Yamaha DVD-S1500 DVD Player

Manufacturer: Yamaha Electronics Corporation, 6660 Orangethorpe Avenue, Buena Park, CA 90620; 800/ 492-6242
Price: $450
Source: Manufacturer Loan
Reviewer: Howard Ferstler.

This review is a bit longer than usual (even for me), because it will deal not only with an SACD/DVD-A player but will also discuss the viability of those two technologies in general. Consider it as a combination player review and one of my regular *Skeptimania* columns lumped together. In addition, Dr. David Rich will offer up a tutorial on DAC design, including comments about the converter in this player.

A number of my upcoming *Scoping Software* recording review columns will deal with specific SACD releases and will involve the use of the player being reviewed here. Some of those will also critique the technology in general.

I reviewed the Yamaha DVD-S795 DVD player back in issue 80 and reviewed the more upscale DVD-S1200 in issue 90. I liked them both, although I did point out that one could at that time get video and audio performance from lower-priced versions that was about as good as what they each offered. The Yamaha units had some notable features that set them apart from lower-priced models from the competition available at the time, however.

OK, now we have still another player from the company and one difference between those earlier Yamaha players and this new one involves price. The DVD-S1500 costs considerably less than both of the others, while at the same time delivering everything that they could and more in terms of picture quality and sound. It still is not dirt-cheap, but the price is in line with what serious audio enthusiasts who would be reading this "get sensible" magazine might care to pay for a good machine.

In addition, the DVD-S1500 can do something that neither of the earlier Yamaha players could do: deliver SACD and DVD-A playback. Its versatility goes well beyond these audio formats, however, because it can also play Video CD, Super Video CD, CD-R and CD-RW (MP3 and JPEG supported), DVD+R, DVD+RW, DVD-R and DVD-RW materials that have been finalized. What's more, it can play back European PAL video DVD source material, in addition to standard, US source NTSC releases. Admittedly, the Euro-disc ability is something that would only matter to a handful of enthusiasts – and probably to only a very small handful indeed when it comes to those who would be reading this magazine.

The player is notable for its very low profile, being only 2.25 inches high. It is a standard 17 inches wide and is a bit more than 12 inches deep, and weighs in at a modest 7 pounds. This is in considerable contrast to the almost elephantine, but still very fine performing, Onkyo DV-S939 player that I reviewed in issue 86. The DVD-S1500 has a big edge even over the earlier Yamaha models when it comes to space/weight issues.

The small front panel is sparse and includes an on/off button and the usual stop, pause, and play buttons. However, it has no scan or skip buttons and leaves it up to the remote control to deal with those and other more esoteric functions. There are also several mode indicators, including one that shows that the "audio-direct" feature punched in via the remote has been activated. (This function, which the Onkyo player also had, disengages the video circuitry to possibly enhance the sound with audio-only source material.) Other front-panel indicators include those that tell you that a multi-channel source is playing or that you are playing a disc that allows auto down-mixing from multi-channel sources. In addition, there is a special indicator that lets you know when a DVD-A or SACD release is being played, as well as an indicator that tells you when the unit enters the progressive-scan video mode.

The rear panel is a bit more expansive than the front. In addition to a detachable power-cable hookup, it includes the usual composite-video and S-Video outputs and also includes the now common, three-jack component-video hookups. There is also a "scan mode" switch back there that would be used if one had an HDTV monitor that accepted progressive-scan inputs. Optical and coaxial outputs for digital signals are also included (for Dolby Digital, DTS, or PCM), as are two-channel analog outputs for a standard audio hookup to a CD player input and a separate six-jack bank of connectors for multi-channel audio outputs. These output left, center, right, left-surround, right-surround, and subwoofer signals.

The latter should be hooked up to the six-jack analog inputs of a suitable processor or receiver and normally they would carry the player's decoded-to-analog SACD or DVD-A program material. These six jacks can also output player-decoded-to-analog Dolby Digital, DTS, and even MPEG signals, mostly from movies. And *very* interesting indeed, they also are able to output Dolby Pro Logic II decoded signals derived from two-channel programs, at least if the sampling frequency of any SACD or PCM sources is below 88.2 kHz.

