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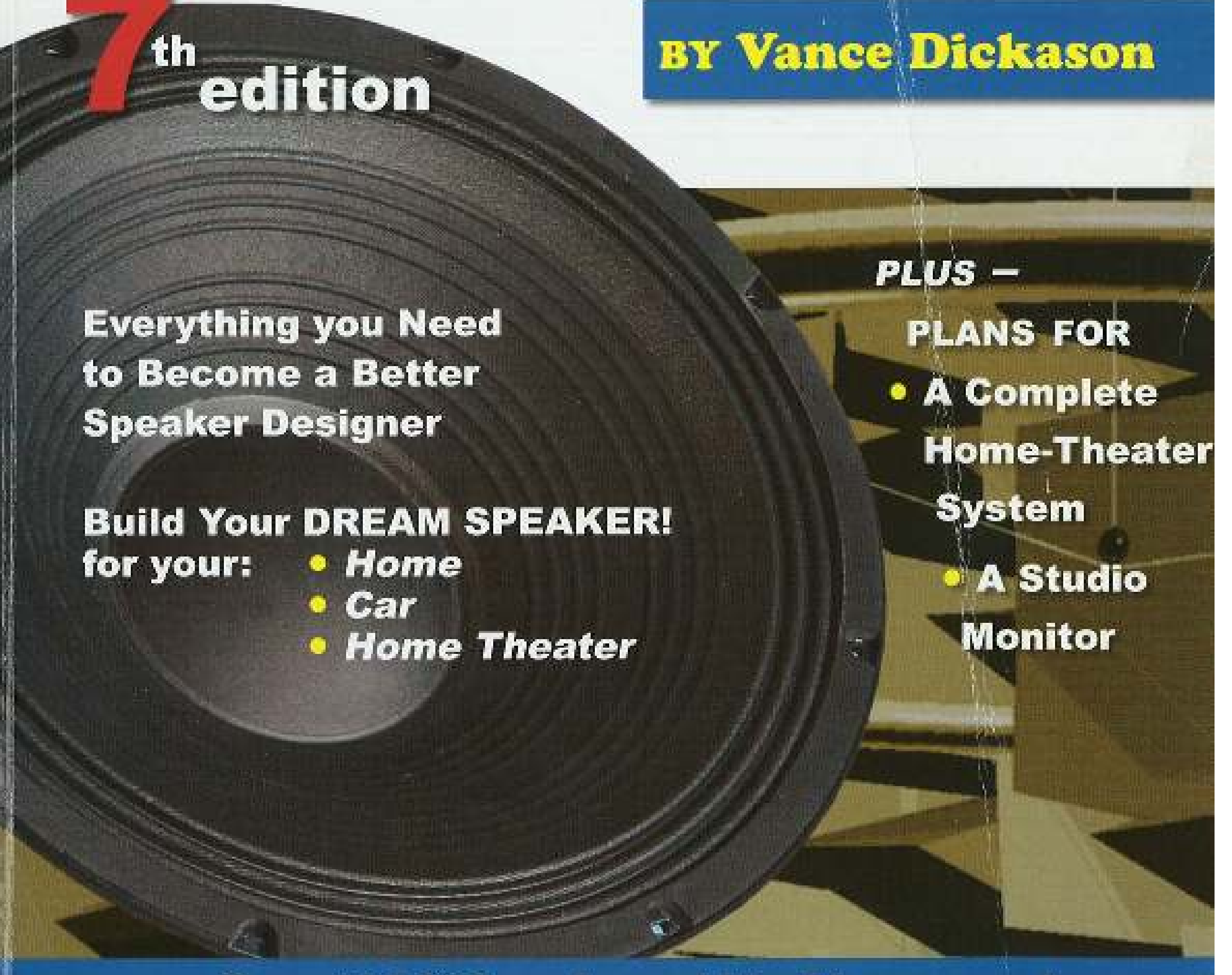
7th edition

BY Vance Dickason



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The Loudspeaker Design Cookbook

7th Edition

BY

Vance Dickason

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This work is affectionately dedicated to my family,

to my lovely mother and departed father,

to my sister Jeanne and brother Steve,

to my children Jason and Jennifer,

and to my grandchildren Jackson and Belle

and their father Arik,

with the remembrance that

the only two things that really count in life

are love and the pursuit of knowledge.

LDC 7 INTRODUCTION

This 7th Edition of the *Loudspeaker Design Cookbook*, first published in 1977, marks its 28th anniversary. Each new edition has brought to light some different aspect of state-of-the-art loudspeaker technology. Several of the editions contained material that was directly the result of new software that allowed the computer simulation of ideas that would be difficult to communicate in any other way. The 4th Edition depended heavily on the release of LinearX's LEAP 4.0 software, and the 5th Edition featured important transducer simulations using Red Rock Acoustics' SpeaD program. This 7th edition was made possible to a large extent by the use of the LEAP 5.0 and Dr. Wolfgang Klippel's Klippel distortion analyzer.

It's been five years since the previous edition was published. As has been the case with all the previous editions, the latest edition has added substantial new material. In terms of references and graphs, the 6th Edition added 83 new references and 214 new graphs. Following this tradition, the 7th edition includes some 42 new references and 341 new graphs, plus 815 graphs on the CD-ROM that is being made available. The CD-ROM has the 125 additional graphs from the Chapter 6 diffraction study (not available in the printed version), plus all the graphs published in print in Chapter 6 that will make the LEAP 5.0 polar plots much easier to read when enlarged on a computer screen and displayed in color.

While the *LDC* has been mostly a précis of available technology, a certain amount of the information has always been produced exclusively for the book, but never more so than in this 7th Edition. The explication on diffraction in Chapters 5 and 6 answers a number of questions concerning the significance of enclosure shape and where to locate drivers on a baffle. However, all this information was presented as computer simulations of single point microphone measurements, and while this gives you an excellent reference for the measurement consequences, it still leaves the nagging question of just how important diffraction and reflection issues are to the subjective sound quality of a speaker. Since I couldn't find any published information on this, I set up a subjective evaluation study in Chapter 6 for the various types of diffraction and reflection phenomena encountered in speaker design. While the results were not really unexpected, they were nevertheless very interesting and informative.

The second original piece of work being offered is a rather comprehensive study of woofer linearity. Working with Pat Turnmire, CEO of Red Rock Acoustics, we had eleven 10" woofers manufactured for the project, starting with a very conventional design, then changing one aspect of the design for each driver sample until the final woofer had most of the bells and whistles of any high performance transducer. The entire group of drivers was characterized using the Klippel DA-2 distortion analyzer to reveal the consequences of each iterative change, and the study also investigated the thermal characteristics of pole vent sizes.

The last new subject, voicing, is a frequently used term among loudspeaker manufacturers. This new section for Chapter 7 gives the reader a solid protocol for making final subjective adjustments to any new design.

As always, producing a new edition of the *Loudspeaker Design Cookbook* is a very rewarding and exciting experience for me, so I sincerely hope you enjoy reading and applying the information in this new edition as much as I have had writing and researching it.

Vance Dickason
August 2005

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Chapter 12 **TWO SYSTEM DESIGNS: HOME THEATER AND A STUDIO MONITOR**

THE
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Mr. Pablapa

HOW LOUDSPEAKERS WORK

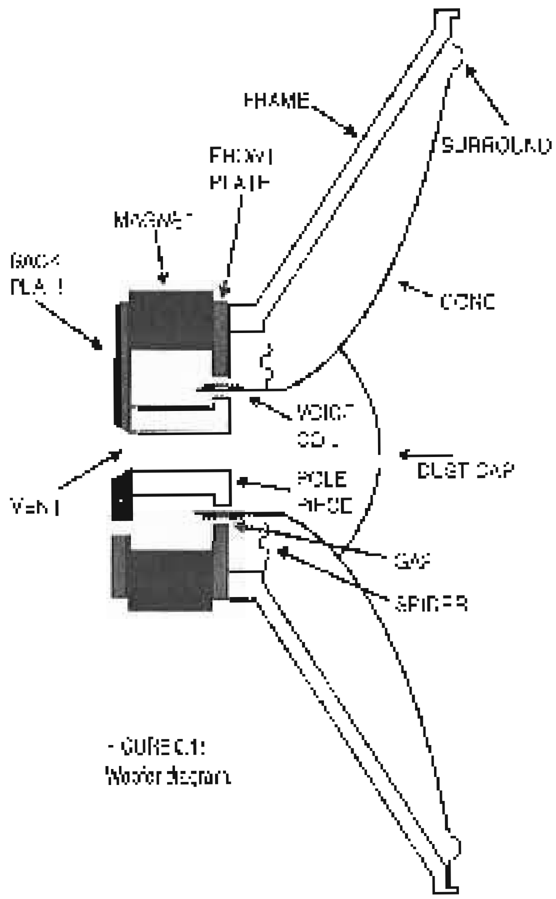


FIGURE 0.1: Woofer diagram.

