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Cables & Accessories

When you have successfully assembled an audio system composed of the best components you can afford (or justify) you can fine-tune that system with accessories to achieve all the performance capability from your investment. You'll want to be extremely careful when venturing into this potential minefield. Some accessories have a small positive effect on sound quality and can be added to an otherwise completed system to squeeze out that last bit of performance. Some accessories are simply frauds that offer only psychological performance enhancements. Some accessories actually have a negative impact on sound quality.

Accessories are among the most heavily promoted products in audio. They are also among the most profitable for manufacturers and retailers. They seldom perform as well as advertised and often make the sound worse, or simply different, rather than better. With religious fervor people keep buying accessories and hoping they will somehow turn a sow's ear into a silk purse. Read my lips folks: it ain't gonna happen. If you get an audible improvement, and that's a big if, that

improvement will be subtle. If you have assembled a high-resolution system that subtle difference may be worth the effort and expense. If your basic system still needs improvement, accessories are a waste of money. Accessorizing a system to add compensatory colorations is a big mistake.

When choosing between a power line conditioner for use with your old amplifier or upgrading to a better amplifier, choose the better amplifier. In fact that's pretty much an axiom. A component upgrade will almost always provide more improvement than a "performance enhancing" accessory. (The exception is in the rare instance when the accessory solves an actual problem.)

If one of your components is flawed, replace it. Don't try to reduce or compensate for the flaw with an accessory. If you start stacking one coloration on top of another you'll become hopelessly mired in a maze of aberrations from which you'll probably never emerge.

"Some accessories actually have a negative impact on sound quality."

Don't take anybody's word for it. Make the salesman prove that his magic moon puck improves the sound of his demonstration system and make sure he'll take it back if it fails to work in your home. Compare the sound of your own audio system with and without the accessory under consideration. The information in this **Journal** will show you how to make valid comparisons and help you make informed decisions.

Gauging the effect of accessories

There is a simple method for evaluating the efficacy of an accessory and I'll show you how to do it. You should never buy anything without proving to yourself that it has a positive influence on sound quality and if everybody did this, many accessory manufacturers would vanish from the industry (as they should). You can weed out many frauds in the dealer's showroom by listening to his demonstration system but the final test should be in your system in your home.

Measuring the effect of accessories

I can hear things I can't measure—yet. That doesn't mean these characteristics can't be measured, it just means that I don't know how. Many accessories that aren't in the signal path fall into this classification and so do most cables, which *are* in the signal path. Don't ignore things we don't yet fully understand. If you can hear it, it matters!

After an objectively accurate and subjectively pleasing system has been assembled it can be fine-tuned with accessories. When discussing components that are not in the signal path—and even some that are—measurements can be inadequate tools for evaluation. When something's going on that is clearly audible but is not indicated by our measurements, that doesn't mean measurements are worthless. It simply means that in certain areas we can't rely as much on measured performance.

If a component is hopelessly wrong, measurements will probably alert us but if a large group of components all have acceptable measured performance, then only listening will allow us to choose. That's probably because less work has been done to identify characteristics that some refuse to believe exist. If you measure the DC resistance, capacitive reactance and inductive reactance of most cables, for instance, you'll find little difference. If you compare them in a bypass test you'll discover amazing sonic variations. I'll show you how.

If you believe that the three parameters of DC resistance, inductive reactance and capacitive reactance completely define audio cables, you're wrong and I'll show you how to prove that to yourself. If you believe that a power line filter that can be proven to remove hash from an AC line will necessarily improve the sound of an audio system, you're wrong and I'll try to show you how to prove that to yourself, too.

If you believe that vibration can't affect a solid-state component with virtually no internal wiring… Well we're getting ahead of ourselves here. This **Journal** won't answer all your questions about accessories but it will get you started on your own listening experiments. When you listen and compare you'll save lots of money that you might have spent on worthless accessories, but don't throw the baby out with the bath water. Some accessories really work and some are realistically priced. And some, like high-quality cables, are indispensable. APJ

Subjective -vs- Objective Evaluation

by Richard Hardesty

*In this **Journal** we're going to examine some of the most important categories of accessories: Cables, acoustic room treatment materials, power line conditioners and vibration control components. Before we begin, I think it's important to lay groundwork by addressing one of the most pressing issues of our day—should we judge components and systems objectively or subjectively? Making that determination requires a discussion of what we're trying to achieve because before they can be achieved, goals must be established. I advocate assembling an accurate audio system and I'm going to tell you why in this article.*

The Effect of Accessories Often Eludes Objective Measurement

Accessories that are outside the signal path may create clearly audible changes that are difficult to measure using conventional tests. Cables are in the signal path and have a tremendous sonic impact, yet they may look almost identical on the test bench when comparing inductance, capacitance and resistance. Measured performance can prove only the *potential* for accurate signal reproduction. Whether that potential has been realized often eludes measurements.

Does Measured Performance Matter?

Those who believe that measured performance is important are often called objectivists. Those who feel that all that matters is how a component or system sounds, are sometimes called subjectivists. You have probably read many articles about the "war" between subjectivists and objectivists. This supposed battle is actually a red herring that obscures a real question which is far more complex and much more important. That question involves the decision about where art stops and science begins.

Playing recorded music at home provides us with the illusion that music is being performed in our living rooms. It's an illusion because real musicians and their instruments aren't present. Recordings and electronic components take the place of musicians but we still want to respond emotionally to their art. We record the actual performance and play it back in our living rooms and hope to respond as if the musicians were there with us. We utilize scientifically designed components to reproduce an artistic expression that has been recorded. Here's where things sometimes get blurry.

The purpose of an audio system is to electromechanically convey an artistic message and the real question about the best way to do this boils down to the following: Whether the system should be a neutral window to the art as recorded or an additional expression of art with colorations that make some recorded music sound "real" to some listeners, some of the time. Both positions can be argued but I'm convinced that the former is the better route to long-term satisfaction and here's why.

Over the course of decades, I have carefully observed as hundreds of people labored to assemble satisfying music reproduction systems in their homes. My observations led me to an inescapable conclusion. More people will be happier more often if they assemble an audio system that is as transparent to the recording as possible and buy recordings that capture the art of music. Recordings should capture the art and those recordings should be reproduced without alteration. Art should prevail until the recording is completed and science should take over when playback begins. What does that mean exactly?

Is a Playback System an Artistic Expression?

Music is unquestionably an art form. Those who compose and play music are artists. We want recorded music to allow us to respond appropriately to their art, in our homes. To make that possible we need to capture the artist's work and reproduce it at home using equipment that is scientifically designed to successfully accomplish these tasks, not equipment that makes an additional artistic contribution of its own. If the equipment makes an additional artistic contribution that enhances some kinds of music, that contribution will assuredly be detrimental to other types of music. Thumping car stereo bass may enhance the impact of rap but it will destroy the sound of a cello.

You wouldn't buy a painting and then alter the colors to suit the décor in your room because the colors are part of the original expression. Why buy an audio system that imprints everything played through it with its own unique sound? That's just like painting a painting.

We utilize science to capture and reproduce art that actually occurred elsewhere. Those who produce and record musical performances are technicians who try to bridge the gap between art (the creation of music) and science (the reproduction of recorded music). Those who make audio components should be primarily engineers, not artists, and that's the basis of the high fidelity approach to home music reproduction.

Two Sides to a Silly Argument

To accurately reproduce the signal, the output from an audio component must closely resemble the input to that component. Although it may be unfashionable to admit, the accuracy of a component can be objectively measured.

Measurements compare the output from a device under test to the input, and deviations are called distortion. Objectivists argue that audio components with substantial deviations from accurate performance cannot reproduce the signal as it was recorded. This is absolutely true but avoids the question of our ultimate goal.

Do we want the audio system to accurately reproduce the signal retrieved from the recording or do we want the system to simply entertain? Do we want to be entertained by all types of music or just the one or two genres we listen to the most?

"Measurements compare the output from a device under test to the input, and deviations are called distortion."

Subjectivists argue that it's all an illusion anyway and a convincing illusion is all that matters. This seems to be a good argument at first but my experience convinces me that it won't hold up in the long run. If a distortion or coloration (or whatever you want to call a deviation from accurate response) were complimentary to all kinds of music it would be hard to argue against it. In fact, if a distortion were complimentary to all music most manufacturers would build it in to their products to make everything sound better. Unfortunately, no such distortion has ever been found. A coloration (distortion) that is very entertaining when listening to one musical genre may be entirely offensive when listening to another.

So which do we want, good measured performance or pleasing sound? Who says we have to choose between the two? Does one necessarily exclude the other?

The answer, of course, is that we should demand both so we can enjoy any kind of music and explore genres with which we are unfamiliar.

"Recordings should capture the art and playback systems should accurately reproduce recordings."

Audio components and systems should deliver impeccable measured performance and good sound. If they don't, perhaps someone is trying to fool you.

Flimflam Men

The so-called subjectivists have taken over a good part of the audio industry today and there are several economic reasons for this situation. When nothing measurable matters, anybody can be a "designer" and everybody can be a "reviewer." The number of manufacturers continues to proliferate. More manufacturers, no matter how unqualified their designers, can buy more advertising in the magazines and provide more products for unknowledgeable and underpaid reviewers to rave about. Dealers don't have to work to demonstrate what's best; they can simply let customers choose a sound they find pleasing, even if that thrill will quickly fade. Dissatisfied customers will probably come back two or three times—resulting in additional sales for the dealer—before they give up.

Don't be fooled! If objective measurements indicate poor performance, that's what you're going to get even if you are momentarily fooled by one or two recordings carefully chosen to deliver pleasing sound. If measurements indicate flawless performance you are only guaranteed the *potential* for good sound. Only listening will determine if that potential has been fulfilled.

So am I recommending that you become a subjectivist or an objectivist? Neither! I'm recommending that you become a critical thinker as you choose components that can bring emotional satisfaction from music into your home.

Choose products by listening but evaluate only those products that are objectively accurate. Why bother listening to components that can't possibly provide an accurate reproduction of recorded music? You may be momentarily fooled but eventually you'll learn to hear those flaws and be disappointed. Don't buy first and learn later.

Choose products that make you smile when listening to music through them but make sure that all types of music make you smile. Why buy something you hate the sound of simply because it measures well? That would be stupid. If a component or system is objectively accurate and it sounds good you're more likely to be happy with it for many years.

Introduction to Cables

by Richard Hardesty

Cables are an important audio component and an audio system can't work without them. You need power cables to get the AC power from the wall sockets to the components. Interconnect cables are required to get the signal from one component to another. Speaker cables are necessary in order to get the output signal from the amplifier to the speakers. Each of these cables has a unique and specific job to perform and we'll describe those jobs next. First let's consider cables as a component category.

There are two ways to look at cables: As important components or as tuning accessories. They can be selected to be as transparent as possible, like active omponents; or colored cables can be used to tune the system like accessories. Audio cable companies can be divided into two primary camps: Those who sell products with significant colorations designed to synergistically compliment the flaws of active components while painting a picture (artist's rendition) of a live musical performance; and those who sell products which are as transparent as possible allowing the accurate reproduction of the recording. I advocate the second approach. I know that cables are an important component in a high fidelity audio system but some still think the entire field of high-end cables is a sham.

Good cables are an indispensable part of an audio system but the high-end cable industry is filled with flimflam men and many of the products sold in this segment cost too much and perform poorly. You need to be very careful when choosing cables or you'll get royally ripped off. I'm going to tell you how to choose wisely but first let's look at the evidence that shows that cables really do make a difference.

Do Cables Really Sound Different?

Cables have been very controversial because they're often expensive and seldom good. People with limited knowledge would have you believe that the only factors that matter are resistance (R), capacitance (C) and inductance (L). Based on these steady-state parameters they can "prove" that there are no significant differences in cables because, if you limit your measurements to resistance, capacitance and inductance, all cables look pretty much the same. Empirical evidence gathered by listening shows that cables have a major impact on sound and after years of arguing with the self-proclaimed “experts” who say all cables sound alike, we critical listeners got some support from Professor Malcolm Hawksford of the University of York (United Kingdom). He could hear that cables make a difference and he set out to find scientific reasons for the differences he heard.

In the early 1800s Faraday arrived at the concepts that described the conduction of electricity. Later in that century, Maxwell developed the equations that quantified these concepts and showed the relationship between electric and electromagnetic fields. In the 1980s Hawksford used Maxwell’s equations to describe audio cables as transmission lines and showed that many factors besides inductance, capacitance and resistance could influence sound. Music is transient in nature and steady-state tests are inadequate for measuring subtle interactions that are clearly audible. Finally we had a “slide rule jockey” who could provide scientific reasons for the differences we had been hearing all along.

If you want to read Hawksford’s work go here:
www.essex.ac.uk/ese/research/audio_lab/malcolms_publications.html
Or here:
www.stereophile.com/reference/1095cable/

Hawksford offered scientific reasons for phenomena we had already discovered empirically. Conductor material makes a difference. The exterior finish of the conductor material makes a difference. Dielectric material makes a difference. Geometry makes a difference. Connectors make a difference. How the connectors are attached to the cable makes a difference. There are sonic artifacts associated with stranded conductors, silver-plated copper conductors, and conductors that are too large in diameter.

“Empirical evidence gathered by listening shows that cables have a major impact on sound.”

Regardless of the technical reasons, the sonic effects of a cable are easy to hear. You simply set up a bypass test and compare the sound of an audio system without the cable under test to the sound of the same system with that cable inserted into the signal path. A bypass test allows you to hear exactly what effects the cable has on the sound of the system. Ideally, if the cable is truly transparent, it should have no effect at all. Realistically it should have the smallest sonic impact possible.

Big lies are frequently based on a kernel of truth and synergy is a minefield you must traverse cautiously. Yes, all the components in an audio system should blend synergistically. Cable colorations can be used to fine-tune an audio system but a truly accurate system must have truly transparent cables. If you start tuning with colored cables how will you ever choose transparent components?

The Purpose of Cables

I have placed cables into a separate class of components. Within that class are unique and specific jobs that can be further separated and examined. Power cables perform a different function than interconnect cables, which perform a different function than speaker cables. Let’s examine the facts about the purpose of each type of cable.

