Subwoofers
Presenting The Fundamentals Parts One - Five

RICHARD HARDESTY

Following numerous requests, we present the excellent technical essay by Richard Hardesty, a former Widescreen Review *technical editor, on the technologies of subwoofer design and performance. This republished work does not include the actual subwoofer product reviews. The original Essential Subwoofer Guide is still available, though limited, in print as a Special Edition backorder. It is also available for purchase as a DRM-free PDF download from www.WidescreenReview.com and as a separate purchase from the iPhone, iPad, and Android mobile platforms.*

As Dr. Boom states in his conclusion, this special edition contains all that you need to know in order to make an informed choice when purchasing a subwoofer. You can also refer to this guide when installing the subwoofer(s) in your system.

Forward

This *Widescreen Review* Special Edition includes a compilation of a series of articles and reviews published in *WSR* Magazine over a period of several years. I have edited the introductory articles in an attempt to make them a little easier to read and added some new material in response to readers' questions. Some individual product reviews are reprinted in their original form and are arranged in alphabetical order, not necessarily in the order in which the products were tested. Many previously unpublished product reviews are included as well.

The review process evolved somewhat during the course of this extended project, and the later reviews have a slightly different written format, but all testing and evaluation was done in the same room with the same associated equipment in order to maintain consistency. Sound pressure measurements represent in-room levels taken at the listening position about three meters (average) from the subwoofers under test. These numbers are useful for comparison purposes only and cannot be directly equated to other measurements taken by other methods in other environments.

Widescreen Review is a home theatre magazine, but many people utilize their home entertainment systems for music listening as well as for film viewing. I have tried to evaluate each of the subwoofers for both their capability to reproduce sound effects in a home theatre system and for musical accuracy.

If you consider bass to be a simple commodity, then how loud a subwoofer will play and how much it costs are your main concerns. These factors disregard sound quality which is very important to many listeners. I have tried to cover all aspects of the performance of the products reviewed by providing measurements of maximum output capability and subjective evaluations of sound quality along with discussions of the technical reasons for engineering compromise.

Home theatre enthusiasts want subwoofers that play very loudly with low distortion at high levels. They are generally satisfied by products with low-frequency limits of 25 Hz to 30 Hz. Music listeners may prefer to trade some of this high-output capability for improved transient response and extended low-frequency performance. Subwoofers that play the loudest seldom sound the best when reproducing music or extend to the lowest frequencies.

The more you know about a given model, the easier it will be to determine if it meets your particular needs. The information in this special edition should help you narrow the field of choice to those products which are most likely to satisfy your personal requirements.

Subwoofers
Presenting The Fundamentals

RICHARD HARDESTY

***Subwoofer**: a specialized device designed to reproduce a small band of frequencies in the most difficult and demanding portion of the audio range—the bass.*

Introduction Part One
(Originally Appeared In Issue 24)

Improved Sound From A Better Foundation

Adding one or two (or more) well-designed subwoofers to your system, and taking the time to make sure that they are properly integrated with the main loudspeakers and the room, can dramatically enhance your enjoyment of music and films at home. Bass is the foundation of most music and there is something primal about our response to it. And nothing can pump up the excitement of film-watching more than adding some power and impact to the low frequency sound effects.

Subwoofers Can Do More Than Just Add Bass

Of course, subwoofers can extend the low-frequency response of the system and add power and dynamics to the bass, but they also offer other very important benefits: they can lower distortion in the midrange, prevent amplifier clipping and potential loudspeaker damage, and make the whole system play louder with greater dynamic range and less audible strain. All this is accomplished by relieving your main amplifiers and speakers of the demands of reproducing very low frequencies.

Allocating The Work

When you add subwoofers to an audio system, the main amplifiers and loudspeakers no longer have to produce low bass frequencies and can be relieved of this duty through the use of high-pass filters.

Using an 80 Hz high-pass filter with even a very small amplifier will make this amplifier seem much more powerful because the major energy demands for music and sound effects occur below this frequency. Reducing the demand for high current at low frequencies lightens the load on the amplifier, and this can reduce distortion and prevent clipping, which is the major cause of loudspeaker damage.

Removing the bass load from the woofer in a full-range loudspeaker will reduce intermodulation distortion in the critical midrange, a good part of which is also handled by the woofer in a full-range loudspeaker.

Preserving The Midrange

The crossover frequency from the woofer to the midrange driver in a three-way loudspeaker typically falls between 500 Hz and 1000 Hz, and in a two-way loudspeaker the crossover frequency may be as high as 3000 Hz. The average range of the soprano voice covers the two octaves above “Middle C” on the piano, or from 261.63 Hz to 987.77 Hz. This is the heart of the midrange, but in many full-range loudspeakers this part of the spectrum is actually reproduced by the woofer.

Removing the bass frequencies below 80Hz from the driver that is also covering these critical midrange frequencies (or part of them) will limit long cone excursions and reduce distortion. Of course, the high-pass filters that are used to accomplish this task can also have negative effects, so the choice of quality, frequency, and slope is critical, as is the proper adjustment of level and phase.

No Guessing Game

Most of us are fascinated by loudspeakers. Even the least discerning listener can hear the differences between one model and another, and we all wish they could sound better. Engineers are no exception, and more scientific investigation has been devoted to analyzing the performance of loudspeakers than all the other aspects of audio combined, particularly low-frequency performance.

Amateurs and professionals alike have experimented with every imaginable method to produce bass, and no practical device has proven to be better than the conventional dynamic driver using a cone diaphragm, mounted in a box-type enclosure. There are, of course, many ways to load this type of driver-enclosure combination, and every one has been explored and the results described in the literature.

Virtually all aspects of loudspeaker performance at low frequencies can be modeled mathematically, and today there are many computer programs available that allow any engineer with a PC to design a subwoofer with predictable results.

There are few mysteries left, but there are significant compromises and trade-offs that are necessary, and consumers should be aware of the sonic effects of these compromises to help separate the quality products from the “boom boxes.”

Designing a subwoofer is no longer a guessing game and the “bad” ones that are for sale are usually not bad by accident, they are made that way on purpose. Why? Because booming, exaggerated bass is seductive to the novice and easy to sell.

Believe me, *you* don’t want it. There are few things that can destroy the fidelity of a system more effectively than a boomy subwoofer. So arm yourselves with a little technical information so that you can make wise purchasing decisions.

Balancing Compromises

If you want to produce low frequencies at high levels, the laws of physics require a large driver (or several smaller drivers) in a *very* large box. An 8-inch driver with a cone area of about 28 square inches would have to move nearly two inches peak-to-peak (which it can’t do) to produce one acoustical watt at 50 Hz (100 dB SPL at eight feet requires about .4 acoustical watts and the decibel scale is logarithmic).

If the frequency is halved, the excursion is increased by a factor of four. It’s obvious that a single 8-inch driver will not play very low *and* very loud, no matter how much it is equalized or *whatever* you do to it. (Several 8-inch drivers used together can equal the cone area of a single larger driver with some advantages, as in the Vandersteen subwoofers.)

If the cone area is doubled, the excursion is halved, so more cone area is necessary to keep excursion within acceptable limits. But bigger drivers with more cone area (or multiple smaller drivers) need bigger enclosures, and *big* is one of the problems.

Big Is Better, But...

If we take a popular 18-inch driver like the JBL 2245H and do the necessary math, we find that for good performance without equalization, a vented enclosure with a net internal volume of 12 cubic feet would be required. (The resulting frequency response and output levels are shown in the illustration.)

This combination would play very loud but would be a very large piece of furniture indeed. Most living rooms could not accommodate one subwoofer of this size, let alone the three or more required for surround sound, so loudspeaker designers have set out to find ways to fool Mother Nature—and maybe you as well.

Making Them Smaller

There are several methods commonly used to make subwoofers with infra-sonic response compact enough to be practical for home use: electronic equalization (including "servos" using motional feedback), compound or "Isobarik®" loading (Isobarik is a trademark of Linn Products of Scotland but the word has been misused so often that it has become a generic term), band pass loading (popular at lower price points) and "high Q" alignments, which sacrifice accuracy and transient response for reduced size and accentuated mid-bass output. Virtually all the subwoofers in this review use one or more of these techniques with varying results.

An Order May Not Be A Command

In electronics, a filter is a circuit that blocks certain signals and allows others to pass. In this article we will be discussing high-pass, low-pass, and band-pass filters. Each of these filters has a *pass* band and a *stop* band.

The pass band is the range of frequencies that are allowed to pass through the filter. The stop band is the range of frequencies that are blocked or rejected.

A high-pass filter allows frequencies that are above the cut-off frequency to pass and blocks frequencies below the cut-off frequency. A low-pass filter passes signals below the cut-off frequency and blocks signals at frequencies above cut-off levels.

A band-pass filter has a high-frequency cut-off point and a low-frequency cut-off point, and passes signals that fall between these two cut-off points and blocks signals that fall above the high-frequency cut-off or below the low-frequency cut-off.

The simplest passive electrical filters contain a single reactive component: a series capacitor for high-pass or a series inductor for low-pass. This is called a single-pole or first-order filter and it gently attenuates the stop band at about 6 dB per octave. (An octave is the range between one frequency and twice that frequency. The range between 20 Hz and 40 Hz is an octave. So is the range between 500 Hz and 1000 Hz.)

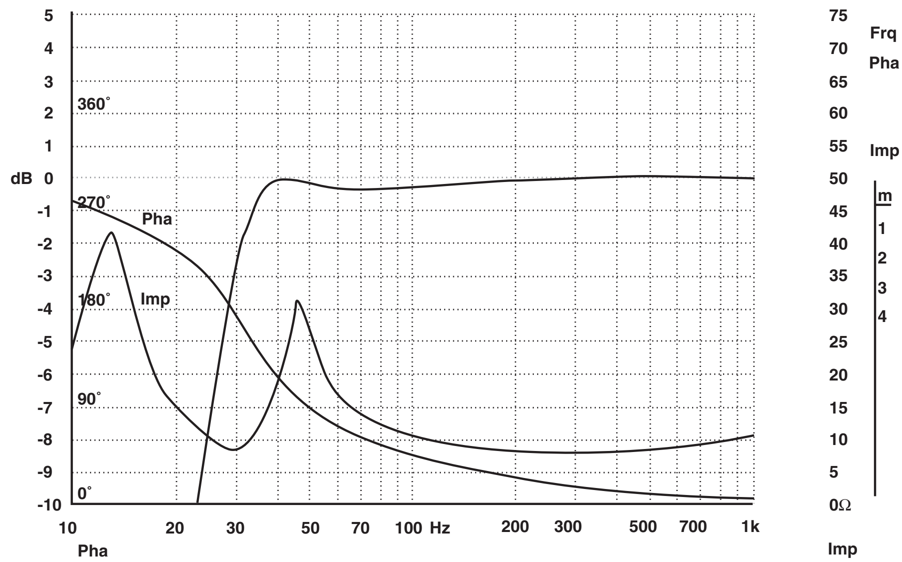
Adding a second reactive component—an inductor in shunt (parallel) for a high-pass filter or a capacitor in shunt (parallel) for a low-pass filter—creates a second-order filter that rolls off at 12 dB per octave. Adding a third component steepens the roll-off slope to 18 dB per octave (third-order) and so on.

First-order filters are considered to be linear phase. Steeper filters cause increasing phase shift and more ringing (oscillation after the signal stops).

All the various mechanical parameters of loudspeakers can be modeled mathematically just like their electrical counterparts.

Subwoofer Equalization

The bass portion of a loudspeaker acts as a mechanical high-pass filter and can be



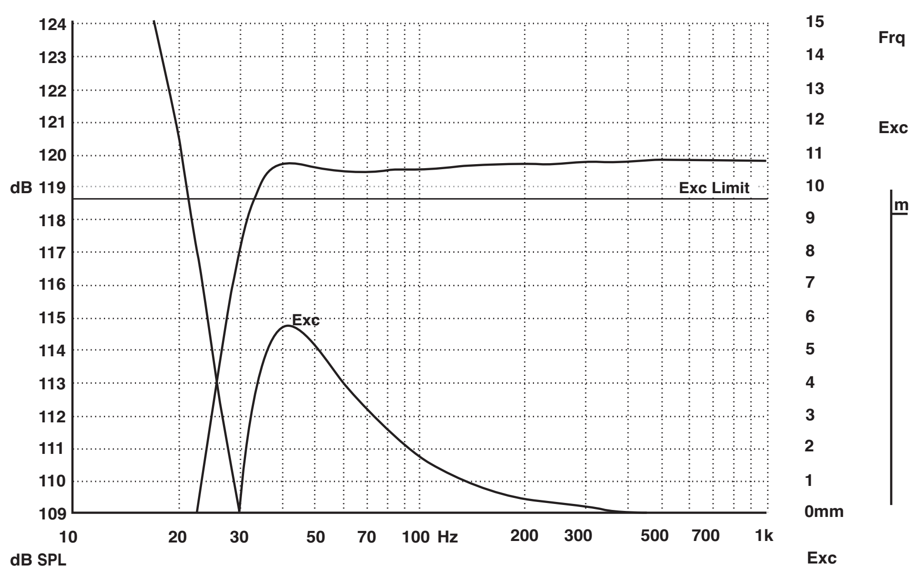
Driver And Subsystem Parameters

Driver Name:	JBL 2245H
Free Air Resonance	f(s) = 20Hz
Total Driver Q	Q(ts) = .27
Equiv. Volume	V(as) = 29 Cu. Ft.
Number of Drivers	N = 1SPL@ 1 m
Input Power (W)	P(in) = 300 (3.28 ft)

Fourth Order Vented Box

Box Vol. (Cu Ft.)	V(B) = 11.937
Closed Box Q	Q(tcA) = .5
Box Frequency (Hz)	F(B) = 30
Min. Vent Area	S(v)MIN = 45
Vent Area (Sq In)	S(v) = 50.3
Vent Length (In)	L(v) = 6.823

Performance Of JBL 2245H 18-Inch Driver
In A 12-Cubic Foot Vented Enclosure
Top: Graph Shows Frequency Response, Phase And Impedance
Bottom: Graph Shows Sound Pressure Level, And Driver Cone Excursion



modeled mathematically just like an electrical filter. Signals with a frequency above the lowest limits of the woofer system are passed, and signals below this cut-off point are not. A sealed box is a second-order filter and response falls predictably at about 12 dB per octave below cut off. This lends itself to simple electronic equalization with a compensating lift of 12dB per octave to extend low-frequency response.

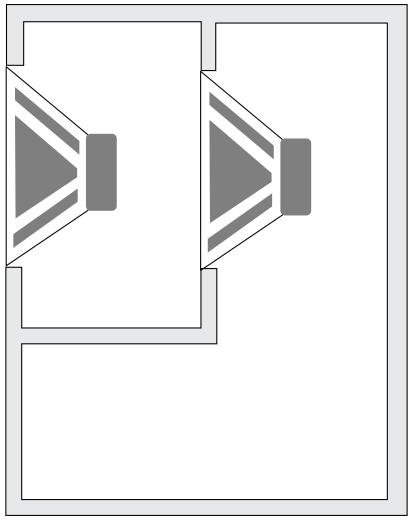
Using some form of electronic equalization is the most common technique for producing compact, powered subwoofers. The drawbacks of equalization include the need for high-powered amplifiers and the requirement for some sort of dynamic compression to prevent excess driver excursion on loud peaks at low frequencies. Servos and other feedback schemes work in a similar way; when they sense that output response is falling, they boost it back up electrically.

