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Do You Know What Your A/V Controller Is Doing?
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by A. Grimani

Ever wondered what actually goes on inside the guts of your A/V controller, the main processing unit in your home theater? Have you chosen the one that provides the best value for your hard-earned greenbacks?

Let's look at the paths audio and video signals take through the large metal box at the heart of a home theater system. Also, let's talk about what influences audio and video performance along the way.

Digital Audio

The digital audio signals from DVD players, satellite receivers, CD players, and music servers go into either coaxial or optical digital ports and their associated receiving circuitry. As I see it, both ports offer essentially identical performance for Dolby Digital or DTS signals. The optical port does have the advantage of avoiding hum loops, though. The digital signals then run through a selector circuit, which should have no ill effects on performance if it passes signal correctly without degradation of the data bit. The digital stream is then converted from Sony-Philips Digital Interface (S/PDIF) into a format compatible with the Digital Signal Processing (DSP) chip used by your controller.

Analog Audio

The analog signals from VHS players, cassette decks, radio tuners, and digital audio devices without digital outputs (like old CD or Laserdisc players), go through input buffers and selector circuits (usually solid-state switchers). Analog input buffering and selector circuits can affect sound quality by introducing noise, distortion, or other interferences, so here's a place to look for good engineering. The signal then passes through Analog-to-Digital (AD) converters, because in most cases the signal ends up having to be converted into the digital domain for surround decoding and Bass Management. The quality and technology of the AD converters have a huge effect on sound quality. These little circuits can be challenging to get right, linear, and noise-free. There are many test discs that point out defects in converters, so go get some of these, and listen to and measure the effects. The outputs of the converters are then presented to a switching network that selects between direct digital sources and those converted from analog sources. Our digital audio stream is then fed to the DSP chip.

Digital Signal Processing

Once inside the DSP chip, the audio signal data goes through tons of computations. The processing includes, of course, decoding of Dolby Digital or DTS bitstreams, algorithms for acoustic soundfield simulations, and the filtering for subwoofer crossovers and tone controls. The mathematical algorithms for all that processing come in several flavors. Some are good and classy, but some are clunky and can affect sonic performance in irrevocable ways. How do you know which products use the right processes? The price of the product, in fact, won't tell you a thing! It doesn't necessarily cost a whole lot more to run the digital computations right. You've got to listen with an open mind, read responsible reviews, and maybe even do some measuring of your own. Listen to test signals that excel at revealing trouble. Some you may want to try are found on the TMH *Hollywood Series* CDs, the PMI/Gold Line *5.1 Audio Toolkit* DVD, the *Avia* DVD, and the very useful EBU *SQAM* CD (available through DB Systems).

The Dolby Digital and DTS algorithms and processing math are pretty standardized and regulated by both agencies, so don't expect much difference between products here – and don't pay attention to those who say theirs is better; look elsewhere. What actually goes on inside the Dolby Digital or DTS decoding stages? Let me go through a quick overview without getting into heavy-duty theory and math on audio compression and decompression techniques. During encoding in the DVD authoring phase, linear digital signals (typically 6 to 8 channels at 20 bit, 48 kHz resolution) are run through an encoder that analyses signal redundancies, spectrum masking, and a number of other psychoacoustic models attempting to record only what is vital to the human ear to hear quality sound signals. The useless or redundant signal baggage is thrown out, and the remaining bits of data are packed in the most efficient way possible onto the stream of multi-channel audio information. The decoder has to undo all this and turn it into linear audio to be decoded by traditional high-performance Digital-to-Analog (DA) converters. Relatively standardized sets of mathematical DSP algorithms are programmed in the core of the DSP engine. Blocks of data flow into the chip to be unpacked, turned from the frequency domain back into the time domain, and processed to follow the metadata instructions that come along with the audio bits. Ultimately, the process, if done well, can restitute the original channels of audio to a degree of transparency that surprises even the most astute sound mixer. A blind A/B comparison of the source program against the encode/decode version usually reveals absolutely no audible difference. Pretty good I say!

Post-Processing

After the signals are recovered, they are ready to go through one of the various post-processing algorithms. The most notable is Home THX processing. Home THX features such as Re-Equalization, Dynamic Decorrelation, and Timbre Matching correct the inherent electro-acoustic differences between the original cinemas for which film mixes are crafted and home theaters. It is crucial to compensate for these vast differences in acoustic space, speaker type, and calibration standards that separate cinemas from home theaters, or else you'll end up with only a rough representation of the filmmaker's intent. If you apply the compensation correctly, you can actually turn your home theater into a pretty neat clone of a studio dubbing stage!

