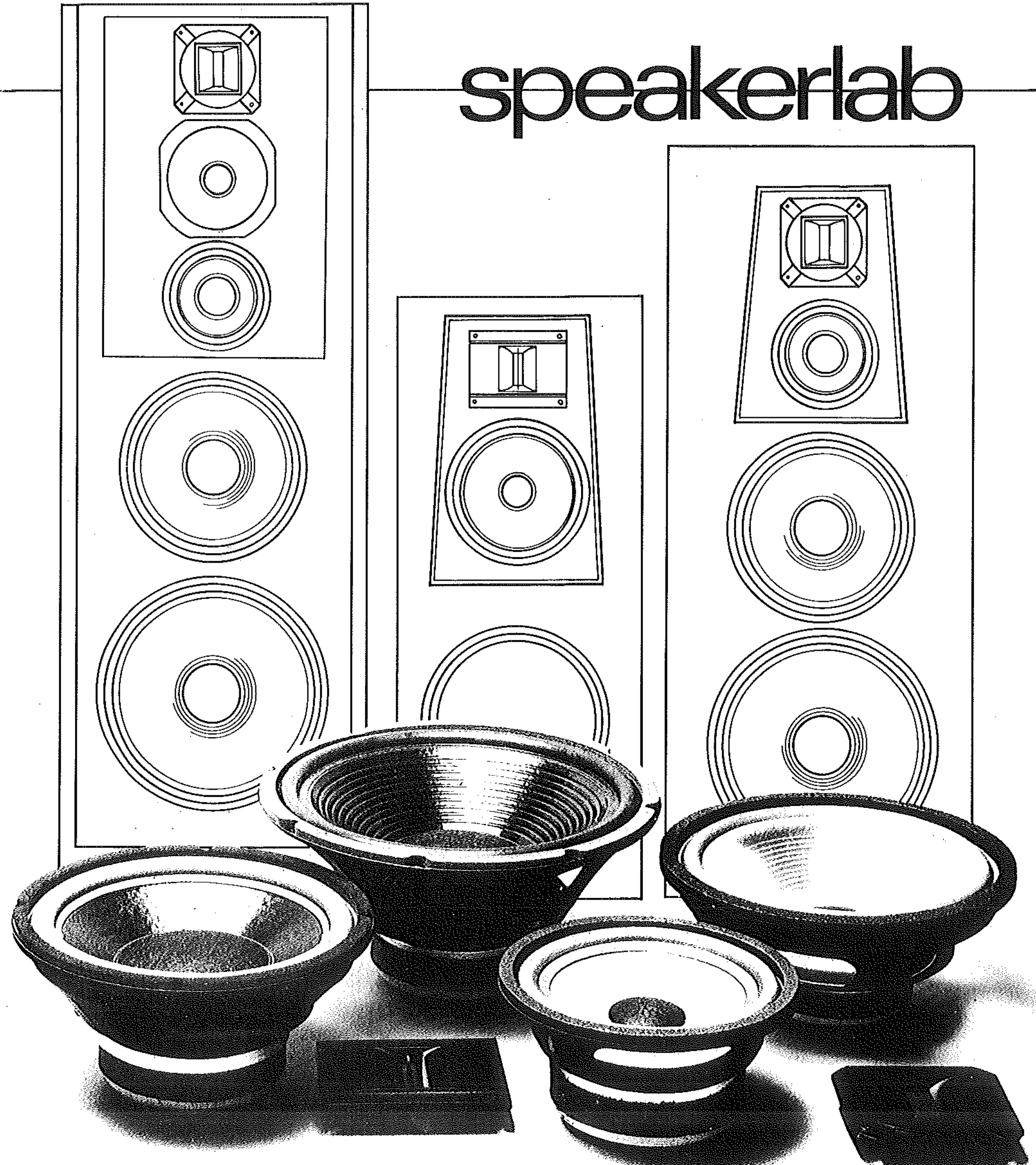
The Loudspeaker Design Guide.

speakerlab



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The Publication for designing your own speaker system:

Designing a speaker system used to be a trial and error affair involving hundreds of tests on different combinations of drivers and crossover parts. This is fine for commercial designers who have the time and resources to do it, but tough on the at-home builder who is building a one-off speaker. He builds it, tinkers a bit, and that's how it comes out.

In the last few years acoustic scientists like Thiele, Small, Ashley, Newman, and Keele have put a lot more engineering into the art of speaker design. It is now possible to design a system, buy parts to specification, build it and be reasonably confident that it will work as designed and that you have hit at (or near) optimum for a given cost, size and complexity.

Designing your own system also sets you free from limitations that trade and shipping requirements set on commercial products. For example, enclosures over about six cubic feet are impractical to sell through stores, but hold no terrors for the at home speaker builder. You can make a ten cubic foot system and assemble it in place, or even install it in your home as a built-in.

In order to support those of you who would like to embark on the journey of designing and building your own systems, we have complied "The Loudspeaker Design Guide" which contains information based not only on theory but practice, as well.

All in all, this publication is an extremely useful collection of nearly every practical aspect of loudspeaker building.

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Chapter One: Fundamentals of Loudspeaker Design.

This chapter contains the basic rules we know that will help you build a set of speakers that get the most out of whatever components you are using. Speakerlab has been manufacturing speaker kits and selling them direct to consumers for many years, so we have a pretty good idea of what works and what doesn't. Good sound results from hitting straight on three factors:

1) Start with good quality components. You can't make a silk purse out of a sow's ear (though the reverse is possible).
2) Base your design on sound principles.
3) Take the time to assemble your system carefully and pay close attention to details. Magic plays no role whatsoever in speaker design. Nor do the sound waves care whether you have a PhD in nuclear physics.

Designing The System

Designing a speaker system breaks down into seven basic steps:

1) Choose the system type-sealed enclosure, vented, etc.
2) Choose your woofer and enclosure volume.
3) Decide on a two-way (woofer and tweeter) or three-way (woofer, midrange, tweeter) system. We recommend using a three-way design whenever possible.
4) Choose your woofer, midrange (if you use one) and tweeter.
5) Select the appropriate crossover for the drivers you will use.
6) Choose box dimensions that will give you the right volume, allow your drivers to fit on the front panel, and fit in your available space.
7) Pick the box material and construction methods.

System type

How the woofer section of your system works depends on how you design the enclosure. Most home systems use either a simple sealed box or a vented one (same thing with a hole). (For more exotic designs such as transmission lines and bass horns, see chapter 2 & 3 respectively).

A sealed-box loudspeaker system is the simplest to build as well as the simplest to analyze theoretically.

Sealed-box speaker systems were first made popular by Edgar Vilchur of the Acoustic Research company, who dubbed them "acoustic suspension" systems. The name is derived from the use of a woofer with high compliance in a totally sealed airtight box. The cone suspension is so loose that it has little control over the cone motion, and instead the air trapped inside the box serves as an "air spring" to keep the cone centered and prevent excessive, destructive cone excursion in response to very low frequency signals.

Many early sealed box systems achieved low bass response by using unusually heavy cones, which also tended to reduce their efficiency--it took extra amplifier power to move that heavy cone around. However, this is not a necessary attribute of the sealed box design--The low bass in most modern designs of this type is achieved by an even higher compliance woofer--and many have quite respectable efficiency figures.

The vented box design is perhaps the oldest, originally patented in 1932 by a Bell Labs scientist, A.L. Thuras. The vent can take the form of a duct, a tunnel, or just a simple hole. The vented enclosure acts as a Helmholtz resonator; that is, the mass of air in the vent resonates against the "springiness" of the air inside at some certain frequency, the Helmholtz frequency. In a properly tuned system the radiation from the vent around the resonant frequency adds to the radiation from the cone, theoretically extending the response of the system to a lower frequency than would be otherwise attainable from the cone radiating alone.

Sealed and vented loudspeaker systems have been studied analytically by two scientists, Neville Thiele and Richard Small (References 1-3). The vented system is the one they have given the most help and attention. One aspect of their theory that has been widely misunderstood has to do with the cone excursion and power-handling capability of a vented system. They showed that vented systems have less cone excursion than sealed systems, and handle more power for signals in the passband. However, for signals below the cutoff frequency of the system--and wide-response stereo program sources do emit considerable subsonic energy from record warps and such--a vented enclosure provides very little cone loading. Air from the rear of the cone escapes from the vent and there is no "air spring" effect to control the cone motion. The woofer is easily overloaded and exposed to excessive cone excursion, particularly if a high compliance woofer is used. Thus a simple vented system will have poor power handling ability unless a stiff-cone woofer is used.

Careful study of Thiele-Small theory shows that for a given woofer, you can build a system of either type--either a small box that is totally sealed, or a larger box that is vented. The larger box will have lower frequency response but (unless equalized) will handle less power.

The key phrase here is "for a given woofer." The well-known dictum that venting will give you a system that is more efficient, or has better bass, or uses a smaller box, is based on the assumption that you also use a different woofer. Box design alone has no effect on efficiency; the efficiency depends only on the woofer. The woofer's motor strength (magnet and voice coil), cone size and weight determine its efficiency.

We recommend sealed systems for first-time builders because: 1) they work best in the limited space most people have; 2) they are easy to design and relatively simple and fool-proof to build; 3) they work well with today's best high-compliance woofers.

Vented systems, on the other hand, are trickier to design and build. You can correct for the "cone unloading" at subsonic frequencies with an equalizer but that makes your system more complex.

For the rest of this chapter we will concentrate on sealed-box systems. The second part of this chapter, on building enclosures, applies to speakers of all kinds.

Sealed enclosure designs

A sealed system works most effectively with a high compliance woofer that has a long-throw, easy-moving cone to pump the large air volumes needed for good bass. As you go down in frequency, a speaker cone has to move farther and farther to put out the same amount of sound; it has to go four times as far at 50 Hz as it does at 100 Hz to put out the same amount of sound. Because of this, any restrictions on ease of cone motion translates into a limitation of low-end frequency response. The sealed system provides the best bass response and efficiency for a given enclosure volume when the woofer cone suspension is so compliant that nearly the only limitation on cone motion is the enclosure "air spring".

The strength of the air spring depends only on the volume of air in the box--the less air, the stronger the spring. So for sealed systems, the most important factor is the internal air volume.

The box volume affects both the resonant frequency (fc) and the Q of the system-(Qtc). In other words, changing volume changes both the POSITION and SHAPE of the low frequency response rolloff curve.

Figure 1 shows how this works. Small volume makes both the resonant frequency (fc) and -3dB frequency (f3) higher, and gives higher Q with a response peak. Large volume gives a low resonant frequency, but makes the response "sag" and gives weak bass and can in fact reduce the cutoff frequency.

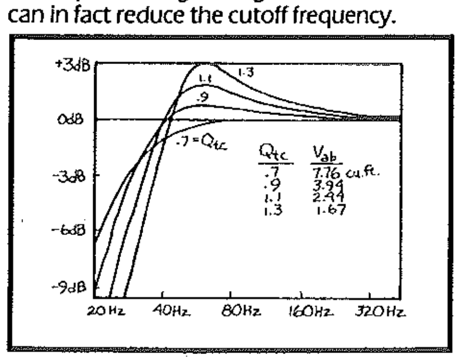

Figure 1. Theoretical frequency response of a hypothetical 12-inch woofer in boxes of various volumes. The hypothetical woofer has parameters of fs 16 Hz, Qts .40, and Vas 16.0 cu. ft. 3. it is somewhat similar to the Speakerlab W1208R.

Earlier we hinted that there wouldn't be any math in this chapter. That was just to lure you in. Fig. 2 takes a little math but is worth it. It will give you the frequency response versus box volume for any woofer. Just for the record, the equations Fig. 2 is based on are:

$$f_c = f_s \left(\frac{\frac{1}{V_{ab}} + \frac{1}{V_{as}}}{\frac{1}{V_{as}}} \right)^{1/2} \qquad (1)$$

$$Q_{tc} = \frac{Q_{ts}}{f_s} f_c \qquad (2)$$

V_{ab} = internal air volume of the enclosure.

V_{as} = volume-equivalent compliance of the woofer.

f_s = natural resonant frequency of the woofer.

Q_{ts} = total Q of the woofer at its natural resonance.

Equation 2 means that Qtc and fc vary together as you vary the box volume. This makes it possible to use the same set of curves for any woofer. Changing woofer parameters changes only the volume for each curve and the frequency scale.

So Fig. 2 is a "do-it-yourself-kit" for home designers. Xerox a few copies of Fig. 2, sit down with your pocket calculator, and have at it. All you need to know are the values of Qts, fs and Vas for the woofer you have in mind. You can regard Fig. 1 as a "worked example" of Fig. 2. To use Fig. 2, follow these steps:

1. Calculate fs/Qts. The answer will be in Hz, of course. Write it below 1.0fs/Qts on the graph.

2. Calculate .5fs/Qts, 2.0fs/Qts, and so on, and write these values on their lines.

3. Calculate Vab values for the different Qtc values from the formula:

$$V_{ab} = \frac{V_{as}}{\left(\frac{Q_{tc}}{Q_{ts}}\right)^2 - 1} \qquad (3)$$

Enter the Vab values in the spaces provided.

Note that physically you cannot have Qtc less than Qts; the system Q won't be less than the woofer Q. So for calculations where Qtc is less than Qts, this formula gives a meaningless answer--the box volume Vab comes out negative.

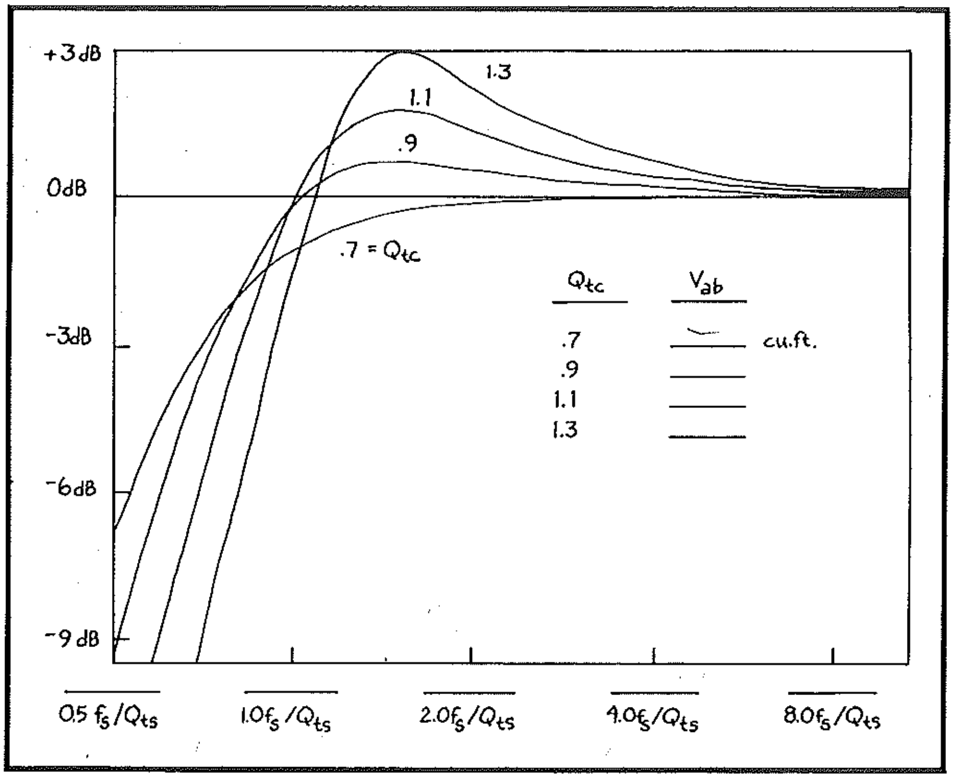

Figure 2. Frequency response graphs for any woofer in a sealed box speaker system. The frequency scale and box volumes depend on the woofer's parameters; fs, Qts, and Vas. See text for instructions marking the frequency scale and calculating Vab.

Also note that Figures 1 and 2 assume that the box is filled with air, not stuffed with damping material which would disturb the theory. A 1-inch thick liner of damping material is OK--its effect is small.

You can interpolate the curves in Figure 1 or 2. For example, for a 2.94 cubic foot box, draw a new curve between the 2.44 and 3.94 cubic foot curves. Draw it closer to the 2.44 curve, 1/3 of the way to the 3.94 curve.

$$\frac{2.94 - 2.44}{3.94 - 2.44} = \frac{.50}{1.50} = \frac{1}{3}$$

If you have a woofer for which data is not available, Table 1 gives typical box sizes for average high quality, high compliance woofers. Lower compliance woofers take bigger boxes.

If you use two or more woofers in a single enclosure, allow adequate volume for each. For example, 10-inch and 12-inch woofers sharing the same enclosure would need 1.5 plus 2.0 cubic feet; 3.5 cubic feet total.

TABLE I.
Box volumes for typical acoustic suspension woofers

Woofer diameter	Volume
8"	.5 - .9 cubic feet
10"	1.0 - 2.0 cubic feet
12"	1.5 - 2.5 cubic feet
15"	6 cubic feet or larger

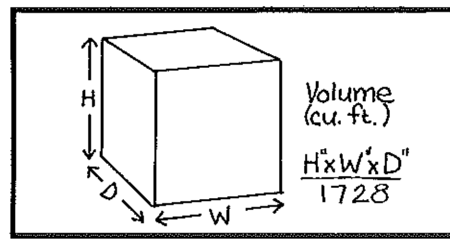

Figure 3. Calculate enclosure volume using the internal dimensions.

Choosing system Q

You might ask, what is the best curve shape- the best value for Qtc? We will let you in on one of the deep, dark secrets of the stereo industry. Speakers designed for a "classical music" audience are generally designed with Qtc equals 1.0 or so. Speakers targeted at "Rock" audiences generally have Qtc equals 1.4 to 2.0, which gives a bass peak of 3.5 to 6.3 dB. Though less accurate, they will sound a lot louder because of the big peak. The tweeter can be adjusted to match the efficiency of the top of the bass peak, resulting in "semi-flat" response (flat except for a big dip around 200Hz.). Many speakers advertized as "high efficiency" are designed this way. Most rooms have a resonant peak around 200Hz, so the 200Hz dip often goes unnoticed.

We recommend that you design for Qtc equal to .9 to 1.2. This gives the most natural, accurate, extended bass response. The .7 - 2.4 dB bass peak that results is fairly broad and its effect on reproduction is less serious than the "missing bass" at lower frequencies that results from lower Qtc values.

Theoretical cutoff frequency

The theoretical cutoff frequency is customarily defined as f3, the frequency at which the bass response is 3dB down from the midband response. You can easily read f3 for the different curves in Figure 2.

Be aware, however, that those bass response curves are theoretical. They correspond to a woofer radiating into open space. Reflections of the low frequencies in a real room of finite size augment the bass and lower the actual 3dB down frequency. For a crude estimate, you can assume f3 in a typical room will be 70% of the theoretical value.

As we mentioned before, for any given design-type (called "alignment" by Thiele and Small) there is a three-way tradeoff among bass response, box volume, and efficiency. For sealed systems, one certain value of Qtc gives the lowest f3 for a given box volume and efficiency. That value of Qtc is 1.1, which gives peaking of 1.9dB.

You can also look at this from the viewpoint of efficiency. The Qtc equals 1.1 sealed box alignment is 1.8dB more efficient (for the same box volume and cutoff frequency) than the simple Qtc equals .707 (Butterworth, or B2) alignment (Ref.2).

This answers one of the riddles of the speaker design art. The vented system has been said to be 4.5dB more efficient than a sealed system of comparable box volume and cutoff frequency (Ref.8). You would think that if this were true, vented systems would have long since made sealed types obsolete. A 4.5dB more efficient speaker takes only about one-third the power to drive--a tremendous advantage. Nevertheless, you can go hear vented and sealed speakers in stereo stores, and both types, of the same size and apparent bass response, play approximately equally as loud. We have certainly never been able to find any three-times-the-power advantage for vented systems.

The 4.5dB theoretical difference is based on a comparison of the fourth-order (unequalized) Butterworth (B4) vented-box alignment to the Butterworth sealed-box alignment. Since the Qtc equals 1.1. The sealed system is 1.8dB more efficient than the Butterworth, the vented system's advantage over the OPTIMUM sealed system is 1.8dB less, or only 4.5 - 1.8 equals 2.7dB.

That famous 4.5dB comparison also assumes that the vented system has zero box losses, that is, infinite Helmholtz resonator Q. Real boxes have a Q of around 7, which makes vented systems about 2.8dB less efficient that the ideal (Ref. 4, p. 308, 310). When you subtract off this 2.8dB, the vented system's advantage disappears altogether. You might hope that other fourth-order alignments would be more efficient; alas, for box Q equal to 7, the Butterworth is about the best.

It is not until you go to equalized (sixth-order) vented systems that you begin to actually achieve the higher efficiencies promised by vented speaker theory.

Measuring woofer resonant frequency

If you'd like to measure the resonant frequency (not the cutoff frequency) of your system, there is an easy way to do it. Connect the equipment as shown in Figure 4. The exact value of the resistor is not important and the voltmeter need not be calibrated, though the oscillator frequency dial should be accurate. As you sweep the audio oscillator through the 20-200 Hz range, the impedance of the speaker will vary and the voltage indicated by the meter will vary. The speaker's impedance is maximum and the voltmeter reading will peak at the resonant frequency. Find the audio oscillator frequency where the voltmeter reading peaks. That's all there is to it.

The 1000 ohm resistor allows a small current through the speaker. The speaker responds, and its cone moves fastest at resonance. (That's what "resonance" means.) Motion of the voice coil in the magnet gap generates a back voltage which is maximum at the resonant frequency. The voltmeter measures the voltage and indicates the maximum, revealing the resonance.

You can easily observe the changes in resonant frequency caused by putting weight on the cone, holding the edge with your finger, or putting the speaker face down on a table, trapping air between the cone and table surface.

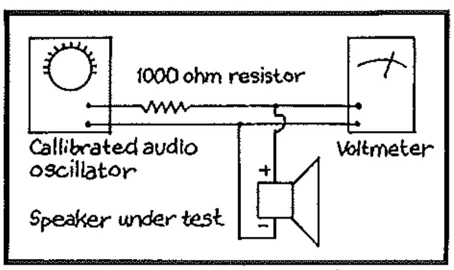

Figure 4. Equipment setup for measuring woofer resonant frequency (in free air or in an enclosure) by the voice coil impedance method.

The same method is used to measure the natural resonant frequency of the woofer itself. This measurement is best made with the woofer on a flat baffle and after it has been played for a few hours (in a sealed box, of course) to "break it in". This most closely matches conditions under which it will be used. Testing the resonance in free air (no baffle) or before break-in will give you a higher measured resonant frequency.

Sealed systems have a single resonant frequency. Vented systems have two, a high one and a low one. At the high resonance, woofer cone and vent air are moving in the same direction; at the low resonance, woofer cone and vent air are going in opposite directions. So the lower resonance is where maximum cancellation between vent and cone radiation is occuring. The same is true for the very low resonance in transmission line speakers.

In a vented system, the box resonant frequency fb (Helmholtz frequency) is the frequency at which the impedance is MINIMUM between the two resonant peaks.

Midranges and Tweeters

We recommend that you build a three-way system if you can afford it. A three-way system (separate midrange speaker) will give a cleaner sound than a two-way system. In a two-way system, the woofer reproduces midrange frequencies and the woofer's cone motion (due to the bass signal) introduces distortion. Using a separate midrange speaker avoids this.

Most important for midranges and tweeters is to buy good units and mount them properly. The design of the enclosure has much less effect on their performance than it does on the woofer's.

The Speakerlab catalog offers only a few midranges and tweeters; we list only the few best among all we have tested, the pick of the world speaker market. If we can make a better unit than any we can buy, we make it. Otherwise, we buy someone elses.

A few comments on the different types of construction of high frequency drivers are in order.

Among direct radiators, dome tweeters have better frequency response, better transient response, and better dispersion than cone tweeters. Cone tweeters tend to "rattle" from waves running back and forth across the cone surface. Domes are more rigid and less susceptible to this effect.

Leaf tweeters are better than both cone and dome types. Their low mass and totally driven membrane combine to offer superb transient response and very extended high frequency response.

Midranges and tweeters should have sealed backs to prevent woofer pressure in the box from pushing the cones around. If one doesn't have a sealed back, make a little box to cover it.

Crossover network

Whatever speaker design you choose, use a real crossover.

The crossover network is an electrical filter that sorts out the different frequencies and sends the lows to the woofer, highs to the tweeter, and midrange frequencies to the midrange. Good crossover networks are made up of coils and capacitors; coils pass lows and block highs, while capacitors pass highs and block lows.

"Cheapo" crossovers used by some villainous speaker makers contain just capacitors to block lows out of the tweeter and midrange so they won't burn out; the full spectrum goes to the woofer. (Chapter 4 covers crossover design).

Here we are just going to cover how you use a crossover, as though you already have one. The crossover has input terminals (for connection to your amplifier) and color-coded wires for connection to the drivers. See Figure 5. Most crossovers mount over a hole in the back of the enclosure with the input terminals and controls accessible from outside through the hole (Fig.9). (This is regarded as very important unless you plan to live inside the speaker enclosure).

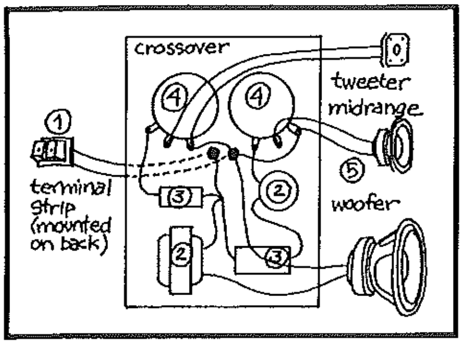

Figure 5. Wiring scheme for a three-way system. The electrical signal from your amplifier enters at the crossover's input terminals (1). Then it goes to the crossover network proper, which is a network of coils (2) and capacitors (3). Here the different frequencies are channeled to the different speakers (5). Lows go to the woofer, mid-frequencies to the midrange, highs to the tweeter. The high and midrange signals pass through L-pads (4), which control the tweeter and midrange volume levels for balancing with the woofer. Parts (1), (2), (3), and (4) are all parts of the crossover.

Many crossovers have level controls for the midrange and tweeter. Variations in room acoustics and frequency response characteristics of other parts of the system (particularly the cartridge) make level controls a practical necessity in many cases. Most home designers will not be using driver types matched for sensitivity. Most tweeters and midranges are more sensitive than woofers and have to be "turned down" for proper frequency balance. In commercial speaker systems this is done by resistors or lossy coils built into the crossover, but in home-built speakers the best approach is to use L-pads on the midrange and tweeter. Unlike switches, L-pads are capable of smooth adjustment over a wide range, and can compensate for driver sensitivity variations, room acoustics, etc.

Enclosure design

Box shape is not too critical so long as the volume is correct. It is the quantity of air inside that does the job, not the shape of the box. But stay away from long skinny column designs- you can get a "pipe organ" effect with waves running the length of the column. Also avoid very shallow boxes, which can give you trouble with reflections off the back wall. A good rule of thumb is to make your maximum dimension no more than three times your minimum dimension. Most bookshelf-style speaker enclosures have height/width/depth proportions equal to about 5/3/2. If you stick to these proportions you won't win any prizes for orginality but your box will look like a speaker enclosure.

When calculating the internal air volume of your enclosure it is usually unnecessary to take into account the space taken up by the crossover, speaker backs, and damping material. The first two actually use only a small percentage of the space, and the damping material is really 99% air and only 1% solid.

Tall enclosure designs are becoming more popular as this allows mounting the drivers in a vertical line, which gives a better defined stereo image—one that has more width and depth, and gives the impression that instruments are more precisely localized in space.

When locating the drivers on the front panel, put the midrange and tweeter near the top so they'll be closer to ear level. Place the woofer near the bottom so reflections near the floor will reinforce the bass. You can make a pedestal to lift the enclosure off the floor later if the bass is too strong.

Building the enclosure

These are the best construction methods we've been able to find to make acoustically correct, good-looking enclosures that are easy to construct and not too expensive.

Construction

Make your box a one-piece unit that is solid and airtight (except for the vent, if your design calls for one). Unless you plan to be using your speaker enclosure to hold up part of the building you live in, strength is not a problem. Airtightness and freedom from rattles are the main things to concentrate on.

The old-fashioned way of making speaker enclosures with the back removable and the front a permanent part of the box is not good for use with modern woofers. To put out solid bass, they pump gallons of air in and out, resulting in high pressure inside the enclosure that will cause wheezes, whistles and rattles around a removable back's edge seal. Instead we use a totally glued one-piece box with drivers that mount from the front of the front panel rather than from the inside.

The enclosure can be made in three major subassemblies; enclosure, front board, and grille cloth frame. Figure 6 shows a typical design for an enclosure and front board. Note the use of cleats for the front and rear panels to rest against. Rabbets (Figure 7) in the front and rear edge of the box rather than cleats are also acceptable, if you have the equipment to cut the rabbets. Cleats or rabbets are not absolutely necessary but make the box more rigid and make it easier to get airtight.