This is something to take seriously if one has a DVD-S1500, because it allows those with older processors or receivers that lack DPL II decoding, but which still have six-channel analog inputs, to now have DPL II playback decoding from their CD and other two-channel source materials. The Yamaha DSP-A1 processor/amp I reviewed in issue 72, and which I still have installed in my middle system is this way, as are a number of other receivers and processors I have reviewed. I consider this feature the DVD-S1500 offers to be a fairly big deal.

Yep, why opt for two-channels only with CD sources when you can get surround sound from them this easy? DPL II surround synthesizing is nearly always superior to standard two-channel playback, unless the center-channel speaker quality or the position of that center speaker stinks. With a player like this, one would hook up the device thusly to get optimum flexibility:

1) Connect the player's two-channel analog outputs to the CD input of their receiver for "pure" two-channel playback from compact discs. One could also use any DSP surround-synthesis modes their receiver offers for a faux surround effect.

2) Connect the optical- or coaxial-digital output to the receiver's digital input, primarily for DD or DTS playback. This would mostly be used for movie sound, although nearly all DVD-A music discs also have DD and DTS alternate sound tracks. This hookup could also be used for CD sources, of course.

3) Connect the six-channel analog outputs to the 5.1 analog inputs of the receiver for SACD, DVD-A, and DPL II playback, or for DD and DTS playback if an older receiver does not have those decoding functions via a digital input.

It is hard to imagine a better hookup arrangement, particularly if that older-model receiver lacks on-board DPL II decoding. The only fly in the ointment is that the DVD-S1500's on-board DPL II processing is factory set and does not offer the fine tuning the technology included in some upscale receivers. Still, it works well and was superior to standard two-channel playback with *all* of the source material I tried.

The DVD-S1500's remote is pretty basic. It includes some cool special-function buttons: audio direct, page turning for DVD-A still pictures, subtitle, angle, zoom, on-screen and front-panel status, shuffle, and an auto-scan feature to check out the first ten seconds of each track. And of course it offers the usual group of standard controls that must be included for decent control of the player: play, pause, skip, menu, etc.

However, there is one exception: there are no slow or fast "scan" buttons on the remote. Instead, if one wants to scan forward or backward they have to hold down the "skip" button for two seconds and then the skip feature is bypassed and scanning takes place. (The manual mentions this fast-scan feature on the remote control's description page, but indicated that it only works in the forward direction. Nope, it can work backwards, too.) Hitting "play" stops the scanning and returns the speed to normal. Unfortunately, there is no way to easily control the scanning speed. To do this one has to access the on-screen menu and awkwardly make the changes.

The minimalist front-panel readout is one of two things that bothered me about the player. For example, if one plays a compact disc the player's readout will briefly indicate which track is playing right after it is selected and then delete that information and present a continuous time-play readout. To check the track being played one has to press the "status" button on the remote. If one wants a continuous view of what is going on with any kind of source material (track that is playing, track-time readout, and total time of the disc) they have to turn on the TV set and read the information on the screen. The menu was at first awkward to navigate (the way the icons shift around is odd), no matter what kind of source material was involved, although after a while I got the hang of it. Still, it was anything but intuitive.

The second sore spot with me involved the player's cue-up time. No matter the source material, it took the device quite some time to access the data and begin playing.

One is normally used to this with movies and SACD and DVD-A sources, but the player was also very slow when initially dealing with compact discs. I mean, when typically accessing a CD after closing the tray the unit whirred and clicked and went on and on like this for a full 20 seconds before the music started. I got similar results with SACD material and DVD-A sources often took even longer. One, Big Phat Band's *Swingin' for the Fences* (Silverline DVD-A 82002) took a full 35 seconds, because the player actually had to access two menus (automatically bypassing the first after pulling it up) before the disc started to play. It is not a great big deal, and probably involves the player's ability to deal with so many different kinds of recordings. However, some users might wonder if the player is having some kind of initial hang-up problem when playing CDs or any other audio-only source materials.

