AUDIO
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JOURNAL

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Source Components

*The **Audio Perfectionist Journal** has been published and distributed in electronic form since 1999. **Journal** #9 is the first one to be printed on paper and distributed by mail and some background information is called for.*

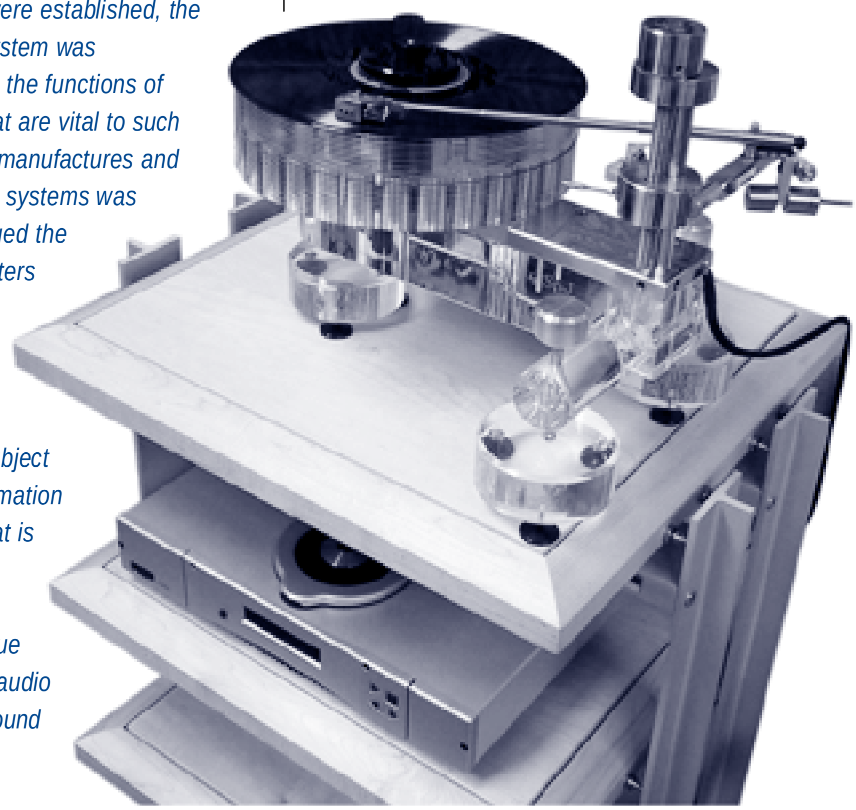
*The content in the **Audio Perfectionist Journal** is based on my high fidelity approach to home music reproduction. **Journals** #1 through #8 defined this comprehensive method for evaluating audio components and systems. Goals were established, the purpose of a home audio system was described and we examined the functions of some of the components that are vital to such a system. The industry that manufactures and sells audio components and systems was scrutinized, and we challenged the expertise of some of the writers who review and criticize the industry's products.*

***Journals** #9 through #12 will delve deeper into our subject of interest and provide information about audio components that is more specific and detailed. You need this information if you want to get the best value for the money you invest in audio components and the best sound from the components you choose.*

*Let's start by restating our primary goal and outlining some of the material that was introduced in previous **Journals.***

Emotion is a Good Thing

We have established that our primary goal is the emotional satisfaction that music can bring and we need to think about how to assemble an audio system that will accomplish this goal. We are confronted by a marketplace filled with a confusing myriad of audio products which are often proffered with hype and misinformation. Those who rely entirely on emotions to lead them to the components necessary to bring about musical satisfaction in the home may



experience years of frustration and financial loss. Wise enthusiasts will also enlist their intellect in order to ferret out those products that are well designed and fairly priced because the marketplace is filled with products that are neither.

We seek an emotional connection with music but we must think about the function of the components we choose in order to assemble an audio system. We don't want to waste money on inadequate or poorly designed hardware and we do want to fully exploit the capabilities of the good products we acquire.

Reason is a Good Thing, Too

The **Audio Perfectionist Journal** is a publication for those who respond emotionally to music and want to learn how to choose audio components which allow that response. There are many ways to go wrong and lots of advertising dollars available to lead you astray.

Music has life-enriching value. Some audio components convey the musical message better than others. Figuring out which components are best requires technical examination and experienced listening and the **Journal** can help you learn how to do both. We evaluate the quality of audio components by listening to them but we also understand how audio components work and we know how to gauge the competence of the designer. Knowledge makes it harder to fool us with spectacular but artificial sound or with impressive but false rhetoric. Remember, misinformation is just like real information but wrong.

> ***"Our primary goal is the emotional satisfaction that music can bring."***

A little knowledge can work for you too. Wouldn't you like to be harder to fool? You have to make the ultimate decision about which products to choose. The **Journal** will provide useful information that will help you avoid costly mistakes.

The Basics

Journal #1, available free on the **Audio Perfectionist** website (www.audioperfectionist.com), explains the philosophy of the high fidelity approach to home music reproduction and home theater. Some common myths are debunked and the basic goals of the **Journal** are established. **Journal** #2, which is also available free on the website, explains how the environment affects the sound you hear and suggests the best ways to improve listening room acoustics and the best ways to position the speakers and the listener within that room. **Journal** #3 begins the discussion of system components that continues through Journal #8.

The Natural Order of Things

The article *The Natural Order of Things* in **Audio Perfectionist Journal** #3 follows the signal path through a home audio system and divides the system into sections based on component function. There are three major component groups, or system sections: source components, which provide information retrieval; amplification components, which raise the signal level; and output components, which convert that signal into sound. Cables are an additional component group that will be considered separately.

Each of the **Audio Perfectionist Journals** #9 through #12 will concentrate on a system section, or component group. We'll examine each section of an audio system in greater depth while discussing the components that provide the specific functions of information retrieval, amplification and reproduction. This journey will begin with **Journal** #9 which is all about source components.

Source components are often undervalued by novices but the source component is vitally important because it sets the limits of resolution for the entire system.

Examining an Audio System, Section by Section

Following the signal path and grouping components based on function makes it easier to understand what each component does and how each individual component relates to other components and to the system as a whole. Understanding these concepts is important because it helps us to properly allocate resources in order to assemble a balanced audio system that provides maximum performance for investment. A lack of understanding can lead to frustration and financial waste.

If you believe that speakers can reproduce a signal that doesn't come out of the amplifier, or that the amplifier can amplify a signal that didn't come out of the preamplifier, you will have a difficult time properly allocating your resources while assembling an audio system that can provide satisfying sound. If you forget that the source component establishes the limits of resolution for the entire system you may spend years and untold sums of money futilely trying to reproduce a signal that was never retrieved from the recording.

Many people spend too much on output components and too little on amplification and source components. They believe that speakers make the biggest difference in sound so they spend most of their budget on speakers and shortchange the rest of the system. The result is likely to be unsatisfactory at best. It is common to see systems that include $10,000 speakers reproducing sound sourced from a $600 CD player. It is common and it's stupid. Factual, logical information can help you avoid costly and frustrating mistakes when choosing components for an audio system or deciding which component to upgrade first in the system you already own. Unfortunately, hype and ridiculous claims are often substituted for facts and logic. We'll try to alert you to claims that are over-the-top or just plain false and present logical arguments instead.

Let's begin with some simple, irrefutable facts.

Truth and Logic

Here are some inarguable truths and some simple logic: Speakers reproduce the signal that is delivered by the amplifier. The amplifier amplifies the signal that is delivered by the preamplifier. The preamplifier processes the signal that is delivered by the source components.

No system component can create a musical signal. Each component is limited to reproducing the signal that is presented at the input to that component. The source component retrieves the signal from the recording medium, establishing the limit of resolution for the entire system.

If an optical disc player or turntable system doesn't retrieve the signal from the disc or record, it's lost forever. The preamp can't process a signal that doesn't exist, the amplifier can't amplify that signal, and speakers—no matter how expensive or exotic—can't reproduce that signal. If you want high quality sound to come out of your audio system you must put a high quality signal in.

Of course each component can add undesirable artifacts like noise and distortion to the desired signal, as described in **Journal** #3, and those undesirable additions must be minimized for satisfactory results.

Analog and Digital

Today our audio systems can play music recorded on analog media like magnetic tape and vinyl records, or music that has been digitally encoded and stored on optical discs. This

"The source component retrieves the signal from the recording medium, establishing the limit of resolution for the entire system."

Journal will provide basic information about analog and digital signals and the components that are used to retrieve those signals from the recording. We will offer suggestions, based on our experience with many components, for achieving the best possible performance from analog and digital sources.

We, of course, haven't heard every product available and information will be provided to help you judge components that aren't mentioned. Products that aren't specifically mentioned in this **Journal** are not necessarily bad (or good). We try to refrain from commenting on components with which we have little personal experience but that doesn't mean that you shouldn't investigate them. A lot of stuff is available for your consideration and we can evaluate only a small percentage of it.

The Current State of Digital

Digital audio has evolved and new formats like SACD and high resolution LPCM (available on some DVD-Video and DVD-Audio discs) provide hope for even better sound. That doesn't change the fact that the audio industry has languished for over 20 years while we have tried to create a silk purse from the sow's ear that is the compact disc. Today's best 16-bit/44kHz recordings played on today's best CD players provide sound that is hard to fault in hi-fi terms. Analog records and SACD discs offer more.

Analog records and SACDs make it easier to become emotionally connected to the music. This emotional connection is the hook that perpetuates the high-end audio industry and perhaps the music business as well. Both are in a slump and I can't help but think that the compact disc is at least partly responsible.

Today's best CD players are barely adequate as a source for a high resolution audio system and decent CD players are still quite expensive. An inexpensive turntable or SACD player will convey the musical message far better than the best CD players available. Hi-fi buffs may never understand this but listeners seeking an emotional response from music will "get it" with their first exposure to good analog or SACD.

I'm not going to tell you to throw away your CD collection and I'm not going to part with mine. But you need to know that you are working harder to achieve emotional satisfaction if compact disc is your primary source for music. The contents of this **Journal** will present a strong case for that position in the hope that people will not simply settle for the mediocrity of CD and will seek superior formats which are coming to market now and, in the case of analog, have always been available.

This **Journal** won't be devoted exclusively to criticizing the compact disc. We will tell you how to get the best sound possible from your CD collection. We'll review an array of players and tell you which ones are worthy of your consideration and which ones aren't. We'll compare the sound from CDs to the sound from high resolution digital formats like SACD and high res LPCM and we'll compare players compatible with these formats.

The Current State of Analog

The general public may think that vinyl records became obsolete in the early 1980s when the compact disc arrived. This point would be hard to argue based on sales numbers. Analog records have never completely vanished from the market but they represent a small niche market much like high-end audio. These niches are of interest to only a small number of discerning music lovers. The fact that records won't die demonstrates a lasting appeal that can only be attributed to sound quality.

Vinyl records are a pain in the butt. They are easily damaged and need to be cleaned and carefully stored. Some surface noise is audible on most pressings and a high percentage of new records are imperfect in some way.

You can't listen to a record while riding in your car or jogging. Records must be played on delicate mechanical devices that are microphonic and easily damaged during house cleaning. These devices (turntables) must be carefully adjusted after installation and meticulously maintained. What a hassle.
Why would people put up with this "primitive" format in a digital age? Because records sound better. The payoff for all the trouble is sound that is alive and deeply satisfying. Give records a try and see if you don't agree.

Source Components

This Journal is all about source components. We'll start with analog and finish with digital. Before we begin our examination of source components I'd like to acknowledge some contributors who helped make this **Journal** possible.

New Members on the Journal Team

Shane Buettner has joined the **Audio Perfectionist Journal** as Equipment Review Editor and he has contributed a large volume of work, including many product reviews, to this issue. Shane is an experienced listener and journalist whose articles and reviews have been published in *Widescreen Review*, *The Perfect Vision*, and *The Absolute Sound* magazines. Shane and I share a common vision about how audio publications should serve consumers and we have been working closely together for several years.

Rick Johnson has joined the **Journal** as Art Director. Rick's artistic skills will make this a more attractive and professional publication that is better able to accurately convey information. We'll now be able to print photographs of products and people along with drawings that will help to clarify complex subjects that are difficult to describe with words alone.

I'd also like to express appreciation to the other contributors to **Journal** #9: Joe Harley of Harley Music Productions, Brooks Berdan of Brooks Berdan Ltd., and Chris Fitzgerald, a **Journal** subscriber. Edith Hardesty has edited the thousands of words of copy to correct our spelling and grammatical errors, as she has done since the beginning. APJ

ANALOG

by Richard Hardesty

An analog signal has continuous physical variables as opposed to a digital signal which is composed of sequential samples that represent numerical quantities at an instant in time. An analog recording of music contains a continuously variable signal that is directly analogous to the air pressure variations, or "sound waves," at the original event. A digital recording of music contains a digital code consisting of sequential samples that represent the amplitude of the analog signal at specific instances in time or the change in amplitude between one sample and the next.

Recordings are created by converting mechanical energy to electrical energy and storing a representation of that electrical energy. During playback electrical energy is converted into mechanical energy to produce sound.

Transducers

Energy conversion is performed by electromechanical transducers, including microphones, cutting heads and loudspeakers. Several conversions may take place—for vinyl records, sound pressure is converted to an electrical signal at the microphone; the electrical signal is converted to mechanical energy at the cutting head; mechanical energy from the record groove is converted to an electrical signal in the pickup cartridge; and electrical energy is converted to mechanical energy by the loudspeakers. While electromechanical conversion occurs during recording and playback, the signal on an analog recording is always a direct analog of the sound pressure and is continuously variable rather than sampled.

Tape and Vinyl

Analog sources available for high-end audio enthusiasts include analog tape and vinyl records. Both are becoming increasingly less common. Analog tape decks for home use are almost extinct and we're not going to spend any time discussing them. We won't ignore analog tape completely because many high quality recordings, even those that end up on a digital medium like CD, are initially captured on analog tape.

Vinyl records, which have traditionally been our highest resolution source, are making a strong comeback in our niche market and we will devote a substantial portion of this **Journal** to an examination of analog turntable systems so you'll know how they work and how to choose a good one should you decide to see what all the fuss is about. I hope that you do decide to investigate records.

Analog Tape

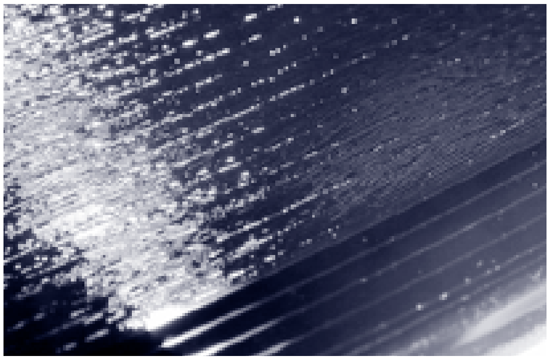
An analog tape recording stores energy in a varying magnetic field created by magnetizing ferrous (containing iron) particles which are bonded to mylar tape. These particles pass by a recording head which magnetizes them using a high-frequency bias current along with the electrical signal that represents the musical information captured by the microphone. During playback an electrical signal is produced in the playback head by the passing magnetic field on the tape. This electrical signal is amplified for conversion back to mechanical energy by the loudspeaker. A tape recorder stores information magnetically utilizing record and playback heads containing coils of wire which become electromagnets for recording and generators for playback.

The record head becomes a variable electromagnet when the signal and bias current flow through its coil. The playback head becomes a generator when its coil is exposed to the variable magnetic field from the ferrous (iron) particles on the recorded tape.

An analog tape recording stores energy magnetically. Energy is stored on vinyl records by physically carving it into the groove walls of a master disc and transferring those groove wiggles to a vinyl record.

Vinyl Records

Vinyl records store the recorded signal in the wiggles of the grooves. The inner groove wall contains the signal from the left



channel and the outer groove wall contains the signal from the right channel. The groove walls have a 90 degree included angle and the stylus moves at ±45 degrees from vertical in response to a signal that occurs only in one channel or the other. If equal amplitude in-phase signals are recorded in both channels the playback stylus will move laterally. Equal amplitude out-of-phase signals will produce vertical stylus movement

A cutting lathe is used to make records which can be created from analog or digital tape recordings or directly from the microphone feed—"direct to disc." The cutter head converts the electrical signal into mechanical motion and carves a groove in

the lacquer coating on the surface of the aluminum master disc. A vinyl record is produced from this master disc (several intermediate stages are involved). During playback, the pickup cartridge converts the groove wiggles into an electrical signal that can be amplified and reproduced as sound.

The process is tedious but uncomplicated and the music remains intact all along the way. The master disc and the stampers which impress the signal on the vinyl discs can be played. (Stampers require a special stylus because a stamper is like a photographic negative—grooves become ridges.)

When vinyl records are made the signal is equalized during recording and playback to limit groove spacing and to minimize surface noise. Low frequencies are reduced in amplitude and high frequencies are increased in amplitude during recording by applying the RIAA equalization curve. An inverse EQ curve is applied during playback where bass frequencies are raised in level and high frequencies are reduced in level to return the signal to flat response.

Vinyl records can have a useful bandwidth of 10Hz-30kHz with excellent phase response. Musical dynamic range is audibly superior to CD even though signal-to-noise ratio numbers might lead slide rule jockeys to believe otherwise. A playback stylus in intimate contact with the groove walls can retrieve recorded information that is remarkably subtle. Listen and compare CD and analog versions of the same recording and hear just how much micro detail—most notably instrumental decay and high frequency "air"—is missing from the CD yet clearly audible on the record. Or attend an **Audio Perfectionist** seminar and I'll demonstrate this for you.

The Analog Difference

If you find it more difficult now to respond to music with the degree of enthusiasm you had in your youth, or if you are a youth and never completely understood those who talk of a deep emotional connection to music, try listening to vinyl

The sound from records is so much more "alive" and involving.

records. They don't have to be new audiophile recordings. In fact, it's probably better if they aren't. Open yourself to this experience and you may find something that's been missing in your life. Let me tell you about what happened to me.

When I advanced into middle age and left the retail audio business, I got lazy. My turntable (a Linn LP-12 circa 1980) needed a new belt and I had a state-of-the–art CD player (a Wadia 860, now an 861). I had duplicated many of my favorite records with CD copies of the same recordings. The CD sound was impressive and everything was sooo easy. No disc cleaning, no stylus cleaning, no hassle. And I could play CDs in my car.

Even though it was now more convenient to select and play a disc, I found that I was listening to my audio system less and less. When I played music that, in years past, had produced a deep emotional response, the sound seemed spectacular and hard to fault (in hi-fi terms) but the goose bumps no longer appeared. I decided that I just had to accept a less thrilling and less satisfying involvement with music—just as I had accepted other diminutions of excitement and joy in my life as a “natural” part of growing older.

CDs were supposed to provide “perfect sound forever.” I had listened to dozens of CD players and I knew that I had one of best available.

The Road Back to Analog

When I started to review audio products and write about audio subjects I needed a complete reference system. I contacted my old friend Ivor Tiefenbrun at Linn in Scotland and asked for the parts to bring my antique LP-12 up to date. I upgraded the turntable to Cirkus status and added a Lingo power supply and the latest Ekos tonearm and Arkiv cartridge. When the turntable system was assembled and operational, I sat down for what turned out to be an ear-opening experience.

I played a bunch of 20-year-old records that used to “get me off” in the old days and tears came to my eyes. The thrill wasn’t gone, after all, or even diminished. In fact, because age and experience have made me less self-conscious and more self-assured, I could really let myself go and I luxuriated in the music like a pig in a mud bath.

The sound from records is so much more “alive” and involving. The rhythm and pace of the music are easier to follow and respond to. (I have no technical explanation for how CDs can have a negative impact on rhythmic integration but I have demonstrated this phenomenon for hundreds of people with universal success.) Records make it easier to connect with the music and the musicians.

Emotions restored

Except for the reintroduction of the turntable, all components in my system remained unchanged. Many of the records that I had chosen for this first listening session were the analog versions of the same recordings that had been emotionally unsatisfying on CD. The excitement wasn’t gone—it had simply been stifled by inadequate digital coding.

I spent the decade of the ‘80s, while I was still selling audio systems, trying to convince people that the CD was a step backwards from the vinyl record, but in the ‘90s I had given in to the convenience of what, by then, had become a much improved digital recording and playback system. CDs are better now since the dark ages of the 1980s but they’re still not good enough.

After returning to my turntable as a source I have been reborn as an analog druid. Although I have an outstanding CD-based audio system, records simply sound better. Vinyl delivers the musical goods like CD never could.

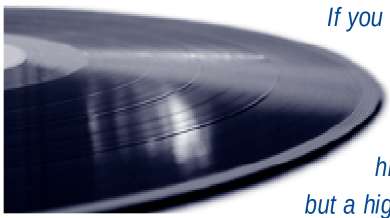
This was not a momentary experience and I continue to choose analog as my preferred source whenever possible. Records deliver music that is more rhythmically involving and more emotionally gratifying. The message of the composer and the skill of the artist are far easier to perceive when the source is vinyl. I have a lot of compact discs and I continue to listen to them but I buy CDs only when a vinyl or SACD version of a recording is not available. APJ

Records Need Record Players

To play vinyl records you need a turntable, tone arm and cartridge (along with a phono preamp—to be covered in Journal #10). Before choosing these components you should know a little about the physics involved. The following articles will provide necessary information.

Turntable Physics

by Richard Hardesty



If you want to play records you need a record player. Record players are called turntables in high-end audio language but a high-end "turntable" is actually a system with three major components: the turntable mechanism, which rotates the record; the tonearm, which caries the pickup cartridge over the record surface; and the pickup cartridge, which generates electrical output.

The job of the turntable system seems deceptively simple but there is more to it than meets the eye. A high-end turntable system must retrieve signals from the record grooves that can't be seen without the aid of a microscope. It must retrieve these tiny signals while both resisting the powerful energy that bombards the system from the outside and dissipating disruptive energy created within the turntable system.

Disruptive energy from within the system includes vibration from the motor and main bearing and excitation of the tonearm tube by the stylus motion in the groove. Disruptive energy from outside the system includes footfalls and other mechanical disturbances, and acoustic energy from the loudspeakers. Acoustic energy from the speakers may enter the turntable system from below through the support furniture or directly through the air into the turntable components, particularly the light weight tonearm tube.

The turntable motor is the only active component in the system. The turntable platter, which supports the record, spins freely on the main bearing and is driven by the motor. The quality of the bearing which supports the platter is critical to performance. Tonearms and cartridges are passive components dependent on the turntable motor for function. The motor drives the platter which drives the groove. The groove drives the tonearm and the stylus.