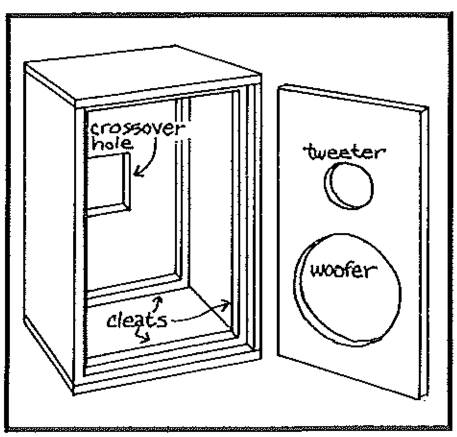

Figure 6. You can build a simple enclosure for a two-way system using a 10-inch woofer like this. All joints go together with 6-penny finishing nails and white glue. First cut all the pieces from 3/4" material; all dimensions are in inches. Join the cleats to the top and bottom. Put in the four cleats for the sides. Install the back.

If you must use a screwed-down removable back on your enclosure, use felt, foam rubber, or some similar compressible, flexible material for a gasket.

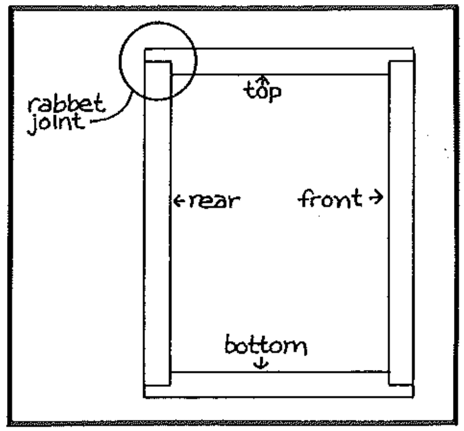

Figure 7. Front and rear edges of the enclosure rabbeted to receive the front and back panels.

We glue in the front and rear panels before mounting the drivers. If we ever have to get into the system again, we remove the woofer and work through its cut-out.

We normally mount our front panels flush with the front edge of the box. We make the grille frame with outside dimensions the same as the dimensions of the box, so it covers the entire front right up to the edge of the box. We use 1/2" thick particle board for the grille frame to allow the driver frames to project beyond the front surface of the box (which they do, since they are surface-mounted) without hitting the grille cloth.

Some builders prefer to have the front panel recessed behind the front edge of the box a half inch or so, and have the grille frame fit in the recess. The edges of the box are not covered by the grille. This is a little more trouble but works fine. You then have to cover the box edges with woodtape or wood veneer, of course, to get a finished appearance.

Figure 6 also shows the hole cut in the back for the crossover network. Our standard crossover and crossover kits fit over a 5-1/2 inch by 4-1/2 inch hole.

Material

Make your enclosure out of plywood or particle board at least 3/4-inch thick. We use 3/4-inch veneered or vinyl-covered particle board for the sides, top and bottom of our enclosures. "Vinyl covered particle board" doesn't sound very exciting, but with modern woodgrain-printed, embossed vinyl it makes an attractive piece of furniture at a reasonable cost. It is indistinguishable from real wood except on close examination. We use 3/4-inch particle board for the front and back of the enclosure.

Don't use solid wood or shop grade plywood. Solid wood warps with humidity changes and will open up cracks that will buzz or allow air leaks. Shopgrade plywood is a reject grade and has loose plies that will rattle. If you use plywood, use either hardwood veneer plywood or AB-interior-grade fir plywood. These types are usually well made with seven plies and only small voids inside. Voids, which can be seen at the edge, should be no more than 1/4 inch in diameter.

Particle board is good acoustically, though rather unattractive in appearance.

We use it for fronts and backs because it won't be seen. Good grades of particle board are identifiable by a fairly smooth edge without crumbly material or air pockets in the center. If you want a solid-wood box, first make an acoustically correct box out of particle board. Then cover it with solid oak flooring or whatever solid wood you want to use.

A table saw or radial arm saw is great for cutting out parts for speaker enclosures. It can even be done with a portable circular saw, sabre saw, or (ugh) hand saw. Historians report that during the Civil War, Abe Lincoln used an axe to make speakers for the White House. For cutting speaker holes, a router is ideal and a sabre saw works fine too.

Assembling the enclosure

Glue every joint. We glue our enclosures together with PVA white glue (e.g., Elmer's glue). A six-sided box is one of the strongest structures known, and the glue makes joints stronger than the wood itself. Contrary to popular belief, extreme strength is not required in speaker enclosures. (The speaker cone undergoes far greater forces than the enclosure, and it's made of paper.) Again, what counts are vibration proofing and airtightness. Rattles and leaks both have the same cure: all joints must be glued together to make an airtight, buzz-proof joint. If you have any cracks or doubtful joints, caulk them with silicone rubber glue, also known as RTV or Silastic. It hardens to pure rubber without shrinkage.

For vibration-proofing, glue is superior to nails or screws. "Point" fasteners (like nails or screws) alone will allow a joint to move and "creak" between the fasteners; so regardless of whatever other fasteners you use, you must also use glue. Screws are very nice but they are a lot of work to use and represent a 200% overkill in strength. The best approach for most people is nails plus glue, relying on the nails to hold until the glue dries.

We use bevel cuts at the corners for appearance, but simpler butt joints are just as good acoustically and may be easier for you to make (Figure 8). If you do make bevel joints, you can use rubber tubing, bungee cord, nails or masking tape to hold the sides together until the glue dries.

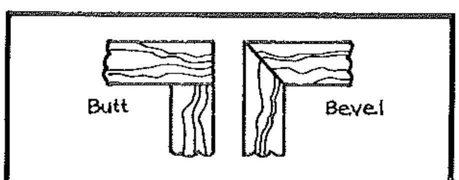

Figure 8. Bevel joint (right) compared to butt joint (left).

Bracing

Often in a multiway system the front board is weakened by cutting several large holes in it for the drivers. Strengthen it if necessary by running a solid wood brace (nailed and glued) from the weakest point on the front board (usually the narrow space between the woofer and midrange holes) to the back. Install the brace at the same time you put in the front board.

Acoustic damping

Line your enclosure on the inside with 1-inch thick fiberglass or Dacron (polyester) to prevent standing waves (echoes) which will give the upper bass and lower midrange response a "singing in the shower" coloration. Cover all sides but the front. If you use fiberglass, use heating duct fiberglass, not the wall or ceiling insulation with foil. The latter sheds glass fibers that can get into the woofer voice coil and ruin it. Most people nowadays use Dacron as it is much easier to work with and does not provoke allergic reactions.

Use more stuffing to pull down the Q of the woofer section (Qtc) if necessary. Weak magnets cause high Q (common) and oversize magnets can cause too low Q (rare). A high-Q peak in the low end frequency response will make the bass sound "boomy" or "flabby".

Stuffing the enclosure will increase the damping in a small-magnet system, so if the bass in your completed system sounds boomy, particularly with a repeated one-note sort of "bomp", the damping is too low. Add stuffing until the bass becomes "solid."

If your bass sounds weak, the damping is too high, and if the box is fully stuffed you should take some material out. What if the bass is weak even with no stuffing? You need a better woofer.

In a vented speaker, you should install the damping material as a liner only. The vented system needs open space in the center of the box to avoid obstructing airflow through the vent.

Grille cloth.

Mount the grille cloth on a separate frame that may be removed from the front of the speaker. That way you can change the grille cloth easily if necessary, and you can operate the speaker without the grille cloth to observe speaker cone motion. Excessive cone motion is symptomatic of turntable rumble, acoustic feedback, low frequency oscillations in your amplifier, or an enclosure that is not airtight. It causes stretching of the speaker suspension and misalignment of the voice coil, resulting in eventual speaker failure.

Mounting the grille cloth on a separate frame also spaces it away from the front board, preventing it from slapping against the front board.

Make the grille frame out of 1/2-inch thick material. You can either cut it out of particle board, with oversize openings for the drivers to radiate through, or you can nail and glue together a frame out of solid lumber.

Velcro hook-and-loop material (available at sewing stores) or Hedlock fasteners are good for attaching the grille frame to the enclosure. Staple one-inch strips of the Velcro hook material to the corners of the front board, and matching pieces of the loop material to the grille frame.

Paint the front board and grille frame black so that the drivers, which are generally black, will not be visible through the grille cloth. Most grille cloth materials that are transparent to sound are also partially transparent to light. Conversely, most lightweight materials that are fairly transparent (hold it up to the light and look through it) will pass sound without appreciable attenuation and work as grille cloth.

The best grille cloth material we have found is double-knit polyester stretch fabric. The polyester yarn is a "hard" material with low sound absorption and double knit is an open weave with lots of "airspace" to pass the sound. In addition, the stretch fabric retains its attractive appearance; it has less tendency to get baggy on the grille frame than other fabrics, and it is easily cleaned by vacuuming.

Assembling the speakers

Most of the details of assembly have already been covered. However, it is important that some steps in assembly be done before others. Installing the crossover after the box is closed would be difficult, to say the least. We recommend that you use the checklist below in assembling your speakers.

Assemble the sides, top, bottom and back of the enclosure together if you haven't done so already, so the enclosure looks like Figure 6.

1. Paint the front and rear panels.
2. Oil the enclosure.
3. Test the drivers and crossover.
4. Fasten the crossover in place.
5. Install the damping material (if rolls of damping material are used, this step may follow the next).
6. Glue in the front board and any braces.
7. Connect the drivers.
8. Fasten the drivers in place.
9. Mount the grille cloth.

Here are a few tips on the individual steps in assembly.

1. Painting the exposed wood

We use black latex paint, which cleans up easily with soap and warm water before it is dry. **CAUTION: Use all paint, oiling mixes, and silicone rubber glues only in a well-ventilated area. KEEP OUT OF THE REACH OF CHILDREN. Do not take internally. Avoid contact with eyes. Wash hands after each use. And all that.**

Paint the front of the front board, the front of the grille frame (including the inside edges of the holes), the back edge of the enclosure top and sides, and the back of the back board (including the inside edge of the crossover hole).

2. Oiling the enclosure

This step assumes you have a hardwood-surface cabinet such as walnut or oak veneer. Vinyl does not need oiling.

For hardwood veneer enclosures use a tung-oil-based oiling mix. Watco and other "Danish oil mixes" are similar.

Oiling brings out the beauty of hardwood and also hardens the wood and makes it much more durable. Unlike paint or varnish which forms a coating that can be chipped off, an oil finish soaks into the wood and makes the wood stronger.

First, wipe the box clean with a damp paper towel. After it dries, pour a small puddle of oil on one side of the box and sand it into the wood with No.400 wet-or-dry sandpaper. Sand WITH the grain of the wood to avoid making scratches across the grain. Sand gently and use plenty of oil. After sanding to a smooth feel, wipe off the excess oil and sanding dust with a paper towel or clean rag.

Do not allow excess oil to dry on the surface; the idea is for it to penetrate and harden in the wood. After oiling all sides, lay the enclosure on its back on newspaper to dry for two hours.

After the wood has soaked up the first coat, you may make additional applications. One coat leaves a "natural" finish. You can re-oil the wood periodically to restore the attractive appearance it had when new. Additional applications over the years continue to increase the beauty of the wood.

3. Test the drivers and crossover

It is a good idea to test the drivers and crossover before you glue them in.

WARNING: When hooking up your speaker, take care to avoid touching together the two wires from the amplifier. Some amps can be damaged if they try to send power through a short circuit. When making these connections, make sure the power to your amplifier is OFF, and the volume is all the way DOWN.

Connect the input terminals on the crossover to your amplifier. Speakerlab crossovers use pushbutton terminals--push in the button on the terminal strip and insert the bare end of your amp-to-speaker wire into the hole. Release the button and the terminal will grip the wire.

Turn on the amplifier, put on some music, and turn up the volume slightly. Do not turn the volume up high---high volume levels before the speakers are installed may damage the woofer by excessive cone excursion. Put your ear to each driver and listen; you should hear sound coming from each driver.

If there is no sound from any driver, check to see that the amplifier is working properly, and that the wiring is good between amplifier and speakers.

Check out the crossover level controls. With your ear to each driver, vary its level control and verify that the driver's level changes. For Speakerlab crossovers with switches, the level changes are small and are easier to hear in a "live" room--one with many hard reflecting surfaces. The level changes are easier to hear on FM interstation hiss than on music.

4. Gluing in the crossover

With the cabinet on its back, squeeze a thin,1/4-inch wide bead of silicone rubber glue around the crossover hole 1/2-inch away from the hole (See Figure 9). Little glue is needed to hold the crossover, and using little will make the crossover easier to remove should you ever need to.

Make sure you get the crossover network in right-side-up. Lay the crossover down over the hole and press it down on the glue. If excess glue squeezes out onto the outside face of the crossover, let it cure fully then trim it off with a sharp knife. Trying to wipe the glue off while it is still wet will only smear it.

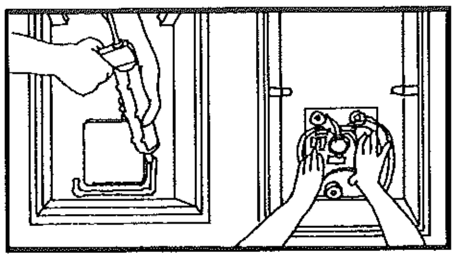

Figure 9: Installing the crossover.

5. Installing damping material

The easiest way to do this is to staple a 1-inch thick Dacron or fiberglass liner to the sides, top, bottom and back. This will prevent internal echoes that cause a "boxy" sound.

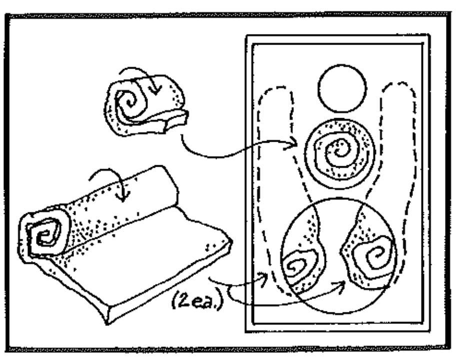

Figure 10: Installing the damping material when rolls are used.

6. Installing the front board.

Make sure you install the front board (Fig. 11) right-side-up with the woofer hole towards the bottom. You can use silicone rubber glue, or most any other kind, to glue it in. If you use a front-to-back brace, mark its position before attempting to glue in the front board. The brace can be held by silicone rubber glue on both ends, or by nails and glue.

7. Connecting the drivers

Points to watch in connecting the drivers: Check and re-check WHICH wire goes to WHICH driver, and be particularly careful of the POLARITY. It is easier to recheck your wires three times than reopen the enclosure

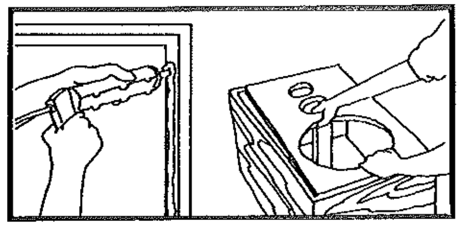

Figure 11: Installing the front board.

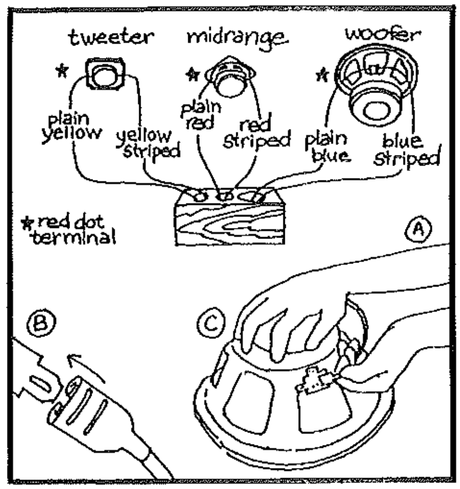

Figure 12: Connecting the drivers.

once. Figure 12a shows a typical wiring scheme. Check the color-code for your crossover network.

Most drivers are marked with a colored dot, number 1, or symbol T1 for the "plus" terminal; the other is the minus (or common) terminal. You can test unmarked drivers for phase by applying a 1-1/2 volt battery to the terminals. When the battery plus terminal is connected to the driver plus terminal, the cone will move OUTWARDS.

Nearly all Speakerlab drivers have the plus terminal marked with a red dot.

Speakerlab crossovers use solderless push-on terminals for the connectors (Figure 12b). When pushing the connectors onto the midrange and tweeter terminals, be careful not to bend or swivel the terminal inadvertently.

Check to make sure that all connectors are on all the way. Most of the tabs on the drivers have a hole that engages a corresponding bump on the connector. (Fig 12c).

On some drivers the terminals stick out and have to be bent back after the wires are attached to fit in the mounting hole.

Make sure the woofer connectors do not contact the metal frame.

8. Mounting drivers with screws and hold-down clips

Position a gasket under the edge of the driver and engage the short end of the mounting clip inside the basket edge. Using an even, tightening pattern, screw down the driver.

9. Gluing in the drivers

Squeeze a bead of silicone rubber glue around each hole (Fig. 13a). Set each driver in its hole and push down into the glue. Do not wipe the excess glue away. Leave the bead of glue around the driver until it dries and hardens. Later you can take a sharp knife or razor and cut and peel away the excess glue, leaving the speakers beautifully finished (Fig. 13c).

Do make sure there are no air leaks around the rims of the drivers.

When mounting the drivers, make sure you keep the crossover wires out of the holes in their frames--if a wire should touch the paper woofer cone, for example, the speaker will buzz.

Let the speaker system dry on its back for at least four hours. The woofer is heavy and may fall out if the cabinet is stood upright before the glue is allowed to reach full strength.

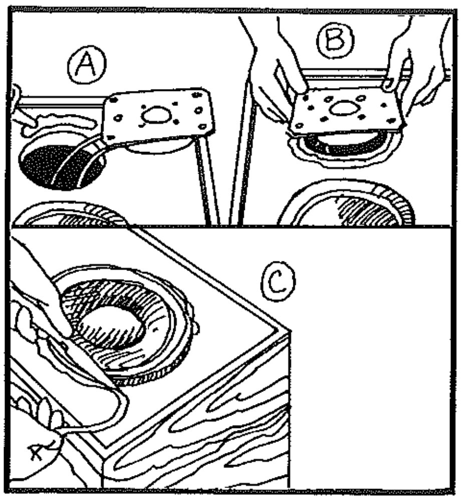

Figure 13: Gluing in the drivers.

10. Mounting the grille cloth

Staple Velcro or Hedlock fasteners to the grille frame before mounting the cloth. Put them an inch or so in from the frame edge to clear the cloth.

Fig. 14 shows how to staple the grille cloth neatly around the grille frame. Staples every two inches or so are about right.

Most grille cloth materials are slightly different on one side than the other and have a noticeable grain pattern. Make sure the grilles for both speakers have the same side of the material out and the grain going the same way.

If you use Speakerlab drivers in your speakers, put Speakerlab nameplates on the fronts so your friends can appreciate your good judgement.

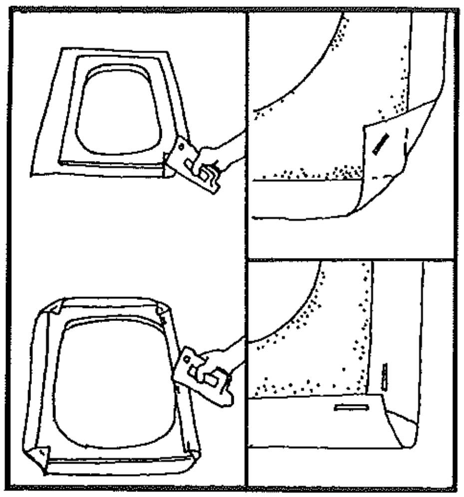

Figure 14: Mounting the grille cloth.

The last word....

Well, we just can't finish this chapter up without throwing in a plug for our kits. So this is it.

A kit is great if you want a proven, optimized design. We tailor the design of the drivers in our kits to work with each other, for much better overall system operation than you can get from driver models chosen at random. As the final step in designing a kit model, we adjust the crossover circuit to correct any residual frequency response amplitude or phase anomalies in the driver types used. This gives the system better response characteristics than the drivers themselves!

On the other hand, if you want to try out a design of your own, the only way to go is to build a system from scratch. So go to it. Most of the new principles and advances in speaker science have been made not by established manufacturers, but by people who were home constructors at the time.

References

1. Neville Thiele, "Loudspeakers in Vented Boxes," JOURNAL OF THE AUDIO ENGINEERING SOCIETY, May and June, 1971. Reprinted in Ref. 4, 181.

Chapter Two: Transmission Line Speaker Design.

We have been interested in transmission line (TL) speaker systems and have been collecting data on them for some time now. We have received many letters such as this one:

"I recently heard a pair of transmission line speakers at a friend's house and was very much impressed. I am toying with the idea of building my own transmission line woofers. My ideal in a woofer is not volume, nor low frequency response PER SE, but rather accuracy (unmuddled, umboomy, distortion-free, having high resolution). The transmission line design seems to fit my ideal. Can you supply speakers suitable for transmission line designs? I would appreciate any comments you might have on what I have said."

Donald A.
Chicago, Illinois

We have assembled the information we have found for the benefit of others interested.

We have not done EXPERIMENTAL WORK at Speakerlab on TL speakers yet. Anyone wishing to use this UNTESTED DATA to experiment on his or her own may proceed at their own risk. We can give you our best wishes but not much else.

Sufficiently forewarned, read on.

A transmission line for sound is simply a pipe with a wave going in one end and coming out the other.

A great variety of speaker designs have been called "transmission lines". Most have a front-firing woofer mounted in a vented enclosure. The vent is so positioned that the length of the sound wave path from the rear of the woofer to the vent is an appreciable fraction of a wavelength at low frequencies. This causes a phase shift of the loudspeaker rear wave before it is emitted from the vent. The woofer rear wave starts out 180 degrees out of phase with the front wave, but after the delay of passing through the transmission line (enclosure interior) its phase is shifted so that it reinforces (rather than cancels) the front wave.

So far as we know, transmission line systems are designed basically by trial and error. "The bigger the better" seems to be the cardinal rule. Bailey's paper (reference 9) is the best writing on the subject we've found but gives no method for calculating enclosure dimensions. We know of no work setting forth theory of operation on a mathematical basis, nor one that gives design methods.

No one has mapped out a quantitative theory for the transmission line as Neville Thiele, Richard Small, and others have done for vented (bass reflex) loudspeaker systems. The radiation impedance of the line load on the woofer, the impedance change from the line impedance to free air at the port, and the delay effects of the line length all make the system very complex to analyze.

Most transmission line speakers use either a line of constant cross-sectional area, or one that tapers and gets smaller towards the port. In constant-area lines the port opening is usually the same size as the cross-section of the line. In tapered lines, the port is usually smaller.

In the constant-area designs, the area generally approximates the area of a square with sides equal to the nominal (outer) diameter of the woofer used. "Simple wave theory" would suggest that the pipe cross-section be no less than the woofer cone area, as a minimum. Thus for 6 to 18-inch woofers the areas would be as given in Table 1.

Table 1

Woofer diameter	Typical Tran. Line Cross Section Area
18"	213-324 square inches
15"	140-281 square inches
12"	85-180 square inches
10"	57-125 square inches
8"	28-80 square inches
6.5"	24-42 square inches
6"	20-36 square inches

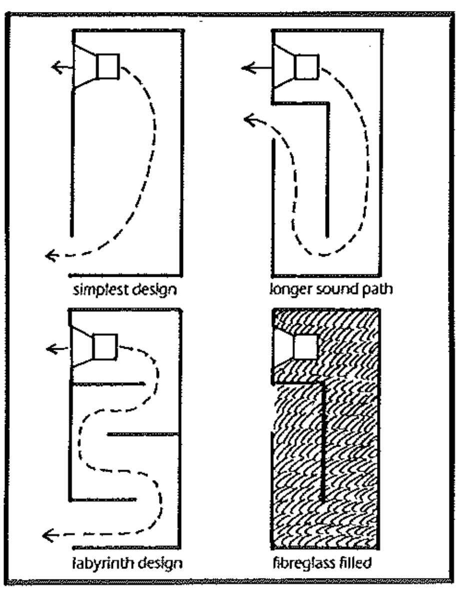

Figure 1: Some typical transmission line configurations.

Theory

We can formulate a half-baked sort of theory by making the simplifying assumption that the main effect of the transmission line is to delay the rear wave of the woofer, after which the delay emanates from the port. This theory ignores troublesome details like woofer characteristics, radiation impedance of the port, reflections inside the line, load of the line on the woofer, etc., etc. "Guesswork" might be a better name for it than "theory".

Anyway, if we assume that the port's output is the same as the woofer output, except that it is delayed by the time it takes a sound wave to travel the length of the line, then we may draw the following conclusions.

At the frequency where the length of the transmission line is exactly one-half wavelength, the vent's sound output is exactly inphase with the woofer's output and total radiation is maximum. The frequency at which this takes place we may call the characteristic frequency of the line fc:

$$f_c = \frac{V}{2L} \qquad (1)$$

where v is the speed of sound (1,130 feet per second), and L is the length of the line in feet. As an example, a line 10.6 feet long (the size recommended later in these notes for a 15" woofer) has a characteristic frequency equal to 53 Hz.

The time delay T for a wave to go from the woofer to the port will be simply

$$T = \frac{L}{V} \quad (2) \qquad \text{Note that } T = \frac{1}{(2f_c)} \quad (3)$$

Assume the output of the woofer at frequency f is equal to $A \sin(2\pi ft.)$

Then the output of the port will be

$$-A \sin(2\pi ft.+\theta)$$

where 0 is the phase shift due to the wave being delayed in the line. The minus sign is because the port wave originates from the rear of the woofer and is, therefore, opposite in phase from the woofer radiation which the phase shift corrects).

So the total radiation is

$$A \sin(2\pi ft.) = A \sin(2\pi ft.+\theta) \quad (4)$$

Our handy trigonomentry handbook says

$$A \sin(x) - A \sin(y) = 2A \sin\left(\frac{x-y}{2}\right)\cos\left(\frac{x+y}{2}\right) \quad (5)$$

so the total output may be written

$$2A \sin\left(\frac{\theta}{2}\right)\cos\left(2\pi ft.+\frac{\theta}{2}\right) \quad (6)$$

We are interested in only the total amplitude part of this, that is the constant part,

$$2A \sin\frac{\theta}{2} \quad (7)$$

Comparing this to the total amplitude of the woofer output, we can define:

$$\text{"line response"} = \frac{\text{total output}}{\text{woofer output}}$$

$$\text{which equals } \frac{2A \sin\frac{\theta}{2}}{A} \quad (8)$$

$$\text{which equals } 2 \sin\frac{\theta}{2}$$

At frequency f, the phase shift in the line, 0, is equal to

$$\theta = Tf \text{ cycles} \qquad (9)$$

$$\text{or } \theta = 360Tf \text{ degrees}$$

$$\text{since } T = \frac{1}{2f_c}$$

this can also be written

$$\theta = \frac{360}{2f_c} f \quad \text{degrees} \qquad (10)$$

$$= 180 \frac{f}{f_c} \quad \text{degrees}$$

So "line response"

$$= 2 \sin \frac{180 \frac{f}{f_c}}{2} \qquad (11)$$

$$= 2 \sin \left(90 \frac{f}{f_c} \right)$$

This function is plotted in Figure 2. As frequency is varied, the vent output goes in and out of phase with the woofer, boosting or cancelling the woofer's output. Whether the vent output adds or subtracts at any given frequency depends on how many wavelengths of phase shift the line delay time causes at that frequency.

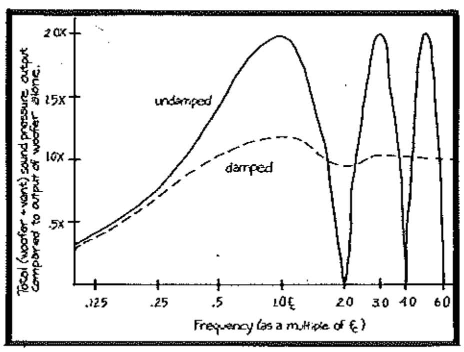

Figure 2: Line response of a transmission line speaker.

This graph covers only the effects of the line and does not take into account the response of the woofer itself; that is why the vertical axis is labelled 1.0X, 2.0X, etc. which means "1.0 times woofer output", "2.0 times woofer output" and so on.

The beauty of a transmission line system is that the vent-radiation boost may be used to extend the response of the system beyond the response of the woofer itself. From about .3fc to about 1.7fc, the graphed response is above 1.0X, meaning that the vent radiation adds to the woofer radiation and the total radiation exceeds that of the woofer alone.

However, the undamped line has terrible resonant peaks and dips at frequencies that are integral multiples of fc (1.0fc, 2.0fc, 3.0fc, etc.).

We need to damp the line to eliminate the high frequency peaks and dips. Damping does this by absorbing high frequency energy in the line so it is not radiated by the vent. In a well-damped line, above fc most of the radiation will be from the woofer alone.

On the other hand, the line damping should not disturb the low-end response; it should allow full vent radiation below .5fc and some peaking around fc, to achieve the desired vent-boost effect. The dashed line on the graph shows what we want to accomplish.

Fortunately, the damping effectiveness of a lossy transmission line is greater when there are more wavelengths in the lossy path-length. Since at higher frequencies more wavelengths fit in the line, the damping effectiveness increases with frequency and we can get the absorption of highs and transmission of lows we need.

