My experience suggests otherwise. When I worked in a CD mastering lab, we converted Philips CD players into CD error analyzers for the factory's QC department. We tapped into the flags in the error correction chips for presentation to a PC running custom software to plot the errors. This allowed us to see exactly the frequency and severity of data errors on CDs. CD data errors are categorized by a letter and two numbers that tell you the error's severity with great precision. I'll spare you the details here, but suffice to say that we could determine with tremendous accuracy exactly what was going on in the error correction circuits, and if any uncorrected errors occurred.

Uncorrected errors (interpolation) not only never occured, but *never even came close* to occurring unless the disc was damaged. In fact, the CD's error correction system is far more robust than it needs to be.

Moreover, I sometimes performed a bit-for-bit comparison between a CD master tape and the CD replicated from that tape. I did this on about ten different CD/master tape pairs, representing perhaps 56 billion bits of data. Not once did the system detect even a single bit error.

Proponents of the "bit error" theory of sonic differences between CD transports suggest that sonic qualities such as soundstage size, timbral liquidity, and dynamics are affected by random bit errors as recovered by the transport. But would random amplitude errors, even if they existed, change the soundstage size?

There's no question that a systemic change in the datastream representing music can affect the qualities I've mentioned—soundstaging, timbral liquidity, and dynamics. Those sonic qualities are not affected by random bit errors, but rather by a wholesale re-arrangement of the data, as can occur in PC-based music servers. The Windows operating system (before Windows 7) will completely resample audio data unless you manually turn off this processing. Many audiophiles have built PC-based music servers only to be disappointed by the sound, and such processing is the reason why. The server must be "bit transparent," meaning that the data on the source are the same data that are output to a DAC. I know that my own server is bit transparent because I have it connected to a Berkeley Audio Alpha DAC which features a front-panel HDCD light that illuminates when playing an HDCD-encoded source. Any data corruption will destroy the HDCD code hidden in the least significant bit and prevent the HDCD light from illuminating.

Another source of a wholesale change in the data that introduces an analog-like variability into digital audio is an asynchronous sample-rate converter. A sample-rate converter, even if operating at 44.1kHz input and 44.1kHz output, creates entirely new output samples that are related to the incoming data, but the output samples are *not identical* to the input samples. Although such a device can remove timing errors (jitter) present in the input signal, it does so at the expense of slightly changing the amplitude of every single sample. In effect, it converts a timing error at the input to an amplitude error at the output. At the last CES I heard an interesting demonstration in the PS Audio room of the audible effect of asynchronous sample-rate conversion. The PS Audio Perfect Wave includes an asynchronous sample-rate converter for those times when it might be needed, but it can be switched out of the circuit. This feature allows one to listen with and without the sample-rate converter in the signal path. Although I was leery of sample rate conversion on purely theoretical grounds, this was the first time I was able to listen to and isolate its sonic effects. Switching in the sample-rate converter caused the presentation to sound thicker and less resolved, smeared the imaging, and made the entire presentation sound synthetic.

Neither of these systemic changes to the audio samples representing the music is the result of the CD transport not recovering all the data from a CD. Virtually all CD transports recover 100% of the data from a disc with zero uncorrected errors, and that the sonic differences we hear are solely the result of jitter.