The tonearm must position the cartridge accurately above the record in three dimensions and act as a sump for energy from sources including the speakers and the vibration from the stylus that is tracing the grooves.

The cartridge generates output when there is relative movement between the cartridge body and the stylus. Ideally the cartridge body will be held motionless above the record while the stylus tracks the groove, but if the stylus is motionless and the cartridge body is moved by external energy which enters the system, or by arm tube resonance reflected back to the cartridge body, a signal will be generated which is unrelated to the recorded information.

We'll discuss the three system components in order of their significance to sound. First the turntable, then the tonearm and finally the cartridge. While this order of sonic significance may seem counterintuitive, experience will prove its validity.

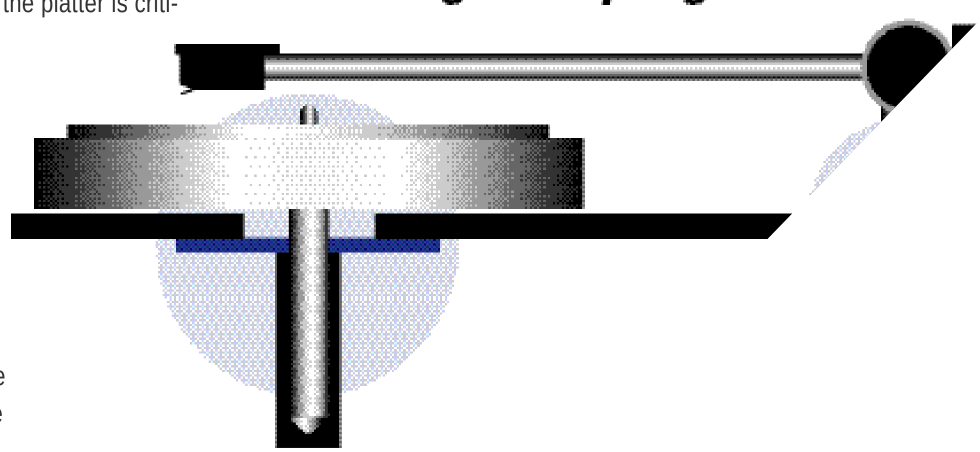
The Turntable

The turntable has a motor which powers the entire system, a rotating platter that supports the record, a main bearing assembly that supports the rotating platter, and a chassis that supports the main bearing and the tonearm and couples the two together over a wide range of frequencies (ideally). This coupling is vitally important because it minimizes the effects of vibration on the retrieval of information. Here's how.

Coupling

If the turntable subchassis and tonearm assembly provide rigid coupling between the main bearing and the pickup cartridge, noise and vibration entering the system will not cause relative

Rigid Coupling



movement between the stylus and cartridge body because that noise or vibration will appear everywhere within the coupled system, in phase.

If the cartridge body is decoupled from the main bearing any noise or vibration that enters the system may cause relative movement between the stylus and cartridge body which will affect the retrieval of information or create a signal that is unrelated to the recording. Bearing noise would appear at the platter and also at the cartridge body in an ideal coupled system and produce no output. Bearing noise would only appear at the platter in a decoupled system and would be interpreted as a signal by the cartridge.

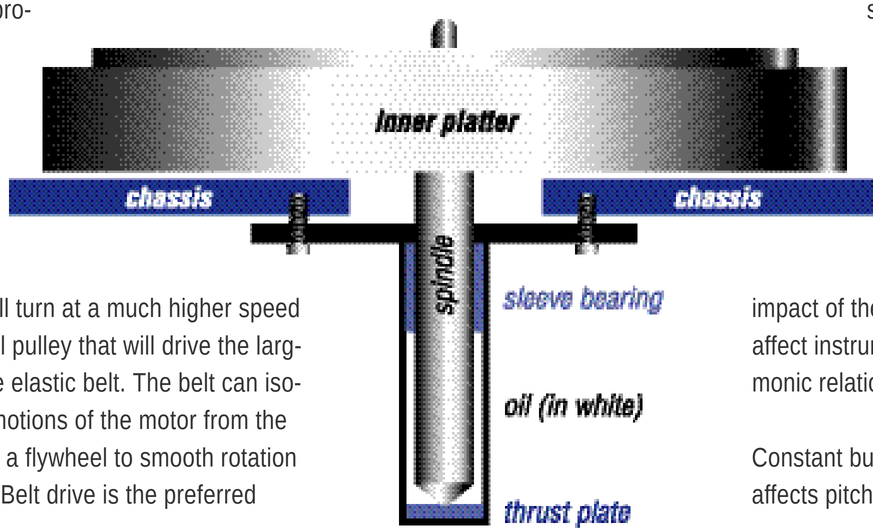
Decoupling can occur at the main bearing-to-chassis junction, at the tonearm-to-chassis junction, at the tonearm support-to-tonearm junction, at the tonearm-to-cartridge junction, etc.

Perfect coupling at all frequencies cannot be achieved in the real world. Energy that cannot be coupled in phase must be absorbed and dissipated rather than reflected. Mass isolated turntables attempt to direct energy into massive components that act as sumps in order to prevent this energy from being reflected back into the system.

The Motor

An electric motor rotates the platter. The motor can be directly attached to the platter—direct drive—or a detached motor can rotate the platter using a belt or string drive. Motors rotate with sequential pulses (in small jerky motions) as poles are energized. Fewer poles mean fewer, stronger pulses (jerks) per revolution. More poles produce more jerks with less energy per jerk.

A direct drive motor must turn at platter speed. A belt drive motor will turn at a much higher speed while rotating a small pulley that will drive the larger platter through the elastic belt. The belt can isolate the small jerky motions of the motor from the platter which acts as a flywheel to smooth rotation and stabilize speed. Belt drive is the preferred drive method for high-end audio but not the only one.



Motors turn with sequential pulses which are smoothed by the flywheel action of the platter and the elasticity of the belt. As the flywheel gets heavier the motor power must be increased. The more powerful the motor the stronger the pulses. A motor with just enough power to start the platter is usually the best choice for smoothest rotation.

In general, DC (direct current) motors are voltage controlled devices and AC (alternating current) motors are frequency controlled devices. DC motors facilitate continuously variable speed contol, which is useful for turntables used by disc jockeys. AC motors are as speed accurate as the frequency of the alternating current that drives them and don't need servos or sensing devices. DC motors use servo speed control systems in virtually all commercial turntable designs. Servo-controlled systems are, by definition, always turning at the wrong speed. Servos can actually create flutter distortion at the sample rate of the servo. Heavier platters smooth the servo action into a more stable rotation. Heavier platters require more powerful motors which create more powerful pulses.

The best turntables I've heard use AC motors and belt drive systems and have platters that are moderately but not excessively heavy.

The Platter

The turntable platter must support the record and rotate at a constant and correct speed. Slow variations from the correct speed are called "wow" and rapid variations are called "flutter." Wow and flutter are descriptive names that accurately convey the negative sonic impact of these distortions. Both affect instrumental pitch and harmonic relationships.

Constant but incorrect speed also affects pitch and harmonic integration. The turntable must rotate at the correct speed to accurately reproduce music. Speed can be verified with a strobe disc or by playing a test tone of known frequency and using a frequency counter or oscilloscope to verify the frequency that is reproduced.

A heavier platter will provide more flywheel effect than a lighter platter but the heavier platter will create far stronger forces in the main bearing and will require a more powerful motor that will introduce stronger pulses (jerks). Heavier is better up to a point.

Clear plastic platters look cool but provide no other benefit that I can determine and they may increase static electric charge because they are non-conductive. A metal platter can be electrically grounded to minimize static build-up in the record. A metal platter can be electrically grounded to the tonearm.

The Main Bearing

The turntable platter rotates on a main bearing that must prevent wobble and create little noise. Low frequency noise, which is created primarily in the platter bearing, is called “rumble.” Rumble is a very destructive type of noise because it occurs at low frequencies which can use up large amounts of amplifier power and speaker excursion, producing intermodulation distortion and reducing headroom and dynamic range.

Every imaginable bearing type has been tried and the vast majority of commercial turntables use a simple sleeve bearing assembly with a thrust plate, which has proven to be the best and longest-lasting solution.

Oil, which is essentially non-compressible, can fill the gap between the spindle and the sleeve bearing providing lubrication and rigid coupling. An oil-bath bearing can be virtually free of wobble. Air bearings rely on air, which is a compressible substance, to support the platter and reduce friction between the spindle and the sleeve and thrust plate. Air bearings increase complexity and reduce coupling while providing no audible benefit in my opinion.

The best turntables I’ve heard use oil-bath sleeve bearings. Inexpensive turntables may effectively use dry bearings that will suffice for several years.

Suspension

Two basic methods are used to isolate turntables from external energy and minimize acoustic feedback. One method utilizes a spring suspension system that acts as a low-pass filter blocking vibration at frequencies above the resonant frequency of the suspension. The other method relies on mass. Because little engineering is necessary, mass isolation is more popular with small manufacturers.

Spring suspension systems are generally tuned to resonate at 3-4Hz. These suspensions are very effective at blocking vibration above 4Hz or so from entering the turntable system through the supporting furniture below. Because they act as low-pass filters they are ineffective at frequencies below 3-4Hz and they make the turntable system very sensitive to footfall disturbance. In a home with wood floors a spring suspended turntable should be supported by a shelf mounted on a load-bearing wall. A suspended turntable should not be placed on an isolation platform that includes another compliant system. Two compliant systems may interact with unpredictable results.

Spring suspension can’t isolate the turntable from airborne energy which enters the turntable system directly rather than through the supporting furniture. Rigid coupling in the suspended subchassis, which connects the platter through the main bearing to the cartridge through the tonearm, is an effective defense against airborne vibration.

A closed, coupled system will minimize interference from airborne energy. Rigid coupling only works at some frequencies. It’s easy to couple low frequency bearing noise to the tonearm but it’s far more difficult to couple high frequency vibration from the tonearm to the platter. Even rigidly coupled, suspended turntables must absorb and dissipate some energy at some frequencies.

Mass isolation turntables rely on high mass as a defense against external energy and acoustic feedback. Manufacturers who use mass isolation simply make everything bigger with the idea that the bigger and heavier it is, the harder it will be to move.

Mass isolation is only partially effective. Bigger, heavier objects are more difficult to set in motion but objects with more mass tend to store energy at lower frequencies than lighter objects. A heavy platter is unlikely to be moved by acoustic energy or footfalls but a heavy chassis may store energy at low frequencies and dump that energy into the tonearm which must have relatively low mass. Rigid coupling is more difficult to achieve with high-mass components and any vibration in the main bearing is likely to create a signal in the cartridge. Acoustic energy will easily affect the tonearm tube but not the rest of the turntable system and undesirable energy that moves the cartridge body is just as bad as undesirable energy that moves the stylus.

The best turntables I've heard use a spring-suspended subchassis that rigidly couples the platter to the cartridge.

The Arm

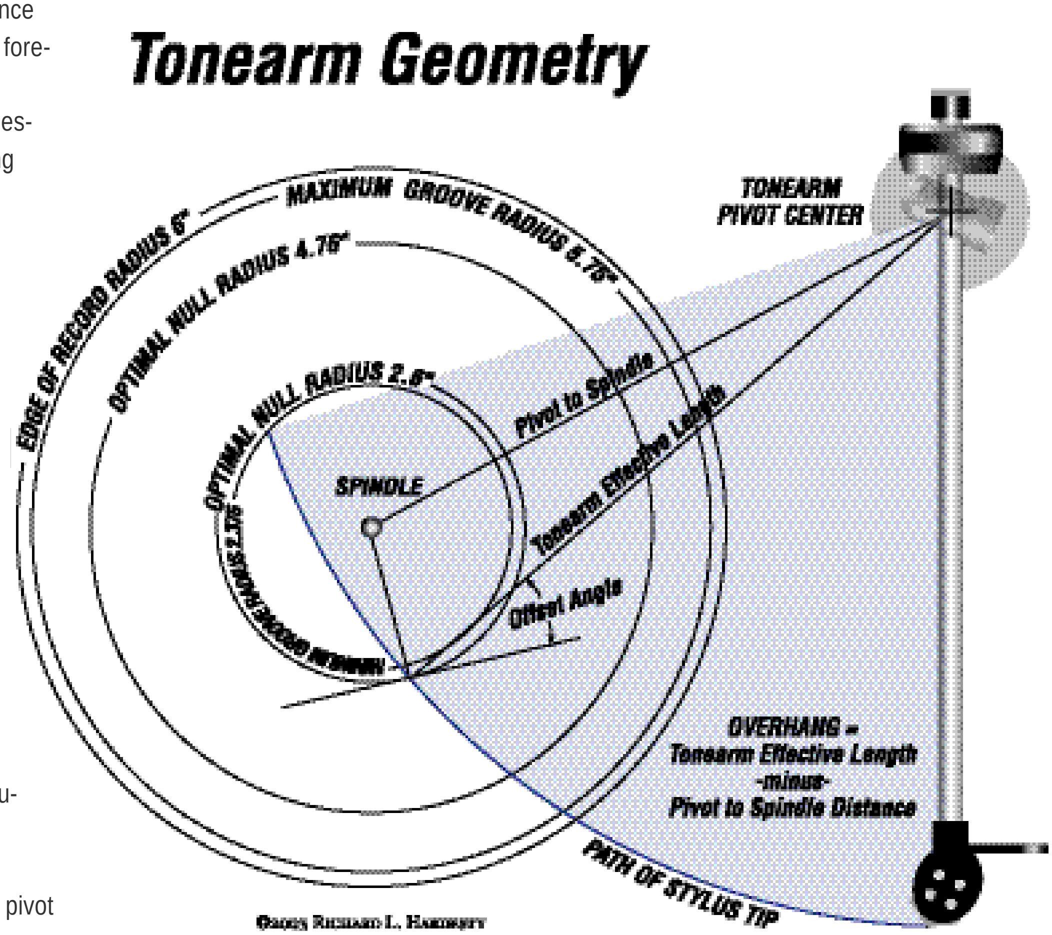
The tonearm positions the cartridge above the record surface in two planes and three directions: vertically (up-and-down), horizontally (side-to-side) and fore-and-aft (front-to-back). The arm must have low inertia and low resistance to up-and-down and side-to-side movement so that the cartridge can track the grooves on a warped record, but the arm must firmly position the cartridge at a specific distance from the pivot and prevent any fore-and-aft or twisting movement. Bearings are used to allow necessary movement while preventing undesirable movement.

The best tonearms I've heard utilize precision ball bearings for horizontal and vertical pivots but unipivot arms are also common. Unipivot bearings are much easier and cheaper to make and require little adjustment for assembly but they provide inferior coupling and allow undesirable movement. Precision ball bearings are costly to produce and assemble and can be easily damaged but provide better coupling and rigidity.

A unipivot bearing has a single pivot point that allows both vertical and horizontal movement (as well as twisting movement that is undesirable). There are only two parts to a unipivot bearing: a shaft with a point at the end and a cup in which this point rests. Gravity holds the parts together and, because the moving part of the tonearm must be relatively low mass, coupling that relies on gravity is compromised. Vertical arm movement when tracking a warped record will affect cartridge azimuth in a tonearm with an offset headshell and a unipivot bearing. (See illustration.)

Precision ball bearings provide rigid coupling while allowing free horizontal and vertical arm movement and preventing twisting or rocking movement which is undesirable. Vertical bearings can be offset at the same angle as the headshell so that azimuth remains constant throughout vertical arm movement.

Properly designed tonearms can track the record with a minimum of angular error and will have zero tracking error at two spots (called "null points") on the playable surface of the record



when geometry is adjusted correctly. Proper geometry requires optimum offset angle and stylus overhang.

The effective length of the tonearm is the distance between the horizontal pivot and the stylus. Overhang is the difference between the effective length of the tonearm and the pivot-to-spindle distance. Offset angle is the angle between the cartridge body and the tonearm tube.

For an arm of a given length, there is an optimum offset angle and an optimum overhang distance to produce minimum tracking distortion. These figures were determined using mathematical analyses by H.G. Baerwald and others. These settings can be adjusted only with precision instruments.

Bias compensation, which is commonly called antiskating force or antiskate, is necessary because of cartridge offset. As the stylus tracks the grooves, friction pulls the cartridge in the direction of record rotation. Because the cartridge is mounted at an angle to the tonearm tube the resulting force pulls the tonearm inward towards the center of the record. Bias compensation is the application of a compensating force which pushes the arm outward away from the center of the record.

Proper bias compensation should result in equal stylus pressure on the inner and outer groove walls which will center the moving elements within the cartridge. This will minimize mistracking caused by unequal pressure and distortion caused by the asymmetric positioning of the generator within the cartridge. Because friction varies with groove modulation, bias compensation is always a compromise.

Vertical tracking angle (VTA) is the angle formed between the cantilever and the record surface. Stylus rake angle (SRA) is the difference between vertical (90 degrees from the record surface) and a line extending from the stylus contact lines. SRA is what we are actually adjusting when we listen for focus. VTA changes a little when we adjust SRA but VTA is determined largely by cartridge design.

All tonearms should allow adjustments for stylus pressure, bias compensation and VTA/SRA. Adjustment may be provided for azimuth, which shouldn't require adjustment if the arm is made properly.

Rigidity is often sacrificed for adjustability. Tonearms that allow easy adjustment of SRA, for example, can't provide the rigidity and coupling of arms with locking height adjustments, using three contact points, that require tools for SRA adjustment.

Tangential tracking tonearms are supposed to eliminate tracking error. They are supposed to provide better sound by more accurately mimicking the movement of the cutter head. Tangential tracking tonearms create many problems in an attempt to solve the tracking error "problem" which is a red herring in my opinion. (A red herring is a distraction from the matter at hand, which in this case is a "problem" that really doesn't require a solution.)

Well-designed pivoting arms actually have very small angular errors over the normal playing surface of the record and zero tracking error at two places (null points) within this range. Pivoting arms can have substantially less angular error in the vertical plane than tangential arms which are usually shorter, and pivoting arms can be much more rigid and can provide much better coupling than tangential arms, which must move sideways on a carrier assembly.

There are two types of tangential tracking arms: active arms, which are motor driven, and passive arms, which are propelled by the stylus. Both types have a carrier mechanism that transports the arm tube and vertical pivot across the playable surface of the record.

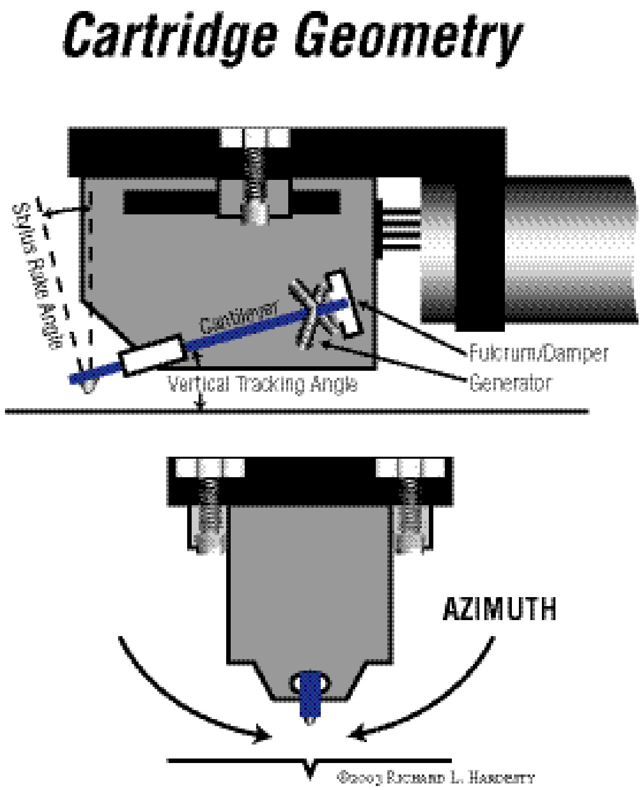
Active arms require some kind of servo mechanism to activate the motor that drives the arm carrier. They must move out-of-tangent with the groove in order to trigger this servo. The result is a sort of dog-tracking movement that may actually create greater angular errors than what would occur with a well-designed pivoting arm.

Passive tangential tracking arms must have very low resistance to lateral movement in order to prevent stylus deflection. They typically use air bearings or roller assemblies to allow the carrier assembly to be propelled by the stylus. Both provide poor coupling.

The best sounding tonearms I've heard are rigid, pivoting arms with precision ball bearings and non-removable headshells. Tangential tracking tonearms offer inferior performance with added complexity, in my opinion, and they make playing records a chore.

The Cartridge

A phono cartridge has a stylus to trace the grooves; a cantilever to conduct the stylus movement to the generator; a generator to produce electrical output; a suspension system, which



acts as both a fulcrum and a damper, to allow the stylus and generator to move; and a body to hold the moving and fixed parts in precise alignment and allow the assembly to be firmly attached to the tonearm.

Stylus

The cross-section shape of playback styli has evolved from conical to elliptical to modified elliptical—called line-contact or some trademark name that means line-contact.

The stylus makes contact with the groove walls only at the sides. The two small areas where the stylus touches the groove walls are called contact patches. As stylus shapes have evolved the contact patches have become longer top-to-bottom and narrower front-to-back. Reducing the front-to-back dimension of the stylus contact patches raises the bandwidth of the system by allowing the stylus to accurately track smaller groove modulations that are closer together—representing higher frequencies.

Cantilever

The cantilever connects the stylus, which is mounted at one end, to the generator, which is usually mounted close to the other end. Moving coil cartridges have generating coils, which are positioned in a magnetic field, attached to the cantilever. In a moving magnet or moving iron cartridge, the cantilever is connected to a magnet or iron armature which is positioned near fixed coils. Generator movements will closely duplicate stylus movements.

Suspension

The stylus and generator are held in position by an elastomer suspension that acts as a fulcrum for cantilever movement and a damper for resonance control (in most designs). This suspension component establishes the compliance of the cartridge.

Cartridge Body

The cartridge body contains all the fixed and moving parts and precisely positions them in relation to each other. It has slots or holes that allow the assembly to be securely mounted to the tonearm with nuts and bolts. Cartridge bodies made from resonant materials, like wood, can add euphonic colorations to the sound which some people enjoy.

Loading

The sonic characteristics of phono cartridges can be modified by electrically loading them with appropriate impedance. Moving magnet/moving iron cartridges can be tuned with capacitive loading, and moving coil cartridges can be tuned with resistive loading.

