Are those Digital Discs All They're Cracked up to Be?

By Len Feldman

As of this writing, it's possible to buy several different brands of CD (compact disc) players. The general press and audio-oriented journals have been, in recent months, extolling the virtues of the laser-read grooveless discs as though they were about to make obsolete everything from Edison's first recorded cylinders to the most recently issued audiophile "digitally mastered" records. (The latter, incidentally, have helped to thoroughly confuse the public who will now have a difficult time differentiating between older "digitally mastered" LPs and the true digital discs becoming available.)

So far, I have had an opportunity to audition some two dozen or more of the first available CD discs on nearly a dozen different players. I have also checked out these disc players, using a test record supplied by Sony Corporation and, more recently, three test records supplied by Philips, the people who invented the basic system that has now become the world standard for digital discs.

Player Differences. Some discerning ears have detected a subtle difference in quality of sound reproduction between various brands of compact disc players due to technical differences in the way these players reproduce sound. The main difference is in the signal-conversion process from digital to analog. The conventional method, used by most manufacturers, involves straight 16-bit digital-to-analog (D/A) converters with steep analog filters. The digital sampling rate established for CD discs is 44.1 kHz, because the industry wanted to be able to record and play back frequencies as high as 20,000 Hz. According to the long-established information theory rules, to be able to do this requires a digital sampling frequency at least twice as high, or 40.0 kHz. So 44.1 kHz was chosen to provide some "clearance" between the highest frequency recorded and the action of the steep filter that must remove all frequencies above the highest audio frequency to "smooth out" the recovered waveform and reproduce it with minimal distortion.

Setting a cut-off frequency for a low-pass filter at, say, 21.0 kHz at the −3-dB point and expecting attenuation at 22 or 23 kHz to be 40 dB or more requires a filter with many "poles" and a very steep slope. Such filters invariably introduce phase distortion and can produce "signal overshoot" (ringing) when the recovered analog waveform consists of steep transients, square waves, or the like.

To produce such "brick wall" filters (as they are sometimes called) requires extreme precision for all filter components. To maintain temperature stability, electronic packaging for the analog filters also becomes extremely bulky.

An alternative approach is one used by Magnavox and several other manufacturers. Three techniques are used to overcome the disadvantages of sharp cutoff filters just above the audio range. The first of these is oversampling, which reduces noise in the audio band by 6 dB. In this technique, the digital signal is sampled at four times the normal rate, which distributes the noise over a four-times broader spectrum than normal.

Digital filtering is used to remove ultrasonic components while maintaining good phase linearity up to 20 kHz. Finally, noise shaping reduces noise in the audio band. Unwanted high frequency noise is then eliminated with simple analog filters that have moderate attenuation slopes.

Published Specs vs. Reality. Reading some of the published specifications for CD players, you might correctly conclude that utopia for audio enthusiasts had truly arrived. "Ruler flat" response from 20 to 20,000 Hz is claimed. The slight attenuation at 18.5 kHz shown in Fig. 1 (−0.8 dB for the left channel and −1.1 dB for the right channel) suggests that the "ruler" is slightly "bent" at one end. However, this response curve is far, far better than anything ever produced from a conventional LP test record tracked by even the finest stylus/cartridge combination.

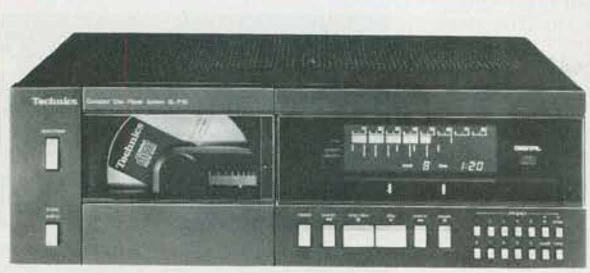
Total harmonic distortion (THD) is usually quoted at about 0.003% to 0.005%. Since most of us have difficulty detecting THD until it reaches at least 0.5% to 1.0%, published THD figures for CD players seem almost irrelevant. In reality, they're not quite that ridiculous, since THD gets *worse* in digitally encoded sound as level *decreases* (the reverse of analog amplifiers, etc.). The quoted 0.005% is for the *loudest* levels of sound the player is capable of handling. Plots of THD versus frequency for that level as well as for lower levels are shown in Fig. 2. As you can see, THD at a −30-dB level is very low but it doesn't have "several zeros after the decimal point" found in the published specs.

