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*This is **Audio Perfectionist Journal** #2. In this edition we will examine acoustics in the home environment. The discussion of acoustics involves everything that happens after the loudspeakers convert the electrical signal into mechanical energy.*

The electrical signal that passes through the components in your audio system must be converted into mechanical energy in order for you to perceive that signal, which represents the original recorded event, as sound. How and where this energy is introduced into the room and what happens to it after that introduction is vitally important to the quality of sound that you will hear. Where you place the speakers and your listening chair, and how you treat the surfaces of the listening room, will have a greater cumulative impact on sound quality than all the sonic characteristics of all the expensive components in your system combined.

Can't Buy Me Love

I have heard truly great—and often very expensive—audio systems making absolutely dreadful sound so many times I could write a book about the subject. In fact, I guess that's what I am doing. The primary reason that so many people get bad sound from good equipment is that nearly all emphasis from dealers and the media has been placed on the quality of the audio components and very little has been said or written about the necessity of proper set-up, placement and use of these components.

The sole objective of the audio industry as a whole is to sell products. The goal of the audio salesman peddling his wares in a retail store is the same as the motive of the magazine reviewers who write about the stuff: they want you to continue to buy new components. Selling products pays their bills and your satisfaction is often a minor concern—if they think about it at all. They may actually hope that you will not be satisfied by the products they recommend. If you are happy with what you hear at home, you may not come back to spend more.

> ***"The sole objective of the audio industry as a whole is to sell products."***

If you are dissatisfied with the sound from your system, the cure they offer is always a new and "better sounding" component or a new "magic" accessory. Many hobbyists are caught in this loop. They spend more time researching components and "upgrading" their systems than they do listening. Like a dog chasing his tail, they will never reach the sonic nirvana they seek unless they first learn how to properly use the equipment they already possess.

Before you can truly benefit from a legitimate component upgrade, you must get the set-up fundamentals right and these fundamentals start with speaker placement and room treatment. Make no mistake about this fact: no component or gadget can completely overcome the effects of poor speaker placement or bad room acoustics. Get these things right and you can have pretty good sound from even a modest collection of well-chosen components. Get these things wrong and you'll be wasting time and money trying to find a magic cure.

Component Sound versus Room Sound

Choosing and refining the system components in order to more accurately reproduce the signal is a prominent part of the search for good sound and future **Journals** will be devoted to these subjects. Proper speaker placement and room acoustic treatment are even more important and this issue is devoted to the subject of acoustics which encompasses these things.
In **Journal #1** we established that the human ear/brain mechanism can distinguish between the sound originating from properly positioned speakers and the sound contributed by the room at frequencies above the bass region. The goal of optimal speaker placement and room acoustic treatment is very simple: make this mental task as easy as possible. The less work your brain has to do in order to hear the desirable sounds of the recording—separated from the sounds contributed by the acoustics of the room which aren't part of the performance—the more pleasurable the listening experience will be. While the goal is simple, accomplishing it may require some experimentation and a little effort on your part.

It would be wonderful if I could tell you exactly where to put your speakers and exactly how to treat your wall surfaces. Unfortunately, the broad variety of speaker types and the virtually unlimited possibilities for acoustic aberrations to be contributed by your unique environment make this an unrealistic goal. What I will try to do is explain the sonic effects of speaker placement choices and various room acoustic treatments, along with some special considerations required by certain speaker types, so that you are less likely to become lost on your journey to good sound. I'll show you a diagram of one of my own listening rooms and tell you what I've done to make the system in that room sound great. I'll tell you which aspects of speaker placement and room treatment have proven to have the most significant audible effects based on my experiences in thousands of listening rooms. And I will tell you how to eliminate almost all bass problems.

After I share some of my experiences and give you some guidelines, you'll have to do the grunt work required to make good sound in your room. Some experimentation will be necessary. Believe me, you will be rewarded in direct proportion to your efforts.

In This Issue

The article titled *Where Do I Put All This Stuff* describes the proper positioning of loudspeakers for optimum stereo reproduction and surround sound listening. It explains the trade-offs that you will encounter with various configurations and it leads into the article titled *Subwoofers From a High-End Perspective* which states my position on the use of subwoofers for high-end audio. This subwoofer article leads to a review of the *Vandersteen 2Wq* powered subwoofer. That review is followed by the article *How to Get Near-Perfect Bass in Any Room*. The title is self-explanatory.

The article titled *Good Acoustics in Real Living Rooms* describes how to get good sound using acoustic treatments with a high spousal approval factor. High SAF simply means "not ugly beyond belief." After all, nobody over the age of twenty wants a living room that looks like a recording studio, or worse.

Coming Up

This issue is primarily concerned with the vitally important subject of acoustics. Future editions of the **Journal** will concentrate on the matter of optimizing the system components in order to reproduce a good replica of the input signal. We'll trace the signal path through the system and describe what happens to it as it passes through each component including the cables.

We'll discuss how each component in a home audio system works and I'll tell you which types I prefer and why. You'll learn about a lot of stuff that you shouldn't waste your money on and why.

We'll examine measurements and discuss the ones that are the most revealing and I'll tell you how to perform some of your own tests. You'll learn how to interpret the test graphs printed by some of the magazines and what to listen for when auditioning components.

I'll review specific products of exceptional merit as we go along, with the emphasis on why these products offer superior performance. The knowledge you gain will help you to become a more informed consumer. That way, you can spend less money on your audio hobby and enjoy it more. APJ

Where Do I Put ...ALL THIS STUFF

It is important to choose the best audio components. Utilizing those components properly is even more important. While it is very easy to make bad sound with good equipment, equipment that is merely adequate can provide excellent sound with proper attention to set up and acoustics.

Anecdotes About Music Lovers & Audiophiles

Brad and Brittany are not rich but they are both intelligent professionals who enjoy music, and their earnings allow them to buy nice things. They consider themselves to be discriminating buyers and they choose their purchases carefully. They seek the best quality but they don't want to spend frivolously. They want products with intrinsic merit, not just prestigious brand names.

After many hours of listening comparisons in the dealer's showroom, Brad and Brittany select an audio system comprised of perfectly matched components that would please any audiophile, and they write a check for $30,000.

When the installer arrives to set up the system, he finds a room with a tile floor, an area rug and huge expanses of uncovered glass stretching up to meet a cathedral ceiling. He is informed of the interior designer's decision: "one speaker goes over there under the stairs and the other one goes against the wall next to the TV set."

You may have met Brad. He's the guy who says "this stuff never sounds the same at home as it does in the store."

Chuck buys the Stereophile issue containing their recommended components list every year. He thinks that if only he could afford all the "Class A-rated" components he would be eternally happy. His listening chair sits against a bare wall across from his speakers which stand at a 110° included angle because they are positioned on either side of a sliding glass door. One speaker sits just a foot away from a room corner.

Chuck has invested thousands of dollars in his "Class-B" components but seldom listens to music because he hates the sound of his system. He plays the same three discs over and over as he compares one new component to another while seeking the magic upgrade that will fix his system and change his life.

If you buy or sell used audio equipment you may have met Chuck. He's always trying to get a deal on products that he can't really afford because he's convinced that his dissatisfaction is caused by his budget limitations.

> ***"...better components sound better under the right conditions, but the sonic advantages they provide can be swallowed up by bad acoustics."***

Bill spends far more time reading audio magazines and hanging out at his local high-end store than he does listening to music. He is the first to buy each new magic audio accessory that the magazines recommend and he has spent over $10,000 on cables alone, but he still tries each new and "improved" model that comes out.

Bill can spend hours discussing the difference in sonic nuance provided by various amplifier designs but the last time he changed his speaker wire he connected one speaker out of phase and didn't know it until a salesman, who was visiting his home to deliver the Ultra Gizmo Signature power line conditioner, discovered the error.

Bill told all his audiophile buddies about the fabulous "spatial" improvements he got by installing the new speaker cables. He used the same ambiguous words coined by the writer whose review had convinced him to purchase the cables in the first place.

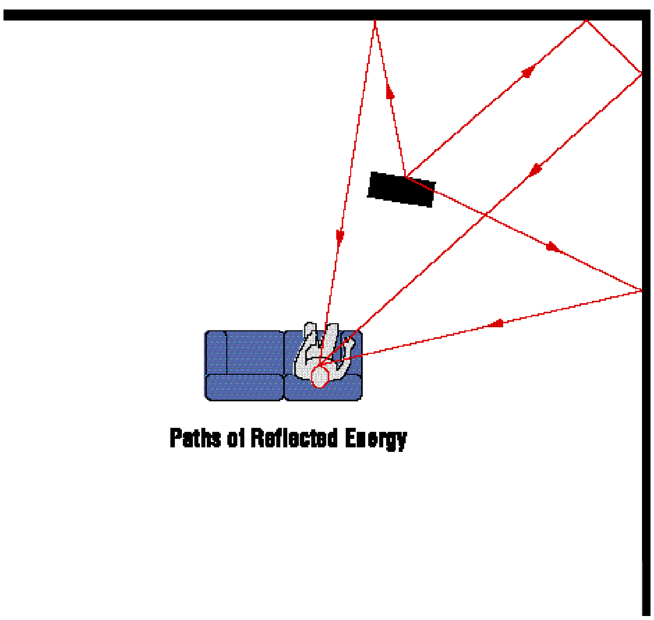
Don't Follow in Their Footsteps

Brad, Brittany, Chuck and Bill are not their real names (or maybe they are). While the names may have been changed, these people and situations are real and they are not unique. I've met hundreds of folks just like these over the years. They have missed the important message that I'll repeat again here: where you put the speakers, where you sit, and how you prepare the surfaces of your room have a greater effect on sound quality than the components you buy to make the sound.

Of course, better components sound better under the right conditions, but the sonic advantages they provide can be swallowed up by bad acoustics (or wiring the speakers out of phase). Let's start off with a discussion of speaker placement. Where you place the speakers determines where the mechanical energy is introduced into the room.

Speaker Positioning

If you have a normally furnished room with carpet on the floor and drapes over the windows, and you position the speakers and the listening chair at least ten feet away from any wall, with a 60° (or slightly less) included angle between the speakers as



Paths of Reflected Energy

viewed from the listening position, you'll probably have good imaging, good detail and good tonal balance above the bass region regardless of room acoustics. If you have any bass at all, the response will be uneven. Moving the speakers closer to the front wall will increase bass output at the expense of midrange detail and image focus. Placing the speakers and/or the listener closer to the room boundaries will cause the sonic contributions of the room to become a more prominent part of the overall sound.

Sound travels at about 1,100 feet per second at sea level, which is about one foot per millisecond (1/1000 of a second). A reflection from a wall that is ten feet away from a speaker will arrive at the listener at least 20ms after the direct signal from that speaker because the reflected sound must travel from the speaker to the wall (10 feet=10 ms) and back (another 10 feet=another 10ms), in addition to the distance from the speaker to the listener. The *Paths of Reflected Energy* illustration below shows some of the many possible paths of reflection.

The intensity of sound (in free space) varies inversely with the square of the distance from the source to the listener, so the reflected sound will be attenuated in amplitude as well as delayed in time. Ten feet between each speaker and the nearest wall will delay and attenuate the reflected signal from these walls (relative to the direct signal from the speakers) and make it easier for your brain to hear the sound from the speakers and the sound from the room as separate entities at frequencies above the bass region.