0.10 ELECTRODYNAMIC SPEAKERS.

This Seventh edition of the *Loudspeaker Design Cookbook*, like its predecessors, aims to describe the operation, application, and measurement of electrodynamic loudspeakers and their associated enclosures and crossover networks. Electrodynamic drivers, the woofers, midranges, and tweeters found in the vast majority of loudspeakers are all based on the same basic concept: a diaphragm set in motion by the mechanical movement of a modulated electromagnetic field. As Mark Gander of JBL puts it, "To make sound, you must move air."

This mechanism is analogous to the electric motor, the rotating armature of a motor being replaced by the moving coil system of a speaker. *Figure 0.1* illustrates a cutaway view of a typical dynamic moving coil loudspeaker. As current is applied to the voice coil, an electromagnetic field is produced at right angles to the flow of current and to the permanent magnetic field. The resulting mechanical force causes the cone or dome diaphragm to move in a motion perpendicular to the gap field and consequently activate the air on either side of the diaphragm.

Three separate but interrelated systems operate in unison in a functional electrodynamic driver:

1. The Motor System: composed of the magnet, pole piece, frontplate/gap, and voice coil.

2. The Diaphragm: usually a cone and dust cap or a one-piece dome.

3. The Suspension System: made up of the spider and surround.

0.20 THE MOTOR SYSTEM.

The motor assembly is composed of five basic parts. These include the frontplate and pole piece, which together form the gap, the magnet, the voice coil, and the backplate. The backplate, frontplate, and pole piece are made from a highly permeable material, such as iron, which provides a path for the magnetic field of the magnet. The magnet is usually made of ceramic/ferrite material and shaped like a ring. The magnetic circuit is completed at the gap, causing an intense magnetic field to exist in the air space between the pole piece and the frontplate.

If an AC current is applied to the voice coil in the form of a sine wave at some given frequency, such as 60Hz, the flow of current in one direction on the positive half of the cycle will produce voice coil motion in one direction. When the current flow reverses on the negative half of the cycle, the polarity of the coil field reverses, and the motion of the voice coil changes direction as a consequence of the alternately attracting and repelling of the two fields.

In order to accurately reproduce the motion induced by the sine wave, the voice coil has to move equally in both directions through the gap. For this to happen, it is important for the magnetic field to be as symmetrical as possible so that motion in one direction will be applied with the same equal force as motion in the other direction. If this were not so, distortion of the signal would result.

If the flow of magnetic flux was confined only to the narrow space across the air gap, field symmetry would be assured and not be of concern. However, the magnetic lines of force "overflow" the gap area and produce stray fields on either side of the gap, known as fringe fields. Several methods are commonly used to ensure the symmetry of the fringe field, and are illustrated in *Fig. 0.2*. The straight pole piece in *Fig. 0.2a* illustrates an uneven fringe field caused by the non-symmetrical gap structure. Although adequate for many applications, this would be the least desirable method of construction. *Figure 0.2b* shows a symmetrical fringe field being created as the result of an undercut pole piece. *Figure 0.2c* depicts the effect of an angled pole piece on the fringe field, which, like the undercut type, results in a more symmetrical fringe field.

The mechanical force developed by the current flowing through the voice coil is represented by the term "Bl." B × l is the force produced by a given number of turns (feet) of wire, l, being subjected to a given flux density per square centimeter, B. Bl is a measurement of motor strength and is expressed in Tesla Meters/Newton. Directions on how to measure Bl are described in *Chapter 8, Loudspeaker Testing*.

0.21 GAP GEOMETRIES AND BL

Two different basic gap/coil geometries are used in loudspeakers, the underhung voice coil and the overhung voice coil. Of the two formats illustrated in *Fig. 0.3*, the overhung coil is by far the most common. The distance labeled X_{max} in the diagram represents the distance over which the coil can travel in one direction and maintain a constant number of turns in the gap. X_{max} can be calculated by taking the voice coil length, subtracting the air gap height, and dividing by 2.

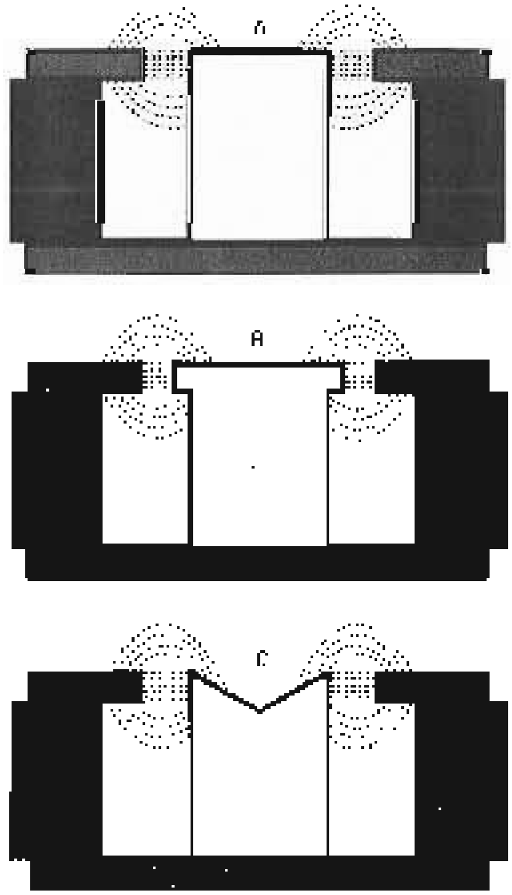


FIGURE 0.2: Fringe field effects for different pole geometries

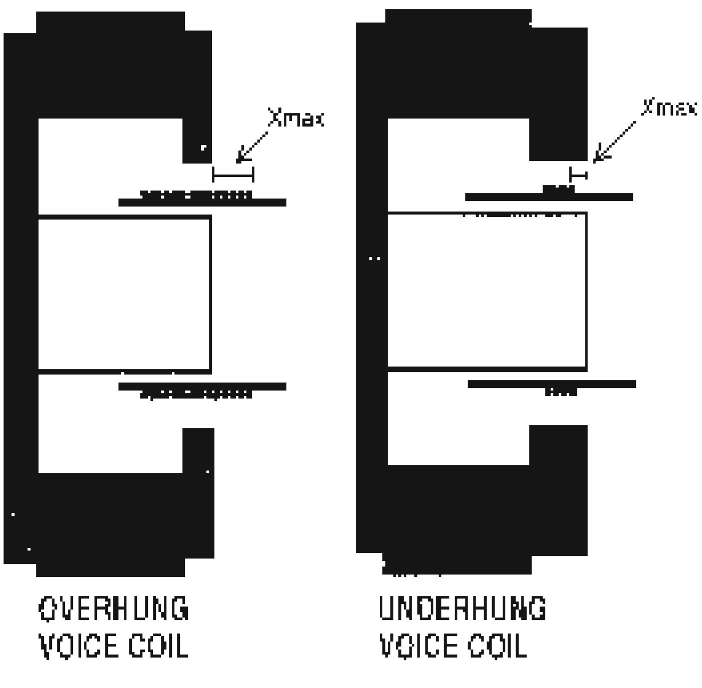


FIGURE 0.3: Overhung and underhung coil geometries

Figure 0.4 shows the graphic comparison of Bl change with increasing excursion between the two gap geometries (this diagram represents movement of the voice coil in one direction through the gap). As increasing voltage is applied to the speaker, the coil moves further and further out of the gap, the number of turns of wire in the gap decreases, and the total Bl motor strength decreases. A speaker is said to be operating in a linear fashion if the number of turns in the gap is constant, and in a nonlinear fashion if the number of turns in the gap is decreasing and changing.