A vented (or ported) enclosure is a fourth-order high-pass filter. (Some "alignments" produce different slopes, but that discussion is beyond the scope of this article.) Response falls at 24dB per octave and transient response suffers. Just as with electrical filters, "the steeper the slope, the greater the ring." (Ringing is continued oscillation after the signal has stopped.)

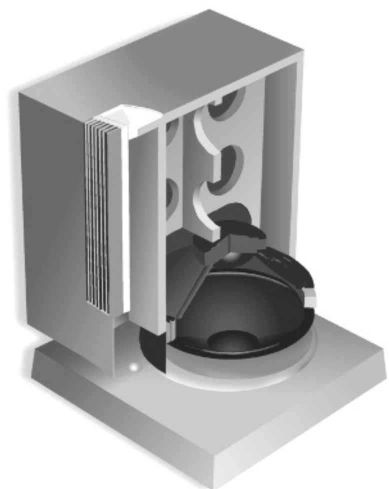
Assisted (equalized), vented alignments become sixth-(or more) order and transient response is degraded even more. A sixth-order slope is 36dB per octave.

Vented enclosures have the advantages of extended, flat frequency response for a given enclosure volume and greatly reduced cone excursion near the tuning frequency. Most inexpensive, passive (unamplified) subwoofers are vented.

There are a few powered, vented and EQed subwoofers available such as the Tannoy and the B&W models in the review.



Compound Or Isobarik® Loading



Linn Isobarik® Subwoofer

Compound Or Isobarik Systems

Compound or Isobarik systems use two drivers: one behind the other. The rear driver, which you can't see, is loaded by the enclosure and "fools" the front driver into thinking that it is in a much larger box.

The enclosure of a compound system can be half as large for the same cut-off frequency, or the enclosure can remain the same size and response can extend much lower than what could be achieved with a single driver system.

By carefully manipulating the space between the drivers, and therefore their coupling, the system can be made almost non-resonant in its pass band, producing very high-quality bass.

The two motors with only one radiating surface (only one of the diaphragms moves air into the room) are 3dB less efficient than a single driver system, but because the drivers are wired in parallel, impedance is halved and the voltage sensitivity remains about the same.

The major drawback of compound loading is cost. Isobarik enclosures are complex and difficult to manufacture, and driver expense doubles. The maximum sound-pressure levels attainable are the same as a single-driver system, but the price will be much higher for equivalent quality.

Bandpass Systems

Fourth-order bandpass loading places the driver(s) in the center of the enclosure with one side of the driver(s) loaded by a sealed box and the other side firing into a ported chamber. All the sound enters the room from the port.

The fourth-order bandpass system has some advantages: low bass from very compact enclosures, a second-order (or greater) roll-off at the top end, eliminating the need for an electrical low-pass filter and the muffling of high-frequency distortion by-products because the driver radiates indirectly into the room through the port.

Many-low priced subwoofers work this way. In fact, I'm listening to one that is connected to my computer as I write this.

Some Polk Audio subwoofers use a bandpass design with a unique and patented set of alignments. Bose holds a patent on band pass enclosures where both chambers are vented.

Poorly designed (sometimes on purpose) bandpass subwoofers have the potential to produce virtually one-note bass, so beware... and listen carefully.

The "Q" Factor

One of the most significant variable parameters that the loudspeaker designer has to work with is the Q factor. A subwoofer is a high-pass filter and the Q_{tc} (the total Q of the system) that the designer chooses as a goal, has a big effect on measured performance, and ultimately on what you hear. Q_{tc} defines the shape of the response curve and the amount of damping to overshoot or ringing (oscillation after the signal stops) that the system will provide.

A sealed enclosure with a Q_{tc} of .5 is considered a "critically damped" alignment with a step response that has no overshoot. For a given driver, a Q_{tc} of .5 requires the largest box.

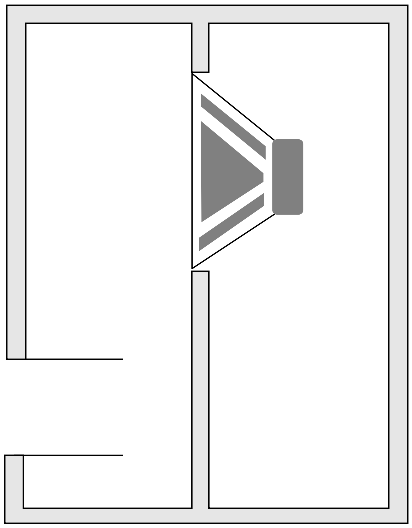
A Q_{tc} = .577 is a Bessel alignment and has the most linear phase response. When Q_{tc} = .707 we have a Butterworth alignment with the flattest amplitude response. Q_{tc} = 1.0 has a peaked response but gives us the smallest box still considered high fidelity.

Unfortunately, there are many subwoofers on the market with Qs much higher than 1.0! If you have any interest in music you should avoid these high-Q designs like the plague.

Why would a designer choose a high Q for his product? Two reasons: a smaller box and more boom. I did some calculations on a JBL 2235H, a popular 15-inch driver, to demonstrate the effects of Q on box size.

For a Q of .5, the net enclosure volume would be 5.4 cubic feet. A Q of .7 gives us a 2.37 cubic foot box. With a Q of 1.0, the box shrinks to 1.08 cubic feet, and at Q=1.2 the volume decreases to only .735 cubic feet. The results on frequency response are shown in the illustration. The results on damping must be heard to be appreciated.

To the novice listener, a peaked response with a lot of under-damped overhang sounds like *more* bass, and when you buy most cheap subwoofers that's exactly what you get. But bloated bass with a lot of overhang sounds "slow," and this can destroy rhythm and pace—and really ruin music.



Bandpass Loading

Choosing a subwoofer that produces a good balance between impressive sounding explosions and a nice, tight kick-drum will require some investigation and then some listening.

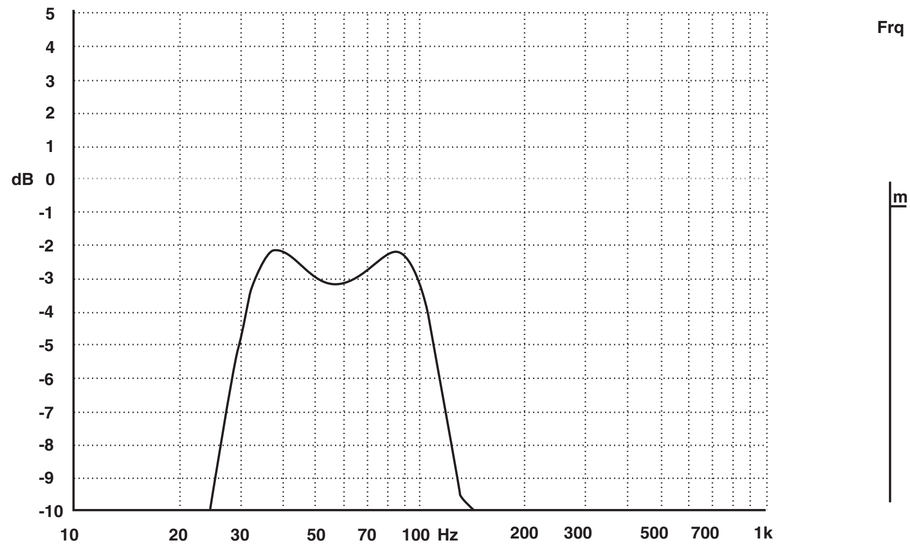
Excursion Is Not Necessarily A Trip

You can displace the same amount of air by moving a large diaphragm a short distance or by moving a smaller diaphragm a greater distance. In theory, a very small driver should be able to reproduce very low-frequency bass, and some can. Whether or not this low-frequency information will be loud enough to *hear* depends on how far the diaphragm of that small driver can travel (this is called excursion), and how that bass *sounds* depends on whether that excursion exceeds the *linear* range of travel for that driver.

Linear excursion is determined by the range of motion of the mechanical suspension parts of the driver and by a mysterious factor called X_{max}, which is calculated by subtracting the height of the magnetic gap from the length of the voice coil and dividing by two. The X_{max} dimension describes how far the diaphragm (cone) can travel in each direction without all or part of the voice coil leaving the magnetic gap.

When the voice coil leaves the magnetic gap the cone is driven solely by inertia, and any resemblance between its motion and the signal is purely coincidental. Then why doesn't everybody use very long voice coils for higher fidelity and lower distortion?

Long, overhung voice coils are more costly and they are inefficient because although current is flowing through the whole thing, only a small portion is in the magnetic gap producing motion, plus long voice coils are heavier. More powerful (and more expensive) amplifiers are required for a given level of output producing more heat.



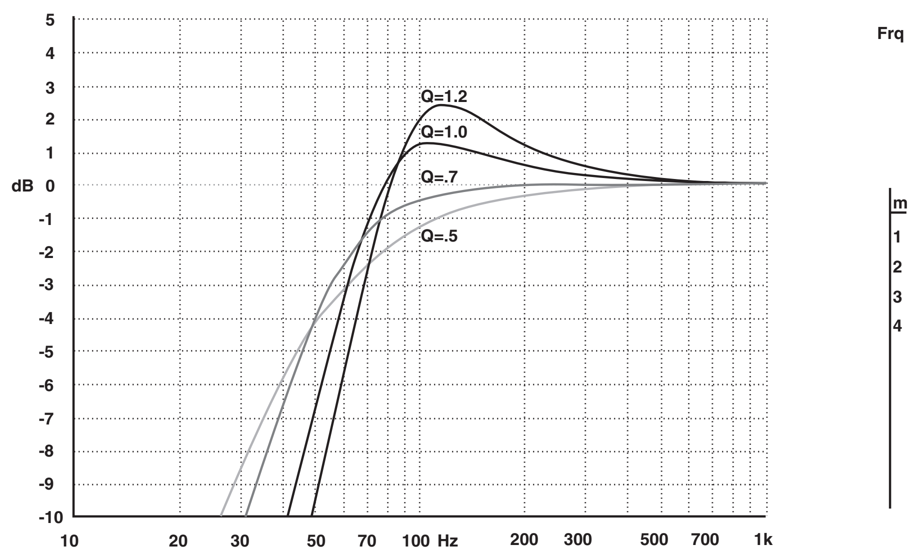
Driver and Subsystem Parameters

Driver Name:	Focal 8V416
Free Air Resonance	f(s) = 24.3Hz
Total Driver Q	Q(ts) = .34
Equiv. Volume	V(as) = 3.7787 Cu. Ft.
Number of Drivers	N = 1 SPL@ 1 m
Input Power (W)	P(in) = 150 (3.28 ft)

Fourth Order BandPass Box

Box Vol. (Cu Ft.)	V(B) = .8329
Closed Box Q	Q(tcA) = .8
Front Box (Cu Ft)	V(B2) = .4761
Front Box Frq (Hz)	F(B2) = 57

Symmetrical Bandpass Frequency Response



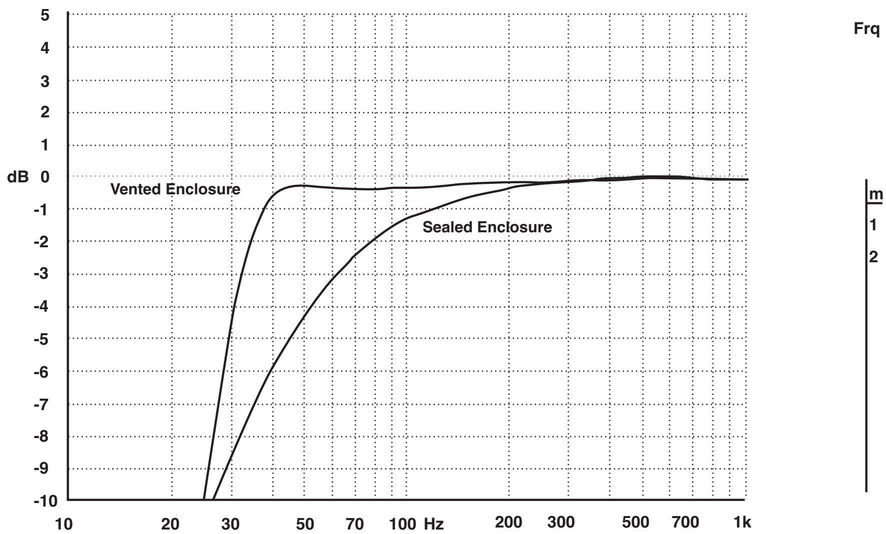
JBL 2235H 15-Inch Driver
In Sealed Enclosure — Q_{tc} Versus Frequency Response

Excursion limits, thermal limits, amplifier cost and size—are another set of compromises that the designer must deal with.

Will One Subwoofer Do?

Many electronic components feature a single subwoofer output with a mono signal that is the sum of the bass from the left and right channels and maybe the center and surround channels as well.

Dolby® Digital and DTS® Digital Surround™ [as well as the Dolby TrueHD and DTS-HD Master Audio] processors can redirect the bass from any or all channels to a single subwoofer. In theory, this is acceptable because low frequencies are considered by some people to be non-directional. Reality



Driver and Subsystem Parameters

Driver Name:	JBL 2235H
Free Air Resonance	f(s) = 20Hz
Total Driver Q	Q(ts) = .25
Equiv. Volume	V(as) = 16.2 Cu. Ft.
Number of Drivers	N = 1 SPL@ 1 m
Input Power (W)	P(in) = 150 (3.28 ft)

Second Order Closed Box

Box Vol. (Cu Ft.)	V(B) = 5.4
Closed Box Q	Q(tc) = .5
Closed Box Res.	F(sc) = 40
Compliance Ratio	alpha = 45

Comparison Of Frequency Response Of JBL 2235H Driver In 5.4 Cubic Foot Enclosure—Sealed Versus Vented Enclosure

frequently differs from theory because things just aren't as perfect as they should be.

Subwoofers shouldn't reproduce signals above the frequency of their crossovers, but crossovers aren't brick walls—they reduce the output level at a sloping rate. A 12 dB per octave low-pass filter (like a subwoofer crossover) gradually reduces output until it is 12 dB down in level after one octave of increasing frequency.

One octave above 80 Hz is 160 Hz, and 160 Hz is directional to most listeners. The average tenor voice can sing lower than that. Can you imagine a tenor singing in your room and you not being able to tell where he was with your eyes closed?

Cones and enclosures produce side band distortion and resonances. A 60 Hz signal will make some sound at 120 Hz whether you want it to or not.

There is some disagreement about this, but I believe that there is important phase information at low frequencies. In theory, all notes start at DC. In my humble opinion, a system should have no fewer than two subwoofers operating in *stereo*, not from a mono subwoofer output. A 5.1- [or 7.1-] channel system needs at least three so that there is a dedicated subwoofer for the low-frequency effects (LFE) channel. And ideally, if you have enough space and money, there should be five subwoofers: one for each primary channel plus one for LFE.

Here's an experiment you can perform for yourself. Visit a specialty dealer and have them demonstrate a complete system with a single, properly set up powered subwoofer. Have the salesperson turn off all the loudspeakers except the subwoofer and replay the same material. See if you can tell where the bass is coming from. If you can, you need stereo subwoofers.

Another factor to consider when deciding on the number of subwoofers that you need is your desired maximum sound pressure level (SPL). Two subwoofers will play much louder than one, and three will play louder than two. You will have to put more energy into a larger room, or a "deader" or more absorbent room, to achieve the same perceived loudness.