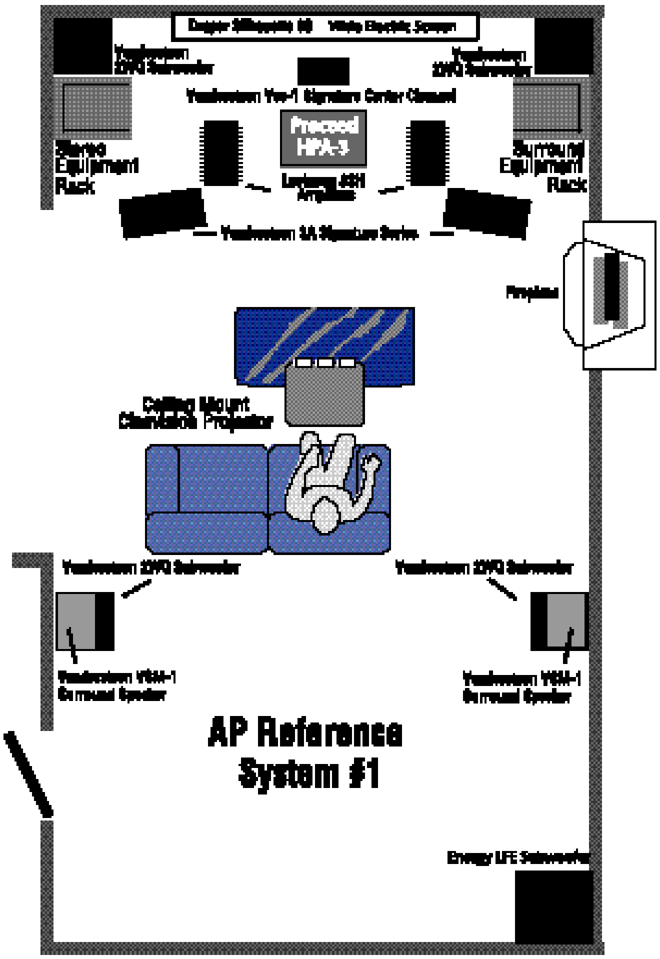
What's that you say? You live in a regular house and the rooms aren't large enough to allow that and, besides, you like bass. Well, I live in a regular house, too, so let's take a look at one of my listening rooms and see where I have positioned my speakers and talk about why I put the components where I did.

My House

I live in a four bedroom, three bath, two story tract house in Southern California. I have complete stereo/home theater systems in two downstairs rooms and a baby grand piano in a third. One of the stereo/home theater reference systems that I use to review components resides in a dining room/den area that is about 13 feet wide and 24 feet long. The AP Reference System #1 illustration on page 5 is a drawing of that room which is roughly to scale. The room design avoids many sonic pitfalls that I have observed in typical homes

In the decades when I was a retail merchant, I got the opportunity to view many living rooms and hear the effects of speaker placement and room acoustics. During visits to customers' homes I have observed that most people sit too far away from their speakers and too close to the rear wall to hear the best possible sound. Stereo speakers are frequently spaced too far apart for good image focus. Main stereo speakers are typically positioned too close to the front wall to deliver the best sound possible.

In surround sound systems, the direct pathway from surround speakers to the listeners is frequently blocked by furniture or other individuals seated side by side, and center channel speakers are often placed in line with the front speakers rather than at an equal distance from the listener which would leave the front three speakers arranged in an arc as viewed from above.



This is a drawing, approximately to scale, of a real system (at the time of this writing) in a real home: mine.

I managed to avoid all these errors in my room by carefully considering the space I had to work with, but I ended up with the one single configuration in which this was possible. In this long, narrow room, there is virtually no alternative to the way the components are positioned. Is this unusual? Hardly. Few people I know can afford either the expense or the space required to have a dedicated stereo or home theater room. Most of us will have to fit our audio toys in among our other furnishings, within the confines of the dwelling in which we live. Is this the end of the world as far as fidelity is concerned? Absolutely not.

If you could hear the system I'm describing here you would know that I have accepted very few, if any, sonic compromises in the room shown in the illustration. The sound is breathtakingly good for both critical music listening and film watching, and it didn't take much alteration to what is a fairly typical room to get that level of sound quality. You don't need a purpose-built room in order to get exemplary audio performance from well-chosen components. You do need to carefully assess the space that you have available and make the best of your circumstances. You can do that on paper.

The room shown in the illustration was designed using a sheet of graph paper on which I made a scale drawing of the room. I then made scale model cut-outs of each speaker, major component and equipment rack from additional pieces of paper. This method allows experimentation with various positions for the equipment by simply moving the paper cut-outs around the paper room. You can do this, too.

Try to achieve the goals described in the rules that follow with emphasis on the rules at the front of the list and attention paid to the special considerations needed for certain speaker types.

Following is a list of rules for speaker placement. They are in order of importance but any one may prove to be impractical in your room.

Ten Rules for Speaker Placement

The goal of optimally positioning the speakers and the listener within the room is to make the sound from the speakers more prominent and the sound from the room less prominent.

Here are some basic rules for speaker placement that have proven to be generally successful in my experience. Any rule may be impractical to implement in your specific circumstances and there is one overriding rule that can negate any of the others: if it doesn't work in your room, don't do it. I have tried to organize these rules in order of importance but it's hard to choose between the first five—they are the most important by far. If you must compromise, simply do the best you can and we'll deal with necessary corrections in the article about room treatment.

1. Place the speakers at least four feet away from the front wall.

Three feet is an absolute minimum distance, in my opinion, and six feet out is usually better than four. The most prominent reflection that you will hear in nearly any room comes from the front wall. If the speakers are too close to this wall, imaging will suffer and midrange detail will be obscured.

Never put bookshelf speakers on a bookshelf! Place them on stands three or four feet out from the front wall, just like floor-standing speakers.

In the past, it was necessary to experiment with the distance between the speakers and the front wall in order to achieve the best compromise between obtaining good midrange detail and imaging, and realizing the best bass performance. If you don't use subwoofers, as I recommend, you'll still have to do this.

I'm going to make a controversial statement right here: trying to get the best bass and the best imaging from the same speaker is like chasing a mirage in the desert. You may think that you are getting close, but you'll never arrive. I gave up trying to do this many years ago when I discovered the benefits of removing the bass energy from "full-range" speakers and redirecting it to a pair of good subwoofers. A true, high-end audio system requires the use of two amplified subwoofers and I'll make my case in an article in this issue. *Subwoofers From a High-End Perspective* explains my position.

Why settle for either compromised bass or compromised imaging? Bass should be introduced into the room in a pressure zone and all other frequencies should not be. A full range speaker utilizing the same driver to reproduce bass frequencies and the lower midrange cannot properly accomplish these conflicting tasks no matter where that driver is positioned.

Put a pair of subwoofers in the corners and place your speakers in the position that provides the best midrange detail and imaging and you can have it all. If you don't have subwoofers yet, move the speakers fore and aft relative to the front wall to achieve the best compromise between bass performance and imaging.

2. Place the speakers so that the included angle between them as viewed from the listening position is 60° or less.

A 60° included angle means that the distance between the centers of the speakers will be equal to the distance from each speaker to the listener. This is the maximum spread that I recommend. I prefer to position my speakers slightly closer together than this. In the room shown in the illustration, the speakers are 9-feet from the listener and 7-feet apart, center-to-center.

Spreading the speakers farther apart does not make the image bigger, it just makes it more diffuse. Time- and phase-correct speakers will produce an image that is not limited in width by the outside edges of the speakers. This image will extend out beyond the outside edges of the speakers. Placing the speakers slightly closer together than the distance from each speaker to the listener usually won't diminish image width and usually will improve image focus.

Think of the speakers as a window through which the performance is viewed, not as the defining boundaries of where the performance is occurring and you'll get a new mental perspective on the question of where to put the speakers.

3. Sit as Close to the Speakers as Practical.

The drivers in my speakers coalesce in time and phase at a point eight feet in front of each speaker, which is the minimum recommended listening distance. I sit nine feet away from the speakers. My mind looks through the speakers to the performance which appears to occur behind, between and beyond the speaker boundaries.

Sit closer to the speakers and farther from the room boundaries for better sound.

4. Put your listening chair as far away from the back wall as possible.

You want the energy that is reflected from the back wall to be delayed and attenuated, too. There's more about that coming up in a few paragraphs under the heading *The Listening Position*.

5. Place subwoofers in the front corners of the room behind the speakers they augment.

This subject is discussed in detail in the articles titled *Subwoofers From a High-End Perspective* and *How to Get Near-Perfect Bass in Any Room*, in this issue.

6. Place the speakers in a symmetrical environment.

Complete symmetry in the environment around the speakers is a desirable goal that is seldom possible to achieve. A symmetrical environment means that the walls beside and behind the left speaker should be duplicated by mirror image walls beside and behind the right speaker. I am fortunate to have that situation in my room but many living rooms simply won't allow it.

In my experience, what's behind and directly beside the speakers is more significant than what's farther in front of the speakers, towards the listener. What's behind the speaker is more important than what's beside it, if you have to choose.

7. Place a center channel speaker behind the line between the front left and right speakers so that all three are the same distance from the listener.

As described in Issue #1, center channel speakers tend to collapse the illusion of image depth. Placing the center channel speaker slightly farther from the listeners than the left and right channel speakers helps a little. Turning it off will help even more. Read Issue #1 for a discussion of the trade-offs.

8. Place surround speakers above and behind the primary listening position.

The theoretical ideal of using identical speakers all around for surround sound is neither practical nor a good use of money. Surround speakers primarily reproduce sound effects and ambient sounds. You don't need the world's best speakers to effectively convey these signals.

Rear channel speakers do need a direct, unobstructed path to each listener's ears. They must be positioned above and behind the primary listening position so that chair backs and other bodies will not be between you and the rear speakers. Floorstanding rear speakers almost never work to provide the best surround sound for this reason.

The International Telecommunications Union (ITU) specifications call for front speakers to be positioned about 30° left and right of center and surround speakers to be positioned about 110° left and right of the center channel speaker. This specification has always worked well as a good point from which to start.

9. Forget about floor bounce and standing waves.

Floor bounce refers to the reflected signal that bounces off the floor between you and the speakers. This is usually the first room reflection that arrives at the listener's ears. Standing waves are the popular explanation for the peaks and valleys that appear in a measurement of "in-room" frequency response.

Floor bounce is a "red herring" in a carpeted room. Every terrestrial-based sound that you have ever heard in your life has included energy reflected from below, or "floor bounce."

The designer of your speakers should have considered floor bounce when the speakers were voiced. The engineers who equalized the recordings you play listened to them in a room with floor bounce. If floor bounce hasn't been dealt with before you get their products it's their fault, not yours.

Floor bounce is a fact of life, as far as a measuring microphone is concerned, but I believe that your brain expects to hear it and has learned to accept it as part of the natural world. I don't think that it is desirable to try to "correct" for it beyond the attenuating effect provided by a heavy carpet, which is required opposite an acoustically reflective ceiling anyway.

Standing waves only cause audible problems at bass frequencies and virtually all these problems will be corrected by placing two subwoofers in the front corners of the room. Read the article *How to Get Near-Perfect Bass in Any Room* in this issue for more information.

Trying to calculate the best positions for speakers using computer software is a less-than-useless endeavor. You will always get the wrong results, but you may be so convinced they are correct that you won't experiment.

