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Introduction

*This is the third **Audio Perfectionist Journal**. In this issue we will continue the discussion of subwoofers that began in Journal #2 and start to examine system components and the signal path through these components. Then we'll consider the question of how many channels we really need to accurately reproduce musical performances in the home.*

The subwoofer articles in **Journal #2** raised many questions from readers so we'll start this issue with a continuing discussion of that topic and offer some additional explanations about the previous articles. Then we'll talk about the signal path through an audio system.

It's important to understand the sequence and function of the basic components in an audio system in order to properly allocate resources for the best sonic results at the lowest cost.

The final article in this issue discusses the number of audio channels necessary for accurate music reproduction. This is another subject to consider when allocating resources to parts of a home entertainment system. Do you really need eight channels of amplification and speakers?

In This Issue

The article titled More About Subwoofers continues the discussion of subwoofers from a high-end perspective that began in **Journal #2** and explains the previous articles in greater detail. This article states my positions in simple terms and presents arguments to support them. It also answers some questions raised by readers following the publication of **Journal #2.**

The article titled *The Natural Order of Things* describes the purpose of various system components and the signal path through them. It attacks some common myths that mislead consumers causing them to spend too much on some components and not enough on others.

The article titled *How Many Channels Do We Really Need?* debates today's popular assumption that more is better when it comes to multichannel audio.

How It Fits Together

This issue begins the discussion of the individual components in an audio system by tracing the signal path through the system and dividing the system into sections. This information stresses the importance of the proper allocation of resources. Future editions of the **Audio Perfectionist Journa**l will concentrate on components from each section of the system, describing what they do.

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

More About Subwoofers

by Richard Hardesty

*The article "Subwoofers From a High-End Perspective" in **Journal #2** provoked many questions from readers. I got e-mail queries like: "What about the REL or Aerial subwoofers?" "Can I use the Vandersteen subwoofers with my small satellite speakers?" "How do I make my own passive high-pass filter?" Many people asked me to define the terminology used in the article. Let's start with some simple statements of my opinions and then I'll try to justify each position.*

Opinions

The sonic benefits achievable by adding subwoofers, as described in **Journal #2**, can only be fully realized using low-Q, sealed-enclosure subwoofers along with passive high-pass filters and full-range main speakers. Vented subwoofers are suitable for home theater only and are not acceptable for use in high-end music applications. Active high-pass filters and the bass management filters built into surround-sound processors are unacceptable for high-end applications. Passive high-pass filters are necessary and can be easily constructed by anyone with a soldering iron and a shrink-tube heat gun.

We'll look at each of these items separately while referring to the vented versus sealed enclosure graphs on page 15 of **Journal #2** and on page 4 and page 5 in this issue. But first I'll define some terms.

Definitions

I unwisely assumed that everybody had read my subwoofer series in *Widescreen Review* and was familiar with basic terminology. This seems not to be the case so here are some definitions.

An electrical high-pass filter is a network that allows signals above the nominal turnover frequency to pass without attenuation. The frequency where the effects begin is called the turnover or corner or crossover frequency. A high-pass filter blocks signals below this corner frequency by gradually reducing signal amplitude (attenuation) as frequency decreases. A first-order high-pass filter attenuates low frequencies at a rate of 6dB per octave below the corner frequency.

That means that when the signal decreases in frequency by one octave it will be reduced in amplitude by 6dB relative to the amplitude an octave above. The turnover frequency is usually the frequency where

the signal is attenuated by 3dB. Therefore an 80Hz high-pass filter will allow a signal at 120Hz to pass without effect, and attenuate a signal at 80Hz by 3dB. A signal at 40Hz (one octave below 80Hz) will be attenuated by 9dB relative to a signal in the midrange. A signal at 20Hz (one octave below 40Hz) will be reduced in amplitude by another 6dB for a total attenuation of 15dB. Get it?

Steeper filters roll off signals in the stop-band at a faster rate. A second-order filter attenuates stop-band frequencies at 12db/octave. A third-order filter has an 18dB/octave attenuation slope, a fourth-order filter has a 24dB/octave slope and so on.

Low-pass filters work the same way but they pass low frequencies and stop higher frequencies by attenuating them at various rates as described above.

Vented subwoofers are suitable for home theater only and are not acceptable for use in high-end music applications.

A loudspeaker driver in an enclosure acts as a mechanical high-pass filter. The characteristics of this mechanical filter can be modeled mathematically just like its electrical counterpart. A driver in a sealed enclosure is nominally a second-order high-pass filter depending on the system Q. (Higher-Q systems roll off a little faster and lower-Q systems roll off a little slower than second-order.) A driver in a vented enclosure is nominally a fourth-order high-pass filter depending on alignment. (As examples, a B6 alignment is a sixth-order filter and a QB3 alignment is a quasi-third-order filter.)

A vented subwoofer uses a resonating column of air to augment the output from the active drive element over a narrow range of frequencies. This type of subwoofer is sometimes referred to by other names such as bass-reflex, ducted port, tuned port or simply ported. Passive radiator designs substitute the mass of the passive radiator diaphragm for the mass of the column of air in a vent but they work the same way.

A passive radiator eliminates wind noise that may occur with a vent and the passive diaphragm can be weighted to allow enclosure dimensions that would be too small to include the port length necessary for low frequency tuning. In all cases, passive radiator designs are considered to be vented systems.

Filters, whether they are mechanical or electrical, have other effects on the signal in addition to attenuation. They cause phase shift, group delay and oscillation (ringing) after the signal stops. All these negative effects increase as the order of the filter is increased. A fourth-order filter causes twice as much phase shift as a second-order filter and doubles the amount of group delay.

How to Interpret the Graphs

The graphs published in **Journal #2** on page 15, and in this issue on pages 4 and 5, compare virtually all the characteristic differences between vented and sealed enclosure designs except transient response and excursion. Excursion will be illustrated later in this article.

You would have to actually measure impulse response or tone burst response to demonstrate transient response differences graphically and we'll get to that later on in the **Journal** series. For now let me just state that sealed enclosure subwoofers typically oscillate one and a half cycles after the signal stops (higher-Qs ring more and lower-Qs less). Vented enclosures typically oscillate three cycles after the signal stops (higher-order alignments ring more and lower-order alignments less).

The other characteristics become clearly evident when examining the traces in these graphs and I'll describe what each trace means here. The graphs on pages 4 and 5 compare the computer predicted response characteristics of a JBL 2235H 15-inch driver mounted in a 5.4 cubic foot sealed enclosure (page 4) with the characteristics of the same driver mounted in a vented enclosure (page 5). The sealed enclosure is designed for a Qtc of .5 and the vented design is a B4 alignment. The enclosure volume is the same in both cases.

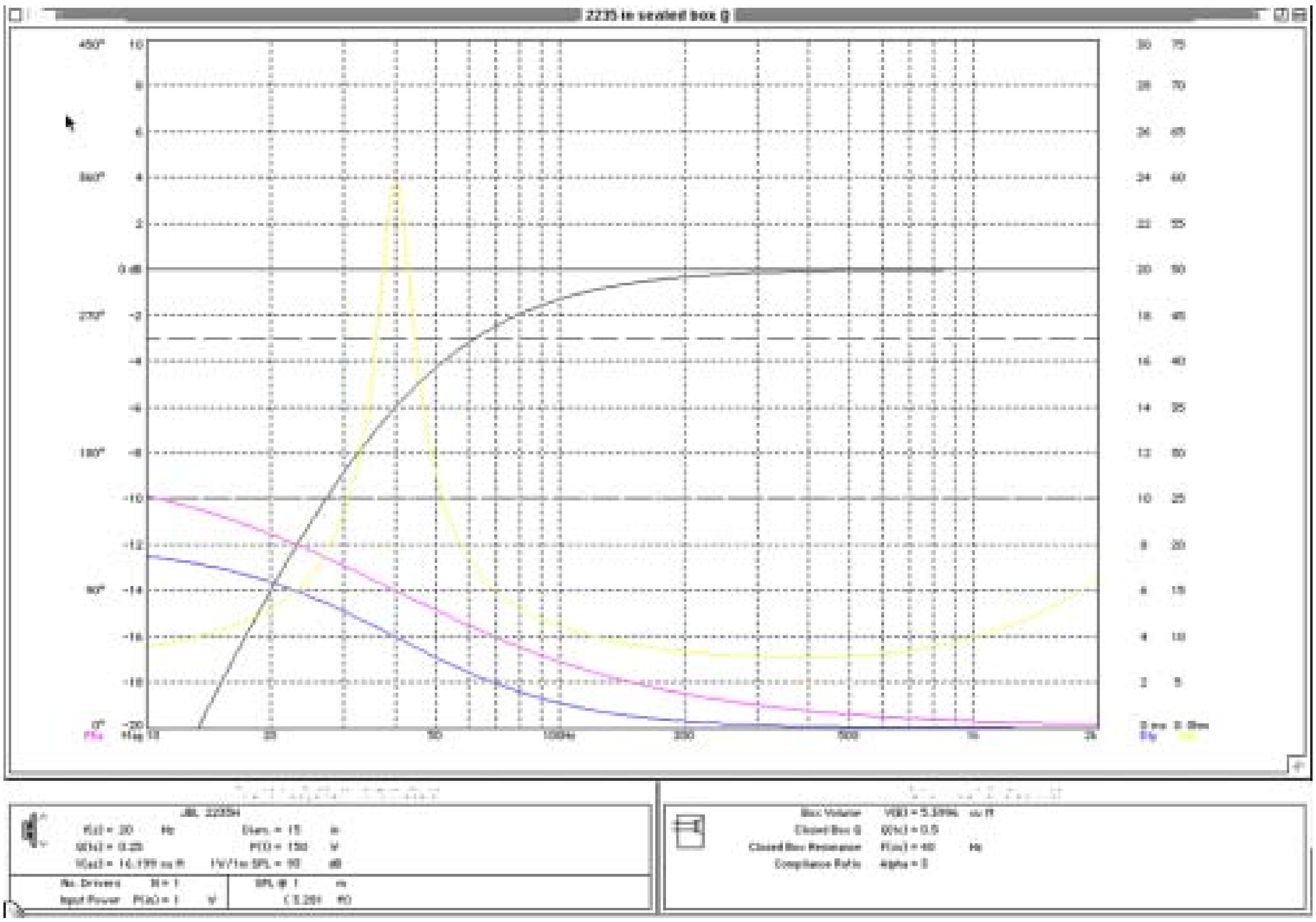
I designed these systems on my computer and I have constructed them and they are sitting in my garage at this moment, in case you think that this is all just theoretic mumbo jumbo. The real test units measure almost exactly as the computer predicted they would, so the graph traces represent real-world performance. Here is what each trace in the graphs means.