We may define the TL cutoff frequency (fo) as the lowest frequency at which the total output is greater than the output of the woofer alone, that is, greater than 1.0X in Figure 2. Above this frequency the port output ADDS to the woofer output. Below "fo" the port output SUBTRACTS from the woofer output. For an undamped line this condition means:

$$2 \sin \left(9 \frac{f_o}{f_c} \right) = 1.0$$

$$\text{so:} \quad f_o = \frac{f_c}{90} \sin^{-1} \frac{1.0}{2} \qquad (12)$$

$$= \frac{1}{3} f_c$$

Since fc is the frequency at which the transmission line is a half-wavelength long, "fo" is the frequency at which it is a sixth-wavelength.

Thus the hypothetical 10.6 foot long line with fo equals 53 Hz mentioned earlier as TL cutoff frequency:

$$f_o = \frac{1}{3} \times 53 = 17.7 \text{Hz}$$

This is rather low, but remember two things- one, we took no account of the woofer's response and it is very unlikely to be flat down to 12.2 Hz; two, contrary to our assumptions some or most of the energy reaching the port will be reflected back up the line rather than radiated.

Nevertheless, the fact that a transmission line is theoretically useful at frequencies for which it is a sixth-wavelength or longer, as shown in equation (12), is an interesting and useful fact. We have never seen it published before.

Line Shape

Low-end cutoff frequency can be reduced by increasing the line length. One may be tempted to reduce the cross-sectional area of the line and crowd a longer line with more folds inside the enclosure in order to get an extremely low cutoff frequency. Unfortunately, a small cross-section raises the impedance of the line. According to "simple wave theory," if the impedance is too high, the motion (and radiation) of the woofer cone will be severely limited. Also the mismatch between line's vent end and free space will be high, and most of the energy going down the line will be reflected back into the line when it reaches the vent rather than be radiated.

This last difficulty may be ameliorated somewhat by making the line tapered, small at the input end and large at the output end (vent). In such a case the design tends to behave more like a horn than a transmission line.

Wave theory notwithstanding, many transmission line systems do use a tapered line that grows smaller near the vent. We suspect that such systems are acting as bass reflex systems with a very long ducted port with some delay.

Damping

The transmission line must be lined or filled with sound-absorbent material such as fiberglass to attenuate the rear wave from the woofer.

The amount of damping material needed can be determined by careful listening tests. If the extreme low bass is "boomy"or "flabby," indicating a poorly damped bass peak, filling the enclosure with an absorbent material will damp down the resonance and make the bass response "tighter". Excessive absorbent material will cause overdamping and bass loss. Some designers prefer natural materials such as wool, while others use only synthetics such as fiberglass or Dacron (polyester) which have less tendency to settle and are immune to mold and insects.

Simply lining the TL interior with fiberglass is a good place to start for trial-and-error damping adjustment. To adjust the damping further, tack in more glass with ripples that extend towards the center of the linespace as shown in Figure 3.

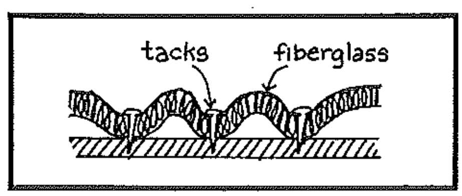

Figure 3. Method of tacking in fiberglass or other sheet damping material to provide damping for transmission line.

For designs like (b) and (d) of Figure 1, installing an access plate on the bottom makes it easy to experiment with the quantity and placement of resistive material. Or you can put the material in through the vent and woofer holes with somewhat more difficulty.

Suggested Dimensions for Transmission lines

Figure 4 shows a suggested transmission line design and dimensions for woofers of various sizes. The dimensions given are the INSIDE dimensions. Don't forget to allow for panel thickness in your calculations of panel sizes to cut. We recommend use of 3/4-inch particle board or plywood except for the 6-inch and 8-inch woofer designs, where 5/8-inch material is acceptable.

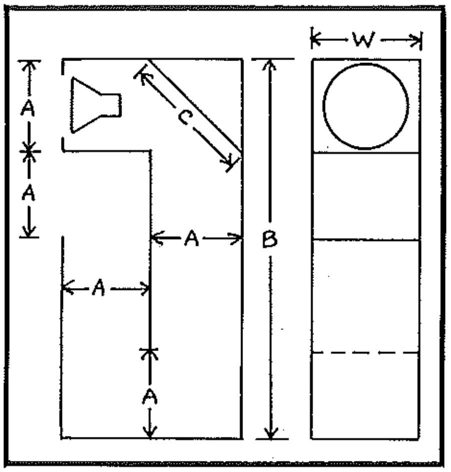

Figure 4. Suggested dimensions for transmission line speaker enclosures. The 45-degree piece (length C) prevents standing waves between the woofer and the back of the enclosure.

Woofer	A	B	C	W
6"	6"	24"	8.5"	7.5"
8"	8"	32"	11.3"	10.0"
10"	10"	40"	14.1"	12.5"
12"	12"	48"	17.0"	15.0"
15"	15"	60"	21.2"	18.8"

Midrange and Tweeter

Since the woofer fires forward in many TL designs, it radiates upper bass and midrange effectively. Therefore a two-way system with only a tweeter to carry the highs will work. However, you'll get better high-end performance with a three-way system. The midrange, tweeter, and crossover are most conveniently placed on top of the transmission line bass section.

Measurements

The woofer will have a low resonant frequency in the transmission line due to the large air mass in the line. The resonant frequency is easily measured by driving the woofer with an audio oscillator (through a 1000-ohm resistor) and measuring the voltage across the woofer. The voltage is approximately proportional to the woofer impedance. The frequency at which the voltage peaks is the resonant frequency. A TL system may have either one or two resonant frequencies. If it has one, theoretically it should be equal to or near fc. If the woofer's free-air resonant frequency is above fc, that will throw off the theory.

If a system is acting like a bass reflex (vented) system in disguise, it will have two pronounced resonant impedance peaks. The frequency of minimum impedance between the peaks is equal to the enclosure resonant frequency, fb, in Thiele-Small theory (References 11and 12).

Suitable Drivers

Woofers used in transmission line systems must have adequate suspension stiffness to prevent excessive cone excursion, since the enclosure is not sealed and provides no "air spring" effect to limit cone motion. Nearly all 15-inch woofers are adequate in this regard and we recommend the Speakerlab W1508S (8 ohms) for transmission line applications. Where a 12-inch or 10-inch woofer is required, the W1208R and W1008B are suitable, though they should be operated at half their normal power ratings.

Woofers for transmission line systems should also have low resonant frequencies to prevent inadequate woofer response from limiting the low-end performance on the system. Acoustic suspension woofers have low resonant frequencies, but most have such high compliance (weak suspension stiffness) that they are liable to be damaged by excessive cone excursion at frequencies below the system cutoff. Such woofers may be used if you employ an active filter to eliminate subsonic energy that can overdrive the cone.

References

This list of references on transmission line speakers is not complete but it contains all the information we have. We welcome any additions anyone may have.

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13. Audio Engineering Society, "Loudspeakers; An anthology of articles on loudspeakers from the pages of the JOURNAL OF THE AUDIO ENGINEERING SOCIETY, Vol. 1 - Vol. 25 (1953-1977)." 1978, Publications Office, AES Inc., 60 East 42nd Street, New York, NY 10017.

Chapter Three: Bass Horn Design.

Horns and direct radiator speakers both employ a moving diaphragm that generates pressure waves (sound) in the air (Figure 1). The difference is that while a direct radiator moves the air directly, the horn driver launches the wave down a tube. The tube then tapers out to launch the waves into the room. The horn is more efficient because a moving diaphragm radiates energy much more efficiently when it is mounted over the end of a tube than when it is merely out in the open. The direct radiator mostly just moves the air out of the way and only a small percent of the energy becomes pressure waves in the air. In the case of the horn driver, the air can't move to the side--the horn walls are in the way--so it has to compress, thereby translating a substantial fraction of the energy into pressure waves (sound). So the horn acts like a transformer, coupling the moving diaphragm to its air load.

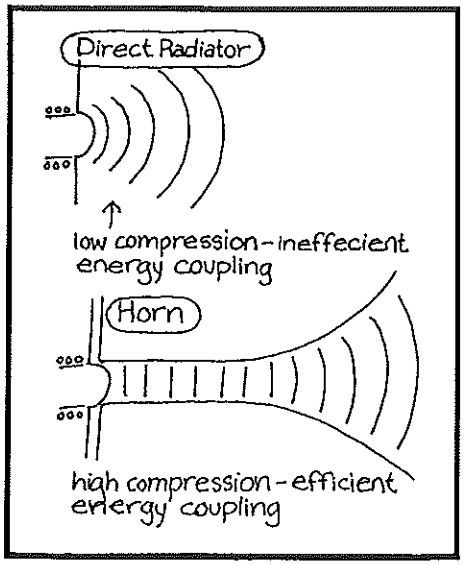

Figure 1. ABOVE--motion of a direct-radiator diaphragm generates low compression and gives inefficient energy coupling. BELOW--motion of a horn-loaded diaphragm generates high compression for efficient energy coupling.

A big disadvantage of horns is their size. To effectively "launch" the sound wave into the room, a horn mouth must be at least one-quarter wavelength in diameter. At 30 Hz a quarter wavelength is 8-feet. A horn with an 8-foot mouth is a heck of a big horn. You can cheat the laws of physics by designing your horn so that it uses the walls of the room in which it is placed as the mouth--most rooms having ceilings 8-feet high or thereabouts. Thus the structure you build that goes in the room can itself be substantially smaller.

A straight bass horn is theoretically the most perfect low-end reproducer. Unfortunately, they are totally impractical to manufacture and sell commerically because of their huge size. However, it is not difficult to make a straight horn as a built-in part of a house; the large size is not so much of a problem and again, you can use the house walls as part of the horn.

Building a straight horn system into your house is a lot of work, but people who have done it report superb results:

"I have finally completed my sound system and I can't describe how completely satisfied I am with its performance.

The sound from your equipment was far beyond any good sound that I was accustomed to hearing, I was completely amazed at what I heard coming out of the whole speaker system.

The photos that I have enclosed might tell part of the story but listening to this thing is unreal. Wish you could be here."

Raymond Magaro
Los Angeles, CA

Raymond's system is diagrammed in Figures 2 and 3.

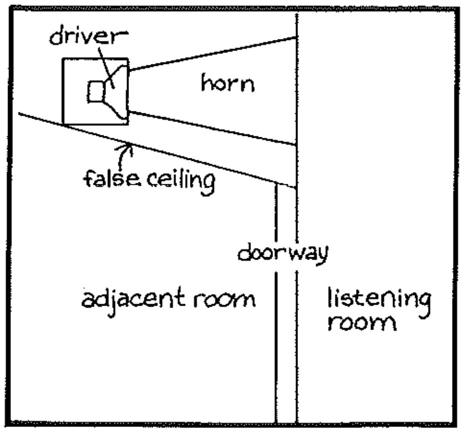

Figure 2. Cross-sectional outline of Magaro's straight horn installation. His home has a stepped ceiling and he mounted the speakers in the empty space above the "step".

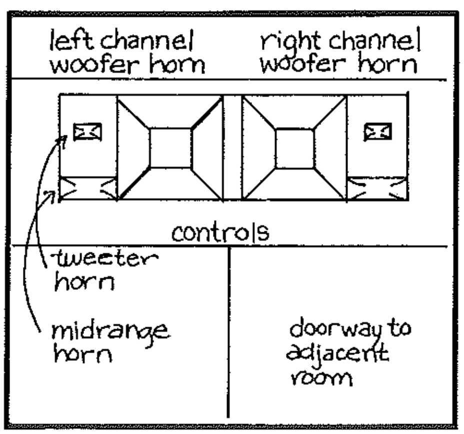

Figure 3. Appearance from listening room of Magaro's installation.

Theory

An exponential horn is a horn whose cross-sectional area increases a certain percentage for each inch of distance from the throat. Its name derives from the fact that the area can be expressed as an exponential function:

$$A = A_0 e^{md} \qquad (1)$$

where e is Eulers constant (2.718), and m is the horn taper factor in inverse inches. Figure 4 shows the definitions of A, Ao, and d.

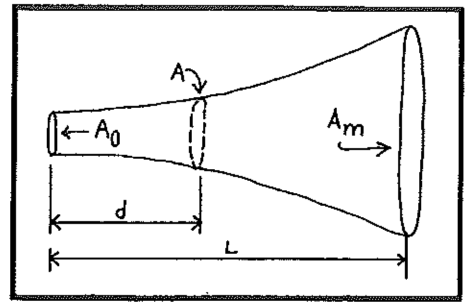

Figure 4. "Ao" is throat area, "Am" is mouth area, "L" is length of horn, "A" is cross-sectional area at distance "d" from throat.

The greater the value of m, the faster the horn expands. For example, if m equals .10 per inch, then the horn expands by a factor of

$$e^{.10} = 1.105$$

for each inch, or about 10% per inch.

The theoretical low frequency cutoff point of a horn is set by the taper factor according to the formula

$$f_c = \frac{mc}{4\pi} = 1080m$$

where fc is the theoretical cutoff frequency in Hz, c is the speed of sound (13575 inches/second), and 4 pi is a constant (12.57).

The "theoretical cutoff" is merely a mathematical quantity and a well-designed bass horn can actually respond down to about 65% of fc in a normal sized listening room. (It is much easier to get good bass in a normal listening room than in the infinite vastness known as "free space". There is no basis for the common belief that you need a big space to reproduce big wavelengths. If that were true, headphones would have no bass at all.)

Mouth Area

To obtain effective low frequency radiation, the mouth must also be of adequate size. If the mouth is too small, low frequency waves will not be "launched" when they reach the mouth but instead will be reflected back up the horn. The horn mouth, if circular, should have a diameter at least equal to one over pi wavelengths of the largest wave (lowest frequency) to be reproduced. If the shape is other than circular, the mouth should have equivalent area. Based on this rule, the mouth area should be:

$$A_m = \frac{c^2}{4\pi f^2}$$

f = lowest frequency to be reproduced.

Throat Area

The third basic parameter of horn design is the throat area. The throat area will depend on the driver used, but a coarse rule of thumb that works well is to make the throat area equal to half the driver diaphragm (piston) area. Beranek (Ref. 2, Ch. 9) goes into this in detail.

A good throat area for low-resonance (15-20 Hz) 15" woofers is about 50 square inches.

Horn Length

The last basic parameter of horn design is length. The length (L) is not independent of the other parameters; it depends on the taper rate (m) and the mouth and throat areas (Am and Ao). The formula is:

$$L = \frac{1}{m} \ln \frac{A_m}{A_o} \quad (2)$$

Once Ao, Am, L and m are set, intermediate values of A may (theoretically) be obtained from equation (1). This is an easy task for those fortunate enough to possess a calculator with log and exponential functions. Those not so blessed can determine the throat areas by a graphical technique using semiloragithmic graph paper. This is paper with a logarithmic scale on one axis and a linear scale on the other, as in Figure 5. This kind of paper has the characteristic that any straight line drawn on it obeys an equation of the form

$$y = ae^{bx}$$

where y is the logarithmic scale and x is the linear scale. A glance will show that this equation is of the same form as equation (1); only the variables are different.

If you know Ao and Am and the horn length, you can use Figure 5 to determine intermediate cross-sectional areas as follows. Put a dot at "d equals 0" and "A equals Ao" for the throat area, and another at "d equals horn length" and "A equals Am" for the mouth area. Connect the two dots by a line. You can then read off the horn area "A" required for a given distance "d" along the length of the horn.

In many cases you will know Ao, Am and fc but not the length. The theoretical cutoff frequency (and taper rate) is proportional to the slope of the line. We have predrawn some slopes for different cutoff frequencies so you can also use the graph without having to know the horn length beforehand--a difficult calculation using equation (2). To use this technique take 3 steps:

(1.) Place a dot at the throat area Ao on the left axis.

(2.) Draw a line of the slope required for the designed cutoff, that is, make your line parallel to the line labeled with the cutoff frequency you want.

(3.) Make a dot where your line passes through your mouth area (Am). Read the horn length on the horizontal axis.

A Shortcut (Pun Intended)

You will find that bass horns designed strictly according to the formula are quite long. Olson demonstrated (Reference 1, Pg. 114) that the horn can be shortened considerably, with little adverse effect on the bass response, by using a faster taper rate in the throat end. In addition this extends the high frequency range of the horn.

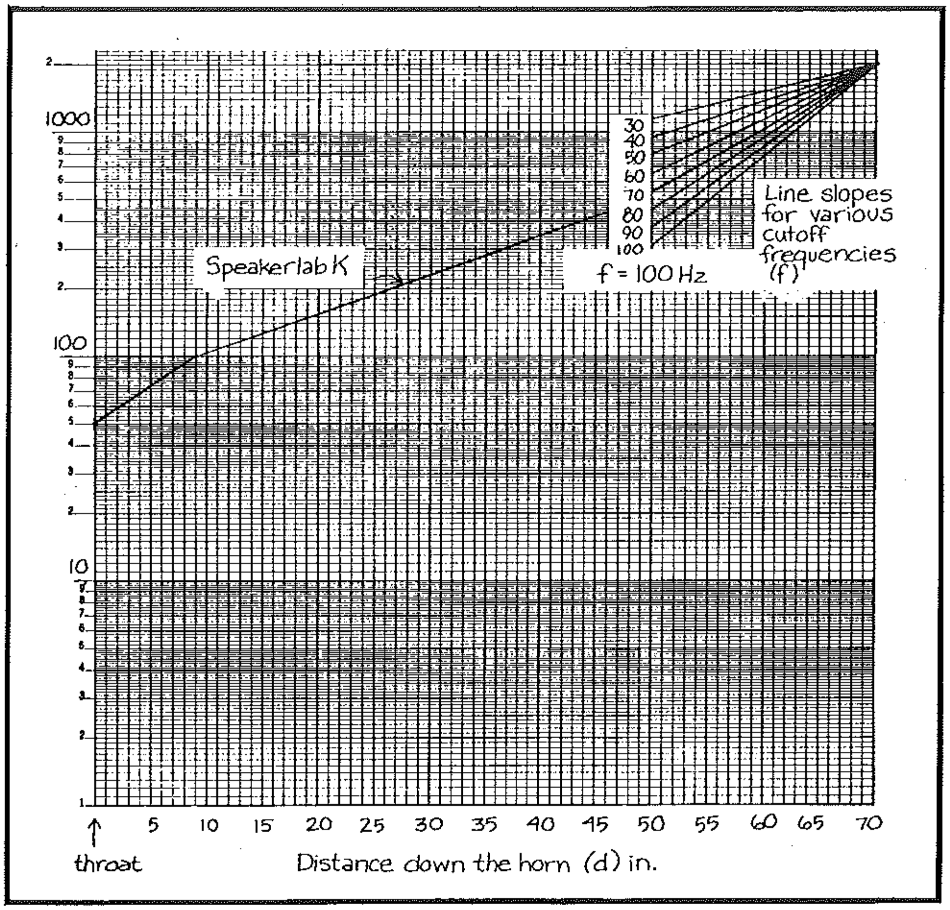

Figure 5: Logarithmic scale for horn length.

Shape

The cross-sectional SHAPE of a horn is of little importance--the cross-sectional AREA is the key factor. Changes of shape along the path of the horn are OK as long as they are gradual, with the exponential taper formula held true in the transition region.

Most horns are round or rectangular (or both, changing shape along their length) because these are convenient shapes to fabricate and install.

Angular Dispersion

Dispersion (uniform coverage over a wide angle) for a horn operating below 500 Hz, particularly a folded horn mounted in the corner of a room is no problem. A speaker in a corner really needs only 90 degrees dispersion because the walls prevent the listener from ever standing more than 45 degrees off axis. Furthermore, low frequencies have little directional characteristic. You can design your horn for construction convenience with regard to dispersion consideration.

Folds and Standing Waves

The width of the horn at folds sets an upper limit to the high frequency performance of folded horns. Sound will not go around a bend if the width of the bend is a half wavelength or more. Standing waves occur between the walls of the horn (sideways in the horn). Because of this, low frequency horns rarely work above 700-1000 Hz and should be crossed over to a midrange driver at some lower frequency.

Behavior Around Cutoff

Horns have a rolloff around the theoretical low end cutoff. In this rolloff, the horn presents a mass load to the driver. It is possible to adjust the total spring constant of the driver (cone suspension stiffness plus the stiffness of the air chamber behind the cone) so that the spring load equals the mass load in the region just below the theoretical cutoff. Then the driver's stiffness reactance will cancel the horn's mass reactance and the low end frequency response will be flat over an extra third or half octave.

The effect probably takes place to some extent in every horn system that has a sealed chamber behind the driver; you can maximize the effect by trial and error, using drivers of different resonant frequencies or varying the air volume of the sealed chamber by partially filling it with air-displacing solid material such as foam blocks. Not fiberglass--fiberglass itself is 99% air.

Some designs use a constriction at the throat to increase the throat mass reactance, which at low frequencies can counteract the stiffness of the driver cone suspension and air chamber.

Construction

The horn itself is not difficult to construct. Build it out of 3/4-inch plywood or particle board. If you use plywood, we suggest AB interior grade. Brace the horn walls with 1" x 2" braces set on edge. Braces keep the horn walls from vibrating. Seal the corners of the horn with silicone rubber glue (RTV). Airtightness is extremely important.

Ringing

A horn is supposed to be a channel for vibrations in the air, which means that the horn wall itself should neither vibrate nor radiate sound. Many horns are made of metal, which is quite susceptible to carrying vibrations--especially if the mouth is unsupported, in which case the horn forms an almost perfect bell. The result is horn wall resonances which make the reproduced sound "tinny" or "harsh.".

When using a metal horn, mount the mouth on a flange of some material with high internal damping (such as wood) to prevent ringing.

Horn Folding

Bass horns are big, and the biggest dimension is the length. Folding the bass horn changes it from a long skinny structure to a more compact shape.

Perhaps the simplest way to fold the bass horn is to coil it around on itself as in Figure 6. The University Classic design is of this type.

The Classic works best on the floor and against a wall.

Paul Klipsch was the first to come up with a really practical folded horn. As well as the horn itself being folded, it used the corner of a room as part of the horn. His classic paper in the "Journal of the Acoustical Society of America" (Ref. 3) describes it fully.

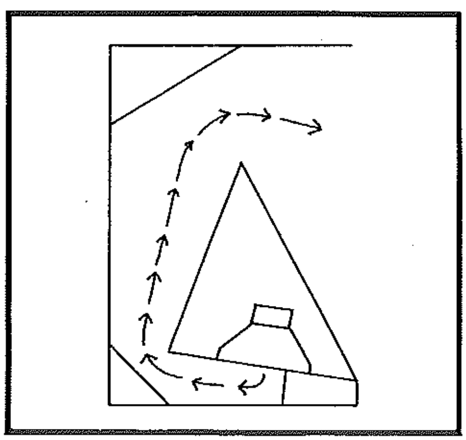

Figure 6. Folded bass horn using the coiled-around configuration.

Corner Reflectors

When laying out a fold, install a reflector in the corner to help the wave turn the corner. (See Figure 7, which shows a method for determining where to place the reflector.) Extend line BC to point D. Extend line AB to point E. Now draw line DE between points D and E. It can be shown from ray optics that a piece placed along line DE will make a perfect reflector to "turn" the wave.

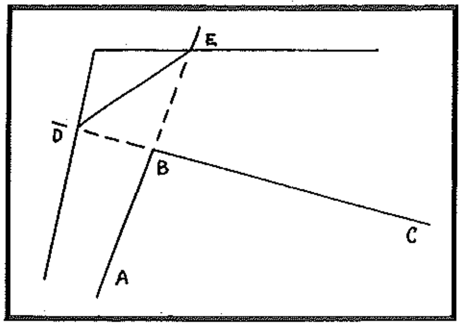

Figure 7. Extend line AB to point E. Extend line CB to point D. Now draw line DE between points D and E.

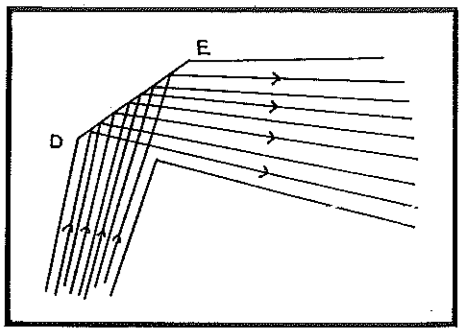

Figure 8. This shows how reflector DE "folds" the sound wave around the corner.

Room Placement

Most folded horns benefit from corner placement but do not require it. However, nearly all require placement on or near the floor and against or near a wall for optimum bass performance. A speaker near the intersection of the wall and floor is radiating into quarter-space (Figure 9); and in a corner, it is into eighth-space. The limited radiation angle and "partial space" provides a more nearly optimum load on the mouth of the horn, which in the case of bass horns is nearly always smaller than optimum for radiation into half or full-space.

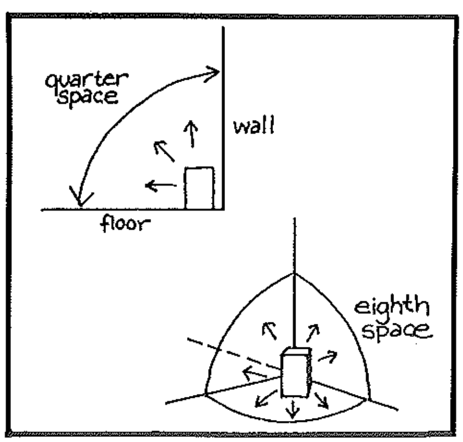

Figure 9. A speaker near the intersection of wall and floor (left) radiates into quarter-space. A speaker in a corner (right) radiates into a solid angle equal to half that, or eight-space.

Frequency Range

A given horn normally will cover only about three octaves at the most, that is, a frequency range of about eight to one from the lowest frequency to the highest. Because of this, horns must be used in multi-way systems--usually a three-way system with separate bass, midrange, and treble sections. It only makes sense to use horns for the high end too.

Equalization

Due to the compromises necessary in bass horns--non-uniform taper rate, folds, less-than-optimum mouth area--they are rarely as flat in frequency response as, say, an acoustic suspension speaker. The horn's virtue is bass that goes low and is powerful. However, a speaker with such bass response will often excite room resonant modes. As a result, the performance of a bass horn can often be improved with an octave-band equalizer. Many installations work well with about 5 dB of boost in the lowest octave (20-40 Hz), 0 dB in the next octave (40-80 Hz), and 5 dB of cut in the next octave (80-160 Hz). The other equalizer sections are set flat (0 dB). Records vary tremendously in the amount and quality of bass in the bottom octave: so judicious readjustment, as necessary, can have dramatic results.

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3. Paul W. Klipsch, "A Low-Frequency Horn of Small Dimensions," JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA, Vol. 13:137-144 (1941).

Why Cross Over?

Most speaker system builders buy the raw speakers from a supplier and then build their own enclosure. When it comes to crossovers, those who wish to buy one ready-made are usually faced with a very limited selection with little to go on to choose the right one. Those who wish to build their own have difficulty finding parts and building information. We hope that this report will provide some of the needed information.

Just about every high quality speaker system in use today has at least two speakers, a woofer and a tweeter. Two speakers can divide the audio spectrum between themselves and each do a better job. One specializes in low frequencies, the other in high frequencies. The crossover network routes to each speaker the part of the spectrum it can best reproduce.

After speakers and enclosure, the most important single factor in speaker system performance is the crossover network. A properly designed crossover can bring out good sound even from cheap speakers, and is absolutely necessary to obtain from high quality speakers their maximum performance.

The most significant advantage of a good crossover is its ability to reduce intermodulation distortion in the speakers to very small proportions. Intermodulation distortion is the worst form of distortion in speakers and is what gives the speaker its generally accepted status as the "weakest link" in the stereo reproduction chain. This distortion is generated by the motion of the woofer voice coil in its magnetic field, which cannot be made perfectly uniform. The woofer's sensitivity changes as the coil moves from regions of high field to low field, and any mid or high frequencies being reproduced by the woofer are modulated by the low frequency that caused the cone motion. This form of distortion is particularly objectionable because it does not occur in nature and is very discordant to the ear. The cure is to exclude the mid and high frequencies from the woofer and have them reproduced by one or more other speakers. A crossover which achieves this will make a vast improvement in sound quality. The speaker designer who takes the trouble to incorporate the right crossover in his design will find that the results repay his diligence immensely.

This chapter covers the basic principles of parallel section crossover networks. Parallel section crossovers have a separate section for each speaker, which makes it possible to tailor a network for any possible combination of speakers.

Two way, 6 dB per octave crossover

The simplest form of crossover network is shown in Figure 1. It is called a two way system because the frequency spectrum is divided into two parts, above 1670 Hz and below 1670 Hz. The low frequency section is merely an inductor which acts as a low pass filter and the high frequency section is a capacitor which acts as a high pass filter. The value of the inductance of the low frequency section is determined by the formula:

$$L = \frac{R}{2\pi f}$$

where L is the inductance, f is the crossover frequency, and R is the impedance of the woofer.

The value of the capacitance in the high frequency section is determined by the similar formula:

$$C = \frac{1}{2\pi f R}$$

Here C is the capacitance, f is the crossover frequency, and R is the impedance of the tweeter. In both formulas the frequency is in hertz and the impedance is in ohms. Capacitance is in farads (F) and inductance in henries (H).

Both of these formulas are expressed in graphical form in Figure 2. Once the crossover frequency and impedance have been chosen, the required parts values may be read directly from the graph.