How loud do you play your system? How big and how absorbent is your room?

To THX® Or Not To THX

Some of the subwoofers in this publication are THX®-certified and some are not. What's the difference? Well, LucasFilm has established a set of arbitrary specifications for subwoofers and they charge manufacturers a licensing fee to certify that their products meet these specifications.

The problem is, they won't tell you what these specifications are, and they require the manufacturers to sign a non-disclosure agreement so *they* can't tell you either. In effect, LucasFilm is saying "we know what's best for you, so just pay more for products with our logo on them and you can be assured that those products meet *our* standards."

Well, here's my opinion (and this is not necessarily the opinion of the publisher): if you are shopping at a box-trading, mass-merchandiser where you can't get professional help and can't listen to what you're buying, then THX certifications on all your components will assure you that everything you buy will work together as a system. But this type of program has no value in the high-end. You shouldn't necessarily avoid THX certified products, you should just carefully evaluate them to make sure that they meet *your* standards.

Room Gain

Subwoofers are typically measured in "half-space." A driver mounted in the middle of one wall of your listening room, away from the side walls and away from both the ceiling and the floor would be radiating into a half-space at some frequencies; 180° in one plane and 180° in the other. If you moved this driver down near the junction of the wall and the floor, there would be an increase in output. If you moved the driver near the junctions of the wall it's mounted on plus the floor plus the side wall (in a corner, near the floor), there would be an additional increase in output.

This increasing output from placement approaching more room boundaries would be fairly linear over a range of frequencies down to about 30Hz in most rooms. As the frequency being reproduced gets lower, and wavelengths get longer relative to the room dimensions, things change. At some point, the subwoofer will actually be modulating the air pressure in the room (as it would in a car).

A very interesting paper entitled "Subwoofer Performance For Accurate Reproduction Of Music" was presented at the 83rd Convention of the Audio Engineering Society in New York during October 1987, by Louis D. Fielder of Dolby Laboratories and Eric M. Benjamin. They measured nine different combinations of subwoofer placement in five different "average" rooms and compared the response at the listening position in the rooms to the response of the test subwoofer measured in a half-space environment at one meter. (An average room was described as about 21-inch by 14-inch by 8-inch, or 2350 cubic feet.) They found that the average level in the room from 100Hz down to 30Hz was approximately 3dB less than the one meter reference sensitivity for the test subwoofer. There was a monotonic increase in level for the frequency range between 30Hz and 5Hz with a total increase of 15dB. This was more than what was expected based on previous studies.

Based partly on this study—and an earlier one by R. V. Waterhouse—many designers believe that a second order roll-off below 30Hz is necessary for flat in-room response down to infrasonic frequencies, and a 16Hz to 20Hz cut off is sufficient for reproduction down to 10Hz in a room.

I think THX adheres to this concept, but of course, they won't tell me. Ken Kriesel of M&K subscribes to the idea, and Jim Wischmeyer of Bag End does not, but products from both designers sound like they go all the way down to subsonic territories without emphasis in my room.

Other than subharmonics, there is little recorded information below 30Hz anyway, so let your ears be your guide.

Blending Subwoofer(s) With The Main Loudspeakers

Getting a seamless blend between the subwoofer(s) and the main loudspeakers can be a difficult and time-consuming task. An experienced and skillful dealer can be a big help with this chore, but for perfect personal satisfaction, a lot of listening and trial and error testing will be necessary. What follows is basic advice that is based on my personal experiences to get your started.

A crossover point from the subwoofer(s) to the main loudspeakers of 80 Hz is about as high as you can go without seriously compromising the fidelity of the midrange. Crossing over above this frequency also makes the position of the subwoofer(s) more noticeable which is undesirable. If your main loudspeakers are truly full range, crossing over at 50 Hz or 60 Hz may sound better. You will lose some dynamic range, however, because the main loudspeakers will still be reproducing some of the low-frequency information, but the important midrange frequencies will be altered less with a very low-crossover frequency.

The main loudspeakers should have solid response down to 60 Hz or 65 Hz for an 80 Hz crossover. Some people think they should extend an octave below the crossover point, or down to 40 Hz. Blending a subwoofer with mini loudspeakers that have low-frequency limits of 100 to 125 Hz is commonly done but the results are never satisfactory in high-fidelity terms. In my opinion this can not be done successfully, but you may disagree.

Steep slope high-pass filters will cause phase shift in your main loudspeakers. If your main loudspeakers are a time- and phase-accurate design, a single pole, passive high-pass filter or no filter at all are your only choices to maintain phase integrity.

Listening

How should you listen when you are evaluating subwoofers? All the subwoofers in this issue will do a credible job of reproducing the sound of a bomb blowing up a car. The major difference between them will be how loud the bomb can be without sounding unduly distorted, and how deep and visceral the sound effect is.

Most of us have, fortunately, not had the opportunity to actually hear a car explode—and even if you have, the sound the subwoofer is reproducing off the film soundtrack is almost assuredly fake. We do, however, have some reference standards with which we are all familiar: the sounds of real musical instruments and the human voice. These sounds can also be found on soundtracks or on music CDs. You do need to find sources that include bass information that is lower in frequency than the crossover point so that you can hear the subwoofer in action and see how well it blends with the main loudspeakers.

Organ pedal tones go very low in frequency. In a standard symphony orchestra, the piano goes the lowest. The low A on the piano is 27.5 Hz. Harps can go down to a low C at 32.7 Hz. String Bass starts at the E above that low C or 41.2 Hz.

In most music, the bass player plays a tune that harmonizes with the melody. The easier it is to follow that tune, the better the job that the subwoofer is doing. Bass and percussion set the rhythm and pace of music. The easier it is to tap your foot to the music and respond to the rhythm, the better the subwoofer.

While this may sound somewhat mystical, once you start comparing subwoofers using music, you'll find it easy to hear the differences.

Set Up

Subwoofers reproduce frequencies with wavelengths that are long in relation to the dimensions of most living rooms. Standing waves may boost or cancel various frequencies at the listening position due to the interaction with room boundaries. Moving a single subwoofer just a little bit in a room may drastically change what you hear at the listening position. Using two widely spaced subwoofers will alleviate most of these problems.

There is a quick and dirty method that can be used to establish a starting position for a single subwoofer. Place the subwoofer at your usual listening position (that's where you usually sit, but you can move the chair). With a signal generator or a processor like the Meridian with a built-in tone generator, or a CD with test tones, play a 50 Hz tone, a 40 Hz to 50 Hz warble tone, or filtered pink noise centered somewhere in that range at a medium listening level. Walk around the area where a subwoofer could be placed and listen. When you find a spot where the signal sounds the loudest and smoothest, mark it. Place the subwoofer on that spot and then play real musical material while fine-tuning the position.

To set phase, play a tone at the crossover frequency and measure the sound-pressure level with an SPL meter. Leave the volume setting the same, switch the phase of the subwoofer, and measure again. The position where the level is the highest is probably correct.

Set the volume level of the subwoofer by measuring signal levels above and below the crossover point and matching the two, then fine-tune by ear while playing a variety of material. I use solo piano recordings to adjust subwoofer levels by ear, but you may prefer something different. Final tweaking should always be done by ear. In real life, test instruments aren't going to listen to your system, *you* are.

The Review

This has turned out to be a major project. Subwoofers are heavy! (But if I can get subwoofer-lifting certified as an Olympic sport, I will have a jump on the competition.) Each product interfaces with the room in a slightly different way, so I had to experiment with placement with each and every one. Matching the subwoofers to the main loudspeakers required more experimenting with crossover points, crossover slopes, and adjusting levels and phase to achieve the best blend.

I tried to get the best performance possible from each subwoofer in every review, but I'm sure that I could have done even better if there had been more time. If I had spent a month tinkering with each product, the series would have taken over three years. [When I originally wrote this no one could have predicted that this review series would actually take more than three years to complete.]

The subjective evaluations are my impressions under the conditions in my home, which incidentally, are very good for bass. My room has been set up by a combination of computer analysis and extensive listening tests. I tuned the low-frequency performance of the room with ASC Tube Traps and furniture positioning to achieve very smooth and extended bass response with virtually no "boom." Your results may vary. One thing is for sure; achieving a perfect setup in your room will require some time and experimentation, but the results will be well worth the effort. And besides, tinkering with loudspeakers is fun.

During testing I used a signal generator to input sine waves into each subwoofer for break-in and to judge the amount of resonant noise emitted from the enclosures. Sine wave stimulus made it very easy to hear the results as each product was momentarily overdriven.

After achieving a good room setup with each unit, I used a variety of signals and instruments to gauge the approximate “real world” in-room maximum sound-pressure level capability before undue distress. “Undue distress” is when I heard audible distortion or strain. Maximum SPL is the level attainable at the listening position; about nine feet from each subwoofer.

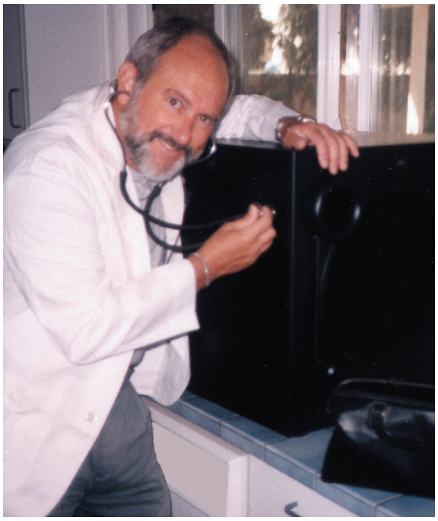
For the listening evaluations I used a variety of music recordings on CD, including but not limited to the following: Bill Elgart *Percussion* (the best solo drum recording I have ever heard), Myrtle Regier *Organ*, featuring J. S. Bach’s Schubler Chorales (you can hear the organ breathe), both made by Mark Levinson in the ’70s and available from Cello Acoustic Recordings in Hamden Connecticut. *Doug MacLeod* from AudioQuest Music (great for rhythm and pace), *Mark Murphy Sings Nat’s Choice* from Muse Records (simply great music with powerful and tuneful bass), *Nojima Plays Liszt* from Reference Recordings (fabulous solo piano recording of music that I wish I could play); classic bass test records like Enya’s *Watermark* and Bela Fleck’s *Flight Of The Cosmic Hippo* and selections from the *Stereophile* and Sheffield test discs.

After trying each subwoofer as a musical bass reproducer, I hooked them up one at a time to the .1 low-frequency effects (LFE) channel of my Dolby Digital system and played sound effects from Dolby Digital LaserDiscs at the standard +10dB level. LaserDisc selections included *True Lies, Outbreak*, *Unforgettable, Heat, Broken Arrow, Dead Presidents, Twister,* and *Executive Decision*. [Many more music titles and film soundtracks were added as the reviews progressed. DVDs weren’t even available when the review series began.]

Each product was then given a subjective rating in each of the following categories: overall music reproduction, overall sound effects reproduction, impact, tonal definition, freedom from overhang, ability to follow the rhythm and pace of a musical selection, midrange coloration, box integrity (freedom from enclosure resonance), distress (can the product be easily overdriven?), subjective deep bass response, and measured ability to play at 105 dB @ 35 Hz in the room-listening position; measured ability to play 25 Hz without substantial reduction in SPL.

A subjective rating of “5” means that this aspect of performance was excellent. A rating of “4” indicates good performance. A rating of “3” means average performance, a “2” indicates below average or poor performance, and a “1” is unacceptable performance and not recommended.

The individual reviews of each product include descriptions of appearance, features, and technical details, plus the subjective ratings from the listening tests. The prices were current when I wrote this but are subject to change. For additional information about specific products, contact the manufacturers at the numbers listed. ■



Dr. "Boom" Examines Another Patient

Introduction Part Two

(Originally Appeared In Issue 26)

More Subwoofers

In Issue 24, we published the first part of the most extensive—and we hope the most comprehensive—subwoofer review ever attempted by a magazine of this type.

Part One of the article defined many of the terms used in the reviews and discussed in some detail, the design compromises with which each subwoofer manufacturer must cope, due to the ever imposing "laws of physics." Awareness of these necessary design trade-offs can make you a more informed consumer and help you make better purchasing decisions, plus becoming familiar with the terminology will make the product descriptions easier to understand.

Fantasy Versus Reality

After the first article was published, I got a lot of calls and letters. The most frequently asked question from readers had to do with my crude, in-room SPL measurements versus the manufacturers' specifications. The questions typically started like this: "Manufacturer X says his product goes down to 12 Hz flat, but your measurements show that it starts to roll-off below 25 Hz. Is he just lying?" Or: "My subwoofer is supposed to produce 117 dB at 14 Hz, but you say it can't. What's the deal?" And then there's my favorite: "Brand X says that the laws of physics only inhibit other manufacturers, but they have a special technique to do what those others think is impossible to accomplish. Is their product really transcendent?"

It's apparent that some further clarification is in order here, so let's talk some more about the difficulty of engineering a device to make loud, low bass in a home entertainment system and the many different ways of measuring the results.

Theoretical Versus "Real World" Performance

Product spec sheets seldom fully describe the testing methods used in producing the published specifications. Some manufacturers don't really measure the finished products at all—they rely on the computer program used in the design process to produce predicted response curves, which are then *augmented* by the *expected* support from the room and used as performance specifications.

Computer programs used for loudspeaker design usually generate "predicted response curves" for a free-field, anechoic environment around the loudspeaker. That means the response that would be achieved when the loudspeaker was placed well away from any room boundary and in a room with no acoustically reflective surfaces. Although some makers measure subwoofers this way, most use half-space measurements, still others use quarter space to more accurately reflect the predicted response of the product as it will actually be used in a home. Note that "free-field" means no boundary support. Picture a loudspeaker suspended in air with no walls anywhere around it, so that it is radiating into 360° in both planes (horizontal and vertical). "Half space" means this radiation pattern is divided in half—picture the same loudspeaker with a wall behind it. For quarter space, imagine the loudspeaker sitting on the floor and against a wall.

Corner placement will produce even more boost at low frequencies, and that is the position in which most subwoofers will actually be used, so isn't it realistic to specify your product's expected response when positioned in the corner of an average room? Maybe—but what's an average room?

Virtually all of the subwoofer manufacturers are relying on the room to "boost" the low-bass response of the device to levels that could not be obtained in a free-field environment.

Although it is a realistic assumption to count on this room-boost to meet your specifications, it means that the actual results achieved will vary tremendously depending on the size and construction of the room in which the product is used, as well as the location of the product in that room.

Assuming a boost from the room also allows a manufacturer to be very optimistic about his expected in-room performance capability, *you* can assume that any specification that includes small deviation numbers like "Frequency Response = 15 Hz-80 Hz + 0/-3 dB," is *not* measured, in-room response. At least not measured in a room like *your* room.

I live in a two-story California tract house with a concrete slab foundation and half-inch drywall on two-by-four construction. All the rooms have large openings into other rooms for that "open, airy look." This wide open and relatively flimsy construction makes pressurization difficult and limits real-low bass extension, and I'll bet your house is similar—unless you live in another part of the world where construction techniques and average room size can be very different.

My measurements are made with the subwoofer under test placed where it would actually be used in the room. I use a signal generator (plus one-third octave filtered pink noise) and a calibrated sound level meter to measure the maximum sound pressure level obtainable at the specified frequency *at the listening position,* before the subwoofer reaches a level where I begin to hear audible distortion. My listening position has been carefully optimized by computer analysis, experimentation, and room treatment to minimize standing wave interaction. This is a simple, realistic and useful measurement, and I will tell you why I do it.