In my experience, most moving coil cartridges perform best when loaded with an impedance between 100 Ω and 500Ω. The lower end of this range seems to provide the most electrical damping and the higher end of the range provides the most dynamic sound. Some cartidges are designed to work into higher impedances up to 47k Ω.

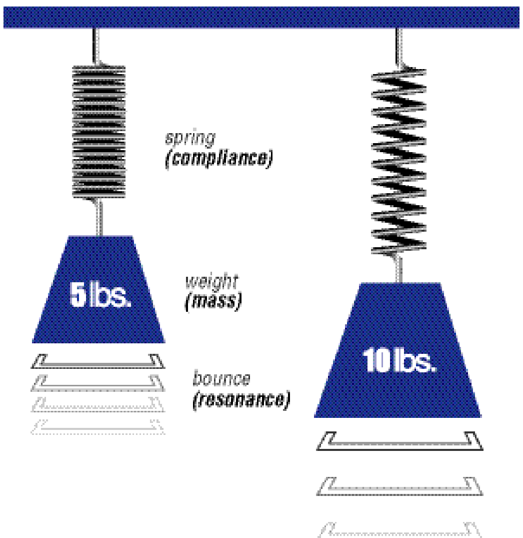
Mats

What do you call a guy who lies on the front porch? Mat. What do you call a 12-inch diameter flat thing with a hole in the middle, made from common materials costing mere pennies, that sells for ridiculous sums of money? Turntable mat. Do mats make a difference? Yes. Is this difference always an improvement in sound quality or accuracy? No.

Your ears must guide you through the turntable mat hype but remember these facts: unless the mat is thinner in the label area where records are thicker, it won't even touch the playable surface of the record. A mat that is the same thickness all the way across will support the record only in the label area and at the outer edge—the playable surface will be hanging in thin air. A soft rubber mat without a record clamp will decouple the record from the platter. If you have a ringy glass platter this may be desirable. If you have a well-coupled suspended turntable it probably will be undesirable.

Resonant Systems

A number of resonant systems have been described in this article. Resonant frequencies must be carefully positioned for maximum turntable performance. These include the suspension resonance, the tonearm/cartridge resonance, and the cantilever resonance.



Suspension Resonance

Spring suspension systems should bounce (resonate) at 3-4Hz. This is about as low as you can go while retaining practical stability. A suspension with a resonance of 3-4Hz will block most frequencies above 4Hz providing good acoustic isolation, but won't bounce too far or too long as it might with a resonant frequency lower than 3Hz. The resonant frequency of the suspension is established by the mass of the suspended parts and the stiffness of the springs and is not user-adjustable in most designs.

Cartridge/Tonearm Resonance

The moving parts of a phono cartridge include the stylus, cantilever and the generator element, which may be a tiny piece of magnet or iron or a set of moving coils. The moving parts are suspended by an elastic fulcrum/damper. The compliance of the cartridge suspension and the effective combined mass of the cartridge and tonearm will establish one of the most important resonant frequencies in the turntable system. This frequency should be as low as possible while remaining above the resonant frequency of the suspension and most warps.

The cartridge/tonearm resonance is very important because it sets the low frequency limit of the turntable system and determines the character of the bass to some extent. This frequency is established by selecting a proper match of cartridge compliance and tonearm mass and can be fine-tuned by varying mass slightly.

As a general rule, a low compliance cartridge should be mounted in a high mass tonearm and a high compliance cartridge needs a low mass tonearm. Playing a test record with a 5-20Hz sweep tone and graphing the results is the best way to determine the actual resonant frequency. Very small changes in mass, like the difference between aluminum and stainless steel cartridge mounting hardware, can alter resonant frequency.

Expert opinions differ on the optimum frequency for cartridge/tonearm resonance. The range usually recommended is 6-15Hz but I will state without equivocation that the magic number is 8Hz. Go higher than that and you'll get fatter bass with less definition. Go much lower and the stylus will jump out of the groove if it encounters the slightest warp.

Cantilever Resonance

The flexibility of the cantilever tube (or rod) and the mass of the stylus and the generator will establish the high frequency resonance of the cartridge. At this frequency there will be a peak in output and above this frequency the output of the cartridge will begin to fall. This frequency should be as high as possible and you can exercise some control over it with your pocketbook.

In the old days cantilevers were made from aluminum tubes and styli were made from big chunks of diamond and mounted in little cups at the end of the cantilever. Lower-priced cartridges are still made this way. Today's better stylus tips are ground on tiny diamonds and "nude" mounted on cantilevers made from extremely stiff materials like boron. This reduces moving mass and creates a stiffer spring, which equates to a higher resonant frequency and broader bandwidth. Sometimes technology isn't just hype. Sometimes you can really buy better performance by spending more. APJ *an interview with...*

Brooks Berdan

by Richard Hardesty

Brooks Berdan is well known in the high-end audio industry. He began applying his mechanical engineering skills to audio components in the 1970s and his meticulous turntable set-ups have established a benchmark to which others can only aspire. Over the years Brooks has created a number of innovative modifications to existing turntable and tonearm designs and inspired new design concepts that have helped to raise the standards of analog performance. Today he is one of our most experienced vinyl advocates. Manufacturers consult with him on new designs and audiophiles from all over the world seek his services.

Brooks Berdan and I used to spend most Saturday mornings riding nearly-legal motorcycles at ludicrous speed through the mountains and canyons above his retail store, Brooks Berdan, Ltd., in Monrovia, California. We'd put 350 miles on the odometers before noon and then go to work. I've known him for a long time and have come to respect his perfectionist nature. This is a man who simply won't settle for "good enough."

Brooks has hands-on experience with virtually every brand and type of turntable, arm and cartridge because his services are available to everyone, not just those who purchase the products he recommends and sells. I asked if he would provide a rudimentary explanation of turntable set-up procedures for **Journal** readers and he graciously agreed. This information will be helpful if you want to try to do the job yourself and it will increase your understanding of turntable systems even if you have no desire to actually work on them.

If you choose to hire an expert to set up your turntable, a little knowledge about the important adjustments and how they're done will help you pick the right technician to do the job. Knowing what's involved will help assure that you get what you've paid for.

Brooks Berdan and I met at the **Audio Perfectionist** laboratories (my house) in February 2003 to discuss the subject of high performance turntable systems for this article. Here's a transcript of our conversation.

Brooks, do you have any caveats for readers who want to set up their own turntable?

Yes, there are some points that I would like to stress before we begin. First, it is essential that a turntable be absolutely level during set up and operation. If the turntable is even slightly tilted, in any direction, all adjustments will be compromised and performance will be negatively impacted. Second, cartridges require 12-15 hours of mechanical break-in to allow the suspension to settle. (Sound will continue to change over a longer period.) A good dealer will break-in a phono cartridge before final adjustments are performed. Readers who set up their own turntables should readjust stylus pressure and VTA/SRA and recheck tonearm geometry after 12-15 hours of play. Third, some critical adjustments are best done by ear and when I talk about adjusting for focus, et cetera, I'm assuming the reader has a properly functioning audio system that allows differences to be heard. Don't try to adjust the turntable to compensate for other flaws in system performance. Fix the flaws instead.

Remember, all turntable set-up errors are cumulative because each adjustment affects all other adjustments. Make each adjustment in the proper sequence and do each one correctly!

That's good advice. How do you prepare?

A firm, level work surface is necessary. Basic hand tools are required along with any special tools needed for tonearm or suspension adjustments. Special tools are usually furnished with the turntable or tonearm. A precision alignment gauge must be used to align the cartridge and set stylus overhang. A carpenter's level is essential. A straight edge level works much better than a round bubble level.

When the work area is ready and the tools are collected how do you begin the set-up?

I start assembling a mass-isolated table by leveling the platter. The feet should be adjusted to be as close to the plinth as possible and locking nuts tightened to eliminate vibration and movement. Next verify that the plinth is level.

If the platter is level and the plinth or arm mounting surface is not level, you've got a serious problem. Fix it before you proceed. All level checks should be done in two perpendicular directions—side-to-side and front-to-back.

That covers the first stage of set-up for tables without spring suspension.

Are suspended turntables assembled differently?

Many suspended tables, like Linn Sondeks, must be elevated for access to the chassis adjustments. These tables should be mounted in a secure set-up jig that allows you to work above and below the plinth. The set-up jig must be level.

Again the first step is to level the platter and verify that the arm mounting surface is parallel to the platter surface. If the arm mounting surface is not absolutely parallel to the platter surface the situation must be corrected. The tonearm will not work correctly if there is even a small deviation from parallel. Fix the problem or replace the defective part. Do not use shims in an attempt to compensate for non-parallel surfaces!

After we're sure that everything is level and true, what's next?

At this point I install the cartridge in the tonearm with the bolts positioned at the midpoint of the headshell slots, mount the tonearm on the turntable and pre-adjust the counterweight. Before suspension adjustments begin, the tonearm cable and belt should be installed and the tonearm should be placed in the midpoint of travel over the playable surface of the record. Then the suspension can be leveled.

Next orient the springs for pistonic bounce—the subchassis should bounce straight up and down with no wobble or sideways oscillation—and test using gentle pressure to bounce the suspension. Make sure that everything associated with the springs is straight and true. If there are bolts in the springs make sure they are straight. Remove any burrs from rubber grommets and seat the grommets in chassis holes. Some grommets, like those used with the Linn Sondek, are eccentric to allow the position of the subchassis to be adjusted. The position should be adjusted so that there is clearance between the arm board and plinth all around.

It's important to understand that the suspension is critical to isolation. It must bounce freely at 3-4 Hz or lower. It must bounce straight up and down. Except for the tonearm cable and belt, there must be no mechanical contact that might allow vibration to bypass the springs. You must use the correct belt and tonearm cable. Incorrect belt tension can pull the subchassis to one side and defeat the suspension. A tonearm cable that is too stiff can alter the bounce frequency and may bypass the suspension.

After the turntable chassis is adjusted we can begin with the critical adjustment of the tonearm and cartridge, right?

That's correct. There are five basic adjustments and they are interactive so I do them all twice. The first time through I get things in the ballpark to minimize cumulative errors and then I go back and set everything perfectly. Let me give you some examples to show why this is very important.

You must get the VTA/SRA (vertical tracking angle/stylus rake angle; see *Turntable Physics* in this issue) close before you adjust overhang. If the back of the tonearm is very high, for example, it will affect the overhang dimension. Overhang must be adjusted before stylus pressure because moving the cartridge back and forth to adjust overhang will increase or decrease stylus pressure. Changing stylus pressure will force the cantilever upward or downward affecting VTA/SRA, etc.

Give us the order in which these adjustments should be performed.

I start by adjusting the height of the back of the tonearm until the top of the cartridge body is parallel with the record surface or the cartridge is slightly low at the rear. This gets the VTA/SRA near the correct setting. I do this by sighting on the top of the cartridge body, not the top of the tonearm tube which may or may not be parallel to the top of the cartridge body. At this point the counterweight should be installed and positioned to provide some stylus pressure. I start on the light side of the range of acceptable stylus pressure and proceed with the adjustments that follow. Final adjustment of stylus pressure will occur later.

After these preliminary adjustments are completed I adjust azimuth. This can be accomplished by observing waveforms on an oscilloscope or by dropping the stylus on a mirror which makes any angular error easier to see by visually doubling the error. If the tonearm does not allow azimuth adjustment do not try to compensate for error by shimming the cartridge or the arm. Accept things as they are or replace the arm or cartridge.

After initial settings have been accomplished I adjust overhang and offset angle. This adjustment must be performed with the aid of a precision mechanical instrument. If you don't have a gauge, find someone who does. I use the null points developed by Baerwald, which have become industry standards.

Now stylus pressure (tracking force) can be set and anti-skate (bias) force applied. If the cartridge manufacturer has a recommended stylus pressure, like 1.7 grams, start there. If the recommended setting offers a range of pressure, like 1.75-2.25 grams, start at the high end of the range. An oscilloscope can be used to observe mistracking on a highly modulated test record but ultimately you must install the turntable in an audio system and listen while making fine adjustments. Gradually reduce stylus pressure and listen for image focus using a simple recording of a solo instrument or voice. Set the final pressure in the middle of the range where the best image focus and most realistic soundstage size is achieved.

Stylus pressure is a critical adjustment because it does more than simply press the stylus to the groove walls. The correct stylus pressure will center the moving element within the cartridge in relation to the fixed elements. In the case of a moving coil cartridge, that means centering the coils in the pole piece. With moving magnet or moving iron cartridges it assures that the magnet or iron element will be centered in relation to the coils for the most linear operation.

Antiskate (bias) force can be adjusted using a test record and an oscilloscope. This takes some experience. If bias is set too high based on mistracking of highly modulated grooves it may cause mistracking on grooves with lower modulation levels like those found on most recordings of music. Antiskate can also be adjusted by playing records with high levels of modulation and listening for mistracking. If you hear mistracking primarily in the right channel, increase antiskate force. If you hear mistracking primarily in the left channel, decrease antiskate force.

Antiskate or bias force cannot be set using a grooveless record because the skating force you are compensating for is created mainly by friction in the groove and there is far less friction when the stylus is gliding across a smooth record surface with no grooves.

Final adjustment of VTA/SRA comes last—not because this adjustment is unimportant but because all other adjustments must be perfect before VTA/SRA can be set correctly. This adjustment is commonly referred to as setting the vertical tracking angle, but what we are actually optimizing is stylus rake angle. (See *Turntable Physics* in this issue.)

First listen for image focus and determine the minimum and maximum settings. Fine-tune within this range of adjustment. Next listen for harmonics and adjust between dull and bright sound. This adjustment is always a compromise and the setting will change when you play records of different thicknesses or records that were cut at different angles.

After these adjustments have been performed, are we done?

After I have done all the adjustments described above, I go back and do them all again, in sequence, to correct for minor errors that have occurred due to interaction between adjustments, and I tighten everything down so the settings will stay set. VTA/SRA and stylus pressure adjustments are interactive and may require some tinkering to arrive at the best compromise setting.

Are there any special considerations for tangential tracking tonearms?

Tangential tracking arms must be adjusted so that the arm carrier is absolutely level across the entire range of travel. The cartridge should be positioned for zero overhang and the arm adjusted to be parallel to the record surface and tangential to the record grooves. The cartridge should remain perpendicular to the record radius across the entire range of travel. Stylus pressure and VTA/SRA are set in the same manner as for pivoting arms.

Some tonearms offer a choice of counterweights. With a pivoting arm a heavier counterweight positioned closer to the pivot is preferable to minimize inertia. With tangential tracking arms a lighter weight positioned farther from the pivot is usually better to facilitate lateral arm movement.

Tangential tracking arms, particularly the passive type, can be affected by lead dressing. Cue the arm up and down in the midpoint of travel and observe whether the arm tends to move inward or outward and dress the cartridge leads for minimum effect. APJ

High Cost of Hi-Fi

It's easy for those of us with middle class incomes to become intimidated by the high prices of high-end audio components. The Journal has tried to dispel the myth that true high-end performance cannot be achieved without spending a king's ransom on an audio system. While good—even great—performing components are available for less than that, outstanding equipment remains relatively expensive for those of us who work for wages.

Dealers are reluctant to discuss alternatives to purchasing new components at retail because they mistakenly believe that this may cost them business. In fact, a healthy used market stimulates business by providing a means for new customers to embrace our hobby. These newbies can get started with used or demo components they otherwise could not afford and they are likely to purchase top-of-the-line upgrade components one at a time when they become aware of the deficiencies of the bargains that got them started. A healthy used market also allows the more affluent buyer to recoup some of the investment in a component that, for whatever reason, has proven to be personally unsatisfying. This encourages that affluent customer to reinvest, perpetuating the business.

I met Chris Fitzgerald when he attended an AP Seminar. Chris has managed to assemble an outstanding system for a remarkably small investment. I asked him to share his experience with other Journal readers who may be working with budget constraints.Here's his story. —RLH

BLUE COLLAR confessions

by Chris Fitzgerald

I really love music but I'm a family man with financial obligations that preclude huge expenditures on high-end audio components. Even so, I have managed, without undue financial sacrifice, to assemble a remarkably satisfying home entertainment system by purchasing used and demo equipment. The total cost of my high-end system was less than half the combined retail price of the individual components.

I am a long-time Journal subscriber and an equipment junkie. I attended **Audio Perfectionist** seminars in August and November (2002). That opportunity changed my perspective on home music reproduction, clarifying much of my earlier confusion about how to attain an accurate and honest sounding set-up in my own home.

The **AP** seminars, of course, included discussions on equipment selection in various price ranges. With this information, and some creative financing, I've managed to put together a system that is top-notch. I assembled a system with a retail value of somewhere around $35,000 for a total expenditure of only about $14,000.

Prior to the **AP** seminar, I already owned what I thought was first-rate equipment—made by TAG McLaren, B&W, Theta, Definitive Tech, Sony, and Mitsubishi—along with an assortment of cables. I had a system that I thought was very dynamic and for me very exciting. As I said, I thought I was doing well until my first visit to Richard's house. After the seminar, armed with my newfound knowledge and a desire to achieve a somewhat identical sound to Richard's, I did what I told myself I wouldn't do for a long time. I began replacing equipment. Much to the dismay of my wonderful wife, I sold everything piece by piece, again.

After hearing Richard's system, I had a much better idea of what type of equipment to obtain. I wanted to have my system built on the same principles that Richard recommends. A set-up

> ***"Frequently, the sellers are people who tend to require the latest, prettiest, or most prestigious toys; I have bought much of my good equipment from such people."***

consisting of time- and phase-accurate speakers, fully balanced amplification with zero feedback, and—something completely new to me—a solid turntable.

Richard has asked me to share with other **Journal** readers my methods for attaining good equipment at reasonable prices. First of all, buying used and demo equipment has allowed me to own better components than I'd be able to afford if purchased new. To achieve this, I buy from internet sites, local dealers, and other contacts in the retail business. I have learned through Richard's seminars how to evaluate equipment based on my needs: what features to look for, quality of workmanship, manufacturer's reputation, etc. I use the internet extensively to research the equipment I am interested in. I get involved in chat rooms on various AV sites to hear different points of view on equipment and, lately, I have read a lot about turntables.

The internet has all of the tools that a thrifty AV shopper needs. Frequently, the sellers are people who tend to require the latest, prettiest, or most prestigious toys; I have bought much of my good equipment from such people. To give you an example, I bought my Theta Dreadnaught amplifier from a lawyer on the east coast. He had the piece listed on an internet site for several months and, by the time I contacted him, he was fairly frustrated that it hadn't generated more interest. He was dissatisfied with it, he said, and just wanted it out of his living room. It turned out that it had a small ding in the top corner of the faceplate. Since I'm generally more interested in how a component performs than what it looks like, I was able to get this fine amp at a great discounted price.

I recently bought a Linn LP-12 from a seller in Florida who had given up on LPs. The ad included a very basic description and some poor quality pictures. This turntable had a broken cartridge and a broken lock on the tonearm rest. Knowing I could have these parts fixed for very little money, I took a closer look. It had an Ekos tonearm, which sells for $1,000 used. I got the whole package for $1,000! When I questioned the owner about the features on the table he knew nothing about it. He admitted to me that he just went to his local dealer and bought it without even a demo.

My system now includes Vandersteen 2CE Signature speakers, a VCC1 center channel speaker, VSM1 surround speakers, a 2WQ subwoofer, a Theta Dreadnaught amplifier, a Classe SSP-75 surround processor, an Ayre K3x preamp, a Linn LP-12 turntable with Ekos tonearm and Troika cartridge, a Sony DVP NS999ES DVD/SACD player, and Audioquest Volcano, Anaconda, and Python cables—all of which were purchased used on the internet, except the Classe SSP-75, Sony DVP, Vandersteen surrounds and Audioquest cables, which were purchased from dealers. All of the components were demos or closeouts, or purchased used on the internet. Several pieces have some sort of minor cosmetic defect—i.e. a scratch on the cabinetry—which doesn't affect performance.

If you're willing to look beyond the superficial, you, too, can obtain first-rate equipment at killer prices. Stay open to buying used and demo pieces from local dealers. Do your homework, research the components and, most of all, know what you want. Be ready to buy the components when they come up—timing is everything!

In a short period of time I have been able to go from a good system to an outstanding system that is everything I ever imagined it would be, one that is time- and phase-accurate, and fully balanced. I just had to overlook two small details: none of it matches, and every piece has scratches. Until next time, happy hunting. APJ *an interview with…*

Joe Harley

by Richard Hardesty

At the 2003 Consumer Electronics Show (CES), as in years past, I found many high-end audio exhibitors demonstrating the capabilities of their products with analog, digital and SACD recordings produced by Joe Harley. Harley and his engineer/partner Michael C. Ross have achieved respect in the finicky high-end audio industry for their abilities to recognize good music and to capture the essence of that music in their recordings so that it can be reproduced in our homes (and at hi-fi shows) with great sound and the emotional message intact.

Joe Harley and I have been friends for many years. I thought that he would make a good candidate for an interview in the ***Audio Perfectionist Journal*** *for a number of reasons.*

Joe makes great sounding recordings of music that you want to listen to, not use just to show off your system. He is a music lover with audiophile sensibilities who is experienced in every area of high-end audio. His knowledge of high-end audio reproduction assures that his music productions are held to high standards—Joe knows the capability of high resolution playback systems and strives to capture the very best sound on his recordings. He has compared the sound of live music to the sound of a monitoring system playing a direct mic feed, giving him an objective basis for comparisons of recording technologies. He has compared analog tape recordings to a direct mic feed and to the sound of live musicians in the studio. He has compared analog master tape recordings to the compact discs and vinyl records made from those tapes. He has compared all these things to Direct Stream Digital recordings on hard disc and to the SACD discs made from these DSD hard disc recordings.

Joe & Michael

I have enjoyed learning about Joe's experiences and I'm sure you will, too. Here are my questions and Joe Harley's responses:

Joe, your recordings have become a staple source for music lovers who care more about content than sound quality, ***and*** *for high-end audio enthusiasts who demand excellent sound. You've managed to capture the musical message and make recordings that sound good.*

Tell us about yourself and the history of Harley Music Productions. How did you start? What motivates you?

I had been working with Bill Low at AudioQuest since 1983, helping to establish the company and set up our dealer/distributor network. In 1990 Bill and I had the idea of doing a "one off" promotional recording using AQ cables. This record turned out to be the Robert Lucas classic *Using Man Blues*. The record was a sensation in blues circles and helped establish Lucas as a major figure in the blues world. It also generated a lot of attention in the audiophile community with Corey Greenberg, then of *Stereophile*, naming it the "audiophile recording of the century." Thinking back on the recording now, I really had no idea at the time what "production" meant. But I loved the music, appreciated good sound and basically just flew by the seat of my pants.