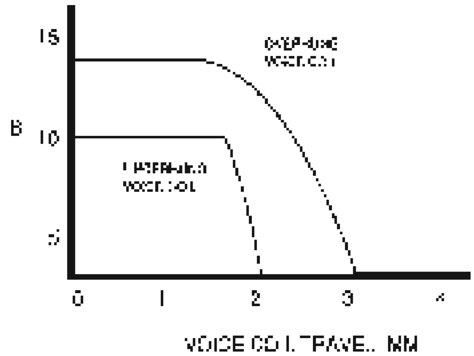


FIGURE 0.4: Comparison of Bl response for different coil geometries

The underhung coil gives extreme linearity over a short distance, but generally has lower Bl than the overhung coil (due to the increased gap height and requirement for a greater magnetic field) and because of the short coil, a lower voice coil mass. The overhung arrangement has the advantage of reasonably good linearity and better efficiency (even with greater coil mass), which accounts for its popularity among manufacturers.

Different combinations of gap height and voice coil length will give the same X_{max} number, but behave quite differently in terms of nonlinear (beyond X_{max}) behavior. For example, a 12mm voice coil length and an 8mm gap have the same X_{max}, 2mm, as an 8mm voice coil length and a 4mm gap height. Although the X_{max} of these geometries is identical, the ratio of gap height to X_{max} is quite different, 4:1 in the case of the 12mm voice coil and only 2:1 in the case of the 8mm voice coil. This ratio governs the rate at which Bl decreases as the coil rides out of the gap.

The curves in *Fig. 0.5* illustrate the variation in nonlinear behavior for geometries with the same X_{max} but with different ratios of gap height to X_{max}, as in the above example (from conversations with Chris Strahm, President of LinearX Systems and author of the Loudspeaker Enclosure Analysis Program, a/k/a LEAP). Looking at the diagram, Bl starts to gradually decrease beyond X_{max} to a point which is about two times X_{max} (double X_{max}) and then dramatically decreases. When the ratio of air gap height to X_{max} is large, the rate of Bl decrease is slower than when the ratio is low. At the furthermost limits of excursion, at the point where the coil is riding a large distance out of the gap, increased excursion does not change Bl significantly, and the

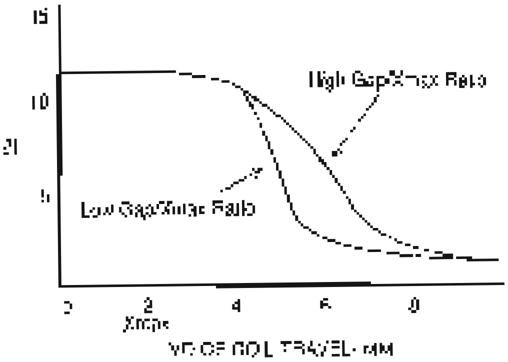


FIGURE 0.5: Comparison of Bl response with identical X_{max} and different coil/gap dimensions

curve becomes more shallow and levels out as it approaches zero.

Although Bl tends to decrease slowly up to about twice the X_{max} dimension, measurable distortion begins much sooner. In terms of distortion, the peak displacement limit of a moving coil can generally be taken as the X_{max} travel distance plus about 15%. Maximum excursion can be determined with a distortion analyzer setup to measure third-harmonic distortion as increasing voltage is applied to the voice coil. As excursion increases and the limits of X_{max} are exceeded, the third-harmonic distortion product increases. The X_{max} + 15% point will tend to coincide with an increase in third-harmonic distortion to a level of about 3%.[1]

0.22 SHORTED TURNS AND FARADAY LOOPS.

The current induced motion of the voice coil also causes an additional current flow in the opposite direction of the drive current, which is known as back EMF (electromotive force). This EMF current is induced in the voice coil as it acts like the armature of a generator. This effect, along with the AC field generated by the program drive current in the voice coil, causes modulation of the magnetic gap field. This phenomenon, identified by W. J. Cunningham in 1949,[2] results in significant second-order harmonic distortion. Further investigation of this effect has shown that the modulation of the field is different depending on which direction the coil is traveling through the field. It is a nonsymmetrical effect.[3]

This nonsymmetrical phenomenon occurs in part because the pole piece, acting like a transformer core, is coincident with the coil throughout its rearward travel, and only partially coincident on the forward travel of the coil as it excurs beyond the limits of X_{max}. It has also been suggested that the voice coil flux interacts with and modifies the shape of the fringe field. That also could, at least in part, explain the benefit gained from push/pull configurations discussed in *Chapters 1 and 2*.

The most obvious solution is to use a sufficiently high level of permeability in the iron next to the voice coil so the metal is always saturated, which results in negligible modulation of the magnetic circuit. This technique is not often used because high-permeability metals are relatively expensive. The most common technique devised to counteract this field modulation/eddy current problem is known as a shorted turn or Faraday loop. Shown in *Fig. 0.6*, the application of the shorted turn has several variations, but all accomplish the same task by generating a field equal and opposite to fields induced by the voice coil. *Figure 0.6a* takes the form of a conductive coating, such as copper, over the pole tip; *Fig. 0.6b* shows a copper cap over the pole piece; *Fig. 0.6c* has a copper cylinder surrounding the pole piece,[2] and *Fig. 0.6d* illustrates the positioning of a shorted ring (flux stabilizer ring), sometimes made of aluminum, around the base of the pole piece.

The shielded pole piece method has the added benefit of causing a decrease in effective voice coil inductance, which results in a rise in high-frequency response. The location and amount of shielding can be juggled to control midband and upper range driver response. The shorted ring at the base of the pole piece acts to reduce second-harmonic distortion, like the shielded pole methods, but does not alter the voice coil inductance and upper range response nearly as much. Although decreasing distortion is one of the benefits of the shorted turn method, controlling the midband upper range frequency response behavior is more often the consideration.

Figure 0.7 illustrates the upper frequency response changes from using a T-pole and copper-shorting ring. The comparison is of the same driver, a Bravox 5.5" poly cone woofer, with and without the T-pole/shorting ring combination. Note that the response of the T-pole/shorting ring version begins rising

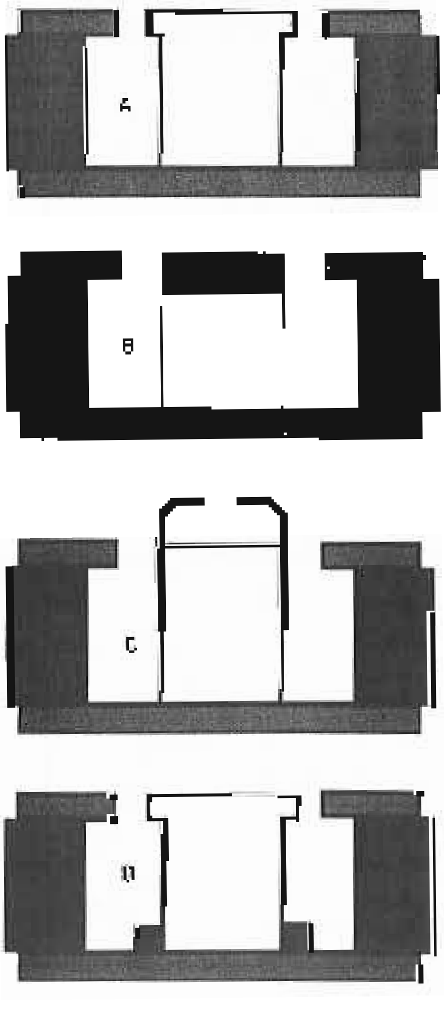


FIGURE 0.6: Comparison of different shorted turn configurations

above 500Hz and has as much as 3-4dB greater SPL than the version without the pole enhancements due to the reduction in losses from reduced eddy currents. *Figure 0.8* depicts the same comparison at 30° off-axis, showing the effect to be not only across the bandwidth above 500Hz, but through the entire radiation angle of the driver, as would be expected from an effect that lowered the voice coil inductance.