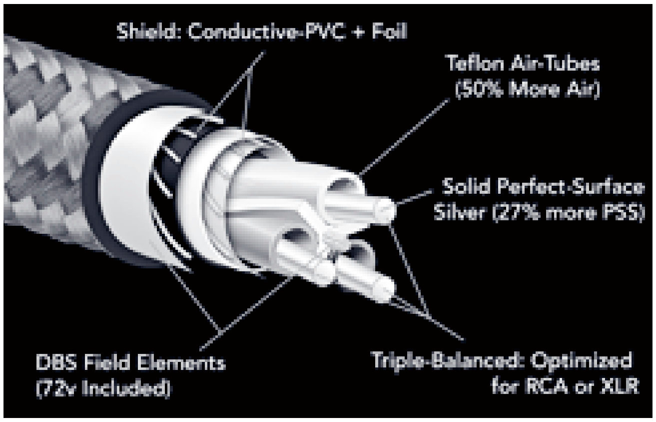
Power Cables

The purpose of a power cable is to transfer AC power from the wall socket to an audio component. The source voltage is high at about 115 volts and high current is available from a (typically) 20-amp circuit breaker. Component power supplies are (typically) designed to operate on alternating current at 50Hz or 60Hz and anything on the line at other frequencies is noise.

Power cables can be further differentiated by the current demands of the component the cable feeds. Amplifiers need lots of power from the wall socket. Preamps, not so much. Some source components even less.

Interconnect cables

The purpose of an interconnect cable is to transfer a phono level or line level audio signal from one component to anoth-



er. The source voltage is low at (typically) <1 volt. The load impedance is high at (typically) 50k ohms or more. Current flow is negligible. In my opinion, which is based on listening experience, bandwidth is critical and phase linearity is important. I believe bandwidth should be high and phase shift should be low. This precludes the use of cables with integrated filter networks.

Speaker Cables

The purpose of speaker cables is to transfer speaker-level audio signals from the amplifier to the speakers. The potential source voltage is medium at (occasionally) >20 volts. The load impedance is low at (typically) 4-8 ohms. Current flow can be fairly high but usually for only short periods of time. Cable impedance should be low so it doesn't limit amplifier damping. In my opinion, bandwidth is less critical (than with interconnect cables) but should exceed the bandwidth of the speakers—I think cable bandwidth should be at least double or triple speaker bandwidth. Phase shift should be negligible below 30kHz. Inductance should be low if your amplifier can remain stable with a low inductance cable (some can't).

My Preferences

I avoid cables with (touching) stranded conductors, cables that use conductors that are too large in cross section (usually 20ga or larger), cables that use silver-plated copper conductors and, of course, cables with integrated filter networks. I try to evaluate cables using bypass testing to hear exactly what effect a cable has on the sound of an audio system. If the system sounds different with the cable in the signal path the cable has distorted (colored) the signal. Less is better.

I've listened to a very large number of audio cables and made some personal observations. Among the cable brands that make the biggest changes in sound (to my ear) are Cardus, MIT, and Transparent. Among the brands that alter the signal the least are Alpha Core, Audioquest and Kimber. All cables degrade the signal to some degree. Less degradation is better. That doesn't mean you can't assemble a good-sounding system using colored cables. You may eventually arrive, by trial and error, at a synergistic blend of colorations that is pleasing. Or you can carefully select the most neutral and transparent components and cables by bypass testing. The choice is yours but I recommend the latter course. APJ

Interview with... Bill Low

by Richard Hardesty

I've known Bill Low for about 30 years and we've had various business dealings during that period. He was a fellow retail merchant, a manufacturer's representative for products I sold, a customer for my modification services and a supplier of audio cables and accessories.

In our early days of selling high-end audio equipment, we discovered that the cables connecting one component to another made a significant difference in the quality of sound we heard. We began to share our experiences and discuss the wire designs that sounded good and those that didn't.

With experimentation Bill found that better interconnect and speaker cables could make bigger audible improvements than many costly upgrades to amplifiers and speakers. And he recognized the opportunities afforded by this new category of audio components. He founded AudioQuest to explore and develop new and innovative wire, connectors and accessories.

He was not alone. At about the same time, Noel Lee started Monster Cable, which has become the biggest of today's wire companies, and the Polk speaker company was distributing a low-inductance wire called Cobra Cable. Since then many others have come and gone, and some have persevered and prospered. These audio cable companies can be divided into two primary camps: those who sell products with significant colorations designed to synergistically compliment the flaws of other components while painting a picture (an "artist's rendition") of a live musical performance; and those who sell products that are as transparent as possible, allowing for the accurate reproduction of the recording. AudioQuest is in the latter group and readers of **Audio Perfectionist Journal** know that I also advocate the high fidelity approach to home music reproduction where an ideal audio system is a transparent window to the recording.

High-end audio cables are controversial. Bill Low is an intelligent and insightful man with decades of experience listening to and designing audio cables. Let's hear his views and see what he has discovered along the way.

Bill, what prompted you to enter the audio cable business?

Gee, I think I have to be humble for a while after your generous introduction. My audio-activity beginnings were certainly humble, including the discovery that I could listen to music on a 10-inch record using a hand-held safety pin. Let's just put my age at that time in single digits, but both an appreciation for the power of music to make me feel good and an interest in controlling sound were already emerging.

As for entering the business, first I spent a decade building Heathkits and Dynakits for record-buying money. Then I started an appointment-only audio store in Oregon while in college. That was followed by complete failure as an independent rep, first in Northern California and then in

Southern California. In 1978 I went back to appointment-only retail with a specialty audio store in my apartment in Santa Monica.

I was progressive enough and informed enough to carry Linn Sondek, Rogers, Meridian, and Koss Electrostatic among others, including Denon, which at that time offered a high-end line of turntables and cartridges imported by American Audioport. Polk Audio had opened up the cable business in June 1976 by showing their imported Cobra Cable at CES in Chicago. I was aware of cable and wanted something special for my store—the best possible cable.

I had already developed this interest on my own, and would have done something on my own had it not been for an opportunity brought to me by another small appointment-only dealer—MWK (Middleton, White and Kemp) located farther south in Anaheim. They had a relationship with Dave Gore, designer of the hot Quatre DG250 amplifier.

Dave brought a cable idea to MWK which was based on a 1978 article about finely stranded litz cables written by Martin Colloms for the British “Hi-Fi News & Record Review.” From that starting point, Dave and MWK used a door handle, a drill and some 180-strand 15-awg litz wire to make an experimental twisted-pair speaker cable. Lo and behold, it blew away the huge welding cable that MWK had been using as their previous reference.

As I was a fellow traveler, in the process of converting from sales rep to retail businessman, and as one who shared the MWK philosophy, I was invited to participate in an order for custom-made speaker cable. We agreed on 435-strands per conductor twisted-pair litz construction—and so was born what I refer to as “the original recipe.”

It wasn’t until two years later, in 1980, after several other dealers and a Japanese distributor had started buying cable from me, that I decided to make cable not just for my retail customers but also to sell to other stores. That’s when I started AudioQuest, and that’s when the evolution of (at the time) LiveWire cables really began. By the end of that year, I had some seriously sophisticated cables, 43 dealers in Southern California, including your wonderful store, and one in Colorado—the still noble Listen Up. I’d put the LiveWire Green Litz of that day up against many of today’s “best” cables (sorry, I’m not always humble).

Tell us about what AudioQuest has become.

What AudioQuest has become is “more of the same and then some.” Today, AudioQuest is a fraction of the size of Monster Cable, but possibly bigger than all the other high-end cable companies combined. That implies a lot of growth along with some changes and, yes, the business is now dominated by video. I make a lot more models of cable these days, but the core concept hasn’t changed at all. That is, how a company—my company—earns a place for itself in the market.

I find it quite ironic that the philosophy considered back then to be as natural as the air we breathed—and I’m talking about MWK, Jonas Miller, Mission Bay Audio, Taylor House, John Garland Audio, Suffolk Audio and Havens & Hardesty, as well as at least a hundred less specialized stores around the U.S.—has turned into something so comparatively unusual and is now a distinct marketing handle for AudioQuest.

Stated simply here, that philosophy was that the best business, the most competitive position, would come from always trying to sell the best performing equipment. It was assumed that a dealer, a salesperson, should have an informed opinion about the performance of the equipment and that was intrinsic to the sales process. Duh! Every manufacturer always claims to make the best stuff, and every dealer always claims to sell the best stuff. What is said is meaningless. Actions are what make all the difference in the world.

There are certainly some excellent dealers out there today, at all levels of the market, who still breath this air. But there’s not as much of that air as there once was. Sometimes—due to loyalty, cozy relationships, or laziness—“high-end’ dealers have stopped paying attention to products they don’t sell. For most of the market, “sales training” (telling salespeople what to say)

has replaced asking the store staff to please listen to the product and, if they like it, sell it—hopefully lots of it.
What AudioQuest has become is a more finely packaged version of what it always was. These days we use a $119 stereo system to give many hundreds of new salespeople every year the opportunity to hear for themselves how simple, obvious and explainable are the differences between speaker cables. I've always tried to tell the truth: the audio-be-faithful truth, the what-I-understand truth, and the why-do-it-this-way truth. But I do recognize that even the truth has to be sold.

I advocate the High Fidelity Approach to home music reproduction and suggest that an ideal audio system be a transparent window to the recording. What is your philosophy on the purpose of audio cables?

I believe that a cable should be as close as possible to no cable at all. Cable can easily be compared against no-cable. Such comparisons are always disheartening, but cable can get close enough to neutral that I believe there is no excuse for any other objective. I cringe every time I hear a reference to cable being like a tone control that must be chosen to be compatible with other errors in a system.

I also cringe at the whole incorrect fixation on amplitude! It's so easy to measure, easy to understand and easy to hear small differences in amplitude—but this does not mean that amplitude inconsistencies, or the lack thereof, are what make a product likable or effective.

Certain combinations of very high capacitance or very high inductance cables can alter frequency response, though in a relatively linear and non-irritating fashion. However, for the most part, amplitude variation, either proportional to frequency or across the range, is irrelevant to understanding the obvious differences between cables.

You probably also do something between laugh and cry when you hear someone accuse a cable of having a "bright" or "hot" high-end, as if the high frequencies have increased in amplitude. The next best thing after cold fusion is a cable that can create energy! In reality, "bright" is due to forms of distortion that cause our computer, the brain, to misinterpret the audio information and then to present our consciousness with aural irritation. Other distortions, such as skin effect, corrupt the audio data to such a degree that the brain cannot interpret the information and therefore presents nothing to the consciousness. The dull top end from skin effect or a poor output transformer is not due to a loss of amplitude but to a loss of information.

Let's go back to some crucial terminology in your question. Of course I agree that an ideal system should be a clear window. But I disagree with the industry's general rephrasing that the purpose of a well-chosen audio system is to be a transparent window. All audio systems (this also includes the room) are so far from real or transparent that the test for success is not whether they sound real or how clear the window is, but whether they are effective emotional transportation, whether they serve the reason we listen to music.

Truth and transparency in a system are absolute values, like the North Star representing an absolutely necessary reference. However the absolute failure of an audio system to sound real doesn't make the system a failure. Have you noticed that the only time a system ever sounds real is when you're not in the same room? An audio system sounds more "real" with the benefit of a gross filter damping one's awareness of a system's misinformation. Our industry's fixation on more information misses the real culprit interfering with believability and with pleasure, and its name is added misinformation.

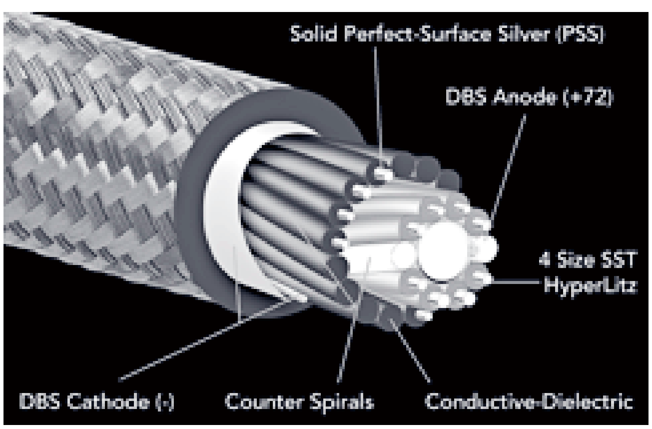
Wires differ in materials and geometry so let's talk about those factors. What have you learned about cable geometry?

One of the first things I stumbled into was the significance of the relationship between conductors in a speaker cable, in cables carrying a significant amount of energy and therefore having significant magnetic fields. I was fortunate that my very first cable experiences taught me that twisted-pair is much better than parallel. This is something the phone company and SMPTE and others had known for ages. So, twisted-pair was my starting point, and my first accidental progress beyond that was thanks to a friend of mine who manufactured subwoofers.

He wanted to be able to recommend something to his dealers as an alternative to the massive, expensive and unwieldy Fulton Gold speaker cable. He asked me to please make a four-conductor version of my twisted-pair cable.

When I received my first production run of this four-conductor cable, I compared it to a double run of the same conductors used in twisted-pairs. The significantly better performance from the four-conductor cable alerted me to the significance of geometry. The difference between parallel and twisted-pair can substantially (though not entirely) be credited to lower inductance in the twisted-pair. However, the capacitance and inductance differences are trivial between two twisted-pairs in parallel versus a single four-conductor cable. The relationship between magnetic fields and the comparative stability possible with four conductors explains most of why a four-conductor cable is so clearly superior.

I could bore you for quite a while with more details of cable geometry, so I'll skip to the other end of my travels. My top models use a counter-spiral arrangement with a circle of (positive) conductors spiraling in one direction, surrounded by a circle of (negative) conductors spiraling in the opposite direction. The relationship between these two tubes of conductors is fixed and non-changing. Stability of electrical values is crucial and generally much more important than the particulars of the electrical value.



However, while positive and negative have a proper constant relationship, the individual conductors of positive and negative are crossing rather than paralleling each other, significantly reducing distortion caused by interaction between the conductors. The contrast between this arrangement and a braided relationship between conductors is enormous. In a braided conductor or a braided cable, there is significant "magnetic disruption" (not my term; normal cable engineers call it that), as each strand or each conductor encounters a constantly changing magnetic environment.

What have you learned about dielectric materials?

You've thrown out a fairly sophisticated term in that question. Insulation is a necessary ingredient in cable construction. Unfortunately, in addition to some degree of insulating ability, insulation also has dielectric properties. The only perfect dielectric is a vacuum, which is nothing at all. So "dielectric" is a property, not a thing.

The various materials available with which to manufacture a cable all have different dielectric properties. They all interfere with the passage of the signal through an adjacent conductor in different ways and to different degrees. Analogies are only analogies, not absolute parallels, so I hope your most technical readers will forgive the liberties I take when I compare the magnetic field around a signal carrying wire to a coin, and when I refer to the insulation as if it were a bottle of shampoo.