We did the early AQ recordings with Kavi Alexander and his fabulous Blumlein/DeParavacini set-up. [Blumlein conceived the crossed/coincident microphone technique and Tim DeParavacini was responsible for the modifications to the recording electronics and tape deck.] I still think it is an excellent way to record, particularly solo and small acoustic ensembles.

I became totally fascinated with trying to capture the best performances…the musical juice…on tape. I loved the simple mic technique for certain projects, but I began to realize that there were obvious limitations to the approach, particularly when the group got larger or drums or electric instruments were involved. I would end up telling everyone to play more quietly, or move an inch here and there. This kind of thing can really put a damper on the creative energy in a session. Musicians end up having to worry about more than their own artistic expression. I decided that for much of the music I wanted to record, a dif ferent approach would be needed. I focused on trying to maximize the traditional studio experience. I insist that the sonics serve the music, not the other way around.

Meanwhile, AudioQuest Music was born as a distinct entity separate from the cable company. I wanted a label to make records that put musical merit as a first priority. I was extremely fortunate to find a few engineers, and in particular Michael C. Ross, who could help bring this vision to fruition. At the time, some so-called audiophile labels had a reputation for making great sounding but musically tepid recordings. I wanted to make recordings that could stand on their own in the blues and jazz world.

AQM jazz recordings featured such artists as Grover Washington Jr., John Abercrombie, Bennie Wallace, Kenny Barron and James Newton. The AQM blues roster found success with such artists as Mighty Sam McClain, Ronnie Earl, Terry Evans, Ry Cooder and Doug MacLeod.

In 1996 I began to do production work for other labels, including GrooveNote, Enja, Telarc and ECM in addition to AudioQuest Music. I left AudioQuest and formed Harley Music Productions at this time, working with such artists as Jacintha, Charles Lloyd, Brad Mehldau, Eden Atwood, Anthony Wilson and Robert Lockwood Jr. The AQM label was eventually sold to Valley Entertainment in 2000.

I also began a long association with the JVC/XRCD program at this time, serving as Creative Director.

How do you choose projects?

Projects come to me in a variety of ways. Sometimes a label will assign a project to me but more often I will bring an artist whose work I particularly admire to the attention of a label. I have also had a number of artist referrals, meaning that someone I've worked with will recommend me to another artist.

The artist and I will come up with a project proposal and I'll draft a budget. If the label approves, we start the wheels turning for a session.

What projects are you proudest of?

Difficult question. Anytime the music is completely happening and you're getting it all on tape or hard drive properly…those are the magic times, when you know that all the effort is worth it. Having said that, there are a few projects that stand out for me. I've worked on the last three Charles Lloyd albums for ECM. The first two included a dream band for a jazz lover like myself….Brad Mehldau on piano, John Abercrombie on guitar, Larry Grenedier on bass, Charles of course on tenor sax and the late, great Billy Higgins on drums. It was one of Billy's last sessions. His health was failing, he was so weak. But when he would get on the drums he would totally transform and be Billy Higgins at his finest! It was something to witness. He would look at his body and say, "This is all gone, it's all spirit now."

What an amazing man.

I've worked with Ry Cooder a number of times on Terry Evans' projects. He really understands that it's the soul of the music that drives the thing, not trying to make it perfect. Ry always wants to get the thing on tape when the band is right on the cusp of learning the material, when the element of surprise and wonder is fresh. He is a master of texture and sound.

I have a special fondness for several artists who are lesser known for whatever reason, but who should be on any best of the best list. Tenor sax player Bennie Wallace, vocalist Eden Atwood, guitarist Anthony Wilson… these artists deserve to have much wider recognition.

There will always be a special spot for R & B master Mighty Sam McClain. Just a few months before we recorded Sam's first comeback album in 1992, he was living on the streets of Boston and eating out of garbage cans. But his talent was intact. The record came out and was a sensation. Sam and I went on to make 7 albums together. Last year one of the tracks from his most recent album was used as one of the recurring themes on the Ally McBeal show. Sam bought a large tour bus on the proceeds from that.

What equipment do you use to make original recordings?

I like to use vintage consoles and vintage mics….they just have a rightness to me that is missing in so much modern gear. For me that means Neve consoles like the 8038 and 8078. At one of my favorite studios, OceanWay Studio B in Hollywood, there is a custom console that I love. It's mostly passive and nearly transparent. For microphones, I like the vintage Neumann's… the 269, M-49, M-50, U-47s. AKG C-12s and C-12As are used frequently on my productions, most often for piano and drum overheads. The Sony C-55P is an amazing mic, especially for string bass and guitar. You pretty much have to record at one of Allen Sides' facilities (OceanWay and Record One) to use them, though, since he has corned the market on these babies.

For years I recorded solely to analog… no digital system I heard could come close to competing with the natural warmth and bloom of analog tape. My machines of choice were made by Ampex… the ATR1002 and 1024 machines.

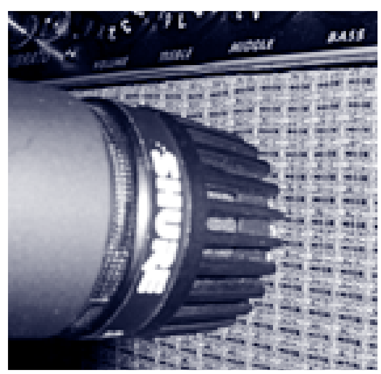
Then, in early 1999 I had my first exposure to Sony's Direct Stream Digital (DSD) format. It changed everything I had previously thought about digital. It rocked my world. I remember very clearly looking over at Mike Ross as we made our first comparison of the direct mic feed to the DSD hard drive to our trusty tweaked ATR 1002 analog machine. The DSD sounded like the direct mic feed! For the first time I had a clear handle on what analog tape DIDN'T do right. The tape still sounded great, beautiful as always. But now I could clearly hear the "rounding" artifacts and loss of transient impact. The DSD sounded so much like the direct feed that I would get confused going back and forth between them….they both sounded so similar.

Since then I have recorded or mixed almost exclusively on DSD and have done a number of projects in 5.1 DSD.

When you record in the studio are you able to monitor with components that provide a reasonable facsimile of the live sound? Does the microphone feed played through the speakers in the studio sound like the live performance?

This is a very good question. The answer used to be "not always." In fact when I first began working at OceanWay I was panic stricken since their own highly touted monitors sounded bright and boomy to me. Dynamic as hell but not a tonal balance I could relate to. I would drive back to San Clemente to check balances on my own system (various combinations of Vandersteens and ARC components). Then my engineer, Mike Ross, brought in some Tannoy monitors that had been modified by the Mastering Lab. Bingo! Here were monitors that were robust enough to live in a studio but had a tonal balance that I felt totally at home with. For many years now we've been dragging them to every session….and now we have 5 to cover us for surround sessions.

When you compare a direct mic feed to the analog tape recording of that feed what differences do you hear? How much is lost in the basic recording process?



Again, the live mic feed is more direct and "open" than even the best analog tape. It isn't a major difference but it is very easy to pick up on. The sound coming back on the analog tape is rounder and slightly blunted by comparison.

When you compare the original master tape recording to the commercial vinyl records and CDs made from those recordings, what musical information seems to be missing? Which provides a better representation of the master tape, vinyl or compact disc? Which one best conveys the musical message to you?

I feel that you have to look at LP and CD separately. There really is no comparison. Yes, it is possible that a bad LP pressing will be beat by a superb mastering of a CD. It's possible. But for the most part LP retains much more of the critical information that makes you feel the music on an emotional level. That thing that makes you tap your foot, nod your head, cry, laugh…whatever it is the artist is intending for you to feel...is just so much more alive on vinyl. CD comes off as kind of dead by comparison. You turn the volume up trying to make something happen, but the deadness just gets louder. You find your mind wandering, you start reading the paper, doing other things. After a while you turn it off. Your brain is being asked to fill in the blanks that 16-bit/44.1kHz can't provide. So listening to music becomes work! Music is supposed to be relief from work.

Tell us about your experiences with Direct Stream Digital recording and SACD discs. How does a DSD master recording on hard disc compare to the mic feed?

DSD is a major advance in my opinion. It is the closest archival medium we have for capturing the sound of the direct mic feed. In fact, if we are talking about the sound of the DSD hard disc, the sound is so close as to be nearly indistinguishable. All the life, the musical "juice" is retained on DSD.

How does a commercial SACD disc compare to the original DSD recording on hard disc?

You ask a very interesting question about how the information is retained once the data is processed for commercial SACD disc. The answer is it mostly is… and more of it is retained now than even a year ago. Those of us involved with DSD recording began to notice that when we backed up the hard drive onto an AIT tape something happened. Something was lost. Sony began to investigate this….minds like Ed Meitner got involved. Now what you get back on the commercial disc is much more like the original hard drive. It was good to begin with and it's getting better all the time.

I know that you have done some multichannel projects. Do you think that multichannel offers an improvement in musical fidelity?

Yes, at its best I believe it can. Of course there's also the opportunity for rampant abuse as we've all heard. I had an absurd mix just the other day by a guy who should know better. He had half the drum kit front right, and the high hat and snare rear left. Whatever could he have been thinking? This totally discombobulates the feel of the rhythm section for the sake of cheap carnival tricks.

But what you can do is to tastefully go for a more immersive experience. You can do this by expanding the front stage left to right, which now also opens up the layering within that stage. Think of it as a horseshoe with the listener's position at the center opening of the horseshoe. Then you bring a sense of ambience, the room mics, into the rears. It can be very effective and involving. The stereo mix just is not as much fun to listen to!

What projects have you completed recently?

I just finished another surround DSD mix for GrooveNote with the popular Asian singer Jacintha. I have also been supervising DSD transfers of older Concord material, again for SACD release on the GrooveNote label. A great Ray Brown album—*Soular Energy*—and LA Four album have already been released. Next week we go in to do the DSD transfers for Stan Getz and Rosemary Clooney albums.

What other projects are coming up?

Next month I will start my first project for DVD-A release for DTS. This will be a surround mix of an album originally released on Blue Note by the band Medjeski, Martin and Wood. I love the band's music so this is one I'm really looking forward to. For this one forget everything I said about surround mixing… this music has no rules; neither will my mixing!

In April we will be recording the wonderful vocalist Eden Atwood again. She is the real thing…she makes you feel the song. We'll have James Moody on board and some other great guests.

In May, I'll be in New York to record Bennie Wallace in a trio with the great Kenny Barron on piano and Eddie Gomez on bass. Later this year I'll be going back in for another Charles Lloyd album on ECM.

What does the future hold for Harley Music Productions?

In October of last year I rejoined AudioQuest after a hiatus of six years. It's great to be back working with Bill Low again. As you can see, part of our deal was that I keep my outside music clients. When a project comes along that I want to do, I go do it. Music and audio… my two addictions. What could be better than that? APJ

DIGITAL

by Richard Hardesty

*I intended to begin this section of this **Journal** with an article that would help to clarify the confusing world of digital audio. I wanted to replace the rhetoric and hyperbole with simple explanations that readers could understand. Obtaining the necessary background information, however, proved to be more difficult than I had anticipated. My research led me to industry "experts" who either couldn't or wouldn't provide me with specific data explaining the actual processes involved. I encountered smoke and mirrors almost everywhere I turned for information.*

Obfuscation

Many of the "authorities" I consulted seem to prefer withholding the truth from me and other consumers. Instead they offer marketing slogans and meaningless terminology along with definitions that make no sense. Why, you might ask, do they want consumers to be confused and bewildered about what's really going on in the digital audio world? Do they feel that providing candid explanations is futile because we simply wouldn't understand? Are they so much smarter than we are?

Is "upsampling" really different from oversampling? Is pulse density modulation really different from delta-sigma modulation? The answer to both questions appears to be no, but manufacturers and industry pundits who have "hung their hats" on new terminology, which doesn't necessarily describe new technology, continue to protest when confronted by evidence and asked for specific information about what distinguishes their "unique" processes from common techniques that are better known and more easily understood.

Differentiating Your Product

Those of us who are familiar with computers know that often there is little to distinguish between the various brands. A dozen or so manufacturers assemble machines around CPU chips made by a small number of companies mounted on mother boards made by a few others. Add video and sound cards from a limited number of sources, and various other specific components like disc drives, and you can assemble a machine that runs the same software and works pretty much like all the others. Digital audio components at the low end of the market are similarly indistinguishable. At the high end, consumers want something different to justify the extra expenditure that high-end audio products demand. Manufacturers often offer rhetoric instead of innovation to differentiate their products from all the rest. They particularly want to distance themselves from competing products that perform the same function but cost less.

Digital audio is a complex and confusing subject that won't be fully explained by a few thousand words even if I could fully explain a subject that I can't claim to fully understand. My original goal won't be completely accomplished in this **Journal** but I won't give up. Consider this an opening salvo in a continuing battle to reveal the truth to our readers as we discover it. I am fairly confident that the information presented here is essentially correct but I will be happy to modify or rescind any statements that prove to be misleading or wrong. I'll continue to learn and present what I discover to you.

The Analog World

We live in an analog world. Daylight fades continuously into darkness and the howl of a lone wolf startles the night and fades continuously into silence. We could record representative samples of the sunset as it occurs with a series of frames (samples) on movie film. Projecting the movie at 24 frames per second would provide an excellent visual simulation of the actual continuous event. We could record the wolf's howl on analog tape or convert the analog waveform to a series of samples in the form of a timed digital code which can be stored on a digital medium. The analog tape will provide a remarkably accurate reproduction of the live sound, but how about the digitally coded recording? Can a series of timed samples allow us to hear more than a pale facsimile of the original event?

Converting a continuous event into a digital code consisting of timed samples provides many advantages for storage and transmission but a sufficient number of samples must be taken, and sophisticated coding techniques must be employed, in order to preserve every nuance of the live event. The digital coding techniques of the past have convinced many audiophiles that digital audio is inherently bad. Newer, higher resolution digital formats provide hope for an audio future that is both convenient and musically satisfying.

Can Digital Audio Work?

Humans perceive the events in an analog world using cells that perform in a digital manner—these cells seem to be either off or on. Neurons, which are the basic functional elements of our nervous systems, communicate with impulses that pass through synapses connecting them. These components of the human nervous system seem to work like the components in digital computers.

We have an enormous number of these cells giving us the capability to perceive events in an almost continuous manner. Our eyes can be fooled by movies that project still images 24 times per second but our hearing is far more sensitive. While many people find that compact discs, which sample 44,100 time per second, can't provide a completely satisfying musical experience there is little question that digital can work because it seems to work in our own bodies. Perhaps what we need is a digital audio system with higher resolution capability.

Just what is a digital signal and how does it differ from an analog signal? How is the resolution of a digital signal determined? How do the new high res digital formats differ from the digital coding on the familiar compact disc?

Digital Signals

A digital signal is composed of sequential samples that represent numerical quantities at an instant in time as opposed to an analog signal, which has continuous physical variables.

The air pressure variations that we perceive as sound are continuously variable. These continuously variable analog "sound waves" must be sampled periodically and converted into a digital code in order to be stored on a digital medium. This digital code must be reconverted to an analog signal for playback. An analog-to-digital converter is called an ADC and a digital-to-analog converter is called a DAC. The sampling theorem defines the requirements for accomplishing the conversion from analog-to-digital-to-analog.

The sampling theorem, developed primarily by Nyquist and Shannon at Bell Laboratories, states that a continuous analog signal can be converted into a stream of timed samples without loss of information if the analog signal is band-limited so there is no content above or equal to half the sampling frequency. Bandwidth limiting is critical to prevent ghost frequencies (images and aliases) from being introduced into the signal.

Bandwidth limiting is accomplished by filtering and the filtering process is accompanied by problems. Filters tend to distort fast transients in music because filters don't respond to impulses in an ideal manner. Filters ring—continue to oscillate after, or in the case of digital filters before and after, the transient.

The signal is low-pass filtered before sampling (analog-to-digital conversion) by an anti-aliasing filter. During reconstruction (digital-to-analog conversion) an anti-imaging filter is applied. The method and application of this filtering has a significant effect on sound quality.

A digital code consisting of sequential samples that represent the amplitude of the analog signal at specific instances in time, or the change in amplitude between one sample and the next, will be created during analog-to-digital conversion. The digital code will consist of many samples representing many instances in time. With linear pulse code modulation (LPCM), each digital sample will have a numerical value that represents the amplitude of the analog signal at the time of the sample. In the Direct Stream Digital (DSD) process, the density of pulses will represent the analog signal.

A large number of digital samples will provide a good representation of the analog waveform over a period of time. Increasing the number of samples in a given time period, or increasing the size of each sample, can provide a better representation of the analog waveform.

The digital code consists of a series of ones and zeros—actually a series of pulses occurring in a time frame. These pulses can represent one of two states. The presence of a pulse can represent one state, and the absence of a pulse can represent the other state. These two states can represent binary numbers. What, you might ask, are binary numbers?

Binary Numbers

Computers, and other digital devices, count and manipulate numbers using the binary system. The binary system works just like the more familiar decimal system except for the number base. The decimal system is base 10 and has ten symbols for integers: zero through 9. When you run out of symbols, you add a column at the left to indicate how many times you have passed the base number and start counting again. Counting goes like this: 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, (out of symbols, add a column and start over), 10. Counting is in groups of 10 and powers of 10. Each column represents a power of 10. The decimal number 111_{10} (base 10) equals 1×10^2, plus 1×10^1, plus 1 (10^0), which adds up to one hundred eleven.

The binary system is base 2 and has only two symbols for integers: zero and 1. Counting goes like this: 0,1, (out of symbols, add a column and start over), 10. The binary number 10 is equal to the decimal number 2. (In binary, I have 10 thumbs but there is still only one on each hand.) Counting in binary is in groups of 2 and powers of 2. The binary number 111_2 equals 1×2^2, plus 1×2^1, plus 1. Add these up and you'll get 7 which is the decimal equivalent of 111 in binary. The binary number system (base 2) can express any quantity just like the decimal number system (base 10). The binary number will probably have more digits but each digit will be either a zero or a one.

Actual quantities can be represented by different symbols in different number systems. I have four fingers and a thumb on each hand. All the fingers (4) plus the thumb (1) on my left hand, combined with two of the fingers on my right hand (2) add up to a number usually referred to as 7 (in base 10). The binary number 111 is an alternative representation for the same quantity. The binary number has more numerical digits but it represents the same number of biological digits (fingers and thumbs).

The binary system can be used for mathematical calculations just like the decimal system. You can add, subtract, multiply and divide in binary. The binary system is an ideal number system for machines because there are only two integer symbols. These two symbols can be represented by a mechanical or electronic switch with two states: open and closed. That's how computers work.

Essentially, a computer is a huge collection of switches. At an instant in time, each switch is either open or closed, off or on. With a sufficient number of switches, and very accurate timing, you can calculate the trajectory to the moon and back, or record music. With sufficient bandwidth you can transfer your calculations, or your music recording, from one computer to another by wire or microwave. Digital codes are very efficient and convenient.

In the audio world, digital codes based on the binary system are used to transform analog audio signals into digital data that can be easily stored on optical discs. These optical discs are the most popular storage medium for our high-end audio systems and they come in several flavors—CD, SACD, DVD-Video, DVD-Audio—each with special capabilities and characteristics.

Recording Music Digitally

Digital recordings are made just like analog recordings with an additional process: analog-to-digital conversion, which converts an analog signal into a digital code. Digital-to-analog conversion, which converts that digital code back into an analog signal, must be performed during playback. To help us choose desirable formats and hardware we need to know a little bit about how the music was converted into digital and about how it will be converted back to analog in our home audio systems.

There are two digital coding methods commonly used for high-end music reproduction: LPCM and DSD. This article will attempt to shed some light on these. Compression algorithms can be used to shrink LPCM data to fit in less space on the disc or take up less bandwidth for transmission. Compressed formats with lower fidelity (data reduced or lossy compression), which are adequate for movie soundtracks and computer music files, will be discussed briefly.

Multibit and 1-bit converters

The amplitude of an analog waveform can be represented digitally by multiple bits at relatively low sample rates (a little more than twice the highest frequency to be stored) or by just a few bits, or even a single bit, at relatively high sample rates (many times the highest frequency to be recorded).

The term "bit" comes from combining the words binary and digit. Sample rate is the number of samples taken each second. Data rate is the total number of bits per second. The data rate equals the number of samples per second times the number of bits in each sample times the number of channels.

Compact discs (CDs) use 16-bit samples to represent signal amplitude levels. Turning on various combinations of these bits provides a representation of the amplitude of the analog signal level at a specific instant in time. Sequential digital samples can recreate the analog waveform. Super Audio Compact Discs (SACDs) use many more samples with just 1 bit per sample to recreate the analog waveform. Here are analogies to illustrate how multibit and 1-bit systems work.

You could make a volume control from 16 switches and 16 resistors of different values. Switch number 1 (and its associated resistor) would allow the smallest signal level to pass. Switch number 2 would allow twice the level of switch number 1 to pass. Switch number 3 would pass twice the level of switch number 2, and so forth. The output from all switches would be summed (added together). Each individual switch would allow different amounts of the signal to pass and combinations of switches would allow the signal level to vary over a wide range from completely off to full level with many steps in between. These switches, like the bits in a CD sample, could be turned on or off in various combinations to produce 2^{16}, or 65,536 different volume levels. 1-bit systems use a different approach.

One switch could turn the volume all the way up or completely off. Volume level could be controlled by turning the single switch on momentarily, many times. If this switch were rapidly turned on and off, with equal time in each position, the average volume would be about half of maximum. Increasing the "on" time would raise the volume level. Increasing the "off" time would lower the volume level. The percentage of time that the switch is "on" is called the duty cycle.

The on or off time could be increased by leaving the switch in the on or off position for a slightly longer period of time. This is analogous to pulse width modulation. Pulse density modulation is similar.

In pulse density modulation, the frequency of pulses or the width of these pulses is irrelevant. Only the average value of the output—the percentage of time that the output is "high"—is significant. If you want to learn how pulse density modulation actually works, read the article *Care and Feeding of the One Bit Digital to Analog Converter* by Jim Thompson from the University of Washington. It's available here:
www.ee.washington.edu/conselec/CE/kuhn/onebit/primer.htm

LPCM

Linear pulse code modulation (LPCM or just PCM) is a coding method using discrete samples that define the amplitude of the analog signal at an instant in time. The analog signal is sampled at intervals and the sample amplitude is expressed as a numerical value. The signal must be sampled at a little more than twice the rate of the highest frequency to be recorded and the size of each sample determines the resolution and dynamic range of the signal.