IM distortion, which many people consider to be more significant than THD, behaves similarly. On a typical CD player, we measured an IM distortion level of 0.003% at maximum recording level but found that it increased to 0.025% at −20 dB. Since at still lower levels (not provided on even our latest test CD discs, but very prevalent in actual musical recordings) we can expect that THD and IM would rise still further.

Dynamic Forms of Distortion. So far, all we've discussed are static forms of distortion that can be measured with repetitive test signals. With such signals, for all practical purposes, CD players and discs can be said to reproduce waveforms that exhibit negligible differences between themselves and the original sounds picked up by the recording microphones. Unfortunately, music waveforms are much more complicated than test signals.

In an effort to approximate such music signals, the new test records produced by Philips include such music-like signals as square waves, tone bursts, and steep transient pulses. In addition, the test disc includes a pair of tones, one at low frequency recorded on the left channel, the other at a higher frequency on the right channel. The purpose of these tones is to settle the question concerning phase linearity of the various CD players—those that use the sharp cut-off filter only slightly above 20 kHz and those that use the oversampling technique espoused by Philips and Magnavox. We'll discuss this problem more later on. First, let's look at some scope photos of square waves.

A reproduced 100-Hz square wave is shown in Fig. 3A, while in Figs. 3B and 3C, we see square waves that have fundamental frequencies of 1002 Hz and 5512 Hz, respectively. Examining Fig. 4, note that what should have been a single pulse of some short duration, followed by a zero-amplitude baseline for the next 127 time periods (compared with the unit pulse width) turns out to be a pulse followed by a decaying series of "ringing" pulses that obviously weren't in the original test recording. Even the relatively mild 4001-Hz tone



The Technics LS-P10 combines compact-disc techniques with a microprocessor system for programming.

> *"These new players . . . produce such magnificent sound with such overwhelming dynamic range, that it's not worth quibbling about phase errors at 20 kHz."*

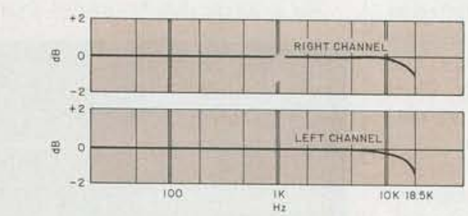


Fig. 1. "Ruler flat" response claimed by some specifications is not always true, as shown here.

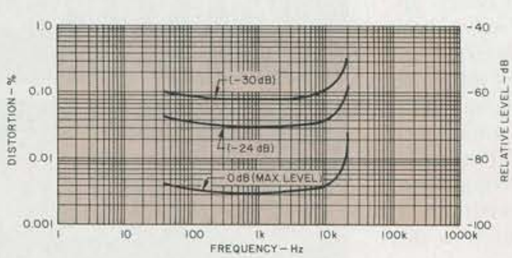


Fig. 2. Harmonic distortion vs. frequency at various recording levels for a typical CD player.

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burst of Fig. 5 (turned on for about 0.026 second) with a repetition rate of 2 Hz, had some trouble settling down to zero after the signal on the disc itself stopped.

As for phase linearity, here's one of the few instances where we could actually see or measure (we didn't say *hear*) a difference between players that use the "brick-wall" filter approach and those that use oversampling. In Fig. 6A, a 2000-Hz left-channel output tone is superimposed upon a 20,000-Hz right-channel tone. The zero-axis positive-going crossing of the lower frequency tone is supposed to occur at exactly the same time as the zero axis positive-going crossing of the high-frequency tone. That's exactly what happens. This perfect phase linearity was observed for a CD player in which oversampling and digital filtering are used.