Flat Versus Maximum SPL

There is a big difference between "flat frequency response" and maximum sound pressure level capability. There is a difference between the real-world performance demands made on full-range loudspeakers and those required from subwoofers.

Full-range loudspeakers are typically measured for amplitude response linearity at modest sound levels. This measurement technique reflects actual performance capability because most sounds in music (and movie dialogue) are not very loud and don't require much power for accurate reproduction under normal listening conditions. A loudspeaker with a sensitivity of 90 dB @ 1 watt @ 1 meter, would only need a couple of watts to play almost any material in the mid-frequency range at normal listening levels. A few watts more would reproduce almost any peak in the midrange.

If you plotted a full-range loudspeaker's response curve at maximum undistorted output at each frequency, it would probably not appear very "flat" because power and excursion demands increase exponentially at low frequencies (and thermal problems would limit sustained output at high frequencies).

In music, the highest note on the piano, a C, has a fundamental frequency of 4186 Hz. This is the highest note there is. All frequencies above 4186 Hz are called harmonic structures or "overtones." These harmonics are very important. Without them, all instruments playing the same note would sound alike.

Overtones give each instrument its own unique sound, even though they can all play the same notes at the same fundamental frequencies. Harmonics are very important for accurate sound reproduction, but they are created by the fundamental and are lower in amplitude. Generally, the higher the harmonic, the lower the amplitude, which works out really well for loudspeakers.

Tweeters need to be fast to reproduce high frequencies. Usually they are light, delicate devices that can't handle as much power as woofers or midrange drivers. Because tweeters are reproducing mostly harmonics (overtones), which are lower in amplitude than fundamental signals in the midrange, a tweeter with a thermal power rating of 20 watts works just fine in a full-range loudspeaker that can handle 150 watts or more when playing music. There aren't any rules about sound effects or synthesized sounds, but high-frequency sound effects at high levels can be painful, so the levels and the duration of non-musical, high-frequency sounds are usually limited. Most full-range loudspeakers, even small ones, don't have a problem producing high-frequency harmonics at levels that match midrange peaks.

Low bass is another story. A full-range loudspeaker that can accurately reproduce the midrange is going to have increasing difficulty as the frequency of the signal gets lower. Subwoofers were invented to allow the low-frequency range to be reproduced at higher sound-pressure levels with less distortion; to more accurately track midrange levels on dynamic music material and movie soundtracks. It's very possible to make a 6-inch woofer reproduce 20Hz, but it's unlikely that it could do so at a level that you could hear, because…

Your Ears Aren't Linear

The studies done by Fletcher and Munson of Bell Telephone Laboratories and others since show that the human ear is far less sensitive to high and low frequencies than it is to sounds in the midrange.

The Threshold Of Hearing is the sound level that the average person can just hear 50 percent of the time. According to Fletcher and Munson, a sound at 20 Hz would have to be almost 80 dB louder than a sound at 1,000 Hz, just to reach the threshold of hearing. The *ambient noise level* in a quiet home is about 40 dB above the threshold of hearing. As you can see from these numbers, if you want to be able to *hear* signals in the first octave (from 16 Hz to 32 Hz), these signals have to be pretty loud.

A subwoofer may have flat amplitude response to very low frequencies when measured in the way that a full-range loudspeaker is measured—with low-level input signals. But how loud it will play at very low frequencies is far more relevant to what you are going to hear in actual use. That's why I measure how loud each subwoofer will play at low frequencies in a real room...at the listening position.

These real-world measurements, though somewhat crude, give some idea of the product's response curve and show what use it's biased toward (music or effects), plus they tell you whether one unit will be sufficient for home theatre levels or if two or more will be needed.

My measurements of all the subwoofers in this review were done in the same way, in the same room. They can be compared to each other, but are not directly comparable to other measurements made by different methods in different acoustical situations. The purpose of the measurements is to give you some idea of the goals of the designer, so that you can decide if his product is suitable for your consideration.

Wishful Thinking About Sound Pressure Level

If someone tries to sell you a subwoofer using a single 10- or 12-inch driver with the claim that it will play at 120 dB at 14 Hz, (or anywhere near this loud at anywhere near this frequency), be suspicious—very suspicious. EQ alone (or in combination with other tricks) cannot accomplish this feat. You would need tremendous cone area, very high-power handling capability, and/or very long excursion capability to produce this kind of output. Look at the following illustrations.

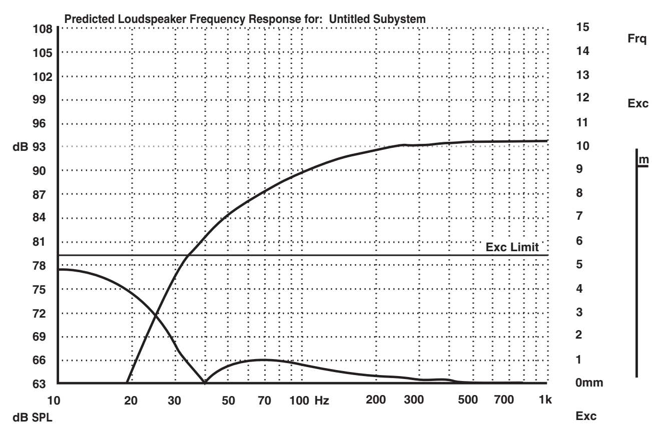
Illustration 1 shows the Focal 5K011, 5-inch driver in a mythical vented enclosure. This tiny unit will produce 20 Hz at 63 dB (30 dB down from its midrange capability). The three-watt input shown is just about the maximum that could be applied at low frequencies without exceeding the driver's linear excursion capabilities and raising distortion to high levels. This is very low bass from a very small driver, but is it loud enough to be useful?

If you insert a simple circuit to attenuate the upper portion of the frequency response down to 69 dB starting at 25 Hz, this loudspeaker would measure flat down to 20 Hz (-6 dB), but its maximum output would still be 63 dB at 20 Hz and only 69 dB at 50 Hz! (This is just an example, not an engineering exercise).

Applying electronic equalization to raise the level of low frequencies would have the same effect as increasing the input power—linear excursion capability would be exceeded, so no additional output would be realized without raising distortion to unacceptable levels.

Illustration 2 shows the popular Dynaudio 30W54, 12-inch woofer in a 3.2 cubic foot sealed enclosure. This driver will handle 210 watts (thermally) and produce output levels up to 115 dB at frequencies above 70 Hz, with rapidly rising distortion below this frequency because its maximum linear excursion limits will be exceeded. Of course, it will play at lower frequencies at lower levels with low distortion by reducing input power, but you get the idea. The cone area and how far it moves are the final determinates for output level. No trick can alter these facts. You can't fool Mother Nature, just gullible people.

Although Mother Nature has set physical limits on ultimate performance, She is open to negotiation. A designer is allowed to trade



Driver Parameters

Driver: Focal 5K011

Nominal Diameter	D = 5.35432 Inches
Free Air Resonance	f(s) = 54.7Hz
Total Driver Q	Q(ts) = .298
Driver Electrical Q	Q(es) = .319
Driver Mechanical Q	Q(ms) = 4.506
Equiv. Volume	V(as) = .420248 cubic ft.
Nominal Impedance	Z(nom) = 8 Ω
DC Resistance	R(e) = 6.5 Ω
Max Thermal Power	P(t) = 40 Watts
Peak Excursion	X(max) = .214566 Inches
Piston Area	S(d) = 13.299 Sq. Inches
Efficiency	n(0) = .5 %
Sensitivity (1W/1m)	SPL = 89 dB
Voice Coil Inductance	L(e) = 0 mH
Magnet Weight	Magnet = 0 lbs.
Voice Coil Diam.	Diam.(vc) = 1.00393 Inches
BL Product	BL = 7.18 T * m
Flux Density	Flux = 0 Gauss
Dynamic Mass	M(as) = .416 (Kg)
Voice Coil Length	L(vc) = .51181 inches
Magnetic Gap Length	L(gap) = .23622 inches

Box Parameters

Box Type: Fourth Order Vented Box

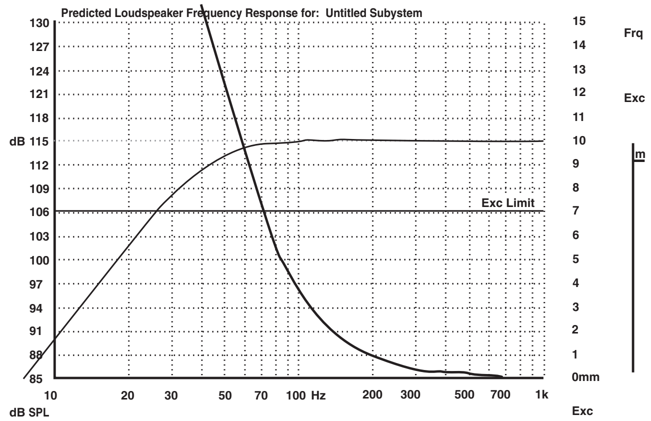
Box Volume	V(B) = 0.231 cubic ft.
Box Frequency	F(B) = 40.00Hz
Min Rec Vent Area	SVMin = 2.2 Sq Inches
Vent Surface Area	S(v) = 0.10 Sq Inches
Vent Length	L(v) = 0.500 Inches
Box Loss Q	Q(B) = 7.00

Subsystem Parameters

Subsystem Name: Untitled Subsystem

Number of Drivers	N = 1
Closed Box Q	Q(tc) = 0.500
Compliance Ratio	alpha = 1.81
Subsystem Input Power	P(in) = 3.0
SPL Distance	D = 1.00 m (3.28 ft)

Illustration 1 Focal 5K011



Driver Parameters

Driver: Dynaudio 30 W-54

Nominal Diameter	D = 12 Inches
Free Air Resonance	f(s) = 22Hz
Total Driver Q	Q(ts) = .357
Driver Electrical Q	Q(es) = .42
Driver Mechanical Q	Q(ms) = 2.39
Equiv. Volume	V(as) = 9.08 cubic ft.
Nominal Impedance	Z(nom) = 8 Ω
DC Resistance	R(e) = 6.25 Ω
Max Thermal Power	P(t) = 210 Watts
Peak Excursion	X(max) = .276 Inches
Piston Area	S(d) = 62 Sq. Inches
Efficiency	n(0) = .977 %
Sensitivity (1W/1m)	SPL = 92 dB
Voice Coil Inductance	L(e) = .25 mH
Magnet Weight	Magnet = 0 lbs.
Voice Coil Diam.	Diam.(vc) = 2.13 Inches
BL Product	BL = 9.44 T * m
Flux Density	Flux = 0 Gauss
Dynamic Mass	M(as) = .0416 (Kg)
Voice Coil Length	L(vc) = .669 inches
Magnetic Gap Length	L(gap) = .394 inches

Note: Voice coil length and gap length are in inches.

Box Parameters

Box Type: Second Order Closed Box

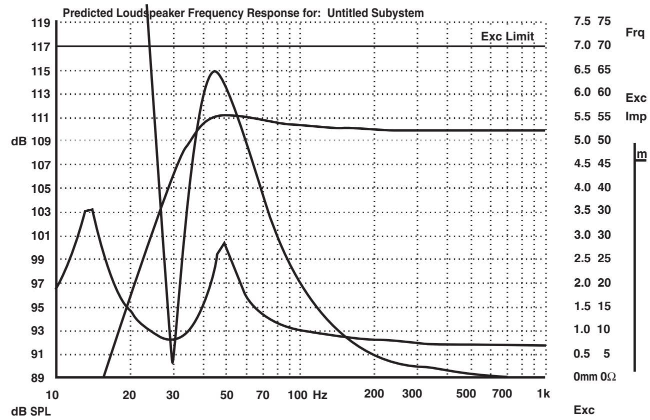
Box Volume	V(B) = 3.192 cubic ft.

Subsystem Parameters

Subsystem Name: Untitled Subsystem

Number of Drivers	N = 1
Closed Box Q	Q(tc) = 0.700
Compliance Ratio	alpha = 2.84
Subsystem Input Power	P(in) = 210.0
SPL Distance	D = 1.00 m (3.28 ft)

Illustration 2, Dynaudio 30W54, Sealed Enclosure



Driver Parameters

Driver: Dynaudio 30 W-54

Nominal Diameter	D = 12 Inches
Free Air Resonance	f(s) = 22Hz
Total Driver Q	Q(ts) = .357
Driver Electrical Q	Q(es) = .42
Driver Mechanical Q	Q(ms) = 2.39
Equiv. Volume	V(as) = 9.08 cubic ft.
Nominal Impedance	Z(nom) = 8 Ω
DC Resistance	R(e) = 6.26 Ω
Max Thermal Power	P(t) = 210 Watts
Peak Excursion	X(max) = .276 Inches
Piston Area	S(d) = 62 Sq. Inches
Efficiency	n(0) = .977 %
Sensitivity (1W/1m)	SPL = 92 dB
Voice Coil Inductance	L(e) = .25 mH
Magnet Weight	Magnet = 0 lbs.
Voice Coil Diam.	Diam.(vc) = 2.13 Inches
BL Product	BL = 9.44 T * m
Flux Density	Flux = 0 Gauss
Dynamic Mass	M(as) = (Kg)
Voice Coil Length	L(vc) = .669 inches
Magnetic Gap Length	L(gap) = .394 inches

Note: Voice coil length and gap length are in inches.

Box Parameters

Box Type: Fourth Order Vented Box

Box Volume	V(B) = 4.000 cubic ft.
Box Frequency	F(B) = 30.00Hz
Min Rec Vent Area	SVMin = 10.2 Sq Inches
Vent Surface Area	S(v) = 14.10 Sq Inches
Vent Length	L(v) = 7.502 Inches
Box Loss Q	Q(B) = 7.00

Subsystem Parameters

Subsystem Name: Untitled Subsystem

Number of Drivers	N = 1
Closed Box Q	Q(tc) = 0.645
Compliance Ratio	alpha = 2.27
Subsystem Input Power	P(in) = 60.0
SPL Distance	D = 1.00 m (3.38 ft)

Illustration 3, Dynaudio 30W54, Vented Enclosure

diminished performance in one area for increased performance in another. Let's take a look at some…

Trade-Offs

A fourth-order vented alignment can reduce cone excursion at the enclosure tuning frequency and allow the same driver to play louder at low frequencies.

In Illustration 3, the response of the same 12-inch Dynaudio 30W54 is shown in a four cubic foot vented enclosure tuned to 30 Hz with two 3-inch ports measuring 7.5-inches long. With this combination we can get 109 dB at 35 Hz, and 102 dB at 25 Hz without exceeding the linear excursion limits of the driver—although we would have to reduce input power to 60 watts.

What are the drawbacks of this approach? The trade-offs are steeper roll-off below the cut-off frequency, diminished transient response, resulting in more oscillation or "ring" after the signal stops and increased sensitivity to infrasonic noise.

Adding equalization will make the alignment go to sixth (or more) order and further depreciate transient response, but it can extend the low-frequency limit of the system. Would another method be better? How about using a servo feedback system to "instantaneously" correct errors in cone movement?

A servo feedback system compares the output to the input in some way. You can mount an accelerometer or a position sensor on the cone or voice coil of the driver and compare the signal that's coming out with the signal that's going in to see if they are the same. (There are many other ways of doing this, but all compare the output to the input in some manner). If there is a deviation, the reciprocal of the error is injected at the input of the amplifier to correct the fault. This works well to correct response errors and reduce distortion on steady-state signals, but what about transients; particularly those that spike in just one direction from the DC reference line?