At the crossover frequency the signal to each speaker will be 3dB down, and each speaker will be radiating half power compared to what it radiates in the center of it's frequency band. Thus the total radiation from both speakers at the crossover frequency will equal the power radiated at the other frequencies, and there will be no hole or peak in the frequency response.

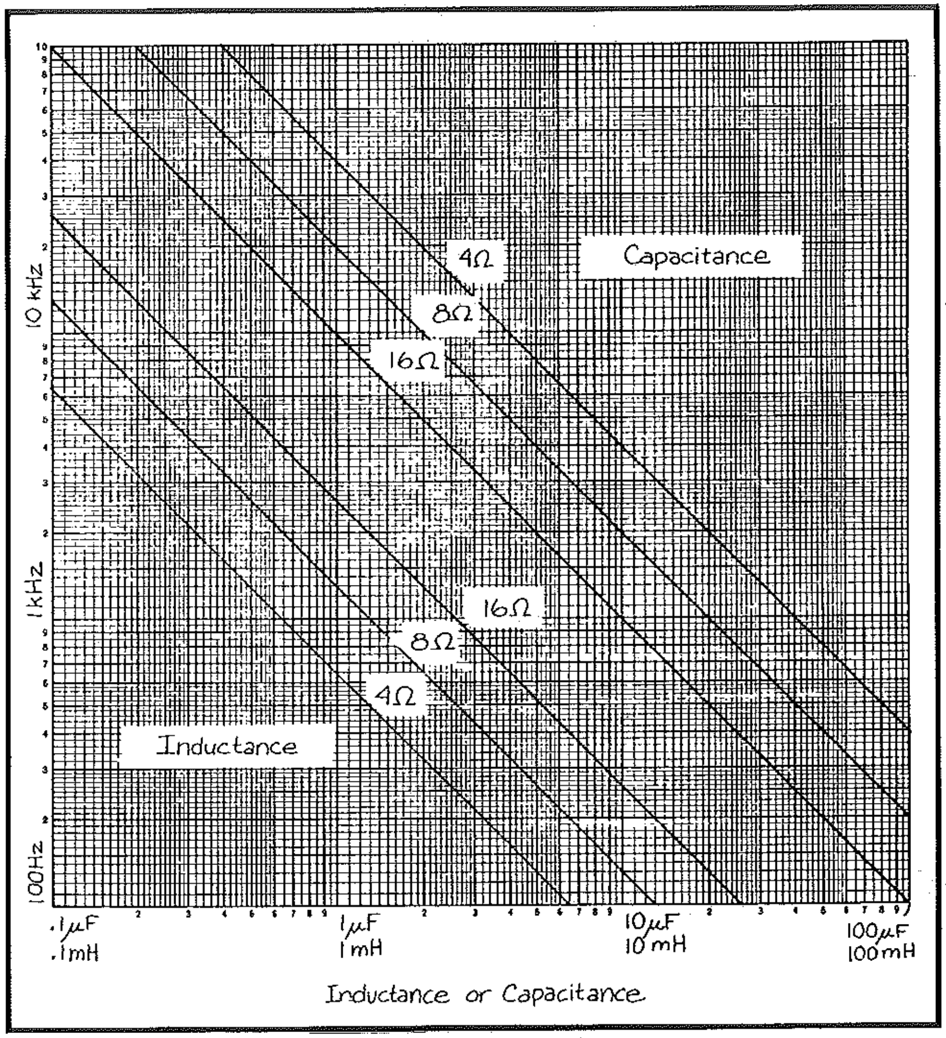

Figure 2. Capacitance and inductance values for 6dB/octave crossover sections. Use the upper three lines to determine capacitance d, the lower three lines to determine inductance.

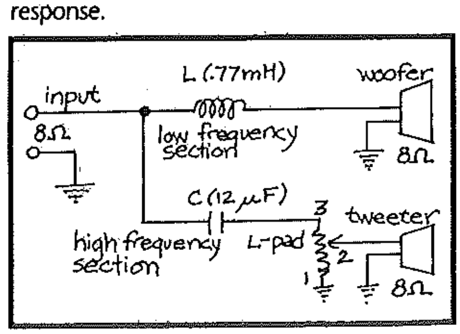

Figure 1. Two-way, 6dB/octave crossover. The ground symbol merely indicates a common connection point inside the speaker system.

This type of network has a rather gradual cutoff beyond the crossover frequency. It is called a 6dB per octave crossover because outside its desired frequency band each speaker receives 6dB less signal for each octave beyond the crossover frequency.

12 dB per octave crossover

A crossover network with a sharper cutoff characteristic, 12dB per octave, is shown in Figure 3. The two systems have this in common: each consists of two sections, one for the woofer and one for the tweeter. The two sections are fed in parallel from the input terminals. All the crossover systems discussed in this report share this feature: each speaker is driven by a crossover section for the part of the spectrum allocated to that speaker, and all the sections are connected to the input terminals in parallel.

Crossover networks may be made for speaker systems of any complexity by connecting appropriate sections in parallel.

Each section passes only signals of frequencies falling in its share of the audio spectrum. It blocks out signals of other frequencies. If the impedances of all the speakers in a system are the same, the impedance of the whole system at the input terminals will be the same as that of the speakers. At any given frequency, only one speaker is drawing power; the other speakers are isolated from the input by their crossover sections. (Right at a crossover frequency, of course, two speakers will be drawing partial power.)

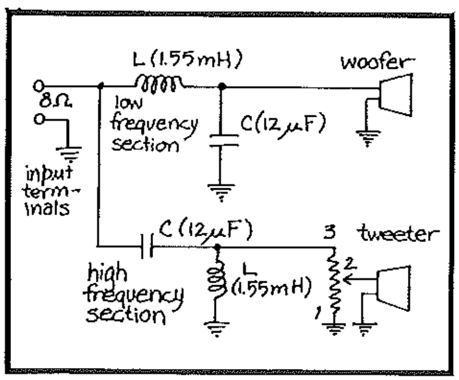

Figure 3. Two-way, 12dB/octave crossover. The parts values shown are for an 8-ohm system crossing over at 1160 Hz.

12 dB per octave midrange section

Highpass and lowpass 12dB/octave sections may be cascaded (the output of one feeding the input of the next), as in Figure 4, to give a 12dB/octave midrange section without difficulty. Use equations 1 and 2 to calculate the parts values.

$$① \ L = \frac{R}{\sqrt{2}\,\pi f} \qquad ② \ C = \frac{1}{2\sqrt{2}\,\pi f R}$$

The formulas are the same as those for 12dB highpass and lowpass sections alone. Also the formulas are the same for both the highpass and lowpass parts, so both inductors have the same value, as do both capacitors.

These formulas hold good for both highpass and lowpass sections. The formulas are similar to those for 6dB/octave sections except that the inductance is 1.414 times the 6 dB/octave value and the capacitance is .707 times the 6dB/octave value.

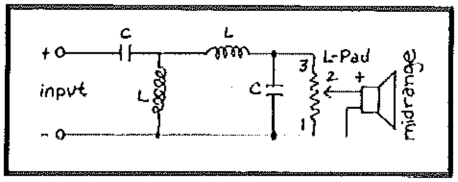

Figure 4. Cascade 12dB/octave midrange section for use in a parallel-connected crossover network. This section is simply connected parallel at the input terminals with the woofer and tweeter sections. Calculate in values for C and L from equation (1) and (2).

18 dB per octave networks

Even steeper rolloff characteristics can be obtained with more complex networks. Figure 5 shows an 18dB/octave system. This type is rarely used except for special applications.

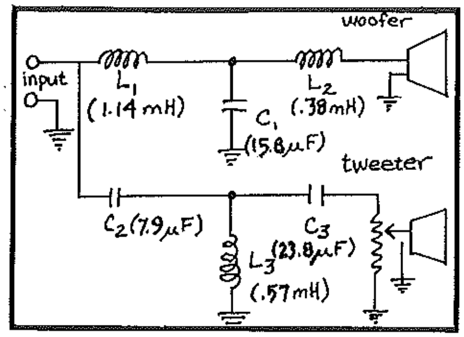

Figure 5. Two-way 18dB/octave crossover. The values shown are for an 8-ohm system crossing over at 1670 Hz.

The formulas for the 18dB/octave crossover in Figure 5 are:

$$L_1 = \frac{3R}{4\pi f} = 1.5 \times 6\text{dB/octave value}$$

$$L_2 = \frac{R}{4\pi f} = 0.5 \times 6\text{dB/octave value}$$

$$L_3 = \frac{3R}{8\pi f} = 0.75 \times 6\text{dB/octave value}$$

$$C_1 = \frac{2}{3\pi f R} = 1.33 \times 6\text{dB/octave value}$$

$$C_2 = \frac{1}{3\pi f R} = 0.667 \times 6\text{dB/octave value}$$

$$C_3 = \frac{1}{\pi f R} = 2 \times 6\text{ dB/octave value}$$

Three-way crossover systems

The crossovers described so far have all been two-way systems. Three-way crossover networks use sections similar to those already presented for the woofer and tweeter, plus a third section of an inductor and capacitor series for the midrange. A typical three-way system is illustrated in Figure 6.

The system of Figure 6 is a 6dB per octave three-way crossover. A 12dB per octave system appears in Figure 7. Note that while true 12 dB sections are used for the woofer and tweeter, the midrange section is of the same type as is used in the 6 dB system. The cutoff slope of this type of section lies between 6 and 12dB and depends on how many octaves wide the midrange band is.

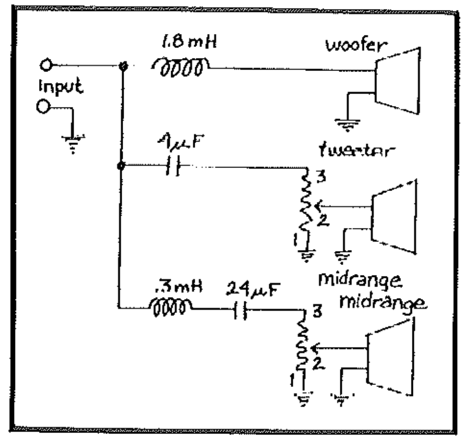

Figure 6. Three-way, 6dB/octave crossover. The parts values shown are for an 8-ohm system with crossover frequencies of 710 and 5000 hertz.

For 6dB/octave midrange sections of the type shown in Figure 6 a special formula is necessary. The midrange inductor and capacitor interact and shift the crossover frequencies that the simple formulas may not be used to calculate their values.

$$L = \frac{R}{2\pi\left(f_H - f_L\right)}$$

$$C = \frac{1}{2\pi R}\left(\frac{1}{f_L} - \frac{1}{f_H}\right)$$

Where FH equals higher crossover frequency and FL equals lower crossover frequency. Highpass and lowpass 12dB/octave sections have a little less interaction but are better used in a cascade-connected crossover.

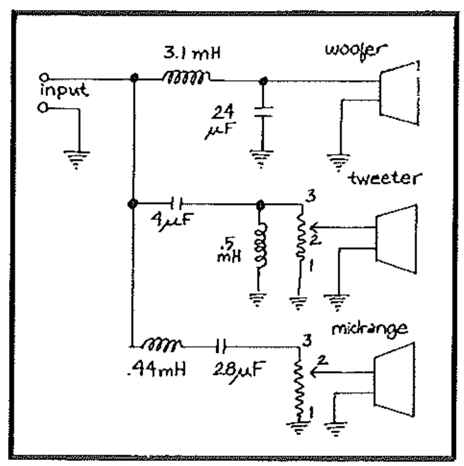

Figure 7. 12dB/octave three-way crossover. The parts values given are for an 8-ohm system with crossover frequencies of 590 and 3500 Hz.

Four-way crossovers

Four-way, five-way, and even more complex crossover systems may be assembled by putting additional sections in parallel. The extreme high and low frequency sections are either 6 or 12 dB per octave networks as described for two-way systems. The other sections are midrange-type networks. A typical four-way system with crossover frequencies of 250, 1300, and 8000 Hz is shown in Figure 8.

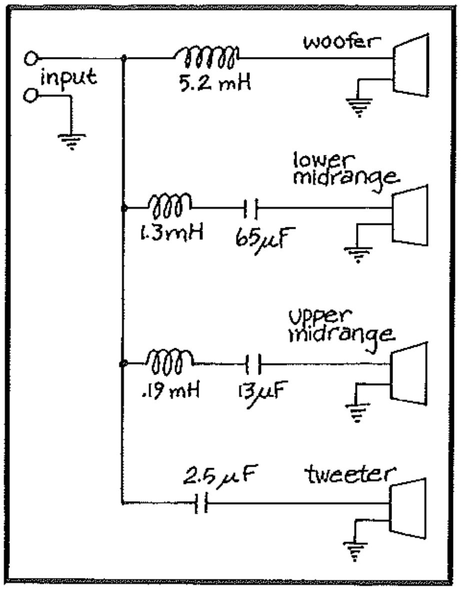

Figure 8. Four-way, 8-ohm crossover system with crossover frequencies 250, 1300 and 4000 hz. The speakers for four-way systems are often called woofer, midrange, tweeter, and supertweeter.

Other types of networks

In situations where speakers all have the same basic impedance, two other crossover network types are useful. They are the cascaded and the series network. Both these network types can offer certain advantages. However, before we start, remember that all speakers must have the same impedance. This is not a serious problem. Take, for example, a system planned with a 4-ohm woofer, 8-ohm midrange and 16-ohm tweeter. We simply parallel the higher impedance speakers with resistors to bring the total impedance to 4-ohms (the lowest value) in each case. The resistor would be a 10-watt unit with the following resitance:

$$R_{parallel} = \frac{Z_{speaker} \times R_{total}}{(Z_{speaker} \quad R_{total})} \qquad (3)$$

Which yields 8-ohms for the 8-ohm unit and 5.33-ohms (5.1 or 5-ohms will do) for the 16-ohm unit. Table 1 shows the correct parallel resistors for various impedance values.

Nowadays 16-ohms impedance is rarely used (except in horn drivers) because transistor amplifiers will generally deliver only half as much power to a 16-ohm load as to an 8-ohm load. Horn drivers have very high efficiency so this fact is of little consequence in their case.

Note that reducing the load impedance by paralleling a resistor will **not** put more power into the driver. The amplifier will indeed put out more power into the lower impedance, but the extra power goes into the resistor, not the driver.

Theoretically, you also can **increase** the impedance a driver offers to the crossover network by adding a series resistor, but the series resistance will also change the frequency response of the driver if it is greater than a small fraction (10-25%) of the driver impedance.

Table 1

To get	With a driver of	Parallel a resistor of
4 ohms	8 ohms	8.0 ohms
4 ohms	16 ohms	5.1 ohms
8 ohms	16 ohms	15.0 ohms

The cascaded network

A 6dB/octave parallel network midrange section needs special formulas— not so in the cascaded network. A crossover in this type is shown in Figure 9. You will note that the high frequencies from the bass/midrange crossover are routed to the midrange tweeter crossover.

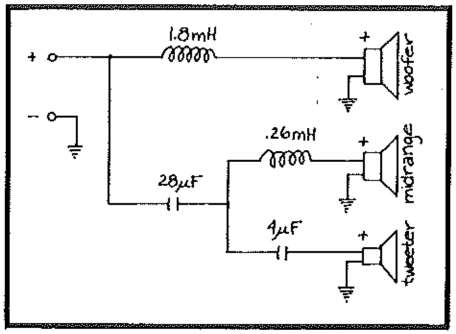

Figure 9. Three-way, 6dB/octave cascaded crossover. The parts values shown are for an 8-ohm system with crossover frequencies of 710 and 5000 Hz. Calculate values for other frequencies from equations (4) and (5). L-pads for the tweeter and midrange are not shown but would normally be used.

The cascade design relies on the principle that a 6dB parallel network with equal load impedances (say 8-ohms) in each branch "looks like" a pure, simple 8-ohms at the input to the network. Since the midrange/tweeter crossover in Figure 9 looks like 8-ohms at all frequencies, there is no interaction between compartments and they are all calculated by the simple 6dB/octave formulas:

$$(4) \quad C = \frac{1}{2\pi f R} \qquad (5) \quad L = \frac{R}{2\pi f}$$

This network also has a further less obvious advantage. The extreme lows (below the woofer/midrange crossover frequency) are eliminated from the tweeter at a 12 dB/octave slope. This is beneficial because some tweeters can create excessive intermodulation distortion with even small amounts of energy fed to them at low frequencies.

A greater advantage is realized with 12dB/octave networks since considerable duplication of values occurs and one can avoid a trip to the heart of Japan for exotic capacitor values. As mentioned before, without the cascaded system, some interaction would occur between crossover components resulting in odd-ball differences in component values and hair-raisingly complex formulas for calculation.

A fully cascaded 12dB/octave network is shown in Figure 10. Note that all components are calculated from the 12dB/ocatve formulas (1) and (2), and that while two more components are required, only four different values are needed as compared with six in conventional parallel designs. Much better midrange performance can be expected with this crossover.

You can expand the cascade principle to create four-way or even five-way systems with little difficulty — so long as all the load impedances are the same. For 12dB cascade networks we recommend that the crossover frequencies be at least one octave apart. For 6dB cascade networks this precaution is not necessary.

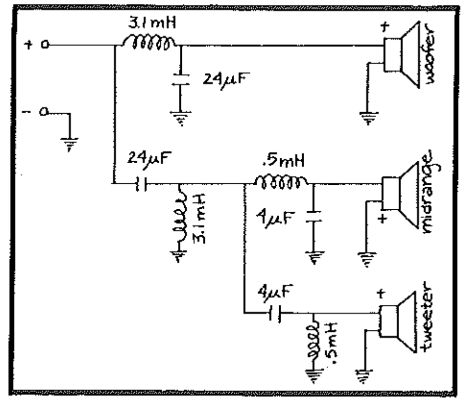

Figure 10. 12dB/octave cascaded three-way crossover. The parts values given are for an 8-ohm system with crossover frequencies of 590 and 3500 Hz. Parts values are calculated from equations (1) and (2). Note that the midrange phase is reversed, as for other 12dB/octave networks. L-pads would normally be used on the midrange and tweeter.

The series network

It is also possible to design a crossover with the woofer and tweeter connected in series (when their impedances have been equalized). Figure 11 shows such a connection for a crossover frequency of 1670 Hz.

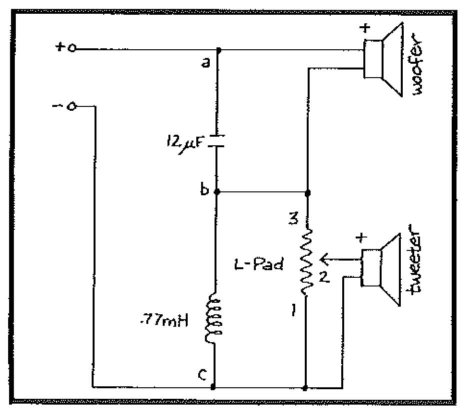

Figure 11. Series-connected 6dB/octave crossover for a frequency of 1670 Hz. Parts values are calculated from equations (4) and (5); the values shown are for an 8-ohm system.

The parts values are the same as for the parallel connected circuit and can be determined from the 6dB/octave formulas, equations (4) and (5).

The last type of crossover we will discuss is the 12 dB/octave series connected circuit. As usual, this network has its own band of devoted followers. For those interested, Figure 12 shows the most popular series configuration. The formulas are:

$$(6) \quad L = \frac{R}{2\sqrt{2}\,\pi f} \qquad (7) \quad C = \frac{1}{\sqrt{2}\,2\pi f R}$$

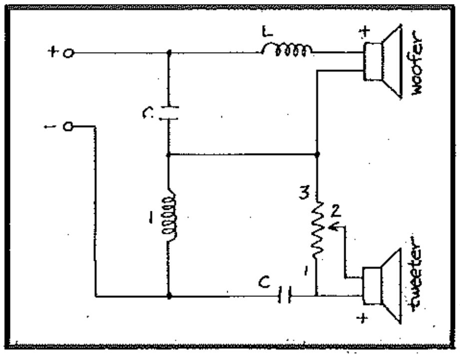

Figure 12. 12dB/octave series-connected network. Parts values are calculated from equations (6) and (7). Note that the tweeter's phase is reversed.

Note that compared to the parallel wired 12dB/octave network, the inductances are cut in half and the capacitances are doubled. Thus this network can be cheaper to make than the parallel form.

Some woofers have a large voice coil inductance and a minor advantage of this circuit is that the voice coil inductance can be "allowed for" as part of the series inductance in the woofer leg of the crossover network. Make the series inductor:

$$L_{actual} = L_{calculated} - L_{voice\ coil}$$

You use a smaller series inductor and avoid the high frequency loss and phase problems that voice coil inductance can cause if not provided for in the design. You can also use this trick with the 6dB/octave and 18dB/octave parallel crossovers.

Is series better than parallel?

Theoretically, any frequency response characteristic and phase characteristic that can be achieved with a parallel crossover network can be achieved with a series network, and vice versa (though the parts values to do it may be different).

The 6 dB/octave networks of both the series and parallel type have constant voltage and constant phase characteristics; that is, if you "add back together" the output from both high and low sections, the total output signal is equal to the input signal in both amplitude and phase.

18 dB/octave networks also are constant voltage — the "added back together" signal is equal in amplitude to the input signal. However, the constant-phase characteristic is not present. While at very high and very low frequencies the output is in phase with the input, there is some phase shift around the crossover frequency. Again, this is true for both the series and parallel configurations.

The 6 dB series network is beginning to come back into favor so it may be appropriate to make a few additional comments on it here. In particular, two myths have grown up around it.

Myth one: It has been thought that the constant-amplitude, constant-phase characteristic is a property of the series network. Examining Figure 11, it is easy to see that the sum of the output voltage (Vab plus Vbc) must be equal to the input voltage (Vac), since the two loads are in series with the signal source. However, as mentioned above, this is also true of the 6 dB parallel network, though more difficult to prove mathematically (Ref. 1).

Myth two: Ashley, one of the leading authorities on crossover network design, in his 1973 paper (Ref. 2), analyzed the series hookup of Figure 11, which he called a "zeta" crossover. He wrote that reducing the inductance and increasing the capacitance in this network (zeta less than 1.0 in his notation) would give some peaking and make the slope around the crossover point greater than 6dB so it "acted like" a 12dB network. Many designers accepted this as true at the time. Ashley brought this up again at the 1976 International Loudspeaker Symposium at the University of Colorado. He observed that the input impedance of this network drops when a larger capacitor and smaller coil than those normally called for (equations 4 and 5) are used. In fact, for zeta equals .5 as he originally suggested to get "12dB/octave" performance, the input impedance drops by a factor of 2, doubling the power drain on the amplifier. So he now recommends that this circuit be used only with the normal coil and capacitor values. He remarked that systems built using normal values also sound best. So used in the best way, the simple series hookup is still just a plain old 6dB/octave network and has no special "sharpness" around the crossover point.

Disadvantages of the 6dB per octave series network.

While some designers favor the 6dB series network, we recommend against it for three reasons:

1) Resistance in the coil prevents attenuation of bass energy to the tweeter from ever exceeding about 70 to 76 dB. See Figure 13. Thus, there is always about 5 to 10% of the bass voltage reaching the tweeter, moving the cone around and causing intermodulation and pushing the tweeter closer to its overload limit for both mechanical and thermal damage.

In the parallel system, on the other hand, the signal to the tweeter continues to decline (at the 6dB/octave rate) indefinitely as you go down to frequencies below the crossover point.

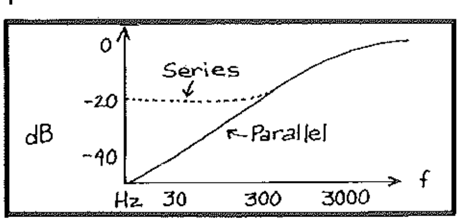

Figure 13. Signal to the tweeter for 6dB/octave series and parallel crossover networks. The crossover frequency is 3000 Hz and the coil resistance is equal to 10% of the tweeter impedance in both cases. Note that the series network allows substantially more bass signal "leakage" into the tweeter.

2) If a wire to the coil comes loose, the tweeter will be in series with the full output current from the amplifier and is likely to be destroyed.

3) 6dB/octave is a very gentle slope for a two-way system in any case, and works the tweeter harder than necessary. A 12 dB section for the tweeter lets it run cooler with less bass "leaking in" to cause intermodulation distortion. Parallel network design makes it easy to match a 6 dB woofer section and a 12 dB (or steeper) tweeter section, which, while not blessed with the mathematical symmetry so dear to theoreticians, works well with real woofers and real tweeters. Figure 14 shows an example of such a "mixed slope" design.

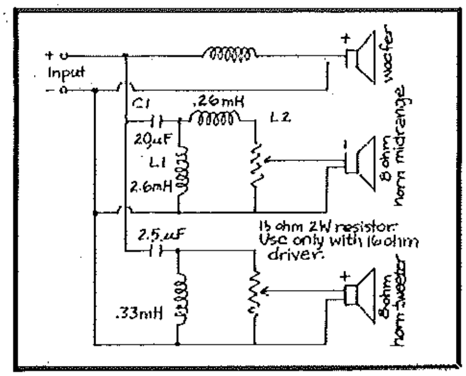

Figure 14. Three-way crossover for horn midrange and horn tweeter. The crossover frequencies are 700 and 5000 Hz. The high-side rolloffs are 6dB/octave and the low-side rolloffs are 12dB/octave. L1 and C1 are calculated from equations (1) and (2) and L2 from equation (5).

Some partisans of the 6dB series network claim that the system can be more efficient because the inductor and its resistance are not in series with the woofer to cause signal loss. The coil may not be in series with the woofer to their eyes, but it certainly is to the signal current. The current has to pass through either the coil or (worse yet) the tweeter to get to the woofer.

Nevertheless, many manufacturers use the 6dB series network. Its simplicity makes it cheap to build, and this network may be necessary where economy is a dominant constraint. Because of its disadvantages, however, we recommend that if you want a series network, you use the 12dB version as in Figure 12.

Crossover tweaking

A speaker with a crossover designed to theory would work perfectly if the loads and drivers on the crossover network were pure resistances and their acoustic amplitude response were flat with no phase shifts within their range. Alas, this is not the case.

The most advanced crossover designers use a theoretical filter network only as a starting point. They measure the frequency response output of the speaker, make listening tests, and modify the crossover away from theoretical to get better actual results. In this way, the speaker system can be made to perform better than the drivers used. Without frequency response measuring equipment, it is difficult to do this in your own designs. However, there are a few things you can do by ear.

If the sound output in the lower midrange of a two-way system seems weak, it may be because the upper-bass output of the woofer is weak. Moving the woofer crossover frequency up half an octave (41%) or so to allow it to overlap the tweeter may help. It will raise the woofer's acoustic output a few dB's at its upper end and change the woofer tweeter phase relationship, which may have been causing cancellation before.

If the extreme highs in your system are weak, and your tweeter uses an L-pad or resistive padding to match its level to the rest of the system, you can boost the extreme highs by bypassing the L-pad with a capacitor as in Figure 15. The effectiveness of this bypass capacitor will vary with the setting of the L-pad, but since you normally use the L-pad at only one setting, you can choose the capacitor (by trial and error) for that particular setting.

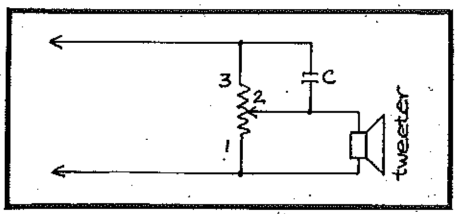

Figure 15. Capacitor shunting the upper resistance of the L-pad to boost the extreme highs. A capacitance of 1-4 uF is usually about right, and should be selected by ear. The arrows connect to the rest of the crossover network.

WIRE GAUGE

Inductance (mH)	#14		#15		#16		#17		#18		#19		#21		#22	
	TURNS	OHMS	TURNS	OHMS	TURNS	OHMS	TURNS	OHMS	TURNS	OHMS	TURNS	OHMS	TURNS	OHMS	TURNS	OHMS
SMALL BOBBIN																
.13													52	.30		
.20															66	
.26									72	.23			75	.46		
.33													84	.48		
.39									90	.29			90	.53		
.52									104	.34			107	.67		
.60					107				112	.37			114	.69		
.77							125	.37	126	.43						
.90	127	.22			132	.32	133	.39	136	.47						
1.1	145		147	.33	152	.35	148									
1.3	153	.29	163	.35	167	.39										
1.6			178	.38	178	.48			185	.68						
1.8			184	.40	180	.51	189	.63	192	.69						
2.2					207	.52	215	.67	216	.79	217	.94				
2.6			224	.53	225	.59	227	.70	229	.84						
LARGE BOBBIN																
3.0									267	.93						
3.7	260	.46	269	.57	276	.69	293	.85							304	2.33
4.4	283	.59	300		300	.83										
5.0			311	.74	317	.85									350	2.79
6.5	330	.70	350	.91												
7.2	343	.76	365	1.00	370	1.02	378	1.34								
9.0					387	1.31	402									
13.5			470	1.40												
20.0															680	6.33

Table 2: Coil winding data. The number of turns required to wind air-coils of various inductance values are given, along with the coil resistance, for various wire sizes. The coils of .13 to 2.6 mH are wound on a bobbin 1.5 inches in diameter (which becomes the inside diameter of the coil) by .75 inches wide. The coils of 3.0 to 20.0 mH are wound on a bobbin or coil former 1.5 inches in diameter by 1.0 inch wide.