The distance between the speakers and the front wall is the most important dimension in terms of audible change and it is likely to be the only one that you can alter anyway.

10. Tweak by ear.

You can fine-tune the angle between the speakers by removing the listening chair, crouching down to the proper listening height and moving backward and forward while listening. If the best image is achieved behind the position where you want your chair, move the speakers closer together. If image focus is better when you are forward of your chosen listening position, move the speakers farther apart.

Toe-in and the distance between the speakers are interactive parameters. Adjust toe-in by ear and then adjust the spacing between the speakers by moving back and forth, and then check toe-in again. You may have to do this a few times to get both adjustments right. Toe-in can affect midrange tonality as well as image focus.

The first-order speakers that I prefer are very critical of listening height and tilt-back. Multiple drivers can be physically aligned and blended together using crossovers with first-order acoustic slopes to deliver all frequencies to the listener with the correct time and phase relationships. This allows the original waveform to be accurately reproduced but it only works through a limited vertical listening window. If your head is higher than it should be or if the speaker is tilted forward more than it should be, your ears will be closer to the tweeter and farther from the woofer affecting the time alignment between these drivers. And vice-versa.

You can fine-tune the degree of tilt-back by raising and lowering your head while listening. If the sound improves with your head 6 inches above your normal listening height, tilt the speaker forward. If the sound is better with your head below the normal height, tilt the speaker back.

All this tweaking may be tedious but it's not difficult and you can be listening to music while you do it. So stop complaining and get to work. You'll be glad you did.

Special Notes About Specific Speaker Types

Not all speakers are created equal and certain types need special attention. Here is a by-no-means-complete list of speaker design types that may require a little more consideration for optimum placement or room treatment.

Dipolar speakers

A dipole speaker has a figure-of-eight dispersion pattern with the rear wave out of phase with the front wave. Little energy is radiated directly to the sides. Full-range planar speakers are usually true dipoles.

Full-range speakers with real dipolar radiation patterns, like planar-magnetic, ribbon and electrostatic designs (not hybrids), are especially critical of position relative to the front wall. The rear wave from these speakers will reflect directly back from the wall and interact with the speaker causing a greater sonic effect than what you'd get with conventional designs. Moving the speakers closer to or farther away from the wall can have a profound effect on the midbass and lower midrange frequencies as well as bass and you'll have to devote more time to experimentation.

True dipoles radiate little energy directly to the sides but tonal balance is very different on- and off-axis, so the wall directly to the side of the speaker is of less concern, but the first side-wall reflection coming off the wall forward of the speaker position becomes more critical.

Dynamic speakers with open-back midrange drivers can't be treated like real dipolar radiators. They have a quasi-dipolar radiation pattern in the midrange (depending on cabinet structure) and remain point source radiators in the bass and treble. They simply splash more midrange energy off the front wall behind the speakers and behave more like bipolar designs in this regard. Less midrange energy is radiated directly to the sides of each speaker but the bass and treble portions of the spectrum are strongly radiated to the sides, so side wall reflections will have a very different tonal signature than the direct sound from the speakers.

Some speaker designs use open-back midrange drivers and a second tweeter aimed to the rear and wired out of phase with the forward-directed tweeter. This type combines all the worst characteristics of dipole and bipole designs. Try to absorb everything but the direct radiation from the front for the best results.

Dynamic speakers with open-back midrange drivers are often touted as being less room-sensitive than other types but in my experience just the opposite is true. Lots of sound-absorbing material on the walls is called for. After you absorb all that rear-directed energy and hear how much better things sound, you may wonder why you bought a speaker that created that energy in the first place.

Bipolar speakers

A speaker with a bipolar radiation pattern directs sound forward and rearward with both waves in phase as they leave the speaker. A bipole is like two conventional speakers placed back-to-back and has very broad dispersion.

Dr. Bose has demonstrated that many undiscerning listeners like the sound of artificial ambience as provided by speakers that purposely direct lots of energy towards the walls. This conclusion was confirmed by those infamous listening tests done at the NRC in Canada. Be that as it may, this is, in my opinion, the antithesis of what is desirable for high fidelity reproduction.

Speakers with a bipolar radiation pattern direct at least half the acoustic energy to the room boundaries where it will be reflected back to the listener from all angles with varying time delay. This added “ambience” smears definition and detail. An artificial sense of spaciousness is created at the expense of a focused, dimensional image and any real resolution.

Here is my advice: If you don’t own bipolar-radiating speakers, don’t buy them. If you already have, try to soak up as much of the rear-and side-directed energy as possible with absorptive room treatment. Positioning the speakers even farther from the room boundaries helps.

In my experience, bipoles work better than dipoles because bipoles have similar tonal balance on- and off-axis. At least the reflections mimic the tonal balance of the direct sound with bipoles, which is not the case with dipoles. In general, dipoles image better and bipoles have better in-room tonal balance. Neither type offers completely satisfying performance in my opinion.

D’Appolito arrays

Speakers with midrange drivers above and below the tweeter are called D’Appolito arrays or MTM (midrange, tweeter, midrange) designs. Depending on the crossover design, there may be a dissimilarity between on- and off-axis response in the horizontal plane. Unlike dipole designs, D’Appolito arrays do radiate significant energy directly to the sides and this off-axis energy may or may not have a similar tonal balance to the on-axis response. This speaker type may need to be positioned farther from the side walls than others, or absorbent material may need to be applied to both side walls directly beside the speakers.

If you have the space, positioning D’Appolito arrays along the long wall of the room may prove to be beneficial.

Listening height and tilt-back are important considerations with speakers that incorporate a D’Appolito driver array. The listener’s ears should be exactly equidistant from each midrange driver for best performance.

Hybrid Line-Source Speakers

Speakers using dynamic (cone) woofers combined with electrostatic, planar-magnetic or ribbon high-frequency drivers, which are long enough to simulate a line-source, have a special problem: bass and treble energy can only be in balance at one specific listening distance.

Sound from a true line-source varies inversely with the distance from the source in a linear manner. Sound from a point-source varies inversely with the square of the distance from the source.

As the listening distance is increased from a point-source/line-source hybrid speaker, bass energy will diminish more rapidly than treble energy. Reflected energy from any given room surface will have a different tonal balance than energy reflected from any other room surface if the path lengths differ. All reflected energy will have a different tonal balance than the direct signal emanating from the speakers. Placement will always be a compromise no matter what you do.

If you have hybrid speakers with long tweeters, start by determining the correct listening distance for balanced sound and then scale the other dimensions from there.

Conventional designs

Conventional multi-way loudspeakers with smooth response on- and off-axis need less special attention than other types. Side-wall reflections will be less problematic because the reflected energy will have tonal balance similar to the direct sound but will come from a different direction. The brain can easily identify this as a reflected replica of the sound from the speakers.

Reflections coming from the front wall may still cause auditory confusion if they arrive from a direction very near the speakers with insufficient delay. Room treatment can fix this problem.

The Listening Position

Pressure zones extend out a couple of feet from each of the room boundaries. If you place a measuring microphone in one of these zones you'll probably measure smoother overall frequency response with a rising bass curve compared to a measurement taken farther from the walls. Based on these facts, the pressure zone near the back wall would appear to be a desirable place to sit but listening tests will prove otherwise.

If you sit near the back wall, you will hear a multitude of reflections from that wall and they will arrive at your ears with very little delay or attenuation, immediately following the direct signal from the speakers. This will smear the desired signal, reducing definition and image focus and altering harmonic structure. Unless your speakers have a response curve that falls from the mid-bass down, you will hear a gradually rising output at lower frequencies when sitting near the back wall.

You can't completely cure this problem by placing absorbent material on the rear wall behind the listener because all frequencies won't be equally affected. Diffusion doesn't work well when distances are small. The sure cure is to place your listening chair well away from the back wall.

Here is some unambiguous advice: never sit against the wall.

Component Placement

Cables are critical components that have an impact on signal integrity. A cable of a given length with more current flowing through it has a greater sonic impact than one of equal length that is conducting less current. Shorter cables have less effect on the signal than longer cables. Cables feeding high-impedance loads have less impact than those feeding low-impedance loads.

Power amplifiers typically have an input sensitivity of 1 to 1.5 volts and an input impedance of about $50k\,\Omega$. Loudspeakers are typically in the range of 4 to $8\,\Omega$ impedance and may require momentary peaks of 100 volts or more. Almost no current flows through an interconnect cable feeding a high-impedance load like a power amplifier. Lots of current flows through speaker cables to drive low-impedance loudspeakers.

You should position your components as close together as practicality allows to minimize all cable lengths. Long interconnects and short speaker cables will generally sound better and cost less than long speaker cables and short interconnects.

Position your amplifiers right next to the speakers and you'll get better sound and better appearance while saving money because, in addition to having a greater impact on the signal, speaker cables are bigger, less attractive and generally more costly than interconnect cables. APJ

SUBWOOFERS

From A High-End Perspective. . .

Suppose I told you that you could add two components to your system that would reduce intermodulation distortion in the midrange by a factor of two or more, dramatically improve the resolution of midrange and high frequency detail, double or triple the dynamic range capability of your system without changing your existing amplifier or speakers and improve imaging more than you can imagine. You would probably be interested, right? But wait, there's more.

These same components would allow the amplifier to maintain tighter control over the speakers in the mid-bass and lower midrange. They could extend bass response to infrasonic frequencies while lowering bass distortion and improving the system's ability to accurately convey the rhythm and pace of music. And these same components could virtually eliminate the uneven response at lower frequencies caused by room standing waves.

Does all that sound too good to be true? Are you concerned about the possible cost of all this improvement? If all this is so easily achievable, are you wondering why you've never heard about it before?

Let me assure you that all these sonic improvements can be yours and I've been conservative in my estimates of the level of audible improvement you'll get. You can have all this for $2,500 and you can upgrade in two steps of $1,250 each. If you are starting from scratch, you may actually reduce the cost *of a complete system by purchasing a less expensive amplifier and a lower cost speaker model, along with these components, and end up with better overall performance. Few people have figured this out and fewer have spread the news, but it's all true.*

Of course the components I'm talking about are a pair of powered subwoofers—but not just any subwoofers. These subwoofers need to have some special characteristics which we'll get to in a minute.

Subwoofers?

I'm sure you are shaking your head in disbelief right now, and thinking that I've lost it. You may have auditioned some popular subwoofer models and been less than impressed with their performance and I won't disagree. Most subwoofers available today are simply unacceptable for use in a system designed for critical music listening.

Yes, we have all heard those thunderous thudpuckers, commonly called subwoofers, that add to the excitement of movie sound and simply ruin the sound of music. How can I claim that these things can actually be beneficial in a high-end audio system?

Here are two reasons that your experience may conflict with my statements: most subwoofers weren't designed for good musical performance, and most dealers set subwoofers up poorly, on purpose.

When properly integrated with the system, subwoofers blend seamlessly with the main speakers and don't make their presence known. But that's a very hard sell to the average consumer and selling is the name of the audio game. Subwoofers are supposed to add bass, right?

After their initial forays into the market, few manufacturers continue to try to make subwoofers that accurately represent music. Why try to educate consumers when it's easier to just give them what they think they want? Boom!

Subwoofer makers soon learned what dealers had already figured out: if they can't hear it woof they won't buy it. Manufacturers started to build subwoofers with high-Q alignments and vents in order to provide more "slam." Dealers started to set up their demonstrations for maximum thump, and maximized sales figures. Awareness of the basic concepts of specialized bass reproducers faded or was suppressed.