LPCM coding is used to make regular compact discs with a sample rate of 44.1kHz and a sample size of 16 bits. That means that a sample is taken 44,100 times each second and that each sample contains 16 bits which describe the amplitude of the analog signal.

LPCM tracks on DVD-Video discs use a slightly higher sample rate of 48kHz and larger sample sizes up to 24 bits. LPCM coding with even higher sample rates can be used to make DVD-Video standard discs—if little or no full-motion video is included—and DVD-Audio standard discs. Sample rates up to 192kHz and sample sizes up to 24 bits can be supported on DVD-Audio discs.

The sample rate establishes the bandwidth of the system because the sampling theorem states that there can be no signal content at or above half the sample rate. To achieve bandwidth of 20kHz (a bare minimum for high fidelity audio) the sample rate must be twice the highest frequency to be recorded with some headroom added to allow for filter attenuation. The sample rate for a standard compact disc is 44.1kHz. This permits a high frequency limit of 20kHz but requires a steep filter (brick wall) to block frequencies just above 20kHz.

Higher sample rates of 48kHz or 96kHz or even 192kHz can be accommodated on DVD discs because of greater storage capacity and higher data transfer rate capability. These higher sample rates extend the range of high frequencies and allow the use of filters with improved transient response.

The compact disc sample size of 16 bits allows for 2^{16} or 65,536 discrete levels of amplitude for a theoretical dynamic range of 96dB. Some DVD disc standards allow samples up to 24 bits for 2^{24} or 16,777,216 discrete levels. Samples approaching 24-bit resolution can provide theoretical dynamic range of 144dB, which exceeds what can be achieved with current electronic components due to intrinsic circuit noise.

LPCM coding is accomplished by analog-to-digital converters. Modern converters with 20-bit resolution (or higher) use oversampling delta-sigma (also called sigma-delta) modulation to raise sample rates and reduce sample size. Oversampling at many times the required rate allows the sample size to be reduced to as little as 1 bit.

The oversampled sigma-delta modulated bitstream is then chopped up into discrete samples by decimation. Decimation filtering, using a finite impulse response (FIR) low-pass filter, quantizes the samples reducing the sample rate and increasing the sample size for storage on the disc. This step is omitted with DSD coding.

It's important to remember that LPCM data on the disc is in the form of discrete samples, even if that data was created using sigma-delta modulation. It's also important to understand that the samples on a CD are 16 bits in size and there are 44,100 of them every second, regardless of the signal resolution during mastering. Oversampling or "upsampling" during playback does not increase resolution or add to the information that is recorded on the disc.

Resolution can only be increased by actually recording more data, which requires more bits. LPCM with 24-bit samples at 96kHz can capture a lot more information than the CD standard of 16 bits at 44.1kHz. So can DSD sampled at 2.8MHz.

DSD

DSD is a 1-bit digital coding process used to make SACDs. The DSD bitstream is similar to an LPCM bitstream before the decimation process since all modern LPCM recordings are made with delta-sigma ADCs and delta-sigma ADCs are pulse density modulators (see the above referenced article by Jim Thompson). DSD offers much higher resolution than 16/44 LPCM and could be compared to 24/96 LPCM but DSD has a simpler process.

DSD is a 1-bit digital stream at 2.8MHz. This digital bitstream is recorded on disc essentially "as is." The decimation step, used to "downsample" the delta-sigma modulated bitstream into discrete samples for LPCM, is omitted. During playback the analog waveform is reconstructed directly from the DSD bitstream.

LPCM can also be reconstructed by sigma-delta modulation and low-pass filtering but there is a significant difference. LPCM data has been decimated into discrete samples for storage on the disc and the DSD bitstream has not. The significance of this difference has often been underestimated.

Compression Algorithms

Even though DVD discs have a relatively huge storage capability, there still is not enough room for everything that content providers want to put on the discs. Full-motion video and multichannel surround sound, alternative soundtrack languages and a choice of audio mixes or compression types are some of the things they want to include to make their products attractive to a wider audience. Sound quality seems to be a minor concern in many cases.

Getting all this stuff on the disc requires data compression. Compression can be "lossless," which suggests bit-for-bit recovery, or lossy (data reduced), which means that some data is discarded in the compression process.

Meridian Lossless Packing (MLP™) is a "lossless" compression scheme used to make multichannel DVD-Audio discs. MLP™ is claimed to provide bit-for-bit recovery with no loss of fidelity. It may work as advertised and it may provide high performance capability but we cannot verify that at this time. The DVD-Audio players we have auditioned have offered poor audio performance by high-end standards. This poor audio performance may be attributable to the players which have come from mass-market manufacturers, or to the recordings which have consisted largely of remastered versions of older material, or to the mixes which have consisted largely of ludicrous surround sound carnival tricks. We must reserve a final judgment of MLP™ for a later date when we have had more experience with a wider variety of software and hardware.

Dolby® Digital (AC-3) and DTS are lossy (data reduced) compression schemes used for multichannel soundtracks on DVD-Video discs. MP-3 is a lossy compression scheme used for portable music devices like iPods and for computer storage of music files. While these lossy compression schemes are perfectly acceptable for the purposes for which they were intended, they cannot provide high fidelity sound and are not suitable for high-end music reproduction.

MP-3 should stay on your computer or portable device. Dolby Digital and DTS can be enjoyed on a high-end audio system if you have a video presentation to distract your attention from sound quality but don't consider either as a music source for critical listening. Here's why.

Compact disc is a marginally acceptable source for high fidelity music reproduction. The data rate for CD is 44,100(sample rate) x 16(sample size) x 2(number of channels) which equals 1,411,200 bits per second. If nine-tenths of this data is discarded using a lossy compression algorithm, the chances are very slim that the remaining one-tenth could provide high fidelity performance.

The "near-CD-quality" data rate for MP3 is 192kbps. Most MP3s offer data rates lower than 192kbps. MP3 files provide two channels of audio. Obviously, MP3 files are not an acceptable source for high fidelity music reproduction but what about Dolby Digital and DTS?

Dolby Digital (AC-3) provides up to 5.1 channels of audio at data rates that start below 400kbps. That's a lower data rate for each channel than MP3! You don't have to be a rocket scientist to figure out that these lossy compression formats can't provide high fidelity performance regardless of the quality of the hardware used for reconstruction. You don't have to be any kind of scientist, just listen.

DTS offers slightly higher data rates than Dolby Digital but the data includes lots of side-band (not signal related) information used to make the process "forward adaptive." DTS sounds different from Dolby Digital but I don't think the sound is necessarily better. It's certainly no more suitable for our purposes than Dolby Digital. Which would you rather have, a broken arm or a broken leg?

Dolby Digital and DTS work fine for movie soundtracks and music videos where your primary attention is on the picture.

Digital-to-Analog Conversion

The digital playback component in our home audio systems must reconstruct an analog signal from the digital code on the storage medium. This reconstruction is critical to the sound quality we hear.

DSD material is a 1-bit pulse stream at a high frequency (2.8MHz). Sony calls it "pulse density modulation," which is another way to describe delta-sigma modulation.

LPCM is a multibit code at lower sample rates of 44.1k to 192k. LPCM can be converted to a 1-bit pulse stream at high frequency before digital-to-analog conversion by delta-sigma modulation.

Linear, multibit DACs can be utilized to decode the LPCM samples directly, or an oversampling interpreter and delta-sigma modulator can be utilized to convert the LPCM samples to a pulse stream at a higher frequency. Delta-sigma DACs include a modulator which raises the sample frequency and reduces the sample size.

A delta-sigma modulator is a circuit that translates a binary number into a pulse stream with a duty cycle proportional to the binary input. The duty cycle of a digital circuit is the portion of time during which the signal is high. The pulse stream is converted to an analog signal by averaging over time with a low-pass filter. A delta-sigma modulator is a pulse density modulator.

A delta-sigma (or sigma-delta) DAC for LPCM usually combines oversampling, delta-sigma modulation and filtering into a single component. Delta-sigma DACs used to be 1-bit DACs and, while there are theoretical advantages in linearity, 1-bit DACs tended to deliver sound that was subjectively softer and less detailed than multibit linear DACs (when decoding 16⁄44 LPCM). Multibit delta-sigma DACs have changed this conception. Today's best multibit delta-sigma DACs can provide sound that seems to equal the resolution of multibit linear DACs.

A DSD pulse stream can be converted to analog without the delta-sigma modulation step because the DSD stream wasn't decimated for storage on the disc.

Oversampling and Upsampling

Oversampling is a term usually associated with conversion from analog-to-digital or digital-to-analog. All CD players provide oversampling at even multiples of the sample rate.

Upsampling is a term usually associated with a digital-to-digital process that may create sample rates that are not even multiples of the base rate—upsampling 44.1kHz to 96kHz for example.

No one seems to be able to offer a complete definition of "upsampling" but it doesn't seem to matter. Upsampling and oversampling are marketing terms with little relevance to consumers.

Both processes, oversampling and upsampling, are mathematically equivalent and both affect the implementation of digital filtering. Cascading digital filters may change the sound and this may provide a subjective improvement but it does not provide higher resolution or add information to the recording.

Upsampling chips are readily available for a few dollars and we've heard some players that use them to good effect but upsampling is not a "magic bullet" that will turn your CD collection into high resolution digital. The recording process determines resolution. Tricks during playback can't add (real) information that wasn't recorded in the first place.

The High Fidelity Approach and Digital Audio

The high fidelity approach allows us to narrow the field of available products by eliminating from consideration those components that are poorly or inadequately designed. Learning a little about how products actually work also helps us to avoid overpriced components that are made from the same off-the-shelf parts as well-designed products costing less. While the high fidelity approach is easily applied to analog components, digital audio components present some special obstacles. Many of these obstacles are based on misinformation.

Digital audio is complicated and confusing to many consumers. Some manufacturers have taken advantage of this fact by creating misleading advertising copy describing how their products work. They have resorted to an old advertising dictum that states "if you can't convince them with facts, baffle them with bullshit." Some of the journalists writing about the subject have failed to see through these smoke screens.

Digital audio components must be designed in accordance with basic scientific principles or they simply won't work. Small companies can assemble digital audio components from integrated circuits designed by qualified engineers at Texas Instruments or Cirrus Logic. There are good and bad aspects to this situation. The bad side is that it may be more difficult to narrow the field of contenders without actually listening to a lot of components. The good side is that even the poorest performing products aren't that bad. So how do we separate the products that are worthy of our auditioning time from those that are not?

There don't seem to be any easy answers to that question. We will tell you what parts we find inside the digital audio products we review, which will help you to determine some parts that are common to the good or bad sounding players or other digital components. Beyond that you'll have to rely more on subjective listening impressions from reviewers you trust or listen to more products than you otherwise might.

The articles that follow will offer our listening impressions of a variety of players and we'll tell you how we arrived at our conclusions. The rest is up to you. **APJ**

optical disc players

by Shane Buettner

*Dick has asked me to say some things to **Journal** readers about disc players—to include my recommendations on which players offer the highest level of audio playback quality and which players should be avoided. I've watched, listened to and reviewed nearly 100 disc players over the last three years and, believe me, I have plenty to say on this subject. Many of the players I've reviewed have DVD-Video, redbook CD, DVD-Audio and/or SACD playback capability. Let's start there.*

Most DVD players by the major manufacturers costing $250 or more offer video playback performance that will satisfy all but the pickiest people with the biggest, highest resolution displays.

I use $250 as a jump-off point because that's the price of Sony's DVP-NS755, a progressive scan DVD player with 12-bit/108MHz video DACs and multichannel SACD playback. That player is good enough that very few RPTVs have the kind of resolving power to differentiate between it and the players that better its video quality. But the audio quality is a different story entirely.

A lot of people I know want a DVD player that's also a reference quality CD player—a single box that performs two critical functions. I'm speaking of reference quality in audiophile terms, not in generic home theater magazine terms. Many people writing for the mainstream AV press have never heard a true reference quality CD playback system and can typically identify only the stuff that's better than the crap they've heard. I was in this camp until I started spending time at Dick's place during my tenure at *Widescreen Review*.

The sad truth is that, at least in audiophile terms, I've heard only two DVD players that are true reference quality CD players suitable for use in high-end audio systems: the Ayre D-1x, and Arcam's FMJ DV27. The simple explanation for this is that it isn't easy or inexpensive to build a good dedicated CD player. And DVD players are inherently different and require a lot more circuitry, costing more to build.

More elaborate drives are required, as are laser pickup mechanisms that can retrieve both DVD and CD standard data. Also needed are MPEG decoders and video DACs in addition to the audio DACs, and in many cases 5.1-channel Dolby Digital and/or DTS decoders along with six channels worth of DACs and analog outputs. On top of that, DVD-Video standards dictate that PCM and Dolby Digital signals on DVDs are at a sample rate of 48kHz, not the 44.1kHz of redbook CD. The better players need to operate at separate clock speeds too, one for each sample rate. That doesn't leave a lot of money for trick power supplies or high quality analog output stages. The bottom line is that there's more stuff in the box, and it costs more to make a box that excels at more than one thing. And DVD players are primarily video products, and that's where the priorities typically lie.

We'll look at the two DVD players that are at home in any high-end audio system as CD players, and a third that's a terrific SACD player. That brings up the question of what makes a good player.

What Makes A Good Player?

In his series on digital controllers for *Widescreen Review*, Dick pointed out that many controllers use similar parts—surround sound decoding/processing chips, DSPs, DACs, etc.—in spite of the wide disparity in prices among many of the products. Why pay more for a given box if it has the same parts inside that a competitor offers for half as much? Disc players aren't quite as cut and dried in this regard as controllers often can be, but there are some things I've observed while performing these reviews.

With digital, the first things I look at and think about are the DACs. You're never going to get any more resolution out of a disc player than what comes out of the digital-to-analog conversion. The best you can do after that is mitigate any further signal degradation, period. Of the few conclusions I can draw, one is of the inherent quality of the Burr Brown PCM-1704 DAC. The best digital controllers I've heard use this DAC, including the Theta Casablanca II Extreme DAC version, CAL's CL-2500, and Integra Research's RDC-7. Several of the best performing disc players I've heard also use it, including Denon's DVM-3700 (the first DVD player I heard that was also a respectable CD player), Ayre's D-1x, Wadia's 861, and Resolution Audio's Opus 21. Every product I've heard that uses this DAC competes with the best products in its category, or establishes the benchmark for a particular category. Odds are in favor of good sound when this component is used.

While many of the initial 1-bit sigma-delta DACs offered soft, ill-defined sound quality in the controllers and disc players that used them, the new multibit sigma-delta designs are a huge step up in performance. Dick first noticed this with the Sunfire Theater Grand II controller, which used Analog Devices' AD1853 DACs. We're hearing terrific sound from other products with multibit sigma-delta DACs sourced from Wolfson and Burr Brown, among others. Although many factors are certainly at work, these DACs seem to represent a huge step-up from 1-bit designs with respect to overall resolution of detail, focus, and soundstaging. Players that we've heard that use them include Arcam's DV 88 and FMJ DV27 players (DACs by Wolfson), Ayre's CX-7, and the Musical Fidelity Tri-Vista (both use the Burr Brown PCM-1738).

The track record for Cirrus Logic's sigma-delta DACs is much more enigmatic. Pioneer's DV-47A uses the Cirrus Logic CS4392, and both Classé's Omega SACD-1 and Philips'

SACD-1000 use the Cirrus Logic multibit sigma-delta CS4397. All offer disappointing audio performance. On the other hand, Philips' DVD-962SA also uses the Cirrus Logic CS4397 and Simaudio's Moon Stellar uses the Cirrus CS43122, a 5-bit sigma-delta stereo DAC with a specified 122dB dynamic range. Both sound terrific. The Simaudio features a massively overbuilt power supply and balanced circuitry in the analog output stages.

The power supply and analog output circuitry are other areas that have enormous impact on a player's sound quality, but exactly how that's accomplished is more mysterious. Some of the Sony players (the $1,500 DVP-9000ES and $3,000 XA-777ES for example) are built more robustly in both regards and sound much better than the other players in their price categories and aren't embarrassed at all by comparison to products that cost many multiples of their prices. Here are a few other things I've observed that may help you make better decisions.

Linear power supplies with larger (and/or multiple) transformers usually sound better than inexpensive switching power supplies. Sony's DVP-9000ES uses a linear supply with dual transformers. Arcam's FMJ DV27 and DV-88P players are virtually identical (including the DACs) with a couple of exceptions—the FMJ player uses an Accousteel chassis and has separate power supplies (and transformers) for analog and digital circuitry. The differences in audio performance between these two players are not subtle—the FMJ player sounds significantly better.

With the analog output stages, the answers are far less distinct. Ayre's D-1x shows how killer discrete circuits can be, but Ayre's CX-7 also shows just how close well designed ICs can come to that level of performance. Sony's DVP-9000ES demonstrates that off-the-shelf op amps aren't necessarily a death sentence for terrific sound either.

When you see XLR connectors on the back of the player make sure to find out, from either your dealer or another source, whether the player is truly balanced. Some products aren't truly balanced, but simply use a phase splitter in front of XLR connectors to create "balanced" connections. This can sometimes result in worse sound quality from the "balanced" outputs than the single-ended outputs.

Typically, DVD/CD/SACD/DVD-A players selling for under $1,000 don't offer any of the above accoutrements. More often than not, these players will have cheap, poor quality DACs, switching power supplies, and poorly implemented analog outputs stages. Caveat emptor. You're more likely to get great video performance and mediocre audio performance from most products in this price category.

Other Buying Tips

I attended an **Audio Perfectionist** seminar and asked Dick what attendees of his other seminars were especially concerned about. One thing that became clear to us is that many enthusiasts lack an appropriate strategy for building a system that's musically satisfying. That's easy to understand; there's a lot of misinformation out there to sort through. Although Dick and I will discuss system building strategies in future issues of the **Journal**, I want to offer a couple of quick tips here.

If picture quality is your priority, many DVD players available for under $1,000 offer good enough video quality for most displays. If you're willing to spend more on a player and display, and if you're also a hardcore videophile, the higher priced components may give you more of what you need. If you're more budget conscious, focusing your money elsewhere in your system will yield higher quality audio performance. Also, the CD's days are numbered. High resolution digital in the form of SACD and DVD-A are here, and at least one of these is likely to survive as a high-end niche format. Think before spending megabucks on a CD playback rig right now. Even if I had that kind of jack (which I don't) I personally wouldn't be considering spending $20K-$30K on a CD rig that doesn't also play back SACD and/or DVD-A, or at least offer an upgrade path to either or both. (You'll read about a $20K dCS CD/SACD rig here that I would spend the money on, if only I could.)

As you'll read below, $3,000-$3,500 can buy you a hell of a CD player. Better performance certainly is available for more money, and we've identified some of the products you should look at. But I think those of you on limited budgets would do better with one of these excellent $3K CD players, investing more money in your speakers, amps, preamps, etc. For an investment of $4K—the cost of a good CD player and $500-$700 for a used Sony DVP-9000ES—you can get surprisingly close to the best CD/DVD/SACD performance available at any price. Then, after the rest of your system is dialed in, you can look into upgrading these source components as necessary.

The Recommended CD/DVD-Video Players

In addition to playing DVD-Video discs another advantage of nearly all recent vintage DVD-Video players is that they will play back DVD-Video standard Audio Discs (DADs) from Classic Records and others. These discs have uncompressed two-channel PCM tracks at high resolution (24/96 in most cases). Although there are relatively few of these discs on the market, those I've heard sound terrific. Some of the audiophile labels, like Chesky and AIX, will be including high resolution, uncompressed two-channel PCM tracks on their DVD-A discs and it may be a significant benefit to you to be able to play these tracks, which sound noticeably superior to redbook CDs.

As I mentioned, I've only used two DVD players that are also reference quality CD players in the strictest sense. Note that, although I do have experience with a massive number of players, I've not yet heard the Meridian players, which also feature DVD-A playback, nor have I heard some of the other high-end jobs from companies like Krell and EAD. This isn't meant to be a totally comprehensive survey, but a summary of my experiences and recommendations of the players I have heard.

Ayre D-1x DVD Player

Ayre's D-1x sells for $8,000 with CD playback capability and no video, and for $11,500 fully loaded with audio and progressive scan video. The D-1x uses an outboard power supply with AC line filtering and separate choke-input filters to further eliminate power supply-related noise and artifacts. It eschews the use of op-amps for discrete circuits.

Charlie Hansen, Ayre's president and chief designer, believes this to be the purer approach, although much more expensive. When I interviewed Charlie, he analogized using discrete circuits with baking a cake—would you rather eat a cake that's made from scratch by a French pastry chef, or would you believe that Betty Crocker's off-the-shelf cake mix would taste just as good?

Two K-Grade 24-bit linear Burr-Brown PCM-1704 DACs are used in a dual-differential configuration for each channel of the D-1x. Burr-Brown's DF-1704 filter is used, which performs 8x oversampling with signals up to 96kHz. Since it is a DVD player, it will play back DVD-Video standard audio discs (so-called DADs) with high sample rates so long as the tracks are uncompressed PCM, not MLP-based DVD-Audio standard discs. The parts for this dual DAC combo cost nearly $46 per channel plus $10 for the filter, which totals just over $100 in parts costs for DACs and filtering alone. That doesn't sound like much, until you consider that many CD and DVD players use sigma-delta stereo DACs with integrated filters that cost $3 a piece, or even less (perhaps much less in mass quantities).

There are two master audio clocks for 44.1k and 48k signals, when one is engaged the other is turned off. At higher sample rates the 44.1k clock is used for 88.2k signals and the 48k clock is used for 96k signals.

Fully balanced circuitry is used, and the D-1x sounds best from its balanced outputs. As with all of Ayre's gear, no feedback is employed, and the chassis is non-ferrous aluminum which resists magnetic fields and eddy currents. Few players are in the Ayre's league with respect to construction and parts quality. If someone tries to sell you a player that's more expensive than the D-1x, beware that you're not likely buying superior parts, or construction, and therefore you're not buying more performance.

Video

For those who care, the D-1x tells an equally compelling story on the video side. Over three years ago when it was first introduced, Ayre's player had tricked-out (and expensive) 14-bit video DACs that sampled at 54MHz stacked in a differential configuration. (I'm still not aware of another player that uses dual-differential video DACs.) The D-1x uses Silicon Image's (formerly DVDO) progressive scan chip set for de-interlacing and 3/2 pulldown detection and compensation. With film-based material this chip set is second to none, and only Faroudja chip sets and the very latest chip sets used by Sony better its performance with video-based material. (Most concert videos on DVD, for example, are video-based.)