Designing coils

A formula for the air-core coils of the type shown in Figure 16 gives:

$$L = \frac{.0008 r N^2}{6r + 9l + 10d} \qquad (8)$$

where
L equals inductance in millihenries (mH),
N equals number of turns,
r equals mean radius in inches,
l equals length of coil in inches,
d equals depth of coil in inches.

Our experience is that the formula is only approximate, giving inductances within about 30% of the desired value. We use an inductance bridge to determine the exact number of turns necessary to get a desired inductance value. This is the tough part — getting the right value. The number of turns of wire required changes with different batches of wire and different winders. Variations in thickness of insulation and winding technique (wire tension, layer compactness) change the outer diameter of the coil, which affects the inductance.

We have found that coils of about the shape shown in Figure 16 (with r equals l equals d) give the most inductance for the least resistance and least weight of wire.

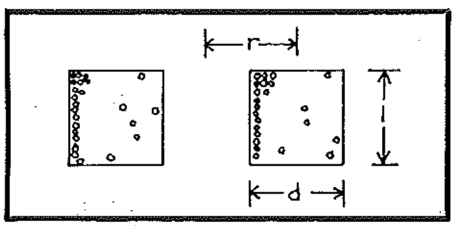

Figure 16. Cross-sectional view of an air-core. Equation (8) gives an approximate formula for the inductance based on r, l, d and the number of turns (N).

The other important consideration for a coil, besides obtaining the correct inductance, is to keep the coil's resistance small to prevent significant power losses in the coil itself. The coil resistance acts like a resistor in series with the inductance. We use a guideline that the coil DC resistance should be less than one-tenth the speaker's impedance. This insures an insertion loss of less than 1 decibel. The resistance is determined by the length of wire you use and its gauge (thickness). For air-core coils under 1mH you can safely use gauge 19, and probably get by with wire as small as gauge 22 (larger gauges correspond to smaller wire thicknesses). To keep the resistance low for coils with much more wire, that is, with many more turns, will require wire of greater thickness (smaller gauge). The largest wire we use is 14 gauge.

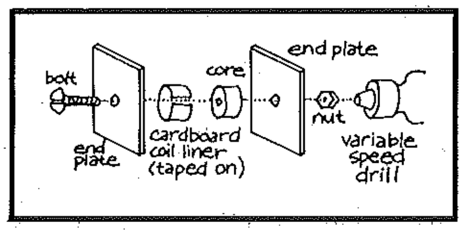

Figure 17. Coil winding jig.

Table 2 gives the data we use for winding our crossover coils. It is based on measurements of the actual inductance and resistance of the coils. Coils wound with standard, round, heavy-film-insulated, magnet wire using this table will have inductances accurate to well within plus 5%.

How to wind coils

You can wind your own coils. Magnet wire is available at electronic hobby stores and electronic distributors.

Figure 17 shows how to make a jig for winding coils with an electric drill.

Make the "core" piece out of dowel 3/4" or 1" long by 1.5" diameter. The bolt is 1/4" by 3" or so. End plates are masonite or 1/4" plywood. Wrap a cardboard coil liner around the core and hold it with plastic electrical tape before winding the coil. The liner enables you to get the coil off the core after winding. Paint a spot on one side of the winding jig and go slowly so you can count the turns. Tie the coil together with electrical tape after winding. You can reuse the coil for as many times as you like since it is not left trapped in the coil after winding (except the cardboard coil liner becomes a permanent part of the coil). Use silicone rubber glue to to mount the coil on your crossover.

Other things you should know

A capacitor may actually be two or more capacitors wired in parallel. The combination capacitance is equal to the sum of the capacitances of the individual units.

The plus terminal of a speaker is marked by a red or green dot, a red insulating washer, or the letters "T1" or "1".

Speaker Impedance

All the speaker systems shown in the figures as examples have had a woofer, tweeter, and midrange of the same nominal impedence. It is not necessary that all the speakers of a system be the same impedance, however. Since the crossover is designed with a separate section for each speaker, it is only necessary that each section be selected for the impedance of its corresponding speaker. Thus, for example, one may use an 8-ohm woofer, 4-ohm midrange, and 16-ohm tweeter in the same system provided that one makes the woofer crossover section for 8-ohms, the midrange section for 4-ohms, and the tweeter section for 16-ohms.

If you have speakers which have unknown impedances you can determine the impedance by measuring the voice coil resistances. Speaker AC impedance is generally 30 to 50% higher than the voice coil DC resistance. Thus 8-ohm speakers will usually have resistance of 5 to 6-ohms; 4-ohm speakers, 2.5 to 3-ohms; and 16-ohms speakers, 10 to 12-ohms. This is not a hard and fast rule and speakers with DC resistances outside the ranges stated are often found. A safe approach is to measure the DC ohms and use the next higher standard impedance value as the speaker impedance. Using this rule, a speaker with a 9-ohm DC resistance would be considered a 16-ohm speaker.

Some speakers use a two number impedance rating system, for example "10/15 ohms." The first number is the DC resistance and the second is the nominal impedance; so this speaker would have a 16-ohm rating by our system.

Some speaker system designs use more than one woofer, or more than one tweeter or midrange. If two woofers of the same impedance are used, they may be wired in series or parallel (usually parallel). The load impedance on the crossover will then be double (series) or half (parallel) the impedance of one woofer. If four woofers are used, they may be wired in series-parallel which results in a combined impedance equal to the impedance of a single woofer. In these cases where woofers (or whatever) in combination are used, the impedance of the combination should be used for the corresponding crossover section.

When the different sections of a speaker system have different impedances, the system impedance which is presented to the amplifier will vary with frequency. This is of no consequence so long as the system impedance never dips below the amplifier's minimum safe load impedance. The latter is 3-ohms or less for nearly all transistor amplifiers rated for use with 4 to 16-ohm speaker systems.

The speaker system impedance will equal the impedance of the woofer section at low frequencies, the tweeter section impedance at high frequencies, and so on. The speaker system builder should rate his system impedance the same as the lowest impedance section. This is the most conservative approach because low impedance loads put the heaviest strain on amplifiers.

L-pads

The L-pads shown in most of the examples are used to set the level of the midrange and tweeter relative to the woofer. Midranges and tweeters are generally more sensitive than woofers and it is usually necessary to "cut back" the signal going to them in order to obtain uniform frequency response. The L-pad is a special kind of volume control which maintains a constant impedance load on the crossover regardless of the control's setting. An ordinary volume control or potentiometer varies its input impedance as the setting is changed. This is unacceptable because it would cause the crossover frequency to change.

Although the L-pad is a three terminal device and is represented by the same symbol as an ordinary potentiometer, it actually contains two variable resistors inside. As one varies the signal level to the speaker, the other equalizes the load impedance on the crossover.

L-pads are wirewound to enable them to handle the high power levels at which speakers operate. When connecting up L-pads it is necessary to wire the terminals properly. The terminal numbers shown in the figures correspond to the numbers on the terminals of the L-pads supplied by Speakerlab. For other types consult the manufacturer's instructions packed with the unit.

If the tweeter and midrange are less sensitive than the woofer, L-pads cannot be used to equalize the speaker sensitivities. L-pads can only reduce midrange and tweeter sensitivities, not increase them. Usually the only time the woofer section is more sensitive is when two woofers in parallel are used. Then it may be necessary to use multiple tweeters and midranges as well.

The proper technique for adjusting L-pads is as follows. First, set all the L-pads to minimum (full counterclockwise). Turn the balance control on your amplifier all the way to one side so you can adjust the L-pads one channel at a time. Turn up the midrange L-pad until the midrange and woofer sounds "blend" so that it is difficult to tell which sound is coming from which speaker. Then do the same with the tweeter L-pad. When the L-pads are properly adjusted you will not be able to hear the "highs" and the "lows" separately; the music will sound all as one. Improper adjustment of the L-pads puts a step in the frequency response to which the ear is very sensitive. It makes the different frequencies sound separated.

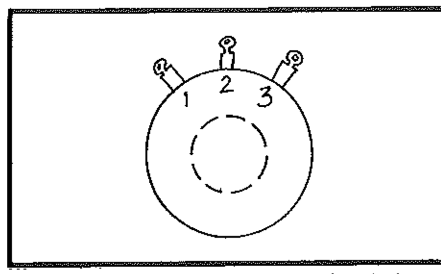

Figure 18. L-pad terminal numbering diagram (rear view).

Coils and Capacitors

Coils for use in crossover networks come in two styles, iron core and air core. The iron core types are generally less costly because they use less copper, the most expensive material. They also are inherently self-shielded and less susceptible to hum pickup.

Air core coils will pick up hum if mounted near a source of AC magnetic fields, such as a power transformer or turntable motor, but this is easily avoided. Crossovers are usually mounted inside speaker enclosures where there are no sources of hum.

Capacitors suitable for use in crossovers are also of two types, solid dielectric and nonpolar electrolytic. Solid dielectric capacitors have two plates separated by a dielectric of paper, mylar, or oil impregnated paper. Nonpolar electrolytic capacitors have two aluminum plates separated by an electrolyte soaked gauze. Passing AC current through the capacitor forms a very thin insulating anodic film on each of the plates. These films are the dielectric in an electrolytic capacitor.

Solid dielectric capacitors provide superior stability and lower power loss compared to electrolytic capacitors, but are much larger and more expensive for a given capacity. Crossovers with a low woofer crossover frequency require large capacitances: usually only electrolytics are practical in such a case.

Nonpolar electrolytics are available in only a few standard sizes and it is often necessary to wire units in parallel to obtain the desired capacitance value. The capacitance of a parallel combination is equal to the sum of the capacitances of the individual units. The voltage rating of the combination is the same as that of the lowest-rated unit.

Unfortunately nonpolar electrolytics are not stocked by all electronics parts houses and may sometimes be difficult to obtain. Speaker builders have tried to "synthesize" a nonpolar electrolytic by wiring two DC (polarized) electrolytics in series back-to-back, but this does not work. DC electrolytics are made with a dielectric film on only one plate, and when the AC audio signal passes through them it begins to form a film on the other plate also. The second film has two bad effects. First, it reduces the capacitance to a low value, changing the crossover frequency. Second, it is very weak and is easily punctured by strong signals, causing pops, clicks and sound level variations in the midrange or tweeter. The unsuspecting speaker builder ends up with mystifying "bugs" in his system that he can never track down.

Whether of the air core or iron core type, the coils must be made with low series resistance to avoid power losses in the coil itself and to avoid changing the effective damping factor of the amplifier. This second consideration requires lower resistances than the first, so it is the determining factor.

The amplifier damping factor is equal to the load impedance (usually 8-ohms) divided by the Thevenin equivalent source impedance of the amplifier. The latter is nearly always .5 ohm or less for transistor amplifiers, resulting in damping factors of 16 or greater. A high damping factor is desirable because it gives the amplifier "tight control" over woofer cone motion at very low frequencies. Interposing any resistance between the amplifier and woofer, either by long speaker wires or by a coil, has the same effect as higher amplifier source resistance and lower damping factor. It changes the woofer bass response and in an extreme case causes a "boomy" peak in the frequency response.

The coil resistance should be no more than a tenth of the woofer impedance in order to avoid these effects.

Power handling capability

The power capability of a crossover is determined by the voltage rating of the capacitors, the power rating of the inductors, and the power rating of the L-pads. Capacitor voltage rating required for a given power level is determined by the formula:

$$V = \sqrt{2RP}$$

R is the system impedance rating in ohms, P is the power level, and V is the capacitor voltage rating. Table 3 gives the power levels obtainable with common impedances and capacitor ratings.

While capacitors must stand off the full speaker system power, inductors and L-pads need handle only the power going to their corresponding tweeter or midrange speakers. This is only a fraction of the system power. How great a fraction of the power it is will depend on the choice of crossover frequencies and the frequency distribution of the energy in the audio signal. Table 4 shows the maximum power which may be expected in a midrange or tweeter section versus crossover frequency.

Table 3.
Effect of capacitor voltage rating on crossover power handling capability.

Voltage Rating	Power Handling Capability 4 ohm	8 ohm	16 ohm
10 V	12 W	6 W	3 W
25 V	78 W	39 W	19 W
50 V	312 W	156 W	78 W
100 V	1250 W	625 W	312 W

Table 4.
Frequency distribution of power in music

Frequency	300Hz	600Hz	1200Hz	2400Hz
Maximum percentage of power above that frequency	50%	25%	10%	5%

For midranges, use the lower crossover frequency to determine the power which the midrange inductor, L-pad, and midrange speaker must handle. For tweeters, use the upper crossover frequency to determine the power which the tweeter and its L-pad must handle. For example, in a three-way system with crossover frequencies of 600 and 2400 Hz, the midrange components will have to be capable of handling 25% of the system power, and the tweeter components 5%. So for a system power rating of 50 watts the midrange inductor, L-pad, and speaker must have ratings of 12.5 watts. The tweeter and its L-pad must have ratings of 2.5 watts.

The woofer and its inductor should be rated to handle the full system power, 50 watts.

Ratings for the speakers, inductors, and L-pads may be obtained from the manufacturer's specifications. Several power rating systems are in use; any may be used so long as it is used consistently. The difficulty arises when one component is specified in DC power and another in "program material" power. The problem is complicated by the fact that none of the speaker power rating methods correspond to any of the amplifier power rating methods.

The most common speaker and speaker component rating method is "watts program material". It equals 2.5 times (approximately) the "watts RMS" or "watts DC". (Watts RMS" and "watts DC" are the same in speaker ratings). A speaker system rated at 50 watts program material may be safely used with an amplifier rated at 50 to 100 watts RMS per channel. This is possible because all those extra amplifier watts are used as reserve power to avoid clipping. A 100 watt RMS per channel amplifier playing music at full output (just on the verge of clipping the music peaks) will actually be putting out only 10 to 20 watts of average (RMS) power.

Thus a 20 watt RMS speaker system may be safely used with a 100 watts RMS amplifier to play music. For reproducing sine waves in the laboratory it's a different story. For sine waves, all the ratings are the same and the amplifier running at full 100 watts capability would burn out the speaker system in a very short time.

We are discussing the maximum power rating of a speaker system, of course. There is also the minimum power required to drive it. Here most speaker makers use the same system and specify required power in terms of minimum amplifier power in watts per channel RMS. The minimum power required is determined simply by the woofer used. Most woofers available require 20 watts or less, and a few 30 watts. Generally the heavier the speaker's magnet, the more efficient it is and the less power it requires.

Mounting

The most convenient method of mounting crossover networks is to glue the capacitors and inductors to a board, then fasten the board inside the speaker enclosure. A suitable glue is silicone rubber "bathtub caulk", such as General Electric type RTV-108. Do not pass metal nails or screws through the coils because currents induced in the metal will change the effective inductance values. For the same reason, mount the crossover at least three inches from any other large metal mass, such as a speaker frame. A metallic enclosure should not be used to house a crossover containing air core coils unless there are generous clearances inside.

The L-pads may be mounted on the same board as the crossover, and the board mounted over a recess in the back of the speaker enclosure so that the control knobs are accessible at the back. Alternatively the L-pads may be mounted where convenient and connected to the rest of the crossover by wires.

References

1. Richard H. Small, "Constant-Voltage Crossover Network Design," JOURNAL OF THE AUDIO ENGINEERING SOCIETY, January, 1971. Reprinted in Ref. 3, pp. 172-179.

2. J. Robert Ashley and Allan L. Kaminsky, "Active and Passive Filters as Loudspeaker Crossover Networks," JOURNAL OF THE AUDIO ENGINEERING SOCIETY, June 1971. Reprinted in Ref. 3, pp. 205-212.

3. "Loudspeakers, An anthology of articles on loudspeakers from the pages of the Journal of the Audio Engineering Society," Vol. 1-Vol. 25 (1973-1977). Audio Engineering Society, 60 East 42nd Street, New York, New York 10017. 1978

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Chapter Five: Using Theile-Small Parameters.

Introduction

The purpose of this chapter is to show you how to design loudspeaker systems using Thiele-Small parameters. There are a number of different types of designs with which you can do this:

Type	**Theoretical name**
Vented box	4th order
Vented box with simple resistor-capacitor equalizer	5th order
Vented box with active equalizer	6th order
Sealed box	2nd order

A given woofer will usually work in several of the design types listed. Other types are also possible; this paper covers these four. Within each type, or "order," there are a number of combinations of parameters called "alignments". Usually a woofer fits directly into only one alignment in each order, determined by one of the woofer parameters, Qts. We will take the approach that you have a woofer with known parameters, and want to see what performance it will give in different "orders". We have included a lot of examples to illustrate the design methods.

Theory

Australian scientist Neville Thiele first worked out a comprehensive theory that relates the bass performance of a loudspeaker system to the parameters of the driver and enclosure (Ref. 1). He applied his theory primarily to vented speaker systems. Another Australian, Richard Small, took Thiele's theory a few steps further. He applied it to sealed systems (Ref. 2), and also extended vented-box theory to take into account losses in the box (Ref. 3).

Thiele and Small based their analysis on certain parameters of the driver and box. These parameters, as they defined them, have come to be called "Thiele-Small parameters." The most important driver parameters are:

f_s Natural resonant frequency of the woofer (measured on a flat baffle)

C_{as} Acoustic compliance of the woofer

V_{as} Volume of air having same acoustic compliance as the driver suspension

Q_{ms} Mechanical Q of the woofer due to mechanical loses

Q_{es} Electrical Q of the woofer as controlled by the electrical damping, which depends on the Bl product and voice coil resistance (Re).

Q_{ts} Total Q of the woofer taking into account both mechanical losses and electrical damping; related to Qms and Qes by the formula:

$$Q_{ts} = \frac{1}{1/Q_{es} + 1/Q_{ms}}$$

The most-used box parameters are:

V_{ab} Air volume of the box

C_{ab} Acoustic compliance of the air volume of the box

f_b Helmholtz resonant frequency of a vented box, that is, the frequency at which the mass of air in the vent resonates with the compliance of the air volume in the box

Q_b Q of the vented-box Helmoltz resonance; taken here as equal to Small's QL (ref. 4, p. 306). Small's QL (ref. 4, p. 306).

Q_{tc} System Q of a sealed box (second-order) system.

Figure 1 illustrates some of these definitions; for their complete derivations, see References 1-3.

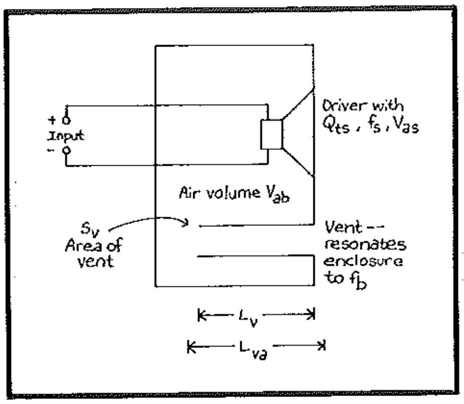

Figure 1. Vented box loudspeaker system. LV is the physical length of the vent. Lva, the equivalent acoustical length, is a little longer because of end corrections.

The terms "fourth order," "second order" and so on, derive from the mathematical equations that describe the system's behavior. The "order" of a speaker system is equal to the number of "energy storages" it has. A sealed box system, for example, has two; the kinetic energy in the moving mass of the cone, and the potential energy in the compressed spring of the cone suspension plus air compression in the box. Thus it is a second-order system.

A vented speaker is a fourth-order system because it has two more energy storages; the potential energy storage in the compressed air in the box is separate from that of the cone suspension spring (the cone can be displaced whether or not the air is compressed and vice-versa), and the vent air mass is the fourth kinetic energy storage.

Adding a simple resistor-capacitor equalizer to the vented box system adds another storage, charge on the capacitor, and makes a fifth-order total system.

Thiele's theory is a mathematical theory, and to really understand its derivation you have to be fairly skilled in mathematics. Fortunately, you do not have to fully understand it to apply it--part of Thiele's genius.

This chapter extends Thiele's approach to take account of box losses in vented systems.

Alas, a number of half-baked conclusions have grown up around Thiele's theory, mixed with old time bass-reflex "lore", to add to the confusion. We will try to clear up some of that, too.

The heart of Thiele's theory is his Table 1 (p. 187, Ref. 4) and gives a number of alignments, or combinations of parameters, which may be used to design speaker systems with flat, extended bass response.

Probably Thiele's most famous alignment is the No. 5, or fourth-order Butterworth (B4) alignment for a vented box. First, you have to start with a woofer that has Qts equalling .383. Let's say you have a 10-inch woofer with Qts equalling .39. The calculations are outlined in Example 1.

EXAMPLE 1. A fourth-order design assuming Q_b = infinity. (Thiele assumed this to simplify his analysis.) Given a woofer with parameters:

Q_{ts} = .39
V_{as} = 7.47 cu. ft.
f_s = 22.8 Hz

From Alignment 5 in Table 1 we have,

V_{ab}/V_{as} = .707
f_b/f_s = 1.000
f_3/f_s = 1.000

Therefore the enclosure parameters and f_3 are,

V_{ab} = .707 x 7.47 = 5.28 cu. ft.
f_b = 1.000 x 22.8 = 22.8 Hz
f_3 = 1.000 x 22.8 = 22.8 Hz

Going through the procedure step-by-step: First we need to know what box volume to use. From Table 1 we have

$$V_{ab}/V_{as} = .707$$

so we may calculate

$$V_{ab} = (V_{ab}/V_{as})\,V_{as} = .707 \times 7.47 = 5.28$$

So our box needs to have volume equal to 5.28 cu. ft.

We also need to know what frequency, fb, to tune the box to. Again referring to the table, we have $f_b/f_s = 1.000$

So we may calculate

$$f_b = (f_b/f_s)f_s = 1.000 \times 22.8 = 22.8\text{Hz}$$

So the box should be tuned to

$$f_b = 22.8\text{ Hz}$$

And as the final step, we have from the table $f_3/f_s = 1.000$

so that

$$f_3 = (f_3/f_s)f_s = 1.000 \times 22.8\text{ Hz}$$

No.	Type	k	Ripple (dB)	Q_{ts}	V_{ab}/V_{as}	f_b/f_s	f_3/f_s	Auxiliary filter f_{aux}/f_s	Q_{aux}	Peak height (dB)
				Fourth-order alignments						
9	C4	.4	.49	.557	2.062	.716	.600			
8	C4	.48	.20	.518	1.789	.757	.641			
7	C4	.6	.07	.466	1.372	.829	.729			
6	C4	.8	-	.415	.948	.927	.867			
5	B4	1.0	-	.383	.707	1.000	1.000			
4	QB3	-	-	.303	.339	1.23	1.45			
3	QB3	-	-	.259	.224	1.42	1.77			
2	QB3	-	-	.209	.134	1.73	2.28			
1	QB3	-	-	.180	.095	2.00	2.68			
				Fifth-order alignments						
14	C5	.278	1.00	1.102	5.236	.781	.685	2.47		
13	C5	.355	.43	.924	4.405	.798	.704	2.07		
12	C5	.4	.25	.810	3.663	.814	.724	1.81		
11	C5	.7	-	.545	1,715	.912	.852	1.22		
10	B5	1.0	-	.477	1.000	1.000	1.000	1.00		
				Sixth-order alignments						
19	C6	.414	.10	.399	.800	.841	.554	.58	1.93	6.0
18	C6	.5	.02	.371	.662	.888	.620	.64	2.38	7.7
17	C6	.6	-	.348	.552	.931	.698	.71	3.15	10.1
16	C6	.8	-	.317	.429	.979	.850	.86	3.78	11.6
15	B5	1.0	-	.299	.366	1.000	1.000	1.00	4.52	13.2

TABLE 1. Design data from Thiele's "summary of loudspeaker alignments." The notation here, which follow's Small's, is slightly different from Thiele's. Qts here corresponds to his Qt; Vab/Vas here is equal to the reciprocal of his Cas/Cab. The two left-hand columns give the numbers and type-names Thiele used to label the alignments. Some of the Chebyshev alignments give some frequency-response ripple in the passband which is specified in dB (peak-to-peak) in the fourth column. "k" is a mathematical constant used in generating the alignments, related to the ripple obtained.

So our cutoff (3dB down) frequency will be 22.8 Hz too.

Thus Alignment 5 gives one combination of box volume (5.28 cu. ft.) and tuning frequency (22.8 Hz) that will give flat response, namely down to 22.8 Hz. As you can see, the B4 design is simple because

$$f_3 = f_b = f_s$$

The next example, Example 2, uses the Speakerlab W1208R, a 12-inch woofer. Its Qts value of .408 comes close to Alignment 6 in Table 1, which calls for Qts equalling .415. (In acoustics, anything within two percent of anything else is dang close.)

EXAMPLE 2. A fourth-order design assuming Q_b = infinity. Given the woofer parameters,

$$Q_{ts} = .408$$
$$V_{as} = 16.3 \text{ cu. ft.}$$
$$f_s = 18.3 \text{ Hz}$$

From Alignment 6 in Table 1 we have,

$$V_{ab}/V_{as} = .948$$
$$f_b/f_s = .927$$
$$f_3/f_s = .867$$

Therefore the enclosure parameters and f_3 are,

$$V_{ab} = .948 \times 16.3 = 15.5 \text{ cu. ft.}$$
$$f_b = .927 \times 18.3 = 17.0 \text{ Hz}$$
$$f_3 = .867 \times 18.3 = 15.9 \text{ Hz}$$

So the W1208R can be used in a 15.6 cubic foot box to get response down to 16 Hz. Both examples provide superb bass response in large boxes--probably not practical designs for most people. Nevertheless, you can see how easy it is to design speaker systems using Thiele's Table 1, and why Thiele is justly famous. Before Thiele's work, speakers were designed on a trail-and-error basis with a lot of tinkering with box sizes and box tuning.

The alignments listed in the Table are not the only ones possible; there are an infinite number of other alignments "hidden between the lines," and you can get them by interpolation. If you have a woofer whose Q lies between .466 and .518 (Alignments 7 and 8), for example, you can find values for

$$V_{ab}/V_{as}\ ,\quad f_b/f_s\ ,\quad f_3/f_s$$

by taking values between those given for the two alignments. More on interpolation later.

Thiele's Alignments 1 through 4, which he calls "Quasi-Third Order" alignments, are really more fourth-order (vented box) alignments for woofers with Qts values below .383. "Quasi" Butterworth means that he used some mathematical trickery to extend the table so you could use it for a wider range of woofers. Another way of doing the same thing that he suggested (p. 189, Ref. 4) is via "sub-Chebyshev" responses. The terms "Butterworth", "Chebyshev", "quasi-Butterworth", "sub-Chebyshev"-- abbreviated B, C , QB, and SC--all refer to mathematical techniques for generating the alignments that we won't go into here. Small covers it in Ref. 4, p. 325 for any of you math wizards that are interested in the arcane details.

Table 1 suffers from a major limitation for practical designers--it assumes zero box losses (infinite box Q, Qb) for the vented system. Actual boxes do have losses, and those losses make Qb finite. Typical well-made boxes have box Q around 5 to 20. Small gives 7 as good typical value for Qb.

Fourth-order system alignments for box Q equal to 7

Small solved the math to take account of box losses (Ref 3). A box designed to Table 1 will have some sag in the bass due to the box losses. Figure 2 shows that sag for the system of Example 1. Small provided the mathematical tools to "rewrite" Table 1 to take account of the box losses. However, he provided his results in the form of graphs, which are not as convenient to use as a table like Thiele's.

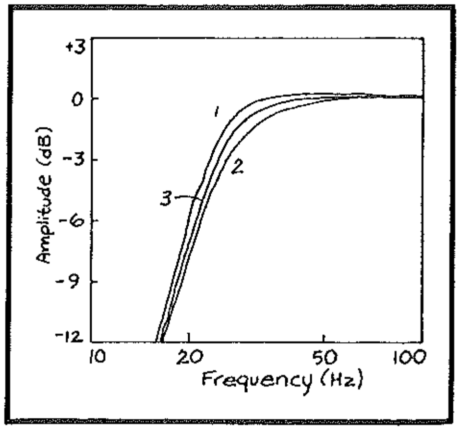

Figure 2. Frequency response of the Example 1 loudspeaker system design. Curve 1--response assuming (as Thiele did) that Qb equals infinity. The slight peak is due to the fact that the woofer's Qts, .39, does not quite equal that called for by the alignment, .383. Curve 2--response with same woofer and enclosure, assuming that the box has Q equal to 7, as most do. The response is poorer. Curve 3--same woofer in the Example 3 system.

(The frequency response data for these curves, and the other graphs in this paper, were calculated using a computer program.)

We have recalculated the data, this time into "alignment table" form, shown as Table 2. As well as assuming box Q equal to 7 rather than infinity, Table 2 differs from Table 1 in two other ways:

1) Rather than quasi-Butterworth (QB3) responses for the low Qts values, we calculated sub-Chebyshev (SC4) responses because they were easier. They provide a slightly different response shape and also work fine.

2) Table 2 also has four extra columns that give additional useful information for advanced speaker design work. The last column lists what we call the "alignment factor", Ka, which is useful for comparing alignments and choosing or designing a woofer to hit a given box volume and efficiency target. The other three columns provide data that is valuable for "forcing" a woofer into an alignment that it doesn't really fit because it has the wrong Qts value. Later on in this chapter, we will tell you how to use these columns.