The black trace shows amplitude response versus frequency with amplitude relative to flat response (or 0dB) shown on the vertical or y-axis of the graph at the left, and frequency shown on the horizontal or x-axis along the bottom. Looking at the black traces you can see that the vented design remains flat down to 40Hz and rolls off steeply below this point. The sealed design begins rolling off gently above 100Hz. (This tapering response will be corrected with equalization in a powered subwoofer design.) The vented subwoofer is -20dB at 19Hz and the sealed subwoofer goes down to 13Hz before reaching -20dB. (A vented, powered subwoofer will need slightly less EQ for flat response but will still roll off at a faster rate.)

The yellow trace shows impedance magnitude versus frequency with impedance in ohms displayed at the far right on the y-axis. You can see that the sealed enclosure has a single impedance peak at 40Hz which shows the system resonance, and the vented enclosure has two peaks at 50Hz and 13Hz. The trough between these peaks shows the vent tuning frequency of 32Hz. At 32Hz the cone excursion (or travel) of the active driver will be minimum and the resonating column of air in the port will produce almost all the acoustic output. The sound that you hear will be produced primarily by the vent.

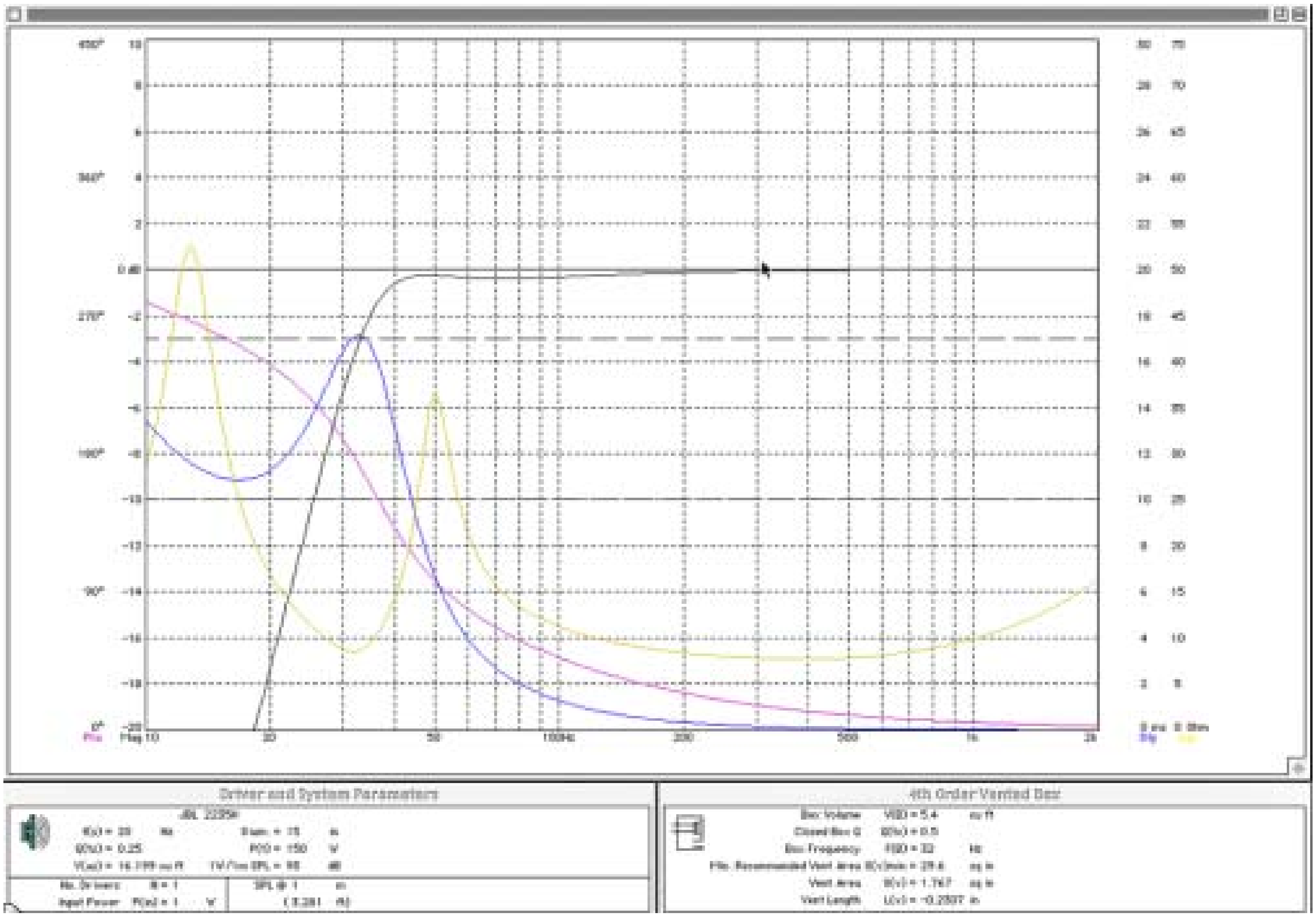
The magenta trace shows phase response versus frequency with the phase angle displayed on the y-axis at the far left. You can see that the sealed enclosure has a phase shift of 90 degrees at system resonance and the vented enclosure has shifted 180 degrees at resonance, almost completely reversing phase between 100Hz and 30Hz.

The blue trace shows group delay with the delay in milliseconds shown on the y-axis at the right. Group delay means that signals around a certain frequency will be delayed in time relative to signals in the midrange. Each millisecond of delay equals about one foot of apparent distance. A subwoofer with 17ms of group delay at 20Hz will appear to be seventeen feet farther APJ



JBL 2235H Driver in 5.4 cubic foot Sealed Enclosure

The illustrations on pages 4 and 5 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 (page 4) and in a fourth-order vented enclosure (page 5). 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.



JBL 2235H Driver in 5.4 cubic foot Vented Enclosure away from the listener than its actual physical position when reproducing a signal at 20Hz. The sealed enclosure has less than 8ms of delay down to 25Hz and the vented enclosure has about double that.

Group delay varies with frequency so the vented subwoofer will appear to be farther and farther behind the main speakers as the frequency of the signal decreases.

Vented Enclosures versus Sealed Enclosures

by Richard Hardesty

Vented subwoofers dominate the home theater market. Home theater subwoofers are hawked by specifying how loud they can play and stating some harmonic distortion number based on a measurement using a steady-state sinusoidal stimulus. These numbers can be very misleading if you intend to use the subwoofer for music. What about the three cycles of oscillation after the signal stops, which is typical for vented designs? Since this oscillation is unrelated to the input signal, isn't it 100% distortion? What about the fact that a musical waveform is transient in nature and is almost never sinusoidal?

If you think of bass as a separate entity that is unrelated to the rest of the spectrum, a specification like 110dB at 35Hz with 1.5% THD might be significant. For film sound it probably is all you need to know because, except for the musical score, movie bass is completely artificial and is created for effect on a Foley stage. Music is different.

Every musical note is made up of a fundamental tone and the harmonics that give the note timbre. Harmonic structure defines the instrument. A piano and a violin sound very different even when they play the same note at the same fundamental frequency because the harmonics from each instrument are different. The time relationships between the fundamental and the harmonics of each note are critical. What if the subwoofer is producing the fundamental and the harmonics are being reproduced by the main speakers? Shouldn't these component parts of the same note be in time with each other?

Music is transient in nature. With the exception of organ pedal notes, you'll be hard pressed to find any sine waves in music. Few musical notes are sustained for more than a few cycles. Most low frequency notes from plucked bass and percussion instruments start with a high amplitude transient spike which settles into a gradually diminishing wave form.

The piano can play the lowest note of any instrument in a standard symphony orchestra. The low A on a piano keyboard is 27.5Hz, which is quite a bit lower than the lowest note a double bass can play—a low E at 41.2Hz. Harmonics of these notes occur at the same frequencies as the fundamentals of the same notes an octave or two higher. That's why you can hear the bass playing on a portable radio that has no real low frequency response. If these harmonics are out-of-step with the fundamentals, as they would be when the fundamentals are reproduced by a "slow" subwoofer, the rhythm and pace of music are negatively impacted and tonal characteristics are altered.

The one advantage that vented enclosure designs offer is a reduction in cone excursion at low frequencies.

The important aspects of subwoofer performance for music reproduction are transient response and how well the subwoofer blends with the main speakers—not how loud the subwoofer will play. To protect your hearing, you should never listen to music at average levels exceeding 90dB. Subwoofers that can reach peaks of 100dB play loud enough for music reproduction.

Vented Advantages?

Vented subwoofers offer inferior performance compared to sealed enclosure subwoofers of equivalent quality in the following areas: transient response, phase response, group delay, and low frequency extension. You might ask why anyone would choose to make a vented subwoofer. There is one very good reason—high output. The one advantage that vented enclosure designs offer is a reduction in cone excursion at low frequencies.

Reduced cone excursion allows vented designs to play louder. A side benefit is that reduced excursion will also produce lower steady-state harmonic distortion measurements. Following are two more graphs comparing sealed and vented enclosures. These graphs show cone excursion versus frequency.