There can't be electricity in a wire without a directly related field outside the wire, and vice versa. Those ferrite filters—that bulge one sees on the cable leading to a video monitor—are able to stop undesirable high frequency energy in the cable by stopping the magnetic field. The wire is uncut. There is no discontinuity in the impedance of the metal wire, and yet the wire's ability to carry a signal is severely changed by the obstacle put in the path of the magnetic field.

Well, the dielectric properties of a wire's insulation are not usually so severe, but they can be dramatic. In other applications, at much higher frequencies and over longer lengths, the absorption and loss of energy due to dielectric can be fatal. The specification of "dielectric constant" is most often used to describe a material's interference with signal amplitude.

However, in our world of fairly short cables carrying audio frequencies, dielectric constant is more deceiving than helpful. Amplitude loss due to dielectric is trivial to irrelevant. But a 10-octave audio signal is incredibly difficult to move around without interfering with its information content—information that is all about frequency and time relationships.

Dielectric specifics such as “propagation delay” get a little closer to the truth of our problem. Propagation delay is specified as a percent of the speed of light. A very well designed 75-ohm coax has a propagation speed just over 80% of the speed of light. But think about it. What difference would it make if it took a minute for the music to come through a cable? It would be like using an early SACD player that took a minute before the music started. As long as the music all arrived undistorted, so what if it came a little late?

What propagation delay does hint at is that dielectric slows down a signal. Much the way shampoo slows down a coin dropped in it, so does insulation slow down the magnetic field of an audio signal and, therefore, slows down the signal itself. Again, this effect alone would be no problem. Unfortunately, the slow-down, time delay or phase distortion is nonlinear. The degree of time delay is different for different frequencies and for different amplitudes, wreaking havoc on a 10-octave signal of ever-changing amplitude.

For generations, cable design engineers have sought to minimize dielectric-induced problems, whether loss of amplitude or distortion, by using better dielectric materials and less of them. “Better” for some applications is not better for others. Sometimes a material that causes less loss of amplitude causes more nonlinear phase errors. This trade-off explains why Teflon is so controversial as a dielectric material. Good for a circuit board, bad for a capacitor, good and bad as wire insulation. Ever heard terms like “cold and analytical” applied to some equipment? Take a look at the variety of dielectric materials used throughout the signal path and you will often find the explanation.

So, besides the choice of material, the other long-standing and well-understood priority is to minimize the use of solid materials and to maximize the use of air (or nitrogen gas or whatever). Foamed dielectrics, air tubes and several other designs all try to minimize the amount of dielectric material near a conductor.

Some people don’t understand battery-biased dielectrics but my system is full of batteries and has been for years. I was really pleased when you put batteries on your cables and understood their purpose immediately. Can you explain your DBS cables for the uninitiated?

Ah, a well-placed setup. Thanks! I’ve just been rambling on about how a dielectric interferes with the information content of an audio signal by introducing nonlinear phase errors. This corruption of the data stream causes our CPUs (brains) to either misinterpret some of the data or to not understand the data at all, in which case it can’t provide our consciousness with any “sound.”

Let’s go sideways for a moment. Skin effect-induced phase shift—which causes no phase error for the majority of the signal (near a conductor’s surface), but which does cause progressively more phase delay at higher frequencies and greater distances from the surface—eventually smears time information to the point that the brain can no longer recognize more delicate ambient information or even an instrument’s upper harmonics. It’s easy to mistakenly think that we can’t hear the missing treble because it must have been turned down, or that the amplitude has been turned down, but it’s not the amplitude that has changed. It’s the information content of the signal that has been corrupted—in effect turned down such that our computer can’t give us an aural picture.

And going sideways again, also for the purpose of laying a little groundwork: What is noise? Most often we would answer with descriptions such as “surface noise on an LP” or “tape hiss.” However, by another definition, this isn’t noise; it’s perfectly well understood information that we wish wasn’t there. Noise could more accurately be defined as unintelligible energy, lacking sufficient organization to be decodable, so we can’t actually “hear” it.

Back to our little problem of dielectric effects getting in the way of our music. Back to the dielectric involvement of a cable's insulation causing nonlinear phase errors. I know you have noticed, and I hope your readers have all noticed, that when they leave their electronics turned on all the time, the system sounds better. This effect is almost entirely due to decreasing the amount of nonlinear phase errors caused by circuit board materials, capacitors, dielectric inside of an FET or transistor, insulation on wires inside and outside of the hardware, and in the very big and distortion-causing capacitors inside of loudspeaker crossovers.

The good news is that while biasing dielectric material doesn't prevent the material from acting like shampoo, from slowing the signal, it does reduce the variation in time delay proportional to frequency and amplitude. And that is the problem we hear.

"Biasing a dielectric" means immersing the material in an electrostatic field. This causes the molecules of insulation to orient themselves relative to the field, much the way iron particles in a junior high science lab orient themselves relative to a magnetic field. When a material is electrostatically amorphous (unorganized) the material causes the most phase errors. The better organized the material, the more completely polarized it is, the fewer nonlinear phase errors. When a left-on audio system sounds better it's because of fewer nonlinear phase errors.

I assume some of those many batteries you are referring to are the whole bunch of batteries in your Vandersteen loudspeakers. Richard (Vandersteen) was putting significant voltage on the capacitors in his Model 5 crossovers before I ever made a cable with a battery pack. Richard harassing me about biasing dielectric material led to the AudioQuest Dielectric-Bias System (DBS)—putting batteries on cable.

Those batteries in your loudspeakers do a far more effective and complete job of biasing the dielectric material in your speakers' crossovers than can ever be accomplished from the outside, even if you played your speakers full blast 24 hours a day. So, too, the DBS system on AudioQuest cables brings cable performance to a level impossible when system components alone are putting a charge on the cable. DBS is much more than just making a new cable sound as good as one which has been in an always-on system for a long time.

Whatever voltage is applied, it takes about two weeks for dielectric to adapt to a charged state. Whether it's the partial effect of leaving equipment turned on or the more complete effect provided by Vandersteen's or AudioQuest's batteries, it takes about two weeks for the dielectric to stabilize, and about two weeks to completely lose the effect by turning off the equipment or disconnecting the batteries. One can't judge the effect of an always-on component or a DBS battery pack by plugging and unplugging. One must have two components or two cables to compare.

So, what is it that most strongly describes the improvement due to a better-polarized dielectric? I really hate to suggest what others will hear or how they will describe what they hear, or how the difference will have meaning to them. But, that said, far and away the comments I hear most often from people who have compared the same cable with and without DBS have to do with praise for the quiet background, for how the sound seems to come out of blackness, often seeming louder thanks to the greater dynamic contrast.

If DBS does that, how can it do that? Remember, now so long ago, when I asked, "what is noise?" When one listens to a non-DBS cable, one certainly doesn't hear any noise, but when one listens to the same cable with DBS, one gains the same improvement in clarity and emotional stimulation as if one had removed noise from the signal.

With hindsight, DBS seems so very simple. It's been known for ages that dielectric effects cause problems. It's been known for ages that partially biasing dielectric materials by leaving the equipment on improves performance. So why not put a full-strength bias on the dielectric material all the time?

Again, with hindsight it seems so simple; hindsight is often like that. If a cable were simply built with the necessary field elements in addition to its signal path conductors, then those elements could be connected to a DC source. This would immerse all of a cable's dielectric material in a sufficiently strong electrostatic field to seriously improve audio performance—both analog audio and digital audio, which is also supersensitive to time-related distortions.

Fortunately, just because something appears to be simple doesn't mean it isn't a new idea and can't be patented! I often compare DBS to a high-speed quad chairlift. Chairlifts have been around for a long time. So have gondolas, with their system of attaching and detaching from the cable so that skiers can get on and off. It seems that a modern high-speed chairlift is not much more than putting the advantages of both systems together, but doing so has increased access to the slopes and revolutionized the economics of the ski industry. Like DBS, the high-speed lift probably is also a patented innovation.

Here's a bit more about the efficacy of DBS. Not surprisingly, analog video and digital video (DVI-D, HDMI) are not sensitive to this type of nonlinear phase error and do not benefit from DBS. Packet information, or information used to assemble a picture that changes a maximum of 60 times a second, just doesn't have the same problems as our humble but much-loved low frequency audio signal.

An AC power cable can be improved by DBS, but only for about the first two days before the effects of 115V or 230V swamps out the DBS effect.

Tell us about the wire in the wires. What materials do you use for conductors?

In the beginning, I specified "high-purity" copper for my cables because…well, why not? It sounded like a good thing. I then learned that there was such a thing as OFC—also called OFHC, which is Oxygen-Free High-Conductivity copper. It turned out that OFHC did sound significantly different. And it turned out that there is as much difference between OFHC from different suppliers as there is between normal electrical grade copper and OFHC.

Key to understanding such disparity is recognizing the "high-conductivity" part of OFHC. It's not OFLD for Low-Distortion! The big, bad normal world manufactures OFHC because it is slightly more conductive, and the test for getting to charge a premium for OFHC is based on the material's conductivity, not its distortion profile.

In our world this difference in conductivity is absolutely irrelevant, but the differences in audio performance are important. I learned that I must listen to the OFHC from different suppliers and choose the material with the best audio performance. I don't design the metals or the processes by which they are made. I am simply a professional consumer in a category most people don't get to play in.

But there's even more beyond OFHC! In 1985 Hitachi introduced LC-OFC, a long-grain copper called "Linear Crystal." Van den Hul Cable introduced a similar concept with their Mono Crystal cables at about the same time. I started using the Hitachi copper in my best interconnect and speaker cables. Hitachi's own cables got almost nothing but bad reviews, despite being some of the best cables on the market. Prejudice can be a very bad thing! Using Hitachi's superior metal in (I believe) vastly superior AudioQuest designs earned some rave reviews for unprecedented performance.

In 1987 I started using copper cast according to Professor Ono's Continuous Casting process, and then drawn through a process like what Hitachi had invented in order to incur as little damage as possible to the cast copper. The result was a material with grains of copper averaging 200 feet in length instead of the huge number of grains per inch in even the best OFHC coppers.

Then a couple of years later, because AudioQuest was the largest customer for OCC copper outside of Japan, a team flew over from Nippon Mining to introduce me to their stress-free 6N copper. It was better and I bought it. Shortly afterwards I also started using solid silver in my top cables, after finally hearing a silver conductor refined enough to provide the transparency advantages of silver without the bright irritation common to less perfect solid silver.

While I was pleased to use first Nippon Mining's and later Dowa's super refined copper in my next-to-top cables, another potential copper customer was equally impressed with the performance of Nippon Mining's copper, but couldn't stomach the price. That dilemma inspired him to spend years figuring out how to get the same or better performance at a lower price.

The delightful fruit of all that thinking and work is the Perfect-Surface manufacturing process now responsible for three of the four grades of metal I use in audio cables. Am I ever glad he brought an example of this effective process to me, scaring me with a material not much more expensive than OCC, which I liked as well as the best silver I had ever heard! Thankfully, in addition to the PSC-grade copper he first brought me, he also accepted my challenge to try this process on higher purity copper and on solid silver, which is much more difficult to work with. And so were born PSC+ copper and PSS silver.

Hawksford showed that the surface finish of conductors was important. Most people think electricity runs through a conductor so this won't seem intuitive. What have your experiences shown and how are AudioQuest conductors finished?

Well, electricity does run throughout a conductor, though with some skin effect-related variation in impedance and inductance-induced phase shift. I can't resist mentioning here that skin effect is what keeps you safe from lightning in a plane or car. Skin effect is the result of a delay in the change of a magnetic field following a change in current. When lightning strikes a plane, it is forced to go around the magnetic field "left over" from the immediately previous state. In an audio cable, higher frequencies represent a faster change, and so they're forced to go around the blockage caused by the previously existing magnetic state, which forces the new current toward the skin of the conductor.

So, skin effect is part of the explanation for why the surface of a conductor is so very important. Due to skin effect the surface of a conductor is the only part of the conductor carrying all frequencies equally, with no variation in impedance or phase. Current density is 100% at the surface for all frequencies. The surface of a conductor is also immediately adjacent to the strongest part of the external magnetic field. Remember that ever-so-important field which allowed insulation material to wreak such havoc with the signal inside of the conductor?

As Hawksford notes, the surface of a conductor is like a rail guide. The electrical current inside a conductor and the magnetic field outside are dependent on the quality (lack of imperfections) in this "rail." Imperfections at the conductor's surface can be thought of as causing turbulence in much the same way that imperfections on a plane's wing cause turbulence, except that in this case the turbulence extends from the very center of the conductor to significantly outside the conductor.

I remember once getting suckered into buying an overpriced paint protection system for my car, a 1994 BMW 540i at the time. The system included thin, clear plastic film over the front of the hood, and also smaller pieces for the side-view mirrors. As I headed out of Denver on the freeway, I was subjected to an astonishingly loud howling sound. I pulled over, pulled the thin film off the side-view mirrors and, for the rest of the way home, enjoyed peace and quiet. It doesn't take much surface disruption to cause a problem.

Are connectors important? What connectors do you use and how are they attached to the cables?

I love your leading questions. Many years ago John Atkinson commented in *Stereophile* that, much to his disbelief, a new XLR from AudioQuest really did sound better than the presumably already near-perfect Neutriks.

Yes, connectors and how they are attached are important. As with the cable itself, metal quality matters. I use OCC copper in my better connectors. Ironically, one of its occasional disadvantages—its hardness—actually makes it more machineable and more appropriate for plugs than normal solid copper.

Plating also matters, especially the quality of application to the metal. Bare copper oxidizes, and because copper oxides are semiconductors their conductivity varies in the presence of a charge, creating a complicated and dynamic distortion mechanism. Silver and tin oxides are conductive and self-healing. Nickel is an effective oxide barrier, and as long as it's plated extremely well it's not as bad as its reputation. Gold is only as good a conductor as nickel and aluminum, about 70% as conductive as copper or silver. But gold is noble and good for protecting materials underneath from oxidizing. However, other than on AudioQuest's direct gold-plated speaker connections, gold is almost always plated over nickel, and is strictly eye candy. Rhodium is very hard so very little is needed to provide an effective barrier layer, but I find even the thinnest rhodium layer causes an irritating grainy distortion that I consider unacceptable.

So, I use a thick silver plating directly over the copper on all my better plugs and connectors. A thin silver plate causes some of the same obnoxious tweeter-in-your-face sound as silver-plated audio conductors, but thick silver plate on connectors becomes a superior parallel conducting path.

