Several less costly systems have been suggested that monitor directly the temperature of a series resistor. The simplest circuit uses a thermal circuit breaker bonded to the resistor. The problem here is that it is impossible to match the highly complex heat-transfer mechanism in the loudspeaker and thus impossible to match the temperature-rise characteristic of the voice coil.

Mechanical Overload. Generally speaking, mechanical overload can be in one of two forms: irreversible damage and cumulative damage. The second form is in the hands of the designer, while the first lies mainly in the hands of the user.

To elaborate, irreversible overload damage can occur when large overload signals cause the woofer cone-coil assembly to "bottom" and the coil and/or cone to buckle. This can be caused by a drastic overload at low frequencies, such as when an organ record is played at an excessively high level. The only real protection against this type of overload damage is plain common sense. Don't attempt to drive a given loudspeaker beyond its capabilities—a point that is usually easy to detect due to the rapid onset of a high level of audible distortion. It should not be forgotten here, too, that the various units in a system can be driven beyond their capabilities by other, usually brief, signals. For example, dropping a pick-up arm onto a record or causing a stylus to jump by

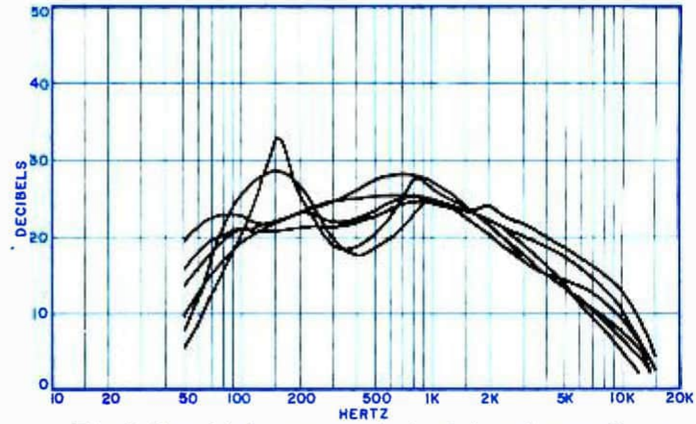


Fig. 5. One-third octave analysis of six rock recordings based on rms levels held for at least five seconds.

jolting the turntable can cause extreme movements in bass units and damage to the voice-coil, especially with very high-powered amplifiers. For similar reasons, if large switching transients occur when controls are changed, care should be taken to turn down the volume before these controls are operated.

The other type of mechanical overload damage is cumulative. It can be produced by work-hardening and eventual fracture of the wires going from the terminal strip to the voice coil on a loudspeaker drive unit. The only answer here lies in the design of the drive unit so that the wires used in this application can withstand continuous flexure. One such type of wire is called tinsel. It's made in multi-stranded form, with the tension being taken by a number of cotton or nylon cores. The only instance in which such precautions do not really apply is in the case of tweeters which operate above about 5000 Hz. Movements here are so small that such work-hardening does not generally occur.

We have covered various areas in which overload can occur and have implied that no single comprehensive protection method exists apart from common-sense precautions. To apply this common sense, however, we need to know what power our loudspeakers can actually handle.

Unfortunately, the foregoing is not simple. Examining loudspeaker specifications will show such power handling terms as: (1) 100 watts rms, (2) 100 watts program material, (3) 100 watts, (4) 100 watts continuous power, and (5) may be safely used with amplifier rated at up to 100 watts rms on normal speech and music program material.

Many other terms may also be found, but only number (5) above or some similar phrasing is of any real value to the consumer. Number (1) may be true at some frequencies but, apart from some specialized speakers, would result in frying the tweeter. Number (2) is not valid either, unless more information is given. For example, does it mean program material not clipping on an amplifier of 100 watts or does it mean an average program material level of 100 watts? The latter would allow full use of an amplifier of about 1000 watts since a typical modern recording has an average-to-maximum power ratio of about 10:1. Number (3) obviously requires more information, and (4) has the same shortcomings as number (1).

The only way out of all this confusion in specifications (if the system is going to be played at high levels) is to get some definitive statement about the maximum safe amplifier power for a loudspeaker system before purchase. Finally, if high sound-pressure levels *are* required, don't forget that a high-powered speaker does not necessarily play louder than a lower-powered system. Loudness is determined not only by how much power can be put into a speaker before damage ensues, but also by how efficient that speaker system is. For this reason it is possible to double power-handling capability by using two loudspeaker systems per channel. However, twice the power handling means only a 3-dB increase in sound level at maximum safe power, a very expensive way to achieve this 3 dB. Moreover, it should not be forgotten that two loudspeakers in parallel (especially if those systems have impedances of 4 ohms) may provide a dangerously low equivalent impedance for the amplifier being used, whereas putting those same speakers in series may adversely affect woofer damping and produce "boomy" bass. These problems are compounded by much more complex ones of placement, interference patterns, and so on, and become worse if *different* speakers are paired up to improve power handling. The obvious answer is to buy the correct system for the purpose in the first place.

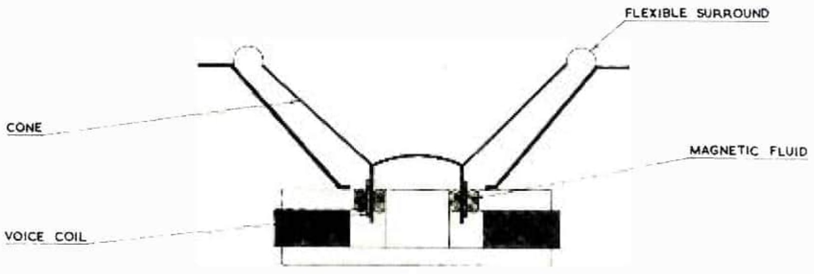


Fig. 4. Cross section of a loudspeaker drive with magnetic fluid to conduct heat across the voice-coil air gap.

Conclusion. If high sound pressure levels are desired from a home hi-fi system, one should adhere to precautions outlined in this article. In general terms, the average user of high-quality speaker systems will never experience overload damage unless some other part of his system fails and "takes the loudspeaker with it." ◇