0.23 VOICE COILS—FORMER MATERIALS AND WINDING CONFIGURATIONS

Voice coils can be wound on a variety of materials, each having an effect not only on the T/S parameter set for a given driver, but also on the upper frequency response. There are two basic types of former materials used for loudspeakers, conductive and non-conductive. Conductive formers are by far the most common and are made from thin sheets of aluminum or Duraluminum (Duraluminum has a higher strength to avoid such voice coil problems as neck deformation during long excursions). Since aluminum is an electrically conductive material, it develops eddy currents in the same fashion as the parts of the motor system (plates and pole). These parasitic "currents" cause losses that are reflected in terms of heat and distortion.

Aluminum formers are not continuous cylinders but have a small slit along the length of the former such that it does not act as a shorting element. (Incidentally, removing the slit and using a continuous aluminum loop does not have the same effect as a shorting ring, although it does lower Q_{ts} by about 10%.) A non-shorting conductive former performance differs in two important aspects when compared to non-conductive former materials such as fiberglass or Kapton™ (a proprietary high temperature plastic material introduced by Dupont). The

FIGURE 0.7: On-axis comparison of two 5.5" woofers with and without copper shorting ring and shaped T-pole (solid = normal motor, dash = motor with shorting ring and T-pole).

FIGURE 0.8: Same as Fig. 0.7 but 30° off-axis.

FIGURE 0.9: On-axis comparison of two 5.5" woofers, one with an aluminum voice coil former and one with a Kapton voice coil former (solid = woofer with aluminum voice coil former; dash = woofer with Kapton voice coil former).

FIGURE 0.10: Same as Fig. 0.9, but 30° off-axis.

FIGURE 0.11: On-axis comparison of two 5.5" woofers, one with a two-layer voice coil and one with a four-layer voice coil (solid = 2L voice coil; dash = 4L voice coil).

FIGURE 0.12: Same as Fig. 0.11, but 30° off-axis.

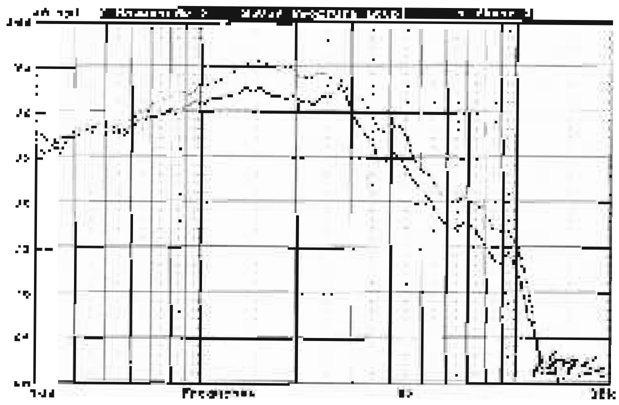
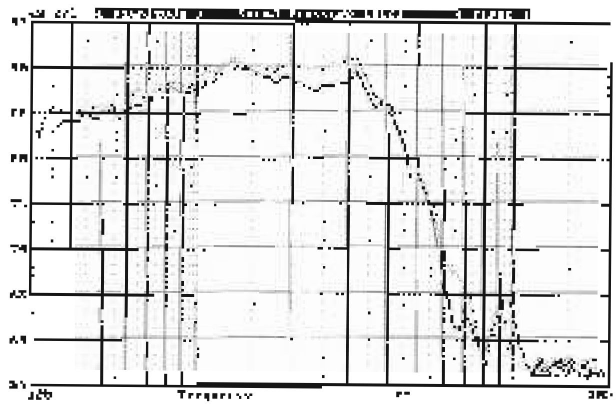


Figure 0.7



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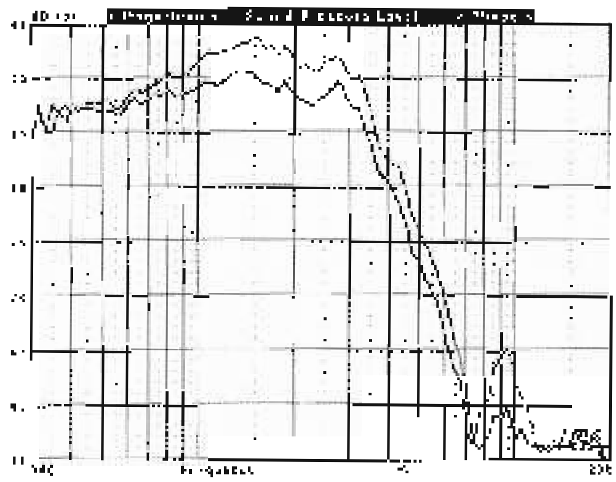


Figure 0.9

Figure 0.11

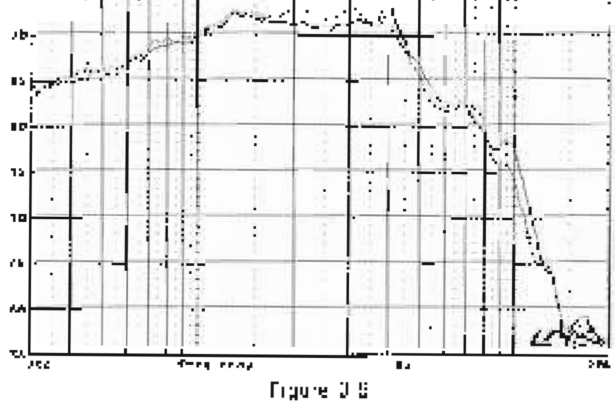


Figure 0.8

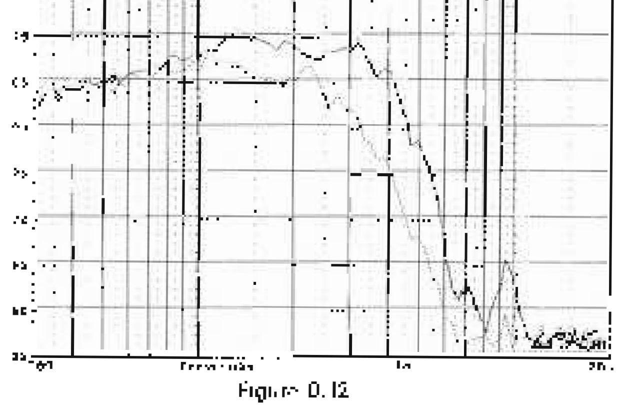


Figure 0.12

primary difference is that Q_{ms} (mechanical "Q") numbers using conductive formers are generally lower in the 2–4 range compared to non-conductive types. For non-conductive formers, higher Q_{ms} numbers between 4 and 12 are typical (the eddy current losses cause the Q_{ms} to be lower in conductive formers). Since non-conductive formers do not exhibit the induced eddy current problem, they also exhibit somewhat lower distortion.

The other performance difference between conductive and non-conductive former materials occurs in the upper frequency response. *Figure 6.9* depicts the comparison of two nearly identical Bravox 5.5" woofers with identical cones, suspensions and motor structures, except that one driver has a Duraluminum voice coil former and the other has a Kapton former. As can be seen, the woofer with the Kapton former has 1–2dB greater output above 1.5kHz. *Figure 6.10* shows the same comparison, but at 30° off-axis showing the effect to be somewhat more prominent. Again, this is due mostly to the difference in eddy current losses of the two materials. It should also be noted that part of this effect is due to mass differences (i.e., Kapton is a lighter material than Duraluminum).

The other notable response variation caused by voice coils is the manner in which they are wound. Obviously, larger voice coils with longer winding lengths have more turns of wire and hence greater inductance that will rolloff a driver's upper frequency response in the same fashion as a series inductor in a crossover. While there are different voice coil inductances for every possible combination of turns of wire and diameters of the former and pole piece, the biggest generalized difference in driver inductance is dependent on the number of layers of wire wound on to the former. The most common layer formats in woofers are two-layer and four-layer. Four-layer formers are frequently used on subwoofers to achieve the required Bl for the target response.