Home theater exacerbated this situation. People today expect a subwoofer to rattle their fillings and the exaggerated bass that most subwoofers deliver is incompatible with accurate music reproduction.

But there is more to bass than boom—bass is the foundation of all music. And there is more to subwoofers than bass. They reproduce bass frequencies to be sure, but bass extension is possibly the least of the sonic benefits offered by good powered subwoofers.

Why Good Subwoofers Improve Sound

In order to provide the benefits mentioned at the beginning of this article, subwoofers must utilize a dedicated bass amplifier, and the main amplifier and speakers must be high-pass filtered using a passive, first-order device.

A high-pass filter does just what you would expect: it allows frequencies above the cut-off point to pass, and blocks frequencies below that cut-off point. The attenuation of a first-order filter is 6dB per octave. That means that the signal will be reduced in amplitude by 6dB, one octave below the crossover or cut-off point. If the crossover point is 80Hz (-3dB), the signal level at 40Hz will be -6dB relative to the signal level at 80Hz, and -9dB relative to the signal level in the midrange. The signal amplitude will continue to fall at a rate of 6dB per octave as the frequency decreases.

A passive, single-pole filter at the input to the amplifier is the only sonically transparent way to high-pass the main speakers.

The subwoofer amplifier will require a low-pass filter to prevent frequencies above the selected crossover point from being sent to the subwoofer. A low-pass filter passes low frequencies and blocks higher frequencies.

A subwoofer with an internal amplifier, commonly called a powered subwoofer, will require electronic compensation to allow both infrasonic response and acceptable dimensions for home use.

Given these stipulations, a pair of powered subwoofers can provide the following benefits:

1. Better performance from your speakers.

Full range loudspeakers utilize the same driver to reproduce both the bass range of frequencies and at least part of the midrange. For optimum reproduction of midrange frequencies little cone movement is required, and a relatively small driver is necessary to provide quick response and good dispersion.

Low frequencies require lots of air movement, demanding greater cone area and more cone movement. In engineering terms, the back-and-forth movement of the cone is called excursion. Cone excursion quadruples with each halving of frequency.

Good midrange reproduction requires the use of moderately-sized drivers and good bass reproduction requires lots of cone area, so most full range speakers compromise the quality of both bass and midrange by utilizing woofers that are too small to provide good bass yet too large to deliver the best midrange quality.

The cone of the 8-inch or 10-inch woofer typically found in a full range loudspeaker will be required to make peak-to-peak excursions of perhaps an inch to deliver audible levels of output at 40Hz and it will have to do this while producing 300Hz (or higher) midrange signals at the same time. A 6.5-inch woofer will make a better midrange driver but it will have to work even harder to deliver low frequencies and IM distortion in the midrange will rise.

Intermodulation distortion occurs when one frequency modulates (alters by its frequency) another. Peak-to-peak cone excursions of an inch or more, which may be required to reproduce a 40Hz signal, will have a substantial effect on a signal at 300Hz. The 300Hz signal will increase slightly in frequency when the cone is moving towards the listener to reproduce the 40Hz portion of the signal, and decrease in frequency when the cone is moving away from the listener. This is only one mechanism of IM distortion, which is sometimes called Doppler distortion. There are other forms of IM distortion.

All dynamic drivers exhibit some nonlinearity in outward versus inward cone movement. High cone excursion exacerbates nonlinear driver response and causes harmonic distortion. Harmonic distortion occurs when a harmonic (multiple) or sideband of the desired signal is produced due to nonlinear behavior of the electrical, magnetic or mechanical mechanism of the driver. If you want to reproduce 40Hz and you get some output at 160Hz as well, that's harmonic distortion.

The results of high excursion of the woofer cone are intermodulation distortion of the midrange signal and increased harmonic distortion of the bass signal. And there's more.

The small woofers required to maintain reasonable midrange performance in a full range speaker don't do a very good job of reproducing the lowest bass frequencies but they do put a lot of energy into the speaker cabinet structure and this is very detrimental to sound quality.

As the woofer cone makes these large mechanical movements to pressurize and rarefy air, an equal and opposite force is applied to the woofer basket, or frame, which is attached to the speaker structure. This force excites resonances in the cabinet structure and tries to move the whole speaker back and forth. Cabinet resonances color the sound in the midrange. Cabinet movement distorts high frequencies.

A backward and forward motion of just a few thousandths of an inch may represent a major percentage of the total excursion of the tweeter diaphragm as it attempts to reproduce subtle high frequency details. The result of structural movement is IM distortion of the midrange and high frequencies.

If you are skeptical about the sonic consequences of woofer energy moving the speaker cabinet, think about speaker spikes. A reduction in cabinet motion is the main reason that spikes beneath the speaker improve sound. Remove the spikes and see (no, hear) what happens.

As you can see, a full range loudspeaker is a bundle of compromises. It is asked to perform many conflicting tasks. There is a proverb that goes something like this: "a man who chases two rabbits has no meat for dinner." By the same token, a speaker that tries to provide both bass and the rest of the spectrum compromises the quality of both.

A single-pole, passive high-pass filter at the input to the amplifier can cure or minimize all these speaker problems and improve performance dramatically. This sonically transparent filter will reduce woofer cone excursion which will reduce distortion in the bass, midrange and treble as described above. The result will be better definition, better imaging, tighter control, greater dynamic range and a better presentation of the rhythm and pace of music. The only thing missing—besides distortion—will be low bass and that will be reproduced by specialized devices designed just for that purpose—powered subwoofers.

2. Better performance from your amplifier.

The major energy demands in music occur at low frequencies. The major current demands from an amplifier are at low frequencies. When an amplifier distorts because of demands for power that it cannot meet, the output waveform is flattened at the top and bottom. This distortion is called clipping because the positive and negative signal peaks have been "clipped" off.

Amplifier clipping becomes evident at high frequencies but clipping is almost always caused by energy demands at low frequencies that exceed the capability of the amplifier.

Clipping is the primary cause of speaker damage because a clipped waveform "fools" the crossover network in the speaker which then passes high power to the high frequency drivers.

An amplifier in normal use will be clipping at least occasionally. The percentage of time that the amplifier is driven to the point of clipping or beyond will have a profound effect on sound quality. As the amplifier approaches clipping the sound will become slightly hard, then harsh, and then, as the amplifier clips, a shattering distortion will be heard. This distortion eventually destroys tweeters and crossover networks.

A single-pole, passive high-pass filter at the input of the amplifier can eliminate all these distorted sounds and make the amplifier sound smoother and more relaxed. The amplifier may seem to be three times more powerful. Removing the huge low frequency current demands from the amplifier, by reducing the level of the input signal at low frequencies, allows the amp to coast along with lots of power in reserve. The system will play at much higher levels with much lower distortion, providing a greatly improved listening experience.

The high current necessary for accurate bass reproduction will be provided by specialized amplifiers designed just for this purpose—the amplifiers in the powered subwoofers.

3. Better bass.

Designing a product to perform a very specific task requires less compromise. Subwoofers are designed to reproduce a small range of frequencies at the lowest audible range. That's about as specific as it gets in audio.

When compared to full range speakers, powered subwoofers can provide the following advantages: more cone area, greater linear excursion capability, more amplifier power at low frequencies, and electronic compensation for falling output at the lowest frequencies. Subwoofers can also have smaller, stiffer, less resonant enclosures and can be placed in the optimum position to introduce bass energy into the room.

Eliminating the compromised bass output from the main speakers by high-pass filtering the input signal to the amplifier will dramatically improve the quality of reproduction in the mid-bass range. Improving the mid-bass provides a better sense of rhythm and pace and makes it easier to follow the tune of the bass.

4. Better room interface.

The pressure-zone microphone (PZM) was developed after it was determined that smooth frequency response at lower frequencies could not be obtained from a stand-mounted microphone due to interactions with the room boundaries. Placing a conventional microphone on the floor smoothed the response curve but caused a gradually rising bass output. Compensating for this bass rise gave us the PZM microphone. A similar effect occurs with speakers.

For good imaging and midrange detail full range speakers must be placed well out into the room. Bass response from these speakers will be uneven due to room interaction. This phenomenon is frequently attributed to "standing waves."

Removing bass from these speakers and redirecting it to a subwoofer placed in the corner of the room will ameliorate most of these room anomalies. The subwoofer will load the room from a pressure zone, smoothing response across the bass range. Adding a second subwoofer, placed in a second corner, allows low frequencies to be introduced from two different positions within the pressure zones of the room virtually eliminating bass irregularities. (You must remove other sources that store and release energy at low frequencies as described in the room treatment article.)

5. Reduced system cost.

In a given manufacturer's amplifier line, the more expensive models usually offer more power and little else. In fact, smaller amplifiers frequently sound better than their big brothers and they always cost less.

The Levinson 33H mono amps that I use cost about $15,000 less than the Reference 33 amplifiers from the same company. Both models are essentially the same design, with the larger version offering only higher output power.

Most loudspeaker manufacturers offer a range of models that differ only in their ability to produce bass. Bigger, more expensive models provide extended bass response with bigger woofers and larger cabinets. Except for bass extension, it's not unusual to find that the smaller models in a given line of speakers actually sound better because they have smaller woofers that offer better midrange performance and the smaller cabinets add less box sound. Compare the smaller Dunlavy models to their larger brothers for example.

The Vandersteen 3A Signature speakers that I use in conjunction with a pair of 2Wq subwoofers deliver 90% of the performance of the Vandersteen Model 5s for 60% of the price (3A Sigs and two 2Wq subwoofers cost about $6,000 and Model 5s sell for about $10,000). My speaker system delivers a time- and phase-accurate response over a usable range of 18Hz to 30kHz. What other speakers can offer that for $6,000?

Some reviewers claim that the 3A Signatures lack the "detail and definition" of the Model 5s. You may find this puzzling because both models share identical midrange and tweeter drivers and use essentially the same crossover network in this range. Why the perceived performance difference? Model 5s have a slightly more inert cabinet structure and they have *built-in, powered subwoofers*.

The use of powered subwoofers can allow a smaller amplifier and a pair of lower-priced speakers to equal or outperform their more expensive counterparts. The result is better sound for less money. Who doesn't want that?

Why Most Subwoofers Don't Work Well for Music Reproduction

Not so many years ago, few people were aware of the concept of specialized bass speakers. Explaining what a subwoofer was and the sonic benefits it could provide were difficult tasks before the home theater craze hit the public. Today, people are rushing to add subwoofers to their audio systems to provide the visceral excitement that only thunderous bass can supply.

Thunderous bass output makes an on-screen explosion or gunshot more physically involving but it can also alter the tonal balance, as well as the rhythm and pace, of music. Most subwoofers seem to march to the beat of a different drummer instead of the one who is playing with the orchestra.

Today the average consumer believes that the only purpose of a subwoofer is to add bass and many music purists derisively refer to them as "fart boxes."