I also offer my speaker connectors in gold, also directly plated, because sometimes one doesn't want to fight every battle and some people are scared of silver oxides. But when I use gold, I use a very thin layer as the-less-the-better applies. My gold-plated speaker connectors are almost as good as clean unplated copper, while the silver-plated connectors are a little better than clean bare copper.

As for how to attach a connector: directly, if possible. All AudioQuest speaker cable connectors are attached with pressure, with a good crimp or multiple setscrews. In both cases a cold weld, gastight connection is made between cable and connector. This is better than any connection requiring solder.

Lower priced AudioQuest interconnect cables use very carefully chosen low-silver content solder (more than 2% silver prevents a good connection). Soldering is efficient. All the audio interconnects over $100/m have their connectors welded on. The process of resistance-welding sends 8,000 amps of low voltage current through the connection between cable and plug, just for a few millionths of a second. The resistance of the cable and plug to so much current, at an inefficient low voltage, heats the metals to the point that their molecules commingle and form an alloy. A welded connection serves as an excellent solder-bypass when I'm evaluating solders.

There is little current flow with interconnect cables and lots of current flow with speaker cables. Power cords conduct only a single frequency. Shouldn't the design of these cables differ?

Yes they should! Many of the mechanisms that cause distortion in cables are present in all these applications, but the hierarchy varies and so, therefore, should the designs vary as well as the allocation of funds for materials and for more expensive constructions.

An audio interconnect essentially carries information rather than power. Most of the detrimental effects of an interconnect cable can be heard in the first inch of cable. The cumulative effect of these distortion mechanisms over length is not so much worse. On the other hand, much of what a speaker cable does wrong, with regard to both inductance and magnetic field interaction problems, continually accumulates. A longer speaker cable sounds more and more out of focus, no matter how good its design and/or materials.

Here's an interesting irony. It's actually easier to compare short speaker cables, easier to be aware of a cable's specific failings. As a longer cable suffers reduced performance because length is like a camera lens going increasingly out of focus, a shorter cable is more like a lens in focus. Distortion in a lens (character flaws, any funny mirror effects) is only more obvious when a lens is in focus or when a cable is shorter.

So, for speaker cable, use the lowest distortion cable you can find in a length no longer than you need to get the job done. On the other hand, as interconnect doesn't suffer nearly so much cumulative damage, don't hesitate to use a longer interconnect when you need to, such as to facilitate the use of shorter speaker cables.

Of course, if the interconnect is high capacitance and it's being driven by a passive preamp, or a rare high output impedance tube preamp, then the driving circuit might misbehave badly, but short of going over this cliff, interconnect capacitance is a minor concern.

AC power cables are a subset of speaker cables. Some of my tricks to reduce distortion in a full-range speaker cable are unnecessary in a power cable. SST, where I use multiple size conductors to reduce awareness of specific conductor size signatures, is irrelevant. But conductor material quality and conductor geometry are exactly as important in a speaker cable.

The detrimental effects in an AC cable of conventional stranding, versus concentric stranding, versus solid conductors, are not surprising as these are power related distortion mechanisms. The benefit of the counter-spiral geometry used in my best speaker cables is proportionally even more important in an AC cable—it's all about controlling and stabilizing the interaction of large magnetic fields.

Are special purpose cables, like those designed to carry digital signals and video signals, significantly different from those designed to carry analog audio signals?

It's the same story again of overlapping sets. As I mentioned before, video signals and some types of digital signals are not sensitive to some of the distortion mechanisms that damage an audio signal.

For example, in a digital audio cable, S/P-DIF or AES/EBU, versus an analog audio cable, dielectric involvement is equally serious. Using better materials and employing DBS are equally effective. However, because digital audio cables are very sensitive to constant characteristic impedance, I must sacrifice the dielectric advantages of air tubes in favor of materials that can guarantee a fixed stable relationship between positive and negative conductors.

Silver-plated audio conductors cause that classic and much-despised tweeter-in-your-face effect due to discontinuity of materials. But for digital audio, and for video, high quality silver-plated copper clearly outperforms even the very finest coppers that I use in my audio cables.

The signal-carrying conductors in a DVI or HDMI cable are essentially a CAT5, consisting of four twisted pairs. But even here, the same cable basics apply: conducting material (including surface finish), solid conductors, dielectric quality and geometry (efficacy of design and stability). You just can't get away from these ever present basics!

What's going to happen in the future? Will we continue to make evolutionary improvements or do you see some revolutionary changes coming?

The evolutionary model is perfect. As "we" are currently discovering, not only are there fewer genes than once supposed, but most of the building blocks for genetic evolution appear to have been around for at least half a billion years. It's the "expression" of these genes that allows for such incredible diversity.

There are very few ingredients that can be manipulated to affect cable performance. The particulars of those ingredients will most likely evolve. Incrementally better materials will become available. And, as with the evolutionary model, I hope I'll continue to encounter the successful result of unintended experiments. Much of what I have learned has come from noticing a performance change when none was anticipated and then, as methodically as possible, working to turn that new awareness into a predictable means and method for minimizing a distortion mechanism. If I knew what was next, I would already have done it! Based on my past experience, bits and pieces of what I can't see now will become visible over time. I look forward to incremental progress.

Thank you, Bill, for a very informative interview! APJ

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Power Line Conditioners

by Richard Hardesty

Manufacturers and dealers have heavily promoted power line conditioners. Consumers have spent lots of money on power line conditioners and cords trying to fix problems that either don't exist at all or, in the rare cases where they do occur, are of minor sonic significance. That money would nearly always be better spent on improved components rather than attempting to deliver improved power to inferior components. Many power line conditioners have no effect on what you hear and some actually make sound worse.

These are bold statements so let me qualify them a little. My experience is primarily in Southern California. We have pretty good power here, delivered by mostly underground power lines, and we almost never have lightning storms. In the 25-plus years that I operated a repair facility within my retail store I never saw an instance of lightning damage to an audio component and I never had a customer complain of a component shutting down due to low voltage. Of course, you may not live in Southern California. If you live in an area where lightning storms and/or brownouts are frequent occurrences you may need protection from these events. If not, you need to be very wary of the claims made by the makers and sellers of power line accessories.

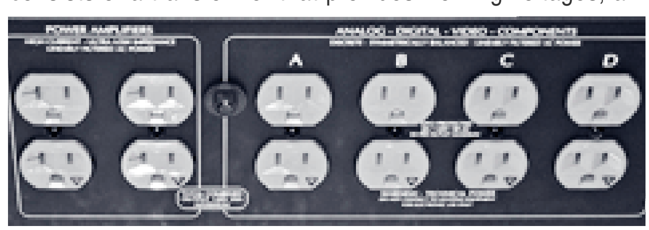
The makers of power line conditioners usually claim two benefits in order to justify their use: (1) prevention of catastrophe due to over- or under-voltage conditions and (2) better sound presumably brought about by removing noise from the AC power line. Both claims should be carefully scrutinized.

Catastrophe is discussed above. Noise filtering is slightly more difficult to understand.

Measurable noise may exist on your power line but some filters that claim to remove it may limit the current needed by your components and/or may negatively impact the sound you hear rather than improve it. Current-limiting compresses dynamics and squashes musical expression. Applying unequal filtering to parts of the noise spectrum may sound like re-voicing an audio system, which is usually not desirable.

Every Power Supply is a Filter

When AC power from the wall socket enters an audio component the first thing it encounters is the power supply in that component. A linear (or conventional) power supply usually consists of a transformer that provides working voltages, a



rectifier that converts AC to DC, and capacitors that filter ripples from the DC. A switching power supply usually has the rectifier first, followed by a DC-to-DC converter and filters to remove the switching noise. If a switching supply has a transformer it may come at the end of the chain instead of the beginning. The power supply is a filter of sorts, and the working voltages from which the component operates are isolated from the incoming AC. Can some noise riding on the

AC get through the transformer or pulse-width-modulator and contaminate the component power supply? Probably. Is this a big deal? Probably not.

"High-end audio is all about subtle improvements and a power line conditioner may slightly improve and refine the sound of your system."

The Audio Signal Modulates the Power Supply

A power supply is a filter but it may not be a perfect one. The audio signal modulates the power supply and the result appears at the output of the component. Some garbage (noise) that was riding on the AC power line may sneak through and contaminate the output signal from the audio component. Removing this contamination without causing other damage to the signal might subtly improve sound.

High-end audio is all about subtle improvements and a power line conditioner may slightly improve and refine the sound of your system. A power line conditioner won't make a poor component sound good and it won't eliminate the need to purchase better speakers.

Pros and Cons

I've presented many of the reasons for avoiding most power line conditioners. Now I think it's only fair to let an expert in the field offer an opposing view on this controversial subject. Following is an interview with Garth Powell of Furman Sound and he will present the reasons in support of the use of power line conditioners in a high-end audio system. APJ

Interview with... Garth Powell

by Richard Hardesty

Garth Powell is a technically astute designer and an accomplished classical and jazz drummer/percussionist who has studied with a host of master musicians and still performs professionally in the US, Canada and Europe. He was employed as a technician-design engineer with AT&T, Hewlett Packard, Analog Audio, Holmes Powell and others. Garth is currently Chief Design Engineer for the line of home theater and audiophile power management products made by Furman Sound, where he has worked for nearly 12 years.

This man knows about music and has closely examined the quality of the AC power that operates our audio systems and its effect on the sound we hear. Therefore, he's ideally qualified to shed light on the often confusing subject of power line conditioners.

Garth, there are several questions that readers might ask about power line conditioners: Is there really lots of noise on AC power lines? Can line noise degrade the sound of our audio systems? Do power line conditioners remove line noise without causing other negative effects? Do audio systems actually sound better when powered by AC lines that have been filtered by power line conditioners? Let's consider these questions separately and in greater depth.

The AC power we use to operate our audio components should be about 120 volts RMS at exactly 60Hz in the United States. Engineers refer to any unwanted signal as noise so any symmetrical or asymmetrical addition to that 120-volt/60Hz signal is noise. Can sensitive instruments detect noise on some AC power lines?

Yes, noise can be detected with an oscilloscope via voltage readings, and with current sensors when measuring an audio system under load. There are more sophisticated means of dissecting the noise as well.

What is the source of this noise?

Radio Frequencies coupling into the AC line, back EMF from appliances and motors, and severe wide-bandwidth harmonic noise from switching power supplies. Current noise on the ground pin (wire) of 3-prong AC cords is also a problem.

Is line noise worse now than it used to be?

Yes, considerably. There has been a tremendous rise in the AC noise level in the last 20 to 25 years. The switching power supplies in personal computers are a huge contributing factor. In 1978 it was common to see considerable noise at high (RF) frequencies such as 1MHz. However, it was unusual to see more than 1mV at much lower audio-band frequencies such as 3kHz. Today you can measure as much as 3 volts of noise within 20 feet of a computer server room! To put that in context, if you induce 1 volt of AC noise in a video monitor, you will lose sync. Three volts of audio noise is nearly five times greater than the average audio signal level in a preamp! Fortunately component power supplies suppress most of this, but they CAN'T suppress it all.

Is line noise as big a problem as accessory manufacturers would have us believe? Does noise really degrade the performance of audio components?

The answer to this question needs to be qualified by comparing the signal level to the noise level. For example, if my music signal is being modulated by noise that is 50dB below 0dBu, the negative impact is inconsequential if I'm listening to a dance mix with 20dB of dynamic range. However, if the source material has a dynamic range of 75dB to 90dB, we have a problem. It's particularly an issue for anyone who cares about sound quality given that the ambient sounds and some harmonic overtones will be at the bottom of the amplitude scale. These low-level sounds are the most prized part of an audio signal and what governs tone color and presence. If these signals are modulated and/or cancelled by noise, we lose resolution.

Of course, the fact that an AC filter can eliminate some noise at certain frequencies is not enough to ensure a superior sonic result. The filter must be linearized for all AC impedances typical in a system's operating range, and it must cover a wide enough bandwidth of frequencies to unearth lost or distorted audio content.

Shouldn't the transformers that feed our houses filter and isolate the power we use from the AC grid?

Not really. If they were isolation transformers specifically designed for noise reduction, then yes, it would help. However, these high voltage step-down transformers are designed purely for power distribution. The 240VAC center-tapped output of the secondary windings of the transformer that feeds your home does not have any faraday shields to eliminate capacitive coupling of noise, and further, the potential reduction in symmetrical noise is compromised because the center tap is probably not true center and the load off either line phase may not be even.

You can often measure 3 to 10 volts differential between center tap neutral/ground and either line phase. Imagine a push-pull tube amplifier with severely unmatched output tubes! This is a recipe for noise, not noise reduction.

A power supply is supposed to isolate the working circuitry from the AC source. Are the power supplies in today's components inadequate?

They always have been and the problem is worse now. In decades past, when power line noise was lower, that noise burst through the power supply as current noise, and the noise on the ground wire that could be capacitively coupled into the circuit was low enough in signal level not to interfere in a manner that was noticed. As noise has increased in both bandwidth and level, there is too much for a power supply or built-in circuit to adequately reject. Further, many fine electronic components employ switching power supplies today. These have advantages in size and weight, and utilize a smaller transformer or no transformer at all to modulate circuitry with a magnetic field. Unfortunately they have relatively poor noise rejection.

The best power supplies for premium audio that I've measured are actually the ones that were championed in the 1930s! An example of this would be a transformer followed by a large tuned choke, built-out with several capacitors and resistors. Selenium rectifiers also allowed less noise than today's typical solid-state counterparts.

Battery supplies for low-current preamps are okay, but only if the batteries are lithium. Lead acid and alkaline batteries actually produce more current noise—or ripple—than typical linear supplies run off the AC line!

Can one component pollute the AC power that another component uses?

Yes! This is why an ultimate no-expense-spared AC conditioner would totally isolate every line level component from every other component. There would be no common hard connection of AC line, neutral or ground.

Think about home theaters. A typical flat screen monitor has 5 to 7 switching supplies in it! Your preamp does not want to see that noise backwashed into it.

Do power line conditioners really remove all noise from the AC line?

That would be impossible. As with many endeavors, physics often reaches up and bites us on the behind. There are many technologies that can be employed for noise reduction, and they all have their trade-offs. The trick is to correct as much as possible without doing harm to the sound of the system—and most AC conditioners certainly do harm the sound.

If power line conditioners work as advertised, why do they often make the sound of an audio system worse?