In addition, after installing a disc most CD players and other DVD players give a total-time readout on their front panels and then go into a hold mode and wait for the operator to press play. In contrast, when a CD is first installed in the DVD-S1500 no total-time readout appears and the player begins playing after the lengthy cue-up procedure. This is not all that critical (unless you are a reviewer wanting to put total-play-time information into his review and it is not included on the CD box information package), but it might bother some users. Sure, I can turn on the TV to get the on-screen menu info, but who wants to turn on a big-screen TV monitor just to get total-time info about a CD?

Player idiosyncrasies aside, for most of us audio nerds the big deal with this player will be SACD and DVD-A performance. Unfortunately, both of those formats are nearly stalled when it comes to marketing, and it is quite possible that they will never amount to much more than niche formats. It is also possible that they both may end up being involved in a long and drawn out DOA situation, at least when compared to hotshot technologies like the CD, DVD-video, and MP3.

If we are going to discuss DVD-A and SACD as practical technologies instead of their sales successes or failures, we first need to come to grips with the bass management and distance compensation issues that involve the DVD-S1500. This can be important, because most surround processors and receivers do not offer these emendations with their 6-channel analog inputs. The signals are passed through unaltered.

With the DVD-S1500 you get full bass management from the six-channel (5.1) analog outputs with SACD source material. The subwoofer crossover points and slopes are user selectable: 60, 80, 100, and 120 Hz, with slopes of 12, 18, and 24 dB per octave. This is a great feature for those with sub/sat systems that have smallish satellite speakers. For those with full-range satellite speakers in combination with a subwoofer, the player offers an SACD "direct" mode that automatically bypasses all bass-management settings from the six-channel outputs. I really like these options.

Unfortunately, there is no bass management with DVD-A source material, no matter what speaker-size settings you choose from the player's menu, meaning that if you use the above-noted (and typical) small-satellite sub/sat system the small satellites will be getting full-bandwidth bass right along with the subwoofer. Strangely enough, there is also no bass management with CD source material when using the six-channel output's stereo-only mode, although the DPL II mode, which is a more viable option in every way I can think of, does manage the bass.

Fortunately, full-bandwidth signals from the player's standard two-channel outputs from CD sources also allow one's receiver to apply bass management in the usual manner. Interestingly, with the six-channel outputs playing CD and DVD source material you do get the ability to independently balance the volume levels on each channel. However, you do not get this with SACD sources.

While DVD-A is shortchanged when it comes to bass management, it is SACD that is shortchanged with distance compensation. DVD-A gets the ability to adjust for different speaker distances to the listening position and full compensation abilities are also provided for DD and DTS sources. SACD gets no compensation at all. For it to work at its best with SACD, the five satellite speakers all need to be similar distances from the prime listening location. This is a weird situation and is probably related to the political machinations that occurred when the parameters for these new technologies were being worked out.

I mentioned above that most processors or receivers do not offer bass management or distance compensation with their six-channel analog inputs. Even the Yamaha RX-Z1 receiver I reviewed in issue 93 lacks these features, because it does its bass management in the digital domain, independently from the six-channel analog inputs. It manages bass from standard, two-channel analog inputs, because those are digitized prior to being run through the amplification and any DSP surround functions. However, the six-channel analog inputs are run directly to the amp sections via the volume-control circuits.

One exception to the bass-management situation with receivers and processors (there are more exceptions, I am sure, but I have no experience with them) is the above-mentioned Yamaha DSP-A1 integrated amp that I use in my middle system. That unit does its bass managing in the analog domain, after digitizing and DSP operations are completed and the signals converted back to analog prior to basic amplification. When you set up the DSP-A1 to bass manage digitally connected DD and DTS sources after they are converted back to analog for amplification it also applies the same manipulations with the six-channel analog inputs.