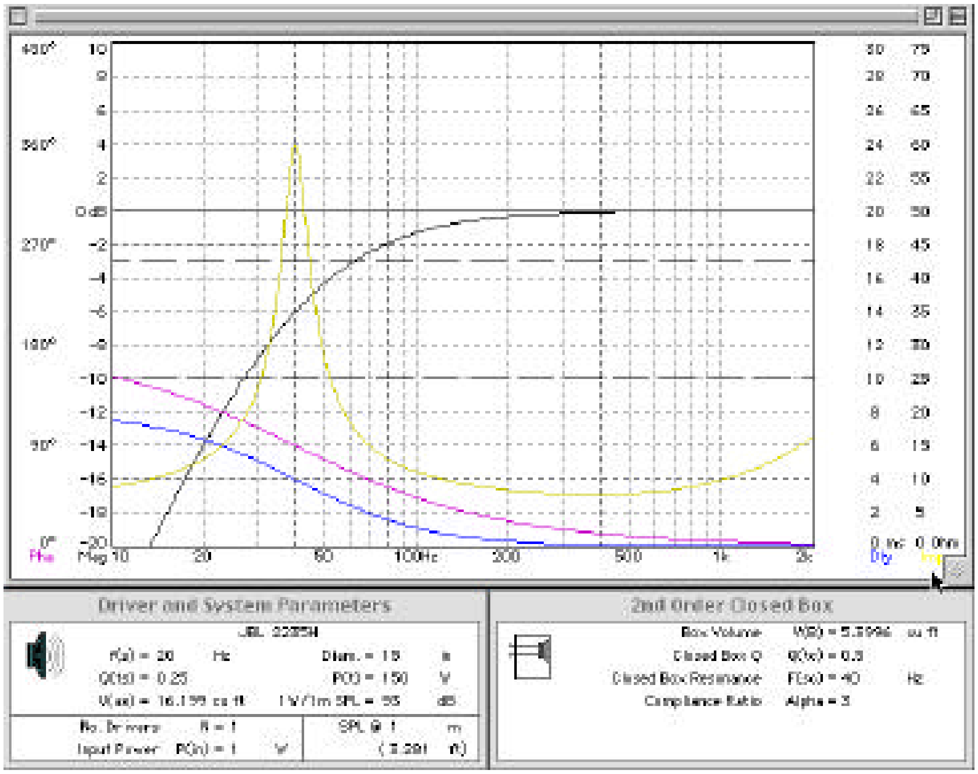
There are many reasons why boom-box subwoofers may do a good job of reproducing explosions and perform poorly when reproducing music.

1. Phase shift and group delay cause subwoofers to start late.

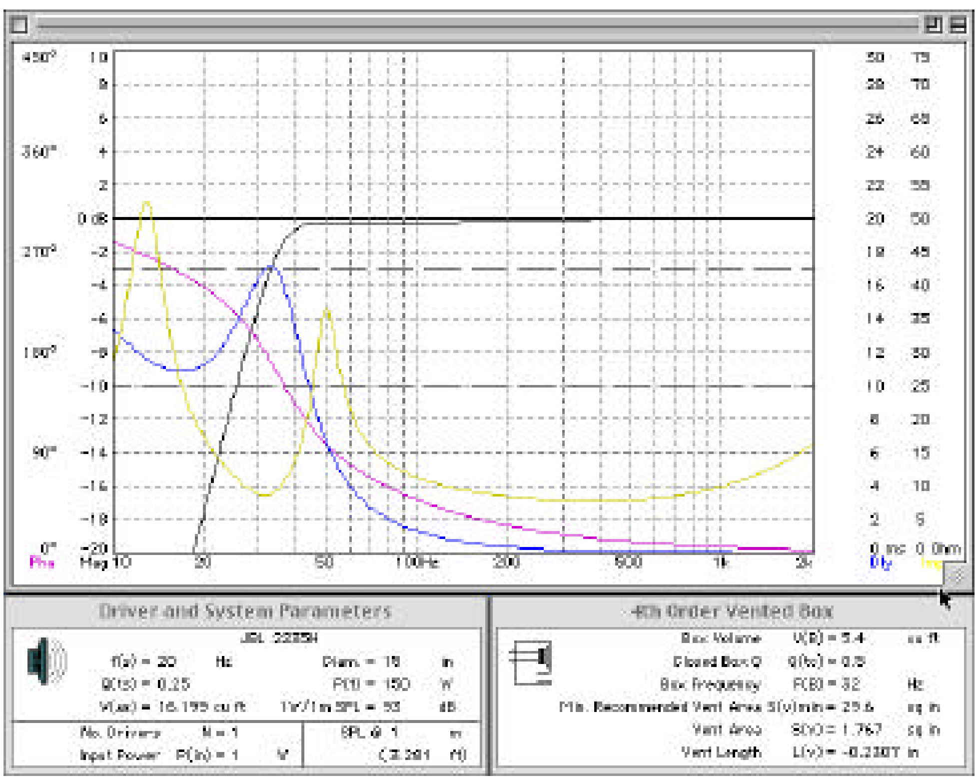
Picture a marching band with the bass drum following about a block behind the rest of the players and you've got a pretty good image of the major problem with most subwoofers—the sound they produce is just out of step with the rest of the music. There are many reasons for this but most revolve around phase shift and group delay.

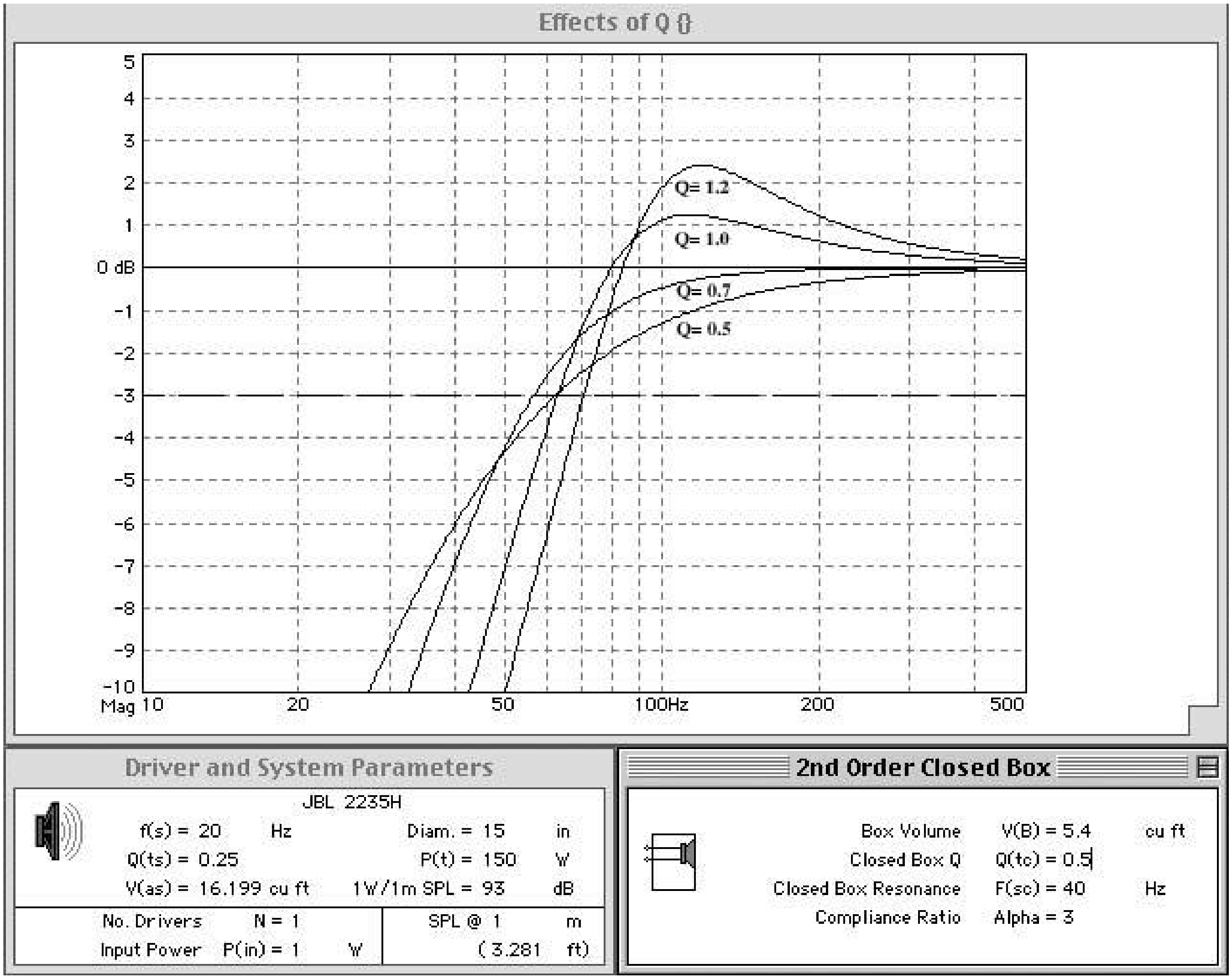
Acoustic phase in this instance has to do with the time relationships of the launch of air pressure waves towards the listener. Phase shift that varies with frequency alters the time relationships between different frequencies. A resonance in the pass band causes phase shift. Filters cause phase shift. A speaker in a box is a filter.

When two elements like a subwoofer and a main speaker have an overlapping frequency range, or are reproducing different parts of a single note, such as the fundamental and the harmonics of that fundamental, you want both to be in step. If the



These illustrations compare frequency response (black trace), phase response (purple trace), impedance (yellow trace), and group delay (blue trace) for the same JBL 2235H 15-inch driver mounted in a second-order sealed enclosure (above) and in a fourth-order vented enclosure (below). You can learn a lot about the performance trade-offs involved in the choice of vented versus sealed enclosure design from studying these graphs. In my opinion, the trade-offs make vents acceptable for use in full range speakers and totally unacceptable for subwoofers that are meant to reproduce music. A conventional vented design offers inferior performance in every area except one: output level capability.





This illustration shows the effect of system Q on frequency response. Notice that the low-Q alignments start sloping down sooner but ultimately extend lower in frequency at the usable low frequency limit of -10dB. Remember that the low-Q alignments provide far better damping to oscillation after the signal ceases.

subwoofer cone pushes out when the main speaker's woofer cone is moving inward, things get out of sync.

Group delay is a complex concept. It is the negative of the derivative of the phase curve with respect to radian frequency. Group delay describes how well the time relationships between a small group of frequencies are preserved within a narrow range of frequencies. Time delay and group delay are not necessarily equivalent but a delay to one group of frequencies changes its time relationship to the rest of the spectrum.

Look at the illustrations on page 15 comparing phase response of vented and sealed enclosure designs. Trying to synchronize the phase of the main speakers and the subwoofer will be difficult with a sealed enclosure subwoofer design and virtually impossible with a vented design because of phase shift as the system passes through resonance in the pass band. Note the 17ms group delay at the 32Hz tuning frequency of the vented design.

Subwoofers with a fundamental resonance in the pass band and a steep slope low-pass filter at the input will produce output that is delayed in time relative to the main speakers, and this delay will vary with frequency.

Subwoofers that start late sound slow and plodding. They distort the overall waveform even if their own distortion products are low.

2. High-Q makes subwoofers stop late.

An electrical filter will oscillate or ring, to some extent, after the signal stops. The steeper the slope of this filter, the more it will ring. The higher the “Q” of this filter, the more it will ring.

Mechanical filters work the same way. In fact, all the mechanical properties of a loudspeaker can be expressed with electrical equivalents and modeled by electrical circuits.

A woofer in an enclosure is a high-pass filter. It passes frequencies above the cut-off or low frequency limit of the design and the signal rolls-off below this point at a rate determined by the design. A sealed box acts as a nominal second-order high-pass filter and a vented enclosure will typically display fourth-order high-pass characteristics. The vented design will ring (oscillate) about twice as much as the sealed design after the signal stops.