However, this can also be looked at in terms of upper frequency response control. *Figure 6.11* shows two nearly identical drivers, both Bravox 5.5" woofers, with the same motor, cone, and suspension, but one woofer with a two-layer voice coil, and the other using a four-layer coil (see *Fig. 6.12* for the same comparison at 30° off-axis). As can be seen, the woofer with the four-layer coil has a much lower frequency low-pass rolloff (−3dB at 2.5kHz) than the two-layer version (−3dB at 4.5kHz). Some manufacturers have developed two-way loudspeakers by taking advantage of this effect and controlling the inductance of a four-layer voice coil to produce a natural low-pass rolloff that will work with a particular tweeter. By doing this, the woofer does not require a separate low-pass crossover filter section and can be operated "wide open" with a crossover comprised of just the tweeter high-pass filter. I did a series of two-way prototypes for MB Quart several years ago using a 5.5" Bravox woofer and a 19mm MB Titanium tweeter. One prototype used a woofer with four-layer coil and a 3kHz mechanical rolloff and the other a standard two-layer coil with a higher rolloff. Both crossovers were computer optimized, but the four-layer woofer prototype had no low-pass filter on the woofer and a third-order high pass topography on the tweeter while the two-layer woofer prototype had a second-order low pass on the woofer and a third-order high-pass filter on the tweeter. In a subjective comparison between these two prototypes, once the levels were adjusted, both models sounded quite good and were very comparable in overall sound quality with the advantage that the four-layer model was cheaper to manufacture and had fewer parts in the crossover. The other difference was that the four-layer model was lower in overall efficiency by 2–3dB due to the extra weight in the four-layer coil, and therein lies the tradeoff.

6.30 THE DIAPHRAGM.

Explaining the physics of speaker cones generally begins with the theoretical discussion of the radiation of an infinitely rigid piston pushing against the air. The transference of motion from the piston to the air would be bounded, in terms of frequency, at the low end of the spectrum by its resonance frequency (below which its ability to transfer energy is limited by mechanical constraints), and the upper frequency limit by the nature of the radiation impedance of the air. Air has resistance to motion, radiation impedance, which decreases with frequency to a point where any additional increase in frequency will be met with the same amount of resistance.

This upper frequency point below which energy transfer will exhibit a steady decrease is a function of both the nature of the radiation impedance of air and the radius of the radiating surface. Smaller radiating surfaces can reproduce higher frequencies than larger radiating surfaces, a fact of nature which accounts for the advent of specialized speakers which cover different frequency ranges.

Real-world cones are not infinitely rigid and will flex to some degree depending upon the characteristics of the material from which they are constructed. Cone flexing has a critical effect upon the high frequency efficiency, SPL response, and polar response of a driver. While different materials have different degrees of stiffness and transmit vibration at different speeds internally, they all tend to produce the same types of flexing, usually referred to as "modes."

6.31 CONE RESONANCE MODES.

Two mode classifications, radial and concentric, are used in analyzing speaker cone vibration, depicted in *Fig. 6.13* (after Beranek with changes). Radial modes extend from the cone center to the edge, occurring mostly at low frequencies and considered secondary in nature. Concentric modes form a collection of waves or ripples that spread outward from the center of the cone. These concentric modes, made visible using holographic techniques, look similar to what

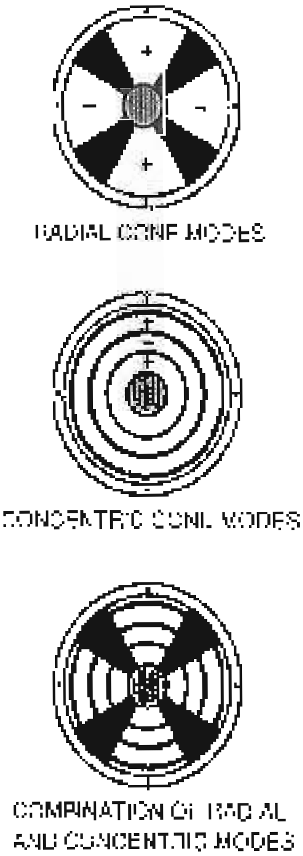


FIGURE 6.13: Cone vibration modes

you see when a pebble is dropped into the center of a bowl of water.

The number of waves varies with frequency, and as frequency changes, some of the ripples are reflected back to the center, forming interference patterns. These waves and ripples push against the air in a complex fashion, and some are in phase with the voice coil signal, while some are out of phase. The "+" and "-" areas in *Fig. 0.6* represent areas of the cone with opposite phase. This complex relationship of addition and cancellation, referred to as cone breakup, creates the many peaks and valleys in the typical loudspeaker SPL curve.

As frequency increases, the effective radiating area of the cone decreases so that very high frequencies tend to radiate only from the center area of the cone. At some frequency the effective radiating mass of the cone becomes small and a steep decrease in output begins which is described as the high frequency rolloff. To achieve a high cutoff frequency, the ratio of the voice coil mass and the cone mass must be as small as possible.[7] Upper frequency rolloff is also controlled by voice coil inductance.

0.32 CONE DIRECTIVITY.

As frequency increases all speakers become more directional and the high frequencies begin to "beam" like the light from an automobile headlight. At frequencies where the wavelength of sound (wavelength being equal to the speed of sound divided by the frequency = c/f, i.e., 1kHz has a wavelength of 1.13 feet) is large compared to the circumference of the cone (about 3 times the diameter), the radiation is spherical. As the frequency increases to the point where wavelength is equal to the circumference of the driver or smaller, the radiation pattern becomes progressively narrower. The chart in *Fig. 0.14* gives the -6dB off-axis points for different diameter speaker diaphragms (after Daniels with changes, [PL Pro Soundwaves, Fall 1985]).

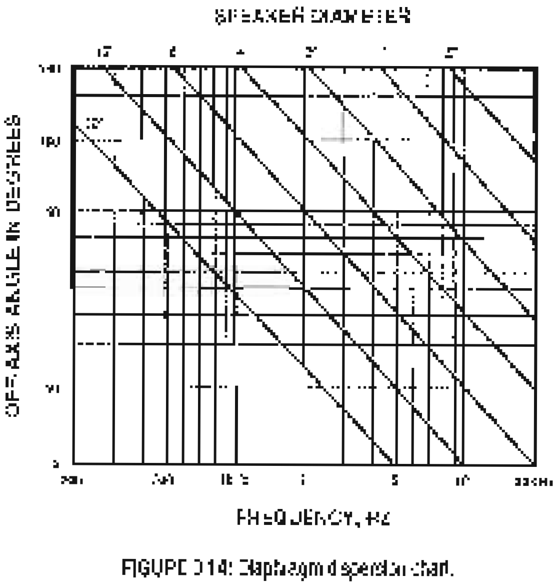


FIGURE 0.14: Diaphragm dispersion chart.

0.33 CONE SHAPE.

Differently shaped cones have different response characteristics. There are two basic shapes used in cone design, conical or flat, and convex. Conical shaped cones tend to have a high peak at the extreme high end of the response range, the location of the peak being in part determined by the angle of the cone. Compared to the convex shape the bandwidth is somewhat wider. Convex cones tend to have a smoother frequency response and only a moderate peak in the upper response (less high frequency efficiency), but with a somewhat reduced bandwidth compared to flat cones.[8]

The frequency response of the convex cones can be altered and controlled by changing cone curvature.

0.34 DUST CAPS.

Gap widths in loudspeakers can vary from several tenths of an inch for large diameter speakers to the thickness of a piece of heavy gauge paper for small cone tweeters. The width is as narrow as practical to maximize flux density while allowing for variations in voice coil alignment and swelling due to heating. When the voice coil is attached to the cone, the area between the pole piece and the voice coil is usually shimmed to accurately align the assembly. This procedure leaves the gap between the coil and pole piece exposed to foreign particles. This being so, it would be possible for small particles to become lodged between the two areas and create obvious problems. The traditional solution is to affix a seal, known as a dust cap, over this area.

Putting a dust cap over the junction of the cone and voice coil solves one problem and draws attention to several others. Two basic types of dust caps are used with speaker cones, solid and porous. A solid dust cap does not allow air to pass through its surface, and creates a small acoustic chamber that will generate air pressure changes as the cone moves back and forth over the pole piece. This compression and rarefaction can have detrimental effects on speaker operation.

Since the area between the voice coil and pole piece is too small to effectively relieve this pressure caused by the motion of the dust cap, manufacturers use two practical solutions to the problem. One is to vent the pole piece, which requires a small hole to be drilled through the pole piece so that air can pass out an opening in the backplate. The other is to punch vent holes into the voice coil form where it attaches to the cone. This will allow air to flow out of the small chambered area and relieve the pressure between the pole piece and the dust cap.