Consequently, when using the DSP-A1 you can zero out the DVD-S1500's bass management with all SACD sources (remember, DVD-A is locked in with no bass management to begin with) and then get uniform bass management from *all* inputs, analog or digital: CD, DVD-A, SACD, DD, and DTS. The result is that a reviewer like me can not only enjoy the musical sounds of these new formats the way they were meant to be enjoyed, but can also do critical comparative listening without being haunted by the prospect of mal-adjusted (or non-adjusted) bass management screwing up my conclusions.

Unfortunately, the DSP-A1 does not have distance compensation with the six-channel analog inputs. (It obviously has them with all DD and DTS digital inputs.) However, and fortunately, in my middle system all of the speakers are nearly the same distance from the listening position, so the distance-compensation adjustment (with DVD-A, in particular) can be zeroed out. Yes, my middle system is ideal for evaluating these new surround technologies. OK, why waste more time. Let's cut to the chase: Yamaha vs. Yamaha.

What does this last sentence mean? Well, it means that I took a very good CD recording (one that I still consider a reference standard for two-channel PCM audio sound), and after applying some very good Yamaha DSP surround enhancements that the DSP-A1 offers, compared it, A/B style, to the same recording produced with SACD surround technology. The recording was the Heinrich Biber and Johann Schmelzer recording *Seventeenth Century Music and Dance from the Viennese Court* (Chesky CD173 and SACD 262). One advantage with this particular comparison was that the relative levels between the two presentations were very close, thereby eliminating the typical "louder sounds better" phenomenon.

The main-channel speakers were Dunlavy Cantatas, reviewed by me in issue 87, and still installed in my middle system, with the Cantatas pulled out from the front wall several feet and positioned about 9 feet apart. The surround speakers were wide-dispersion Allison Model Fours, located well out to the sides, somewhat behind the listening position and about six feet from the floor. (Room-power response curves I ran on all of these speaker systems can be found in issue 95.) The contest also included the front "effects" channels the DSP-A1 offers with its hall-simulation modes, and the speakers up there (on the front wall, six feet up and essentially flanking the Cantatas) were a pair of modified Radio Shack mini-speakers. (I had modified them by replacing the tweeter and crossover network with Allison versions and using better acoustic stuffing inside of the box.) The subwoofer in this system is a Hsu TN1220, reviewed by me in issue 67.

The listening room is roughly 17 x 22 feet, with an 8-foot ceiling and the listening position was about 10 feet from the axis between the main speakers. If this face off were not able to highlight the surround, bandwidth, and noise-level advantages of SACD, nothing else would be able to, either.

In this case, the SACD release was a 4.1 job, with no output from the center channel. Because of this, rather than let the DSP-A1 apply surround enhancements to the CD that included a derived center feed, I chose to make use of one of its standard Yamaha DSP hall simulations, notably the one labeled "Hall C." (The processor's manual says that Hall C simulates a European, "classic shoe-box type concert hall with approximately 1700 seats.") This function has the left and right channel signals go to the left and right main speakers unaltered, with DSP applied to the side/rear surrounds *and* the front "effects" channels. This is in contrast to the SACD version playback, which had only the left and right mains and the left and right side/rear surrounds active. This channel-count difference will mean a lot, as we shall see.

The results? Well, the surround-sound SACD sounded better than the CD when the latter was played back with only two-channels in operation. (It is easy to switch to straight-stereo CD playback from any of the Yamaha DSP modes to compare processed and unprocessed playback.) It would be hard to see how the result could be otherwise, given the impact of the surround channels with the SACD version.

However, I think that the CD won the contest by a slight margin when the Yamaha DSP hall-simulation circuits were engaged.

Yep, with the CD, I believe that the incorporation of those two "effects" channels up front, in addition to the standard hall ambiance applied to the rear/side surrounds, managed to simulate a concert-hall sense space better than what the SACD release could deliver.