The “Q” of the mechanical system affects oscillation, too. System “Q” 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 of .5 is considered a “critically damped” alignment with a step response that has no overshoot. For a given driver, a Q of .5 requires the largest box. This low-Q alignment has a downward-sloping response curve but offers the best possible transient performance and the lowest frequency extension at -10dB.

A system Q of .577 is a Bessel alignment which has the most linear phase response and offers slightly less damping.

When Q =.707 we have a Butterworth alignment with the flattest amplitude response. This is the most common alignment for “high-end” subwoofers because it offers a “full” sound which is still well controlled.

System Q near 1.0 delivers a peaked response but allows the smallest box size still considered by some to be high fidelity. A subwoofer with a system Q over 1.0 is a boom box with a peaked response curve and lots of overhang. Guess where most home theater subwoofers fall.

Subwoofers that play on after the signal has stopped (due to oscillation) sound slow and muddy.

The Q and slope of the high-pass filter formed by the subwoofer acoustical system have a major effect on the sound of the bass the subwoofer produces, but there’s more. The subwoofer is a mechanical high-pass filter but it must be used with an electrical low-pass filter and those cause problems as well.

3. Steep filter slopes and direct-radiating drivers.

Most subwoofer designs include one or more drivers that radiate directly into the room. It’s commonly assumed that subwoofers shouldn’t be allowed to encroach on the midrange where they perform poorly so the conventional wisdom mandates a steep-slope low-pass filter to prevent output above the bass region. There are some drawbacks to this approach.

A low-pass filter with a fourth-order slope will cause another complete phase rollover, increasing signal delay. Even with this steep attenuation curve, side band distortion will still be audible if the driver radiates directly into the room.

All drive elements will break-up (display non-pistonic cone behavior) at some frequency. Filtering below this frequency prevents the signal from stimulating this behavior. Many other anomalies—cone resonances, surround reflections and “flapping,” magnetic nonlinearities and basket ringing—will remain audible as side-band distortion even without frequency-specific stimulation. And some midrange frequencies will still pass through the filter at attenuated levels.

Midrange signals (even at low levels) and side band distortion detract from the quality of sound from the main speakers and draw attention to the position of the subwoofer which should be spaced away from the main speakers.

4. System resonance in the pass band.

Removing resonances from audio components is generally considered to be a good idea but subwoofers are designed to create resonance.

A vented subwoofer has two resonances right in the middle of its pass band. The vent resonance is tuned to play at frequencies where the output of an unassisted driver would be falling. Much of what you hear from a vented design is a production of the subwoofer rather than a reproduction of the signal.

Sealed enclosures are better with only a single resonance in the pass band. One is better than two, as you can see from the graphs on page 15, but none is better yet as we'll see later.

When the subwoofer passes through a resonance a big shift in phase occurs. Look at the graphs of phase response for sealed and vented enclosure designs and see the effects of resonance on phase. (System resonance occurs where the impedance peaks. Note the single peak in the graph of the sealed enclosure and the dual peaks in the graph for the vented enclosure. The dip between the impedance peaks indicates the tuning frequency of the vented enclosure.)

Subwoofers with a resonance in the pass band will tend to emphasize the frequencies around this resonance. The higher the Q the greater the emphasis. Ever hear the term "one-note bass"?

5. Dissimilar amplifiers for high and low frequencies.
A powered subwoofer may have an internal amplifier that is designed for optimal performance when driving the specific load of the subwoofer drivers. This amplifier will almost always have electrical compensation for the falling response of the subwoofer driver(s) which will typically be housed in an enclosure that is smaller than ideal—because nobody wants a subwoofer the size of a refrigerator in their living room.

A perfect subwoofer amplifier and the amplifier that is best suited for driving the main speakers may be very different electrically and sonically. For instance, a bipolar solid-state amplifier will offer the best performance for bass but a MOSFET or tube amplifier may sound better driving the main speakers.

Transfer function is a measurement that compares the frequency and phase response of the output from a device under test to the input signal. If the transfer function of the main amplifier is very different from the TF of the subwoofer amplifier, this sonic dichotomy may have a negative impact on overall sound quality.

Making a Subwoofer to Play Music

A subwoofer should march in step with the rest of the band and stop playing when the song has ended. Most do neither.

Subwoofers should be positioned in the room corners to properly load the room from pressure zones, creating the smoothest bass response. If the subwoofer has a "Q" higher than .5 (most do) it will exhibit a rising response when placed in a corner.

So how can a subwoofer be designed that doesn't suffer from the performance deficits described above?

We can eliminate the problem of subwoofers that start late by blending the subwoofer and the main speakers using a first-order crossover for transient-perfect phase response. This filter system can be implemented in an unusual way: the high-pass section can be placed at the input to the amplifier driving the main speakers and the subwoofer amplifier can sample the output from this amplifier, including its sonic signature and phase characteristics. The subwoofer amplifier can have its frequency response tailored to compensate for the falling response of the input signal and the falling response of the acoustic system that is operating primarily below system resonance. (More about that in a minute.)

Blending the amplifiers in this way will ameliorate the discontinuous sound created by dissimilar amplifier designs driving different parts of the spectrum.

We can eliminate the overhang of subwoofers that stop late by designing for a target Q of .5 to achieve critical damping, along with the greatest usable bass extension, and to allow corner placement without a rising low-end response.

The driver(s) can be slot loaded to mechanically filter out sideband distortion and midrange frequencies. And the system can be designed to operate primarily below the fundamental resonance so that no resonance can cause sonic emphasis or phase shift in the pass band.

This all makes perfect sense and I'd like to take credit for thinking it up all by myself. But I didn't—Richard Vandersteen did. What I have just described is the Vandersteen 2Wq powered subwoofer which has been subtly but continuously refined since it was first brought to market sixteen years ago. It is the most sophisticated product of its kind available today. APJ

the VANDERSTEEN ***2Wq*** SUBWOOFER

The Vandersteen 2Wq subwoofer is completely unique in a number of ways. It takes advantage of the fact that loudspeakers in sealed enclosures offer very predictable amplitude and phase response characteristics at frequencies below the fundamental resonance of the system. The 2Wq operates primarily below fundamental system resonance to provide frequency and phase linearity that cannot be achieved by conventional designs with resonances in the pass band.

The 2Wq uses a phase-perfect first-order crossover with special characteristics. It samples the output from the amplifier that is driving the main speakers for better system integration.

It uses feed-forward error correction to prevent output errors before they occur and a unique protection circuit that does not compress signal dynamic range. The 2Wq will not produce audible distortion regardless of the frequency or level of the input signal.

It utilizes three small drivers instead of one larger unit for greater power-to-weight ratio and better diaphragm control. The 8-inch drivers in the 2Wq are slot-loaded to linearize pressure on the front and rear of the cones and to mechanically filter side-band distortion.

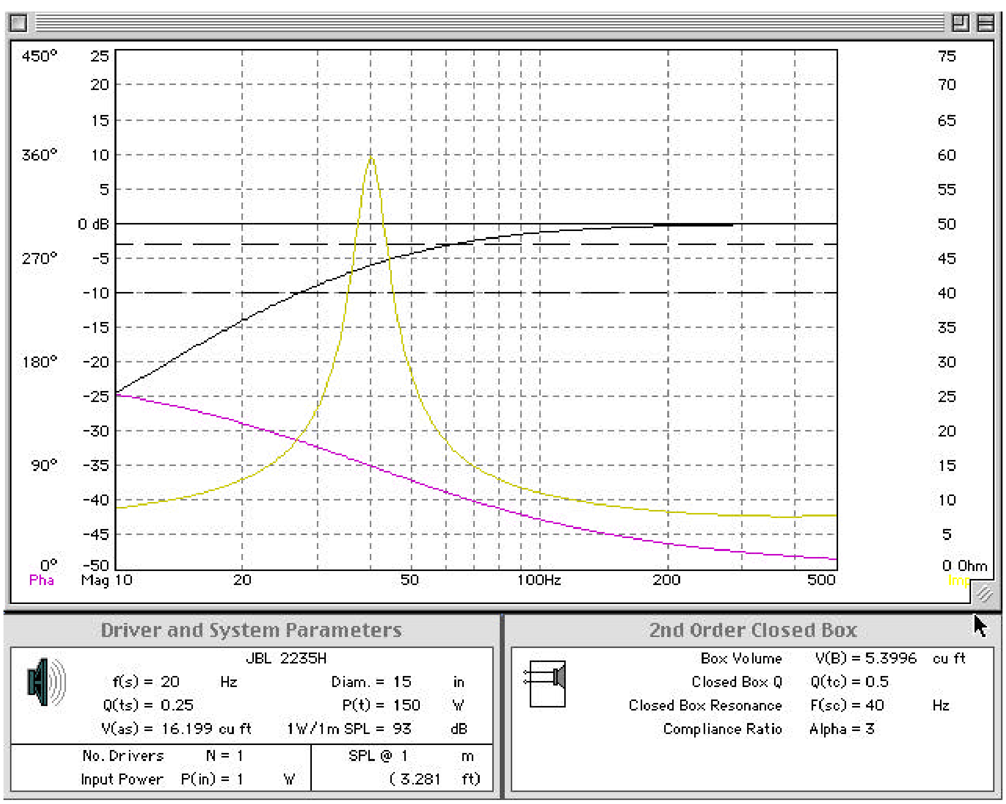
The cabinet is constructed using constrained-layer lamination techniques and cross-bracing, resulting in the most inert, resonance-free subwoofer enclosure that I've ever seen.

It has adjustable Q. You can adjust the output response curve of the 2Wq to suit room acoustics and placement.

No other commercial subwoofer offers all these features and virtually all high-quality competitors cost more.

The 2Wq Operates Primarily Below System Resonance

Conventional subwoofers operate above system resonance. They exhibit uneven response and lots of phase shift as they pass through the fundamental system resonance, which typical-



This illustration shows the amplitude and phase characteristics of a sealed enclosure subwoofer with a Qtc of 0.5, at frequencies below system resonance which, in this example, is 40Hz.

ly occurs at the lower end of the pass band. Designers try to get the resonance frequency low because output falls off steeply below this frequency. A vented design with a B4 alignment falls at 24dB per octave and a sealed design with a Q of .7 falls at 12dB per octave below resonance. Take a look at the illustrations in the previous article to see what happens to phase.

A driver mounted in a sealed enclosure with a Qtc (total system Q) of .5 will have an amplitude response curve that falls in a linear fashion at about 8dB per octave below system resonance with limited and linear phase shift. Output will theoretically extend down to DC without the sudden drop-off and phase shift that occurs when conventional systems pass through resonance. This predictable and linear frequency and phase response is easily compensated for with electronic correction in the amplifier to produce ruler-flat output to subsonic frequencies.

Since output is more linear below system resonance and flat response can be achieved with amplifier compensation, why not design a subwoofer that operates below resonance rather than above it? That's just what Vandersteen has done.

The result is a subwoofer with virtually no resonance in the pass band, minimum group delay, linear phase response, flat amplitude response to subsonic frequencies, critical damping and a low system Q making it suitable for corner placement.

A Better Blend with the Main Speakers

The Vandersteen 2Wq subwoofer is integrated with the main speakers using a unique system that is not a crossover in the usual sense. Transitions between the subwoofer and the main speakers are made with gentle 6dB per octave slopes using phase- and transient-perfect first-order filters that are completely transparent.