For one thing, many filters are not linear! Most AC filters were designed for use in a laboratory, not your in-home audio system. Typical AC filter attenuation curves change radically with the current drawn by the system, and the frequency of the noise. This creates resonant peaking in most AC filters, which can actually amplify noise over part of the audio bandwidth.

If the masking effects of AC noise are not reduced as evenly as possible, it's akin to removing a veil from your loudspeaker's tweeter, while the midrange and woofer are still covered up. Many audiophiles have noticed that traditional AC conditioners from many sources have re-voiced their systems in a very displeasing way. Sometimes defensive electrical engineers have told them that they're delusional. Far from it!

What we audiophiles prize most is low-level signal content. If that content is uniformly modulated we suffer but we accept what we cannot have or what we haven't yet heard. If we attempt to remove this noise, but do so inconsistently—say reducing noise at 3kHz by 5dB, 5kHz by 40db, 7kHz by 10dB, and 10kHz by 1dB—we will have a system that sounds like it's been poorly equalized or severely re-voiced! Not on the loudest peaks, but in the low-level signal harmonic content, which is the part of the signal that audiophiles and musicians prize the most.

Secondly, we cannot treat all components as if they require the same thing. Power amplifiers will suffer from some types of AC noise modulation (particularly asymmetrical noise). However, current compression will be the primary offender given today's relatively inefficient loudspeakers and extraordinarily dynamic audio content. Amplifiers strain for current at every moderate-to-high volume transient signal they attempt to reproduce. Their power supplies are robust and well designed; it's simply an unfortunate truth that the harder they're pushed, the lower their supply impedance becomes, and the more it appears to the transformer that it's feeding a dead short (albeit for a very short duration). When this happens a huge surge of current is drawn from the wall's AC supply until the amplifier's power supply is again stable and free of core saturation and distortion.

If a typical AC conditioner is placed in series with the power amplifier, the series components—transformers, inductive chokes, relays, and others—will actually raise the AC impedance making current compression WORSE.

Furman's Transient Power Factor Technology actually runs the power amplifier through a parallel circuit with tuned capacitance that will function properly with our series-parallel circuits for the other audio components, while providing power amplifiers with a current reservoir (for up to 25ms) from a high frequency source that has lower impedance than a 40-amp dedicated service! This extends the dynamic range of even the largest monoblock amplifiers rather than crushing their performance.

Typical amplifiers sound faster, with greater extension and control. Power amplifiers are only as good as their power supplies—that's a great deal of what you're hearing! Our transient power factor circuit gives these supplies the current-on-demand they need.

What is "balanced power?"

A one-to-one isolation transformer with a very precise center tap can provide balanced AC. Ideally the transformer will have precisely matched windings on either side of the center tap. The top winding will be half the voltage of the incoming line voltage relative to center tap ground, while the bottom winding will constitute the other half. It is critical that these windings are in opposite polarity to each other so that symmetrical (common mode) AC noise will cancel. The only drawback other than weight and expense to this technology is that it should NOT be used for power amplifiers, as ANY transformer will raise AC impedance, which may create current compression. Also, this technology will not work alone to reduce AC noise from line level audio components because only symmetrical noise is cancelled. Reducing asymmetrical noise will typically require a low-pass filter.

What about parallel inductors?

That's a poor man's way of creating what balanced power or a good isolation transformer would do. The problem is that the parallel inductors are hard to incorporate in a linearized circuit, and they cover a VERY limited range of RF frequencies.

Are re-generators better?

It depends on what's meant by "re-generation." Some manufactures have used this term to describe isolation transformers. For comments on that see the preceding discussion. If we're talking about active circuits that convert AC to DC then re-generate an AC signal, that's different.

An active re-generator is typically a voltage amplifier with a limited bandwidth. They're very inefficient, run quite warm as a result, and are large and heavy. Ten amps RMS is extraordinarily high current availability for such a design, and in a dedicated sound room such a device will eliminate the need for central heating in the wintertime. These designs are excellent for voltage regulation—delivering output that is typically ± 0.1 VAC!

The frequency is adjustable as well, which is superb for motors. Most audio and video products do not need AC voltage regulation, however, unless the voltage is well below 114 VAC or above 126 VAC.

The problem with active regeneration is limited output that can create current compression by considerably raising the AC impedance. When re-generators run out of peak current they will clip far more severely than your AC outlet ever would. Also, if an AC re-generation power amplifier and its power supply could eliminate all forms of noise, nothing could get past your components' regular power supplies as they are virtually identical circuits. Unfortunately this is NOT the case. Even common-mode noise won't be eliminated from signals outside the frequency bandwidth of the active circuits in a power re-generator. A well-designed passive filter has the clear advantage for eliminating AC noise. Additionally, an AC re-generation amplifier will not be able to isolate ground noise—it's passed straight through to your components.

Thank you, Garth, for this illuminating interview! APJ

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Vibration Control Products

by Richard Hardesty

A large group of accessories can be classified as vibration control products. Included are equipment racks and amplifier stands, points and feet that go under components, record mats and clamps for turntables, damping rings that go around vacuum tubes, and similar devices. Vibration affects all audio components and the impact that mechanical vibration has on sound may surprise you. Most accessories don't help but some can improve sound to a remarkable degree.

It's obvious that a turntable is susceptible to vibration. Everybody who has done any experimentation knows that subtle changes to turntable suspension and record damping devices, like mats and clamps, can make major changes in the sound you hear when playing vinyl. It's fairly obvious that the stylus that reads the information engraved in the record grooves can be disturbed by vibration. Susceptibility to mechanical vibration is less apparent with digital disc players but almost as important. Preamps and power amplifiers, especially solid-state units, appear to be even less susceptible but this is an illusion. Vibration affects everything, even cables.

You might ask how vibration can affect a solid-state component with virtually no internal wiring. Think about accelerometers and strain gauges. Conduction through these silicon-based components changes as they are bent or moved. The output devices in amplifiers are usually mounted directly on large extruded heat sinks. Run your finger down an amplifier heat sink to see just how easily these structures can be excited. Vibration will have some sonic effect on all audio components. High-gain components will be most vulnerable.

"Acoustical energy from the speakers vibrates everything."

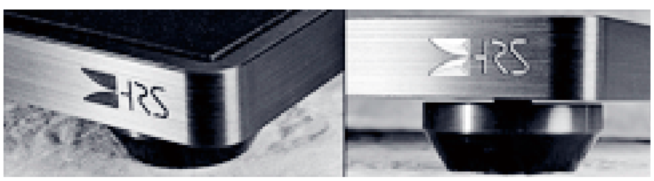
Sources of Vibration

There are some internal sources of noise. Turntables and disc players have motors and bearings that generate mechanical vibration. Most audio components have big power transformers that vibrate mechanically. These internal noises are usually dwarfed by the effect of the loudspeakers, which are the major source of energy that shakes our audio world. Audio components are attacked externally by vibration created by the loudspeakers as they convert the electrical audio signal into mechanical energy you can hear. Speakers move air, which in turn moves everything else.

Acoustical energy from the speakers vibrates everything. You can feel it by gently placing fingertips on the coffee table or nearby bookshelves. It can be detected with a stethoscope placed on or near an amplifier heat sink. This energy can enter and affect all audio components.

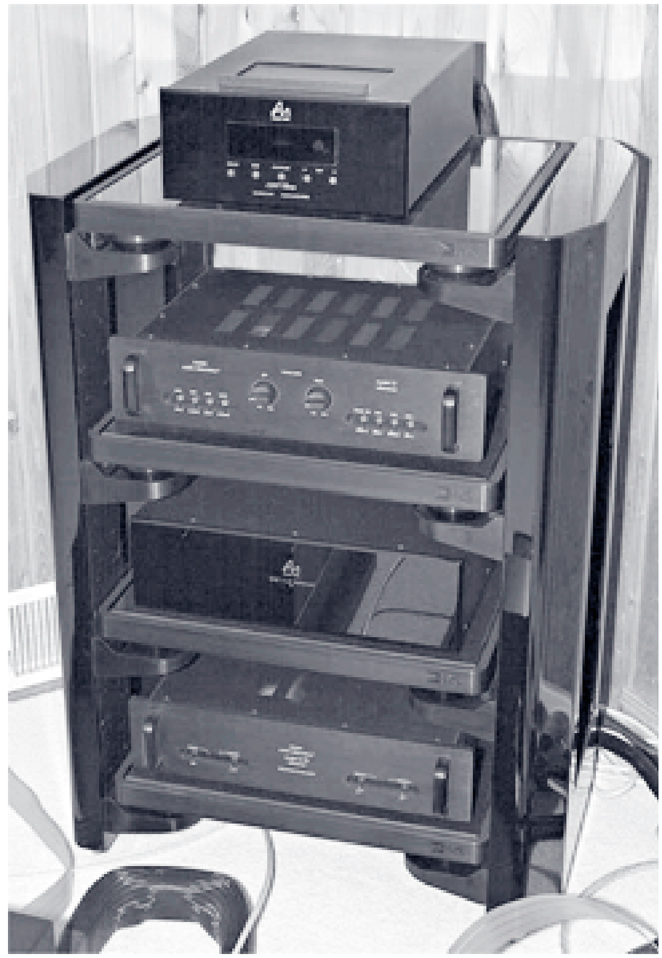
Paths of External Vibration

External vibrations emanate primarily from the loudspeakers and enter audio components by two paths: 1) through the support structure upon which the component sits, and 2) directly through the air from the speaker to the component.



In an ideal world expensive audio components would be designed to operate in this real environment. Most audio components, however, are designed by electrical engineers, who are sometimes aided by mechanical engineers. Electrical engineers have no training in vibration control and most mechanical engineers have little experience in this field. Some audio designers are self-trained and haven't been educated much at all. That opens up a fertile field of vibration control accessories, most of which were discovered by accident and many of which simply change rather than improve the situation.

There are exceptions that have been carefully designed by experts in the field. The vibration control products made by Mike Latvis of Harmonic Resolution Systems are among the accessory products that I've heard that make a profound difference in the sound of audio components—even very expensive high-end components. I'll interview him next and let him make his own case. APJ



Interview with... Michael Latvis

by Richard Hardesty

Michael Latvis has devoted his career to vibration control. He has degrees in mechanical engineering and design, over twenty years of experience and thousands of hours of professional training including product design, development and manufacturing in industries focused on interior noise reduction and vibration isolation systems for commercial aviation; helicopter rotor bearing, engine and transmission vibration isolation systems; army and navy weapon systems shock and vibration isolation products; and industrial computers and electronics vibration isolation systems. He holds three US patents and various international patents related to shock and vibration isolation products and has published and presented a number of papers about vibration and noise control products in various industries.

Mike plays the trumpet and has always loved music and home audio systems. In 2000 he founded Harmonic Resolution

Systems (HRS) to develop products specifically for high-end audio and video equipment. His decades of technical experience combined with a dedication to music make him a perfect candidate for an **Audio Perfectionist Journal** interview.

Readers may want to know: is the listening room really filled with lots of vibrating energy? Where does this vibration come from? Does this vibration degrade sound produced by an audio system? Can vibration control devices eliminate vibration? Will eliminating or minimizing vibration actually make sound better? Let's consider these questions separately and see what Mike Latvis has to say.

Can vibration cause audible degradation or is this just another way for accessory manufacturers to make money by selling products that "solve" a non-existent problem?

Vibration and mechanical noise cause a significant audible degradation of the signal. That vibration causes this degradation is somewhat less obvious to many people but the end result is signal damage. The loss of information and added artifacts caused by vibration and noise can remove the essence of what makes a high-end audio system a special experience for the listener. The sonic improvements made with well-designed noise reduction products are not only significant but also unique in nature and cannot be achieved by other means.

What are the sources of vibration that affect audio components? In other words, what shakes our audio components? Does this energy enter the components from the supporting rack or directly through the air?

There are several sources of vibration in an audio system. The most significant and obvious sources of noise in the vast majority of systems are the speakers. The energy from the speakers takes more than one path to your components where it becomes an issue.

Speaker-generated noise may take one of two primary paths to the components. (1) Structure-borne noise is vibration transferred from the support structure to the audio components. The structure can be excited directly from the floor or from the air. (2) Airborne noise is energy transmitted from the drivers of the speakers to the air which in turn energizes the components, the room, and all of the structural devices (shelves, etc.) supporting the components.

Can vibration really degrade the performance of audio components other than mechanical devices like turntables and perhaps CD players?

Our engineering tests, listening tests, and experience with a very wide range of Harmonic Resolution Systems customers,

using a very wide range of systems and configurations, all confirm that vibration and noise can significantly degrade the performance of an audio system. We find that a majority of components that have electronic circuits that generate, transfer, or carry the signal directly can be audibly improved by addressing the impact of vibration and noise that reaches the component circuit.

How can vibration affect solid-state components, like preamps, that don't appear to be microphonic?

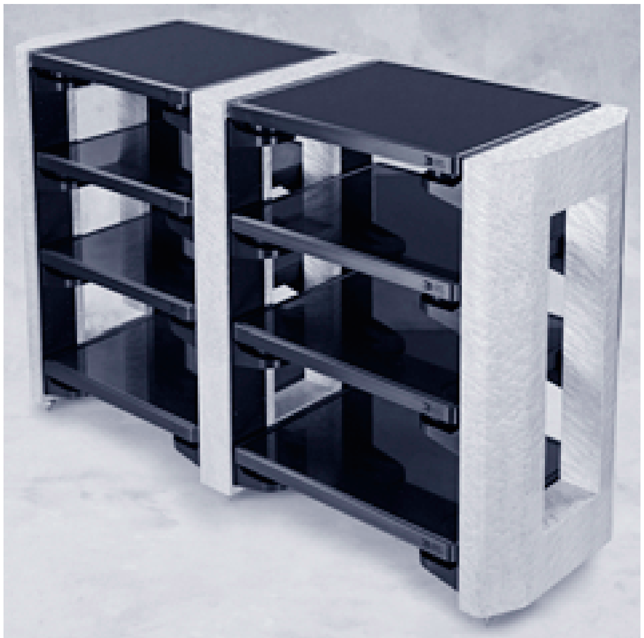
This is a great question because at first glance this may not be obvious to many people. There seems to be a more general acceptance that turntable performance can be improved by reducing mechanical noise that reaches the stylus. It is obvious that the turntable cartridge is an accelerometer reading the frequency and amplitude of the grooves in the album. The cartridge converts the mechanical motion to an electrical signal, which is then amplified to drive the speakers. Because the cartridge is converting mechanical motion to an electrical signal, most people can see clearly that erroneous mechanical noise that also reaches the stylus will be converted to electrical noise and that this noise will likely damage signal quality.