Here is the bottom line. If you want to reproduce the sound of bombs exploding at 120dB in your living room, you'll need a vented subwoofer. If you want to accurately reproduce the sound of musical instruments, you'll need two sealed enclosure subwoofers.

It's Just a Phase

Phase response is a key element here. When speakers are out-of-phase they work against each other and don't blend together. When left and right stereo speakers are wired out-of-phase one pushes while the other pulls. There is no image and bass is greatly diminished due to cancellation. Tonal accuracy goes out the window. Switch the plus and minus wires to one of your stereo speakers and see for yourself. (Switching the polarity of one speaker shifts the phase of that speaker's output by 180 degrees.)

What do you suppose happens when a subwoofer changes phase by 180 degrees within its pass band? Signals at 80Hz may be out-of-phase with signals at 20Hz. How can musical fundamental tones be integrated with their harmonics when the phase of the fundamentals is a moving target which is constantly changing?

Do you see why loudspeaker manufacturers want you to believe that phase is inaudible and unimportant?

Remember, the mechanical filter of the subwoofer driver in its enclosure is one element to consider and the electrical filter of the crossover network is another. Both filters, electrical and mechanical, cause phase shift and group delay. The steeper the slope, the greater the effect. This is why I recommend low-Q sealed enclosure subwoofers and passive first-order filters.

Small Satellite Speakers With Subwoofers

It would be wonderful if satellite/subwoofer systems provided satisfactory high-end performance. Manufacturers could quit making big, full-range speakers and we'd have lots more space in our living rooms. Unfortunately, achieving a seamless blend between a small satellite speaker and a subwoofer is virtually impossible. You can't get a completely satisfying blend because of the same phase effects that we have been discussing.

As a small satellite speaker reaches the lower end of its frequency range it will approach or pass through system resonance just like the subwoofers described earlier. As the resonance frequency is approached, phase shift begins to occur.

A vented satellite speaker will experience a complete phase reversal at a frequency near the crossover point between it and a subwoofer. Add the phase shift of the steep-slope, active high-pass filter required to prevent overloading the small speaker with low frequency energy and you have a situation that precludes a seamless blend between the satellite and subwoofer. A satellite speaker in a sealed enclosure will perform a little better but not enough to work as well as a full-range speaker with response to at least an octave below the crossover frequency. Even a sealed enclosure satellite speaker will probably require a second-order high-pass filter.

Do satellite/subwoofer systems work at all? Of course they do. THX has built their whole program around the concept. You can get lots more bass and much higher output from small satellite speakers by adding a subwoofer or two but, in my opinion, you'll never achieve the seamless blend between the satellites and the subwoofer that allows true high-end performance suitable for critical music listening.

Do you see why loudspeaker manufacturers want you to believe that phase is inaudible and unimportant? Phase is a difficult problem for them and they'd like to ignore it and want you to ignore it, too.

Active High-pass Filters

A passive filter contains only passive circuit components. Passive elements include capacitors, inductors and resisters. These elements require no external power but may drop some signal voltage, which is referred to as insertion loss. A single pole high-pass filter consists of only a series capacitance and has negligible insertion loss above the turnover frequency. Filters with steeper slopes are generally implemented with active circuitry to minimize insertion loss, size and cost. Active filters contain transistors, or more commonly ICs, and use power.

Small speakers must be filtered more aggressively in order to achieve the higher output levels and potential midrange benefits provided by adding subwoofers. This means active high-pass filters with steep slopes of second- to fourth-order. What do you suppose happens to the phase response of your main speakers when you run the entire signal through an active third-order high-pass filter? What do you think happens to the fidelity of your expensive amplifier when you put an inexpensive integrated circuit at the input and pass the entire signal through this circuit?

Active high-pass filters are simply an unacceptable option for a high quality audio system. Commercial crossover networks typically utilize cheap IC op-amps that add noise, phase shift and distortion to the output from your main speaker system. These negative effects completely offset any of the benefits provided by adding subwoofers in my opinion.

This same argument can be used against the use of active loudspeakers which use the same op-amps in thier crossovers.

Some have advised using no high-pass filter at all with the subwoofer(s) added simply to augment the bass of the main speakers. This is a bad idea. Using no high-pass filter at all precludes all the benefits of higher output capability and distortion reduction in the midrange and you still have the problem of phase shift in the satellites.

High Q Won't Do

In **Journal #2** I described the advantages of corner placement for subwoofers. This usually won't work when using subwoofers with a system Q of .7 or higher (which describes the vast majority of commercial designs) because in most rooms you'll get exaggerated bass response due to room gain. Corner placement will create a rising bass response that compensates nicely for the falling free-field response of a subwoofer with a Qtc of .5 but higher-Q designs will sound boomy with the additional bass lift provided by the corner. Higher-Q subwoofers have a steeper high-pass characteristic and inferior transient response as well.

Build Your Own High-Pass Filters

A passive, first-order high-pass filter consists of a value of capacitance placed in series with the input to the amplifier. You can either replace the input coupling cap in your amplifier with one of the proper value or you can build an external filter.

You can easily build a passive high-pass filter using a female RCA or XLR connector, a couple of high-quality capacitors, a male RCA or XLR connector, and some heat-shrink tube to hold it all together. Connect the ground terminals of the male and female connectors together with a piece of wire and solder the capacitor(s) between the hot terminals. The proper value of capacitance should be placed in series with each leg of a balanced circuit. Cover the whole assembly with shrink tube and heat to shrink. Simply insert the completed filter between the interconnect cable and the input to each amplifier channel (you'll need one for each channel). Plug the interconnect cable into the filter and the filter into the amplifier.

The formula for determining the necessary value of capacitance in farads is: $c=1/(2\pi fz)$ or capacitance in farads equals one divided by (6.28) x (crossover frequency in hertz) x (input impedance in ohms).

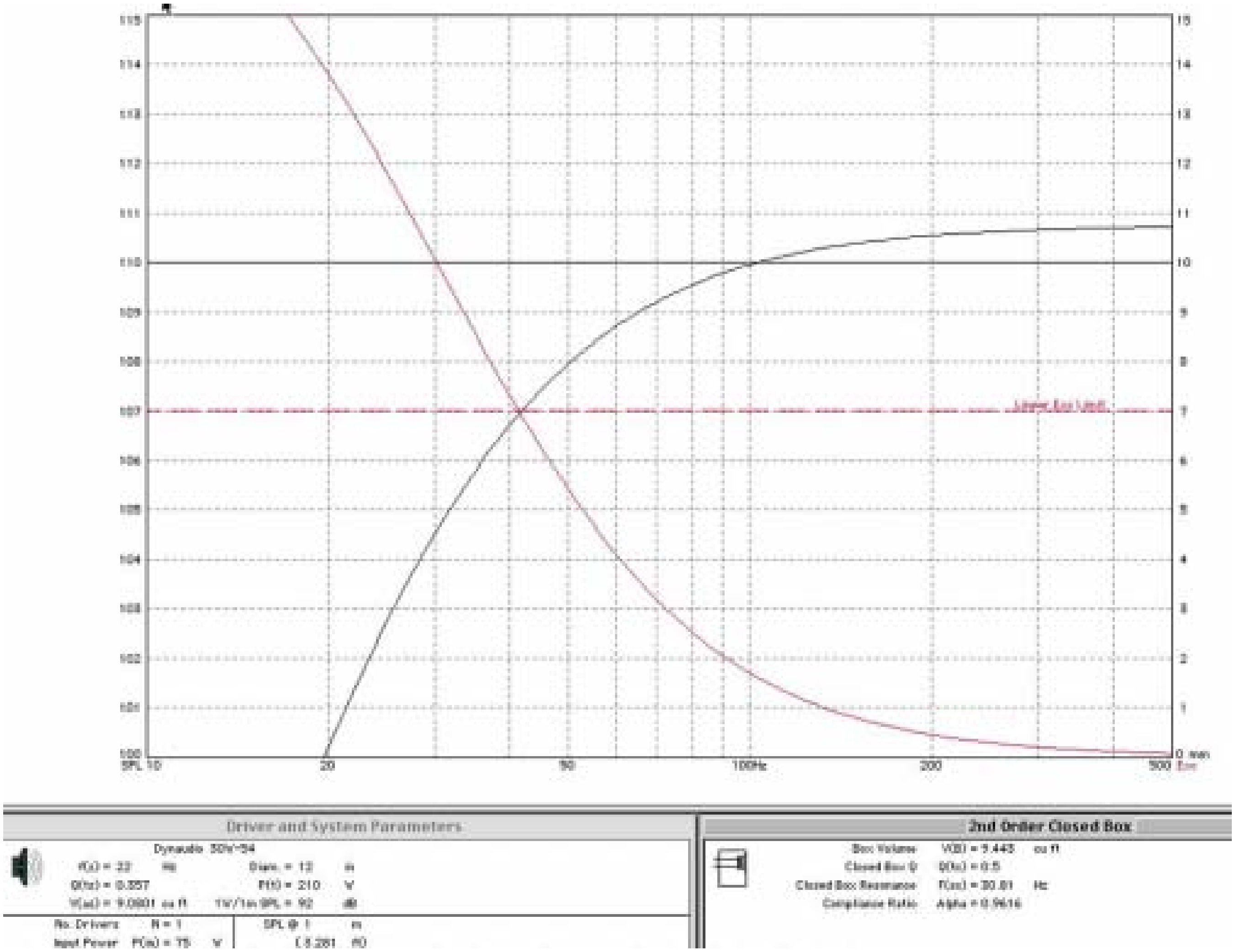
With a tube amplifier or a solid-state amplifier using FET inputs, impedance remains fairly constant with frequency and the manufacturer's specs will allow you to calculate the value of the necessary capacitors with reasonable accuracy. Manufacturer's specifications for balanced inputs sometimes state the impedance for each leg and sometimes state the total impedance. You'll need to know which is which if you make balanced filters.