Porous dust caps also readily relieve the air pressure created above the pole piece, but create other problems. First they provide a leakage path from the inside of the enclosure. This is not terribly significant since the volume of air leakage through the gap is small, especially compared to that of a lossy enclosure. The other problem area happens as the cone moves inward on the pole piece, and air is forced through the dust cap toward the cone's radiating surface. This sudden squirt of air will be out of phase with cone radiation and can cause a frequency response problem.[9] It is probably not a good idea to seal porous dust caps that cause offensive response anomalies, since the original design may have specified the porous dust cap for cooling purposes. The air flow through the gap area can provide significant cooling for the heat generated by the voice coil. Sealing the dust cap can also cause compliance and Q changes which may or may not be desirable.

Dust caps also modify a driver's upper range frequency response. Since the cone tends to radiate near the center at high frequencies, the dust cap can play a critical part in shaping the upper end response of a driver depending on its material composition and shape. Solid caps tend to cause greater changes in frequency response than porous ones. Occasionally you see solid caps which have small round vents with screens to relieve air pressure, giving them the benefits (or detriments) of both methods.

Figure 0.15 depicts the frequency response comparison for the exact same Bravox 6.25" woofer (same motor, suspension, cone, voice coil, and so on) but with five different types of dust caps: porous cloth, doped cloth; soft PVC (Poly Vinyl Chloride); hard polypropylene and an inverted hard poly type dust cap (all other dust caps in this study were standard convex types). Since this graph is somewhat difficult to read with this much information, *Figs. 0.16–0.19* give a more meaningful comparison, with each of the graphs comparing the standard cloth porous dust cap to the four other types of dust caps.

Figure 0.16 compares the porous cloth dust cap with a doped cloth dust cap (the same cloth cap with a soft damping material painted onto the surface). Not quite intuitive, the doped cloth actually increased the output at the upper frequencies with somewhat more attenuation above 4kHz with an overall smoother response. The overall response of the undoped dust cap is also fairly smooth and even and has the added advantage of providing increased voice coil cooling by providing a passage for air moving past the voice coil.

Figure 0.17 compares the cloth dust cap to a soft PVC dust cap, which is a favorite among many manufacturers. As can be seen, the response of the PVC dust cap is smooth and even with no significant response anomalies, but a little less extension above 4kHz, probably due to mass and density of the material. Not only do manufacturers frequently choose this type of dust cap for its benign response characteristics, but also for its cosmetic appearance which gives a more coherent "look" in this era where so much emphasis is placed on industrial design esthetics.

Figure 0.18 compares the cloth cap to a hard poly dust cap. The hard plastic material in this case has a prominent resonance that is producing over 6dB more output centered on 3kHz, not a real convenient location for such an anomaly if you were trying to cross this woofer over at 3kHz to a tweeter. This is not necessarily typical of all hard poly dust caps as the nature of this anomaly is dependent on the diameter, shape, and density of the dust cap, but this is somewhat typical of what I have seen over the years and generally speaking, I never order a woofer sample deliberately with a hard plastic dust cap for use in a two-way application. For subwoofers or woofers intended to cross over to a smaller diameter driver at frequencies at least one or two octaves below the dust cap induced response anomaly, it really is not relevant and in these applications hard plastic dust caps are fine.

Figure 0.19 gives the last comparison between a

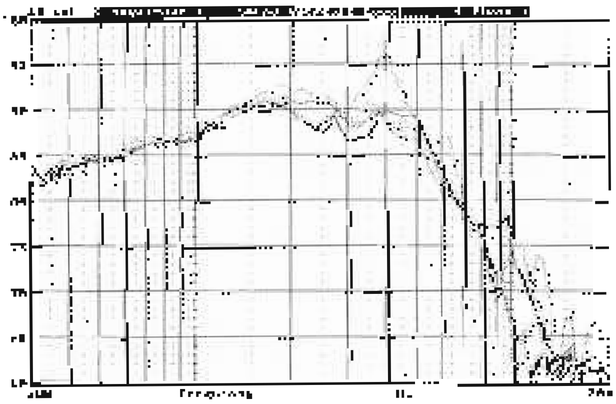


Figure 0.15

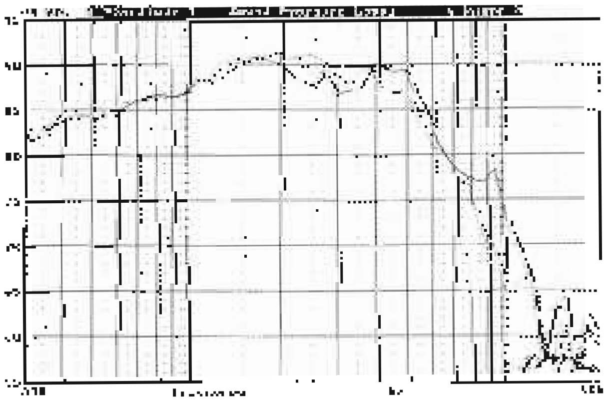


Figure 0.17

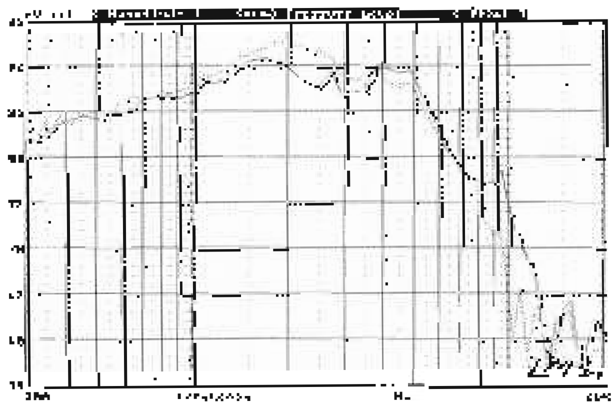


Figure 0.16

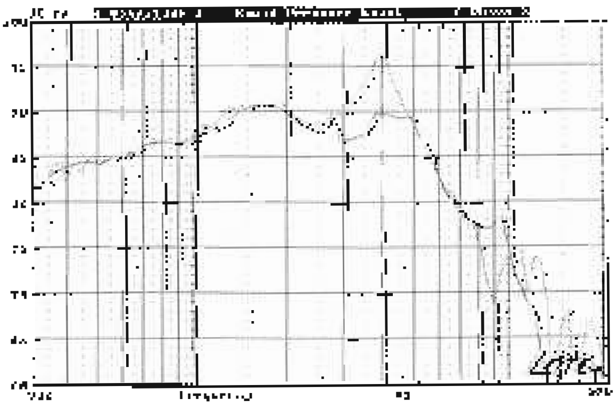


Figure 0.18

FIGURE 0.15: On-axis comparison of five identical 6.25" woofers, each with a different type dust cap (solid = cloth dust cap; dot = doped cloth dust cap; dash = soft PVC dust cap; dot/dot = hard poly dust cap; wide dash = inverted hard poly dust cap).

FIGURE 0.16: On-axis comparison of two 6.25" woofers, one with a cloth dust cap, and one with a doped cloth dust cap (solid = cloth dust cap; dot = doped cloth dust cap).

FIGURE 0.17: On-axis comparison of two 6.25" woofers, one with a cloth dust cap, and one with a soft PVC dust cap (solid = cloth dust cap; dot = soft PVC dust cap).

FIGURE 0.18: On-axis comparison of two 6.25" woofers, one with a cloth dust cap, and one with a hard convex poly dust cap (solid = cloth dust cap; dot = hard poly dust cap).

cloth dust cap and a hard plastic inverted dust cap. The inverted hard plastic dust cap has a number of advantages and has gained in popularity over the years. As can be seen in the response comparison, the inverted cap has a similar and smooth response as does the cloth dust cap. The other attribute that an inverted cap has is that, if they are made reasonably small so that they fit into the area near the neck joint of the cone (the junction of the voice coil former and cone), the cap can help strengthen the joint area and decrease the tendency of the cone neck to deform on hard excursions.