In general, box losses mean that to get truly flat response for a given woofer, you

k	Ripple (dB)	Q_{ts}	V_{ab}/V_{as}	f_b/f_s	f_3/f_s	$V_{ab}/(V_{as}Q_{ts}^2)$	$f_b/(f_s/Q_{ts})$	$f_3/(f_s/Q_{ts})$	K_a
.25	2.26412	.894044	8.88417	.535773	.464929	11.1147	.479004	.415667	.927638
.3	1.46383	.76605	6.217	.580319	.499555	10.5942	.444553	.382684	.840484
.35	.934504	.681993	4.86258	.621722	.534629	10.4546	.42401	.364613	.797263
.4	.586276	.623071	3.97991	.661364	.570459	10.2517	.412077	.355437	.772137
.45	.359654	.579697	3.33743	.699598	.606931	9.9314	.405555	.351836	.75627
.5	.214477	.546488	2.84357	.736397	.643816	9.52143	.402432	.351838	.745722
.6	.067859	.49883	2.1371	.805066	.717922	8.58857	.401591	.358121	.733393
.7	.016847	.465808	1.66705	.866144	.79132	7.68305	.403457	.368603	.72734
.8	.002648	.441012	1.34219	.918939	.863142	6.901	.405263	.380656	.72472
.9	.000133	.421251	1.111	.963409	.932954	6.26081	.405837	.393008	.724344
1	0	.404814	.942225	1	1.00059	5.74967	.404814	.405055	.725651
1.1	0	.390727	.816171	1.02946	1.06605	5.34608	.402237	.416533	.72833
1.2	0	.378402	.720066	1.05267	1.12938	5.0288	.398334	.427359	.732173
1.5	0	.348853	.539198	1.09375	1.3074	4.43062	.381558	.456091	.749099
1.7	0	.333598	.469178	1.10495	1.4167	4.21589	.36861	.472608	.76348
2	0	.31542	.402065	1.10841	1.56698	4.04126	.349616	.494256	.787271
2.5	0	.293918	.340676	1.09848	1.78118	3.94356	.322862	.52352	.82711
3	0	.279571	.308129	1.08364	1.95232	3.94229	.302955	.545813	.862237
3.5	0	.269676	.288831	1.06977	2.08639	3.97153	.288491	.562649	.891026
4	0	.262639	.276451	1.05816	2.19077	4.00775	.277913	.57538	.913948
5	0	.253629	.262051	1.04127	2.33643	4.07367	.264097	.592587	.946413

TABLE 2. Fourth-order loudspeaker system alignments based on enclosure Q (Qb) equal to 7.

need a little larger box and higher fb tuning. So systems designed to Table 2 turn out to have a little larger Vab and higher f3 than Table 1.

In Example 3, Table 2 is used to design a system for the same woofer used in Example 1 with Thiele's table.

Comparing Example 3 to Example 1, you will see that Table 2 makes the box a little bigger and fb and f3 are a little higher. The frequency response shape is also a little better, as curve 3 in Fig. 2 shows. Example 4 on the next page gives a practical design for a real woofer.

EXAMPLE 3. A fourth-order design assuming Q_b = 7. Given the woofer parameters,

Q_{ts} = .39
V_{as} = 7.47 cu. ft.
f_s = 22.8 Hz

From the k = 1.1 alignment in Table 2 we have,

V_{ab}/V_{as} = .816
f_b/f_s = 1.029
f_3/f_s = 1.066

Therefore,

V_{ab} = .816 x 7.47 = 6.10 cu. ft.
f_b = 1.029 x 22.8 = 23.5 Hz
f_3 = 1.066 x 22.8 = 24.3 Hz

EXAMPLE 4. A fourth-order design assuming Q_b = 7. The Speakerlab W808R, an 8-inch woofer, has parameters:

Q_{ts} = .354
V_{as} = 4.24 cu. ft.
f_s = 24.2 Hz

From the k = 1.5 alignment in Table 2 we have,

V_{ab}/V_{as} = .539
f_b/f_s = 1.094
f_3/f_s = 1.307

Therefore,

V_{ab} = .539 x 4.24 = 2.29 cu. ft.
f_b = 1.094 x 24.2 = 26.5 Hz
f_3 = 1.307 x 24.2 = 31.6 Hz

Fifth Order Alignments

Many people find this an attractive alignment because it calls for a fairly compliant woofer in a vented box. The woofer suffers from a great deal of "cone bounce" in response to turntable rumble and other forms of subsonic noise.

In his paper, Thiele had assumed that the designer would eliminate subsonic energy, if necessary, by filtering out out-of-band signals elsewhere in the signal chain. Alas, such filtering does not appear in conventional stereo equipment. The rumble filters, subsonic filters, and low filters in stereo receivers and preamplifiers are notoriously imprecise things with considerable effect on the low and midbass--up to as high as 200Hz, believe it or not--as well as on the subsonic (below 20 Hz) energy.

Thiele suggested using a fifth-order alignment, such as Table 1's Alignment 10, rather than a fourth-order one. A fifth-order system is a vented speaker system with a simple resistor-capacitor auxiliary filter (Fig. 3), usually called an equalizer, ahead of the power amplifier. Since the filter is passive, it introduces no noise, hum or distortion into the system. It eliminates subsonic noise without disturbing the bass response, since the design of the speaker has taken it into account; fifth-order alignments make the filter an integral part of the system, rather than a tacked-on "extra" with uncertain effect on the frequency response.

The auxiliary filter also saves amplifier power and allows use of a high-compliance, low-Qts woofer without subjecting the woofer to excessive cone excursion and possible suspension damage.

Unfortunately, Small did not apply his analysis taking account of finite Qb to the fifth or sixth-order cases. We have done so, and the resulting fifth-order alignments appear in Table 3.

Thiele's fifth-order alignments are in Table 1, Alignments 10 through 14. Our Table 3 differs from Table 1 in that:

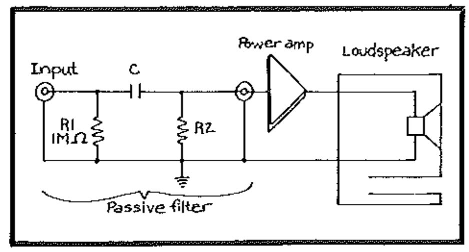

Figure 3. Fifth-order loudspeaker system. The speaker and enclosure are similar to the fourth-order system. The new element is the passive resistor-capacitor 6dB octave high-pass filter, generally placed ahead of the power amplifier. The filter's cutoff frequency will be

$$f_{aux} = \frac{1}{2\pi(R2)(C)}$$

We recommend a value of 51 Kohms for R2, for compatibility with standard impedance levels at the interface between preamp and amplifier, or in a tape monitor loop. Then the capacitor needed will be

$$C = \frac{1}{2\pi f_{aux}(51000)}$$

The 1 megohm resistor, R1, eliminates switching transients.

1) Again, Table 3 assumes Qb equals 7.

2) Sub-Chebyshev (SC5) responses are included to cover Qts values down to .26.

Comparing the values of Vab/Vas and f3/fs in our fifth-order alignments to those for the same Qts values in the fourth-order alignments in Table 2, you will see that the fifth-order systems take larger boxes and have higher cutoff frequencies. This is the cost of the subsonic filtering action of the equalizer, which, being a gentle 6dB/octave filter, has some effect on the high bass.

The most important parameter of the filter in designing the system is faux, its cutoff frequency. The filter may be inserted between preamplifier and power amplifier, or in the tape monitor loop of a preamplifier, integrated amplifier or stereo receiver. If the system includes a tape deck, the auxiliary filter should go after the tape deck in the loop--between the deck's LINE OUT jacks and the receiver's TAPE IN or TAPE MON jacks.

Example 5 shows a fifth-order design using the same woofer as Example 3. Comparing the two examples, you will see that the fifth-order system takes a 12%

k	Ripple (dB)	Q_{ts}	V_{ab}/V_{as}	f_b/f_s	f_{aux}/f_s	f_3/f_s	$V_{ab}/(V_{as}Q_{ts}^2)$	$f_b/(f_s/Q_{ts})$	$f_{aux}/(f_s/Q_{ts})$	$f_3/(f_s/Q_{ts})$	K_a
.25	1.35356	1.48698	40.2284	.673029	2.52728	.637451	18.1938	1.00078	3.758	.947875	2.49302
.3	.786891	1.2192	17.5466	.709679	2.18898	.66063	11.8044	.865239	2.6688	.805438	1.83392
.35	.449613	1.04311	11.682	.739093	1.9411	.682071	10.7364	.770955	2.02478	.711475	1.56956
.4	.251183	.919124	8.71077	.765092	1.75418	.70347	10.3112	.703214	1.6123	.646576	1.40731
.45	.136404	.827551	6.83614	.789266	1.60939	.725422	9.98208	.653158	1.33186	.600324	1.29259
.5	.071489	.757452	5.5307	.812323	1.49462	.748108	9.63984	.615295	1.1321	.566656	1.20599
.6	.016965	.657776	3.8445	.856115	1.32536	.795629	8.88555	.563132	.871789	.523345	1.08397
.7	.002973	.590694	2.83151	.89711	1.20727	.845412	8.11509	.529917	.713129	.49938	1.00353
.8	.000294	.54256	2.17915	.935007	1.12033	.89661	7.4027	.507298	.607847	.486465	.948088
1	0	.477735	1.42659	1	1	1.00048	6.25067	.477735	.477735	.477962	.880445
1.2	0	.435305	1.03141	1.04962	.918928	1.10334	5.44306	.456906	.400014	.480289	.844858
1.5	0	.392139	.721099	1.09775	.834275	1.25206	4.68938	.430469	.327152	.490983	.82181
1.7	0	.371074	.603712	1.11586	.792115	1.34661	4.3844	.414067	.293933	.499692	.817847
2	0	.346494	.492733	1.12858	.742231	1.48102	4.10411	.391047	.257179	.513164	.821605
2.5	0	.317445	.392711	1.12771	.682387	1.68421	3.89706	.357987	.21662	.534644	.841351
3	0	.297657	.34038	1.11546	.641141	1.86003	3.84177	.332024	.19084	.553651	.867121
3.5	0	.283654	.309628	1.10068	.611747	2.00854	3.84825	.312211	.173524	.56973	.892806
4	0	.273455	.290027	1.08669	.590246	2.13191	3.87852	.297161	.161406	.582981	.91596
5	0	.260044	.267367	1.06419	.561863	2.31696	3.95378	.276737	.146109	.602513	.952731

TABLE 3. Fifth-order loudspeaker system alignments based on enclosure Q (Qb) equal to 7.

smaller box and gives 17% (.23 octave) higher cutoff frequency.

As long as you are providing a filter anyway, you may as well make it a more effective 12dB/octave one and go to sixth-order system alignments which provide greatly increased performance.

EXAMPLE 5. A fifth-order design assuming Q_b = 7. Given the woofer parameters,

Q_{ts} = .39
V_{as} = 7.47 cu. ft.
f_s = 22.8 Hz

From the k = 1.5 alignment in Table 3 we have,

V_{ab}/V_{as} = .721
f_b/f_s = 1.098
f_{aux}/f_s = .834
f_3/f_s = 1.252

Therefore,

V_{ab} = .721 x 7.47 = 5.39 cu. ft.
f_b = 1.098 x 22.8 = 25.0 Hz
f_{aux} = .834 x 22.8 = 19.0 Hz
f_3 = 1.252 x 22.8 = 28.5 Hz

Sixth-order system alignments

Sixth-order systems also use an auxiliary filter, in this case a 12dB/ocatve filter as in Figure 4.

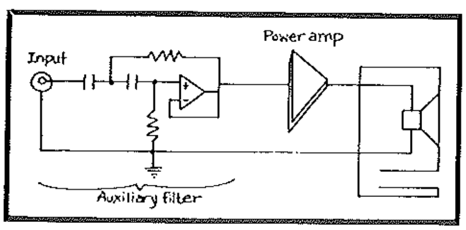

FIGURE 4. Sixth-order loudspeaker system. The auxiliary filter is a 12dB/octave, second-order, high-pass filter. It must be an active filter to get its Q, Qaux above .5. Ref. 7 describes a practical design for the filter.

The filter is second order; with the vented speaker itself, which is fourth order, it raises the order of the total system to six. Our sixth-order alignments appear in Table 4 on the next page. Note that:

1) Again, we use sub-Chebyshev alignments to cover Qts values below .312.

2) Thiele calculated three types of sixth-order alignments, which he called Class I, Class II, and Class III. (We included only the Class I alignments in Table 1.) A sixth-order system can be analyzed as three component second-order filters in cascade. Each of the component second-order filters has a different Q; for the Butterworth case, the three Q's are 2.0, .707, and .5. You choose one of the imaginary component sections to make real in your auxiliary filter. The other two "recombine" (mathematically) into the vented speaker's fourth-order equivalent filter.

If you make your auxiliary filter the high-Q one, your system is Class I; the mid-Q one, Class II; the low-Q one, Class III. We computed alignments only for Class I systems.

3) Thiele originally specified the filter's peaking in terms of a parameter he calls Yaux. We have used Qaux throughout this paper since Q is more familiar to most people. Yaux and Qaux are related by the formula:

$$Q_{aux} = \frac{1}{\sqrt{Y_{aux} + 2}}$$

A design using Table 4 appears in Example 6. Comparing it to Example 3, the fourth-order system using the same woofer, you will see that the box is a little bigger (6.41 versus 6.10 cu.ft.) but the cutoff frequency is nearly an octave lower (14.0 versus 24.3 Hz.) Thus the sixth-order design gives a tremendous improvement in bass response compared to the others.

This improvement is obtained at the cost of first, more complexity; you have to provide an active auxiliary filter. And second, it draws more power from the amplifier. The auxiliary filter provides bass boost which the amplifier has to supply as additional output power.

The filter goes between your amplifier and preamplifier or in the tape monitor loop. Reference 7 describes a simple filter you can build.

EXAMPLE 6. A sixth-order design assuming Q_b = 7. Given the woofer parameters,

Q_{ts} = .39
f_s = 22.8 Hz
V_{as} = 7.47 cu. ft.

From the k = .5 alignment in Table 6 we have,

V_{ab}/V_{as} = .858
f_b/f_s = .876
f_{aux}/f_s = 1.386
Q_{aux} = 3.77
f_3/f_s = .616

Therefore,

V_{ab} = .858 x 7.47 = 6.41 cu. ft.
f_b = .876 x 22.8 = 20.0 Hz
f_{aux} = 1.386 x 22.8 = 31.6 Hz
Q_{aux} = 3.77
f_3 = .616 x 22.8 = 14.0 Hz

Example 6 calls for Qaux equals 3.77, which demands 11.6 dB of bass boost. This is impractical for most systems, which would not have that much reserve amplifier power. Generally, Qaux around 2 is desirable. It corresponds to a maximum boost of 6 dB, which most amplifiers can handle. The elimination of wasted low frequency energy in the subsonic region makes available more amplifier power for use as boost. Example 7 shows a design using a woofer with Qts equalling .312, which takes Qaux equal to 1.93 and 6.0 dB of peak boost.

Second-order system alignments

Most of the attention paid Thiele's and Small's work has centered on the vented-box alignments. However, their results also show that the second-order (sealed enclosure) alignments have some very useful properties and can provide quite respectable performance. The theory for second-order systems, though simpler, was actually fully developed by

k	Ripple (dB)	Q_{ts}	V_{ab}/V_{as}	f_b/f_s	f_{aux}/f_s	Q_{aux}	Filter peak height (dB)	f_3/f_s	$V_{ab}/(V_{as}Q_{ts}^2)$	$f_b/(f_s/Q_{ts})$	$f_{aux}/(f_s/Q_{ts})$	$f_3/(f_s/Q_{ts})$	K_a
.27	.626827	.531489	1.72085	.696606	.44708	6.92927	16.8364	.431682	6.09192	.370238	.237618	.229434	.419028
.3	.423505	.499197	1.54138	.725809	.47085	6.24015	15.9319	.455253	6.18538	.362322	.235047	.227261	.41717
.35	.216443	.45944	1.30002	.770769	.511086	5.3549	14.6131	.495199	6.15874	.354122	.234813	.227514	.417034
.4	.107644	.430758	1.11447	.810776	.55161	4.69178	13.4764	.535512	6.00624	.349248	.23761	.230676	.419311
.5	.023830	.391663	.858442	.875715	.631992	3.7654	11.5935	.615893	5.59612	.342985	.247528	.241222	.428265
.6	.004241	.365705	.69901	.922453	.710241	3.14998	10.077	.694981	5.22664	.337345	.259739	.254158	.441073
.7	.000525	.346892	.595355	.954747	.785956	2.71224	8.81672	.772622	4.94753	.331194	.272642	.268016	.456692
.8	.000033	.332543	.525085	.976455	.859293	2.38552	7.74677	.849137	4.74827	.324713	.285752	.282374	.47461
.9	0	.321253	.475628	.990753	.930551	2.1328	6.8245	.92494	4.60864	.318282	.298942	.29714	.494483
1	0	.312187	.439667	1	1	1.93185	6.02059	1.0004	4.51122	.312187	.312187	.312311	.516042
1.5	0	.285764	.352892	1.01315	1.32439	1.34074	3.19724	1.3795	4.32142	.289521	.378464	.394212	.642103
2	0	.274057	.322139	1.01163	1.6112	1.05854	1.59057	1.76547	4.28905	.277245	.441561	.48384	.786121
3	0	.26453	.300112	1.0069	2.06514	.798057	.205426	2.52397	4.28876	.266355	.546291	.667667	1.08477

TABLE 4. Sixth-order loudspeaker system alignments based on enclosure Q (Qb) equal to 7.

EXAMPLE 7. A sixth-order design assuming Q_b = 7. Given the woofer parameters,

Q_{ts} = .312
V_{as} = 11.67 cu. ft.
f_s = 18.2 Hz

From the k = 1.0 alignment in Table 4 we have,

V_{ab}/V_{as} = .440
f_b/f_s = 1.000
f_{aux}/f_s = 1.000
Q_{aux} = 1.93
f_3/f_s = 1.000

Therefore,

V_{ab} = .440 x 11.67 = 5.13 cu. ft.
f_b = 1.000 x 18.2 = 18.2 Hz
f_{aux} = 1.000 x 18.2 = 18.2 Hz
Q_{aux} = 1.93
f_3 = 1.000 x 18.2 = 18.2 Hz

Small (Ref. 2) after Thiele developed the theory for vented systems.

The formulas for sealed-box systems don't work quite the same way as the others, since a given woofer with a given Qts will work in a range of box volumes. (for the fourth-order and higher designs, you will recall, each alignment called for one specific Qts value.) Table 5 presents the second-order design formulas in an "alignment table" similar to the others. Note that Vab/Vas and fb/fs in the table are not constants, but formulas involving Qts. This is because in second-order systems, as we said, each alignment will fit many woofers and vice-versa. The formulas using Qts "adjust" the alignment to fit the Qts value of whatever woofer you are using. Also observe that this is not "alignment jamming;" second-order alignments are inherently able to accept a wide range of Qts values.

Unlike vented systems, which have two Q's, Qts and Qb, the second-order system has only a single system Q, Qtc. Woofer damping and box losses combine in this one Q value.

There is one limitation in fitting different woofers into different second-order alignments; Qtc must be greater than Qts. If you try to use an alignment for Qtc less than your woofer's Qts, the elf in your calculator will call you on it by giving you a negative value for Vab.

The best known alignment in the table is the Qtc equals .707 one, which is the Butterworth (B2) alignment. It gives "maximally flat" (maximum Q with zero peaking) frequency response--see Fig. 5.

EXAMPLE 8. A second-order design with Q_{tc} = .707. Given the woofer parameters,

Q_{ts} = .39
V_{as} = 7.47 cu. ft.
f_s = 22.8 Hz

From the Q_{tc} = 1.0 alignment in Table 5 we have,

V_{ab}/V_{as} = $1/(.5/Q_{ts}^2 - 1)$
f_3/f_s = $.707/Q_{ts}$

Therefore,

V_{ab}/V_{as} = $1/(.5/.39^2 - 1)$ = .437
f_3/f_s = .707/.39 = 1.81

and

V_{ab} = .437 x 7.47 = 3.26 cu. ft.
f_3 = 1.81 x 22.8 = 41.3 Hz

EXAMPLE 9. A second-order design with Q_{tc} = 1.0. Given the woofer parameters,

Q_{ts} = .39
V_{as} = 7.47 cu. ft.
f_s = 22.8 Hz

From the Q_{tc} = 1.0 alignment in Table 5 we have,

V_{ab}/V_{as} = $1/(1.0/Q_{ts}^2 - 1)$
f_3/f_s = $.786/Q_{ts}$

Therefore,

V_{ab}/V_{as} = $1/(1.0/.39^2 - 1)$ = .179
f_3/f_s = .786/.39 = 2.02

and

V_{ab} = .179 x 7.47 = 1.34 cu. ft.
f_3 = 2.02 x 22.8 = 46.0 Hz

Example 8 applies the Qts equals .707 alignment to our exemplary woofer. This alignment is not the most efficient one (in terms of best f3 for a given box volume and efficiency), however. The most efficient alignment is the Chebyshev (C2) alignment with Qtc equal to 1.1, which gives a peak of 1.83 dB (Fig.5). The Qtc equal to 1.0 Chebyshev alignment is very nearly as efficient, and has a peak of only 1.25 dB. Example 9 applies this alignment.

Let's compare the Qtc equals 1.0 alignment to the fourth-order system of Example 3 and the B2 alignment we just calculated in Example 8 (all using the same woofer):

	4th order Ex. 3	**B2 Ex. 8**	**C2(Qts 1) Ex. 9**
Vab(cu.ft.)	6.10	3.26	1.34
f3 (Hz.)	24.3	41.3	46.2

The shape of the low-end frequency response curve for a sealed-box system depends only on Qtc which in turn depends on Qts, Vas, and Vab. This graph shows the response curve for a woofer in various box volumes. To apply it to a given woofer: 1) calculate .5fs/Qts, fs/Qts etc., and write the answers in the spaces provided along the frequency axis at the lower edge of the graph. 2) Using the formula, calculate Vab for Qtc equals .7, Qtc equals .8, etc., and write the answers in the Vab column.

Thus for a given woofer, the Qtc equal to 1.0 sealed-box alignment dramatically reduces the box volume compared to the B2 alignment, at some cost in bass response. This alignment may be used effectively with high-Vas (high-compliance) woofers which would otherwise demand impractically large box volumes.

The sealed system's more gentle rolloff below f3 (12 dB/octave) compared to fourth, fifth, and sixth-order systems (24, 30, and 36 dB/octave) gives better response to the extreme lows than simple comparison of f3 values would suggest.

The sealed-box system has three other factors in its favor:

1) The enclosure can be partially filled with damping material, which slows down the speed of sound and can increase the "effective" volume. For example, when you use the Speakerlab W1208R 12-inch woofer in a 2.2 cu.ft. box, filling the box one-quarter to one-half full with Dacron will effectively increase the volume by about 12% and increase fs/Qts by about 6%. Vented systems cannot be

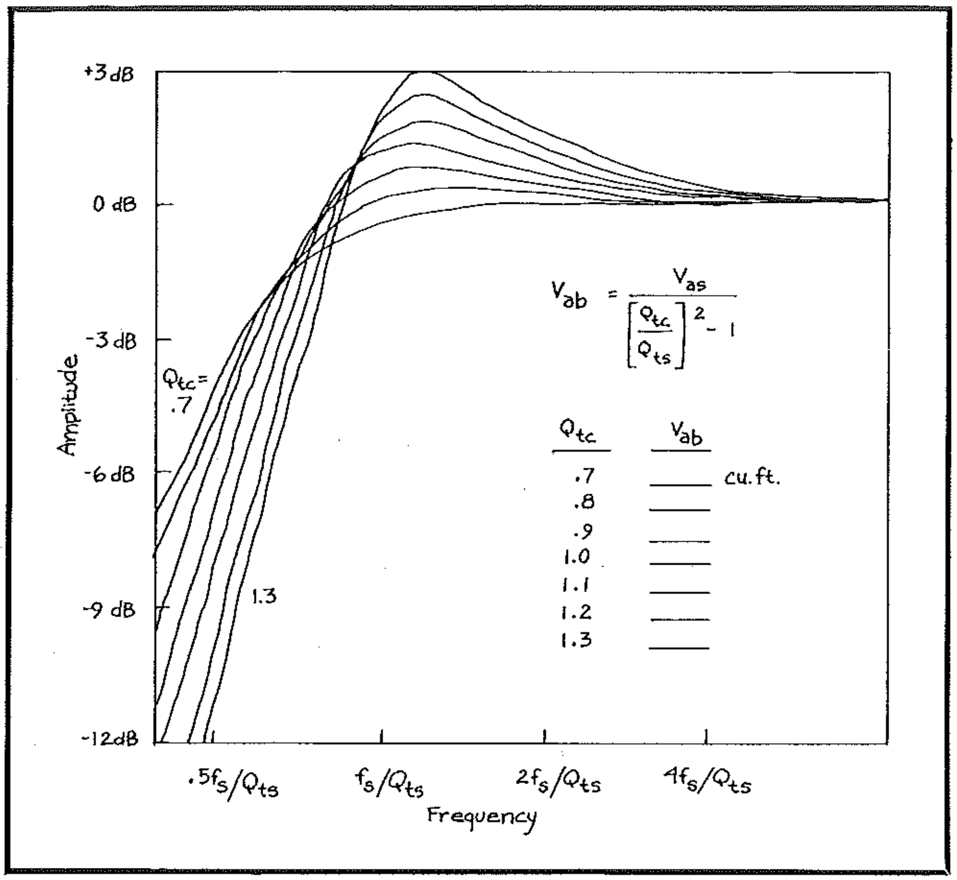

Figure 5. The shape of the low-end frequency response curve for a sealed-box system depends only on Qtc which in turn depends on Qts, Vas and Vab. This graph shows the response curve for a woofer in various box volumes. To apply it to a given woofer: 1) calculate .5fs/Qts, fs/Qts etc., and write the answers in the spaces provided along the frequency axis at the lower edge of the graph. 2) Using the formula, calculate Vab for Qtc equal to .7, Qtc equal to .8 etc., and write the answers in the Vab column.

Q_{tc}	V_{ab}/V_{as}	f_3/f_s	K_a	Peak height (dB)
.5	$1/(.25/Q_{ts}^2 - 1)$	$.777/Q_{ts}$	1.311	
.6	$1/(.36/Q_{ts}^2 - 1)$	$.725/Q_{ts}$	1.084	
.7	$1/(.49/Q_{ts}^2 - 1)$	$.707/Q_{ts}$	.953	
.707	$1/(.50/Q_{ts}^2 - 1)$	$.707/Q_{ts}$	.952	
.8	$1/(.64/Q_{ts}^2 - 1)$	$.718/Q_{ts}$	.885	.21
.9	$1/(.81/Q_{ts}^2 - 1)$	$.747/Q_{ts}$	.851	.69
1.0	$1/(1.00/Q_{ts}^2 - 1)$	$.786/Q_{ts}$	.835	1.25
1.1	$1/(1.21/Q_{ts}^2 - 1)$	$.832/Q_{ts}$	.830	1.83
1.2	$1/(1.44/Q_{ts}^2 - 1)$	$.883/Q_{ts}$	.831	2.41
1.3	$1/(1.69/Q_{ts}^2 - 1)$	$.936/Q_{ts}$	.835	2.97

Table 5. Second-order loudspeaker system alignments. The values of the design ratios Vab/Vas and f3/fs depend on the Qts of the woofer used, so the table entries for those ratios are formulas rather than constant numbers.

filled with damping material because it spoils Qb, the box Q. (Though a one-inch thick liner on the walls to eliminate echoes and "boxy" sound is OK.)

2) It doesn't have a vent to take up air volume. In a vented system, the vent takes up some volume (Fig. 1) so the actual box volume has to be larger than Vab by that amount.

3) The sealed system's gentler rolloff means the phase shift in the bass region is less. This seems to give the bass more realism and "punch," but we have not made controlled tests.

Theory does not address the question of which response "shape" is sonically the most accurate. We have found that for acoustic suspension (C2 alignment) speakers, the Qtc equal to 1.0 alignment (as in Example 9), which has a 1.25 dB peak, gives the most natural sounding reproduction.

Some speakers with a reputation for "high efficiency" have Qtc as high as 2.0. The resulting 6 dB peak makes the bass sound loud, though not accurate to the critical ear.

You can study efficiency versus Qtc value simply by running down the Ka column in Table 5. As you can see, Ka (which is proportional to f3 for a given volume and efficiency) rises drastically for Qtc values below 0.8.

Unlike the higher-order designs, the second order system is simple enough that there are straighforward (though complex) algebraic formulas to express the cutoff frequency, peak height, etc. The following information is from Small, Ref. 2.

The speaker's gain at the system resonant frequency fc for a second-order system is simply equal to Qtc, the system Q. Thus in Figure 5, for example, for the Qtc equal to 1.0 curve, fc is equal to the frequency where the curve crosses the 0 dB axis.

The cutoff (-3dB) frequency is a function of Qtc and is given by:

$$\frac{f_3}{f_c} = \left[\frac{(1/Q_{tc}^2 - 2) + \sqrt{(1/Q_{tc}^2 - 2)^2 + 4}}{2}\right]^{1/2}$$

The frequency of the bass peak top, for $Q_{tc} > 1/\sqrt{2}$, is:

$$\frac{f_{Gmax}}{f_c} = \frac{1}{[1 - 1/(2Q_{tc}^2)]^{1/2}}$$

The height of the bass peak, in dB, is:

$$G(jw)_{max} = 20\log_{10}\left[\frac{Q_{tc}^4}{Q_{tc}^2 - .25}\right]^{1/2}$$

Interpolating between alignments

Unlike the hypothetical woofer with Qts equal to .39 that we have been using in the examples, most woofers will land with Qts values between alignments. The way you handle this is by interpolating.