What is less well known to audiophiles is that there are many other devices that convert mechanical noise to an electrical signal. In industry these devices are commonly used to measure stress in parts, the dynamic response of mechanical systems, and the frequency and amplitude of vibrating systems. A host of other measuring devices are used in many processes and product applications. All of these devices take a mechanical event and convert it to an electrical signal in order to easily process and use the information.

One of the most common devices used by engineers to measure mechanical motion is a strain gauge. A strain gauge is very simple and consists of a relatively thin wire bonded to a film. The film is then bonded to a mechanical part. A charge amplifier is used to put an electrical signal through the wire. When the part under test moves, the thin wire changes form and the result is a change in resistance. The change in resistance is measured as a change in voltage through the device. The mechanical motion is now read as an electrical signal. This mechanical-to-electrical conversion through a simple wire is so reliable that engineers developing critical aircraft systems and many other products use it to measure stress, frequency, and vibration amplitude. Companies all over the world use strain gauges every day.

Another way to convert mechanical motion to an electrical signal is by use of piezoelectric-based materials. The piezoelectric devices are often man-made or naturally occurring crystals, such as quartz crystals, that produce a charge output when they are compressed, flexed or subjected to any force. The electrical signals from piezoelectric materials are used daily to measure mechanical acceleration by monitoring the electrical response of these crystals.

The crystals used for vibration and noise measurement are similar in nature to the parts used in digital-to-analog conversion (DAC) circuits. The fundamental understanding that the most common way to measure force, pressure, vibration, and shock is by taking advantage of the many electrical devices



that change characteristics in response to these mechanical events, provides the necessary insight to see that a high performance electronic audio circuit that contains many different electrical components is very likely to respond to the vibration and noise to which these components are subjected.

With this understanding we can then begin to imagine the number of different locations in a high performance audio component where the signal quality may be degraded by vibration and noise. We can then also see that if the environment is changed in any way we might hear this change. More importantly, if this vibration and noise are reduced in a significant way we can reduce or even eliminate the damage to signal quality by reducing or eliminating the mechanical noise affecting the component. Reducing noise that degrades the signal provides the listener with a new level of information. Now he can hear just the signal and a unique level of detail and nuance from the actual recording.

Do points, spikes and rubber feet placed beneath audio components reduce vibration or just change the frequency? I've heard the sound change when Jenga Blocks are placed beneath a solid-state preamp but this eludes most explanations based on science I understand. What's really happening here?

Our experience at Harmonic Resolution Systems Inc. is that with a wide range of high performance audio systems you cannot change anything that touches the chassis of an audio component without changing the "sound" of the component. Based on our discussion of how electrical circuits are sensitive to vibrations, you can see that any time you touch a component chassis with another object you change the response of that structure to vibration and noise. The change that you hear when you place an object in contact with a component is the change in the chassis' response to the new object or the change in location of that object. This change may or may not improve the sound.

Whether you have just changed the environment or truly improved the environment depends on how well the product being used was engineered and how effective this solution is with the component chassis it is being applied to.

It seems intuitive that heavy objects are more difficult to move than lighter ones. Wouldn't a big, heavy support rack stop vibration problems?

This intuition is actually true to some degree because Newton's second law is that F=MA. For a given force (F) the level of acceleration (A) on an object will be lowered as the mass (M) increases. However, mass has another effect on system response. As mass increases, changes occur in the natural frequency of a system because mass is part of that relationship as well.

If the natural (or resonant) frequency of the object coincides with the input frequency (vibration) then you might actually amplify vibration. Mass also decreases acceleration of an object in a linear manner. If we want to significantly reduce the vibration by multiples of 10, 100 or even 1000, mass alone may not be the most practical way to achieve high levels of noise reduction. So in general you can make improvements by increasing mass in a knowledgeable way. But with a more complete approach that takes advantage of mass along with a

number of other engineering principles you can reduce vibration and noise to a much greater extent.

The reference level MXR Audio Stand by Harmonic Resolution Systems does weigh in at about 500 pounds for a four-shelf system but it also contains many other materials and technologies that contribute significantly to the performance of this system. Without these other technologies the performance is significantly reduced.

Can vibration control products actually eliminate vibration that a heavy rack can't?

Yes, the application of many different technologies can actually be much more effective than just adding mass. If you look at the Harmonic Resolution Systems M3 Isolation Base we do have a shelf weight of approximately 65 pounds, but we also use a custom-designed primary isolation stage that is set based on the load of the component and greatly reduces the magnitude of the noise before it reaches the mass. The M3 Isolation Base also contains seven different materials and more than forty parts to make up a complete system that reduces noise and resonance within the frame itself. I firmly believe that a well-executed complete system approach will outperform a design approach that maximizes only one design variable.

In the 1970s people were building isolation boxes for turntables. Is this still a good idea?

It is a very good idea if they are built in a manner that reduces the level of noise reaching the stylus in a known way. The principles we have discussed are not new. Accelerometers and the knowledge of mechanical-to-electrical conversion go back well before the 70s as do the principles of reducing vibration and noise. The fundamental principles involved are not new but the detailed understanding and the dedicated application to high performance audio, which started with turntable suspension systems many decades ago, is now being taken to a new level.

At Harmonic Resolution Systems we have spent the last seven years developing a complete line of materials and products specifically for high-end audio systems that allow you to reduce signal degrading noise to an entire system at each and every component. This work, and work by other companies dedicated to the high-end audio industry, has allowed us to achieve a new level of performance that until recently was not available to audiophiles.

Will we hear an audible improvement that justifies the cost of vibration control products?

Based on our experience and feedback from our customers over the past seven years, the answer is yes. You will definitely hear a very significant improvement that more than justifies the cost of the products. You must set your well-thought-out and carefully selected components on something. This is not an option. What you select to put your components on and in contact with will definitely impact their performance. I think your expenditure should be proportionate to the rest of your system. We use a general rule that 10% to 30% of your system cost should be invested in well-designed mechanical noise reduction products. This will ensure that the rest of your investment is delivering peak performance and that you will hear the music in a way that can only be achieved with the application of these products.

Thanks, Mike, for an edifying interview. APJ

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Acoustic Treatments

by Richard Hardesty

Acoustic treatment materials and systems, often called "room tuning accessories," can literally make or break the sound of a home audio system. Attending to room acoustics is necessary in order to achieve maximum performance from an audio system but room treatment can be easily overdone. Adding excessive acoustic treatment materials can depreciate the sound created by even the finest audio components.

The home theater craze exacerbated the trend to excess because we were assaulted by an avalanche of misinformation which resulted in many "custom home theaters" that were virtual anechoic chambers. The goal, as advocated by many home theater experts, was to eliminate all natural room acoustics and deliver all sounds, including artificial ambience cues, from as many speakers as could be included to surround the listeners. Spectacular sound effects sold home theaters. During the process of delivering more spectacular sound effects, stereo imaging and natural music reproduction were forgotten, or at least relegated to a position of less importance.

Sound effects are fine for movies and video games but have nothing to do with natural music reproduction, in my opinion. Well-made stereo recordings can provide all the spatial information necessary for a realistic musical experience if the room doesn't get in the way. Touching up the acoustics of a normal room is not that difficult or mysterious. Overdoing it can render the room useless.

Any room in your home should be a comfortable and attractive place in which to sit and converse or read or whatever. Such a room will probably be a good place for a music reproduction system as well. You may have to attend to some acoustic adjustments to achieve an excellent listening environment but you shouldn't have to go to extremes and you may be disappointed if you do.

Anecdote

Let me share a personal experience. Shane Buettner is a friend and colleague who has written for several major periodicals and contributed product reviews to **Audio Perfectionist Journal**. A couple of years ago, Shane built a custom home with a purpose-designed media room for his audio and video equipment. He had a specialist company install a complete acoustic treatment system in that room. This installation involved extensive measurements and adjustments that included suggestions about speaker placement and fine-tuning the ratio of behind-the-cloth treatment panels that provide absorbsion, reflection and diffusion. The retail cost of the installation was about $30,000. Yes, that's thirty thousand dollars! The appearance of the room, which included designer fabric covering all wall surfaces, was beautiful. Sound was acceptable but not great.

Shane installed state-of-the-art components and we started to listen, more than casually. We reviewed several product categories, like preamplifiers, by directly comparing components in groups, using both his listening room and mine. For many of these comparisons, the major components in Shane's audio system and mine were virtually identical—the only major differences could be attributed to room acoustics. His room was purpose-built and "acoustically treated" by a major purveyor of acoustic room treatments. My room was a regular living room with a few hundred dollars worth of readily available materials to adjust acoustics. Shane's tastes are slightly different and I'm writing this so the following impressions are mine alone.

Shane's system/room delivered better than average sound that was slightly less lively—more dynamically compressed—than mine. The performers seemed more distant in his environment; midrange detail was good with a slightly reduced sense of intimacy. Compared to mine, his bass was a little soft, slightly rubbery and lacked impact. I preferred the sound in my room and the cost advantage was enormous.

Shane wasn't entirely happy and, after several attempts by the company to adjust the installation, he started to slowly remove acoustic panels from behind the cloth wall covering. The more material he removed the better the sound, in my opinion.

The speaker positions recommended by the treatment company were based on a computer analysis of "room nodes" and were supposed to provide flatter frequency response. Their recommended placement was detrimental to stereo imaging.

I think you can save thousands of dollars and make better sound if you just think about what you want to accomplish and use common sense to achieve your goals.

Room Treatment Goals

A good listening room should be as lively as possible with a relatively short decay time. The reasons for this are simple: a dead room requires much more power input to achieve the same perceived listening level and higher power means higher distortion from your amplifier(s) and speakers. An overdamped room will cut the maximum sound level peaks and reduce the perceived dynamic range.

Damping can be achieved by introducing materials that absorb rather than reflect sound. The more sound reflected, the livelier the room. The more sound absorbed, the deader the room. You'll probably hear slightly more detail in a dead room but dynamics will be restricted and the punch necessary for an involving sense of rhythm and pace may be depreciated.

A good listening room will have a decay time that minimizes, but does not eliminate, reverberation. A room with little or no reverberation sounds closed in and uncomfortable. A room with too much reverberation sounds cold with blurred detail as new sounds are smeared by the lingering reverberations from previous sounds.

Reverberation can be adjusted by adding absorption and diffusion. Diffusive materials reduce reverberation by reflecting sound at multiple angles so it gets bounced around and attenuated rather than directly attenuated by absorption. Diffusion is generally preferable to absorption but the critical word is balance.

Reflected sound from behind and beside the speakers is more confusing to the ear because it comes from a similar direction as the direct sound from the speakers and is only slightly delayed in time. You want to hear the signal from the speakers primarily and the sound from the room secondarily.

A measuring microphone integrates the sounds from the speakers and the room but your ear/brain differentiates between them because of timing and direction. The goal of all room treatment is to make this task easier. There's nothing mystical about it. You want to hear more from the speakers and less from the room without making the room an uncomfortable place.

Remember, virtually all that you've read about acoustics was based on large venues where sound “waves” could be completely developed, even at low frequencies. Dimensions in domestic rooms are often smaller than complete wavelengths. When room boundaries don't allow complete wave development the room starts to act as a pressure vessel and the laws of hydraulics may be more appropriate than those of acoustics. Large venue acoustic “rules” don't always stand up to scrutiny in small rooms. For some intriguing information read the studies from Dolby Laboratories about bass from a single location (re: the development of 5.1 channel audio systems).

Room Treatment Materials

The most effective treatment material for acoustic absorption is Owens 705 rigid, compressed fiberglass. You know those panels that define your cubicle at work? They're probably made from Owens 705 rigid, which is very effective for reducing sound transmission from one open-top cubicle to another. Also very effective for absorbing rather than reflecting sound from your living room walls and providing fairly linear absorption over a wide range of frequencies above about 200Hz.

There are many aesthetically pleasing materials that can provide acoustic diffusion. My fireplace is covered with stone veneer that is very stiff for good bass and has irregular angles of reflection for a highly diffusive surface. My room has windows front and rear covered with plantation shutters that can be adjusted to deliver the desired amount of diffusion.

Stiff boundaries mean better bass. Flexing walls act like bass traps. Dispersive panels made from Pegboard™ backed by soft fiberglass are often detrimental to bass performance because they make boundaries spongy.

My Usual Method

I typically place absorptive material directly behind and beside the speakers (usually not in the center) because it is more difficult to differentiate between the direct sound from the speakers and sounds reflected from nearby surfaces. These reflected sounds arrive very soon after the direct sound and from similar angles and usually need to be attenuated.

I try to use dispersive materials everywhere else unless the room is too reverberant. Opposing reflective surfaces can be problematic. Carpet the floor opposite a hard ceiling. Break up reflections between two opposing hard wall surfaces with dispersive material. Don't try to kill all reflected sound with absorption. Absorb the worst problems and diffuse the rest. If you can't hear a specific problem leave the room alone and enjoy the music. APJ

Bypass Testing

by Richard Hardesty

Bypass testing is often misunderstood but it's definitely the most effective method of evaluating audio components subjectively. Bypass testing, as described below, can be used to evaluate the sound of active audio components as well as passive devices and cables. First you have to understand the principle.

The goal of bypass testing is to determine what effect a device under test has on the sound of an audio system. This is accomplished by comparing the sound of an audio system with the device under test in the signal path to the sound of the same audio system without the device in the signal path. The difference you hear is the effect the device has on the sound of the system. If that sounds simple it's because it is! Too bad more people don't actually do it.

Active components and speakers can be easily evaluated on the test bench to determine which ones have the potential for accurate reproduction and which ones don't. You can measure the signal-to-noise ratio, bandwidth and common types of distortion but there are limitations to so-called objective testing.

Objective Testing

The performance of most audio components can be objectively measured with test instruments. The signal at the output of the device under test is compared to the input signal and deviations are called distortion. These tests are compromised to some degree because the stimulus is usually a simulation and some kinds of distortion have not been thoroughly identified or investigated.

An audio system is designed to reproduce music but test signals (stimuli) are often steady-state sine waves or pseudo-random noise. Bandwidth, signal-to-noise ratio, harmonic and intermodulation distortion are well understood. Other distortions, which may be clearly audible, are not as completely understood and often not measured at all.

Poor measured performance guarantees that the component under test can't accurately reproduce the input signal. Excellent measured performance only guarantees the potential for good sound. After potential accuracy has been established, listening tests can begin. Only listening can determine which potentially good components actually sound pleasing.