In order for a deviation to be sensed by a servo mechanism, it must occur. Once it has occurred, it is in the room with you even if the problem is corrected in the very next instant. The next instant may not be soon enough to prevent audible time smear on transient material, which can negatively affect rhythm and pace.

We discovered years ago that negative feedback in amplifiers was okay if used sparingly but sounded bad when used to excess—even if steady-state distortion measurements were improved. I think the same is true with subwoofers.

A feed-forward servo is another matter. With feed-forward systems, the input signal is analyzed to determine if it will produce a distorted output if allowed to pass unaltered. If a signal is too high in level, at too low a frequency, for example, it must be dealt with, or distorted output will result. Correction techniques can include dynamic changes to the electrical parameters of the system, as in the Vandersteen subwoofers, or simply a reduction of the level of the input signal. Correction happens before the error is actually produced mechanically.

Feed-forward schemes work well in preventing audible overload distortion and have few sonic drawbacks, but they can't correct for a deficiency in basic product design, and nothing changes the fact that you must move a certain surface area a certain distance in order to produce a given output level.

What's Really Down There?

Although 20 Hz is generally considered the lower limit of human hearing, many people can hear signals below this frequency. I can. Frequencies below 15 Hz are universally considered to be infrasonic.

Large pipe organs can play tones with fundamental frequencies below 20Hz, but they are the only acoustic instruments that can. The first note on an 88-key piano is an A at 27.5 Hz. (I've played the piano for almost 50 years and have never used the lowest key.) No other instrument in a symphony orchestra can go this low. The lowest note the Bass can play is an E at 41.2 Hz. The lower limit of the Bass Tuba is the same. The Contrabass Sarrussophone can play a half tone lower, but who's ever heard one of those?

Low-frequency special effect sounds are

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NOTE: iOS subscriptions do not include webzine full-access passes and are completely separate from web and print subscriptions. Subscribe to the print edition for full web-access passes by calling 877 398 3367 usually in the 40 Hz to 60 Hz region because that's where they have maximum impact. Most anything that occurs below 30 Hz is likely to be a subharmonic, or just noise.

Even if a subwoofer could produce high-level sound at 10 Hz, do you need that kind of frequency extension for satisfactory performance in your home theatre system? Since almost nobody really has it, and there are a lot of happy people out there, the answer is probably not. But only you can decide. The argument for subwoofers with bandwidth below what can generally be heard is that they can more accurately reproduce *audible* signals with less phase shift than those products with cut-off points within the range that you *can* hear.

Do the subwoofers with the greatest output at the lowest frequencies sound better in the range where most signals actually occur? Go audition some and find out. It's fun.

More Evaluation Material

Here are a few more software selections you might want to try when evaluating subwoofers. Music CDs: Sarah McLachlan's *Fumbling Towards Ecstasy*, and Rob Wasserman, *Trios*... try "Dustin' Off The Bass" and "Spikes Blues." For DVDs and LaserDiscs try: *Independence Day*, *Se7en* and *Heat*—the shoot-out after the bank robbery is fantastic. ■

My advice is very simple: put the subwoofer(s) as near to as many room boundaries as you possibly can, and seat yourself as far away from the walls as you can get.

"Dr. Boom" Tips The Driver

Introduction Part Three

(Originally Appeared In Issue 27)

Better Sound At Home

Subwoofers can be a very important part of any high-quality home entertainment system. A well designed unit—properly integrated into the system—can greatly enhance your enjoyment of both music and film sound. A bad one, or even a not-so-bad one that is poorly implemented, can reduce fidelity and really mess things up.

Our goal with this series of articles is to provide the reader with the technical understanding necessary to make wise purchasing decisions. We want to ensure that adding a subwoofer (or subwoofers) will *improve* the sound of your home theatre system and give you a real sonic return on your investment. We think that the more you know about how the available products work, the better your chances are for success.

The Heart Of The Matter

All of the subwoofers reviewed so far utilize the conventional moving coil, dynamic driver. Although many other means of moving air at low frequencies have been tried, nothing has proven more effective or more practical than the good old permanent magnet motor, cone diaphragm driver that we all think we know so well.

The dynamic loudspeaker was invented in 1925 by C. W. Rice and E. W. Kellogg. The first units used electromagnets that were energized by the DC power supplies of the amplifiers that drove them. These field coils are now obsolete and have been replaced by permanent magnets.

Early permanent-magnet motors used an alloy of aluminum, nickel, and cobalt called Alnico, which is virtually unavailable now. Most current subwoofers have drivers that use ceramic magnets.

All moving-coil drivers work in much the same way, and their construction is similar. I think a basic understanding of the way in which these drive elements perform may be helpful to our readers as they try to learn about subwoofers and the design compromises that they must consider when choosing a commercially available unit. So now let's take a look at dynamic drivers and the function of their basic parts.

Not All Drivers Have Cars

The diaphragm of a dynamic driver produces sound by compressing and rarefying air through its back and forth movements in response to an alternating electrical current flowing through its voice coil that is suspended in a powerful, fixed magnetic field. Simple, huh?

The frame, or basket, of almost every driver holds a permanent ceramic magnet

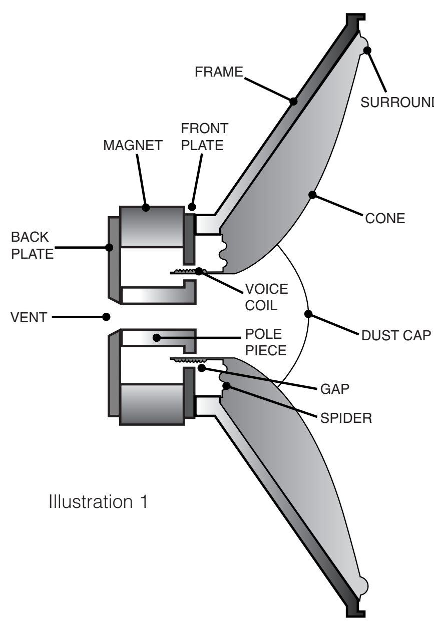


Illustration 1

shaped like a donut. One magnetic pole of this magnet is attached to a back plate and a cylindrical iron pole piece (with or without a central vent) that extends through the hole in the center of the donut and is surrounded by the voice coil. The other pole of the magnet is connected to a plate with a hole in it that closely encircles the *outside* of the voice coil. This front plate completes the magnetic circuit.

The force produced by the magnet is focused in the gap between the two pole pieces, and the voice coil is centered in this gap. The "spider" keeps the voice coil centered between the pole pieces and acts as a spring to return the moving parts—the voice coil and the cone—to the starting position in the center of the magnetic gap, following an excursion which creates sound. The "surround" terminates the cone at the big end and seals the enclosure, isolating the back wave from the front wave when the cone moves in and out.

The spider provides the main restoring force that brings the cone back to its rest position with the surround providing some assistance. The stiffness, or softness of this combined spring is referred to as compliance.

Since compliance is the reciprocal of stiffness, the higher the compliance, the softer the spring and the easier it is to move the cone back and forth.

When a dynamic driver is mounted in a sealed enclosure, the compliance of the air

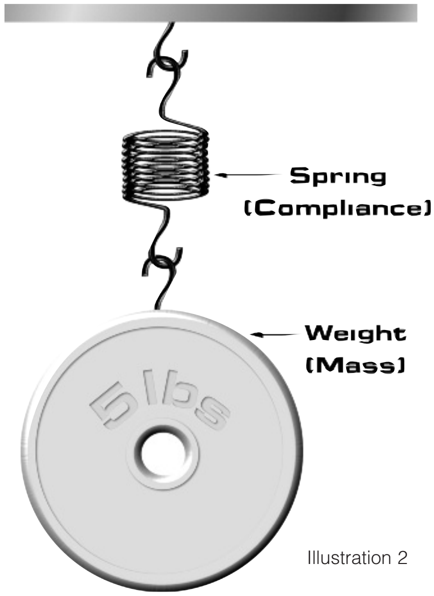


Illustration 2

in the box adds some restoring force to the cone and helps to damp oscillation like a shock absorber in a car.

Fundamental Resonance

Every driver has a fundamental resonance frequency, and this frequency defines the lower limits of its operating range. Obviously, this is an important factor in subwoofer design.

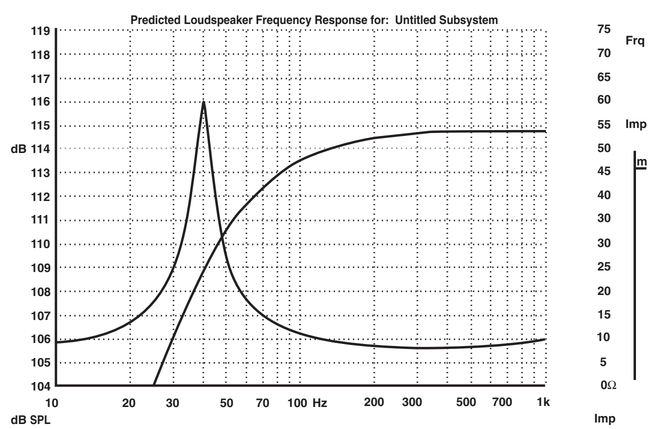
Compliance and mass establish the resonant frequency of a driver or a system. As compliance decreases, resonant frequency increases. As mass increases, resonant frequency decreases...picture a weight hanging at the end of a spring (see Illustration 2).

If you pull down on this weight and then release it, the weight will then bounce up and down at a set frequency. If you make the weight heavier (more mass), it will bounce slower (lower resonant frequency). If you make the spring softer, the weight will also bounce slower (again, a lower resonant frequency). Conversely, making the weight lighter or stiffening the spring will raise the resonant frequency.

The enclosure "loads" the driver in order to control and modify the fundamental resonance, thereby extending and smoothing the bass response of the system.

Drivers designed for use in sealed enclosures will generally have a lower free-air resonance frequency (due to higher compliance and/or higher moving mass) than those designed for use in vented enclosures, because the air in a sealed box will stiffen the restoring force on the cone and raise the in-system resonant frequency.

Vented enclosures use the mass of a column of air in the vent (port), and the "spring" of the air in the enclosure to create another



Driver Parameters

Driver: JBL 2235H

Nominal Diameter	D = 15 Inches
Free Air Resonance	f(s) = 20Hz
Total Driver Q	Q(ts) = .25
Driver Electrical Q	Q(es) = .28
Driver Mechanical Q	Q(ms) = 2.5
Equiv. Volume	V(as) = 16.2 cubic ft.
Nominal Impedance	Z(nom) = 8 Ω
DC Resistance	R(e) = 6 Ω
Max Thermal Power	P(t) = 150 Watts
Peak Excursion	X(max) = .33 Inches
Piston Area	S(d) = 138 Sq. Inches
Efficiency	n(0) = 1.3 %
Sensitivity (1W/1m)	SPL = 93 dB
Voice Coil Inductance	L(e) = 1.2 mH
Magnet Weight	Magnet = 0 lbs.
Voice Coil Diam.	Diam.(vc) = 4 Inches
BL Product	BL = 20.5 T * m
Flux Density	Flux = 12 Gauss
Dynamic Mass	M(as) = 0 (Kg)
Voice Coil Length	L(vc) = 1.2 inches
Magnetic Gap Length	L(gap) = .28 inches

Note: voice coil length and gap length are in inches

Box Parameters

Box Type: Second Order Closed Box

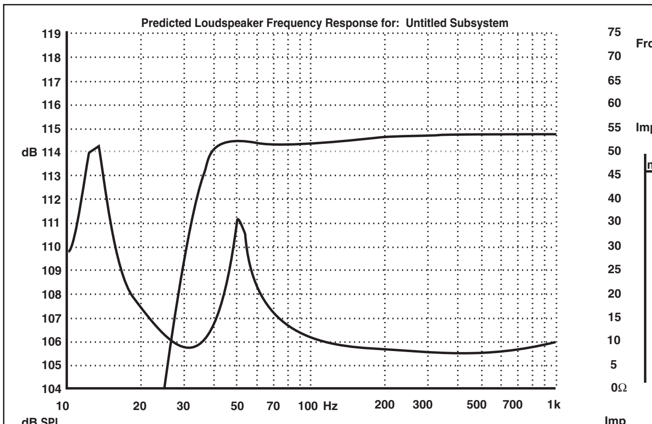
Box Volume	V(B) = 5.400 cubic ft.

Subsystem Parameters

Subsystem Name: Untitled Subsystem

Number of Drivers	N = 1
Closed Box Q	Q(tc) = 0.500
Compliance Ratio	alpha = 3.00
Subsystem Input Power	P(in) = 150.0
SPL Distance	D = 1.00 m (3.28 ft)

Illustration 3 JBL 2235H, Sealed Enclosure



Driver Parameters

Driver: JBL 2235H

Nominal Diameter	D = 15 Inches
Free Air Resonance	f(s) = 20Hz
Total Driver Q	Q(ts) = .25
Driver Electrical Q	Q(es) = .28
Driver Mechanical Q	Q(ms) = 2.5
Equiv. Volume	V(as) = 16.2 cubic ft.
Nominal Impedance	Z(nom) = 8 Ω
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Magnet Weight	Magnet = 0 lbs.
Voice Coil Diam.	Diam.(vc) = 4 Inches
BL Product	BL = 20.5 T * m
Flux Density	Flux = 12 Gauss
Dynamic Mass	M(as) = 0 (Kg)
Voice Coil Length	L(vc) = 1.2 inches
Magnetic Gap Length	L(gap) = .28 inches

Note: Voice coil length and gap length are in inches.

Box Parameters

Box Type: Fourth Order Vented Box

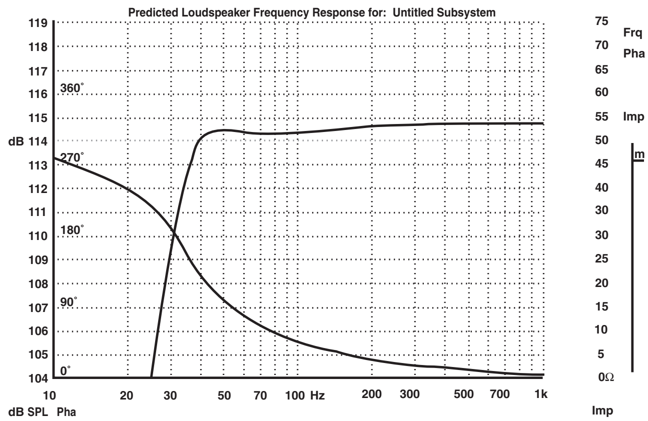
Box Volume	V(B) = 5.400 cubic ft.
Box Frequency	F(B) = 32.00Hz
Min Rec Vent Area	SVMin = 29.1 Sq Inches
Vent Surface Area	S(v) = 30.00 Sq Inches
Vent Length	L(v) = 10.165 Inches
Box Loss Q	Q(B) = 7.00

Subsystem Parameters

Subsystem Name: Untitled Subsystem

Number of Drivers	N = 1
Closed Box Q	Q(tc) = 0.500
Compliance Ratio	alpha = 3.00
Subsystem Input Power	P(in) = 150.0
SPL Distance	D = 1.00 m (3.28 ft)

Illustration 4, JBL 2235H, Vented Enclosure



Driver Parameters

Driver: JBL 2235H

Nominal Diameter	D = 15 Inches
Free Air Resonance	f(s) = 20Hz
Total Driver Q	Q(ts) = .25
Driver Electrical Q	Q(es) = .28
Driver Mechanical Q	Q(ms) = 2.5
Equiv. Volume	V(as) = 16.2 cubic ft.
Nominal Impedance	Z(nom) = 8 Ω
DC Resistance	R(e) = 6 Ω
Max Thermal Power	P(t) = 150 Watts
Peak Excursion	X(max) = .33 Inches
Piston Area	S(d) = 138 Sq. Inches
Efficiency	n(0) = 1.3 %
Sensitivity (1W/1m)	SPL = 93 dB
Voice Coil Inductance	L(e) = 1.2 mH
Magnet Weight	Magnet = 0 lbs.
Voice Coil Diam.	Diam.(vc) = 4 Inches
BL Product	BL = 20.5 T * m
Flux Density	Flux = 12 Gauss
Dynamic Mass	M(as) = (Kg)
Voice Coil Length	L(vc) = 1.2 inches
Magnetic Gap Length	L(gap) = .28 inches

Note: Voice coil length and gap length are in inches.