Interpolating is like filling out your income tax forms, a challenge but not much fun. It is easier to do than to explain. But it is also easy to make mistakes. Most people already know how to interpolate; mainly what we want to do is give you two rules that will help you avoid mistakes.

For an example, let's recalculate Example 1. To refresh your memory, it applied Thiele's Alignment 5 (from Table 1) to a woofer with Qts equal to .39, even though the alignment called for Qts equal to .383. This caused some error, which showed up as a slight peak in the response curve (curve 1 of Fig. 6). Let's use interpolation to create a new alignment for Qts equal to .390 exactly. The new alignment will lie between Thiele's Alignment 5 and 6.

Refer to Example 10, which is a "worksheet" for interpolating. Lines 1 and 3 give the design ratios for Alignments 6 and 5.

EXAMPLE 10. Worksheet for interpolating.

	Alignment	Q_{ts}	V_{ab}/V_{as}	f_b/f_s	f_3/f_s
Line 1	6	.415	.948	.927	.867
Line 2	new	.390	.760	.984	.971
Line 3	5	.383	.707	1.000	1.000
Line 4		.032	.241	-.073	-.133
Line 5		.007	.053	-.016	-.029
Line 6		.219			

Here is the procedure:

1) Subtract each entry in Line 3 from the corresponding entry above in Line 1. Write the answers on Line 4. (You thought we were kidding when we said it was like doing income tax, didn't you?)

2) In the Qts column only, write the new Qts value needed, .390, on Line 2. Subtract Line 3 from Line 2 and write the answer on Line 5.

3) In the Qts column, divide Line 5 by Line 4. The answer is your interpolation factor, IF. Write it on Line 6.

4) For all columns except Qts, multiply the Line 4 entries by the IF and write the answers on Line 5.

5) For all columns except Qts, add Line 5 to Line 3 and write the answers on Line 2.

All done. Line 2 is the new alignment.

Here is the mathmatical basis for the above procedure.

The general interpolation formula is:

$$IF = \frac{W(Q_{ts}) - B(Q_{ts})}{A(Q_{ts}) - B(Q_{ts})}$$

where

$W(Q_{ts}) = Q_{ts}$ value wanted

$B(Q_{ts}) = Q_{ts}$ of line below

$A(Q_{ts}) = Q_{ts}$ of line above

So for Example 10,

$$\frac{.390 - .383}{.415 - .383} = \frac{.007}{.032} = \frac{7}{32} = .219$$

The above may be expressed verbally as "390 is 7/32 (.219) of the way from .383 to .415." Then fb/fs will also be 7/32 (.219) of the way from the line-below value to the line-above value:

$$\frac{f_b/f_s - 1.000}{.927 - 1.000} = .219$$

Note that the denominator here is (.927 - 1.000), not (1.000 - .927). Solving for fb/fs,

$$f_b/f_s = .219(.927 - 1.000) + 1.000 = .984$$

or in general terms

$$W(f_b/f_s) = IF\left[A(f_b/f_s) - B(f_b/f_s)\right] + B(f_b/f_s)$$

where $W(f_b/f_s)$ is the wanted value of f_b/f_s and so on.

Similarly for V_{ab}/V_{as}, f_3/f_s, etc.

$$W(V_{ab}/V_{as}) = IF\left[A(V_{ab}/V_{as}) - B(V_{ab}/V_{as})\right] + B(V_{ab}/V_{as})$$

$$W(f_3/f_s) = IF\left[A(f_3/f_s) - B(f_3/f_s)\right] + B(f_3/f_s)$$

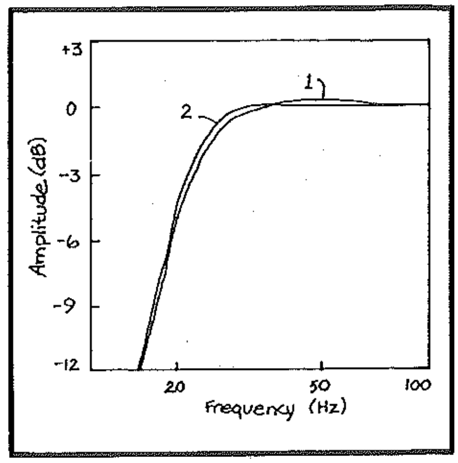

Figure 6. Curve 1--woofer with Qts equal to .39 in a system aligned for a woofer with Qts equal to .383, that is, with Vab equal to 5.28 cu.ft. and Vas equal to 22.8 Hz. Curve 2--same woofer in the alignment obtained by interpolation in Example 10; Vab equals 5.68 cu.ft., fb equals 22.4 Hz.

Did you see where we used (.927 - 1.000) in the denominator, not (1.000 - .927)? **RULE 1:** the position of the numbers in the formula depends on their position in the table, not which number is larger or smaller. Observe in the example that while Qts gets smaller, fb/fs gets larger. The most common mistake in interpolating is using the smaller number as though it were lower in the table, that is, mixing up .927 and 1.000. This is particularly easy to do if you don't use a worksheet as in Example 10.

When you're done, check your result. Qts equal to .390 is closer to the lower line value of Qts in the example; then the value of fb/fs should also be closer to the lower line value. Is it? Same way for f3/fs- is it closer to the lower line value?

RULE 2: always "eyeball" your answers like this to catch simple errors.

If you have never done interpolation before, we suggest that you check your calculations the first few times by "backterpolating". Backterpolating is just interpolating back from your answer to the givens. You recalculate IF using the Vab/Vas column, just the same as you did with Qts. You should get the same IF as before. Then you use the IF to recalculate Qts. You should get the same value as you had at the start. If your backterpolated IF and Qts don't agree with your original numbers, you know you made a mistake somewhere.

Applying the new Vab/Vas and fb/fs ratios from Example 10 to the Example 1 woofer gives flatter frequency response, shown by curve 2 in Fig. 6.

Culinary secrets of the great chefs

Most of the stuff so far we have presented "cookbook" style: "do thus-and-so and you will get a speaker system with such-and-such characteristics." For people who like to know where the recipes come from, here are some of the physical principles behind speaker operation.

The sealed enclosure system is easiest to analyze. When a woofer is put in a sealed box, the sealed "air spring" in the box adds more "spring constant" to the cone-and-voice-coil moving system and raises the resonant frequency of the moving structure. So fc, the system resonance, is always greater than fs, the woofer's free air resonance. The smaller the enclosure air volume, the stronger the "spring", and the higher fc. The frequency is proportional to the square root of the total spring constant. For the woofer alone, the total spring constant is

$$1/V_{as}$$

With the woofer in the box, it is

$$1/V_{as} + 1/V_{ab}$$

So putting the woofer in the box increases the total spring force by the factor

$$\frac{1/V_{as} + 1/V_{ab}}{1/V_{as}}$$

and the frequency increase ratio (fc/fs) is given by

$$\frac{f_c}{f_s} = \left(\frac{1/V_{as} + 1/V_{ab}}{1/V_{as}}\right)^{1/2}$$

The Q of the resonating system is proportional to the frequency, so

$$\frac{Q_{tc}}{Q_{ts}} = \frac{f_c}{f_s} = \left(\frac{1/V_{as} + 1/V_{ab}}{1/V_{as}}\right)^{1/2}$$

or

$$\frac{f_c}{f_s} = \frac{Q_{tc}}{Q_{ts}}$$

Thus, to summarize, putting the speaker in a box raises the resonant frequency (called fc rather than fs when in the box) and also the Q (now called Qtc rather than Qts). In an infinite box, we would have

$$f_c = f_s \text{ and } Q_{tc} = Q_{ts}$$

This leads to the one limitation in fitting different woofers into different second order alignments: Qtc must be greater than Qts. If you try to use an alignment or Qtc less than your woofer's Qts, the formulas will give a negative value for Vab.

To the sealed box system designer, the most important number describing a given woofer is probably not any one of the standard parameters, but rather the ratio fs/Qts, which will be in hertz. It represents the system resonant frequency for the woofer in an enclosure volume that gives Qtc equal to 1.0. The cutoff frequency f3 is then equal, as Table 5 shows, to .786fs/Qts in such a system.

The box volume that will give Qtc equal to 1 is

$$V_{ab} = V_{as}\frac{Q_{ts}^2}{1-Q_{ts}^2}$$

thus

$$V_{as}Q_{ts}^2/(1-Q_{ts}^2)$$

is the second number of most interest in sealed box design.

Alignment Jamming

We get a lot of letters from speaker designers, and one of the things they like to do is use a different alignment than the one called for by the Qts of their woofer. Since Qts is one of the most critical parameters, this sort of "alignment jamming" can lead to serious errors.

Here's a hypothetical example. Let's say we want to build the Example 3 system, which called for a 6.1 cu.ft. enclosure, in a smaller box, say 4 cu.ft. This suggests that we need an alignment with

$$\frac{V_{ab}}{V_{as}} = \frac{4.0}{7.47} = .535$$

The first impulse of your typical energy-wasting, alignment-jamming American citizen is to apply the k equal to 1.5 alignment from Table 2 as in Bad Example 11.

As you can see, the calculations give values for Vas and f3 that don't look too bad. Alas, the response is that the woofer Q is all wrong. The Q is directly related to the cone mass, which is the most critical factor in relationship to the box volume and box tuning frequency (fb).

The cure is to use the "compound parameters"

$$f_s/Q_{ts}^2 \quad \text{and} \quad V_{as}/Q_{ts}^2$$

to do your alignment jamming. This makes the box correct for the cone mass and damping, so the errors fall into the compliance. System performance is least affected by compliance errors, particularly high compliance woofers (usually indicated by low Qts values). D.B. Keele describes the theory behind this in Ref. 5 and it is also covered further in Ref. 6.

Here's how to do it with a fourth-order design:

1) Calculate fs/Qts and VasQts2 for your woofer.

2) Pick the alignment you want to use. As a rule of thumb, your woofer should have Qts within 20% of the value called for by the alignment.

3) Calculate the design parameters Vab, fb, and f3 using the factors from the table.

$$\frac{V_{ab}}{V_{ab}/V_{as}Q_{ts}^2}, \quad \frac{f_b}{f_s/Q_{ts}},$$

$$\text{and } \frac{f_3}{f_s/Q_{ts}}$$

Our woofer has VasQts2 equal to 1.136 cu.ft., so to use it in a 4 cu.ft. box, we need an alignment with

$$\frac{V_{ab}}{V_{as}Q_{ts}^2} = \frac{4.0}{1.136} = 3.521$$

Looking back at Table 2, there is no alignment with this VasQts2 equal to 3.521, meaning (ouch) it can't be done. However, the k equal to 2.0 alignment doesn't look too bad, since its Vab/VasQts2 factor is only a little larger than the one for the k equal to 1.2 alignment—meaning it will take only a little larger box. Example 12 applies this to our usual woofer. As curve 2 in Figure 7 shows, this gives a flatter response shape.

The procedure for fifth and sixth-order designs is the same, with one addition. Calculate faux using the factor faux/(fs/Qts) from the appropriate table, in the same way you calculated fb and f3.

Alignment factor

The last column in Tables 2-5 lists what we call the "alignment factor", Ka. It is useful for comparing the performance of different alignments. As is well known, there is a tradeoff between bass response, volume and efficiency. Some alignments give better "bargains" in the tradeoff than others; Ka tells you which.

BAD EXAMPLE 11. A fourth-order design assuming Q_b = 7. Given the woofer parameters,

Q_{ts} = .39

V_{as} = 7.47 cu. ft.

f_s = 22.8 Hz

From the k = 1.5 alignment in Table 2 we have,

Q_{ts} = .349

V_{ab}/V_{as} = .539

f_b/f_s = 1.094

f_3/f_s = 1.307

Therefore, applying some arithmetic,

V_{ab} = .539 x 7.47 = 4.03 cu. ft.

f_b = 1.094 x 22.8 = 24.9 Hz

f_3 = 1.307 x 22.8 = 29.8 Hz

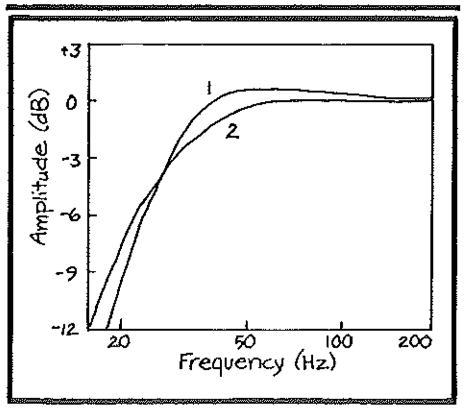

FIGURE 7. Curve 1--Bad Example 11 design. Curve 2--Example 12 design.

EXAMPLE 12. A fourth-order design assuming Q_b = 7. Given the woofer parameters,

$$Q_{ts} = .39$$
$$V_{as} = 7.47 \text{ cu. ft.}$$
$$f_s = 22.8 \text{ Hz}$$

Therefore,

$$V_{as}Q_{ts}^2 = 1.136 \text{ cu. ft.}$$
$$f_s/Q_{ts} = 58.5 \text{ Hz}$$

From the k = 2.0 alignment in Table 2 we have,

$$V_{ab}/(V_{as}Q_{ts}^2) = 4.041$$
$$f_b/(f_s/Q_{ts}) = .350$$
$$f_3/(f_s/Q_{ts}) = .494$$

Therefore,

$$V_{ab} = 4.041 \times 1.136 = 4.59 \text{ cu. ft.}$$
$$f_b = .350 \times 58.5 = 20.5 \text{ Hz}$$
$$f_3 = .494 \times 58.5 = 28.9 \text{ Hz}$$

Here's how it works. Let's say you have a design requirement for a 2 cubic foot system operating at 1/2% efficiency (radiating into halfspace). You want to see what kind of cutoff frequency (f3) values you can get with the d equal to 1.5 and k equal to .5 alignments in Table 1.

It can be shown that f3 is the product of three factors:

$$f_3 = K_a \times K_m \times K_{ev}$$

Km and Kev are explained further in the next section. For now, you should know that for a 1/2% efficient, 2 cubic foot system, and assuming typical mechanical losses in the woofer, the product KmKev is equal to about 49 Hz.

Using Ka, we can calculate f3 for the two alignments:

k	$K_a \times K_mK_{ev}$	$= f_3$
1.5	.749 × 49	= 36.7 Hz
.5	.746 × 49	= 36.6 Hz

Thus the two alignments give nearly equal bass response. Note, however, that they demand quite different woofers. The k equals 1.5 alignment has

$$V_{ab}/V_{as} = .539$$

so that

$$V_{as} = V_{ab}/(V_{ab}/V_{as}) = 2.0/.539 = 3.71 \text{ cu.ft.}$$

The k equal to .5 alignment, on the other hand, has

$$V_{ab}/V_{as} = 2.843$$

so that

$$V_{as} = V_{ab}/(V_{ab}/V_{as}) = 2.0/2.843 = .703 \text{ cu.ft.}$$

The alignment factor also makes it easy for driver designers to work backwards from system performance specifications (N0 and f3) to the driver parameters needed, which is otherwise rather difficult. We just showed how to find the woofer's Vas; finding fs is easy. Using Ka gives us f3, and the tables give f3/fs for each alignment, so

k	$f_3/(f_3/f_s)$	$= f_s$
1.5	36.7/1.307	= 48 Hz
.5	36.6/.644	= 23.6 Hz

We now have all the woofer parameters. fb (and faux if appropriate) can be obtained by applying the coefficients from the tables as usual.

Remember that these computations are subject to several conditions:

1) You need some knowledge of your woofer's mechanical losses. In the examples we have been assuming that Qes/Qts equals 1.25.

2) It assumes that you can design (or select) a woofer with parameters to fit the alignment. We are not talking about comparing different alignments for the same woofer now; that would mean different Vab for each alignment. Here we are talking about different woofers for a given Vab.

3) It assumes the woofer is radiating into ideal halfspace, that is, there are no room acoustic effects. Real rooms will actually make the system perform better because of bass reinforcement and sound pressure level (SPL) increases due to reflections.

In spite of these limitations, the alignment factor is very effective for making comparisons between alignments.

Efficiency-volume factor

This section is mostly math analysis. Math analysis helps some people understand physical phenomena, and for others it just gets in the way. You can skip this part without missing too much if you belong to the second (larger) group.

The formula for the efficiency-volume factor is

$$K_{ev} = \left(\frac{c^3 N_0}{4\pi^2 V_{ab}}\right)^{1/3}$$

where N0 equals the midband efficiency as a pure ratio (N0 equals 1.0 is 100% efficiency) C equals velocity of sound equals 1131 a second.

The famous tradeoffs between bass response, efficiency and box volume appear in this factor. You can see that if you double N0 (holding everything else constant), Kev will increase by the cube root of 2, 1.260 times. Doubling the volume will decrease Kev by a factor of one over the cube root of two, or .794. And, as we have seen, f3 is proportional to Kev (assuming the other factor's constant).

This all derives from work by Small (Ref.3). He gives the efficiency as:

$$N_0 = \frac{4\pi^2 f_s^3 V_{as}}{c^3 Q_{es}}$$

this can be rewritten

$$f_s = \left(\frac{c^3 N_0 Q_{es}}{4\pi^2 V_{as}}\right)^{1/3}$$

Therefore,

$$f_3 = \frac{f_3}{f_s} f_s = \frac{f_3}{f_s}\left(\frac{c^3 N_0 Q_{es}}{4\pi^2 V_{as}}\right)^{1/3}$$

The part in parentheses can be expanded into three other factors so that

$$f_3 = \frac{f_3}{f_s}\left(\frac{V_{ab}}{V_{as}} Q_{ts}\right)^{1/3} \times \left(\frac{Q_{es}}{Q_{ts}}\right)^{1/3} \times \left(\frac{c^3 N_0}{4\pi^2 V_{ab}}\right)^{1/3}$$

or $$f_3 = K_a \times K_m \times K_{ev}$$

We have been calling the first factor, Ka, the alignment factor because the three factors in it come from the alignment data. Thus Ka depends only on the alignment chosen and is useful for comparing alignments. It is a pure ratio. Assuming the other factors (box volume, efficiency, and mechanical losses) are held constant, f3 for different alignments will be proportional to this factor.

We may call the second factor, Km, the mechanical loss factor. It is also a pure ratio. It depends primarily on the driver construction, though anything else that affects either Qts or Qes or both will affect it. Small points out (Ref 3) that the Q of the woofer in the system, which he calls Qt, can be different from Qts, the Q of the woofer itself. Stuffing in the box around the woofer will add mechanical damping and make Qt lower than Qts. In that case Qt should be used in place of Qts in design work and in the previous formula above. Resistance between the amplifier and speaker will raise both Qes and Qts, and if the change is appreciable the modified values should be used in calculations.

We have called the third factor, Kev, the efficiency-volume factor. It has units of hertz. In comparing systems with different efficiency and volume specifications (assuming the same alignment and driver mechanical losses) f3 will be proportional to Kev.

(The subscript "ev" here has no relationship to the Electro-Voice Company, where coincidentally Ray Newman and Don Keele made some of the first practical applications of Thiele and Small theory, described in Refs. 11 and 4.)

For frequent calculations it is convenient to have Kev defined for N0 in percentage points and Vab in cubic feet:

$$K_{ev} = 71.56(N_0/V_{ab})^{1/3}\ \text{Hz}$$

If you examine the first equation in this section, you will see that the value of Kev can also be reduced by lowering c, the velocity of sound inside the enclosure. Filling the enclosure with a gas heavier than air, or partially filling it with damping material in the case of a second-order system, will reduce the velocity of sound and therefore reduce f3. (Increasing the value of pi would also have the same effect).

The preceding analysis describes the details of the efficiency-volume-bass response tradeoff, but does not explain physically how it comes about. How it arises is easiest to visualize in the second order case.

Assume a 2 cu.ft. system with Qtc equals 1.0. Both fc (and hence f3) and efficiency are controlled mainly by the same factor--cone mass. Increasing cone mass reduces fc, reduces efficiency, and increases Qtc. Bl product (magnetic field strength times the length of voice coil wire in the field) also has an effect on the efficiency, but much less effect than the cone mass has. When you increase cone mass, you have to increase Bl to correct the Q. But the Bl increase is not enough to fully offset the efficiency decrease due to the additional mass. Result: lower bass response takes more mass which costs you efficiency.

Note that, given constant Q, higher Bl product is associated with lower efficiency--a surprising result. Doubling the cone mass (and increasing Bl by 41% to keep the Q the same) reduces fc a half-octave and reduces efficiency 4.5 dB.

Also let us remind you that the "1/2 octave costs 4.5 dB" law assumes a given alignment, in this case the second order Qtc equals 1.0 alignment. Between alignments the law does not hold; one alignment may provide much better f3 for a given volume and efficiency than another. Hence the relentless search for better alignments by relentless searchers like John Hoge, Mile Nestorovic, and J. Robert Ashley.

Comparing alignments

We have used the same woofer for most of our examples, one with Qts equal to .39. In the sixth-order case, We preferred a woofer with Qts equal to .312 because it allowed a more reasonable degree of bass peaking in the auxiliary filter. This led to designs for the different system-types (orders) calling for much different box volumes and performance. To summarize them here:

Ex.	Order	Vab	F3
3	4th	6.10 cu.ft.	24.3 Hz.
5	5th	5.39 cu.ft.	28.5 Hz.
7	6th	5.13 cu.ft.	18.2 Hz.
8	2nd	1.34 cu.ft.	46.2 Hz.

Order	Q_{ts}	K_a	f_3 (Hz)	f_s (Hz)	V_{as} (ft^3)	f_b (Hz)	f_{aux} (Hz)	Q_{aux}
4	.39	.728	35.7	33.5	2.45	34.4		
5	.39	.822	40.3	32.2	2.77	35.3	26.8	
6	.312	.516	25.3	25.3	4.56	25.3	25.3	1.932
2	.39	.851	41.7	21.8	8.65			

Table 6. Four loudspeaker system designs for N0 equals 1/2% Vab equals 2.0 cu. ft. f3 was determined using Ka. fs, fb and faux were determined using f3 and the factors f3/fs, fb/fs and faux/fs from Tables 2 - 5. Vas and Qaux were determined using factors from the tables.

Thus the fourth, fifth, and sixth-order alignments all used boxes in the same volume range, around 5-6 cu.ft. The fourth and fifth order designs gave comparable performance with f3 around 25 Hz. The sixth-order system gave much better performance with f3 at 18 Hz. The second-order design called for a much smaller box and much higher f3 than any of the others.

All these designs had the same efficiency, of course, since they used the same woofer; the woofer with Qts equal to .312 we used in Example 7 was the same as the Qts equals .39 woofer, except with the compliance increased. The moving mass, cone area, and Bl product--which determine the efficiency--were the same.

The woofer designer's work is made more complicated by the fact that changing one of a woofer's mechanical properties--compliance, cone mass, mechanical resistance, voice coil resistance, etc.--can cause several of the Thiele-Small parameters to change. It is interesting to see the effects of the change in this woofer.

Increasing the compliance shifted all three of the Thiele-Small parameters; Qts, Vas and fs. Increasing compliance by a factor of 1.563, as we did, decreased Qts by a factor of 1/1.563, or 0.800, from .39 to .312. It decreased fs by the same factor, from 22.8 to 18.2 Hz. And of course 1.563 times 7.47 cu.ft., the old compliance, was 11.67 cu.ft.

That the efficiency was unchanged by these shifts is reflected in the fact that the quantity

$$\frac{f_s^{\,3} V_{ab}}{Q_{ts}}$$

which is proportional to the right side of Small's efficiency equation, stayed constant.

Another useful way of making comparisons is to choose a certain box size and efficiency, and then apply different alignments and see what sort of f3 we can get. This requires that we be able to vary woofer parameters to suit.

Let us assume $N_0 = \frac{1}{2}\%$

$$V_{ab} = 2.0\ \text{cu.ft.}$$

Let's use Qts equal to .39 alignments for the fourth and fifth-order cases, and the practical Qts equal to .312 alignment for the sixth-order case. For the second-order case let's use the Qtc equal to .9 alignment, which has fairly high efficiency and desirable shape, with Qts equal to .39 woofer. Assume, as we have been, that Km equals 1.077.

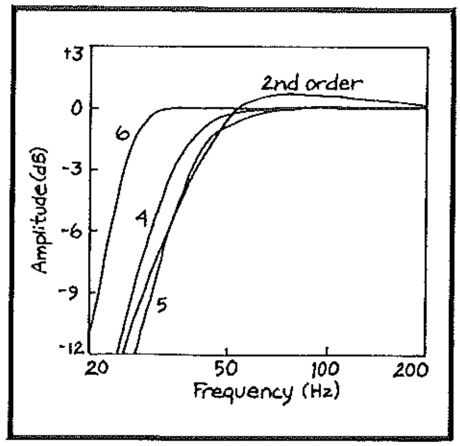

Figure 8. Frequency response curves for the four systems in Table 6. Curve 2--second-order system. Curve 4--fourth-order system. Curve 5--fifth-order system. Curve 6--sixth-order system.

Table 6 gives the design values that result.

The frequency response curves for the four systems are plotted in Figure 8. The fourth and fifth-order designs give similar response. The second-order design has cutoff frequency comparable to the last two and a peak of .69 dB. The sixth-order system gives about a half octave more bass than any of the others.

Vent Design

A vent has acoustic mass that depends on its physical length (Lv) and cross-sectional area (Sv). The equivalent acoustic length (Lva) is a little longer than the physical length because the air just outside the ends of the vent moves too and contributes some mass.

The acoustic mass is proportional to Lva/Sv. Note that this is not simply the mass of air in the vent, which would be proportional to LvaSv. The area Sv goes in the denominator because smaller vent area means higher acoustic mass; in a small vent, the air has to move faster through the vent for a given volume flow.

Thiele has given us (Ref. 1):

$$L_{va}/S_v = 2697/(f_b^{\,2} V_{ab})$$

where Lv is in inches, Sv in square inches, and Vab in cubic feet.

Most vented systems use a duct mounted in the front panel as in Fig. 1. For this type of vent,

$$L_{va} = L_v + .823\sqrt{S_v}$$

where the second term is the correction for the air at the ends.

If the vent is simply a hole in the front panel--which is sometimes the case for large enclosures--then:

$$L_{va} = L_v + .958\sqrt{S_v}$$

where Lv is just the thickness of the panel and the second term represents the air around the hole, which contributes most of the mass.

Thiele got these formulas from Langford-Smith, Ref. 13. Langford-Smith got them from Plach and Williams, Ref. 14. The formulas agree with Beranek's data (Ref. 15, p. 132). Nevertheless, it's a good idea to make your vents adjustable, just in case. You can adjust plain-hole vents with a sliding shutter to vary Sv, and duct-type vents by trimming the duct length.

Let's design a vent for the fourth-order system using the W808R that was described in Example 4. See Example 13.

For unobstructed airflow, we need clearance between the end of the duct and the back panel at least equal to the duct diameter, 3 inches. So if the inside front-to-back dimension of the enclosure is 13 inches or more, that will be fine.

We have made plans for a narrow, tapered enclosure as shown in Fig. 9. (When people ask why it's narrow at the top, tell them that the narrow front panel area around the tweeter reduces diffraction effects from the cabinet edges and provides better imaging. (The real reason is that we want a speaker that looks like one Starbuck and Boomer would have.)

Examining the equation for Lva/Sv, the first equation in this section, you will see that the greater the area of the pipe, the longer it must be. While it might seem the smaller the duct the easier it is to make, in small ducts the air velocity gets high and "wooshing" noises and lower box Q can result. We recommend that you make the vent as large as practical, 3 inches in diameter at least if possible.

Once you've built the enclosure and vent, it is not difficult to measure fb and adjust the vent to correct it if necessary. At the box resonant frequency, the woofer cone motion is minimum, so any method of detecting minimum cone motion will enable you to identify fb if you have a calibrated sine wave oscillator.

Two possible ways:

1) Put a sound level meter (or microphone with amplifier and meter) up to the woofer cone and find the frequency at which woofer output (not total system output) dips.

2) When the woofer's cone motion is minimum, its impedance is minimum. Find the minimum-impedance frequency between the two resonant impedance peaks a vented system displays. Reference 12 tells how to measure the impedance.

EXAMPLE 13. A vent for a speaker system with

V_{ab} = 2.29 cu. ft.

f_b = 26.5 Hz

Therefore:

$$L_{va}/S_v = 2697/(f_b^2 V_{ab}) = 2697/(26.5^2 \times 2.29) = 1.677 \text{ in.}^{-1}$$

Note that L_{va}/S_v has units of reciprocal inches, in.$^{-1}$

If we use a 4-inch diameter vent, then

$$S_v = (\pi/4)D^2 = 12.57 \text{ sq. in.}$$

$$L_{va} = (L_{va}/S_v)S_v = 1.677 \times 12.57 = 20.95 \text{ in.}$$

Too long. We can either use a passive radiator--it's easier to get enough acoustic mass with a solid diaphragm in the vent than air--or try a smaller diameter duct.