Solid-state amplifiers with bipolar input stages can have substantial changes in input impedance at low frequencies. Here you need to insert the estimated capacitance and measure the -3dB point and adjust. You can determine the -3dB point by measuring voltage output from the amplifier using a signal generator.

Some adjustment is usually necessary to achieve perfect results and this is facilitated by the temporary box that comes with the Vandersteen subwoofers. It has dip switches to select various capacitance values for experimentation. If you make your own filters you'll have to cut-and-try to achieve the best sound.

Conclusion

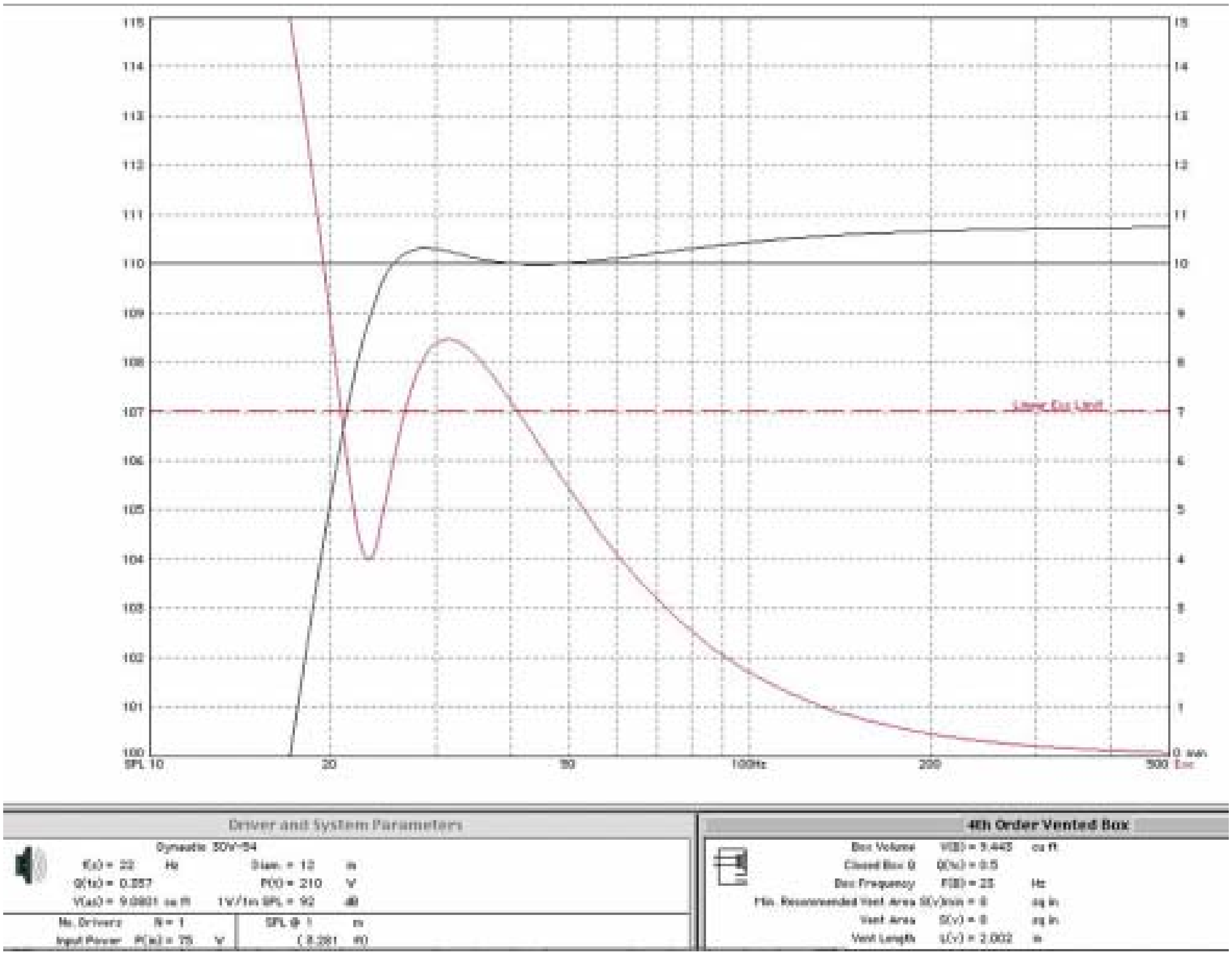
My advice is simple. Choose quality over quantity. Avoid steep-slope filters whether electrical or mechanical. Remember the facts about phase shift and group delay that were presented here. There will be lots more discussion about these topics in future **Journals**. APJ



Dynaudio 30W-54 in Sealed Enclosure

The graph above shows the predicted response of a Dynaudio 30W-54 12-inch driver in a 9.4 cubic foot vented enclosure. The black trace represents output level in dB SPL as shown on the vertical or y-axis at the left. The purple trace shows excursion in mm on the y-axis at the right. Frequency is shown in Hz along the x-axis at the bottom. Frequency range is 10-500Hz and is shown logarithmically. As you can see, an output of 107dB at 22Hz can be obtained without exceeding the linear excursion limits of the driver.

The graph on the next page shows the predicted response of the same driver mounted in a sealed enclosure with the same volume. The sealed enclosure design will reach the linear excursion limits of the driver at 42Hz without EQ, using the same input power as the vented example above. Equalization to compensate for the falling response of the system will require a reduction in power and output to prevent distortion caused by driving the speaker beyond the linear excursion limits. The sealed enclosure design will probably sound better but won't play nearly as loud as the vented design.



Dynaudio 30W-54 in Sealed Enclosure

The Natural Order of Things

by Richard Hardesty

Recording a live musical event requires the conversion of acoustical energy into electrical energy by the microphone(s). Air pressure waves from instruments and voices move the diaphragm in the mic and a tiny AC voltage appears across the microphone cable. The resulting analog waveform is then amplified, processed and converted again to some other form for storage.

If the storage medium is an analog record the electrical energy is converted back to mechanical energy by the cutter head, which makes physical grooves in the lacquer-coated master. If the storage medium is a compact disc the electrical energy is first converted to a digital code and then engraved into the aluminum foil of the CD. If the information is to be stored on magnetic media, such as recording tape or hard disc, the energy is converted to a concentrated magnetic field by the record heads, which magnetize ferrous particles on the tape or disc surface. If the original event is going to be broadcast over the airwaves the electrical energy is used to modulate radio waves.

To reproduce the musical performance in the home the information that represents the original event must be retrieved from the storage medium, converted back to electrical energy and then processed and amplified to a level which is sufficient to drive a loudspeaker. The speaker converts the electrical energy to mechanical energy in the form of sound waves. Then your ears convert this mechanical energy back to electrical energy for interpretation by the brain. And you thought this was simple.

The electrical signal that represents the acoustical information from a recorded musical performance flows through the individual components in a home audio system in a prescribed order. Understanding the signal route and what happens to the signal as it follows this path through the chain of components is important. It allows us to efficiently allocate our resources when purchasing or upgrading an audio system.

Most of us will have to set limits on the total cost of our audio systems and spending too much on one component and not enough on another can lead to disappointing results.

Too Much Here, Not Enough There

The most common error that budding audiophiles make is spending too much on speakers and not enough on the amplifier that drives them or the CD player that sets the limits for signal resolution well before the speakers get a chance to reproduce anything.

The speakers are the final components in the signal chain and they certainly are an important factor in the sound of an audio system, but they cannot reproduce musical information that doesn't come down the speaker cables in the form of an electrical signal. Speakers convert electrical signals to mechanical energy. They do not create information, they reproduce it. Speakers cannot improve signal quality.

If a low-quality CD player fails to retrieve musical information from the CD, the best amplifier in the world can't amplify that lost information and the best speakers in the world can't reproduce that information—it's gone forever. If the CD player successfully retrieves every bit of the stored data from the CD but parts of the signal get lost or distorted by some component along the signal path, the speakers can't replace that lost information or correct a colored or distorted signal in any way.

An accurate speaker accurately reproduces the signal that appears at the speaker input. If that signal is bad the sound that comes from the speaker must be bad, no matter how good the speaker.

Speakers, the Weak Link?

Speakers are almost always considered by novices to be the weak link in the chain of audio components. This idea is based on the ancient concept that harmonic distortion measurements determine the level of perfection that components achieve. Since speakers have more harmonic distortion than amplifiers or other system components they must be less perfect. This is a misconception as we'll see.

The idea of the speaker as the weak link has been supported by dealers, some of whom know better, because dealers make lots more profit on speakers than on other components. They want you to spend a larger portion of your budget on speakers to maximize their profits.

Magazines have also supported the idea because there are more speaker manufacturers than makers of other components and the magazines want to print as many reviews and sell as much advertising as they can.

Speakers get the most attention from hobbyists and magazine reviewers because even a novice listener can hear big differences between various models. It's easy to convince inexperienced listeners that speakers are the only component that makes an audible difference and many people steadfastly believe this. Any amplifier will do because they all have low distortion, they'll tell you—just get good speakers and you'll have good sound. All CD players sound the same, they say—bits are bits; only speakers make a difference.

The facts about speakers paint a different picture and here they are: Loudspeakers are the final components in the signal chain. Speakers reproduce only the signal that is fed into them. If that signal is bad and the speakers are accurate, the sound will be bad. If you put lots more information into a simple speaker you'll get lots more information out. If you put garbage into an outstanding speaker you'll get garbage out. Placing a better speaker at the end of a system of flawed components may actually make the sound worse because a better speaker may be more revealing of the flaws in the components that precede it in the signal chain.

Speakers reproduce only the signal that is fed into them.

Are speakers an important component? Absolutely. Are speakers the only component that really matters? Absolutely not.

You can easily hear the difference between a good amplifier and an excellent amplifier through modest speakers. When I was a merchant, I sold many $10,000 amplifiers by comparing them to less expensive amplifiers using a $1,200/pair speaker system. The demonstration was far more effective because of the modest cost of the speakers used in the comparison.