Other DSP post-processing choices also show up in most controllers. They usually aim to create multi-channel effects from 2-channel material (Lexicon's Logic 7, Meridian's Trifield, etc.), or to recreate larger acoustic spaces from multi-channel tracks. Some of the processes are good and bulletproof, while others royally mess up the original intent. Again, listen to test signals and repeatable musical passages such as those available on the *SQAM* CD or the *Avia* DVD.

Bass Management

Once the multi-channel signals exist in linear digital form, and before we present them to the loudspeaker system in the room, it is time to do some Bass Management. Why manage the bass? Because bass is the hardest region of the frequency spectrum to reproduce accurately in a small room. Bass from some speakers might overwhelm the room, while bass from other speakers may be lacking altogether. Also, some speakers just can't handle deep bass at high volumes. What you ultimately want is strong, tight, consistent bass throughout the room, and one of the best ways to get that is to feed all the low frequency signals in common to well-situated subwoofers. There are also other permutations for handling the low frequency spectrum to best optimize the sound system's power handling and bass smoothness. The more sophisticated controllers give you lots of choices, and it can take an advanced degree in electronics and acoustics to wade through them to the best combination! In fact, Bass Management comprises multiple functions, including high-pass filters (that cut off low frequencies), low-pass filters (that cut off high frequencies), dynamic signal overload limiters, and summing networks. There are serious differences between algorithms, and some filter structures can cause noise and distortion, or produce insufficient rejection. Which ones are good? Listen to test signals and music. Measure the filters and their distortion. Note that low frequency distortion not only shows up as audible "dirt" in the bottom end, but it also creates audible localization cues because the subwoofers are now reproducing more than just the spectrum below 80 Hz.

Time Delay

The next phase of signal processing is usually called Time Synchronization (which is actually a trademark of THX). In an idealized speaker layout, all speakers would be at identical distances from the listening area, forming a circle. In fact, that's not practical in the majority of installations. Instead, the surrounds are usually closer to the listening area and the Center might be further away. The subwoofer could really be anywhere that aesthetics – hopefully coupled with some room acoustics considerations – dictate. Also, consider that placing the speakers in a circle around the listening area might not actually lead to the best sound from each speaker individually due to speaker-room boundary interface issues. All in all, the utopian goal of "speakers in a circle" is virtually impossible to achieve. The solution is to delay all the early-arriving signals from the nearer speakers. The process is simple. Store the digital signals of the early birds in memory long enough for the latecomers to catch up. How long is that? The good controllers make the calculation for you by asking you for the distance to each speaker, then handicapping each foot of earliness with 1.13 ms of delay time. (Sound travels 1.13 ft. per ms.) In the end you can make a system with completely asymmetrical speaker locations sound almost right – amazing!

Back to Analog

At this point, the signals are ready to go back to the world of analog waveforms by passing through DA converters. These little buggers have come a really long way in the last ten years. They usually apply sophisticated mathematics to the digital signal. The noise floor is pushed out beyond the audible range by noise shaping. Signals are over-sampled so that re-entry to the analog world is smoother, and in most cases the digital word length is kept to 24 bits for lowest noise and maximum headroom. After the actual converters comes the reconstruction filtering, which was the source of much debate and study in the early days of digital audio, but is now done with advanced algorithms and gentle analog filters to avoid perceived problems with high frequency roughness. (There are dozens of research papers on this topic alone in scientific journals!)

Level Adjustment

Our 6 to 8 analog signals are now waiting to run through the level control mechanisms. There are many approaches here, and this stage alone probably contributes the most to sonic signature and quality. The typical approach for higher-end controllers is a set of precision, digitally-controlled attenuator ladder networks. These offer 1dB attenuation steps over at least 80 steps. The master volume control talks to the microprocessor, which talks to the level attenuator and sets which position the resistor network taps into. The setting of individual channel gain (a.k.a. level calibration) also talks to the ladder network by putting an offset into the actual master volume control. As you can see, the microprocessor has lots of work to do just remembering and calculating the proper attenuation setting for volume and level calibration! The performance differences between circuit topologies used are vast. Some circuits are noisy, some introduce distortion, and some are hard to control. Less elegant controllers have dual stages of volume control: one stage for the calibration level setting and one for the master volume. This results in excessive noise and sonic degradations. Listen carefully using test discs, and try running some measurements if you have test equipment. Even try asking the product's design engineer about it; he or she might have some useful insights on the particular topology.