Objective measurements are very helpful in exposing poorly designed active audio components and speakers but may be inadequate when discussing accessories because subtle audible improvements may not produce measurable change and we haven't yet figured out all the parameters which cause clearly audible effects in components like cables.

Testing Accessories

Want to learn which accessories work and which ones don't? It's simple, really. Listen to the system without the accessory and then with the accessory and compare the sound. Take the accessory out and listen again. Set up a simple bypass test if you can, and compare the sound with and without the accessory. This sounds ridiculously easy but that's really all there is to it!

When you compare the sound with and without the accessory you'll hear just what that accessory does to the signal, good or bad. Sometimes you'll be hard-pressed to hear any difference at all. Sometimes you'll hear a small difference that you may consider an improvement. Be sure the difference is an improvement and not simply a difference.

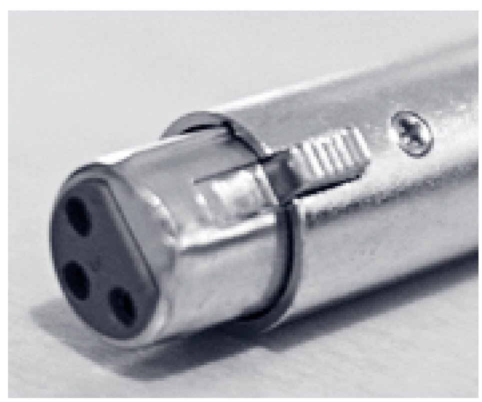
In most cases the improvement in sound won't be worth the price of the accessory and in some cases the sound will actually be worse when using the accessory. I've auditioned many accessories that made the sound much worse, many more that did little or nothing and a very few that actually made a small sonic improvement. With the exception of cables, I have seldom heard an improvement that justified the cost of the accessory, but when you've purchased and set up the best active components and speakers you can afford, even minor enhancements can be worth their cost.

Bypass Testing

A bypass test allows you to quickly compare the sound of an audio system with and without the component under test in the signal path. Bypass tests are usually slightly compromised because an audio system won't work when a major component or cable has been removed from the signal path but there are ways to get around this compromise.

Your goal is to hear what the device under test does to the sound of the system. You'll hear what it does by listening with the device in the signal path and listening again to the system without the device and comparing the sound. With a little practice you'll be able to hear exactly what sounds the device adds to or subtracts from the signal. This is the only valid way to evaluate and compare components subjectively and it assumes that you have assembled an audio system from components chosen to be as transparent as possible.

Want to hear the sonic effects of interconnect cables? Get an integrated amplifier with "preamp out" and "power amp in" connectors. Listen with jumpers between these pairs of connectors and then remove the jumpers and insert interconnect cables. You should be able to hear what the interconnect cables do to the sound of the system. Don't have an integrated amp? Put the cables under test in series with the usual cables between your components and listen. Then remove the cables under test and listen again. Balanced cables with XLR connectors can be daisy-chained directly. Unbalanced cables with RCA connectors will require female-to-female RCA adapters.



It's harder to use bypass testing with speaker cables. I use mono amplifiers and the shortest possible speaker cables to simulate the bypass condition. Then I use eight or ten feet of the cables under test to be sure I'm hearing the characteristics of the test cables. Put the cables under test in series with the short wires and listen. Then remove the cables under test and listen to just the short wires. You should be able to switch between the cables under test and the bypass cables fairly quickly but you won't be able to go immediately back and forth between the two. You won't have to.

You can test lots of components this way but you may have to use some ingenuity. My VTL 7.5 preamp has inputs that can be configured to unity gain. I can place another preamp in a unity-gain loop (preamp out to unity-gain input), adjust the volumes with a meter and compare sound with and without the preamp under test in the signal path.

Distortion

Any change in the signal is a distortion to that signal. Additive distortions include noise and coloration. Subtractive distortions include lost detail and imaging clues. The audio signal can pass through a perfect component and emerge unchanged. This is the goal of the high fidelity approach to music reproduction and the definition of transparency in audio.

Striving for "synergy" means stacking colorations one atop another in an attempt to achieve a "musical" sound. You might end up with a pleasing system this way but then again you might win the lottery or be struck by lightning. Anything's possible. APJ

Aesthetix Callisto Signature Line Stage Preamplifier

by Shane Buettner

I've been waiting for this one for a long time. APJ readers know the respect we've held for Jim White and his product design work. Jim's day job used to be at Theta, where he worked on the Dreadnaught and Citadel amplifier projects, and we've also lauded his Saturn series of components for his own company, Aesthetix.

But our firsthand experience with Aesthetix' gear hadn't yet extended to its flagship Jupiter series components, comprised of the Callisto line stage and Io phono stage. For years great sound has followed these components at trade shows and dealer demos alike. After reviewing and loving the Saturn series, the Callisto especially has been at the top of my review wish list, and I finally got my shot.

Aesthetix sent over the top-of-the-line $14K Callisto Signature tube line stage, which is a purist design to the extreme. It's a three-box design, with a separate power supply, separate input selector, and separate volume control for each channel. And that doesn't include a remote control, which is a $2,000 option.

From what little I've given you, it's obvious this isn't a product for the uninitiated or those squeamish about tubes. But for those who aren't intimidated by this design, the sonic benefits are unparalleled in a number of key respects.

Jupiter Series Overview

As mentioned, the Jupiter series consists of the Callisto line stage and Io phono stage, and each can be configured in a variety of ways. The Callisto is available in single (though still separate) power supply versions as the Callisto MK II at $9K. The Signature version, featuring a number of component upgrades, is $11K, and the second power supply costs $3K, so the total cost of the preamp reviewed here is $14K.

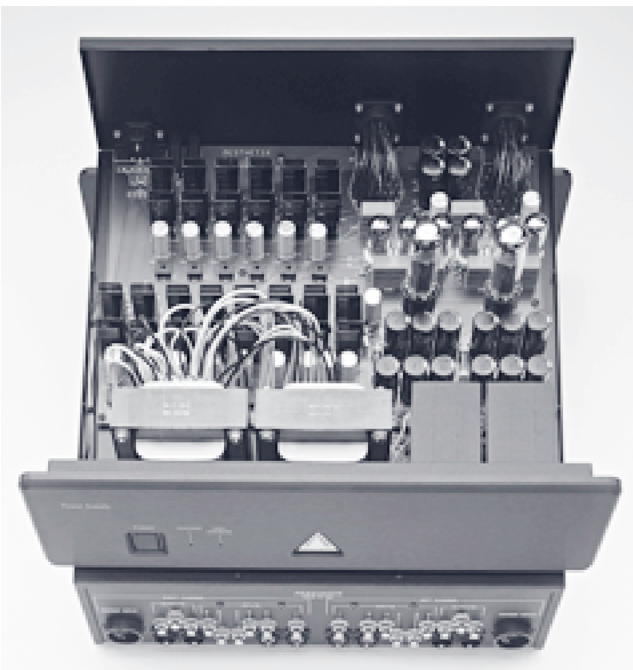
The remote control option is an additional $2K, making an already expensive preamp even more so. While it doesn't offer input selection, it does make the Callisto far more convenient to use by controlling the volume levels of both channels simultaneously. This option can also be added later, so if you can't spring for the $2K up front, you can still buy and enjoy your Callisto knowing that you can upgrade the ergonomics later with the remote option.

The Io is rather fascinating. It too features an outboard power supply in even its base MK II configuration, which is $6,500, or in an upgraded Signature version for $9K. However, users can also purchase the Io with a second input and the same trick balanced volume control that's used in the Callisto for $9K. This allows you to use the Io without a separate preamp, although you're limited to a phono source and one other input source. Kicking that up a notch to the Signature takes you to $11K, and if you're still not done, you can add another power supply for an additional $3K.

So, if your sources are a turntable and a single optical disc player, you can do all that for a price that's comparable to the Callisto line stage by itself. If you're a total madman, you can buy the full-on, dual power supply Signature versions of the Callisto and Io, and you're into Aesthetix to the tune of $23K. For that you get a total of six boxes to find homes for in your equipment rack, and over 40 tubes! Now that's a tube front-end!

Design

The Callisto is a three-box design with separate power supplies dedicated to each channel connected to the line stage unit by umbilical cords. This approach has its supporters and detractors. To that my only reply is that several of the very best preamps I've heard use this approach, including Ayre's K-1xe and VTL's TL7.5. Of the preamps I've heard that perform at the elite level, only ARC's Ref 3 eschews this approach.



Many stereo components claim to be dual-mono. The Aesthetix Callisto, in its dual power supply mode, really is. Inside the line stage unit, the two circuit boards are clearly mirrors of one another and each is clearly isolated in the chassis. On top of that, there are individual input selectors and volume controls for each channel. The downside of this purist approach is that every time you change a source or the volume level, you have to do it twice, and in the case of the latter you need to make sure the gain for each channel is matched. More on that later.

Each power supply is full tube, with tube rectification and a choke input filtering network. There are eight vacuum tubes in each supply (for a total of 16 between the two supplies), and each power supply box weighs more than any other preamp I've used, and more than a lot of power amplifiers out there. This three-box design is Heavy-Duty with a capital "H" and a capital "D," and there is no question that this rig has serious curb appeal when people walk into your music room.

The Callisto is fully balanced from input to output, and when the balanced inputs are used it employs no global feedback. Six decibels of local feedback are introduced when the single-ended inputs are used, and it's audible. The Callisto sounds much better with balanced sources, and can accommodate two of them.

The Aesthetix high resolution volume control is a 46-position stepped attenuator that Jim White believes is unique in configuration. According to Jim, it's common for a resistor ladder volume control to have more resistors in the signal path at certain volume positions than at other positions, often at lower volumes. For example, if his 46-position resistor ladder were configured in this fashion, assuming full Off is 0, the signal would be going through 35 resistors in series at volume step 10, with 10 resistors in series going to ground.

In the Aesthetix volume control, two switches are used in the attenuator, and at each position on the volume control there is only a single series resistor and a single parallel resistor in the signal path, from steps 1-45. Jim believes that this delivers superior sonic results and eliminates any disparities in resolution and sound quality throughout the entire volume range. The downside? Expense and complexity. There are eight very expensive, 46-position Shallco silver contact switches used in the two volume controls needed in the Callisto. For each channel there are two switches for each attenuator, and two attenuators for each leg of the balanced signal. There are 360 Roederstein resistors—total—in the two volume controls. Wow!

Living With A Three-Box Tube Line Stage

It would be a gross understatement to call the Aesthetix collective an impressive piece of kit. The line stage unit still weighs more than a lot of single-box preamps out there, even though it contains no power supply or transformers or any of the other stuff that typically adds weight to a component.

There are start-up and power-down sequences that must be observed. Each power supply needs to be turned on and warmed up before the power amps are turned on (an LED on the power supply tells when it's time) and conversely, the power amps should be powered down first and the Callisto's power supplies second.

I think the candidates for this preamp already know who they are just as the people who'd never consider such a beast know who they are so I don't want to belabor the point, but this is a purist design, with three boxes, 24 tubes, two input selectors and two volume controls. And, unless you pay extra, no remote control.

This is obviously not the product for anyone looking to have a two-channel system share space with a home theater system. Although Aesthetix can tell you which volume position would be unity gain for a processor pass-through, you don't want to be burning 24 tubes to watch football on Sundays. I didn't use the processor pass-through at any time during the months I spent with Callisto, choosing instead to swap cables at the power amplifier end when I wanted to go from audio to video or vice versa.

I was worried, going into this review, that level-matching between the two volume controls would be cumbersome, in addition to not having the remote control option installed on my review unit. But I had no issues on either front. The volume steps are easy to feel as the volume is increased/decreased, and there are markings all around each volume knob like a clock face and it's only difficult to be sure the channels are aligned if the lighting is quite dim. Of course, the remote control option solves this issue entirely.

During my time with the Callisto the tubes did get noisy. But the truth is that I spent almost a year with the thing, alternating for months between it and the ARC Ref 3, and I replaced the tubes in the line stage once. A few times I took the top cover off the line stage and reseated the tubes and heard it quiet down, only to gradually ramp up in noise again. I'm not particularly phobic about tube hiss but when I heard the tubes during quiet passages of music, I took action.

This situation isn't foreign to tube components, but in my limited experience with modern tube gear it's not necessarily a given either. VTL's TL7.5 contains two tubes, and I replaced them a few times during the two years or so I owned that preamp due to excessive noise. (VTL has since changed the tubes used in the TL7.5 for that very reason.) On the other hand, I've lived with BAT's VK-51 and VK-51SE and Audio Research's Ref 3 for months on end without hearing any tube noise at all, let alone excessive noise. If this kind of thing bothers you, you know who you are and it's unlikely that you'd be considering this preamp anyway.

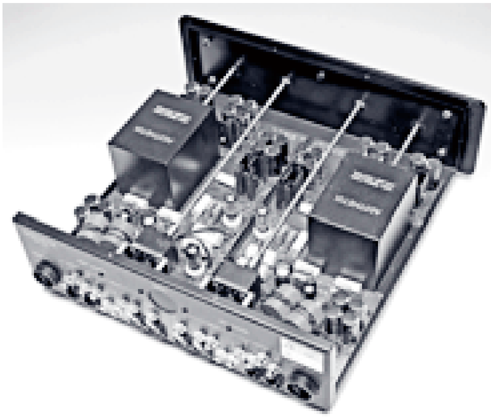
Sound

If you read a fair number of recent reviews you'll see references to the closing of the sonic gap between tube and solid-state components. Even here in the pages of **Audio Perfectionist Journal** we've commented that today's best solid-state amps do so many of the seductive things that tube power amps do. But for the most part, I think it’s just wrong to suggest that tube and solid-state power amps sound the same. For example, while I haven't done the head-to-head comparison myself, I have a hard time believing a high-feedback solid-state Halcro amp and a top-of-the-line VTL all-tube power amp sound more alike than not, as one reviewer suggested.

The reference to a convergence of tube and solid-state is meaningful in that the artifacts typically associated with these components are disappearing. The best tube gear isn't soft, warm and grungy and the best solid-state isn't analytical, sterile and non-musical. And yes, the Callisto is representative of this. The Callisto is as fast, neutral, dynamic and authoritative as any preamp I've ever heard, solid-state or not. If you want a tube preamp to add a vintage, warm glow to your music, this ain't the one for you.

In fact, I'd call the Callisto spectacularly neutral, if there is such a thing. What I mean is that sometimes the more neutral a component is, the less exciting it is in a short demo. VTL's TL7.5 is an example of a component that I had to listen to at length before it engaged me entirely due to the fact that nothing it did immediately jumped out. Not so here. While the Callisto is as tonally neutral top to bottom as any component I've ever heard, it's immediately commanding in listening for its see-through level of transparency and image focus and extension at both frequency extremes.