An Audio System is Not a Chain

If you hang too much weight from a chain the chain will break at the weakest link. Hence the saying a chain is only as strong as its weakest link. Making other links in the chain stronger will not make the chain stronger than the weakest link because that is where it will fail. Audio systems don't work that way.

Each link in a chain is the same and has the same purpose. Each component in an audio system is different and performs a different job.

An audio system is made up of a series of components and the signal passes through them sequentially. The components in the signal path can't make the signal better, they can only make it worse. Each component is imperfect and each one degrades the signal somewhat by losing information and adding noise and distortion. Additive degradation includes noise, distortion and coloration. Subtractive degradation involves the loss of information or resolution and the compression of dynamic range.

Each component is somewhat dependent on those that precede it in the system but an improvement in any component will probably be audible because a better component will cause less damage to the signal providing a better sounding result overall.

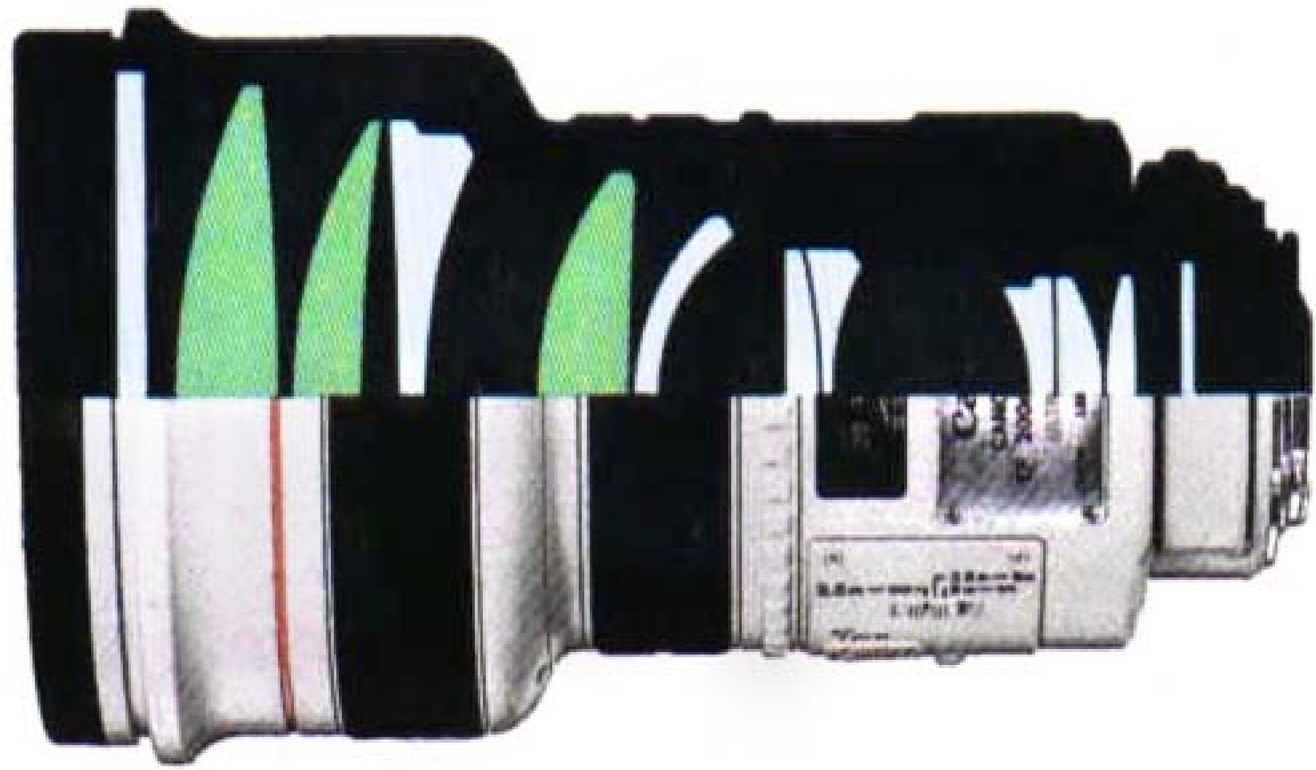
Strengthening some individual links in a chain won't make the chain stronger but improving any of the individual components in an audio system will make the system sound better. Every component makes a difference and improving any one will improve the overall sound. Some improvements will cost a lot more than others so we should study the purpose of each component in order to maximize fidelity and minimize expenditures. Let's consider the components in an audio system.

Elements/Components

The elements in a film or video projector lens make a good visual analogy to the components in an audio system. Each glass or plastic element in a multi-element lens bends or focuses the light rays passing through it. Each element must perform its particular modification to the light rays that pass through with a minimum of light loss. Cleaning the surface of any lens element will improve the image. Audio components work in a similar way.

Each component in an audio system has a specific job to do. Each transforms or amplifies the signal in some way. Each one must perform its task while losing as little musical detail as possible, and adding as little noise, distortion and coloration as possible.

No lens element is completely perfect and some light will be lost and some distortion will occur in each element. If any element is tinted, the entire image will be tinted. A better lens pro-



vides a better image because it does less damage to the light passing through it. Because the image starts out small at the film frame and gets larger at the screen, distortions that occur early-on in the light path will be magnified.

No audio component is completely perfect either. Each component will lose some musical information or detail, and add some noise and distortion or coloration to the audio signal. If any component adds coloration the overall system sound will be colored. Because the signal is being amplified, distortions that occur early in the signal path will be magnified.

The projector lens can't improve the image on the film and an audio component can't improve the quality of the recording.

No audio component can make the signal better. An accurate amplifier, for instance, should produce an output signal that is a higher-powered replica of the input signal. A better amplifier will make a more accurate replica causing less degradation to the original signal.

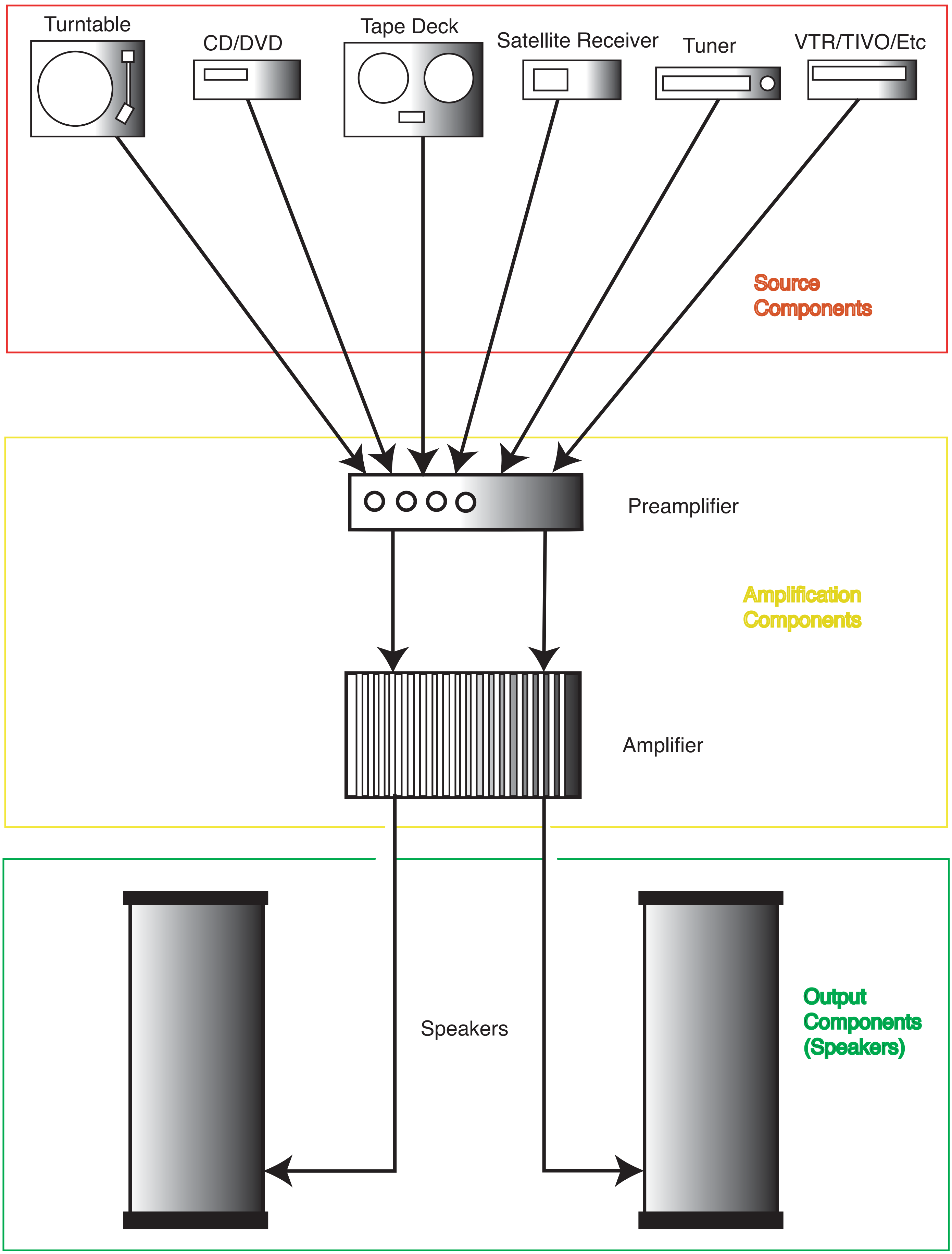
It can't improve upon the quality of the signal because there is no mechanism for signal improvement.

A better amplifier will pass along more information (less signal loss) and add less distortion, noise and coloration. Replacing an amplifier with a better one will improve the sound of the system regardless of the quality of the speakers at the end of the signal path because the signal will be degraded less by the amplifier along the way.

System Sections

An audio system can be divided into basic sections: the source components, the amplification components, and the output components. The cables that connect these sections together are important components, too.