A passive, first-order filter is inserted at the input to the main amplifier. This filter causes the signal to the main amplifier and speakers to roll off at 6dB per octave below 80Hz.

A 300-watt subwoofer amplifier, designed specifically to deliver high current into the low impedance load of the three drive elements, samples the signal at the output of the main amplifier and compensates for the roll-off to produce flat output from the subwoofer. The output from the subwoofer amplifier is tailored to produce first-order low-pass response above 80Hz and a rising response below 80Hz to compensate for the falling response curve of the filtered input signal as well as the falling output response of the subwoofer which is operating primarily below system resonance.

Output that could exceed the power limits of the amplifier, or the linear excursion limits of the drivers, is prevented by dynamically raising the low frequency cut-off point rather than compressing the signal. A unique circuit analyzes the input signal and dynamically alters the feed-forward error correction characteristics to accomplish this feat. The 2Wq cannot be driven to produce audible distortion under any conditions, yet it never compresses the dynamic range of the signal, maintaining the natural rhythm and pace of music regardless of level.

A passive, first-order high-pass filter at the main amplifier input is completely transparent so the sound from the main speakers is not negatively impacted in any way and all the positive benefits listed in the *Subwoofers From a High-End Perspective* article can be realized.

Sampling the output from the main amplifier passes along the sonic and electrical characteristics (particularly propagation delay) of that amplifier to the subwoofer system for a better blend between the subwoofer and the main speakers, even if the main amplifier is a tube or MOSFET design. This results in superior integration between the subwoofer(s) and the main speakers.

Less Audible Distortion

In addition to the feed-forward error correction system and the specially designed internal amplifier, the 2Wq uses several other distortion reducing techniques.

Three 8-inch drivers have the combined cone area of a single 14-inch unit but three motors can provide far better control over the lighter, stiffer cones. Smaller cones produce less side-band distortion than larger, more flexible cones, and any distortion that remains will be at higher frequencies which can be mechanically filtered by the indirect radiation path.

These three drivers are slot-loaded providing an indirect radiation path into the room. Slot-loading the front of the drive elements equalizes the pressure on the front and back of each diaphragm making resistance to fore and aft movement more linear.

A driver in a sealed enclosure "sees" a diminishing volume of air and increasing pressure within the box as the cone moves inward, and an increasing volume of air and reduced pressure as the cone moves outward. Covering the front of the driver(s) with a plate so that radiation from the front of the drive elements enters the room through a slot or slots between this plate and the enclosure is an attempt to compensate for this phenomenon.

Slot-loading provides a reduction in distortion by linearizing cone motion and also acts as a mechanical low-pass filter to absorb residual distortion products at higher frequencies. This mechanical low-pass filter is far more effective than a steep-slope electrical filter for the reasons described earlier.

The cabinet is elaborately constructed using constrained-layer laminates and cross bracing to completely eliminate panel resonances and spurious noise. The 2Wq enclosure feels like a solid block of material. Rapping on any panel is like banging your knuckles against a rock. Panel flexing is simply out of the question.

Caveats

The Vandersteen 2Wq subwoofer provides tightly controlled bass that is "critically damped" and limited in output level compared to a typical home theater subwoofer. Two units will be required in all but the smallest rooms to provide the THX-recommended output level of 105dB at 35Hz. I recommend using two subwoofers anyway and 105dB is much too loud for music listening so I don't see these as problem areas.

Tightly controlled bass that is perfect for music may not satisfy explosion fans who use their audio systems for both music and home theater. Vandersteen makes another subwoofer, the V2W, for these folks. It looks the same but trades some control and integration for the ability to play much louder.

Other subwoofers that offer excellent performance for those with a strong home theater bias include many of the M&K models and the Bag End InfraSub. These subwoofers will still perform well on music and deliver more visceral output. Don't use their high-pass filters. Choose a passive single-pole filter instead.

Best Value

In my opinion, the Vandersteen 2Wq is the best subwoofer available for reproducing music regardless of price, and the price is a mere $1,250 each! If that's not a bargain, I'm a bad shopper. I have four 2Wq subwoofers and they're not on "long-term loan;" I bought them. I want the best possible performance and I'm willing to pay for it. If the product that offers the best sound quality also costs less, I won't complain. APJ

how to get. . .

NEAR-PERFECT BASS

I'm sure you've heard about computer software that will analyze "standing waves" based on your room dimensions and determine where to put your speakers. Placing your speakers in these positions is supposed to minimize the excitation of room modes and produce the smoothest overall response.

Then there are those magic boxes that supposedly analyze and correct aberrations in response caused by room interaction. These devices allow you to put your speakers anywhere and then fix response by equalizing the input signal (see Issue #1).

There is a kernel of truth in each approach but neither one offers a complete solution. You do need some control in order to adjust response at bass frequencies but moving the speakers is not enough and electronic compensation has too many negative side effects, not the least of which is cost.

Homogenized Bass

At very low frequencies the sound from the speakers and the sound contributed by the room cannot be separated by the ear/brain mechanism. You perceive the combined signal as a single entity and total amplitude becomes the overriding factor. You need to be able to adjust amplitude with frequency to smooth overall response and you can't do that with a passive system and full range speakers unless you employ an equalizer. Accepting the negative effects of placing an equalizer in the signal path just to smooth bass response is unwise and equalizers cost money. The best solution is to employ powered subwoofers and divide the spectrum between specialized reproduction devices.

Powered subwoofers provide the benefits we've already addressed and they facilitate fine tuning of the low frequency performance of your system in your room.

Facilitating Adjustment

When you add powered subwoofers you create a biamplified system. With these subwoofers placed in the front corners of the room, bass is introduced from two locations within pressure zones. This will provide remarkably smooth bass response that will require little correction. A biamplified system with a crossover point at about 80Hz provides all the tools you need to fine-tune this response in the room.

Nearly all the audible irregularities in bass response that you will encounter in the home occur between 60 and 120Hz. A biamplified system and a nominal crossover frequency of 80Hz makes adjustment of amplitude anomalies within this range a piece-of-cake because of the overlapping output of the subwoofers and the main speakers.

Near-perfect bass can be achieved by removing any sources that store and release energy at low frequencies, as described in the article on room treatment, and by tuning bass amplitude linearity by adjusting the high-pass and low-pass frequencies and slopes that blend the main speaker and the subwoofers. Here's how to do that.

Beginning Adjustments

Get a suitable test CD with sine wave signals or warble tones covering the range of 20Hz to 200Hz or so. A signal generator is better but I don't recommend pink noise as a stimulus.

Using a sound level meter, match the output level at the listening position at 50Hz and 150Hz by adjusting the volume control on the subwoofer(s). Make sure the volume control on your preamp remains at the exact same setting. Adjust the volume to produce about 85dB at the listening position so you won't damage anything.

Using a signal at the nominal crossover frequency, set the phase of the subwoofers to deliver the highest output at the listening position. Readjust the balance between the subwoofer(s) and the main speakers.

Smoothing Response

With a fixed input signal level, carefully measure the output level at the listening position for each 5Hz interval between 30 and 150Hz and write it down or make a line plot on a sheet of graph paper. A Radio Shack sound level meter works fine for this purpose.

With a crossover point of 80Hz you'll have output from both the main speakers and the subwoofer(s) over the range of 60 to 100Hz or so. Crossovers with steeper slopes (18-24dB/octave) will have less overlapping output and those with slower slopes (6-12dB/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, not the meter. Remember, how it sounds is more important than how it graphs.

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

If total output in this frequency range (60Hz to 100Hz) is low, 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 with the main speakers.

If you need more output at the lower portion of this overlap region, adjust the range where the overlap occurs downward without increasing the amount of overlap. To do that, lower the frequencies 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? APJ

Note: The Vandersteen subwoofers will self-adjust the low-pass response to some degree when you change the high-pass frequency (because their input is a sample of the filtered output from the main amplifier and you are changing that by varying the high-pass frequency). You can vary the high-pass frequency and tailor the overall response using the Q control on the Vandersteen subwoofers to accomplish the same results as described above.

A suck-out in the 60Hz-80Hz 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 speakers and the subwoofers due to phase problems 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) in order 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 40Hz to 120Hz in most any room, along with a good blend between the main speakers and the subwoofers.

Response below 40Hz is largely determined by the output capability of the subwoofers and the ability of the room to contain pressure. Stiffening the walls can extend bass response and bass traps can remove some low frequency energy if necessary. See the following article for more information. APJ

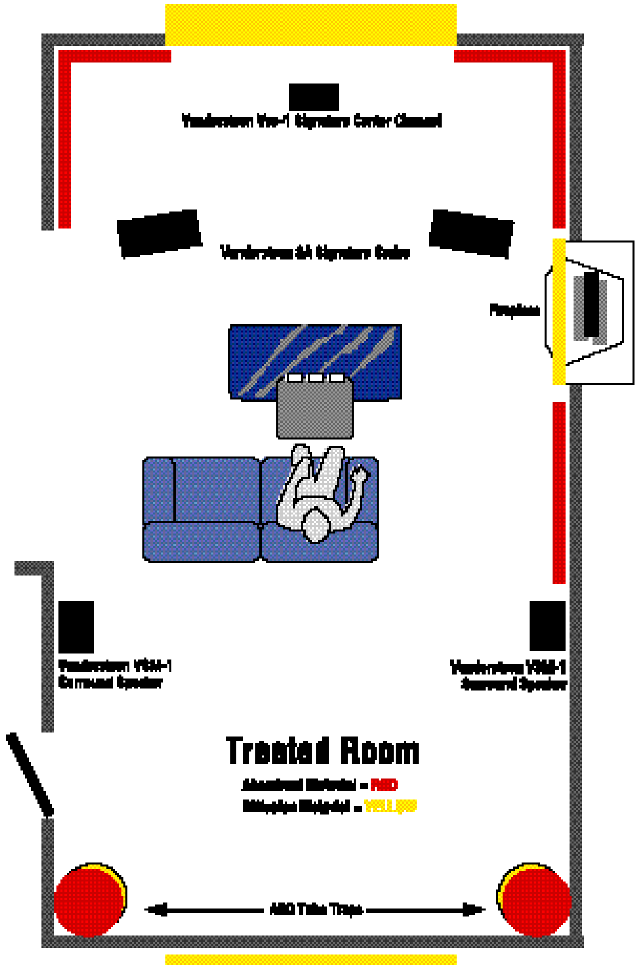
Good ACOUSTICS in REAL LIVING ROOMS

The goal of acoustic room treatment is to make the direct sound from the speakers more prominent and the reflected sound, or reradiated energy from the room, less prominent—without making the room an uncomfortable place to be.

Reradiated energy comes from room surfaces or objects in the room that reflect the signal originating from the speakers or from other surfaces, and from wall panels or objects that store and release low frequency sound by mechanical flexing. Reverberation is energy bouncing back and forth between reflective surfaces which produce coherent replicas of the original sound.

A Room Can Be Too Dead

You could make the direct sound from the speakers most prominent by eliminating all sounds contributed by the room.



Reflections coming from surfaces near the speakers can confuse the ear and I like to place absorptive material beside and behind the speakers. In this room there is a bay window between the speakers and I covered that with wide-blade, plantation shutters which act as sound diffusers for stereo listening, and block most of the light for home theater viewing. Shutters also cover the window at the rear of the room.