The same pair of outputs for a player that uses the "brick wall" filter approach, is shown in Fig. 6B. Note that the positive crossings of the zero axis for both tones are no longer at precisely the same spot; some phase displacement has taken place.

Moment of Truth. Having established that there's at least one measurable difference between the two types of players, we set out to conduct a battery of listening tests to find out if we could hear the "subtle" differences referred to earlier. We listened to difficult passages from all two dozen of our CD discs. The recordings that we especially liked sounded *great* on both types of machines! The recordings that we didn't care for (either because they weren't really made from digital master tapes or because we didn't like some of the tinkering that the recording engineers had done) sounded bad on both types of machines!

Taking everything into consideration, the various artifacts introduced at the recording studio are far more significant and tend to mask any subtle differences arising from the two design approaches to CD players we've been talking about. These new players, when playing good CD recordings (of which there are an ever-increasing number), produce such magnificant sound with such overwhelming dynamic range, that it's really not worth quibbling about phase errors at 20 kHz and transient overshoot at frequencies even a dog would have trouble hearing.

To answer the question posed at the beginning, "Are those digital discs all they're cracked up to be?" You *bet* they are, and then some! ◇

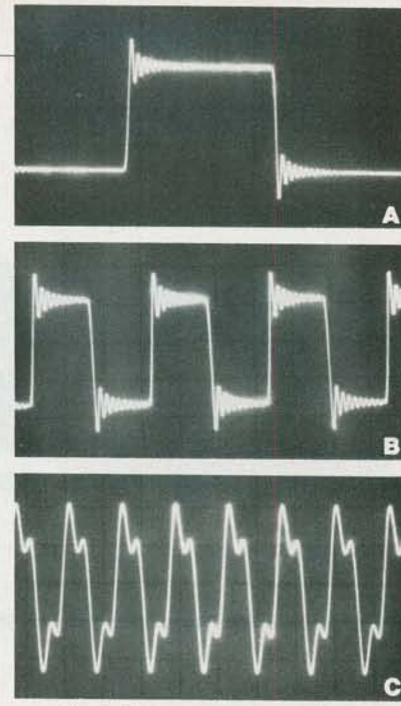

Fig. 3. Square waves reproduced by CD player at: (A) 100 Hz, (B) 1002 Hz, and (C) 5512 Hz.

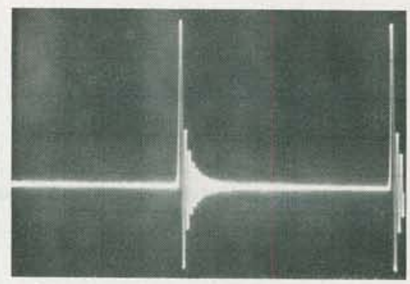

Fig. 4. Single pulse produces ringing.

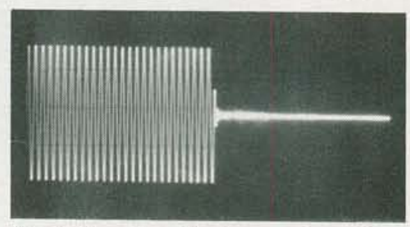

Fig. 5. Tone burst had some after-effect.

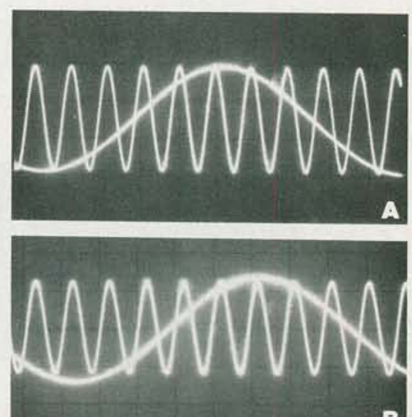

Fig. 6. Perfect phase linearity (A) and minor phase error with "brick wall" filter (B).