The source components are all those devices that recover stored or broadcast data. These may include an analog turntable, a CD player, a cassette deck, an open-reel tape deck, a VCR, a DAT deck, a satellite receiver or set-top box, a


Turntable
CD/DVD
Tape Deck
Satellite Receiver
Tuner
VTR/TIVO/Etc
Source Components
Preamplifier
Amplification Components
Amplifier
Speakers
Output Components (Speakers)

DVD player, a laserdisc player, or a tuner—any component that retrieves information from a storage medium or the airwaves. These components may have two or more signal channels.

The source component sets the limit for signal resolution. The amount of signal detail that comes out of the source component is the maximum amount that subsequent components in the signal path have to work with. A better turntable system retrieves more information from the record and creates less noise and distortion by producing less rumble and contributing less wow and flutter. A better CD player retrieves more data from the CD and distorts this data less with lower jitter. A better tuner resolves more of the broadcast information and adds less garbage.

The amplification components include a preamp and an amplifier. The preamp may be part of a surround-sound processor. An integrated amplifier is a preamp and amplifier on a single chassis. A receiver is a tuner, preamp and amplifier in one box. A stereo system has two channels of everything and a surround-sound system usually has five or more.

The amplification components take the low-level signal from the source components and make it bigger. The preamp is the control unit that provides switching, buffering and volume control along with some amplification—in the case of a phono stage, lots of amplification. The amplifier receives the signal from the preamp and raises it to a level that can drive the speakers.

The signal is actually amplified over and over again until it reaches a power level that is high enough to drive the loudspeakers. The concept of a straight wire with gain is a misleading one. Every stage of an amplification component completely recreates the signal as you'll see later. The amplification components can't replace information that was not retrieved from the storage medium by the source components, and the amplification components can't remove noise or distortion introduced by the source components. If the turntable distorts the signal from the record with speed fluctuations (wow and flutter) the amplification components will amplify these distortions along with the desired signal. If the CD player produces a distorted replica of the signal during digital-to-analog conversion due to jitter, the amplification components will simply raise the level of these distortions.

The output components include loudspeakers or perhaps headphones. A stereo system has two speakers and a surround-sound system has more.

The output components receive the signal in whatever condition it's in after amplification and convert the amplified electrical energy into mechanical energy to move air and make sound.

There is no component in any speaker that can make the signal better. By the time the signal reaches the speakers it has passed through all the other components in the audio system. Information that got lost along the way cannot be recreated by the speakers, but noise and distortion can be. Noise and distortion in the signal will be clearly revealed by a high-resolution speaker. You'll hear any noise and distortion that was contributed by any component in the signal path.

If you trace the signal path through the system it should become clear that each component can only work with the signal it gets from the component that precedes it in the signal path. But that doesn't mean that replacing a downstream component with a better one won't improve the sound of the system.

Since each component degrades the signal a little, a better component will provide a better overall result because the system as a whole will now degrade the signal a little less. Added coloration is a slightly different matter.

If one component adds coloration to the signal the sound will be colored. Trying to cover up that coloration by adding components with "synergistic" colorations is a very bad idea. A coloration will remain until the last colored component is replaced. Each component should be as neutral as possible in tonal quality.

You can also see that adding more components to the signal path will degrade the integrity of the signal even more. Every additional device will add noise and distortion and lose information. Every additional device will require another set of cables. The simplest signal path is almost always the best sounding and the least expensive. Consider adding components carefully.

Adding a stand-alone D-to-A converter to improve the performance of the DAC in the CD player adds another component with another power supply along with another set of cables. These additional components will add noise and distortion and may lose signal information. Be sure that the outboard DAC is a lot better than the one inside the CD player or this expenditure may not result in an audible improvement.

In future **Journals** we'll delve more deeply into each major system component and describe how each one works in greater detail, but in this issue we will concentrate on resource allocation. Let's take a brief look at some of the individual components along the signal path.

Sources

The quality of the original recording or broadcast sets the absolute limit for resolution no matter how good your reproduction system may be. The best audio system in the world won't make a bad recording sound good. The source components must retrieve the recorded or broadcast information and they set the limits for the level of resolution possible from your playback system. If you don't get all the information off the disc (or other source) that information won't come out of your speakers as sound. Read that last sentence again.

Time and time again I have seen disgruntled hobbyists with expensive speaker systems listening to $400 CD players and wondering why their speakers don't sound like they did in the store demonstration. They're only bits, they say. *Sound & Vision Magazine* says they all sound the same.

A CD or DVD player is the most common source for high-end audio or home theater systems today. These players read tiny pits on the discs with a laser beam and perform digital-to-analog conversion on the data that is retrieved from the disc. This is an incredibly difficult and demanding task that requires a precision instrument to achieve high quality results. A $79 DiscMan can read a CD but the analog signal that comes out is an inadequate source for a high-end audio system.

The signal from the source component sets the limit for resolution for the entire audio system. No component in the amplification or output sections of the system can recreate musical information that was lost or damaged by the source component.

The CD player I use (Wadia 860) costs more than my speaker system. I consider this a proper allocation of resources. I use a separate high-quality DVD player (CAL CL 2500 DVD) for movies.

Amplification

A preamp may not appear to do much more than switch input signals and provide volume control and with line-level sources that's basically true. (With phono sources, a preamplifier can provide far more gain than the amplifier.) They may not seem to be important but preamps make a significant sonic contribution and, because they work with tiny signal levels that will later be amplified many times over, the sonic character that preamps impart to the signal will be a major factor in the overall sound of the system. Good preamps are expensive but worth it.

Amplifiers are the most underrated and possibly the most important single component in an audio system. Many people believe that amplifiers are nearly perfect because they all have low harmonic distortion. There is a large and very vocal group in the audio world that believes there are virtually no audible differences between high quality amplifiers. Believe me, nothing could be farther from the truth.

The straight wire with gain analogy, which is offered as a description of the ideal amplifier, is completely misleading. The input signal to the amplifier and the output signal from it are merely cousins, and in most designs distant cousins at that.

Each stage of the amplifier recreates a replica of the signal from the stage that precedes it. The amplifier doesn't take a tiny signal and make it bigger—the amplifier creates a big signal that replicates the tiny one. You could say that the signal you actually hear is created in the amplifier rather than passed through it. (Future **Journals** will consider amplifier design in detail. I'll make a strong case proposing that amplifiers rather than speakers are the weak link in an audio system. How's that for controversy?)

I'm listening to a variety of amplifiers these days including Levinson 33H monos, a Proceed HPA-3, a Theta Dreadnaught and a CAL CL 2500 MPA. Each of these amplifiers is outstanding in its own way, but I am still searching for the perfect one to live with forever. When I find it **Journa**l readers will be the first to know.

Speakers

There are many terrible speakers available, of course, but the good ones are better than most people think. Vandersteens, Thiels and Dunlavys have better impulse response than the compact discs we play through them! These speakers are demonstrably more accurate in time and phase than the primary recorded music source of the day.

Speakers, like all other audio components, are not perfect. But their faults are remarkably agreeable with the human hearing mechanism. The acoustic errors that they make are similar to and usually no greater than the contribution from the natural environments in which we listen.

Modern high-quality speakers have very low distortion above the bass frequencies. Depending on how and what you measure, speakers may have less harmonic distortion than certain amplifier types such as single-ended triode tube amps. Most high-quality speakers have amplitude response that is sufficiently linear to deliver good sound in real rooms. A good speaker may have minor frequency response errors, but if these are balanced so that octave-to-octave energy is balanced, the errors are relatively benign and will be swamped by the room's acoustic contributions. Good speakers won't deviate from flat response by more than ±1.5dB over the full audible range.

Most speakers have gross phase response errors which have been accepted in the era of the compact disc because the CD is not capable of delivering a linear-phase signal. 96kHz/24-bit LPCM on DVD and Super Audio CD discs are about to change all that. Dunlavy, Thiel and Vandersteen speakers are time- and phase-correct today and will be ready for tomorrow's improved recording technologies. This whole phase thing will be explained in great detail in future Journals.

I own Vandersteen 3A Signature speakers and 2WQ subwoofers and this $6,000 combination beats anything else I've heard regardless of price, with one exception. I can't use the Vandersteen Model 5s for amplifier comparisons because they have an internal bass amplifier and can't be operated full range like the Model 3s. I have a pair of Vandersteen Reference Monitors on order.

Cables

High-end cables are the biggest scam in audio and some of the most expensive ones perform very poorly. But that doesn't mean that cheap ones will do or that cables aren't very important. Cables can dramatically change the sound of an audio system.

Cables are an important component in the system. If you connect the world's best preamp to the world's best amplifier with cables that lose information or add distortion and coloration, the signal will be degraded by these cables, just as it would by any other component with similar aberrations.

Good cables are extremely important and the marketplace is a minefield of scam and hype. Anybody with a crimping tool can call himself a cable engineer these days. I'm going to devote lots of space in future Journals to the subject of audio cables but in the meantime be very careful. Don't buy any expensive cable product without carefully listening to it in your own system. Most of the mega-dollar cables are really bad and should be avoided.

Conclusion

A smart buyer will consider individual audio components in the context of a complete system. The proper allocation of resources requires a balance of expenditure between the major sections of the system. If you spend too much on one section you will surely short-change another.

If you are upgrading an existing system and you have one component that is of substantially lower quality than the others, change that first. If everything is well balanced quality-wise, start at the beginning with an improvement in the source section of the system. Then move down the signal path and improve amplification and finally the speaker system. Trust me, you'll get far more bang for your buck this way. APJ

How Many Audio Channels Do We Really Need?

by Richard Hardesty

As home theater becomes the ubiquitous basis of mass-market home entertainment, we're going to see a big demand from the general public for multichannel music recordings. When your neighbors have invested all their spendable cash in 5.1 channels of amplification and speakers for their home theater surround sound systems, they will want sound to come out of all those speakers when they play music. I predict that content providers will jump on this bandwagon and offer surround sound music recordings. More is better, right?