Box Parameters

Box Type: Fourth Order Vented Box

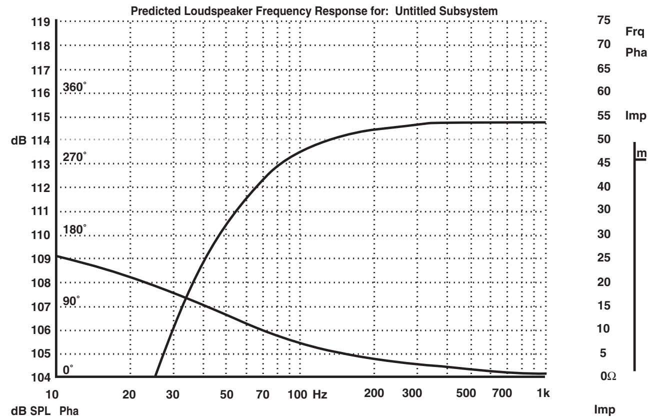
Box Volume	V(B) = 5.400 cubic ft.
Box Frequency	F(B) = 32.00Hz
Min Rec Vent Area	SVMin = 29.1 Sq Inches
Vent Surface Area	S(v) = 30.00 Sq Inches
Vent Length	L(v) = 10.165 Inches
Box Loss Q	Q(B) = 7.00

Subsystem Parameters

Subsystem Name: Untitled Subsystem

Number of Drivers	N = 1
Closed Box Q	Q(tc) = 0.500
Compliance Ratio	alpha = 3.00
Subsystem Input Power	P(in) = 150.0
SPL Distance	D = 1.00 m (3.38 ft)

Illustration 5, JBL 2235H, Vented Enclosure



Driver Parameters

Driver: JBL 2235H

Nominal Diameter	D = 15 Inches
Free Air Resonance	f(s) = 20Hz
Total Driver Q	Q(ts) = .25
Driver Electrical Q	Q(es) = .28
Driver Mechanical Q	Q(ms) = 2.5
Equiv. Volume	V(as) = 16.2 cubic ft.
Nominal Impedance	Z(nom) = 8 Ω
DC Resistance	R(e) = 6 Ω
Max Thermal Power	P(t) = 150 Watts
Peak Excursion	X(max) = .33 Inches
Piston Area	S(d) = 138 Sq. Inches
Efficiency	n(0) = 1.3 %
Sensitivity (1W/1m)	SPL = 93 dB
Voice Coil Inductance	L(e) = 1.2 mH
Magnet Weight	Magnet = 0 lbs.
Voice Coil Diam.	Diam.(vc) = 4 Inches
BL Product	BL = 20.5 T * m
Flux Density	Flux = 12 Gauss
Dynamic Mass	M(as) = (Kg)
Voice Coil Length	L(vc) = 1.2 inches
Magnetic Gap Length	L(gap) = .28 inches

Note: Voice coil length and gap length are in inches.

Box Parameters

Box Type: Second Order Closed Box

Box Volume	V(B) = 5.400 cubic ft.

Subsystem Parameters

Subsystem Name: Untitled Subsystem

Number of Drivers	N = 1
Closed Box Q	Q(tc) = 0.500
Compliance Ratio	alpha = 3.00
Subsystem Input Power	P(in) = 150.0
SPL Distance	D = 1.00 m (3.38 ft)

Illustration 6, JBL 2235H, Sealed Enclosure

resonating system that interacts with the driver resonance, modifying the character of the overall system resonance. You can see what happens in the graphs of Illustrations 3 through 6.

A passive radiator system works just like a vented system but replaces the mass of the column of air in the vent with the mass of the passive cone.

Look at the graphs shown in Illustrations 3 and 4. These graphs show frequency response and impedance for a JBL 2235H 15-inch driver in enclosures with a volume of 5.4 cubic feet. This driver has a free air resonance of 20Hz.

In the sealed box of Illustration 3, the graph shows that the resonant frequency of the system is 40 Hz, as indicated by the sharp impedance peak at that frequency.

In Illustration 4, a vented enclosure tuned to 32 Hz shows the two impedance peaks typical of a vented design with the valley between them at the box-tuning frequency. This JBL driver appears to be best suited to a vented enclosure design, but adding weight to its cone would lower the resonant frequency and probably improve performance in a sealed box. Note that in the vented enclosure, one of the impedance peaks is at 50 Hz—prominently located in the middle of the pass band.

Now look at Illustrations 5 and 6. These graphs show frequency and *phase* response of the same driver in the same enclosures (one sealed and one vented) as above. Note that with the vented enclosure in Illustration 5, we have 180° of phase change at 30 Hz. The phase response of the sealed enclosure system in Illustration 6 looks much different.

These phase plots are just for the driver in the enclosure. Phase changes due to filtering and equalization are not considered. The relevance of this stuff will be discussed in the reviews of the Bag End and Vandersteen subwoofers. These two companies believe, as I do, that phase response is important in any loudspeaker system, even one designed only for low-bass reproduction.

Waves You Can't Surf

When the cone of a dynamic driver moves outward, it increases the air pressure in front of the cone and reduces the pressure on the other side at the same time. When the cone moves inward, the opposite occurs. These pressure waves in the air modulate your ear drums and are perceived as sound if they occur in the frequency range that we can hear—roughly 20 Hz to 20 kHz.

A driver suspended in free air couldn't reproduce low frequencies very well because a pressure wave created on one side of the cone would wrap around and fill the low-pressure area simultaneously developed on the other side and cancel out what otherwise would have become sound. A very large, flat baffle would prevent this cancellation but would add little in the way of damping to smooth the driver's natural resonance peak—the driver would have to rely on its own mechanical and electrical properties for damping.

A sealed "air-suspension" enclosure isolates the front wave from the rear wave to prevent cancellation and smooths and extends the response by damping resonance and oscillation after the signal stops.

A vented enclosure "delays" the rear wave so that it can be phase inverted and added to the front wave in phase (at some frequencies) to extend response and reduce driver excursion at the tuning frequency of the enclosure.

Some enclosure designs attempt to damp out resonance so completely that it is almost imperceptible, acoustically or electrically.

What Do I Like?

I've listened to and written about so many subwoofers in the last few years that almost everyone now calls me "Dr. Boom."

"Well, Dr. Boom," you might ask, "after listening to all those subwoofers, have you come to any conclusions? What design works best? What do *you* like?" In addition to the subwoofers in this special edition, I think that I have heard just about every product that has achieved any level of commercial success in the last thirty years or so, and I have designed and built a number of subwoofers myself in those years so I do have some general preferences. I'm going to tell you what those are, but first I want to remind everyone that my opinions are just that. They're just *my* opinions.

You may have different tastes in music. You may have different tastes in film. You may have different tastes in a lot of things...and that includes subwoofers.

You might use my reviews as an aid to help establish your own merit system. Listen for the characteristics that I describe and see if you can hear these things, and then decide just how important each of these sonic properties are to *your* enjoyment. Modify some of my biases to fit your personality and you'll get what *you* want, not just what sounds good to me.

So, what do I like? In general, I prefer the tighter transient control of sealed enclosures to vented boxes, although a well-designed vented system can sound much tighter than a high-Q sealed box.

I usually don't care for servo feedback systems, but the conservative use of feedback in conjunction with other techniques can work very well, as in the Mirage BPS-210 subwoofer, which I like a lot. And the products that I have enjoyed the most over the years are all attempts at producing aperiodic response in the pass band. Examples are transmission line, resistive port, and really well-designed Isobarik systems; and those subwoofers that operate primarily below system resonance like the Bag End and Vandersteen products.

The less equalization the better in my opinion, and that means that bigger boxes usually sound better to me than smaller ones.

Band-pass designs have never been completely pleasing to me. I find that band-pass subwoofers usually have a characteristic "the bass is in the closet" sound that is distinct and different from the sound of the main loudspeakers, but Polk Audio has produced some band-pass designs that perform well. (The Polk designs use patented alignments so they are unique among band-pass systems).

It is very hard to make broad generalizations about subwoofer design types because they all have performance tradeoffs, and good products have been delivered utilizing a variety of techniques.

In the reviews completed so far, we have given "Spotlight On Performance" awards to sealed box designs, vented designs, band-pass types, push-pull driver arrangements, Isobarik systems, and two designs using passive radiators. There are many different ways of achieving high performance and, as with all audio components, your ears are the final arbiter.

I can say for sure that I like subwoofers that are transparent to the rest of the system. If you are always aware that a subwoofer is present, then things aren't working as well as they could be. Good subwoofers, when set-up properly, blend seamlessly with the main loudspeakers and make the sound bigger and more natural, not more bass heavy.

More Demo Material Suggestions

While listening to the subwoofers in this installment of the review, I have enjoyed the music from many CDs, especially the following: Doug MacLeod *Unmarked Road* from Audioquest, Chris Isaak *Baja Sessions* on Reprise, Sergio Mendes *Brasileiro* on Electra, (thanks, Tony), and the Louis Prima *Collectors Series* from Capitol. Suggested DVDs and LaserDiscs include: *Mission: Impossible*, *The Rock*, and *Men In Black*, plus the DTS version of The Eagles *Hell Freezes Over*.

If you are primarily interested in film sound and music is secondary, a vented subwoofer will generally get you more slam. If high fidelity music reproduction is your first concern, stick with sealed enclosures.

The Old And Wise Dr. Boom

Introduction Part Four

(Originally Appeared In Issue 31)

Practical Advice Vs. Technical Theory

The first three subwoofer articles in this series offered lots of technical information to help readers understand low-frequency loudspeaker design concepts and the engineering trade-offs that are required to make a commercially viable subwoofer.

This installment is aimed more at providing practical advice on how to get good bass performance in your home, using the readily available products offered for sale at your local dealer. The advice presented here is strictly my opinion and is based on my considerable real-world experience in the implementation of home audio systems.

Subwoofers should make your home entertainment system sound better, not boomier; so I'm going to start this segment of the subwoofer series by sharing some of my observations about the nature of sound. I'll then offer some simple, practical suggestions about how to set up your home theatre system and the room that contains it. I hope that this will help you to get the deep, tight, well-defined and controlled, low-frequency performance—aka good bass—that all discerning listeners seek.

We definitely live in the information age. Unfortunately, much of the information that is so readily available is either flawed or completely wrong. I've included some cautions about the misinformation that was disseminated in the past and advise that you carefully scrutinize the advice that flows freely from so many sources today.

Folklore

Thirty years ago or so, some pundit stated that you couldn't reproduce low-bass frequencies in

Mutt & Jeff Subwoofers The Velodyne HGS-12 (Left) & Energy ES 18XL (Right)

a small room if the wavelength of the frequency in question exceeded the maximum room dimensions. As silly as this may sound now, if you refer back to the writings of those days you will find repeated references to this supposed fact.

Headphone manufacturers and car stereo installers, apparently unaware that their feats were impossible, went ahead and made deep bass in small spaces. How can this be? If a 40 cycles-per-second (40 Hz) bass tone has a wavelength of 27.5 feet, how can it fit in a car with a passenger compartment that is only six feet long?

Ear Drums Vs. Microphones

Any force that makes your ear drums move in and out will be perceived as sound if the frequency at which they oscillate is within the range of frequencies that we can hear.

Visualizing acoustic pressure waves as the sinusoid that represents them electrically can be misleading. Ripples in a pond surrounding the spot where a rock fell present a more realistic depiction.

The water level in the little circular waves that form around the point of entry is higher than the surface of the pond at the crest of each wave and lower than the pond surface in the troughs between the crests. A leaf floating on the water will rise and fall as the waves pass by. Sound waves work in much the same way.

Sound is just a tiny, recurring increase and decrease in the ambient air pressure around your head that takes place at an audible frequency. If the barometric pressure could (repeatedly) rise and fall rapidly enough, you would hear it.

The vibrating surface of the skin on a bass drum in the concert hall makes pressure ripples in the air, which travel out to your ears at 1,100 feet per second (at sea level) and cause your ear drums to vibrate in sympathetic response. You perceive this as the sound of a drum in a large hall. Things work a little differently in a smaller venue.

Picture a leaf floating on the surface of water contained in a bucket. You can make the leaf rise and fall by inserting and removing your foot. When your foot goes in, the water level rises and then drops as your foot is removed. The leaf doesn't have to wait for the ripples to travel as it did in the larger pond. In this smaller container it will rise and fall in nearly immediate response to the insertion of your foot.

When a car stereo woofer cone (or a headphone diaphragm) producing a bass note moves outward, the enclosed volume of air between it and your ears becomes slightly smaller and the air pressure increases. This causes your ear drums to move inward. When the sound-producing diaphragm moves inward, the space that contains the air gets slightly larger and the pressure drops—your ear drums move outward. In a small space (relative to the wavelength of the sound), the sound-producing element and your ear drums are almost directly coupled at low frequencies.

A microphone has a diaphragm that simulates one of your ear drums, and it appears to work in an analogous manner. When the microphone diaphragm moves in and out in response to air pressure changes, an electrical signal is generated. This all seems very simple, and "objective measurements" will confirm that it is, but careful observation shows us that while the microphone works pretty much as expected, human perception is far more complex.

The sound of seven plucked double basses behind a symphony orchestra in a large hall makes the microphone diaphragm move in exactly the same way that a recording of the same performance would, when played back in a living room, through a high-resolution audio system with a subwoofer capable of reproducing the full bandwidth of the original event. The microphone might represent these two events as identical, but they sure won't sound the same.

All the traditional rules of acoustics that apply to the large space of the concert hall change subtly when we start the switch to the pressure-vessel environment of the smaller room at about 130 Hz. No discerning listener will ever be completely fooled by the subwoofer(s), but the microphone may be.

Even a laboratory-grade calibrated microphone is no match for a pair of good ears connected to an intelligent brain.

Now I don't pretend to know all of the answers to all the questions about what is really happening here. I have spent a lot of time listening to both live performances and recorded music, and comparing real-world voices and sounds to the recordings of those same things on film soundtracks, and I have learned a lot about which of today's accepted facts don't stand up to scrutiny and which things really do work in a home environment. That is what we're going to discuss next.

Some Things I Know, Some Things I Guess....

There is a great song lyric that goes something like this: "Some things I know, some things I guess, some things I wish that I could learn to express...." ("Southbound Train" by Julie Gold ©1994 Julie Gold Music).