9.38 DOME SHAPES.

Dome tweeters and midranges have problems similar to those of those in cones. The two basic shapes are convex and concave. Concave dome radiators usually have much greater efficiency in the high frequency range, but a narrower directivity pattern. The higher efficiency is due in part to a wide peak caused by cavity resonances (although this can be damped to some extent) and the fact that convex domes are usually made of hard materials. Convex domes have a wider directivity pattern in the upper frequency range, and lack the efficiency of concave domes in that range.

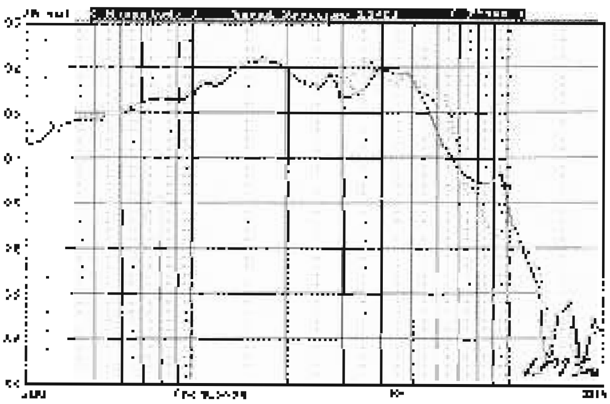


Figure 9.19

FIGURE 9.19: On-axis comparison of two 5.25" woofers, one with a cloth dust cap, and one with an inverted concave hard poly dust cap (solid = cloth dust cap; dot = inverted hard poly dust cap).

Figure 9.20

FIGURE 9.20: On-axis comparison of three 5.25" woofers, one with a rubber surround, one with a foam surround, and one with a Santoprene surround (solid = rubber surround; dot = foam surround; dash = Santoprene surround).

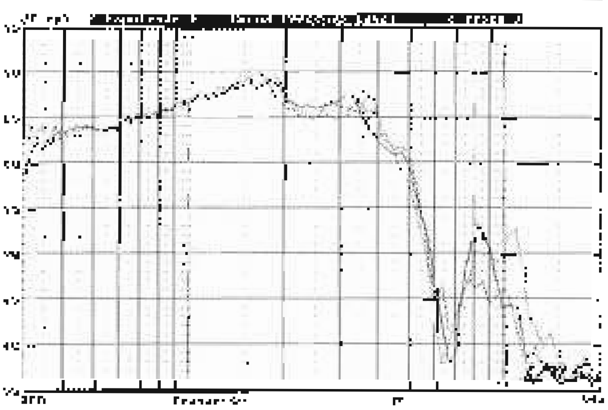


Figure 9.21

FIGURE 9.21: Same as Fig. 9.20 but 30° off-axis.

9.40 THE SUSPENSION SYSTEM.

The suspension system in any speaker is comprised of two elements: the surround and the spider. The surround, usually made of rubber, foam, or treated linen, performs several tasks. The surround helps keep the cone centered and provides a portion of the restoring force that keeps the coil in the gap. The surround also provides a damped termination for the cone edge. The spider, usually made of corrugated linen, likewise keeps the voice coil centered on the pole piece and also provides the restoring force that keeps the coil in the gap.

9.41 THE SURROUND.

The stiffness provided by the surround and spider is usually presented in terms of ease of motion, or compliance (compliance is the reciprocal of stiffness). In terms of the total compliance of the speaker, the spider provides about 80% and the surround perhaps 20% of the total compliance. The surround has two important functions. Its primary job is to keep the voice coil centered over the pole piece; however, damping the vibration modes at the outer edge of the cone is also critically important. The choice of thickness and type of material used in a surround can dramatically alter the response of the speaker. The ability of the surround to damp cone modes and prevent reflections back down the cone can alter both the amplitude and phase of modes combinations, making it an integral element of cone design and a viable response shaping tool.

Figure 9.20 illustrates the response comparison of different surrounds materials attached to the three samples of the same 5.25" Bravox woofer[4] (same cone, spider, voice coil, motor, and so on). The three different materials used were rubber (in this case, not pure butyl, but a commonly used rubber compound for surrounds called NBR), foam, and injected Santoprene (this is a sophisticated process that Bravox uses to insert mold the Santoprene surrounds by injecting them onto the cone edge). Rubber gives the smoothest response with the least anomalies and overall one of the best edge damping materials to use for a surround where smooth upper frequency response is a consideration. The only downside to using rubber as a surround material is that it has to be made using a vulcanizing process that is slow and more costly than the heat forming process used with foam-type surrounds. The foam surround response is not quite as smooth by comparison and, at around 10kHz, it has obvious problems damping the cone's upper frequency modes, which is of course not too relevant to a woofer that is being crossed over between 2–3kHz. Foam surrounds are relatively easy to fabricate and inexpensive, but suffer from degradation over time due to exposure to light and various air impurities found in large metropolitan areas (and, even in small towns sometimes). Overall, they do not perform as well as rubber when it comes to edge damping, but foam is still one of the most frequently employed materials for surrounds. Less common, but becoming more prevalent, is Santoprene. Santoprene looks like rubber, and can be either heat formed (which is done a lot for subwoofer surrounds) like foam surrounds, or injection molded. This material is as inexpensive as foam, but does generally not have really good edge damping characteristics at upper frequencies, as can be seen in the graph. *Figure 0.21* shows the same comparison of all three surround materials, but at 30° off-axis. From this it is obvious that a lot of the response problems are relatively on axis and that many of the response anomalies are less pronounced viewed off-axis.

0.42 THE SPIDER.

The spider has several functions. Its secondary tasks are to keep the voice coil centered over the pole piece and provide a barrier that keeps foreign particles away from the gap area. The primary purpose, however, is to provide the main restoring force (compliance) for the speaker. It is the stiffness of the spider which determines the speakers' resonance. Speaker resonance is a function of compliance and mass and can be related by:

$$f_s = [6.28 (C_s \times M_D)^{0.5}]^{-1}$$

Where f_s = the driver free air resonance frequency, C_s = the driver compliance, and M_D = the total mass of the driver (the weight of the entire cone assembly, voice coil, cone, spider and surround, plus free air mass load).

0.43 LINEAR AND PROGRESSIVE SUSPENSION SYSTEMS.

It seems intuitively evident that the best type of suspension would be one which would provide uniform restoring force throughout its range of travel. While this can be true for closed box type speakers where the compliance of the air within the box acts as a restoring force on the cone, the exact opposite is true of drivers in vented cabinets. This anomaly, dubbed the "oil can effect" by Don Keele,[11] results in dynamic offset of the voice coil. The offset problem, a nonlinear phenomenon, occurs as the driver is being driven towards its X_{MAX} limitation. As the coil moves to a position where more turns are out of the gap, Bl decreases, back EMF decreases, and the coil draws more current, pushing the coil even farther out of the gap and thus creating distortion.

A progressive suspension system can counteract this nonlinear offset problem. This type of spider and surround combination provides increasing stiffness at the same time the Bl is decreasing (*Fig. 0.5*). If the breakpoint for the increase in stiffness coincides closely with the breakpoint for Bl decrease, then the voice coil is prevented from accelerating out of the gap. Suspension systems of this sort are found frequently in professional sound woofers intended for high SPL applications. Unfortunately, many amateur audio designers seem to be unaware of this fact, since it is not uncommon to find a woofer with an extremely linear suspension system being used in a vented application.

0.50 MODELING LOUDSPEAKER IMPEDANCE.

All of the systems I have described can be mathematically modeled by an electrical circuit diagram which is analogous to their operation. This technique is at the heart of all the box calculation methods described in *Chapters 1–4*. The electrical analogy of a driver is represented by a circuit with an impedance which duplicates that of the actual driver. The details of the actual measured impedance of a typical loudspeaker are shown in *Fig. 0.22*. *Figure 0.23* gives the analogous electrical model of a speaker. The circuit's elements are as follows:

R_E = DC resistance of the speaker

R_{EVC} = frequency dependent resistive component of the voice coil reactive rise (real part of voice coil inductance)

L_{EVC} = frequency dependent inductive reactance component of the voice coil reactive rise (imaginary part of voice coil inductance)

M_D = mechanical parameters due to mass

C_S = mechanical parameters due to compliance

R_{MS} = mechanical parameters associated with damping

Z_R = rear radiation impedance of the driver

Z_F = front radiation impedance of the driver

This model is similar to the one described by Beranek,[4] with the exception that the voice coil reactance was taken to be a fixed value as opposed to being frequency dependent[13] as shown in this diagram.