The only downside to the D-1x's video prowess is that, as great as it is, it's a 480p player. People with the kinds of systems required to see how great this player really is (i.e., expensive front projection systems) need the video scaled up to resolutions higher than 480p for optimum playback. Ayre's recent announcement of an SDI (Serial Digital Interface) output for the D-1x is a positive step. That will allow the D-1x to be connected to high-end video processors entirely in the digital domain, and may make the D-1x even more attractive to the high-end video crowd.

Performance

In my opinion, the D-1x is quite simply the best CD/DVD player available in terms of pure performance. I've heard nearly all of Ayre's products, and the first thing that jumps out is the complete absence of noise and distortion. Such is the case with the D-1x. The music is presented from a pure black background, and words like clarity and purity come immediately to mind. The audio quality is detailed, revealing nuances of your best and most familiar CDs that you've not heard before. There are layers of space and detail, front to back, with great spatial separation of musicians on stage. Image focus is razor sharp with amazing delineation. Sound permeates far beyond the speaker boundaries—the size of the soundstage will be determined by the rest of your system and your room, not the D-1x. Tonally, its performance is beyond reproach, with weight and authority at the low end, airy extension at the high end, and precision and focus at all frequencies in between.

The D-1x is so good that its only fault may be that it's just too good for the compact disc! It ruthlessly reveals the shortcoming of the format, especially when you put on a 24/96 DAD from Classic Records and hear how much better it sounds than redbook. As long as the rest of your system is up to the D-1x's standard, you'll get layers of spatially precise images front to back, and way out to the sides of the soundstage. But along with that resolving power comes a sound that can be a bit lean, and a little on the cold side.

Ayre's two players (the new CX-7 is covered later in this issue) seem to ratchet the music upward in frequency just a bit, lacking some warmth and therefore spotlighting the midrange and treble to a degree. Using cables with a warmer midrange will ameliorate this somewhat, but other deleterious side effects may accompany that desirable characteristic.

By comparison, Wadia's 861 shows nearly the same resolving power but images in a softer, rounder fashion and sounds more musical, more full-bodied and enjoyable to listen to in non-hi-fi terms. The Wadia, however, doesn't do DVD-Video, nor can it play back 24/96 DADs without using a compatible player as an external transport.

But listening to the music swell and become fuller and richer with the 24/96 material, while maintaining the same mind-boggling resolution, I came to think that the D-1x simply delivers everything the CD has, ugly warts and all. I personally hope that there's a high resolution player (SACD and/or DVD-A) in Ayre's future—a player of the D-1x's level could establish a new benchmark for high res players. On the other hand, the D-1x probably wouldn't be a good match for a system built around speakers tipped toward any degree of brightness. And note that my evaluation here is based upon the "listen" low-pass filter setting; the "measure" setting is noticeably less musical in comparison. Listen was the one that felt right to me and cemented in my mind the D-1x's place at the very top of available CD players. I've heard players that evoked a more favorable emotional response (including Ayre's own CX-7, which you'll read about later) but I've never heard one I could unequivocally say was better in terms of absolute quality.

With 480p DVD-Video the D-1x is just as resolute. Every detail is revealed, which imparts as much depth and dimensionality as I've seen from a progressive DVD player. The images take on that rarified illusion of looking into real three-dimensional space. If 480p is enough to drive your display, the D-1x is the best you'll find, although I must add that recent players, like Sony's DVP-NS999ES and Pioneer Elite's DV-47A, are more than good enough for most displays for a mere fraction of the price. Spending just $1,100 on one of those players would allow you to put a lot of money into the rest of your system.

Disc access and transport speed are not equivalent to the Sony and Denon players. But if you've got the money to spend, the D-1x is the undisputed heavyweight champion of pure performance CD/DVD players.

Ayre Acoustics
2300-B Central Ave.
Boulder, CO 80301
www.ayre.com
303-442-7300

Arcam FMJ DV27

The Arcam FMJ DV27 is just $2,599 and performs so well as a CD/DVD player that it will be difficult for many people to justify spending any more.

Technical refinements include Wolfson WM8740 multibit sigma-delta DACs, separate power supplies for analog and digital circuitry, a disc drive suspended on four semirigid tiers and a "Full Metal Jacket" chassis with two layers of Acousteel surrounding a layer of rubber to damp vibrations.

Three clock speeds are used: one for the video DACs and two for audio DACs—one for 44.1kHz and its multiples (88.2k and 176.4k), and another for 48kHz and its multiples (96k and 192k).

Performance

The Arcam FMJ DV27 is simply wonderful. While it offers very good overall resolution, its main strength is its exceptionally natural sound quality, especially with vocals. It soundstages and images right up there with the best of them, allowing sound that's not at all confined to the physical loudspeaker boundaries. It is slightly rough around the edges in the midrange, but I found that to be an attractive quality. That little edge makes for a visceral and immediate vocal presentation that I really get into.

Dick and I initially compared this player to the Wadia 861 and, while the Wadia was better, it took a while to figure that out and longer still to quantify the differences between the two players. The Wadia is more refined, with better definition in the midrange and extension through the treble, and a grander, more precise spatial presentation. Note that all these are subtle differences that will only be noticed in high resolution systems.

The Arcam's video presentation is also phenomenal, only bested in fine detail by the Ayre D-1x and the most recent vintage of mainstream flagship players like Sony's DVP-NS999ES, Pioneer Elite DV-47A, and Denon's DVD-3800. These newer players offer an extra dollop of resolution that will be noticed by those with the highest resolution displays. Those who aren't using high-end front projection rigs will likely never see anything to be concerned about from the DV 27. Its progressive scan solution is provided by Silicon Image and is excellent.

The DV27 is bested in audio quality by Arcam's own FMJ CD23 and some of the other (much) more expensive dedicated CD players in this survey. What sets it apart is its complete package of high performance with DVD-Video and CD at a price that's so much lower than Ayre's D-1x, which is the only player that outperforms it with both video and audio. At $2,599 it's nearly the best in both categories, and many people will be able to make a case for spending the extra bucks in other areas of their system. Also note that Arcam is working on a DVD-A upgrade for the FMJ DV27. Existing players will be upgraded in the field.

US Distributor: Audiophile Systems Ltd.
8709 Castle Park Dr. Indianapolis, IN 46256
www.audiophilesystems.com
888 272 2658

DVD-Video Players That Also Play DVD-A/SACD

Universal players (those that play CD, DVD-A, SACD and DVD-Video) are thus far a better idea than reality. In other words, I haven't heard one yet that offers more than mediocre performance with all formats. The typical pattern I've seen is decent performance with CD and DVD-A respective to the player's price category, excellent video performance, and poor performance with SACD. Players in this category include Pioneer Elite's DV-47A, Marantz's DV-8300, and Onkyo's DV-SP800.

Multichannel just compounds these problems—it's hard enough to get high quality two-channel out of a player let alone six channels with multiple formats (and that doesn't even address the issue that there aren't any reference quality six-channel preamps on the market). High quality DACs and analog output stages are expensive, and the cost increases threefold with multichannel. As an example, the Pioneer Elite DV-47A uses three Crystal CS4392 DACs for PCM and DSD in order to hit its $1,199 price tag. These integrated DAC/filters have an OEM quantity cost of about $3 a piece. I've heard only one DVD/SACD player I can unequivocally recommend, and no universal or DVD-A players. As a result, I own a Sony DVP-9000ES SACD player but not a DVD-A player. Let's discuss some specifics.

Recommended: Sony DVP-9000ES

Any discussion of this category of products has to start with Sony's DVP-9000ES. This DVD/SACD player was a steal at its introductory price of $1,500. It's overbuilt with a big power supply and tons of copper shielding, and slick good looks with a slim-line disc tray. Its DACs are LSIs built by Sony. It offers excellent performance with all formats except the CD, with which it offers merely competent (soft but inviting) performance.

This player is now discontinued, but can be found on the internet for well under $1,000 and it's tough to beat even by spending much more. The video performance is good enough that only the most recent flagship-level players outperform it, and even then not by a significant margin.

Its performance with two-channel SACD is so good that only the dedicated audio players like the Sony XA-777ES and Musical Fidelity Tri-Vista outperform it, and then only by a relatively slim margin. Although it is only respectable with CDs, it does err on the soft and inoffensive side, not the harsh, edgy side. And it sounds remarkably good with 24/96 DADs, suggesting that its sound quality increases commensurately with the quality of the incoming signal. The DVP-9000ES is an SACD player that will stand on its own in the highest resolution systems. Richard Hardesty and Richard Vandersteen each also own a DVP-9000ES, which says something.

For those who are curious, the new Sony flagship DVP-NS999ES is a better video player, and a slightly inferior SACD player. Although it offers multichannel SACD (compared to the DVP-9000ES, which is two-channel SACD only) it is not the overbuilt, over-achieving monster its predecessor was with two-channel SACD. At this point multichannel doesn't interest me nearly enough to want or recommend a new DVP-NS999ES over a used DVP-9000ES.

Almost Recommended Players

The following players are solid bargains and offer excellent video performance, but aren't quite as suited to dedicated high-end audio installations as the Sony DVP-9000ES.

Denon

I've always appreciated the audio quality of Denon DVD players starting way back with the DVM-3700 carousel player, which was a good CD player. As far as the mass market players are concerned, Denon's are always respectable. They typically offer as much resolution as any of them, if not more, but also have the relaxed, non-mechanical sound of the Sony players. Their recent players, the DVD-9000 ($3,800) and DVD-3800 ($1,199), are two of the very best video players available—both use 12-bit/108MHz video DACs and Silicon Image's vaunted SiL504 progressive scan solution.

Both are also very good with CD and DVD-A, and Denon players are right up there with Sony players on disc access speed and ease of use. The DVD-9000 uses Burr Brown PCM-1704s in a dual differential stack on the front channels, while the DVD-3800 uses the new PCM-1738. In spite of its build quality advantages, the DVD-9000 doesn't offer enough of a performance edge over the DVD-3800 to justify the massive price differential. The DVD-3800 is a superb machine and does everything very well.

Philips DVD 962SA

Although I was not at all impressed with the Philips SACD 1000, that company's first and most expensive DVD/SACD player ($2,000 upon introduction), the DVD 962SA is much better at just $599 MSRP. It has Faroudja progressive scan, with overall video performance that's just short of the Denon DVD-3800 and Sony DVP-NS999ES. More germane to this conversation, its SACD audio quality is right up there with the Sony DVP-NS999ES, although short of the excellent performance of the DVP-9000ES. A very solid double duty player; its clunky user interface is the only detractor.

Players To Avoid

These players aren't necessarily awful—though some of them are—but they don't represent good value in terms of performance and cost. These are not recommended for purchase but if you already own one you probably won't get cancer because of it.

Classé Omega SACD-1

I had high hopes for the $12K Classé Omega SACD-1. It's a two-box solution with a separate power supply in one box and a beautiful, top-loading player in the other box. The fit & finish of the shipping container is nicer than most players!

It uses three Cirrus Logic CS4397 stereo DAC chips for PCM and DSD—two chips stacked in a dual-differential configuration drive the balanced outputs and a single stereo chip for the single-ended outputs. The sample I heard was awful. It made both CD and SACD sound like some of the worst, most non-musical digital I've heard. It sounded so bad that I wondered if something was wrong with the player.

I've never used a player that malfunctioned in such a way that everything operated perfectly with bad sound being the only symptom but I suppose anything is possible. In any case, if a dealer auditions this player for you listen carefully to the player, not the salesman's accolades, and certainly don't be swayed by the fact that *Stereophile* includes this player in their annual recommended components issue.

Philips SACD 1000

You'll see this player at closeout prices of less than $1,000, but don't fall for it. It was overpriced at its original $2,000 price and the new players smoke it in audio and video quality.

With video it doesn't have progressive scan and frankly isn't very good. Like the Classé Omega player, it too uses the Cirrus Logic CS4397 DACs. Its audio quality, like some of the players below, is so poor that it makes SACD sound like mediocre CD playback.

Pioneer Elite DV-47A/Marantz DV-8300/Onkyo DV-SP800

All three of these universal players are based on Pioneer kits and offer similar performance. The Pioneer 47A is $1,199. The Marantz beefs up the power supply a bit and costs more at $1,599, while the Onkyo costs $200 less at $1,000 MSRP.

The video for all three players is red hot and among the very best available at any price although Pioneer's proprietary de-interlacing solution is now slightly behind those offered by Faroudja, Silicon Image, and even Sony's latest generation of video processing.

The CD performance is best with the Pioneer Elite player, which uses Pioneer's Legato Pro upsampling technology to eek some extra air and detail out of PCM material making it a respectable performer among DVD players in its $1,199 price range. The DVD-A performance of all three players is nothing to write home about, and SACDs sound mechanical and digital, like merely decent redbook CD playback. Dick once said of an early DVD/DVD-A player that it does everything it's supposed to do except sound good. That's pretty much the story here. Note that Pioneer has two more recent universal players that use different DACs—the Elite DV-47Ai and the Pioneer DVD-45A. I've not yet had a chance to audition these players, and perhaps they are improved. APJ

Dedicated Audio Players

The following players are for music only and we're giving them more extensive coverage with complete reviews. All are recommended and each has some special strengths.

Ayre CX-7

by Shane Buettner

The Ayre CX-7 costs $3K and boasts a new industrial design that matches Ayre's latest line of products, like the K-5x preamp. The chassis is aluminum and it's a pretty look, albeit spare. I admit to being a less-is-more kind of guy when it comes to this kind of thing, so its aesthetics speak to my sensibilities in particular. The CX-7 is a single box with an integrated power supply and uses ICs as opposed to the discrete circuits of the D-1x.

The digital front end of this player uses Burr Brown's PCM-1738 stereo DAC, which is billed as an "Advanced Segment" DAC, and a Burr Brown DF-1706 digital filter. The DF-1706 employs 4x oversampling, and is cascaded with the PCM-1738's integral 8x filter.

According to Burr Brown, 16-bit signals are interpolated to 24 bits, which then are split, or “segmented,” into the upper six bits of the signal and the lower 18 bits. The lower 18 bits are converted in a multibit sigma-delta process, while the upper six bits are converted in a linear conversion. This stereo DAC is configured for differential (balanced) output for each channel, and is a current output.

Ayre’s Charlie Hansen believes it’s a huge advantage to have a current output DAC, which allows him to design zero feedback current-to-voltage converters that are superior to the op amps used within typical voltage output DACs. Whatever the reasons, the early returns show the PCM-1738 to be something of a price/performance breakthrough. It costs just $5 (list price) per stereo chip and, as noted, only one chip is required for a balanced stereo output. That’s a fraction of the price for a dual-differential PCM-1704 set-up, and the players that use the PCM-1738 absolutely hold their own with the PCM-1704-equipped players.

The CX-7’s analog output stage is truly balanced. The CX-7 also has user-adjustable low-pass filter settings of “listen” and “measure,” and like the D-1x it sounds best in “listen” from the balanced outputs.

Performance

Although I did not have both players on hand for A/B comparisons, the Ayre CX-7 gives up remarkably little ground to my memory of its big brother, the D-1x. The CX-7’s sound is clean and pure, with plenty of resolution. Frequency extension on top seems roughly the same, but the bottom end of the CX-7 is a little fuller, although perhaps just a bit looser and less defined, resulting in a slightly (but decidedly) warmer overall character than the D-1x. I reiterate that the CX-7 also shares that characteristic of shifting the music up in register a hair—it’s not exactly bright, but it’s certainly not reticent in the midrange/treble, and the body of the music lightens just a hair.

The dynamic contrast is also staggering, and better than any player I’ve heard other than the D-1x. It’s spatiality and focus are terrific but somewhat flatter from front to back than the D-1x or Wadia 861. The CX-7 places musicians and vocalists on the stage with impressive solidity, although it’s slightly less dimensional and holographic than the Wadia 861. Still, at $3K the CX-7 is excellent in all respects and in some ways I find it even more emotionally engaging than its big brother, the D-1x, which too often reminds me just how flawed 16/44 CD really is.

If I were shopping for a CD player in the $3,000 range, the CX-7 and the Arcam FMJ DV27 are the two I’d consider. The Arcam doesn’t quite match the striking sense of clarity and overall transparency of the CX-7, but it has more body, enough detail, and it unquestionably makes music. And the FMJ is a kick-ass progressive DVD-video player and will also play DVD-video standard discs with uncompressed 24/96 tracks, with a full DVD-A upgrade in the works.

These are both excellent players and are close enough in audio performance that your preference for one over the other is likely to be system and taste dependent. Personally, I could live happily with either one. Audition both and see which one fits your needs and sounds best in your system. APJ

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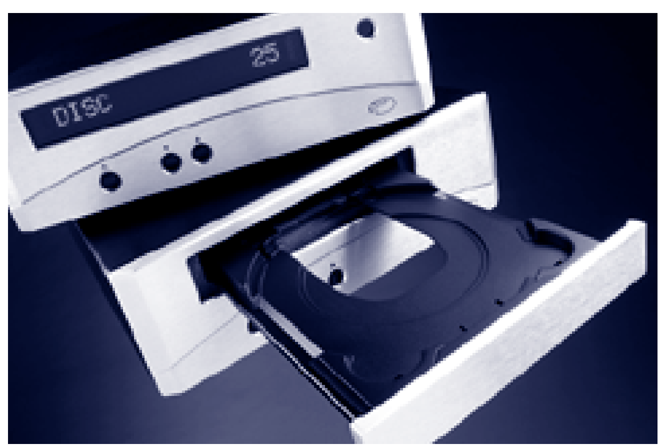
Boulder, CO 80301
www.ayre.com 303-442-7300

resolution audio OPUS 21

by Shane Buettner

Resolution Audio sells the Opus 21 direct over the internet and through a few select distributors in Europe. Selling direct allows Resolution Audio to offer this collection of engineering and high quality parts for a modest $3,500 (this price reflects an anticipated increase).



The Opus 21 is a two-box system, with the power supply and display electronics (i.e., the noisy stuff) separated from the transport, DAC and analog output stages. Both boxes are attractive, and the overall fit & finish is hot for a $3,500 player.

The chassis are both aluminum to eliminate magnetic interference. The Opus 21 is now shipping with an improved disc drive that is said to be quieter than its predecessor. It has a digital input that will accept external PCM signals up to 96kHz, and it also has its own volume control, which allows users to drive an amplifier directly from its analog outputs. Although the volume is controlled digitally, the volume control is implemented in the analog domain to avoid truncating any resolution.

The Opus 21 uses Burr Brown PCM-1704 24-bit DACs stacked in a dual-differential (balanced) configuration. Although these are the same DACs used in Ayre's D-1x, they aren't the K-grade variety. The K-grade DACs are guaranteed to meet certain measured performance specs (note that there is no guarantee that the ungraded DACs don't meet the same specs). In front of the DACs are Motorola DSP-based filters using software-based algorithms written by Resolution Audio that oversample at 16x with 44.1k and 48k signals, and at 8x with 96k signals.

A passive current-to-voltage converter is used, followed by an instrumentation amplifier to achieve the proper gain characteristics. Resolution Audio believes this approach offers superior common mode rejection compared with discrete circuits, although part of this tradeoff is that the instrumentation amplifier employs feedback. All in all, you'd be hard-pressed to find a player with a more unique design, or with as much money in parts/construction anywhere near a $3,500 retail price.

Performance

The Opus 21 has knock-out resolution from top to bottom, with excellent extension at both frequency extremes. The midrange is not only well defined, but smooth in character. Low level musical details are clearly audible. To a certain extent this resolution is a double-edged sword.

On the one hand, with good recordings it can be a striking experience to hear so much more of what's on the disc—especially with 24/96 material via the digital input. Most people have never heard the kind of resolution this player delivers, let alone at the $3,500 price point. I've heard much more expensive players that aren't as revealing.

The only criticism I can level at this player is that, as high in resolution as it is, the musical picture doesn't hang together in as coherent a fashion as I've heard from other players. This is a difficult phenomenon to explain. The best I can do is offer the example of a guitar playing.

While I could certainly hear all the pieces of its sound with great clarity—the strings being strummed, the hands sliding on the strings and hitting the body of the guitar occasionally, and the decay of the string's sound—those individual elements never coalesced into a coherent image of a guitarist playing his instrument.

Ayre's CX-7 is in the Opus 21's price range, and while that player doesn't have quite the same level of resolution in hi-fi terms, it does present images of musicians (and the music overall) as more of a whole, coherent picture that I found more convincing.

Consider a few things though. I use time- and phase-accurate Vandersteen speakers. Time- and phase-accurate speakers image in a more convincing, dimensional, and holographic fashion than conventional designs and I'm convinced they also reveal things about components that other speakers don't. In any case, I think that the $3,000-$3,500 price range is a killer elbow of price/performance for enthusiasts.

High res digital is here, and if you're an audiophile you're going to want that capability in the future. But for now, I think this is a reasonable price range in which to shop for something that will help you get the most out of your CD collection while the library of high res titles grows. Although I didn't respond emotionally to this player as I did to both the Arcam FMJ DV27 and Ayre CX-7, it is outstanding in many ways and many audiophiles will get off on the staggering amount of musical detail they're hearing and forget the rest.

Resolution Audio makes it easy for you to decide for yourself—they offer a 30-day in-home audition period. If you decide to return the player they'll charge you a 5% re-stocking fee to cover the credit card transaction fees and shipping charges. Check it out. APJ

Resolution Audio
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review

dCS verdi & delius

by Shane Buettner

I've been an audio enthusiast and hobbyist for years. To make a living doing something that is also my hobby is immensely enjoyable and satisfying, and I'm ever thankful for that. But, as with any job, a sense of monotony can occasionally set in, making things feel like a grind. Then a dream assignment like reviewing this dCS gear drops into my lap and bam! I'm charged with a sense of purpose and renewed (wild!) enthusiasm.

I imagine the anticipation I felt while waiting for the dCS gear to arrive is akin to what the Car and Driver guys feel when Ferrari's latest super car is scheduled for delivery—Corvettes are fun to drive, but they're not quite up to the same level of hedonistic thrill. Like the Ferrari, dCS' gear is nothing short of an attempt at defining state-of-the-art performance. I may never find out what it's like to fly around the race track in one of those 12-cylinder land rockets, but those guys probably aren't going to get to listen to dCS' Verdi/Delius combo either!

