Hi-res fallacy?

Neville Roberts takes a look at recent research that seems to indicate that higher sampling frequency rates are little more than a waste of space

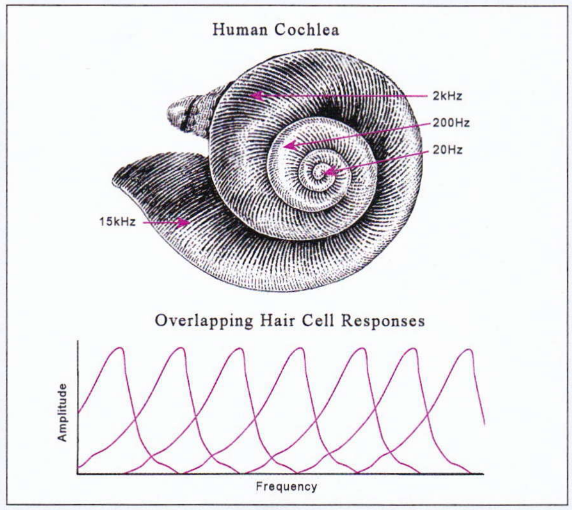
There is a view that the higher the sampling frequency and bit-length, the better the audio quality. Uncompressed 24-bit/192kHz digital files have long been considered the ultimate in sound quality, but some recent research conducted by the Xiph.Org Foundation indicates that 24-bit/88.2kHz or 96kHz may actually be preferable.

The problem with CD is that the format uses just 16-bits, whereas it's widely recognised that a bit-length of at least 20-bits is required to accommodate the full dynamic range of human hearing. As CD's sampling frequency is 44.1kHz it only just manages to include the upper frequencies of human hearing, with no chance of recording desirable instrument ultrasonics that give instruments their characteristic sound.

When a signal is sampled, it is inherently band limited in frequency. In other words, when it is sampled by a finite number of points, it cannot represent an infinite range of frequencies. The sampling rate determines this frequency range because it sets the maximum frequency that can be recorded by a specific sampling rate. This is known as the Nyquist frequency. A conventional audio ADC (analogue-to-digital converter) will only create signals up to the Nyquist frequency and if there are any above that (such as ultrasonic frequencies produced by musical instruments), they are interpreted by the converter and mapped to frequencies lower down. When one frequency is decoded as a different frequency (a process called aliasing), it results in undesirable intermodulation distortion. This process is similar to filming a moving vehicle where the spokes on the wheels spin at a speed close to that of the frame rate. The result is that the wheels appear to slow or even go backwards. To mitigate aliasing in audio, digital convertors have anti-aliasing filters.

An optimum digital sample rate is lower than 192kHz and closer to 88.2kHz

Very high sampling rates may offer a desirable extended frequency response, but can result in a spray of intermodulation distortion across the audio spectrum. Choosing a sampling rate is a balance between not recording any ultrasonics, such as with CD, and recording too many. The problem with digitising at a lower sampling rate means the anti-aliasing filters have to work closer to the audio frequency. However, most modern DACs have oversamplers to allow filtering to take place well away from the audio frequencies, making sampling at very high rates entirely unnecessary.



Each hair cell's frequency band is determined by its position on the membrane

The human ear has hair cells that sit on the resonant basilar membrane in the cochlea. Each cell is effectively tuned to a narrow frequency band determined by its position on the membrane. Sensitivity peaks near the middle of the band and falls off to either side in an asymmetrical shape. Although this sensitivity overlaps with the bands of the neighbouring hair cells, a sound is inaudible if there are no cells tuned to hear it.

Hear hear!

A young person's hearing range is said to typically span from 20Hz to 20kHz. As hearing is a biological sensory system, an individual's hearing is never going to be perfectly flat and will be unique to them. Furthermore, any damage to an individual's hearing will result in a drop off in the frequency response around the hair cells where the damage has occurred. Age-related hearing loss is usually a result of the higher frequency cells losing their sensitivity or their hairs becoming damaged over time. This means the brain attempts to compensate for these non-linearities in sensitivity, but this process can result in a form of analogue aliasing, which can add to the digital aliasing effect I mentioned earlier.

This seems to support a view that an optimum digital sampling rate is actually lower than 192kHz and closer to 88.2kHz. In fact, a friend of mine has conducted ABX tests (double-blind listening tests) using a high-end system to digitise some of his LPs and found that his listening panel preferred the playback of music digitised at 24-bit/96kHz to a 24-bit/192kHz sample. Clearly more research is needed, and in the meantime I'll hold off buying an extra capacity hard disk to accommodate my hi-res music collection ●

NEVILLE ROBERTS
Sounding off