I *know* from practical experience (empirical evidence) that certain accepted "truths" aren't completely true and I can *guess* at the reasons why. (A guess could also be called a hypothesis.)

Some concepts, about which I have little doubt are difficult to *express* in the limited allotment of space that is available for a magazine article. But I'm going to try anyway.

Let me give you some examples. I know that the traditional idea of standing waves is not completely correct when applied to smaller spaces, and that computer software that takes your room dimensions and calculates where to put your loudspeakers based on these concepts almost never works to give you the best sound. (Did you ever hear anyone talk about standing waves in a car?)

I know that two widely spaced subwoofers will improve the quality of bass everywhere in the room (and eliminate most, if not all, of the standing wave problems) but that the results won't measure quite as well using single mike techniques. Four subwoofers, in four different locations, sound better yet. (Editor Gary Reber uses six powered subwoofers in his primary home reference system and there are six in the Polk *Super* Signature Reference Theater system at *Widescreen Review's* production offices!)

A subwoofer for each main loudspeaker, crossed over from a full-range signal to that specific channel sounds much better than summed bass from the subwoofer output on a processor/controller even when an equal number of subwoofers are used in each case. (Some testers claim that mono bass is better, based on their simplistic measurements. They're dead wrong!)

And I know that interpreting measurements (particularly from subwoofers) is highly subjective and if the tester relies solely on his instruments he can come up with a measurement that "proves" his preconceptions (which are frequently misconceptions).

So what then is the value of this subwoofer series? If all the "expert" opinions are in conflict, how do we get at the truth?

I believe that the goal of the vast majority of our readers is simply to get good bass performance from their home theatre systems and they read these articles to find out how to do that. Well, I know how to obtain good, tight bass in my living room and I can share some of that knowledge with you. How do I prove that my advice can be trusted even when it may differ from the advice that you have read elsewhere? Well, look at my picture. Don't I look old and wise?

If I were a politician or a custom home theatre guru, I would advise you to hand over your money and just trust me. But I'm just a magazine writer now (although I have tweaked over 10,000 audio systems in real homes). So I'll give you some advice based on personal experience and you can try it. If it works, you will have better bass for free (almost). If it doesn't work, you can hire one of those guys who write those personal ads disguised as technical articles in those other magazines, to set up your room for you.

Intelligence + Experience = Wisdom?

When I was young I loved to use the phrase "wisdom is in the head, not in the beard." Now that I'm old and have a beard (along with a few less brain cells and a lot less hair), I realize that while smart is good, there is no intellectual substitute for practical experience. Since I can't get any younger, that's my story now and I'm sticking to it. I have also learned that trusting your senses frequently produces more satisfying results than relying solely on your intellect, especially when evaluating systems designed specifically to stimulate those senses. To put that in other words: if it sounds good, it is good.

As a youth I charged into the world armed with a science education, lots of theoretical knowledge, and an ego as big as my intellect. I found that working as an engineer in the computer industry brought me little personal satisfaction and decided to share my love of music with others and make my living by designing accurate audio systems for their homes. This gave me the opportunity to listen to a broad variety of audio components in a very large number of different acoustic settings, almost all of which were in actual homes.

I met a lot of nice people. I sold a lot of outstanding equipment, set it up properly, and made most of them very happy.

I used to take tons of test gear into client's homes and I measured everything in sight until I learned that I nearly always got better sound without relying entirely on this equipment. While measurements were sometimes useful for initial set-up, actually listening to an audio system proved to be the best method of gauging and adjusting sound quality. After all, a sound system is supposed to sound good, not just produce good graphs.

Measurements should be used to discover *why* something sounds the way that it does so that flaws can be corrected and successes can be repeated. It is very easy to fall into the trap of using test results to *gauge* sound quality, which is a backwards concept in my opinion.

I am certainly not condemning all measurements—I've made a lot of them. I'm just advising that tests remain subordinate to the real goal of achieving good sound and not become the determining factor.

By the time that I had visited 100 homes, I was using the test equipment far less and my ears much more,and achieving better sonic results. After I had visited 1,000 homes, I could set up rooms and equipment intuitively, based on past experience, and achieve measurably better results much more quickly.

I had learned by this time that many of the acoustic theories that I had been taught in school just didn't work in real living rooms.

The homes I visited looked different, one from another, but acoustically they were remarkably similar. The average room size and method of construction didn't vary much, and consistent results could be achieved from similar set-up techniques. It's likely that your home is very much like mine, acoustically.

When my system count reached 10,000, I stopped actually going to the site and began to set up rooms and systems using telekinesis.—OK, I got a little carried away there for a minute, but you get the idea. I didn't just fall off the turnip truck. It would be very hard for anyone to suggest an experiment involving home audio systems that I haven't tried personally.

I learned from experience just how much I didn't (and still don't) know. I learned about a lot of accepted acoustic concepts that simply don't work in small rooms (like traditional room EQ), but experience has also taught me about many things that do work to make better sound in a home environment. What I'm going to try to do here is share some of that positive experience with you. The suggestions that follow are based on empirical evidence—these are things that I know because I have observed them to be true, over and over again. Give them a try and see if they work for you.

Now obviously all rooms aren't the same and while my suggestions are based on techniques that have generally proven to be effective, some acoustic problems will require specific attention beyond the scope of this article. However, simply setting up your system properly will get you nearly all the performance that your equipment is capable of, almost every time.

You will read opposing viewpoints from other sources and many of them can be traced back to "the force" in Northern California—Lucasfilm's THX division.

The THX Phenomenon

The high-end segment of the home high-fidelity audio business was a mature industry when home theatre first appeared on the scene. We had learned a great deal about how to make accurate sound in a home environment over a thirty year period of testing and experimentation.

The increasing popularity of home theatre spawned new self-proclaimed experts, many of whom had no audio experience at all, and little technical knowledge except for what they were taught at Skywalker Ranch in the THX training courses. These THX evangelists went forth preaching the gospel according to Tomlinson Holman, the primary architect of the THX standards.

The THX philosophy is based on many cogent principals that were developed over the years for sound reinforcement in large venues like movie theatres, and THX tries to simulate the sound of these professional PA systems in the home. Here is where we differ.

While I wholeheartedly support the work that the THX division of Lucasfilm has done to improve the sound at the local cineplex, I don't think that this level of sound quality is sufficient to act as a standard for the home theatre enthusiast (and certainly not for a music lover), and I know that many principles that are useful in large-venue public address systems don't work well in the living room, if high-fidelity sound is your goal.

THX operates like a secret society. They want you to join their club and pay extra for products that display their logo but they will not tell you (or me) about the standards that these products are required to meet. After measuring many THX-certified subwoofers and processors I can tell you this much: THX advocates mono bass obtained from the subwoofer output of a surround processor (even when multiple subwoofers are used), and steep-slope crossovers between the subwoofer(s) and the main loudspeakers.

They require that THX-certified subwoofers be capable of very high-output levels in the mid bass region. These high-output levels are frequently achieved by limiting deep bass response capability and sacrificing some fidelity and transient performance to obtain product certification. I disagree with these choices.

The Bad Old Days

Early high-fidelity systems were based on the theatre sound systems of the day. The whole hi-fi industry started out with JBL horns or Altec *Voice Of The Theatre* loudspeakers. We quickly learned that we could do much, much better in the home, and we did. Modern high-end hi-fi systems offer natural sound reproduction that is light years beyond the fidelity of the PA systems used in commercial movie theatres.

With the advent of home theatre, a new myth was created. It proposed that we should go back to the days of primitive audio systems so that we could "accurately replicate the sound of the dubbing stage" where the film soundtracks were created. These dubbing stages still use loudspeaker systems based on sound reinforcement designs that are decades old.

Making our home audio systems sound as bad as a dubbing stage is supposed to deliver sound that "the filmmaker intended."

Don't buy this false premise if you want good sound in your home. *Real* sounds are the true standard for fidelity, not the sound from the PA system used in the dubbing theatre. A voice should sound like a voice, not like a loudspeaker using a horn-loaded compression driver as a tweeter.

An audio playback system should not be required to alter the recorded signal to suit some "expert's" opinion of what a film soundtrack should sound like. DVDs and LaserDiscs are strictly for home theatre use. It is the filmmaker's responsibility to put the sound that he or she "intends" to be heard on the DVD or LaserDisc, so that an accurate home playback system can reproduce it without alteration.

Are Measurements Ever Useful?

I have been measuring audio components for over three decades, and I have accumulated a huge data base for my own reference.

In the early days, I employed primitive tools and did calculations by hand (I am proficient with a slide rule). I have graduated to sophisticated computer-based instruments and I can push *Enter* on a computer keyboard as well as the next guy.

While I believe that measurements are invaluable to a designer and can be very useful to a reviewer if he knows what he's doing, (most don't), I don't believe that publishing test graphs offers much value to the reader. I do it from time to time if I think that a graph can illustrate a point without requiring subjective interpretation, but I don't print graphs just to demonstrate how clever I am or to support my own preconceptions.

Test graphs have relevance to the tester who knows how they were made and then only in relation to others made in the same way. How a graph relates to perceived sound quality requires *subjective interpretation*. And graphs seldom tell the whole story anyway. At best, graphs have *relative* relevance and then only to the experienced tester who also uses his *ears*.

Measurements should attempt to determine *why* something sounds the way it does, and that requires that the tester listen first.

Subwoofers are particularly difficult to measure in any meaningful way. A 15 Hz sound wave is nearly 74-feet long and I don't know of any anechoic chamber that is large enough to contain a full cycle at or near this frequency. You can't window-out room reflections at the low frequencies we're dealing with and because of the high ambient sound level in the city, outdoor tests are highly impractical.

Outdoor and anechoic tests are suspect anyway. Can any free-space (or half-space) measurement really gauge the performance potential of a subwoofer that will actually be used in the confines of a relatively small room, and probably be positioned in the corner of that room?

You can measure single-driver, sealed-box designs, by using a close-mike technique and then compensate for diaphragm size. The results correlate somewhat with the perceived sound. But what about vents, passive radiators, and multiple driver systems?

Do you add the output from a passive radiator or a vent to the output from an active drive unit even though the two may be out of phase over most of the subwoofer's pass band? How about an algebraic add or "complex sum," will that accurately reflect how the two signals combine in the room?

Does a distortion measurement using a steady-state sinusoidal stimulus correlate to any perceived sonic characteristic, and how do you accurately gauge transient response, a performance factor that can be easily heard?

What about in-room amplitude response (how low does it go and how loud will it play)? Isn't that all that really matters to me as a subwoofer user?

Will any test help me choose, position, and adjust a subwoofer so that it will improve rather than diminish the performance of my home theatre system?

Listening is the most important tool, of course, but there are some simple measurements that can help you get the best sound in your home. In fact, one of the oldest and simplest tests—the sine wave sweep—is not only valuable but it's easy for you to simulate yourself.

Pros use a sweep generator and a chart recorder or a computer-based measurement system, but you can get the same information from a test CD with discrete tones, or warbles, a sheet of graph paper, and a Radio Shack sound level meter. (I don't recommend LED spectrum analyzers and pink noise signals.)

Knowing the approximate response curve of your subwoofer, through the crossover region, as measured at the listening position, will help you to identify and correct problems with acoustics and set-up.

Use the slow response setting. Don't place the meter on a tripod. Move it around a little and take an average reading. Look for major peaks and dips and adjust accordingly. Don't play sine waves too loud for too long or you will see smoke.

How loud the subwoofer will play in a real system can also be measured this way, but you may damage it and I've already done that for you (measured how loud they will play, not damaged them) in this series of reviews.

Good Bass, How Do We Get It?

Subwoofers cover a fairly small range of frequencies—generally just the two octaves from 20 Hz to 80 Hz plus the roll-off region slightly above and below this range. As luck would have it, most of the low-frequency acoustic problems that you will encounter in an average home occur between 60 Hz and 100 Hz.

Because the crossover between the subwoofer and the main loudspeakers is right in the middle of this range, you can correct many of these problems by simply adjusting the individual crossover points (high-pass and low-pass) and the relative level and phase of the subwoofers versus the main loudspeakers.

Boomy Bass, Or No Bass

Walls that flex too much will suck-up bass. Suspended ceilings with drop-in panels are a disaster in this regard. Surfaces that store and release low-frequency energy cause "boom," as do pressure zones near the room boundaries (don't sit in one).

If you live in a typical tract house like I do, and your walls are made of half-inch drywall nailed to 2 x 4 studs, pound your fist on the wall between the studs and listen to what happens. Most room-boom comes from these wall panels (not from standing waves) and the only cure is to make them stiffer.

You can attach a second layer of drywall with screws, line the interior surfaces with brick or block, cover the walls with heavy wood paneling sandwiching absorbent material between the dry wall and the wood, or attach surface studs and absorbent panels to stiffen the walls and damp some of the re-radiated energy. I've chosen the last option for my room because it's the cheapest and it serves the dual purpose of also absorbing reflected higher-frequency energy from problem surfaces.

Choosing Subwoofers

With a given enclosure size, vented designs (all non-resistive port, passive radiator, and band-pass types) will generally provide greater output, and sealed box units will provide higher fidelity and improved transient response.

For the highest fidelity, choose drivers that are 12-inches in diameter or smaller, mounted in sealed boxes. Larger drivers (15- and 18-inch) will sometimes offer higher output but will produce more side band distortion from cone break-up.

Vented systems look good on paper if you consider just amplitude response. A well-designed vented system can extend the range of high-level bass output, but it does this by adding another resonating mechanical circuit. Much of what you hear is a resonance, not the signal. Below this resonance, the driver becomes unloaded, distortion goes way up, and response falls off very rapidly.

The steep-slope, high-pass characteristic of a vented system degrades transient response and limits the ultimate low-frequency extension of the subwoofer. If you are primarily interested in film sound and music is secondary, a vented subwoofer will generally get you more slam. If high-fidelity music reproduction is your first concern, stick with sealed enclosures.

Besides improved transient performance, sealed enclosures have another seldom mentioned but significant advantage over vented systems for high-fidelity reproduction: substantial usable output well below the nominal cut-off frequency.

Equalized vented designs will exhibit a sixth-order, high-pass characteristic (or thereabouts) producing a very steep roll-off with little output below their low-frequency cut-off point. Sealed designs roll off more gently (second-order or less for low-Q designs) providing output well below the range of flat-frequency response, albeit with reduced level. Sealed boxes make bass that sounds much more natural to me.

Trust me on this: Two smaller subwoofers (of comparable quality) operating in stereo will always sound better than one big one using a summed mono signal.

If your goal is maximum output at low frequencies for home theatre, and music listening is a secondary concern, choose one or two 15- to 18-inch subs in vented enclosures driven by a summed mono signal.

If you want the highest-fidelity bass, choose sealed box subwoofers with 8-inch, 10-inch or 12-inch drivers (or multiple drivers of these sizes) and use at least two subwoofers in stereo. Position the subwoofers as far apart as your room allows.

If you have the space and the money, use a subwoofer for each primary channel (not the center channel) crossed over from a full-range signal to that channel (not from a summed subwoofer output on the processor), plus a dedicated LFE subwoofer. A subwoofer for the center (or dialogue) channel is always a bad idea in my opinion.

I have four Vandersteen 2W-Q subwoofers in my primary reference system: one for each of my main channels, and an Energy ES-18XL (an 18-incher) for LFE. The big Energy will deliver 114 dB at 40 Hz for involving sound effects, while the Vandersteens, with their triple 8-inch drivers and adjustable Q, provide the highest musical fidelity possible. I can have my cake and eat it too. So can you.