For the last two years at CES, the UK's dCS has been making spectacular digital sound in the various rooms occupied by Audiophile Systems Ltd., their US distributor. Among the products I've heard making killer sound at the shows are the Purcell digital-to-digital upsampler, which is said to convert 16/44 CDs to a 1-bit/2.8MHz DSD stream, and the Elgar Plus DAC. The depth and dimensionality this combo was extracting from CDs was truly remarkable. With this in mind I literally jumped at the chance to review dCS's Verdi CD/SACD transport and Delius DAC (little brother to the Elgar Plus).

While dCS's flagship playback system is the $34K Verdi/Purcell/Elgar Plus (yep, that's $34K with three zeroes—why do you think I went with the Ferrari references?), the Verdi/Delius combo comes in at just $20,990 ($10,995 for the Verdi and $9995 for the Delius). That's not chump change, but given that Linn, Levinson, Krell, and others sell CD-only players for as much or more it's not out-of-this-world either. Audiophile Systems considers the Verdi/Delius combo a statement of functionality and performance at this price point, and I agree. Before I get to all that though, just who is dCS?

dCS: From the Military to Your Listening Room

dCS takes great pride in regarding most manufacturers of high-end digital playback devices more or less as "assemblers." The distinction they offer is that, where most manufacturers source their DACs, filters, ICs, and other components from Burr Brown, Crystal, et al, dCS has designed their very own patented D-A conversion architecture called a dCS Ring DAC®. I'll get into what specifics I know about the device later in the review, but for now what's important to know is that dCS developed the technology, which eventually turned into the Ring DAC, as part of their work designing high performance A/D converters for British military airborne radar systems. The accuracy of these systems was the difference between a successful mission and big splat.

The reasons that high-end manufacturers don't make their own D-A converters are the expense and level of expertise involved. A lot of R&D, and a lot of money. Consumer electronics giants Sony and Yamaha, both of whom use their own LSIs for D-A conversion in some products, are the only other companies I can think of who make their own DACs. High-end companies typically don't have those kinds of resources, financial or otherwise. Due to their past in the defense industry, dCS does and that is a significant distinction for their products.

Verdi Transport

The Verdi CD/SACD transport is based on a Sony broadcast standard transport mechanism that costs a lot more than the mechanisms used in most players. How much better that makes it than a typical disc player I don't know, but considering that most consumer players use transport mechanisms that cost just a few dollars (OEM parts cost) it's obvious Verdi is a different animal. This drawer loading mechanism is equipped with dual lasers for CD and SACD. The drawer and mechanism are suspension isolated for vibration control.

The Verdi has every brand of digital output known to man,

including three AES/EBU outputs, one each RCA, BNC, toslink optical, and ST optical interface. There are two IEEE 1394 outputs for DSD data and wordclock inputs and outputs on BNC.

The 1394 designation defines the physical interface for the DSD data stream—the security protocol is proprietary to dCS, and the 1394 DSD digital output functions only with their gear (no open high res digital standards yet!).

The wordclock ins and outs allow the Verdi, or a connected DAC, to be used as the master clock depending on how the connections are made (more on this in the setup section below). There are also CH1 and CH2 data outputs (on BNC) that can be used in conjunction with the wordclock output to transmit 16/44 signals, which is more common in the pro world (dCS has their toes firmly planted in both the pro and consumer sides of the biz). A 9-pin Software Upgrade Connector for updates via PC is also included, and the power cord is removable.

That last point is an important one. The Verdi's digital processing and control functions are software based and can be upgraded with relative ease. Some upgrades can even be performed by simply putting a CD disc in the player to upload the software changes. I think upgradeability is a critical feature for a product at this level and price, and the Verdi clearly delivers on that score. Verdi also ships with a 1-year "guarantee" that is automatically extended to 5 years when purchased from a dCS dealer and registered with dCS. What's more, the guarantee is fully transferable to subsequent owners who will then be notified as updates are made available. How's that sound?

Delius

Delius is in many ways a scaled down version of the $15K Elgar Plus. Audiophile Systems describes the main differences between the two pieces as being the Elgar's improved fit & finish (although Delius obviously isn't lacking in good looks), and more advanced power supply and balanced output circuits. Delius is also specified as a 24-bit/192kHz-DSD DAC, but some explanation is required. Although its digital inputs accept high res signals (up to 24-bit word lengths with samples rates of 48k, 88.2k, 96k, 176.4k, and 192k) and it'll certainly play 'em, the Verdi transport (at least as my sample was configured) recognizes only 16/44 CDs and SACDs. It would not recognize 24/48 or 24/96 tracks from the video zone of DVD-Video standard discs. Using a Purcell digital-to-digital upsampler you can convert 16/44 to a "higher resolution" signal to send to Delius, but you must use a different external transport (like a DVD player) as a source for native high res PCM.

Delius uses software-based DSP-based processing that oversamples PCM signals to 5-bits at 2.8MHz for 44.1k sources (and its multiples, 88.2k, 176.4k) and 3.07MHz for 48k sources (and its multiples, 96k and 192k), respectively, with noise shaping employed to shift noise out of the audio band. Sounds familiar doesn't it? Substitute the 5-bits for 1-bit and it sounds a little bit (pun intended) like DSD, doesn't it? On the surface of it, this isn't much different than what happens in other players. The players covered here by Wadia and Resolution Audio also use DSP-based oversampling filters, only at different sampling rates than dCS uses. (See the reviews of these other products in this issue.) The difference is that, like the Elgar Plus, Delius uses dCS' vaunted Ring DAC® board.

As opposed to the off-the-shelf DAC chips used in virtually all other digital components, the dCS Ring DAC® is a whole daughter-sized board full of discrete resistors controlled by gate arrays. Ring DAC's 5-bit architecture operates at 2.8MHz and 3.07MHz and keeps the inherent linearity advantages of 1-bit sigma/delta D-A systems while minimizing (or seemingly eliminating) the disadvantages—namely clocking errors and noise. Its resistors are exceptionally accurate and don't drift with time or temperature changes. Errors are randomized and effectively cancelled by samples being taken from a broader number of current sources. The results are (claimed to be) linearity and resolution that are virtually unknown in any other D-A conversion system.

If you read *Stereophile*'s measurements of digital source components you know that the very best of them measure around 20-bits resolution at the DAC output, even if the DAC in question is specified as 24-bit. That's why I generally specify a DAC as being "compatible" with 24-bit signals instead of just calling it a 24-bit DAC. Notice that I didn't do that in this case. dCS claims the Ring DAC® to be capable of output resolution of no less than 24-bits (29.5-bits according to Audiophile Systems). That's a bold statement. Resolution this high is difficult to measure—some existing test gear apparently wasn't up to the task so dCS designed and built their own!

So, a few questions still come to mind. The dCS Ring DAC® operates at 5-bit/2.8MHz (or 5-bit/3.07MHz with 48k). What does it do with DSD, which is 1-bit/2.8MHz native? Or with the Purcell's upconverted DSD stream? Is the signal decimated to 5-bits? I don't know, but I can assure you that I asked dCS, who wouldn't tell me—trade secrets, you know. So, much as I'd like to tell you what this player is doing to make such ground-breaking sound, dCS is apparently afraid to let the cat out of the bag, lest their competitors take advantage.

Now that I've devoted so much space to the logical side of things, let's get the physical world. There is one set each single-ended (on RCA) and balanced outputs, Delius has two digital inputs each on AES/EBU and coax/RCA, one coax on BNC, and one each toslink and ST optical. There are two IEEE 1394 digital inputs, and wordclock input/output on BNC. Also included is a digital output on BNC that will output digital signals up to 24/96 resolution. Delius is also software upgradeable, and a 9-pin Software Upgrade Connector is included and like the Verdi, the power cord is removable. Hardware changes are sometimes required, but dCS makes them as inexpensive as they can for their customers. Delius comes with the same transferable warranty outlined above for Verdi.

Setup, Controls and Functions

The Verdi/Delius combo is made to stack. Verdi's chassis is literally two Delius chassis stacked together, creating the overall impression of three boxes on top of one another with the disc tray in the top box. Aesthetically, the dCS stack is as good looking as gear gets, in a slightly sci-fi way. A conversation piece(s) nevertheless.

Not only does the Verdi/Delius present a dazzling array of features, dCS also offers exceptionally well-written and understandable user manuals and even goes the extra mile by offering quick setup guides that show how to use each product together with dCS's other products. The dCS Verdi with dCS Delius guide shows how to make the recommended physical connections, and how to engage the logical factory settings. For simplicity's sake I followed the quick setup exactly, connecting the AES 1 and 1394 digital outputs of the Verdi to the AES 1 and 1394 inputs on the Delius, along with connecting the wordclock "out" of Verdi to Delius' wordclock "in." With the exception of the wordclock connection, this worked out famously. Verdi and Delius operate at two clock speeds. One clock is used for 44k sources and its multiples, 88.2k, 176.4k, and DSD at 2.8MHz. The other clock speed operates for 48k and its multiples, 96k and 192k. As mentioned, either Verdi or Delius can be used as the single, master clock for the entire system. The quick setup guide recommends a connection method that uses Verdi's clock as the master, which is how I used the system for a few weeks before I began putting this review together. A closer look at the Delius user manual and the dCS web site revealed their recommendation that Delius be configured as the master clock to reduce jitter (and increase sound quality). I tried this configuration and did hear perhaps a bit better sound. In any case, maybe amends should be made on the quick setup guide.

The dCS stack then functioned as the ultimate plug & play—putting a 16/44 CD into Verdi automatically switches Delius to the active AES 1 input, while putting an SACD in switches to the 1394 input and vice verse. Unlike a trip to the dentist it's quick and painless! Hybrid SACDs automatically boot to the (two-channel) SACD layer, unless instructed otherwise in Verdi's "favorite layer" menu selection. Indicators on Verdi let you know what kind of disc is playing (CD, SACD, hybrid SACD, etc.) and show the title and track text programmed onto most SACDs. The display is difficult to read at a distance.

The front panel display of Delius tells which input is active and by default displays the bit depth and sample rate of incoming PCM signals and merely shows "Verdi" with DSD. The only aspect of interacting with the dCS stack that isn't a snap in some respects is the remote control. Although it's one of those substantial and heavy-duty aluminum jobs, direct track access is a little weird. You have to hit one button to activate the number keys, and then you need to enter the number zero before the track number, even before a double digit number. To make a single digit track access quicker, two zeroes are required. Once a disc is playing, the choices increase further. For CD and SACD there are four separate filter algorithms that can be selected, and for higher resolution PCM two more filter settings become available. Delius remembers the last filter setting used for each format, and for each PCM sample rate specifically.

The filter settings can be implemented from remote control as well as the front panel display menus. For PCM I chose Filter 4, which offers the gentlest low-pass slope with the greatest time domain accuracy. For SACD, after some experimentation I ended up with Filter 2. The user manual offers no specifics on its characteristics beyond noting that Filter 1 offers the widest bandwidth and Filter 4 is for measuring, not listening. The concern with SACD is that the high frequency noise might damage wide bandwidth preamps, amps and/or speakers. Maybe it was just a sense of cautiousness playing with my mind, but I thought Filter 1 sounded a bit hashy in my system.

Trust me that what's described above is not even a summarization of Delius' functionality, but a mere mention of just some of its capabilities. Among other things Delius can also drive amplifiers directly from its analog outputs, and comes with a fully functional volume control. I didn't have a chance to fully evaluate this aspect of its capabilities as I'd been able to spend only a few weeks with it at press time. I plan on writing more about this product, and Audiophile Systems has mentioned that I might get an opportunity to use a Purcell with the Verdi/Delius stack. Either way, look forward to reading more about dCS' gear in future issues of the **Journal**.

Performance

From the first moment I loaded an SACD into the dCS stack I knew I was hearing more layers of resolution and clarity from my favorite SACDs than I'd ever heard before. It was stunning. Front to back, side to side, I felt like I was being taken farther and farther into the recording space. I've never heard any source component with more soundstage size or focus beyond the speaker positions. Tonally the dCS stack is extended at both frequency extremes and resolved in the midrange, but totally neutral as far as I can tell. What goes in comes out, with no unwanted emphasis anywhere. As thrilled as I've been by SACD sound in general, I didn't know just how much more information was there on the discs, waiting to be retrieved. Virtually everything I admire about the SACD playback I've heard from other players was there—the rhythmic "rightness," the natural detail and ease, the three-dimensional and convincing images—but I heard more of everything with the dCS stack.

Before reviewing the dCS stack I'd regarded SACD as a close runner-up to good vinyl playback, and had yet to be convinced that DSD could sound better than the very best analog. SACD doesn't sound digital in any way on the better players (particularly the Sonys) and shows an obvious and vast improvement in resolution of low level musical details and spatial accuracy and dimension (and anyone who tells you this isn't the case is really telling you a lot more about their system than the SACD format!).

I've regarded SACD as good enough that the convenience factors of an optical disc system started to make sense, but I still felt I was hearing more air, transparency and overall involving sound from vinyl. There has to be something going on if I'm still willing to go through the ritualistic abuse required for playing records! My experiences with the dCS gear and a new recording by the Eighty-Eight's label have changed my perception of SACD's place in the recording media food chain for the first time.

Clark Terry and Max Roach's *Friendship* was recorded simultaneously in DSD and analog. I compared the SACD version played on my Sony DVP-9000ES SACD player and the dCS stack against the vinyl playback version on a Linn Sondek LP12/Linto combo. Comparing both versions of this recording showed me two interesting things. First, I couldn't hear significant differences between the LP and the SACD when I played the SACD on the Sony, which in itself is impressive. Second, playing the SACD on the dCS stack, I heard some significant differences between it and the LP, and the SACD sounded better!

Friendship is an audiophile's dream—sensational music that's recorded to the highest aesthetic standard, with sound that's just amazing. Although there are a lot of great tracks here, "The Nearness of You" (track #13) is just stunning in its ability to conjure the image of Clark Terry playing his horn in my room. The nearness of *him* on this recording is unbelievable, an almost physical presence. The image is rounded and holographic with more convincing air than CDs ever show. You can hear Terry inhaling before every breath into the horn and the keys tapping into the instrument's body. Other quirky room sounds are apparent, enhancing the illusion of being in the recording space. The air into and out of Terry's horn has the most natural dynamics I can imagine, and that's one of the ways in which the DSD recording steps above the LP.

The excellent sounding LP is softer, slightly muted and somewhat compressed in dynamic contrast compared to the SACD. Terry's mini-crescendos keep going and going, and on the SACD those peaks hit a little higher in level with more openness and impact. For the first time, I had the distinct sensation that DSD could deliver playback that's more transparent than analog and freer of any form of coloration. That the SACD could better such a fabulous sounding LP was remarkable, and it took this dCS playback system to reveal that superiority.

Listening to SACDs through the dCS stack is consistently revelatory. Ray Brown's acoustic bass on Groove Note's SACD of *Soular Energy* is fat and natural, with rollicking pace—SACD gets the timing of the music just right. With Beck's *Sea Change*, the degree of texture and shading of the instruments and Beck's vocals are mesmerizing. Yet another standout is Analogue Productions' SACD of Bill Evans Trio's *Waltz for Debby*. Although the mix is very stereo—as in very left and right with little in between—this SACD has a startling live quality that captures the club sounds, as well as an incredible amount of airy detail in the cymbals and snare brushes. Whatever SACDs I played, the dCS stack communicated to me more of the music and the feelings behind the music.

With CDs the dCS stack is no less impressive, delivering massive but precisely focused soundstage, and layering the musicians on distinct planes front to back. As with SACD, the program material and the other gear in your system will determine the boundaries of the soundstage. The degree of clarity is striking, and the sound is highly resolved with excellent extension at the frequency extremes. It's very transparent to the source and yet there's a good degree of musicality as well, especially with the Filter 4 setting. The resolving power of the dCS stack is right up there with the Ayre D-1x and Wadia 861 as the most revealing CD playback rigs I've heard.

In my opinion, even with SACDs, the dCS stack leans just the teensiest hair to the analytical side. Buried deep in all its transparency is the slightest hint of coldness, a condition that is more noticeable with CD playback. As with Ayre's D-1x, I wonder if this system is simply revealing more about the source material than I care to hear. (I also wonder how far a Purcell upsampler might go toward eliminating this perception.)

Certainly that's a possibility because my perception did change with 24/96 material from an external source. Also, the dCS stack doesn't create the palpably holographic and convincing images delivered by the Wadia 861. The Wadia's images are simply more cohesive, rounded and convincing—more like analog in that respect than any CD player I've ever heard. Note too that I'm talking about the slimmest of margins here. The Wadia and this dCS stack are the two most astonishing playback systems I've heard with respect to combining high resolution and convincing musicality and imaging. I've not heard a playback system with more resolution than the Verdi/Delius combo, and only the Wadia eclipses it in musicality. This is one of the finest CD playback systems available, period.

Conclusion

The dCS Verdi/Delius is a state-of-the-art playback system and that is not the only important thing going on here. This dCS system didn't just reveal itself as an outstanding playback system. It revealed more of the vast potential of the SACD format itself than anything I've yet heard. The Verdi/Delius rig represents the best SACD playback I've heard and is competitive with the very best CD playback systems available at any price, even if it falls slightly short of the clear-cut benchmark status it has established with SACD.

Judging by what I've heard at trade shows, the addition of a Purcell to the Verdi/Delius system might change this perception and I hope to report to you on that in the future. And this isn't even dCS' flagship system. The Delius sounds so good that it's difficult to imagine what the Elgar Plus sounds like! Once a certain level is attained incremental improvements in audio quality often come with disproportionate price tags. Considering that it's still somewhat commonplace for CD playback systems to cost upwards of $20K without the ability to play SACD and/or any form of upgradeability, I think the dCS offers performance and flexibility commensurate with its admittedly gaudy price. Even at $21,995, the Verdi/Delius stack is what I would have if I could afford it. APJ

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review Musical Fidelity

CD/SACD TRI-VISTA

by Richard Hardesty

One player in this survey stands apart from the others. The Musical Fidelity Tri-Vista CD/SACD player is unique in a number of ways. It will play CDs, SACDs, and high res music on DVD-Video standard discs (DADs)—it's the only player that we've seen without a video output that will play DADs. It has a choke-regulated power supply with multiple transformers and uses miniature vacuum tubes in the output stage. It provides near state-of-the-art performance for both CD and SACD playback, and it's reasonably priced for a product with this build quality and level of sophistication.

The Company

Musical Fidelity is a UK company founded in 1982 by a classically trained clarinetist named Antony Michaelson. MF makes a variety of high-end audio components and has established an excellent reputation for performance and value. Michaelson demands that all MF products deliver musically satisfying sound and the samples that I've heard do.

The Tri-Vista CD/SACD player is a limited edition component with analog circuitry that utilizes miniature vacuum tubes. Only 800 Tri-Vista players will be made.

Outside

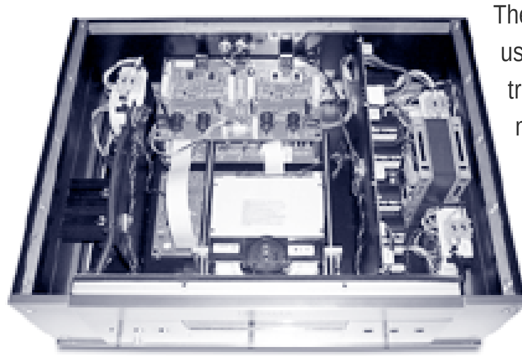
Hi-fi components made in the UK are often more compact than their Japanese or American counterparts with industrial design that could be flatteringly described as subtle. The Tri-Vista is

neither compact nor subtle. It's a brawny beast with bold styling. A very attractive component in my opinion.

The unit is quite large, weighing more than 50 pounds, and is beautifully finished with a thick silver face plate and black fluted sides. The front panel has six buttons to control transport functions, plus a power switch. The disc tray is centered just below the bright display, which shows the usual information and can be easily seen from across the room. The chassis sits on four large feet with translucent rings in the middle of each. These rings glow red while the power supply stabilizes immediately after the unit is first turned on, then fade to amber as the unit un-mutes and becomes operational. After about a half-hour, when the Tri-Vista has completely warmed up, the glowing rings around the feet turn blue making the product seem to levitate above the shelf. This trick looks unbelievably cool and everybody who witnessed it at my house was suitably impressed.

The Tri-Vista rear panel is very simple. There are coaxial and optical (toslink) digital inputs and outputs (one each). A single pair of single-ended analog outputs is provided on RCA connectors and the power receptacle is an IEC-type allowing power cord replacement. The remote control allows the user to access all player functions.

Inside



The Tri-Vista uses a Philips transport mechanism with dual Burr-Brown 1738 DACs. These DACs have differential output but MF, like many UK manufacturers, eschews balanced circuitry and provides only single-ended analog outputs. Two Crystal 8420 sample rate conversion (upsampler) chips are incorporated in what I presume is a cascaded array of filters. MF claims that the signal paths for CD and SACD are separate but offers no additional information. The Burr-Brown 1738s are claimed to be compatible with PCM and DSD data so perhaps one is used for each signal type.

The power supply in the Tri-Vista is very impressive indeed. Multiple transformers and choke filtering are employed. The power supply in this disc player looks more substantial than what you'll find in many amplifiers.

The output stage incorporates four miniature 5703 mil spec vacuum tubes. A small amount of negative feedback is utilized in the analog circuitry.

Sound and Performance

I listened to two samples of the Musical Fidelity Tri-Vista CD/SACD player, both borrowed from owners because there wasn't time to obtain a review sample from the manufacturer. Both samples performed in a consistent manner and delivered excellent, satisfying sound. The Tri-Vista offers uncompromised CD and SACD performance that rivals the best players we've heard for each format.

With regular compact discs the Tri-Vista presented sound that was highly detailed yet smooth. Images were well defined and three-dimensional with excellent focus. The sound was vivid with just a touch of character in the upper midrange.

For the first several days I perceived an upper midrange push bordering on aggression. After a few days of continuous play this settled into a very pleasant musical presentation with just a touch of a distinctive thumbprint added by the player. The presentation was always exciting and musical with what I would describe as a slightly "technicolored" enhancement.

The Wadia 861 is slightly more transparent and neutral and seems to retrieve slightly more detail from CDs. The Wadia creates a more palpable image with even more depth but the Tri-Vista delivered sound that was emotionally engaging, very dynamic and nearly as refined. I preferred the Wadia overall but the Tri-Vista was not far behind and it also plays SACDs and costs about two grand less.