Now, this is obviously a matter of subjective opinion and some listeners would have no doubt preferred the SACD version, mainly because of the somewhat tighter soundstage focus. However, I felt that the front "effects" channels opened up the sound and allowed the CD to sound better than the SACD version, particularly with tight-focus main speakers like the Cantatas. Please note that no matter which version might be preferred, the contest was no walkover. We are discussing a taste-related issue here.

OK, let's think about this a bit. It is possible that the SACD was not originally mastered with surround sound in mind, being engineered by the very talented Miguel Kertsman way back in 1997. My guess is that the Chesky engineers had to work with the material in such a way when producing the later multi-channel version that they essentially had to simulate surround sound from a mix that was originally set up to deliver a fine two-channel program. That with their obviously very good studio processing hardware they still could not match what the DSP-A1 could do at home (at least in *my* room) with the CD version is a credit to the Yamaha DSP technology.

Of course, it is also possible that the master tapes were indeed configured for future surround-sound productions, and in that case the Yamaha DSP ambiance simulation processing looks even more impressive.

I do have to make one point. Yes, with multi-channel materials that use the surround channels *only* for simulating hall ambiance, I believe that good, home-based DSP ambiance-generating technologies (this would also include DPL II and DTS Neo:6 ambiance-extraction technologies) in combination with good two-channel source material will be as successful at simulating a live-music space as good SACD and DVD-A surround source materials. This certainly would apply to most classical material and a lot of acoustic jazz, too. However, with recorded pop music all bets are off.

Most pop music is *not* recorded with the intention being to simulate a live performance in a hall, club, auditorium, etc. Rather, they are typically engineered to be ends in themselves. This means that the engineers often place performers (vocalists, drums, horns, and even pianos) in all of the channels, essentially putting the listener into the middle of a musical soundfield. With that kind of material, SACD and DVD-A surround recordings (as well as music recorded with DD or DTS technology) have a major advantage over synthesized ambiance from two-channel inputs.

There are classical-music exceptions, of course. Think of some of the stuff Berlioz did, as well as some church-choral music, and of course something like the *1812 Overture*, would probably sound really impressive with the cannons coming from all around you. And if the listener also wants audience sounds (applause, chair squeaks, coughing, etc.) around him when listening to classical performances recorded live then obviously surround-sound recordings have an edge. Still, for most acoustic-music recordings that are to simulate live performances, DSP ambiance simulation working with good two-channel inputs will almost always sound as realistic as the surround-sound versions.

Now, this leaves only one other item to deal with regarding the supposed superiority of SACD compared to the compact disc: *per-channel sound quality*. Well, if it was there I did not hear it. The results were similar to what I experienced with DVD-A materials. The extended bandwidth and lower noise floor of the SACD simply did not mean anything as best I could tell. The CD was more than quiet enough (the major background noise involved non-obnoxious hall artifacts and possibly very low-level microphone noise and not the digital technology) and the extended bandwidth above the top audible octave provided by SACD technology remains, in my opinion, laughable overkill.

Of course, this only involves one comparison. It is possible that other contests would lean in favor of the SACD versions. However, I have pointed out before that this particular CD is one of the very best sounding concert-hall recordings I have ever encountered (I often use it when doing my speaker testing A/B comparisons), as well as one of the best baroque-ensemble performances that you will ever hear. I think that the technological excellence of any CD is important in a face off of this kind, because it reduces the chances that it would sound worse than an SACD because of poor mastering done with the former. If SACD is to better what CD technology can offer it has to be able to surpass the very best example of that technology.

In this case, the only way the SACD surpassed the two-channel CD version involved the additional channels. However, once the Yamaha DSP-A1's ambiance-simulation circuits were called into play to assist the CD by simulating additional channels the contest was over. The processor/CD collaboration delivered the superior goods. The good news about this is that this kind of assistance can be applied to every CD already in one's collection if one is willing to spring for a good DSP device.