Ready for Output

After the volume control stage, the signals are run to output amplifiers, circuits that have enough poop to push the audio signals out of the box and deal with the unpredictable loads that the interconnect wiring and the device down the line often present.

Some controllers have balanced outputs, which is really an additional stage of circuitry. Two identical but opposite signals are output at the connector (typically an XLR where Pin 2 is "hot," meaning it has the same polarity as the original unbalanced signal, and Pin 3 is "cold," meaning that it has inverse polarity to the original signal). Balanced signals have huge advantages when used in complex A/V systems because they inherently reject interference noise and hum. But beware! There are good circuit topologies and not-so-good ones. The latter have different output impedances on the two pins. This leads to poor rejection characteristics. Look at the specs, and ask some questions. Try talking to the design engineer and asking how they ensured equal impedance on the plus and minus pins. If they go blank, or say something that sounds like a bunch of male bovine refuse, you've got a hint… Again, there is no significant difference in cost for doing it right vs. wrong, but you'll be the one left tracking down hum interference if Mr. Propeller-head chose the wrong circuit topology.

Caveat Emptor

One last thing to mention before we close the chapter on audio signals: Along the sinuous path through a controller, signals can pick up induced hum from bad transformers; noise from microprocessors, fluorescent displays, and video circuitry; and be polluted by inadequate circuit design practices. Your only protection from products that suffer from these problems is to listen to and measure the audio signals fed out of the box. So, let me ask this question. How many of you actually run a battery of tests on products before you buy them? Pat yourself on the back if you said, "I do." The rest of you might want to add this practice to your process!

On to Video

Now that we've followed all the steps in the long voyage of audio signals, let's look at the video signals. Most A/V controllers do nothing more to the video than select it from the array of choices at the inputs and send it to the monitor output. The selector circuitry can be good or bad. The most common error is to cut off signal bandwidth at both the high frequencies and low frequencies. One sure way to check for this is to run test patterns from a DVD (like *Digital Video Essentials* or *AVIA*) through the unit. Compare the patterns at the controller's monitor output to the direct output from the DVD player. Ideally you will do this with a waveform monitor (available cheap on eBay), but you can also look at the video display if you have a trained eye. You will need to sit through the tutorials on both test discs to learn what to do.

Another degradation I have seen at times is clipping of the synchronization or the peak white levels. Both of these are problems that a waveform monitor will display right away. There are all kinds of other ills possible like ghosting, ringing, color phase shifting, or saturation changes – you name it. Look at test patterns, learn from the tutorials, and reject the use of any video switching paths that in any way degrade the video signal.

Most products have on-screen menus to make configuration and calibration steps easier, but those video summing networks can also pollute the signal. Test these, too. You can try comparing an output that contains OSD with one that doesn't. If there are marked differences, you have some clues. Since most products today offer switching for composite, S-video (Y/C), and component video signals, you had better run tests on all those paths.

More Coding

Then there are some products that transcode between all the video signal choices. I love the concept of transcoding because you can set up only one video path from the A/V controller to the display. You no longer have to switch inputs on the display every time you change a source. Beware, though, because it just ain't that easy to turn composite signals into decent component signals! Some of the more expensive controllers I have seen out there do a terrible job of the conversion. Some basic display devices actually do a better conversion, and that's just too bad! Measure the conversion, look at it, and if it doesn't work right send a complaint to the manufacturer's engineering department. They usually listen to this form of criticism, especially if it is backed up with measurement data or visual observations made using specific test patterns from known test DVDs. You actually have the power to make a difference!

Out at Last

After source selection, OSD summing, and possibly transcoding, the video signals are ready to leave the box. A set of video driver circuits pushes the signals out of the ports on the back of the controller. These driver circuits should also be well designed in order to preserve video quality. Your observations on test pattern will tell you if they affect the pristine quality of your DVD player's output. Remember that, in the world of video, signals are to run on transmission lines with predefined output and input impedance values (75 Ohms). This means the outputs of the controller have 75 Ohm series resistors tied to them, and the inputs of the display device have 75 Ohm termination resistors tied across them as well. This also means that you can't double up on display devices using a Y-connector. You need to have one device per port, so if you must feed several displays you will need a distribution amplifier. (Note that some professional monitors have switchable termination resistance, but it's not a common feature on consumer gear.)

Now It's Your Turn

So what's the point of all this? Both audio and video signals have lots of room for degradation along the maze that is a home theater controller. You are the one to verify that the products you buy perform to your expectations. I wanted you to get a sense for the places to look for problems. Did I succeed?

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