For a 3-inch diameter vent,

$$S_v = (\pi/4) \times 3^2 = 7.07 \text{ sq. in.}$$

$$L_{va} = 1.677 \times 7.069 = 11.86 \text{ in.}$$

This is more reasonable. The end correction will be

$$.823\sqrt{S_v} = .823\sqrt{7.069} = 2.19 \text{ in.}$$

Therefore,

$$L_v = L_{va} - 2.19 = 11.86 - 2.19 = 9.67 \text{ in.}$$

Limitations of the theory

All this theory is based on the assumption that the woofer is a rigid piston, that is, all parts of the cone move in unison and the cone does not flex (except at the surround, of course).

Perhaps, we should have titled this section "Finding woofers that fit the theory". The theory applies best to woofers with heavy, strong cones. It works better with small drivers, which tend to have fairly rigid cones compared to their weight, than with larger drivers.

Also the theory applies so long as the pressure in the box is not too high; in a very small sealed box, for example, the air pressure tends to "clamp" the outer parts of the speaker cone while the center region and voice coil move independently. The same thing happens with very-high-Q box resonance; the cone becomes immobilized due to large box "back pressure" but the voice coil can move anyway. Thus alignments that result in very high box pressures are the most critical. Alas, these are also the alignments that provide the most extended bass in small boxes; copious low frequency bass radiation is associated with large air volume flow in and out of the box, creating high pressure. The better the alignment, the more important the need for a rigid cone structure.

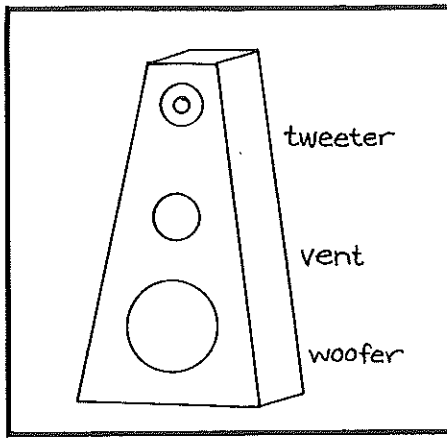

Figure 9. Speaker system designed according to Examples 4 and 13. The enclosure is 30 in. high and 16 in. deep. It is 6 in. wide at the top and 16 in. wide at the bottom which gives (with 3/4-in thick walls) an internal volume of 2.27 cu.ft.

Woofer cones made with a "sandwich" type of laminated cone construction, or reinforced with some material such as Speakerlab's Polylam™ cone treatment, are more rigid than plain paper.

For most woofers 12-inches in diameter or less, with strong cones, systems designed to the theory will work well; errors just change the performance numbers a little. In many calculations the errors are self-cancelling. Cone flexing raises measured fs, Qes, Qts, and Vas. These errors tend to cancel out in calculations involving efficiency; Small's equation

$$N_0 = \frac{4\pi^2 f_s{}^3 V_{as}}{c^3 Q_{es}}$$

suffers the most. It tends to give unrealistically high efficiencies—off by a factor of two or more--for woofers with flexible cones. Designers sometimes calculate the efficiency of available woofers from published specifications for fs, Vas, and Qes, but the results can be misleading. A woofer with a large diameter, light cone will have large fs and Vas and get a high rating from this equation; unfortunately, such woofers are most prone to cone flexing, and the actual efficiency will be much less than calculated. Sensitivity ratings (usually expressed in terms of dB SPL at 1 meter, 1 watt) are a more reliable gauge of woofer efficiency.

Units

We have used English units (cubic feet, etc.) throughout this chapter because that is what the table saw is marked in. Since the alignment tables give the design parameters (f3/fs and so on) in dimensionless ratios, however, you can also use metric units and everything will come out all right; if your woofer specs have Vas in cubic meters, for example, then when you multiply that by Vab/Vas to get Vab, the latter will come out in cubic meters.

Pitfalls

There are a number of pitfalls that often trap first-time speaker designers.

To beginners, Neville Thiele's Alignment No. 5 (Ref.1) looks the best because it calls for a very simple design--a woofer in a vented box. However, as we pointed out earlier, speakers designed to this alignment are unusually susceptible to turntable rumble because the cone is unloaded at subsonic frequencies.

Half-Truth no. 1: "Vented speakers have less cone travel". This one keeps appearing in print by authors who should know better. It's hard to see how it got so far, since there are plenty of vented systems around where you can see the cone bouncing like crazy. We can only guess this is another case of the power of the printed word over visual evidence.

The truth is that only in their useful frequency range do vented systems have less cone travel. Below the low frequency range limit cone motion rises again, as can be seen in Thiele's Figure 10 of Reference 1. So in order to get maximum power-handling capability and minimum distortion, a filter or equalizer of some kind is usually needed to eliminate subsonic noise.

Vented-system cone displacement continues to rise as you go down in frequency, being limited at DC only by the cone suspension compliance. In the sealed system, on the other hand, cone motion is limited by suspension compliance plus air stiffness in the seled box and is held to a reasonable limit with or without additional filtering.

Half-Truth no. 2: "Vented speakers are 4.5 dB more efficient for the same low frequency response and volume."

This one started with a truth published by Richard Small in Ref. 8. He compared Thiele's fourth-order Butterworth alignment, No. 5, to the sealed-box second-order Butterworth (B2) alignment, and observed that the Alignment 5 system was 4.5 dB more efficient. This conclusion is perfectly valid, so far as it goes. Alas, many people have gone a step further and assumed that this means that vented systems are in general more efficient than sealed ones.

Not so. All fourth-order vented systems are less efficient than Thiele's theoretical Butterworth alignment, and most sealed systems are more efficient than the B2 alignment. Vented systems suffer because the theoretical alignment does not take account of energy losses in the vent. It is these losses that make Qb finite (typically 7) rather than infinite. Vent losses reduce the efficiency by about 2dB.

In sealed designs, on the other hand, a very efficient alignment, and the one most acoustic-suspension speaker makers use, is the alignment with Qtc equal to 1.0. It is about 1.7 dB more efficient than the B2 (Qtc equal to .707) alignment.

Thus the efficiency difference between real vented and sealed systems is reduced to only about 0.8 dB, or 20%. or for the same efficiency and low-end cutoff (-3dB down) frequency, the vented box's volume can be 17% less. However, this does not take account of the volume taken up by a ducted vent, and in most small systems the vent indeed must be ducted, adding to the volume required. Under close scrutiny, the 4.5 dB advantage of the fourth-order vented system over the sealed system nearly disappears.

Half-Truth no. 3: "Some higher-order (equalized) alignments are extremely efficient." This one comes from the definition of "efficiency" used, where efficiency is taken as the midband efficiency. Some of the alignments with very high midband efficiencies require considerable bass boost. This demands extra power from the amplifier, and the efficiency rating makes no allowance for this extra power required. When you do allow for it, it becomes obvious that the amazing efficiency rating is only illusory. Thiele made note of this in his original paper (p. 188, Ref. 1), but many other people have neglected this point.

Untruth no. 4: No. 3 also comes in another, even worse version: "You can make a speaker system more efficient by porting the box". Given the woofer, the type of enclosure has no effect on efficiency; the efficiency is determined by the woofer alone. What you can get by indiscriminantly porting a box is a nice boomy resonant peak, which will indeed make the speaker sound louder if you like maximum noise and minimum accuracy.

Where venting a system really pays off is in the sixth order (equalized) systems. These do indeed provide substantially more bass response for a given system, at the cost of increased design complexity.

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Chapter 6: Bass Reflex Equalizer.

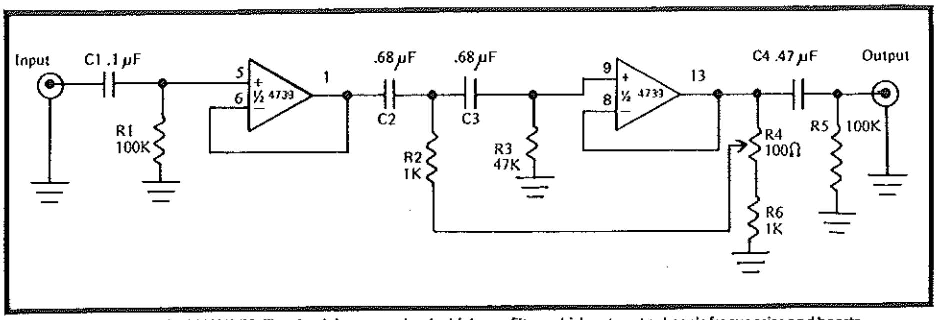


Figure 1. BASS REFLEX EQUALIZER. The circuit is a second order highpass filter, which cuts out subsonic frequencies and boosts frequencies around the bass reflex port resonance. Each operational amplifier is one section of the 4739 dual op amp integrated circuit. Capacitors C2 and C3 control the characteristic frequency and potentiometer R4 controls the peak height. The values of C2 and C3 shown give a characteristic frequency of 34 Hz. The maximum peak height is about 9 dB.

The bass reflex, once famous for the stomach-shuddering bass it could deliver from a gargantuan box (and the hernia you got from moving the box) has been long overshadowed by the miniscule acoustic suspension speakers. But now bass reflex is back again, smaller but better.

The old boom-box builders used to live by one rule: BIG. Their favorite saying was, "the only substitute for cubic feet in a speaker enclosure is more cubic feet." And it worked. Those immovable monoliths really put out the bass, if you had some place to put them.

Now an Australian, A. N. Thiele, has found out how to build the old rafter-rattlers in a reasonable size (Ref. 1). Others attempted to reduce the size by going to ducted ports and high compliance woofers, but all they did was find a new way to make a "leaky acoustic suspension." Acoustic suspension (high compliance) woofers have a nasty tendency to self-destruct when used in a non-airtight box. They count on the sealed air to act as an "air spring" and provide the restoring force that the ultra-loose cone suspension lacks. In an unsealed box the cone travels too far and the woofer eventually tears itself apart.

Thiele made two discoveries through mathematical analysis. First, the self-destructive cone-flopping you see in bass reflex occurs at ultra-low subsonic frequencies mostly generated by turntable rumble and other forms of noise. You can filter these frequencies out with no loss in reproduction accuracy.

Second, the port of the small bass reflex has the ability to put out super bass under special circumstances. Although the low frequency port radiation is normally a few dB lower in amplitude than radiation over the rest of the spectrum (because the port is ducted), the port can still put out the huge air volume flows needed for solid low bass. The port must be ducted in the small bass reflex because the mass of air in the port must be increased to counteract the effect of smaller box volume. The inertia of the increased air mass makes it harder to move, but can be overcome.

Thiele's cure for both these problems—cone-flopping and reduced bass sensitivity—is an equalizer. The equalizer cuts out the destructive subsonic energy and boosts up the bass around the critical port radiation frequencies. With conventional speakers, bass boost often just overdrives the cone and increases distortion. Thiele demonstrated that if the boost is applied to the bass reflex around the port resonant frequency, where cone motion is minimum and port radiation maximum, the speaker can radiate tremendous energy with little increase in distortion.

For the equalizer Thiele recommends second-order highpass filter (which calls an auxiliary circuit). He gives three reasons for using the equalizer:

1) it reduces the speaker cone motion

2) it reduces signal energy below the cutoff frequency of the system, thereby avoiding wasting amplifier power in this region.

3) it extends the low frequency response of the system

Equalizer Design

The equalizer goes in the signal path between the preamplifier and amplifier. Nearly all modern preamplifiers and stereo receivers come equipped with a tape monitor circuit which is very convenient for accomplishing this.

A suitable design for the equalizer is shown in Figure 1. This circuit is described as a "high pass second order filter of the controlled source type" in the standard texts on active filters (Ref. 2). This kind of filter has three characteristics that make it ideal for use with the bass reflex: a sharp cutoff at the low end to eliminate subsonic energy, a controllable peak in the response at low bass frequencies to exploit the port radiation, and flat frequency response from 100 Hz up that leaves the rest of the frequency spectrum unaffected.

Use of an integrated circuit dual operational amplifier, the 4739, makes construction of the circuit relatively simple. The pinout of the 4739 is shown in Fig. 5.

The circuit shown in Fig. 1 has a characteristic frequency, which Thiele denotes faux, the auxiliary frequency, is controlled by the two capacitors C2 and C3 which are equal in value. To obtain any desired characteristic frequency use the formula

$$C2 = C3 = \frac{23.2}{f_{aux}}$$

where faux is in hertz and C2 and C3 are in microfarads. Figure 2 compares the responses of the filter for different peak height settings. We set the characteristic frequency at the unusually high value of 154 Hz (C2 and C3 equal to 0.15 uF) for this measurement so the low frequency roll-

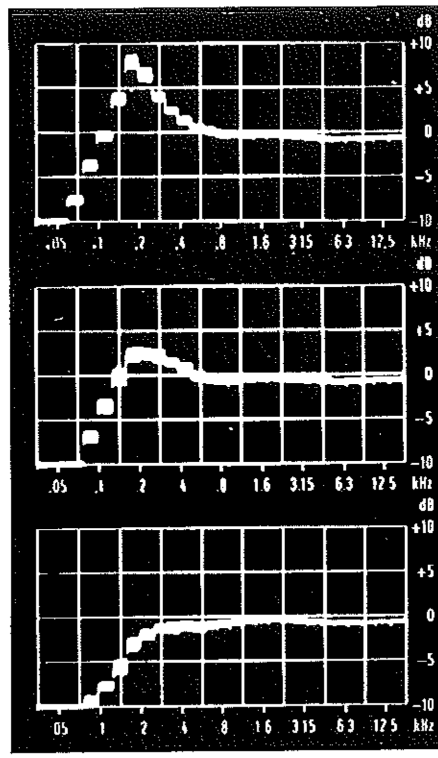
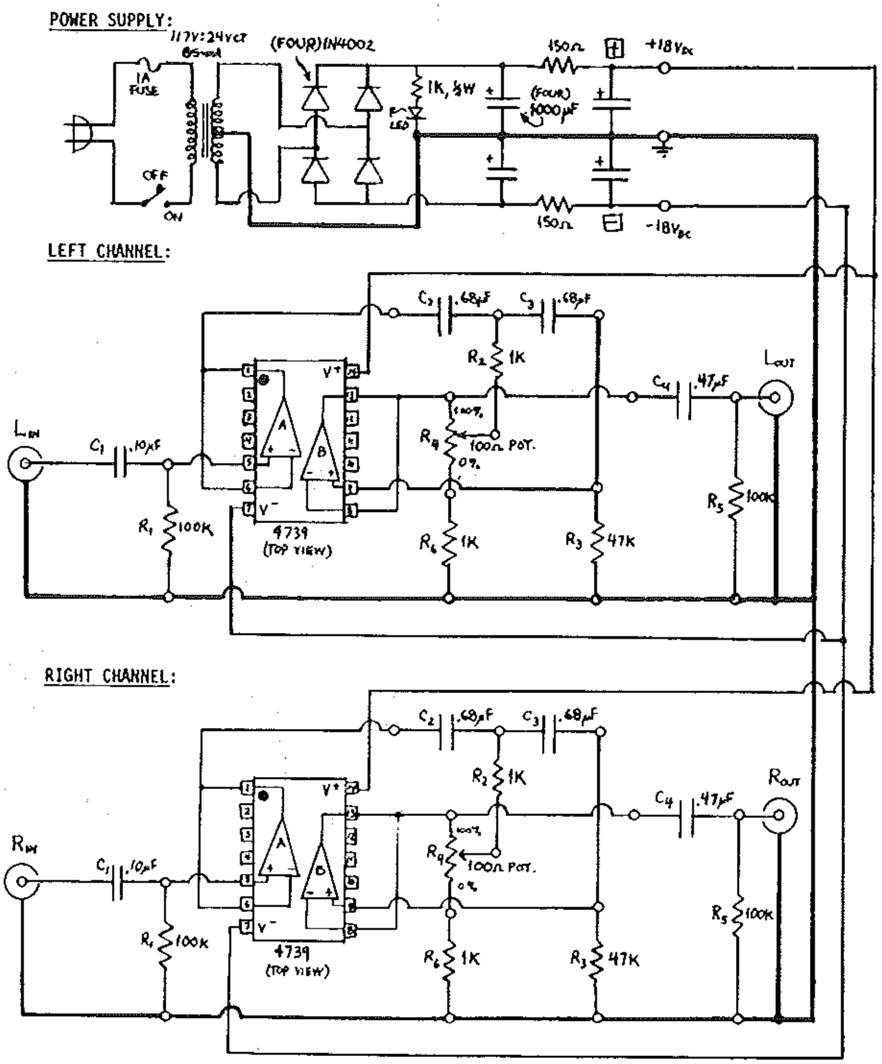


Figure 2. Frequency response of the equalizer with different peak height potentiometer settings. The characteristic frequency is 154 Hz in all three cases. The slight rise at 2.2 kHz is an error in the test equipment, not a characteristic of the equalizer.

off of the filter would be well within the range of the analyzer and fully displayed.

At large peak heights the peak frequency approaches faux, the characteristic frequency of the filter, but at lower peak heights the peak frequency tends to be considerably greater than faux. This is a normal characteristic of a second order highpass filter. The peak frequency becomes very difficult to measure accurately for peak heights below about 5 dB.

The two RC networks C1-R1 and C4-R5 also contribute some low-frequency rolloff—about 2 dB at 3 Hz. Their main effect is to make the cutoff of subsonic energy a little sharper.



BASS REFLEX EQUALIZER 575—PARTS LIST

1- Circuit card (optional)
1- 117 V primary, 24 or 25 VCT secondary, 85 mA (or more) transformer.
4- 1N4002 or other 100 V, 1 A (or greater) rectifying diodes.
4- 1000 uF, 25 V or greater electrolytic capacitors.
1- LED (Light Emitting Diode) for ON indication.
2- Raytheon or Exar 4739 integrated circuit operational amplifiers.
2- 14-pin dual-in-line IC sockets (optional).
(R4) 2- 100 ohm trim pots for mounting directly on printed circuit board (alternatively you can use one 100 ohm dual-ganged, 1/4W or greater, pot for panel-mounting).
2- 150 ohm, 5%, 1/4W carbon film resistor (color code: brown-green-brown-gold).
(R2, R6) 4- 1K, 5%, 1/4W carbon film resistor (brown-black-red-gold).
(R3) 2- 47K, 5%, 1/4W carbon film resistor (yellow-purple-orange-gold).
(R1, R5) 4- 100K, 5%, 1/4W carbon film resistor (brown-black-yellow-gold).
1- 1K, 5%, 1/2W carbon composition resistor (brown-black-red-gold)—goes next to LED.
(C1) 2- .10 uF, 25V or greater capacitor.
(C4) 2- .47 uF, 25V or greater capacitor.
(C2, C3) 4- .68 uF, 25V or greater capacitor.
4- phono (RCA) jacks for inputs and outputs.
1- on-off switch (optional if unit is plugged into a switched outlet).
1- fuseholder and 1 A fuse.
1- line cord and plug.

Figure 3. COMPLETE SCHEMATIC of the Bass Reflex Equalizer 575. This is a convenient parts layout for those of your handwriting the circuit.

Constructing the Equalizer.

The complete schematic and parts list for the Bass Reflex Equalizer is shown in Figure 3. We have given it a title—the 575. It is stereo and includes the power supply needed to power the integrated circuits.

For those who have equipment for etching their own printed circuits, a proven design is given in Fig. 4, shown in actual size. Fig. 5 shows where the parts fit on the board.

Here are a few hints on construction. Beginners should use only first-quality parts. Bargain integrated circuits supplied by electronics hobby stores and mail order outfits usually owe their low price to the fact that they were rejected by somebody for substandard performance. Electric current is invisible, so unless you have test equipment (and the experience to use it) you will not be able to debug a circuit that does not work because of a defective part.

Use good soldering techniques. To get good solder joints, use 60/40 rosin core electronic solder and make sure the conductors you are soldering are clean. The more actual metal-to-metal contact between the conductors, the stronger the joint will be. Put a little solder on the tip of the iron to wet it so it will conduct heat to the joint. Put the iron against the joint, heat it, then flow solder into the joint. The joint itself should be hot—hot enough to melt the solder. However, when the copper foil conductor of a printed circuit card is over-heated, its bond to the board relaxes and the foil lifts off, frustrating any good mechanical joint. Do not melt solder onto the tip of the iron and then carry it to the joint—this burns the solder and makes a poor joint.

Troubleshooting—if you get no signal at the output of the equalizer, measure the voltage at the output pins of the integrated circuits (IC's). The output of any operational amplifier (op amp), with no signal at the input, ought to be within 1 volt of ground. If it isn't, the IC will probably be "locked up" close to either the plus or minus supply voltage. If so, check the DC current paths at the inputs of the op amp.

If the bias is OK (output voltage near 0 volts at each op amp with no signal present) and the dang critter still won't work, put a test signal on the input (a sine or square generator is handy here) and follow the signal through the circuit with an oscilloscope. At some point the signal will disappear—check that stage carefully.

Most of the bugs we find are simple mechanical problems—shorts between IC pins or to the chassis, one voltage supply not connected, etc. Have a friend look at the thing—they might see something you can't.

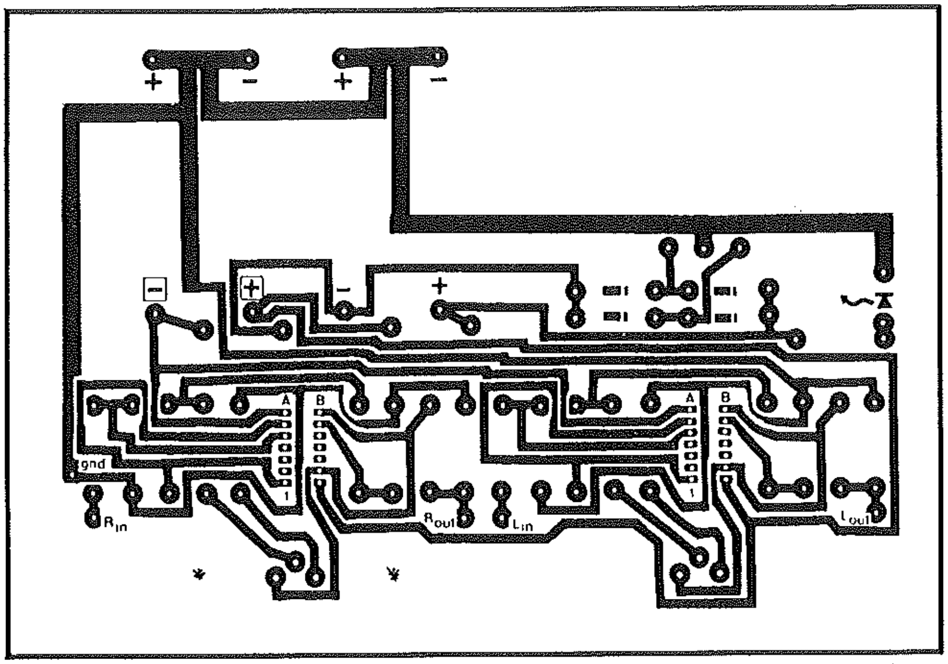


Figure 4. BASS REFLEX EQUALIZER 575 printed circuit card. This board is 4-3/4" x 7-3/8" measured from corner targets. View shown from bottom, copper clad side.

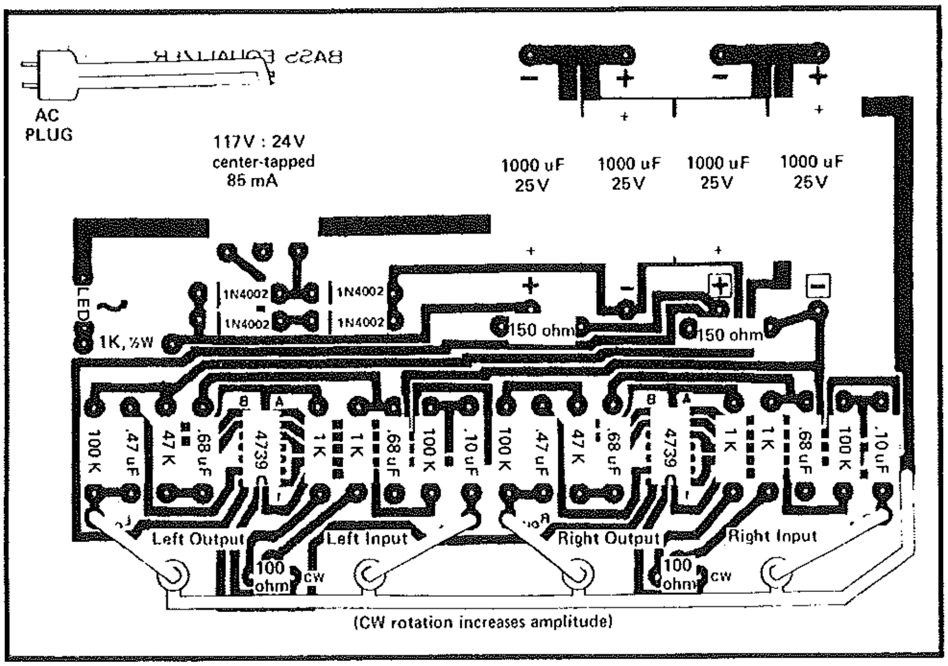
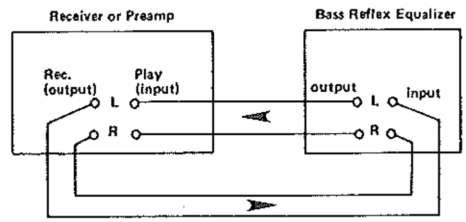


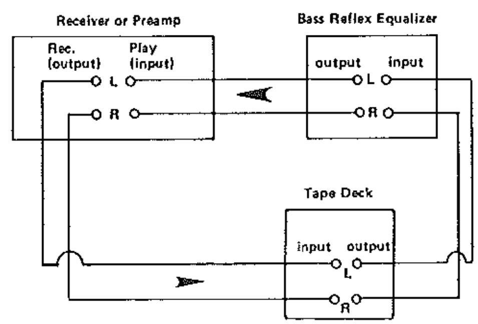
Figure 5. BASS REFLEX EQUALIZER 575 parts layout on circuit card (top view). Be sure to orient the integrated circuits, electrolytic (large 1000 uF/25V) capacitors, diodes (1N4002), and LED correctly as shown. The integrated circuits are oriented by the notches at one end. The electrolytic capacitors have positive or negative signs at one end or the other. The diodes have a band at the cathode end. The LED's cathode will be the lead marked by a flat spot or notch on the rim on the lens, or will be the shorter lead. The phono jacks can be either mounted through holes centered on the asterisks shown on the circuit card pattern or can be chassis-mounted. If mounted on the circuit card, be sure to connect the phono jack common terminals together, and then make a connection from there to the point marked "grid" on the printed circuit card, shown here by a line along the bottom edge of the card.

We have built, tested and used the circuit of Figure 1 and 3, and the accompanying circuit card of Figures 4 and 5. Whereas the Bass Reflex Equalizer works well for us when properly constructed, we assume no responsibility for damages incurred as a result of anyone constructing or using the Bass Reflex Equalizer.

Hooking up the Equalizer.



HOOK UP THE BASS REFLEX EQUALIZER 575 into the tape monitor circuit of a receiver or preamp as shown above. The arrows show the direction of signal flow. Labels used on receivers for the tape circuit input are PLAY, MONITOR, ON, or TAPE IN. Labels used on receivers for the tape circuit output are RECORD, REC, or TAPE OUT. If you have no tape monitor circuit you can insert the equalizer between the preamplifier and power (main) amplifier by using the PREAMP OUT and MAIN IN jacks on the receiver or integrated amp.



IF YOU WANT TO USE A TAPE DECK as well as the equalizer, and your receiver has only one tape monitor circuit, you can hook up both units as shown above.

Using the Filter.

Thiele gives design formulas for bass reflex systems that use the equalizer. However, from our point of view his major accomplishment was showing the tremendous bass potential of reflex designs, not his measurements and formulas on how to do it. His design methods are quite complex and require knowledge of speaker parameters supplied by only a few manufacturers. (Speakerlab is one that does supply the parameters.) You may want to build a system using a woofer which has unknown parameters. For trial-and-error design we have a real time analyzer that gives us a continuous display of a speaker system's frequency response while we make adjustments on the system. We can vary the vent of a bass reflex and find the optimum size in a few minutes.

You may not have a real time analyzer but you can still work by trial and error and get some pretty good results if you have a good ear. This may not sound very scientific but it is in fact the way many speaker systems, and even some great ones, were designed.

Take a bass reflex system that already performs well and hook it up to the filter, or even just put a likely looking speaker in a fairly plausible box and adjust the vent until it sounds best.

Then hook up the bass reflex equalizer. For a start you might use C2 equals C3 equals .47 uF so faux equals 49 Hz. Run the peak height up and down and see what happens. If this has no effect on the sound, faux is too low—raise it. If varying the peak height definitely affects the sound but does nothing to improve it, faux is probably too high—reduce it. If large peak heights make the bass over strong but solid, you're in the ballpark. Reduce the peak height to where you get maximum realistic bass. When you get it set right, bass will go low and be solid, and the visible cone motion will be substantially reduced below what it is without the filter.

This approach will not give an "optimum" design, merely one that is dang good. Each of Thiele's alignments is optimum in the sense that each satisfies some different mathematical criterion. A trial-and-error design will satisify only one criterion—excellent performance.

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