SACDs were presented with a similarly enhanced sound. SACDs sound a lot better than CDs and that fact was clearly evident when comparing the formats on the Tri-Vista player. SACDs allow the top end to open up providing a more natural presentation of air and space and the MF player got all the musical information right. The sound was fuller and more substantial than what the Sony 9000 can provide and slightly more refined, although the Sony, which costs far less, was not embarrassed in the comparison. The dCS stack, costing more than 3 times as much as the MF player, delivered sound from SACDs that was slightly more transparent but the Musical Fidelity sound, while not as neutral and colorless, was perhaps slightly more musically engaging.

Conclusion

The Musical Fidelity Tri-Vista CD/SACD player is a remarkable performer. It can stand up to the very best CD players in head-to-head competition and it rivals the best SACD player I've heard. It has a slight character of its own but is never offensive while providing a listening experience that is always fun and exciting. At $6000 it's a bargain and a benchmark that sets new standards for performance and value. API

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Wadia 861
INTEGRATED CD PLAYER

by Shane Buettner

Over the last three years I've spent a lot of time listening to disc players at my home and have brought many of them to Richard Hardesty's listening room to listen to them in his system and compare them with one another and with Dick's reference gear. During that time, Dick's reference CD player has been Wadia's 860, which was then upgraded to the 861 (the 861 sells for $7,950). During this time many players have come and gone; few have even approached the Wadia's performance, and none have unequivocally surpassed it. In fact, I've yet to hear a single player that I prefer overall to the Wadia. Let's take a look at this reference quality player.

Design and Construction

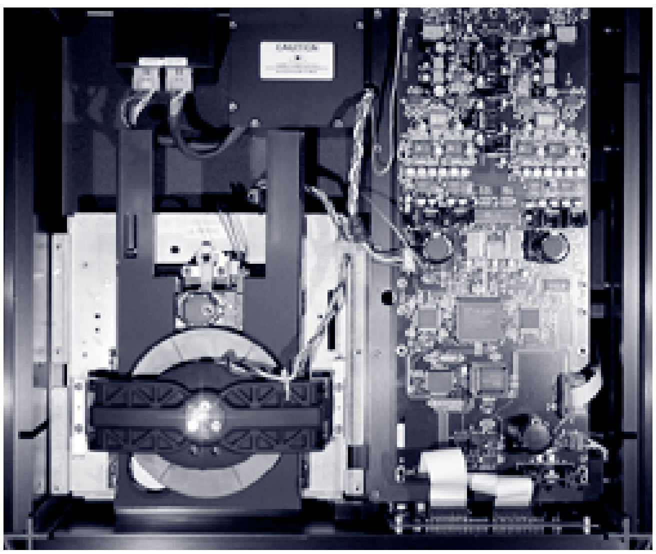
The heart of Wadia's digital technology is the DigiMaster™ filter system. This software-based filtering solution is implemented on high-powered dual Motorola DSPs. Software-based solutions are inherently flexible, allowing relatively easy running changes and improvements. The 861's DigiMaster version is 2.4, which engages 32x oversampling with 44.1k and 48k signals.

The 861 is also compatible with signals up to 24/96 resolution (oversampling at 16x with 88.2k and 96k sources), but, since its drive is a CD mechanism, signals with resolution higher than

16/44 must come from a player with a DVD transport via the 861's digital input. Wadia makes a point of oversampling at a direct multiple of the incoming sample rate as they believe that asynchronous oversampling (oversampling a 44.1k signal to 96k instead of 88.2k, for example) has deleterious effects on sound quality. I don't have such a rock solid opinion on this philosophy. I've heard some products that perform asynchronous oversampling that sound bad, but I've also heard others that sound excellent, such as Musical Fidelity's Tri-Vista.

The 861 uses two K-Grade Burr Brown PCM-1704s per channel in a dual differential stack for each channel. (The 860 used Burr Brown PCM-1702s, and I'll explain the differences I heard between the two iterations later in the review.) Following the DACs the 861 uses Wadia's proprietary Swift Current technology. Based on a "current conveyer" IC (Integrated Circuit), this solution performs current-to-voltage conversion without negative feedback.

Many DACs incorporate their own I/V (current-to-voltage) conversion with op amps that employ negative feedback. The players I've heard that avoid feedback in the analog output section image in a more coherent and convincing fashion than nearly all the players I've heard that use feedback. Whether this is evidence of feedback introducing timing errors (feedback is a time domain phenomenon) or simply that the I/V conversion internal to DACs is generally poor I can't say for sure. That the Wadia eschews feedback and sounds this good probably means something though.



The 861 has user-selectable low-pass filter algorithms with different roll-off characteristics. The setting that Dick and I both prefer is algorithm "A," which has the gentlest roll-off and the best time domain performance, resulting in the most organic and musically satisfying sound.

The 861's analog output stages are truly balanced, maintaining balanced operation from the DACs to the output. It also has a sophisticated digital volume control and high current output stage that allow the 861 to drive an amplifier directly as a digital preamp.

"The Wadia's construction materials are heavy-duty, and I mean that literally."

The Wadia's construction materials are heavy-duty, and I mean that literally. The 861 tips the scales at 48 lbs. and its remote control is heavier than a lot of *players* on the market. It has a fit & finish and overall look and feel that instill the kind of pride of ownership one expects from an $8K CD player. The body is a "monolithic" machined aluminum piece and the disc drive mechanism is a trick job by Teac with its own internal clamping mechanism to minimize vibration and a self-aligning digital servo to adjust tracking errors. Wadia claims that this "full" clamping mechanism offers superior performance even to top-loading mechanisms that use a manual clamp that fits only around the disc's spindle hole.

The 861 uses two separate power supplies with their own toroidal transformers. One supply is dedicated to the transport mechanism and servo, while the other powers the D-A circuits. The transformers are shielded in an aluminum subchassis with neoprene lining on the top and bottom of the enclosure to prevent any hum, noise, vibration and other transformer-related artifacts from contaminating the audio circuitry. As Dick's experience of upgrading his Wadia 860 to an 861 indicates, upgradeability is a key tenet of Wadia's philosophy.

In addition to upgradeable software-based DSPs, the Wadia 861 is physically constructed in a modular fashion with separate, replaceable boards for inputs/outputs, DACs, display/controls, and the transport mechanism. The rear panel is also easily replaceable to facilitate new interface options. (The same is also true of Wadia's more affordable 301 CD player.) Wadia takes pride in making sure your cheese isn't left out in the changing winds of the digital world!

Performance

The Wadia 860 was smooth, refined, and musically involving in a way that nearly all CD players just aren't. Images were full and round with superb spatial delineation. But, compared to the most detailed and resolving players (like Ayre's D-1x), the 860 was noticeably soft and warm. And I don't mean to imply that was a bad thing. Playback of 16/44 redbook CDs can benefit from the right amount of softening around the edges. Over the long haul I've often felt it would be more pleasurable not to hear as many of the CD's flaws (particularly the hard, glassy sound that bright recordings exhibit) even if that means losing some degree of resolution. The Wadia 860 was still competitive with the best players out there, and while I may have found it preferable in some ways to players with more resolving power, it was obvious that there were sonic tradeoffs for the "listenability" and musicality I enjoyed so much.

Being so familiar with other products that use Burr Brown's PCM-1704 DACs, Dick and I both expected that the 861 upgrade (which switched out PCM-1702 DACs for the PCM-1704s, among other things) might be the cat's pajamas if it bridged the gap between the 860's musical righteousness and the more highly detailed response of players like the Ayre. Man, did it ever work out that way!

The revamped 861 has it all—extended response at the frequency extremes, a midrange with incredible resolution of low level musical detail and texture, and a grand soundstage with precisely drawn layers of image focus. The 861 retained all that was engaging about the 860 and added more of everything. The 860 got the music right, but the 861 can take you into the recording space in a way its predecessor couldn't. And it does so with a deliberate grace, making instruments and vocals feel more natural, real and vital than any other CD player I've heard. Although dCS' Verdi/Delius stack and Ayre's D-1x at least match the 861's resolving power, they don't quite match its vitality. Images of musicians and vocalists are fuller, more three-dimensional and more holographic. It's as though there's more air around the images, which have mesmerizing solidity and substance.

Tonally, the 861 is rich, full and warm but doesn't lack any degree of air or extension on top. It captures the full foundation of the music, along with all the midrange and top end detail, without the slight leanness that accompanies the Ayre D-1x's resolving capabilities. The 861 also has a remarkably deep soundstage, front to back, as well as incredible ability to focus far out to the sides of the stage. At all times even the subtlest differing planes are revealed to create most convincing images of musicians in front of you, spatially differentiated from one another and clearly defined.

All of the 861's considerable attributes improve commensurately when a 24/96 signal is fed into its digital input. 24/96 is much more rhythmically right than redbook CD, with greater resolution of the timbres of instruments and the recording space. The ability to focus instruments way out to the sides and far back behind the speakers is particularly improved with 24/96 and the Wadia does more justice to these recordings than any player I've heard.

Conclusion

The Wadia 861 is one of the very best CD players available at any price, and is unquestionably the one that I enjoy listening to the most of all the players I've heard. The 861 is one of the players that you should look at very closely if you have the $8000 to spend on a CD-only player. Musical Fidelity's $6K Tri-Vista, for example, is nearly as good a CD player and not only plays SACDs, but is a cut above the Sony players with SACD performance. That being said, for those who demand the best and can afford it, the Wadia 861 is an absolute killer CD player. Its pure performance edge and Wadia's commitment to upgrades for its customers make this player a compelling option.

Special Note: As we were going to press Wadia informed us that a special edition version of the 861 is about to become available that offers an even more advanced transport mechanism and makes other incremental improvements to the player. We can't wait to hear it, and will be reporting on the 861 SE in a future issue of the **Journal**. APJ

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Methodology & Demo Software

by Shane Buettner

Here are just a few words on how these reviews were conducted and the software frequently used to evaluate the players. Much of the review work was done by me at my house in northern California, and I also brought many of the players with me to Dick's home in southern California so that we could listen together in his system.

My system includes a BAT VK-51SE tube preamp, and Audioquest interconnects and speaker cables. For the past several months, I've been using Arcam's FMJ DV27 as a reference for CD playback and have also used Sony's DVP-9000ES and XA-777ES SCAD players. Dick and I both use Vandersteen 3A Signature loudspeakers and 2Wq subwoofers high-passed with Theta Dreadnaught power amplifiers. We both use two multichannel Dreadnaughts in a vertically biamped configuration for each front channel. We both use Linn Sondek LP12 turntables, although Dick's is more up-to-date than mine. While his is a full Lingo/Arkiv/Ekos rig, mine is a 1985 vintage table with the fluted body and original power supply, Ittok tonearm and stock tonearm cable. I have updated to the Arkiv cartridge, and I use Linn's Linto phono stage. Dick uses the phono stage in his Ayre K-1x preamp. Both our rooms share double duty as home theater environments, and Dick's is acoustically treated more extensively than mine. (I couldn't get away with any more while maintaining a harmonious household.) I'm about to move into a new house that has a completely treated, dedicated listening room/theater (can't wait!).

I've been to Dick's place so often and listened to his system so much that it really is a second reference system for me. My opinions here are amalgams of my experiences in my room, gathered over several weeks or months of listening, combined with relatively infrequent but intense sessions listening with Dick in his room. For example, during a three-day period we listened to the Ayre CX-7, Resolution Audio Opus 21, and the dCS Verdi/Delius stack. We compared these players to Dick's Sony DVP-9000ES and Wadia 861 players, and to one another. We generally compared two or three players at a time, switching between inputs on the Ayre preamp. We typically listened for longer periods with a lot of material to gather our initial impressions before focusing down to switching between players with one or two cuts to discern and quantify the subtlest differences between the players. We took frequent breaks from listening to avoid fatigue.

In my own listening room, I typically listen to a component over days, weeks, or even months. During that time I respond to it emotionally or I don't. In either case, after that's established, I'll do some switching between the review component and a reference (or references) to quantify the differences between them with distinction. Neither of us performs blind testing because we both feel that lends itself to relying too much on the analytical side of the brain, which is the opposite of the side of the brain that music appeals to. As simple as it seems, listening to the music and seeing if you tap your feet or get goosebumps on your arm can tell you a lot.

System Revealing Software

I thought many of you would like to know what we listen to, and what we listen for when evaluating equipment. As I started putting this list together and speaking with Dick about it, this became a more puzzling effort than it initially seemed. Dick and I both respond to image focus, specificity, and palpability. So many of the best recordings for revealing the performance attributes of an audio system are small ensemble pieces and jazz recordings in particular. Although I like jazz very much, it's not the only thing I listen to, and in fact doesn't even embody the majority of my listening. And, when we're reviewing components, Dick and I both do what we always do: we listen to music we like for pleasure. All kinds of music.

We listen to classical and jazz, classic rock and new rock, some folk and bluegrass, and virtually all things in between. My quandary for this article was whether to write about music I like, or music that I can recommend to our readers that will allow them to determine if they can hear all the things in these recordings we're hearing. The answer is the list below that represents some of both.

The fact is that like most of you I listen to a lot of music that isn't "audiophile quality" in any way. And I'm still just as moved by it. Listening to these recordings on a revealing system might not be something I can wax poetic about in the fashion that's sometimes useful in communicating how a product under review sounds, but I can assure you that some of my poorer recordings connect me to the better components I review.

One of the recordings I own that will never make a list of CDs with great sound is *The Boatman's Call* by Nick Cave and the Bad Seeds, and I'll never forget the first time I heard that CD on a revealing digital audio rig. It still wasn't *Belafonte at Carnegie Hall*, but so much more of the feeling of the words was communicated to me through the inflections of the vocals. Cave writes the songs, sings, and plays piano, and his lyrics and playing took on an even more profound resonance. I'm getting the same goosebumps on my arms and neck that I had that evening as I write this.

Neil Young's recordings affect me the same way. His music and lyrical poetry mean a lot to me; they're evocative to me in almost the same way that well written short stories are. While they contain none of the story arc that novels and movies can portray, they present clear pictures of emotion, feeling, relationships, and situations. Among the many things I admire in his work is his ability to re-interpret his decades-old tunes and make them fresh again, and often simpler.

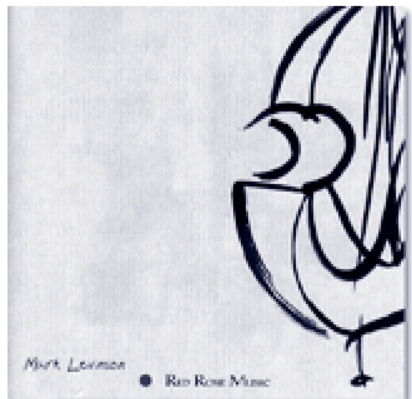
Neil Young Unplugged features the best takes I've heard on "Pocahontas," "The Needle and Damage Done," and "Helpless." This mostly poor recording was done on the spot at Universal Studios for MTV. I know that it means a lot to me when a component can give me just a little more of the guitar's sound, the body and hands on the strings, and more air in Young's harmonica. But I'm not sure there's enough to say about that to give you anything meaningful to listen to in order to evaluate your system.

And then there's the issue of genre. Fanatics of both classical and jazz can be Nazis about whether reviewers listen to their brand of music to evaluate gear. Classical by itself can certainly reveal a lot about a system's dynamic capabilities, its tonality, and its ability to transport the listener into a very large space. But people who listen exclusively for which gear best recreate the air and space of Carnegie Hall can often end up choosing components that don't provide the kind of focus that better serves other, more intimate forms of recorded music. Jazz suits me not only because I like it, but because the often intimate nature of its recordings is revealing in a way that lends itself to a number of different genres.

I guess what I'm getting at here is that we listen to music that's all over the map; we're just eclectic kind of guys. Many, many forms of music help us to form our opinions, but not all of them lend themselves to being written about as reference material. Trust though that no matter what your bag of musical taste is, we listen to it and it does enter into our opinions even if you don't find it mentioned in our reviews or lists of reference software.

This initial list includes a few digital recordings you might want to try in your own systems and listen for some of the attributes described here. We will be updating and adding to our software lists from time to time, both in print and on the web site. I'm starting here with just some of the recordings Dick and I both use of music we both like. Note that I'm sticking with digital because that's what was reviewed in this issue. Get ready for a truckload of vinyl recommendations in the next Journal!

Mark Levinson *Live Recordings at Red Rose vol. 1–SACD*



According to the liner notes, this pure DSD recording was made by Mark Levinson (the man, not the company) in his Red Rose Music store in New York City using two microphones, a preamp and a Sony DSD recorder. No editing, no jive of any kind. The result is perhaps the most stunningly natural set of recordings I've ever heard. The entire disc is full of tunes that I enjoy listening to from start to finish, and still do even though this disc is my most often-used demo piece! Here are some of the highlights.

It opens with Chico Freeman on tenor sax doing a riveting, breathy interpretation of Duke Ellington's "In a Sentimental Mood" accompanied by George Cables on piano. You can hear every part of the sax, from Freeman's mouth blowing through the reed (sometimes in a spitty fashion) to his fingers working the keys. And you can clearly hear (if not practically see) him shifting on his stool while he plays.

There are three solo piano pieces that are riveting: "Misty," "It Might as Well be Spring," and "Alligator Crawl." The piano sounds unbelievably lifelike, especially the weight of the lower registers. You can hear the player's fingers on the keys, and there's amazing flow to the timing of the music and the pedal work. The piano decay seems to just float onward and upward forever. On high resolution systems you'll not only hear the players humming along with the tunes or clearing their respective sinuses, you can hear some traffic noise and other ambient clues that take you beyond the music and into the Red Rose store.

There's also a killer reading of a Rupert Brooke poem, "Little Dog's Day," by Kim Cattrall accompanied by Levinson himself playing an old Italian bass. A very distinct image of crickets outside the house is drawn way up toward the ceiling, about halfway between the left speaker and the listener—you'll think they're outside your listening room! Levinson's double bass has weight, and you hear the cavity of the instrument as well as the strings and its snappy pace. Cattrall's voice is warm, full of subtle inflection and very natural. This disc dramatically demonstrates the superiority of DSD over CD and is an absolute must have for any audiophile—it's also a hybrid disc with a CD layer, so it's mandatory even for those without an SACD player.

***Ray Brown Trio** Some of My Best Friends are the Trumpet Players–Telarc CD*

Ray Brown was a world-renowned jazz bassist who played with many of the greats, and always put together first-class musicians to play on his records. This set features trumpet players like Terence Blanchard, Nicolas Payton, and Clark Terry, among others. The whole disc rocks but I listen most often to track 3, which reveals the full measure of Brown's nimble work on the acoustic bass. You can hear the body and strings, and Brown's hands sliding up and down as he plays. You can hear how well your woofers are integrated with your main speakers, with the right amounts of weight and speed when your system is dialed. The piano is set off to the left, on a plane slightly behind the bass, and the drum kit stretches out to the right from the middle of the stage. The instrument images and timbres are natural, and the horns just rip out of the background at a slightly higher point in space. On the better players/systems you can hear the trumpet player blowing air into the horn and other low level details. This is a terrific recording that's sourced from a direct-to-DSD recording down-converted to 16/44 PCM by Sony's Super Bit Mapping Direct process. Unfortunately, Telarc has told me that they don't plan to release this recording on SACD because it's not a multichannel master. I hope they come to their senses! I'd love to hear this disc on SACD, but until I do I'll just have to recommend this one on CD.

***Lyle Lovett** Joshua Judges Ruth–MCA Records CD*

The jacket of this CD indicates this is a full digital recording.

The music has emotional resonance and the sound is wonderful. Pianos and strings have natural foundation and decay. Russ Kunkel's percussion distinctively punctuates the music, with cymbal splashes moving throughout the soundstage. In addition to the moving (and often morose) lyrics, Lovett's voice and the way it's recorded here can tell you a lot. You can hear not only his voice, but the microphone and his proximity to it (especially on track 8). You can hear his intake of breath before he sings a verse, and even the parting of his lips. On brighter players his voice loses some depth, and the details noted above take on a spitty quality. Great music, great sound.

***Clifford Jordan Quartet** Live at Ethell's–Mapleshade CD*

Mapleshade's recording of Clifford Jordan's band is charged with the very essence of the club this music was recorded in. You can hear some buzz in the onstage electronics, tinkling glasses and murmurs in the crowd. The musicians are very clearly drawn in stage, on noticeably differing spatial planes front to back. The piano and cymbals have life and air, and the recording of Jordan's tenor sax is stellar. Like Levinson's recording of Chico Freeman, you hear every part of the performance, from Jordan's air through the mouthpiece, to the body and keys. Remarkable resolution and feeling, one of the best CDs I've ever heard and the performance of Jordan's band matches it with soul-felt intensity.

***Beck** Sea Change–Geffen/Universal SACD*

Beck's *Sea Change* has hardly left my SACD playing rig(s) since I bought it a little over a month ago. This disc sounds as good as a fairly production-heavy pop recording can sound, and I think artistically this is one of Beck's most compelling

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efforts—right up there with *Odelay*, my favorite of Beck's previous releases. Beck's lazy vocals and the guitars sound warm and natural with sensational liveliness. As densely layered as the mix is, this SACD never loses focus of any of the instruments. The drums and percussion in particular have a rich, fleshy texture. On track 6 there's a shaker that's delivered with mesmerizing clarity—you can practically see (and feel) the little beans moving inside their shell with the music. This is one of the rare pop recordings that manages to put the musicians in your room—great music bordering on audiophilia.

Alison Krauss *Forget About It–Rounder SACD*

Alison Krauss, a frontline foot soldier in the latest bluegrass movement, lands on the pop side of country folk with *Forget About It* and absolutely pulls it off. As with the Beck release noted above, this recording captures Krauss's strings and vocal work in particular, which is the heart of the record as one would expect. There's an incredibly natural feel to the guitars, with the body cavities vividly portrayed and the lid completely lifted with the strings' decay. It's very open and involving, and the emotional weight and quality of the music is phenomenal. Krauss's vocals are appropriately warm and cozy, which fits her style. The highlights for me are "Maybe" and "Ghost in this House." It's a gift to have a recording this inviting of such enjoyable music.

Nojima *Plays Liszt–Reference Recordings CD*

Dick introduced me to this recording by calling Franz Liszt the Eddie Van Halen of his time. This music, composed by a blazingly talented piano virtuoso, is certainly intended to be played by one. And Minoru Nojima is the one! This is an excellent natural sounding recording of a piano. It captures in great detail the percussive power of the instrument along with the authority and speed of the piece and Nojima's ability to express it to us. The piano covers the broadest area of the frequency spectrum of any instrument and, with this particular recording, you'll learn a lot about your system's abilities at all ranges. You'll hear low level details—the sound made as the keys are struck—and the wild variations in intensity of the notes that are played. APJ

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