I did manage to listen to a number of other SACD recordings, and those will be reviewed in my *Scoping Software* column, possibly in this issue. I also listened to a number of DVD-A releases on the player, most of them previously reviewed by me after being played on the Onkyo DV-S939 installed in my main system. The sound was notably good if the discs were recorded well and often a major flop if they were not. (Flop status is not uncommon at all with quite a few DVD-A and SACD releases, due mainly to them being remastered from rather old source materials.)

One excellently recorded DVD-A release, the *Swingin' for the Fences* jazz item noted previously, has a genuine center channel feed and it actually sounded terrific even with the NHT VS1.2 center speaker in my middle system mounted fairly high up on a big-screen TV monitor. (Fifteen inches higher up than the vertical source center of either of my Cantata main-channel systems.) The centered up trumpet section actually sounded like it was at the back of the ensemble, located fairly high up on risers.

However, speaker-location issues aside, SACD and DVD-A sound quality will have far, far more to do with the recording and mastering techniques involved (particularly involving microphone quality and placement, as well as mixing judgments) than with the disc technologies themselves. And that's a fact.

OK, so what do I think of this Yamaha DVD-S1500 player. Well, I think it is a really nice unit. I went over some of the problematic characteristics previously, and while they might bother some users, I think that most people will be thrilled with what this device can do. For one thing, it can be shoehorned into a squashed-down rack space that many other upscale players can only dream of achieving. Yes, it is expensive by Best-Buy budget player standards, but it is cheap compared to some other upscale units and it can almost certainly play SACD and DVD-A releases as well as any of them. Actually, it probably can do this better than most, due to the admittedly not quite comprehensive bass-management and distance-compensation features.

The video performance was first-rate, and so was the DD and DTS audio performance. I gave it a defect-tracking test with the Canadian *CD Check* disc and it tracked cleanly to defect check level 3, with substantial interference (clicking) noises at levels 4 and 5.

Level-3 performance will handle just about any kind of defect one might encounter with a CD, although I have to admit that some of the other players I have reviewed in the past could track level 4 cleanly. (The earliest Yamaha player that I reviewed, the DVD-S795, could do this, and the later DVD-S1200 had only minor problems with level 4.) Given that players I have fooled with that could only track level 1 cleanly still sounded superb with musical source materials, any hair splitting over greater defect tracking abilities is just that: hair splitting.

Yep, the DVD-S1500 is a fine player, and I have no problem recommending it to individuals who want to play back SACD and DVD-A materials, in addition to compact discs and DVD movie releases. Topping things off, its musical performance with compact discs is equal to that of any other players I have auditioned.

I pulled the cover off of the unit and gave the digital to analog converter (DAC) number and some other internal information to Dr. David Rich to research. The model CS4382 DAC was built by Cirrus Logic. These are the folks who now own Crystal Semiconductor, a division of the company that does data converter design work, and Dr. Rich checked the Cirrus data sheet to come up with some info. He indicated that the most important thing that separates the great players from those that are merely good involves the quality of the DAC. Below is some technical info that is a review all by itself, courtesy of Dr. Rich. You will not see technical summary like this in our magazine very often, so pay attention.

-HF

Dr. David Rich on DACs, including the one used in the DVD-S1500: The CS4382 DAC delivers better than average behavior. To make life easier for our readers I am converting specification from the data sheet into effective number of bits (ENOBS). From the data sheet we find the CS4382 has typical 19-bit performance for noise, and with worst case 18-bit performance. When a manufacture states that a parameter is "worst case," this means that another manufacturer using the part in his product can test that part and expect it to meet the specification.

Unfortunately, a specification marked "typical" on a specification sheet can be a problem, because many manufacturers do not say what "typical" means. However, any decent designer designs only with worst-case numbers in mind if the parts performance for what he is designing can affect the stated specifications of the final system. For example, the motor in a DVD player had better be able to spin fast enough so the player can read the DVD. A "typical" spin speed in the motor's specification world not cut it. In other words, the designer would want the motor manufacturer to supply a worst-case motor spin speed to make sure that the player can always meet minimum standards.