Today, surround sound music recordings require data reduction and that means perceptual coding schemes like Dolby Digital and DTS, which, regardless of the advertising claims to the contrary, are a giant leap backward from CD-quality sound. I won't even bother to discuss Dolby Pro Logic matrix encoding, which has been a dismal failure for music recordings.

The relative merits of Dolby Digital versus DTS Digital Surround have been debated in the press but in my opinion this is like asking "which would you rather have, a broken arm or a broken leg?" Dolby Digital sounds pretty good considering that at least nine tenths of the original recorded data is thrown away, but it is certainly not a high fidelity process. DTS claims to be "completely transparent" to the original master recording but consider this. If you put six channels in the space formerly filled by two, the compression ratio required is at least three-to-one. (It's really worse than that because a lot of side-band data has to be transmitted along with the recorded information to facilitate decoding.)

Data compression at a three-to-one ratio means discarding two thirds of the data from the original recording. That means that two thirds of the recorded information is deemed to be inaudible and is simply thrown away. This is done by a process called perceptual coding. Perceptual coding algorithms analyze the recorded information and retain data that is thought to be necessary and discard data that is thought to be inaudible due to masking and other effects.

Data rate comparisons are somewhat misleading because DTS is a forward-adaptive system which includes more side-band data required for decoding. DTS does operate at a higher data rate than Dolby Digital, providing just enough additional information to allow the listener to hear just how bad data-reduced recordings sound compared to uncompressed recordings. (An experiment you can try for yourself is described later in this article.) The recorded information that is discarded may be inaudible to the folks at DTS but it's certainly not inaudible to me.

What about tomorrow when technologic advances will bring us the capability of greater fidelity along with more channels of information from sources like DVD-Audio and Super Audio CD? How many channels will it take to provide the ultimate spatial effect? Dolby is already pushing 6.1 channel surround sound with the EX system, many surround sound processors offer matrix-derived side channels for a count of 7.1, and Tom Holman (of THX fame) has demonstrated a 10.2-channel system at trade shows. Will more channels actually improve spatial realism or is this just another excuse to sell more hardware? Does surround sound really provide a better spatial effect than good stereo can?

Two Eyes, Two Ears

We perceive the world around us using many senses. Our visual and aural perceptions utilize dual receptors—we have two eyes and two ears—so that we can extrapolate the third dimension of depth or distance. We see and hear from two slightly different locations simultaneously, allowing our brains to calculate not only the precise direction of the sight or sound, but the proximity of its source as well. We can tell by sight or sound that something is standing before us and just how far away it is. We can tell by sound when something is positioned in front of us, to either side, or behind us and whether it's approaching or moving away.

But we can be fooled under the right conditions. If you photograph a scene from two slightly different camera positions simultaneously, and project the resulting images so that each eye sees only one of them (like the IMAX 3D process where the viewer wears a hood which determines what each eye sees), these flat images projected on a flat screen seem to become three-dimensional. A high quality stereo recording, played back on a properly configured stereo system, delivers a holographic sonic image in front of you. This image is three-dimensional in every sense of the word (to a listener who is positioned correctly, equidistant from each speaker).

Do you think that the IMAX picture would be even more dimensional if three cameras were used? How about four? Since you only have two eyes, it's easy to understand intuitively that three or more images, projected together, would simply blur the result rather than improve it. Shouldn't sound work the same way? In fact it does, but there are some additional considerations.

Sound Is More Primal

Although we cannot see what is behind us, we can hear and precisely locate the direction and distance to the source of a sound anywhere around us, including its height or distance from the ground. This is important for survival. It allowed our ancestors to hear predators sneaking up behind them in the jungle, and it allows us to hear the mugger approaching us from behind while we're at the ATM machine. Our field of vision is a little more than 180 degrees horizontally, and far less than that vertically, but we can accurately hear over a 360 degree field in both planes.

If you block one eye while watching an IMAX 3D movie, the third dimension collapses and the image looks just as flat as any conventional film. If you move off-axis from a position that is perfectly centered in front of your stereo speakers, the same thing occurs: the sonic image collapses. Can this be remedied by adding more channels of audio?

Sight and Sound in the Real World

We see the world through two eyes that are spaced slightly apart allowing us to observe objects from two different perspectives simultaneously. From these two vantage points we can mentally calculate the distance to an object by triangulation and we can see slightly around the object to observe its depth. The IMAX 3D movie system captures two images from two different perspectives just like our eyes do. These two images are projected on the screen with alternating frames from each perspective. First a frame from the left image, and then a frame from the right image is projected on the screen and so on.

The viewer wears a hood with synchronized LCD lenses over each eye. These lenses switch between clear and opaque states to block the left eye while the right image is being projected on the screen, and then block the right eye while the left image is being projected. The left eye sees only the image from the left camera perspective and the right eye sees only the image from the right camera perspective. The result is assembled in the brain as a remarkably life-like, three-dimensional visual experience.

Our hearing is even more sensitive. We locate the direction from which sounds originate by several mechanisms. Inter-aural time delay is a primary method and the amplitude differential between the ears is an additional aid. Stereo simulates these cues in reverse.

Real sounds generally come from a single place in space. The sounds from a musical instrument or a human voice emanate from the single location of that instrument or person. The sound from a stereo system emanates from two stereo speakers and this can create the illusion that the sound source is somewhere other than the position of the speakers, usually between and behind them. The speakers have to be spread apart at a fifty to sixty degree included angle relative to the listener for this illusion to work. If the speakers are placed closer together, the illusion disappears and the listener perceives a single sound source. What do you think happens when we place additional speakers for added audio channels in between the primary stereo speakers?

Real World Directional Determination

The widely accepted concept of inter-aural time delay says that we determine the direction of the source of sounds by the arrival time differential between each of our two ears. If a sound arrives at the left ear before it arrives at the right ear the sound must come from a place towards the left of the listener. How far left is supposedly determined by how much time elapses between the arrivals at each respective ear. This seems plausible for transient sounds but how about continuing signals? What are we mentally timing here, the arrival of the pressure peak on each cycle?

If you have ever tried to locate a cricket chirping in your kitchen you know that the direction of origin of sharp, transient sounds is very difficult to determine. The position of a bird warbling at a similar frequency is easily discerned, however.

I believe that the brain performs a phase null on the arriving signals to each ear and calculates the direction of the source, queuing on the first arrival to establish whether the origin is left or right of the listener.

The concept of head-related transfer function says that the shape of the ears and head alter the spectral balance of identical sounds originating from the front and to the sides of the listener. THX makes much of the necessity of altering the signal to the rear channel speakers to "timbre" match the sound of the front and rear speakers due to this spectral shift.

This idea doesn't stand up to scrutiny. Turn your head ninety degrees or more while listening to your stereo and see if the spectral balance of the sound is altered. If you hear any frequency-related change at all, your brain is wired differently from mine.

I believe that the shape of the ears is very significant to our ability to localize the origin of sound, but I think that phase plays an important role here as well. When I turn my head relative to the source of sound I don't hear a spectral shift. I believe I hear a phase shift.

So what is my point here? I'm trying to make it clear that when it comes to human perception, all is not known. But that hasn't kept many experts from proclaiming that it is.

Many studies have been undertaken and papers written about inter-aural effects and head-related transfer function. These tests are presented as objective assessments and cloaked in jargon that makes them seem very "scientific." At the bottom of all these tests is a simple experiment: arbitrarily chosen listeners are asked "can you hear that?" These listeners' subjective opinions become the basis for all these supposedly objective truths. If you think that these tests prove anything, think about this.

If one hundred percent of the members of a group of listeners can't hear a tone or other stimulus under test conditions that doesn't prove that the tone or stimulus is inaudible. It only proves that those people under those conditions couldn't hear it. What about more experienced listeners with more sophisticated audio systems listening under more natural conditions?

Scientific "truths" about the function of the human senses are based on tests that almost always rely on the subjective response from humans who are asked for their opinions. How does the optometrist with all those instruments correct your vision? He alternately places two lenses in front of your eyes and asks "which is better, A or B?"

If all these scientific tests are based on the subjective impressions of unskilled listeners, aren't your own personal experiences just as valid? Can't you experiment and determine what works for you personally? I knew you could.

In my opinion—which is based solely on my observations—the human ear/brain mechanism is very sensitive to phase and uses this factor to analyze all sounds for direction of origin, distance from the listener and proximity to surrounding reflective surfaces. A time- and phase-accurate stereo system can do a remarkable job of recreating these phase cues using two properly spaced speakers.

Testing in the past has concentrated almost exclusively on amplitude and spectral differences and, for the most part, ignored phase response when analyzing how we hear. I know that phase is an important aspect that has been overlooked and it will be discussed extensively in future **Journals**.

Surround Sound—Brute-Force Spatiality?

In a commercial movie theater, stereophonic audio is virtually impossible. People sit too far from the speakers and no one is perfectly centered between them. The front speakers are placed behind the screen and the rear speakers are mounted against the wall, so that all sounds arriving at the listener become a mishmash of direct and reflected energy that is very smeared in time and phase. The center channel speaker can be twenty feet closer to the listener than the left and right front channel speakers because all are arranged in a row against the front wall. If sound is panned hard left, it will appear to come from the left side of the screen, and vice versa, but that is not imaging, it's multitrack mono. A home stereo system can place an image to the left of the left speaker, if the proper phase information is reproduced by the right speaker and the listener is positioned properly between the speakers. That's imaging.

It takes two speakers, both reproducing an acoustically related signal with small differences in amplitude and phase, to create an image. There is nothing "spatial" about a mono signal coming from a single speaker. It sounds just like a mono signal coming directly from one speaker, which is what it is.

We can create a semblance of spatial effect in a movie theater by adding more speakers and redirecting the sound from one to another. This is a brute-force method of simulating a spatial effect, but there simply is no other way to provide "enveloping" sound under these conditions. How about in the home?

The Sound of a Space

Your brain can easily and precisely locate the source of a mono signal regardless of the direction from which it arrives. That's how you hear real sounds in the real world. A stereo system can fool you and make sounds appear to come from a position other than the two speakers if the speakers and the listener are placed properly.

