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Music Playback From A Computer – The Audiophile's Concerns

Introduction and main requirements

With the rising popularity of computers as a music source, more and more audiophiles are asking themselves whether a computer can be a decent playback system, similar to e.g. a CD Transport. To answer this question we have to identify the potential sources of disturbances. In digital audio there are two main factors responsible for the quality of a playback system:

1.) The proper transmission of the bits, i.e. a 0 must stay a 0 and a 1 must stay a 1.

2.) The proper sampling clock timing during sampling of the analog signal (A/D Conversion) and during D/A conversion.

Item 1.) can be achieved relatively simple. The CD uses a powerful error correction mechanism which can correct, i.e. do a perfect bit reconstruction, of error bursts of about 4000 bits. 4000 bits corresponds to a scratch on the CD of about one tenth of an inch (2.5mm) length. In addition, using the redundancy in music signals, it can conceal error bursts of up to 12300 bits or a scratch length of three tenths of an inch (7.5mm). "Conceal" is the term meaning interpolating the music signal out of error free data. Of course this is not bit perfect anymore, it is an estimation.

Hard disks used in computers are even better in terms of data recovery, as a single bit which is wrong can render a file on a hard disk useless. So one can safely assume that it is easily possible to get the bits unchanged from the data carrier to the final destination, the D/A Converter.

Item 2.) is more difficult. The "proper sampling clock timing" means that the samples which are taken of the analog signal during the A/D conversion process, ideally are taken at absolute equidistant times, i.e. in the case of the CD at exact multiples of 1/44100th of a second. Deviations from those ideal sampling points are called jitter. In real world systems this jitter can not be zero, as it is an analog process which generates the sampling clock and thus is subject to noise, which results in sample timing jitter. As music consumers we trust that during the A/D Conversion care has been taken to minimize the sampling jitter, as we can't (easily) do anything about it once the signal has been digitized. But during playback, we can take care that the D/A Converter we are using has low intrinsic jitter and in addition is not sensitive to jitter of the sampling clock fed from the source, e.g. the CD Transport or Computer.

Consequently the essence is that the D/A Converter used should be (virtually) immune to jitter. If this is the case, then the computer can be a playback system as decent as a CD Transport. Potentially it is even better than a CD Transport, as it is capable to play back high resolution files with e.g. 24 bit word length and a 192 kHz sampling rate.

Jitter

Jitter usually is suppressed with a so called PLL (Phase Locked Loop) circuit which can be viewed as an electrical equivalent to a flywheel. The PLL is fed the sampling clock (e.g. 44.1 kHz) and generates the same clock "phase and frequency locked" to the incoming clock. I.e. it follows the incoming clock on a long term basis, but short term fluctuations (jitter) are suppressed. This is similar to a flywheel which does not react much to changes in the driving force. Our D/A converter units use a scheme with two cascaded PLLs for even larger jitter reduction.

Alternatively the D/A converter can be made the master clock for the whole system. That means that the audio source is slaved to the D/A converter. This is not easily achieved with a CD transport as it has to have an appropriate clock input. With computer playback via FireWire or USB on the other hand, such a scheme can be easily implemented. Our FireWire or USB based D/A converters for instance can be the clock master and thus are controlling the computer.

Detrimental data modifications

One thing which has to be kept in mind though is the fact that the player program on the computer can easily change the audio data during playback. With today's player programs (e.g. iTunes, Windows Media Player, Foobar2000, MediaMonkey, Jriver, Amarra etc.) one has to be careful that no unwanted signal processing is applied to the signal. Potential harm can come from:

- A badly implemented level control in the digital domain, i.e. one without dithering applied.
- A sampling rate conversion going on in the background because the sampling rate the D/A Converter is running at does not correspond to the sampling rate of the file played. This rate conversion can go "unnoticed" as the operating system engages the rate conversion as required.
- Any other processes like equalization or other sound effects.

Bit Transparent

If the player is capable of feeding the audio files unaltered to the D/A Converter it is said to be "bit transparent", a term used in pro audio circles.

ASIO drivers make it simpler to have bit transparent transfers, i.e. one therefore would prefer D/A converters which can work with ASIO drivers (applicable to Windows based systems).

A nice thing would be to have a means of determining whether the player software is bit transparent. One way to check for this is with a D/A converter which supports HDCD decoding. If the HDCD LED comes on upon playing a HDCD encoded file, the player is bit transparent for sure. Some of our interfacing and D/A Converter units allow the user to test whether the player is doing bit transparent playback.

Digital Level Control

Another question is whether the digital level control in those player programs can be used for a "high-end" level control and thus a preamplifier down the chain could be omitted. The answer is "it depends". With today's 24 bit D/A converters there is a huge dynamic range available which gives room for a digital level control. Usually level controls are not required for huge level changes, but mostly to adjust the volume between different tracks. I.e. a range of e.g. 0 to 24 dB of attenuation is sufficient in most cases. A 24 dB attenuation means that the bits are shifted four places to the right (6dB per bit). E.g. a 16 bit source which occupies bits 1 through 16 in a 24 bit word, will occupy bits 5 through 20 after the attenuation. So still plenty of room in a 24 bit word.

Another example: A 24 bit source which occupies bits 1 through 24 in a 24 bit word will occupy bits 5 through 24 after the attenuation, the 4 least significant bits are shifted out of the 24 bit format. This is where the dithering process comes in. If there wasn't any dither applied, the 4 least significant bits in our example are simply cut off (truncated). This truncation generates a quantization error which exhibits itself in the so called quantization distortion. On the other hand if dithering is applied, the quantization error gets de-correlated from the music signal, i.e. it is wide-band noise. So the music does not get distorted, it rather can be heard completely undistorted down to levels way below the 24 bit limit. There is only noise added – much more pleasant to the ear than distortion.

The advantage of a level control in the digital domain is that a preamplifier can be omitted for a truly minimalist setup. Many D/A converters can easily drive a power amplifier and thus a typical setup would consist of a player, the D/A converter, power amplifier and speakers. It can't get much simpler.

CD Ripping

Another issue is the ripping of CDs to the computer's hard disk.
A very popular program to do this in a decent manner is the http://www.exactaudiocopy.de/ program. This program makes sure that the bits are properly read off the CD.

For OSX systems iTunes can be used to rip CDs, preferably with the error correction mode turned on. Or the free MAX program (http://sbooth.org/Max/). MAX can also convert between lots of audio formats. This can be very useful if a download is in FLAC format but needs to be played back on iTunes which does not support FLAC.

For audiophile quality ripping, never use any of the lossy compression schemes like MP3, WMA, AAC etc. Use either uncompressed data (e.g. WAV or AIFF files) or lossless compressing formats like FLAC, WMA lossless, Apple lossless (ALAC) etc. Losslessly compressing formats have the same audio quality as uncompressed formats.

Wrap-up

Despite all the technical issues mentioned above, a computer based player has some other nice features, like:

- The capability to convey audio data via networks, be it wire based or wireless. I.e. the music collection can be accessed from any room within the house.
- The precious music collection can be backed up on a separate hard-disk and put away.
- The access to the music is so much simplified that one all of a sudden rediscovers those "never heard" tracks.

To summarize, there is no reason why a computer, combined with a proper D/A Converter can't be a decent playback system. There are pitfalls, but those will become less and less with the emergence of computer playback.

Recommended sites

A site dedicated to audiophile computer playback:
http://www.computeraudiophile.com

Directory of music download sites:
http://www.findhdmusic.com
See this page:
http://www.findhdmusic.com/high-definition/directory/