With some DA converter specification listings, only the "typical" specs are given. If the final-product manufacturer uses those chips and has done no assembly line testing to insure that DVD players delivered to consumers meet some minimum specification, many of those players may not do so. This is the case, unless the manufacture indicates the specification is guaranteed, as is the case under federal law for amplifier power output under a specified load, frequency range and THD. To my knowledge, the only hi-fi company that currently states that the all the specifications they issue are guaranteed minimums (or maximums) is Accuphase.

The noise figures I have just given in ENOBS are measured by Cirrus Logic with an "A-weighted" filter. That kind of weighting is applied by placing a filter before the noise meter. The filter is said to allow the measurement to better reflect how the ear perceives noise level changes, meaning less sensitivity at the high and low end. Weighting also makes DAC noise performance look better since some noise has been filtered away. However, it really has no application here, since a properly designed DAC should have a flat noise floor, although cheap DACs can have an increase in noise at the low end. This is a sign of a design compromise that we do not want hidden under an A-weighted rug.

Even when no weighting is used, the DAC in this player has a little more than 17-bit resolution. Distortion at -20 dB input level is a little better than 18 bits typical. At full scale it drops back to 16.5 bits. Worst-case, full-scale distortion is slightly below 16 bits. These distortion numbers are for 1 kHz. The numbers usually get worse at higher frequencies, but the data sheet does not list distortion results at higher frequencies. The decline of the ENOBs with increasing input signal level frequency at full scale is one of the key benchmark tests for evaluating the usefulness of a general purpose DAC in a particular application space. For an application specific chip like the CS4382, the information sometimes does not make it to the data sheet.

That said, Cirrus supplies almost all other dynamic specification about this chip, whereas data sheets for chips that do not perform as well, but are likely less expensive, may contain no worst-case numbers, and numbers without A weighting will not be listed by manufacturers. Providing more detail on the data sheet shows the company has confidence in the design's ability to deliver these numbers when the devices are mass produced.

How many bits of noise level headroom do we need? Well that depends on how quiet your room is and the maximum loudness you will tolerate in that room. (This is the all-digital-ones level of the CD – the largest signal level the DAC can reproduce.)

Of course, other components in one's audio system must also have noise specs as good the DA converters. Most home electronics will not hack it, since a maximum signal to noise ratio of 110 dB is the equivalent of 18 bits. And of course the recordings must also have been created with very low microphone noise. In addition, they must have low noise in the analog signal path that follows the microphone (analog level adjustment and equalization may sometimes be used in the production of a modern CD), have low noise in the studio or concert hall, and also have sufficiently low A/D converter noise.

Data in conference and journals have presented information that points to 18-bit equivalent signal-to-noise levels as the required *minimum* for a professional studio (designed for very low background noise, which may only be achieved with special construction techniques and materials). I do not recall the maximum signal level (all ones) was used for the tests but I assume it was at least movie theater level loud.

With respect to distortion, again we must consider the rest of the equipment in the system. Almost all of you have seen THD vs. level graphs and will recall they rise as the level gets higher. At higher signal levels, more nonlinear effects of the electronics are uncovered. This is true with the DAC as well as analog components. In most cases the power amp will dominate a systems-distortion level at maximum signal level – the point where the power amp is about to clip, which in a digital system should be set to correspond to the all ones digital representation of the loudest signal level on a CD. Only the very best power amps could match this Cirrus converter's distortion level at its worst-case distortion specification.

Before we get too excited, please recall the DAC in a DVD player is only in use in SACD or DVD-A modes when the analog pass-through of your AV receiver is active. In this mode, all the good things your AV receiver can do (advanced digital bass management, multi-band EQ, multi-channel synthesis etc) are bypassed. When playing normal CDs (via the digital hookups to the receiver and not the analog outputs) and DVDs, it is the DA converters in the AV receiver that count and not the one in the player.