If you were surrounded by a gazillion speakers driven by a gazillion channels of discrete audio, you would not have much spatial effect unless these speakers were fed an acoustically related signal in pairs, and the pairs were placed about sixty degrees apart. A 360-degree circle can be divided into only six, 60-degree sections.

Spatial effect is created by two sources reproducing related signals that differ slightly in amplitude and phase. The effect occurs between the speakers, not at them. Beyond a certain point, adding channels will actually depreciate this effect, not improve it. Don't believe me? I'll give you some experiments to try and then you can decide whether you agree with me. If you decide that you don't, just keep buying more stuff until you run out of space or money. I won't mind.

More is Better, Right?

I'll assume that you have a high quality home theater system with the front channels set up in an acceptable manner for stereo reproduction (about 60° included angle between the front left and right speakers, with the speakers positioned well away from the walls), or know someone who does. Your seated position must be exactly centered between, and an equal distance from, each of your front speakers. If this doesn't describe your setup, the following experiments will be meaningless.

Listen to a good stereo recording using only the front left and right speakers and no signal processing. Unplug one speaker and listen again. (You can switch the signal to mono if you want to, it doesn't matter.) Is there any question that two channels are better than one? Two speakers can create a sound field with three dimensions. One cannot. A single speaker sounds like a single source whether the signal is mono or stereo or multichannel.

Now switch the signal to mono and listen through two speakers again. All sounds should appear to come from the center, midway between the speakers. No depth, no spread, no third dimension, no space between individual sound-producing elements, but it still sounds more natural than a mono signal coming from just one speaker, doesn't it?

Play some multichannel material that you are familiar with, using either three channels or five channels for reproduction, and listen to the sound of three front speakers. (Ignore the sound from the surrounds or turn them off.) Now reconfigure your controller for "phantom" center channel and play the same material again. Are three channels across the front better than two? Or does that center channel sound like a flat, mono source, just stuck in the middle, and never fully integrated with the overall sound?

Many listeners have found that even matched center channel speakers don't sound quite right and they have experimented with methods to improve the situation. Adding excess delay to the center channel helps. Using a bleed box to leak some center channel information to the front left and right speakers and vice versa will sound a little better but not as good as turning off the center channel speaker altogether. As long as you are sitting in the middle, adding a center channel speaker depreciates the spatial effect of the system, in my opinion, but what if you invite friends over to share your home theater system?

Move over to one side and listen in phantom mode again. Time- and phase-accurate speakers will deliver a pretty good image well off-center, but if you keep moving to the side the sound will eventually begin to pull to the side in the direction that you have moved. If that sound is dialog, which should appear to come from a character on screen, the effect can be disconcerting. Adding a center channel speaker will definitely improve this situation for a listener seated off-axis while watching a film.

A Center Channel For The Rear?

As we have discussed, you can't create much true stereo effect in a commercial movie house, so the more channels utilized to simulate spatiality the better. A center rear channel, matrixed from the discrete left and right surround signals, will probably offer a significant improvement in sound directionality for most listeners at the movies. Do you need this in your home? Not if you and a companion sit in a position centered between your surround speakers. Do you want to have it? Maybe.

How many people usually watch when you play a movie on your home theater system? Do some of them sit off to the sides, well away from the "sweet spot" required for good spatial effects? If so, then you may want to invest in a rear center channel speaker and the network and amplifier to drive it.

If you are like me and share your system with only one or two others, my advice is to take the money that you would spend reproducing additional channels and invest it in upgraded components elsewhere.

Surround Sound For Music?

Live musical performances, almost without exception, occur on a stage in front of the listeners. The sound of a live performance has depth and dimension and each instrument has a specific position in space but all this occurs in front of the listener. In a small venue there is little or no contribution from the room and the only "ambient" sounds come from the crowd surrounding you. These sounds will probably be more of a distraction than an addition to the natural recreation of the original event, but it is conceivable that a tastefully done surround sound recording could actually add to the sense of realism. What are the chances that surround sound recordings will be tastefully done?

The Lyle Lovett CD *Joshua Judges Ruth* on Curb/MCA records is a very good stereo recording. It has also been released in 5.1-channel DTS Digital Surround and comparing the two versions is very interesting.

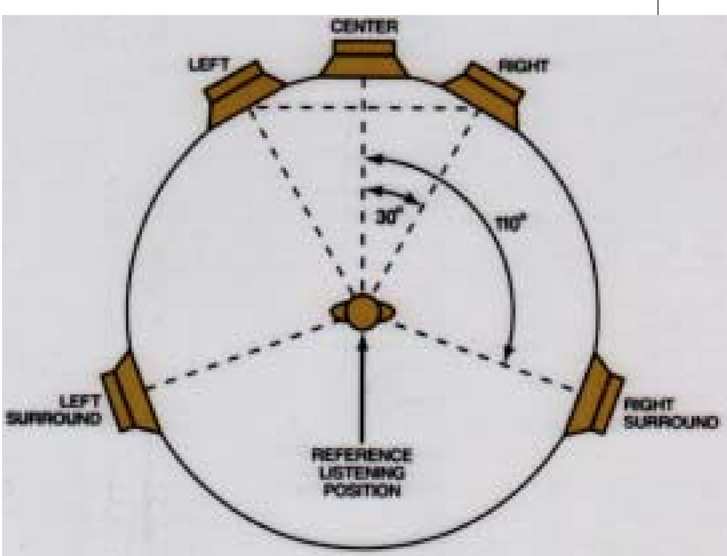
The stereo version has natural sound and good spatial effect with the band appearing to be spread across the stage in front of the listener as it would be in a live performance. The multichannel version is very gimmicky with backup singers and percussionists coming from the rear speakers in a manner that I find disconcertingly unnatural. I find the stereo version to be convincing and the multichannel version to be humorous. But there's more.

The piano and Lyle's voice remain in front where they belong on the 5.1 disc but are stripped of all the subtle sonic details that provide natural harmonics and decay. These things were deemed to be inaudible by DTS' perceptual coding algorithm and discarded for data compression. The lack of these details is clearly audible to me on my system. Try this comparison for yourselves and see if you agree.

The Columbia Music Video DVD release of James Taylor, *Live At The Beacon Theatre* has a Dolby Digital 5.1-channel sound mix and a 48kHz/20-bit PCM stereo track, too. Comparing these two different recording technologies can be enlightening.

The 5.1 track puts audience sounds all around you, which is appealing, but the performers on stage don't sound very good and their sonic images are not in focus. The stereo PCM track offers clearly superior fidelity and, in my humble opinion, a much better spatial presentation. You may find it helpful to turn off the picture when you do this comparison so that your attention is focused on the sound.

You may decide that you enjoy the multichannel mix enough to accept the sacrifice in fidelity, but I think you should be aware of the trade-off required.



The ITU Standard.
One valuable guideline for multichannel recording and speaker set-up is the ITU standard. It positions the listener at the center of a circle of speakers. The center speaker is located at 0°. Left and right are at ±30°, forming an equilateral triangle with the listener. Left surround and right surround speakers are positioned at ±110°, with tweeters at or slightly above ear level.

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So, How Many Channels Do We Need?

I think that music recorded in a small venue sounds best using only two channels for reproduction. It's hard for me to understand how anyone could imagine that hearing bongos over your left shoulder and back-up singers over your right shoulder simulates a real musical experience and sounds natural. Adding a bunch of artificial reverberation simply blurs definition and detail and defocuses the frontal image, but this is less offensive than putting instruments in the rear channels and is a matter of taste.

Surround Sound for Music is Like Bowling Balls for Fish

A large venue like a concert hall naturally adds a lot of reverberant sound to the direct sound emanating from the instruments on stage. Accurately reproducing some of this "hall sound" may actually improve the reproduction of the original event. To get the sound from the back of a large venue like a concert hall requires double stereo—two channels in front and two channels behind. A tastefully made recording might sound much more spacious with four channels. And four is the minimum requirement for film sound, too. (Plus subwoofers, of course.)

Film sound is distinctly lacking in spatial effect and fidelity and was designed for exaggerated impact. Surround sound for movies can be a lot of fun and it helps to overcome the sonic limitations of film soundtracks by distracting the listener with special effects. Fidelity is a moot point because there is no "real" counterpart for most film sound effects. They are fabricated on the Foley stage and are totally artificial.

If you like to share your home theater system with several friends, you'll need to add a center channel in the front, and maybe a rear center too. If you have a very large room, side channels may improve the experience of film sound as well. Remember, film sound is almost always multitrack mono with no real spatial effect.

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Conclusion

More may actually be better, or maybe not. If you are a serious music listener who values fidelity, probably not. Don't be overly enthusiastic and rush out to buy every new gadget that is offered for sale. Start by determining how you will actually use your home entertainment system. Then listen and decide for yourself. Will more stuff really give you better sound? APJ

New Surround Sound Formats for 2004

by Richard Hardesty

*This **Journal** was written before multichannel DVD-Audio and SACD discs were readily available. Now we can have fairly high resolution surround sound audio recorded with lossless compression. I've heard demonstrations of all the latest surround sound formats and systems and my opinions have not changed. I enjoy surround sound for movies and music videos and prefer higher quality two-channel reproduction for serious music listening.*

If you are considering multichannel music reproduction you should remember these facts: six channels will either have to cost three-times as much as two channels or be one-third as good. Six crumby speakers may sound better than two crumby speakers but two good speakers will win every time.

The ITU Standard

When setting up a system ignore anybody who says that there is no standard for recording.There absolutely is a standard and it's shown below.

I have installed hundreds of surround sound systems and tried all possible configurations. This is the one that I arrived at through trial and error and it's the one used by every recording studio that I've visited. There actually is a standard for recording and you should position your speakers the same way for accurate playback.

You should seek speakers that sound the same when used in the positions where they will actually be used, not speakers that are the same. Enjoy. APJ