Main Loudspeakers And Crossover Slopes

It's wishful thinking to believe that a mini loudspeaker (satellite) and a subwoofer, combined using a steep-slope crossover, can equal the performance of a full-range main loudspeaker augmented by a subwoofer, with the two blended together using gentle crossover slopes. You need at least an octave of response beyond the crossover point for a seamless blend between any two elements. That means that a main loudspeaker should have solid bass performance to below 40 Hz for a subwoofer crossover point of 80 Hz. A subwoofer should have pistonic response without cone breakup to at least 160 Hz, and breakup that occurs well above this frequency will be audible as directional sideband distortion, regardless of the crossover point or slope. Huge drivers can't deliver true, high-fidelity results in my opinion.

Steep slope filters ring and cause other deleterious effects and should be avoided if possible. Third and fourth-order (18dB and 24dB/octave) filters will be necessary when using small main loudspeakers with limited bass response. First and second-order filters are preferable (mandatory in my opinion) for the best fidelity. A steep slope high-pass filter will depreciate the sound quality of the midrange and should be avoided if at all possible.

If space and budget allow, start with full-range loudspeakers and add subwoofers using gentle crossover slopes for the highest-fidelity results.

Positioning The Subwoofers And The Listener

Because we are dealing with a confined volume of air when we are talking about low frequencies and small rooms, you need to be aware of the pressure zones that occur near the room boundaries. The subwoofers should be positioned in a pressure zone and you should not be. My advice is very simple: put the subwoofer(s) as near to as many room boundaries as you possibly can, and seat yourself as far away from the walls as you can get.

Computer software that determines the best listening position is worse than worthless. Move around and determine the best listening position by *listening*. Beware of anyone who says "*look* at how much better this sounds."

The front corners of the room, behind the main left and right loudspeakers (on the floor of course), are almost always the two best positions for subwoofers. Don't put the subwoofers nearer to the listeners than the main loudspeakers that they augment. Don't position multiple subwoofers in the same location— spread them as far apart as practical. In the unlikely event that you end up with too much low-bass output, move the subwoofers away from the corners, along the front wall.

Never, ever put your listening chair(s) against a wall. And don't tell me that you have to. If your room is so small that you need to put your chair up against a wall, either move or take up needlepoint and forget about good sound.

Beginning Adjustments For Subwoofer Level And Phase

Match the levels of the main loudspeakers and the subwoofers by adjusting for equal output at 150 Hz and 50 Hz using a test CD and a sound level meter. Make the test tones measure about 85 dB at the listening position. This initial adjustment will get the subwoofer volume-level setting in the ball park.

Using a warble tone or a sine wave at the crossover frequency, measure the total output sound level with the subwoofers wired in positive phase (plus to plus). While maintaining the same level settings, reverse the phase to just the subwoofers (not the main loudspeakers) and measure again. If you have two subwoofers, make sure that they always stay in phase with each other. (When reversing phase relative to the main loudspeakers, reverse both subwoofers at once.) Leave the subwoofers connected in the polarity that produces the most output at the crossover frequency. After determining the correct phase setting you may have to readjust the subwoofer volume level.

Smoothing Response From 40 Hz To 120 Hz

With a fixed signal level, carefully measure the output levels at 5 Hz intervals (or smaller) between 30 Hz and 150 Hz. With a crossover point of 80 Hz you will have output from both the main loudspeakers and the subwoofers over the range of 60 Hz to 100 Hz or so. Crossovers with steeper slopes (18 to 24 dB/ octave) will have less overlapping output and slower slopes (6 to 12 dB/octave) will have more. Adjusting the amount of this overlap and the frequencies over which it occurs will adjust the level of response in this region.

Listen while you measure. You hear differently than the sound level meter does. If you hear something that you can't measure, trust your ears. Remember, your system is supposed to sound good, not just make good graphs.

If there is an excess of energy in this region (60 Hz to 100 Hz), raise the high-pass frequency and/or lower the low-pass frequency to shrink the range of overlapping output from the main loudspeakers and the subwoofer. Less overlap means less total output in this range.

If there is not enough output in this frequency range (60 Hz to 100 Hz), do the opposite—raise the low-pass frequency and/or lower the high-pass frequency to increase the overlap. More overlap will increase the total output in this range if the subwoofers are properly phased relative to the main loudspeakers.

If you need more output at the lower portion of this overlap region, adjust the range where the overlap occurs downward. To do that, lower the frequency of both the high-pass and low-pass filters. If you need more output higher up in the overlap region, raise the frequencies of both the high and low-pass filters. Get the idea?

A huge suck-out in the 60 Hz to 80 Hz region may require repositioning of the subwoofers to lift the output level in this range. You may have some cancellation in the overlap between the main loudspeakers and the subwoofers, which can be corrected by repositioning the subwoofers. A huge bump in output over this same frequency range may require that the phase of the subwoofers be reversed (creating some cancellation) to diminish output over this band of frequencies.

If you tinker with these adjustments awhile, you should be able to achieve a fairly flat response curve over the range of 40 Hz to 120 Hz in most any room, along with a good blend between the main loudspeakers and the subwoofers. Response below 40 Hz is largely determined by the capability of the subwoofers and the interface with the room, and other than placement experiments and the use of bass traps (you should be so lucky as to have so much bass output below 40 Hz that you need traps), you won't have much control over the results.

Listen And Tweak

Once you have achieved basic measured accuracy in the transition from the subwoofers to the main loudspeakers, you can make minor adjustments by listening to actual recordings of the material that you will be using the system to reproduce.

Play a variety of software (I start with music recordings and then go to film soundtracks) and tweak the adjustments that you have made, until you achieve a sound that pleases you. That is, of course, the ultimate goal of a home entertainment system—pleasing the listener.

Does The Actual Crossover Point Matter?

While 80 Hz is a good nominal crossover point, it is a compromise and you can feel free to vary this number at will to get the best possible sound. Here is what you're dealing with: (1) A higher crossover point will allow more midrange information to pass to the subwoofer. Subwoofers are optimized for low-frequency reproduction and they don't do midrange well. We get a greater sense of directionality from higher frequencies, and it is not desirable to be made aware of the location of the subwoofer, particularly if only one is used. (2) A lower crossover point allows more low-frequency energy to pass to the main loudspeakers. This will increase intermodulation distortion in the midrange and decrease overall dynamic range by adding stress to the bass driver and the cabinet structure of the main loudspeakers. The greatest sonic benefit offered by adding subwoofers to an audio system is due to the reduced demand for low-frequency output from the main loudspeakers.

External Crossovers

An external crossover that allows adjustment of both the high-pass frequency and the low-pass frequency will provide the most flexibility in set-up and is likely to produce the best sonic outcomes.

Most of the crossovers that are built-in to powered subwoofers have fixed high-pass frequencies and slopes, but you can still adjust everything else and usually get pretty good results. Some high-end controllers offer adjustable crossover points and slopes but most don't. Using the fixed filters in your processor/controller offers a greatly diminished potential for success.

Subjective Tweaks

Want more "slam" and impact? Adjust for more output at about 40 Hz. Want music to sound a little warmer and fuller? Try a little more output at about 80 Hz. Want to damage dishware in the cupboards? Go for emphasis at 25 Hz or so. Above all, have fun!

"Absorbing" Low Frequencies

It is possible to "absorb" low frequencies. Bass traps like the Tube Traps™ from Acoustic Sciences Corporation convert excess low-frequency energy to heat, effectively absorbing the acoustic energy from the room. Removing energy at very low frequencies requires really big traps. It's much better to try to reduce output levels at the problem frequency than to try to remove excess energy from the room.

Flat panels can not effectively absorb acoustic energy below about 200 Hz, regardless of the claims of advertising. Flat, compressed fiber glass panels can damp drywall "boom" somewhat, if the soft material is glued directly to the drywall. Hang-on-the-wall products have no effect at all on low frequencies.

Highs And Lows

They say that everything is relative and nowhere is this more true than in your home theatre system. The amount of bass energy that you perceive is relative to the amount of treble energy (everything else but bass) that is present.

If you have a very bright-sounding room with a lot of high-frequency energy bouncing all around, it will sound like you have less bass. Treating the room to absorb some of this higher-frequency stuff will not only improve the sound overall, it will have the effect of increasing perceived bass response.

Over-treating the room with material that absorbs high-frequency energy may result in a dull, sterile, bass-heavy sound that will be very hard to correct. An anechoic chamber is not a pleasant place to hang out, and neither are many of the over-damped, professionally-built home theatres that I have visited.

Summary

Choose the best subwoofers for the job, based on your priorities. Stiffen the walls in your listening room and absorb problem reflections. Position the main loudspeakers and the subwoofers correctly for their respective purposes. Adjust the crossover to achieve the best balance and blend between the main loudspeakers and the subwoofers. Tweak all the adjustments by ear to make sure that everything sounds good to you. Enjoy. ■

Introduction Part Five

(Originally Appeared In The Essential Subwoofer Buyer's Guide)

More About Filters

After these articles were published in *Widescreen Review* magazine, I received a lot of correspondence asking for clarification about many of the points presented in the series. I have re-edited the four introduction articles in an attempt to make certain concepts a little easier to understand.

One subject that seemed to be particularly confusing to our readers involves the interfacing of subwoofers with the rest of a home theatre system. Further discussion of the crossover networks that are used to combine subwoofers and main loudspeaker systems seems to be appropriate.

Subwoofer Crossovers

The proper blending of the output from one or more subwoofers and the output from the main loudspeakers in a home theatre system requires the use of some kind of crossover network.

A crossover network consists of a high-pass filter that prevents bass frequencies from entering a main loudspeaker, and a low-pass filter that keeps the subwoofer from trying to reproduce frequencies above its optimum range.

Choosing "small" as a designation for any loudspeaker(s) from the set-up menu of your digital controller will cause a high-pass filter to be activated, which will remove bass frequencies from that loudspeaker(s) and redirect the bass from that channel to the subwoofer(s). Some controllers allow the high-pass frequency and slope to be adjusted. Refer to the owner's manual for information.

Subwoofer outputs on most digital controllers are low-pass filtered, although some, like the Acuras, feed a full-range signal to the subwoofer. A full-range signal will require external filtering. Some controllers allow a choice of low-pass frequencies and slopes. Refer to the owner's manual for information.

If you use the low-pass filter provided in the digital controller, you will want to bypass the built-in low-pass filter included in most powered subwoofers. Almost all powered subwoofers provide an unfiltered or "direct" input for this purpose.

With most digital controllers, setting all loudspeakers to "large" will feed full-range signals to each loudspeaker and will pass only LFE bass to the subwoofer output. If you select "no subwoofer" from the set-up menu, the LFE bass content will be mixed back into the other channels. (Check the owner's manual to be sure that your controller works this way.) This configuration allows the use of external crossover networks to divide the signal between the subwoofers and the other loudspeakers.

Using external crossovers offers the advantages of adjustment and tuning described in the introduction to Part Four of the subwoofer review series.

Pitfalls

While low-pass filters are relatively innocuous, high-pass filters may not be. Audiophiles have been reluctant to insert any filter before their main loudspeakers in the fear of screwing-up the sound.

This fear is not completely unfounded, but the practice of allowing the main loudspeakers to run full-range while adding a redundant subwoofer to augment the low bass is a big mistake. I've already told you why but I think that this information bears repeating.

Removing bass energy from your main loudspeakers provides the single greatest sonic benefit obtainable by adding a subwoofer to your system!

Adding a subwoofer to a full-range loudspeaker system without high-pass filtering will probably make the system sound worse. Even with the low-pass filter in the subwoofer set at a ridiculously-low frequency, like 40 Hz, there will still be a thickening of the mid-bass if your main loudspeakers have response extending to below 50 Hz, and you will not benefit from the reduction of intermodulation distortion that can be achieved by reducing bass output from your main amplifier and loudspeakers.

Inserting an active high-pass filter in front of your main loudspeakers, including those filters found in most digital controllers, will probably make the system sound worse too. The entire signal, with the exception of the low bass, will pass through this filter, adding noise, distortion and phase shift.

Digital filters are not the panacea that they are purported to be. A steep-slope digital filter sounds just like a steep-slope analog filter to me. Some high-end surround-sound processors offer first-order digital filters that might work well, however.

So how do you reduce bass output from your main amplifier and loudspeakers without negatively impacting the sound? With a passive, single-pole, high-pass filter.

Passive High-Pass Filters

A passive high-pass filter consists of a single value of capacitance applied at the input to the amplifier. This capacitance, in series with the input impedance of the amplifier, attenuates (rolls off) the lowest frequencies of the signal as it enters the amplifier. Attenuating the lowest frequencies shrinks the amount of current demanded from the amplifier and reduces the excursion of the woofer in the main loudspeaker, lowering intermodulation distortion.

There is probably a coupling capacitor in series with the input of your amplifier already. (And if there isn't, there should be.) Adding a little more capacitance to raise the high-pass frequency to about 80 Hz will improve the sound from your system so much that you won't believe it 'till you try it. All that will be missing will be the low bass, and the subwoofer(s) will take care of that.

Will a passive high-pass filter work with an active low-pass filter even if the low-pass portion is third or fourth-order? You can bet it will. I listened to every subwoofer in this series configured in that manner. Many high-end outboard crossover networks work just this way—with a passive first-order high-pass section and an active third or fourth-order low-pass section.

Almost every powered subwoofer has a built-in, active low-pass filter, so you probably already have one of those. All you really need is the high-pass filter. So where do you get one?

You can buy passive high-pass filters from M&K, Mirage, and Vandersteen dealers, and many other manufacturers offer them as well. Check with a knowledgeable dealer and try to find one who will loan you some filters with different values so that you can experiment with crossover frequencies. Don't let the dealer talk you in to an active high-pass unit unless you have small satellite loudspeakers. If you have small satellite loudspeakers you are probably better off using the high-pass filter built-in to your digital controller. Why buy anything else?

A passive high-pass filter is dependent on the input impedance of the amplifier to establish a crossover frequency. You must know the input impedance of your amplifier in order to determine the value of capacitance needed to achieve a suitable crossover point. Most manufacturers will specify an amplifier's input impedance in the owners manual. 50 kΩ is typical for solid state amplifiers.

Bass Management

What about the bass management system built in to your surround sound processor? Wouldn't it be much simpler to just set the loudspeakers to "small" in the processor controls and forget about it? Of course it would be, and you don't have to feel bad if you do it that way. If your system is used primarily for film watching and you're not seeking the highest fidelity sound, just use the bass-management system and be done with it. But if you want the best sound don't use the crossovers in your digital controller.

As I described in the fourth installment of the subwoofer series, the main problem with the filters in processors is the lack of adjustability. If you can't vary the frequency and slope of both the high-pass and low-pass segments of the filter, you are unlikely to get a good blend between the subwoofers and the main loudspeakers along with flat response in the room.

Another problem with bass management is the drastic way in which the signal is manipulated. Read my article about digital controllers in Issue 32 of *Widescreen Review* to see just what happens to the signal when it is sifted through a multitude of filters during bass management. This process may be good enough for filmmakers but it's not good enough for me.

Conclusion

This special edition contains all that you need to know in order to make an informed choice when purchasing a subwoofer. You can also refer to this guide when installing the subwoofer(s) in your system.

So what are you waiting for? Get in your car right now and go buy at least two powered subwoofers. Your life will be happier from the moment that you hook them up. ■