This Cirrus chip has a balanced output, and this is found only on the better converters and requires more analog electronics. The digital filter preceding the DAC is a complex design providing a digital frequency response of +/-0.01 dB and a 90-dB stop band for digital signals at a 44.1kHz sample rate. Although the +/-0.01 dB spec noted in the Cirrus info sheet may look silly, it is an important indicator of FIR filter tap length, and it correlates with the very important stop band attenuation. The best chips are +/-0.002 dB with more than 100 dB of stop-band rejection. The CS4382 chip has a true DSD inputs for the SACD disks. However, I cannot tell whether it passes it through to the analog output directly or turns it first into PCM, in which case any advantage of SACD signals having no digital signal processing is rendered moot. The data sheet is unclear—in one section on the frequency response of the chip I find the heading "Combined digital and on chip analog filter response -DSD mode."

The chip also has a slow-roll-off mode that trades stop-band attenuation for improved group delay flatness in the passband (improved by a factor of 3). In the 44-kHz mode the slow rolloff starts slightly in-band, at 18.3 kHz instead of 20 kHz. Both fast and slow rolloff modes bring the signal level down 3 dB at 21.9 kHz. More significant is the change in the rejection of the 20 kHz first-folding tone that results in the reconstruction process of the sampled signal (24 kHz for a CD). This is down only 20 dB, which is 10 times the value of the 40 dB in the fast mode. (Remember, decibels are in logs; hence a doubling in decibels is a 10x increase.) Maybe a teenage kid can hear it. The slow mode will make the ringing of the filter to an impulse look better in the time domain but at the cost of a potential audible effect, at least for teens. In addition I note that no scientific study has shown the ear is not sensitive to group-delay flatness.

In the 96-kHz mode, the rolloff moves from 42 kHz to 28 kHz. The -3db point is constant at 48.9 kHz. In the 96-kHz mode the first fold tone for 20 kHz is 78 kHz and this is well rejected with both fast and slow filters (and your ears). In the 96-kHz mode, group delay flatness is 14 times better than CD with the filter in the same mode. As was the case for CD, the slow mode makes things three times better. However, now we are starting with very small variations in group-delay flatness as a result of the reduced requirements on the filter to have an extremely steep transition band when the signal is sampled at 96 kHz.

The effect of the shape of the impulse response of the filter in the time domain that results from moving from 44 kHz to 96 kHz sampling is easier is to appreciate directly in comparison to looking at group delay curves vs. frequency. Some people I know who are experts in sampled data systems (but not in audio) say the reduction in the ringing before and after the impulse might have some effect on the reconstruction of signal in the time domain. Audio Engineering Society conference papers have been presented giving more details on this, but I have not seen them make it to the society's *Journal,* which critiques the materials in greater detail prior to acceptance.

The fast and slow filter responses will change the shape of the impulse. I have no recommendations on which shape would be preferable. 96-kHz sampling is such overkill that small details of filter rolloff become inconsequential. From an engineering point of view, that is a good thing. The difference between reconstructing a signal at 96 kHz and reconstructing it at 44 kHz is clear for all to see on a scope. Whether the difference can be heard is still an open question, but given a choice I will go for the thing that measures well, if it does not cost me any more. In the case of the hardware, the cost impact is small. Software costs are still a major issue.

The bottom line with this DAC is that you lose only a bit in comparison to the best. However, you have at least an extra bit over the lower-cost universal DVD players as well as the AV receivers I am currently testing. The players I am currently testing are, under worst case specs, just 16-bit engines. However, that is all you need for CD playback, and that type of performance in mid-line products would have been impossible a few years ago.

Conclusion: at $450 this unit is a good deal without question, assuming the analog stage is not messed up.

-DAR

T$S

Excerpted with permission from The Sensible Sound, Issue 104, July/Aug. '05. Subscriptions to TSS can be purchased by calling 1-800-695-8439 or writing to: 403 Darwin Drive, Snyder NY 14226.

The $ensible Sound