I have never heard a preamp that can match the Callisto's level of resolution and transparency, nor have I heard one that took me as far into the listening space, or was as astute at recreating these



spaces in my listening room. Two listening experiences in particular stand out as testament to this. When Classic Records reissued The Who's *Who's Next* on 200-gram vinyl I borrowed a copy of an original British pressing to compare the two. Not only could I hear every minute difference between the two, on the British pressing I could hear the very dimensions of the booth in which Roger Daltrey recorded his vocals. It was startlingly convincing, and the kind of listening experience audiophiles spend years chasing.

Another stunner was an original pressing of *Sinatra at the Sands* with Count Basie's orchestra on vinyl (also on a now out-of-print DVD-A disc that's nearly as good). Although a touch dry, this is an amazing recording. Frank's vocal and Basie's piano are fully realized, with thoroughly convincing image density. People talking, glasses clinking, the audience is just there. But where the Callisto steps beyond is in the way instruments and vocals bloom and decay, reverberating through and defining the dimensions of the Sands. As I listened, it was like the music was active sonar, pinging and drawing the interior of the Sands in three dimensions. I felt like I could draw a sketch of the place even though I've never been there.

Comparing the Callisto to today's best preamps is fascinating. There are additional nuances in resolution, and the extra carriage into the recording space is subtle at this level—the level at which the ARC Ref 3 and VTL 7.5 dwell. But even so, any additional resolution over these other fine preamps is weighted differently based on how superb those other preamps are. The Callisto betters the resolving power of those preamps, even if incrementally, and that is startling. The Callisto's closest competitors are very recent ground-up designs. The Callisto's performance against these formidable preamps is serious testament to its design. The Callisto has been in production since 1994, with the only significant design update occurring with the introduction of the Signature version in 2001 and the remote control option introduced last year. What's more, even a Callisto manufactured in 1994 can be upgraded to a fully current Signature version. That's staying power.

Going further in comparison, the ARC Ref 3 resolves nearly as much musical detail, and is more of a relaxed fit in listening without being soft in the slightest, which is very seductive. It's also got something beyond that ease in the way that it's completely non-mechanical. The Callisto is detailed as all get-out, but is also not hard or analytical in any sense. While it doesn't match the Ref 3's relaxed qualities, neither can the ARC quite scale the heights of the Callisto's tremendous resolution of detail.

The Callisto is unequaled in dynamic realism, micro and macro. And really, nothing I've heard comes close. The jump and staggering, genuine dynamic contrast is not quite like anything I've ever heard. Music played through the Callisto has a raw, almost "live" intensity, even when the music is studio recorded.

Overall, to my ears the Ref 3 and the Callisto stand tall at the top of the preamp mountain. The Callisto has no rivals in pure resolution of detail and dynamic punch. The Ref 3 is nearly the Callisto's match in that regard, and has a non-mechanical purity and complete lack of musical grain that at times makes the Callisto sound almost like too much of a good thing.

These two preamps are addictive in different ways, and the simple truth is that I know I could live very happily with either of these preamps because I have, for months on end. Sonically, when each was in the system I wanted for nothing, but would occasionally miss some attributes of the other. Although the remote control option closes the gap somewhat, the ARC is still less intimidating in being a single box with only five tubes, and it still boasts greater ease of use and convenience in functionality. Even with the remote, the Callisto is a purist design that's more demanding as an audiophile lifestyle choice.

Conclusion

The Aesthetix Callisto is entirely worthy of the high esteem in which it's been held for the years it's been in production. I've had the very good fortune to have a number of the finest preamps in the world in my listening room over the last few years. Even in this heady company, the Callisto is in a class by itself in many respects, and does things that the other outstanding flagship preamps I've heard simply can't do. If you can acquiesce to the commitments that this preamp demands, the sonic rewards are beyond reproach. APJ

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Audio Research Reference 3 Preamp Review

by Shane Buettner

The Denons and Pioneers of the consumer electronics world change out their entire lines, year in and year out. Companies like Audio Research don't. It's a big deal when Audio Research releases a new flagship preamp. ARC's previous top-of-the-line preamp, the Reference 2 MKII, was introduced in 2000. Its successor and the subject of this review, the Reference 3, didn't make its debut until 2005. And man, it was worth the wait.

The Ref 3 maintains its predecessor's $10K price point, and its looks are mostly classic ARC. But the design is brand-spanking-new from the ground up. The Ref 3 represents a complete rethinking of the preamp, as conceptualized by Bill Johnson and company, not an incremental change.

While ARC's reputation is certainly enviable, it's also got to be a daunting task to go about replacing something like the Ref 2 MKII, a preamp that's not only among the best of the best, but truly special. ARC has exceeded all of my expectations with this preamp.

A New Take On The ARC Look

ARC's classic handles and knobs for input selection and volume control are still there, but gone are the balance and tape monitor source knobs. Instead of a bank of toggle switches there are four buttons for Power, Processor Input, BAL/SE, and Mute.

But the most obvious and controversial way in which the Ref 3 deviates from the classic ARC look is in the single, massive fluorescent display that now sits smack in the middle of the front panel. The readout is huge and green—the kind of thing that someone who's legally blind could read from the other room. And people seem to either love it or hate it. I don't feel strongly one way or the other about the display, but I do feel that the preamp sounds silkier and quieter with the display off. I like very much that when the display is set to off, it turns on very briefly whenever changes are made to input, volume, etc., and then turns back off. Perfect!

The display is a reflection of the Ref 3's more logic-driven and "modern" operation. One knob selects input, another controls volume. A button push selects whether the balanced or single-ended inputs are in use; another button push can turn any input into a unity-gain, processor pass-through input. With the aid of its small but properly detailed remote control, the Ref 3 is as functional as it needs to be to fulfill its roll as an expensive flagship preamp, and then some, and yet it's simple in setup and day-to-day use. For this modern age the Ref 3 is an ergonomic triumph for sure.

Design

ARC has a remarkably informative web site, so I'm not taking credit for much investigative reporting here. The audio circuit is pure tube, but the power supply is referred to as hybrid, and also spec'd for 50% more energy storage than the Ref 2 MKII. The circuit boards and power transformers are outfitted with parts new to this preamp. Other eye-catching specs include bandwidth increased to 200kHz compared to the REF2's 60kHz, and distortion lowered by a dramatic (and audible) 40%.

As significant as all this certainly is, perhaps the most radical element of the new design is that it no longer uses any global feedback. Eschewing feedback has led to a renaissance of sorts in solid-state designs, in my opinion, and resulted in performance in these designs that has much more in common with the musical righteousness of tubes (particularly the zero feedback designs of Ayre and Theta that have been lauded in these pages for years).

Ayre's Charlie Hansen is one of feedback's biggest detractors, and in my conversations with him he's described global feedback as a time-domain phenomenon. While ARC's previous flagship didn't use much feedback, it used a little and after hearing both preamps I believe the REF 3 sounds better for eliminating feedback entirely. I think it's possible that this is even more audible when used with other components that minimize or eliminate feedback, and when time- and phase-correct speakers are used. This describes my system exactly.

Touching further on one of those other improved specs, let me talk about the REF 3's bandwidth expansion to 200kHz. I've written about this before, but bandwidth has somewhat mysterious effects on audio gear. Conventional wisdom would have it that 60kHz is well above our range of hearing and that going beyond that is sonically inaudible and therefore unnecessary. And yet, far more often than not the higher bandwidth products I've heard have sounded more open, transparent and dynamic, especially power amps. And indeed, the Ref 3 betters the Ref 2 MKII in all those respects.

Also, as with the Ref 2, the Ref 3 uses a chip in its volume control circuitry. A number of flagship preamps out there use discrete solutions—including those by Aesthetix, Ayre, VTL and others. The REF 3 does better than hold its own with these designs, which says Bill Johnson and his crew must know something about getting the most out of such a design!

The Ref 3 is balanced from input to output, and all of its inputs can be selected as either single-ended or balanced. Unlike the Ref 2 there is no tape monitor, so you can't listen to one source and tape another. The AC inlet has been changed from a 15-amp connector to a 20-amp connector.

Fitting with its 21st century flagship status, the Ref 3 is outfitted with discrete commands and is thus compatible with AMX or Crestron remote control systems, if that's your thing.

Sound

I wish that I had the experience to place the significance of the Ref 3 back through the decades, comparing it to all of ARC's past designs. (I'm not that old, but maybe Dick will write a companion piece to this review doing just that!) But I do know the Ref 2 and the Ref 2 MKII intimately. The former drove two systems I listened to years ago that brought me out of the digital cold and back into vinyl's warm glow.

How the Ref 3 stacks up against the Ref 2 and Ref2 MKII is unequivocal: the Ref 3 is the best yet. And it's something of a departure sonically. The Ref 2 preamps sounded magnificent, splashy and engaging in all the right ways. Music sparkled with these preamps. But compared to the very best, the Ref 2 preamps weren't as high in real resolution or detail, with the midrange just tipped a little forward.

Ref 3 trades some of ARC's trademark pizzazz for a quieter, more laid back and yet more sophisticated sound that's higher in resolution and contains much more musically significant detail. It's no less musical; it's just musical in a different way. Tonally, there's more foundation in the bass, a less forward but more revealing midrange and a very sweet, extended top end.

The Ref 3 isn't as flashy in a short demo as its predecessors, but is far more satisfying over long-term listening. It doesn't jump up and grab you with a *ta-da,* but as you sit and listen there's simply more music there. The Ref 2 preamps sounded like excellent hi-fi; the Ref 3 just sounds more like music.

When the Ref 3 came in, VTL's super 7.5 was my reference preamp. That design is nearly all solid-state, with two tubes in the gain stage. It's dead quiet, and dynamically very powerful. While I didn't find the Ref 3 to be equal to VTL's dynamic swing, it didn't lack in that area. The Ref 3 actually sounded quieter and lower in distortion in direct comparison. Related or not, the speakers are dead quiet when the ARC has its volume cranked and there's no music playing. Not even a soft hiss.

But getting away from these typical audiophile references, the ARC did things more immediately noticeable and important. I became instantly aware of each individual recording space as I changed from recording to recording in a more immediate and profound sense. I didn't have to think about it or cock an ear to listen for it, it just happened.

And "happen" is a word I like a lot in relation to the Ref 3. This is the first preamp that made me hear the VTL as what it is: a solid-state preamp with a pair of tubes in it. This is very subtle, but the REF 3 definitely sounded less mechanical and gave a purer sensation of music simply happening in the room. It didn't sound like a hi-fi, it sounded more like music and less like electronics.

Compared to the Ref 2 preamps, I'd also say that the Ref 3's imaging is superior, more three-dimensional and fleshed out in living space. The soundstage is deeper and wider and yet also more precisely drawn in between. It's spectacular in these regards, which I hold very dear as a listener.

I've made some points here that make it sound like I prefer the sound of this preamp to the VTL 7.5 and the fact is that I do. But that reads as more significant than it is. My system is ultrahigh in resolution, and I'd refer to my preference for the Ref 3 in these respects as just that—a reflection of preferences, not an absolute, qualitative judgment. In direct comparisons the differences in these preamps are very subtle and more than a few factors go into the VTL's favor. For instance, the VTL is superior in dynamic authority and extension at the frequency extremes. That usually results in greater overall transparency, but not in this case. I actually felt the ARC was more transparent to the recordings I played.

But my emotional response is far simpler. I responded more to the Ref 3 and, truth be told, I've been happiest with my hi-fi when I've had an all-tube preamp in the system, regardless of all other factors. And overall, when day-to-day use and performance are taken into the equation, there isn't a preamp I've enjoyed having in my system more than the Ref 3. But I'm not ready to wrap up just yet either. Tubes evoke a certain degree of fear and loathing among some people. Even for you diehard solid-state types, I wouldn't hesitate to recommend the Ref 3 as your first foray into tubes. There simply isn't any fussiness or downside to the tubes with an ARC. You don't hear the tubes, they don't fail very often, and when they do they're not overly exotic or expensive to replace. Bulletproof is what it is.

The Ref 3 uses four 6H30 tubes, and one each 6550C and 6H30P in the power supply. During the many, many months I had the Ref 3 one of the big tubes in the power supply went out. I replaced it and that was that. The tubes never got noisy, and in fact remained dead quiet, and the sonic picture remained unchanged to my ears over all those months. In short, you won't even know the tubes are there. But if you get curious, one of the nifty new pieces of info that the big, fluorescent front-panel display can give you is a readout showing the hours on the tubes. Tubes in the Ref 3 are rated for 4,000 hours and a complete set costs only about $300 to replace. I don't know of a more pain-free tube component to own than this.

Conclusion

The Audio Research Reference 3 is everything a flagship preamp should be. It has all the functionality a modern piece should have, including the ability to blend simply and seamlessly into a full surround-sound system. Yet, in spite of its complex feature set, setting up and using the Ref 3 is downright easy.

All of that would be mere window dressing if the Ref 3 didn't offer spectacular sound. Like the very best components that we've raved about here at **Audio Perfectionist Journal**, the Ref 3 transcends the terminology typically associated with equipment reviews and requires simpler, broader terms. The Ref 3 sounds more like not having a preamp in your system, while simultaneously making some of its excellent competitors sound merely like excellent hi-fi. It's breathtaking in how much musical information it reveals, and yet it's effortless and utterly relaxed in doing so.

I've heard preamps that have a little more of this and a little more of that. But none that's as complete, top to bottom, or as satisfying in its presentation of the entire musical picture, weaving each distinctive element of music into a coherent whole. The Ref 3 goes beyond hi-fi; it's a 21st century classic from ARC. APJ

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Last Thoughts

by Richard Hardesty

*After reading this **Journal** you might be asking yourself, Should I spend money on accessories or is the whole product category a big fraud designed to rip me off? The correct answer to both questions is a qualified yes. Well-chosen accessories can make subtle improvements in sound but no accessory can make a silk purse out of a sow's ear. But subtle improvements in sound are what high-end audio is all about. Be realistic about the potential for improvement and choose carefully.*

Don't spend any money on any accessory until you have assembled a balanced audio system from the best components you can afford. When you can't improve the components any more, squeeze the most performance from them with accessories. Listen to be sure an accessory actually provides an improvement to your sound.

If you expect an accessory to make a subtle refinement in the sound of an audio system, not a miraculous change, you are likely to be pleased with a few products offered for sale. APJ

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