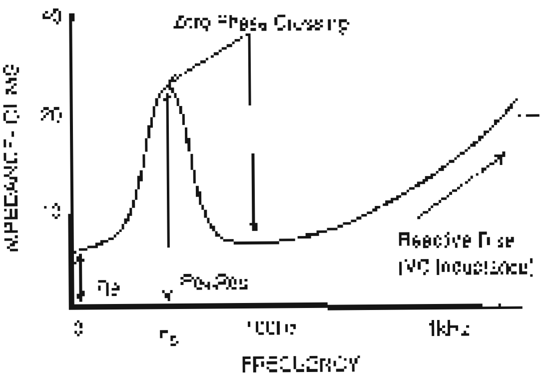
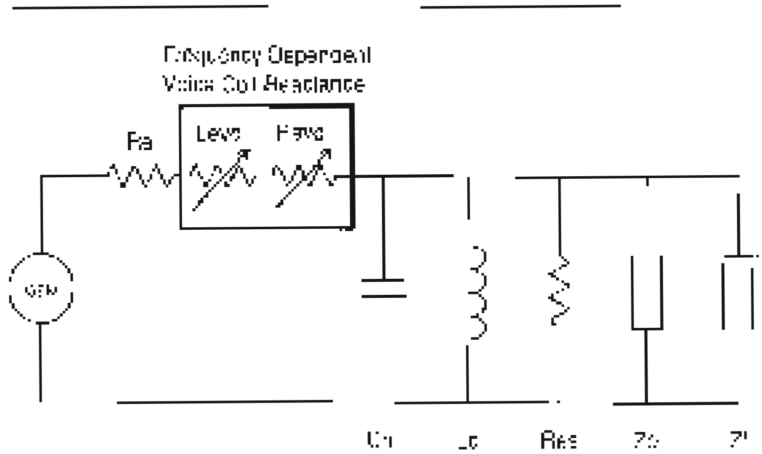


FIGURE 0.22: Loudspeaker impedance diagram.

FIGURE 0.23: Schematic model of a dynamic loudspeaker



0.60 POWER, EFFICIENCY, AND ROOM SIZE.

The loudness produced by a given amount of amplifier power is a direct function of the efficiency of the loudspeaker and the volume of air it is trying to excite. Deciding how much loudspeaker capability you need to achieve a target volume level in a given room is an important question you should consider prior to building your own speakers. Since most loudspeakers are rather inefficient devices, usually on the order of 0.5 to 2%, coming up with the appropriate amount of acoustic power may not be calculated simply. If we consider a typical infinite baffle driver with efficiency of 0.5% (calculation of loudspeaker efficiency is discussed in *Chapters 1* and *2*), and an amplifier capable of delivering 50W RMS, then the acoustic power available from this system would be 0.25 acoustic watts (0.005 × 50W = 0.25W).

The graph in *Fig. 0.24* can be used to establish the approximate program material SPL produced for a specified amount of acoustic output in a given room volume. If we take our 0.25 acoustic watts and put it into a typical 20′ × 20′ × 8′ living room, which is approximately 100 cubic meters of space, it would achieve an SPL of about 97dB. To produce an additional 3dB for 100dB SPL, we would have to double the amplifier power to 100W.

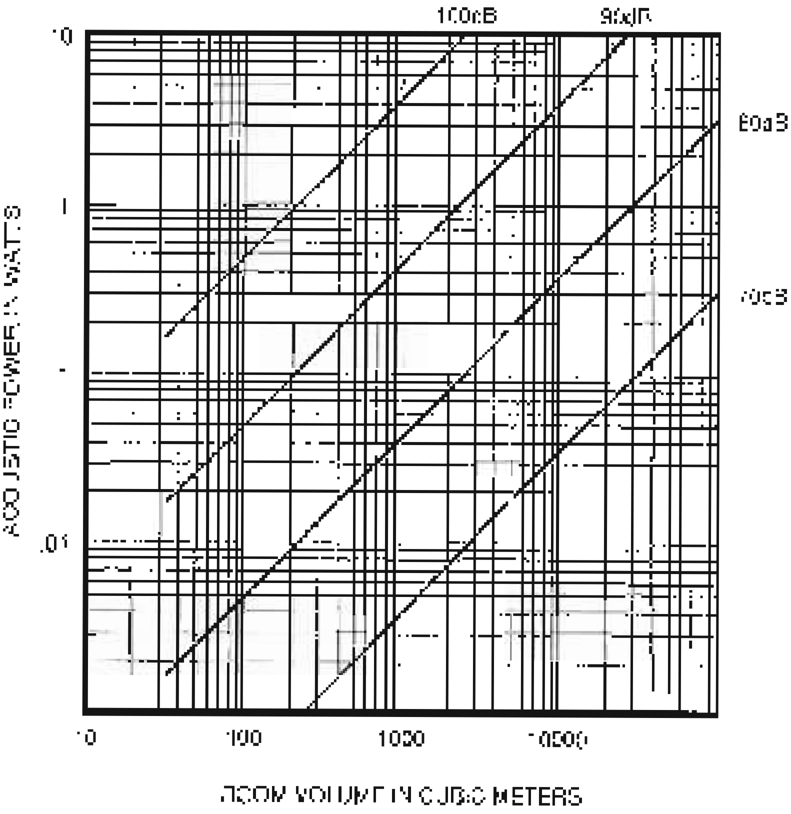


FIGURE 0.24: Acoustic power vs. room size chart.

Depending on the RMS rating of the speaker, and providing the rating is in some way adequately related to the thermal capabilities of the driver (many are not), obtaining a 0.5% efficiency speaker may not be sufficient. Rather than double the amplifier power, another solution would be to use an additional driver. The addition of a parallel connected second driver doubles the cone area and increases acoustic power by a factor of four. The one acoustic watt of the combined drivers could produce nearly 103dB in the same room driven by the original 50W of output power. To reach the same SPL level with a single driver would require 200W.

0.70 ADVANCED TRANSDUCER DESIGN TOPICS.

Most all of the discussion about designing speaker enclosures in the *LDC* (see *Chapters 1–4*) and elsewhere in the loudspeaker industry literature is from the perspective of predicting box performance from a specific set of driver parameters. This, however, is the perspective only of a system designer. The other perspective not frequently considered in print is that of the transducer engineer, whose job is to come up with the combination of parts, cones, voice coils, magnets, top and bottom plates, dust caps, surrounds, and spiders to produce a woofer that will perform in a specific enclosure type.

For the most part, transducer engineers are a rare breed and good ones are hard to find. There is no specific curriculum at the university level that teaches this skill, so becoming a professional in this field is either self-taught or passed from one practitioner to another, usually within the corporate confines of either the larger speaker companies or OEM driver manufacturers.

System design was at one time considered a "black art" until the advent of the professional CAD software packages (such as the LinearX LEAP software, which I rely upon for most of the simulations in this book) for doing this type of work. Until recently, transducer engineering still fell into the "black art" category, because the only way to develop a new woofer was a lot of trial-and-error and years of experience. This process has been radically changed by the introduction of new transducer modeling software released about the time I was putting together the 6th edition of the *LDC*. A design sequence that once would require a lot of experience, often at least three to four sample iterations, and perhaps months of R&D time to produce can now be done much faster in a computer simulation.

The first of these new CAD programs was authored by Red Rock Acoustics and is titled SpeaD (Speaker Designer) (see *Chapter 9* for details about SpeaD and another example of this type of software, WinMotor). *SpeaD* consists of a set of software tools that makes transducer design much faster and easier. Essentially, it lets you play "what if" with your designs and build a woofer in your computer.

The SpeaD tools include two separate programs:

- Reverse *SpeaD*—software that can synthesize the required T/S parameters required to achieve a set of performance targets such as f_3, box size, enclosure type, and tuning.
- SpeaD—generates speaker parts specifications that will produce a driver whose performance matches the desired target T/S parameters.

The exciting part of this type of transducer simulation software is that you can make changes in seconds and easily explore possibilities that were once simply too time consuming or difficult. With the sample content of Loudspeaker Design Cookbook

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