A floor-to-ceiling veneer of rounded stones was installed in the area above and around the fireplace opening and this acts as a sound diffuser. Tube traps in the rear corners absorb low frequencies and diffuse higher frequencies. Equipment racks and other furniture (some not shown in this illustration) prevent reflected sound waves from bouncing back and forth between hard room surfaces. The end result is great sound and a pleasant environment in which to read or simply relax.

Making the room an anechoic chamber (literally a chamber without echo) is not a good idea for many reasons, the most important of which is that an anechoic chamber is not a pleasant place to be and you want your listening room to be comfortable and inviting.

Removing too much reverberance from the room by over-damping the wall surfaces causes other negative effects as well. These include increased distortion due to the higher input power required to produce the same perceived volume level, and tonal balance shifts due to nonlinear absorption.

As more energy is absorbed from the room, more power must be applied to the speakers to provide the same perceived volume. More power means more distortion and strain from the amplifier and speakers.

Absorptive material can soak-up energy only at frequencies above 150Hz or so, and has little effect on low frequencies. Excessive application of absorptive material will make the room sound dead and bass-heavy, even if bass output levels are reduced. Diffusion can reduce reverberation time with little impact on tonal balance.

Here is my advice in a nutshell. Do as little as possible to the natural room acoustics while attempting to eliminate coherent reflections from the room boundaries. Make sonic reflections incoherent by using absorption and diffusion. Balance absorption and diffusion to minimize reverberant decay time while maintaining a bright, lively acoustic characteristic in the room.

The Party Test

In Issue #1 I told a story about a trip abroad and some advice I received there about room acoustics. I'll repeat it here.

While visiting Scotland I asked my friend, Ivor Tiefenbrun, Managing Director of Linn Products, for his opinions about room acoustics. His advice was simple. "Throw a party and watch where people gather to converse," he said. "Put your hi-fi in that room." He spoke with a Scottish accent, of course, but I don't know how to write that way.

We think of ourselves as visual beings because we are so conscious of the things we see, but our hearing is far more sensitive and primal than our sight. The sounds we hear influence us on a more emotional or subliminal level. We may be less consciously aware of the sound around us but its effects are powerful nonetheless.

We tend to add furnishings to a room until it feels comfortable to be in there. Much of that feeling of comfort has to do with how the room sounds. A room that is too bare sounds cold and barren like a hallway at the courthouse or the restroom at the airport. Too many furnishings make you feel closed-in and claustrophobic. An anechoic chamber is not a comfortable place to be and neither is an over-damped custom home entertainment room. You want to hear the music, not the blood circulating in your head.

Most domestic rooms that provide a comfortable place for conversation will also provide acceptable acoustics for a piano recital or for hi-fi listening. Some additional work may be necessary to achieve the very best possible sound but don't overdo it.

You should be careful not to absorb too much energy from the room with acoustic treatments. Try to diffuse some reflections rather than absorb them.

Choosing a Room

If you build a special room for your audio system, you can choose dimensions that spread resonances and minimize the frequency response aberrations caused by them. That's another topic for another place. Here, we are going to discuss a subject that is far more practical to most of us—choosing the best place for your audio system from the rooms available in your existing house.

First, some good news. Huge rooms and high ceilings are not necessarily an advantage, and large openings into other areas are. That means that regular rooms in normal houses can provide a good environment for music listening.

The small, narrow room that I have been describing in this **Journal** is perfect as a test laboratory for my equipment reviews and a fun listening and viewing room for just my wife and me. If you entertain, you'll need a larger space, but bigger rooms require more power and offer more surface area to reflect sound.

Sound bounces off the walls just like a cue ball bounces off the side cushions of a pool table. The angle of reflection will duplicate the angle of incidence. More walls provide more angles but not necessarily better sound.

Cathedral ceilings can double the cubic area of a room and all that additional space will be devoid of sound-absorbing furniture and drapes. Large areas of plaster and glass will extend above the normal living area and sound will reverberate up there like bats bouncing around your belfry. You can make good sound in a room with vaulted ceilings but it will take more work. Contrary to popular belief, a high, flat ceiling is usually better than a sloping one.

A room that is sealed-off from the rest of the house will have some strong, resonant modes just like a speaker enclosure. A room with large openings into other areas, like what you'll find in most modern houses, will support fewer strong resonances and they'll probably be at lower frequencies where they may actually be beneficial. The room I've been describing in the previous articles has a nine-foot gap in the left side wall leading to a tiled entry area and other rooms on the ground floor.

Reverberation

If you place two large mirrors on opposing walls and stand between them, you'll see an infinite series of reflections that gradually diminish in size. Sound reflections between two hard surfaces work much the same way. This is reverberation.

Reverberation is the persistence of sound in an enclosure after the original sound has ceased. A little bit tells your brain that you are indoors. Too much will blur the signal from the speakers with an echo of a signal that preceded the one you want to hear now.

An RT60 measurement quantifies reverberation. It is a measurement of the time required for a transient sound to die away to one-millionth of its original intensity or decay by 60dB. A good-sounding room will have a bright, lively characteristic with a short reverberant decay time in the midrange.

You can measure RT60 with test instruments and consult tables constructed by who-knows-who to adjust for someone's idea of the ideal decay time for the volume of the room, or you can clap your hands together and listen for an echo. If you hear a ricochet sound when you clap your hands together, fix it with wall treatment. Keep treating until you hear just a single clap with no repeating echoes.

The spectral plot of my hands clapping together peaks over the range of 1kHz to 2kHz. This is a perfect test signal to gauge reverberation.

The problem of excessive reverberant decay time can almost always be cured by absorbing or diffusing reflections from the walls or other hard surfaces. The general rule is: don't allow two reflective surfaces to oppose each other. If you have a hard ceiling, carpet the floor. If you have a large area of glass, cover it with drapery material or damp the wall opposite the glass.

You can shorten reverberation time by either damping or diffusing reflections. Damping absorbs some of the energy and diffusion disperses the energy by spreading it out over more area.

The cue ball in the earlier example would not bounce off the cushion if you placed a pillow in its path, and sound will be damped in a like manner by absorptive material placed on the room surfaces or between the source and the surfaces. You can't diffuse a cue ball, but if you could it would shatter into many pieces as it bounced back from the cushion, and the pieces would scatter in all directions and each one would have less energy than the whole.

Look again at the treated room (page 23) and see where the surface modifications were applied (shown in red). Following are some options for absorbing or diffusing problem reflections.

Before my room was treated, there was a mid-frequency ring that made a hand clap sound like many ricocheting bullets, and a bass boom centered around 70Hz. I applied absorptive and

diffusive material to most wall surfaces and stiffened the walls around the speakers to correct these problems. The only commercial (read expensive) products that I used were four 12-inch diameter ASC Tube Traps.™ *Audio Amateur* has published articles on how to make devices like the Tube Traps but that was more effort than I was willing to exert, so I bought some from ASC.

Absorption

Absorptive material was applied to the walls from floor to ceiling beside and behind the speakers and along one side wall.

I used multiple 2x4-foot by 1-inch thick Owens #705-rigid compressed fiberglass panels glued directly to the wall surfaces between a stiffening frame made from 2x2-inch pine studs. The 2x2s were attached to the wall with drywall screws through the sheetrock (drywall) and into the interior studs. Each of these 2x2 frames included two vertical and three horizontal studs to prevent the drywall from flexing.

I covered these framed panels of compressed fiberglass with fabric and finished the edges using mahogany molding. This damping method serves a dual purpose: it absorbs reflected energy at frequencies above 200Hz and prevents the storage and release of low frequency energy by the flexing of the drywall panels. More about this later.

There are many commercial products available to damp wall reflections and if you are not a do-it-yourselfer check the following web sites for information about these products:

www.acousticsfirst.com
www.acousticscience.com
www.echobusters.com
www.kineticsnoise.com
www.owenscorning.com
www.whisperwalls.com

Here's a word to the frugal. Commercial sound-damping products are surprisingly expensive and none are more effective than the compressed fiberglass material that I used which is cheap and readily available from builders' supply stores everywhere. Most commercial products actually use the same or equivalent compressed fiberglass material. Foam is far less effective but cheaper.

Plaster and glass reflect sound very well. Cover as much of these surfaces as you can with absorptive or diffusive material.

Some furnishing items that work well to absorb reflections include upholstered furniture, bookcases filled with books (not CD jewel cases), and thick, fabric wall coverings.

Diffusion

To provide diffusion, I used rounded stone veneer from floor to ceiling over and around the fireplace opening, and wide-blade plantation shutters which completely cover the windows at each end of the room. The cylindrical objects in the rear corners of the room are ASC Tube Traps.™ They absorb low frequencies and have mylar panels on one side that reflect and diffuse higher frequencies. Their effects are adjustable depending on how they are oriented. Face the reflective side into the room and they absorb low frequencies and diffuse high frequencies. Turn the reflective side towards the wall and they absorb a broad spectrum of frequencies.

Decorative shutter panels can be placed on walls to provide diffusion and semi-cylindrical objects work well, too. Decorative columns are available from builders' supply stores and interior

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designers. Commercial diffusers are available from RPG Diffusor Systems, Inc. Check their web site at: www .rpgdiffusors.com

Note: my dictionary says that the word "diffuser" ends in "er" but RPG spells it "or."

Any material with an irregular (textured, not flat) surface may work to diffuse sonic reflections. You want to break-up the reflection and disperse the signal in many directions.

Energy Storage

To eliminate sources of stored energy, I used ASC Tube T raps in the rear corners of the room and made the walls stiffer and heavier in the front of the room near the speakers and subwoofers.

My house is constructed in the typical tract-house manner: half-inch drywall nailed to 2x4 studs placed on 16-inch centers. These "rubber walls" are a significant source of "room boom."

If you strike a wall that is constructed this way with the heal of your hand, hitting between the studs, you'll hear the problem. The spectrum of the boom from the flexing dry wall in my room was strongest between 60Hz and 80Hz confirming my listening impression of a 70Hz emphasis from the room. This emphatic boom was delayed as the energy was stored and released. The result was a blurring of detail and a loss of definition from the lower midrange down. Making the walls stiffer and heavier corrected the problem and improved the sound.

The fragility of the cloth over fiberglass materials prevented me from striking the treated wall to measure the spectral effects of stiffening and damping, but it surely sounds better now.

Bass energy can build up in corners which act like megaphones to project this energy back into the room. Tube T raps can soak up this excess low frequency energy and dissipate it as heat. This is seldom a major problem in rooms with large openings like mine, but I have five subwoofers. It can be more problematic in closed-off rooms.

That's All Folks

I began this issue by describing why people get bad sound in their homes and I want to finish on a more positive note. I've been in many homes where dedicated music lovers have taken the time to squeeze every bit of performance out of their audio systems. The set-up suggestions presented here are based on empirical evidence gathered from my own experiments and my observations of what others have done to make good sound.

It would be impossible to answer all the questions about acoustics in a 16,000 word essay even if I knew all the answers. But if you start by implementing as many of these suggestions as you find practical in your home, I guarantee that you will be impressed by the amount of control you have over sound quality. When you discover how much tuning you can do without spending a dime, you may find yourself doing a lot more listening and a lot less shopping.

With a little trial-and-error effort, you'll be well on your way to audio bliss. In the next issue of the **Journal** we'll start to discuss components and how they work, as we continue our journey. APJ

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