Upsampling, Upconversion and Oversampling:

A Marketing Synonym Game

Introduction

Consumers are attracted to new ideas. It can be trends, places, products or simply features. Usually new features aren't really new, they're nothing more than marketing hype. Every year we are exposed to new models of televisions and new models of cars, and more often than not, there is nothing really new or better inside. Digital audio “Upsampling” and “Upconversion” also falls into this reality. This kind marketing drive has subtly entered the world of digital audio and some companies have managed to increase the sales of their digital audio products by claiming the use of the "new" upsampling technology. The goal of this article is to explain the reality of these technical concepts, leaving all myth aside, and to provide a better understanding of what is really happening with this aspect of digital audio technology.

Basic Digital Audio Theory

Let’s start with the basic Pulse Coded Modulation (PCM) theory. PCM is a method of converting an audio signal from its native analog format into the digital domain where it is comprised of only zeros and ones. This is done on a Compact Disc (CD) for example, and then the original analog signal is recovered by playing the CD in a CD player. PCM works as follows: The analog signal is defined by a value on two axis, amplitude and time. At the sampling frequency, 44.1kHz for CD audio, the amplitude of the signal is encoded into 16-bits words. This means that every 1/44,100th of a second, the closest numerical value to the amplitude of the original analog signal is stored. Since 16-bit words are used, there are 65,536 possible values (2^{16}). When the analog signal is recreated using a digital filter followed by a digital-to-analog converter (DAC), found inside every CD Player, all of these encoded numerical values are restored to their original amplitudes with respect to the time axis, recreating a signal with a staircase shape; each step’s limit corresponds to the finite value in time and amplitude of the encoded digital signal. This signal is then sent to a reconstruction filter that “smoothes” out the staircase shaped signal, creating an output signal without all the jagged steps. This result is an analog signal, which should closely resemble that of the original analog signal.

What are the limitation of this system?

One limitation results from the approximations required to encode the real amplitude of the signal into a discrete value. In reality, the signal always lies between two consecutive levels. This causes noise in the digital domain and the limits dynamic range; with more bits, you have better amplitude precision and an equivalent increase in dynamic range. Another limitation is the sampling frequency. With a higher sampling frequency, you have more samples of the signal in one timeframe. This extends the available bandwidth of the system. Imagine a decrease in the size of the steps as the amplitude resolution (number of bits) increases and/or sampling frequency increases. A finer staircase shape (less jagged appearance) allows the use of a simpler reconstruction filter, which is directly related to sound quality. The less effect the filter needs to have, the less it affects sound quality. In more scientific terms, the sampling frequency dictates the highest possible frequency the system can reproduce. Nyquist has demonstrated the mathematical law which states that a sampling system can reproduce frequencies of up to half the sampling frequency. The system has to filter out all frequencies above half the sampling frequency, which is where unwanted artifacts of the digitized signal (the staircase) reside. The audio CD has a specified bandwidth ceiling of 20kHz, leaving a small gap between that upper frequency and half the sampling rate which is 22.05kHz. The filter must be very steep (high order) to remove information above 22.05kHz but still leaving information under 20kHz. Such a filter was developed at the inception of CD playback and was named the “brickwall filter”. This filter had a terrible impact on sound quality.

Oversampling

The oversampling technique was developed to get away from a “brickwall filter”. A digital system interpolates new points between the different original samples to obtain an artificially higher sampling rate. This allows the use of a less aggressive filter because it doesn’t have to eliminate frequencies as close to the frequencies it must not affect. It first began as four times (4x) oversampling (i.e. 176.4kHz), then later eight times (8x) oversampling (i.e. 352.8kHz). The digital filter must perform many mathematical calculations to determine the value of the point it must add to the original digital signal. Often, this calculated value may fall between two discrete values, so the oversampling system must round off the value to the closest discrete value. To increase the precision of the resulting calculated value, DACs and digital filters with more than 16-bits of resolution were therefore introduced. We have seen 18-bit, 20-bit and 24-bit digital filters and DACs. It is important to note that oversampling creates an artificially higher sampling frequency, which does not extend the real frequency response of the original media or the system, but simply extends the frequencies that need to be filtered out, allowing for a simpler and better sounding analog filter.

Upsampling & Upconversion

One of the latest storage mediums is the popular the Digital Versatile Disc (DVD). When developing this new standard, a higher-than-CD resolution PCM format was adopted with a maximum resolution of 24-bits/96kHz. For the professional market, this new format had to be compatible with the CD's 16-bit/44.1kHz resolution. This would allow the conversion of original recordings to the new standard. So a sample-rate converter chip, which is nothing more than an oversampling digital filter, was created to actually convert any digital signal from one standard format to another format. For example, a 16-bit/32kHz signal can then be converted to 24-bit/96kHz and 24-bit/96kHz can also be converted to 16-bit/48kHz. This gave rise to the marketing hype with the concepts of upsampling and upconversion, which claims could upsample or upconvert your 16-bit/44.1kHz CD to a 24-bit/96kHz resolution digital signal prior to the digital to analog conversion, resulting in DVD-audio like quality from CD. While this statement is a great idea for marketing purposes and is surely impressive to most consumers, it is technically only half true, and is not the best way to improve the audio quality that can be derived from CDs.

Why?

Digital filtering is digital filtering regardless of name assigned to it, and how the interpolation is made still relies solely on the arithmetic calculations implanted in dedicated hardware or software. The main difference is how well the “mechanics” of the mathematics will assist in the signal’s reconstruction. When changing the sampling rate, it is better to maintain an integer multiple of the original signal’s sample rate, so the processing is kept simple. More importantly, the end result is more accurate, thus enabling a higher fidelity of sound reproduction. A two times (2x) oversampling system will double the sampling rate, by adding one easy to find numerical value in between each actual sample. For example, when a 44.1kHz digital signal is processed, a 88.2kHz digital signal is obtained. It is simple, effective and precise because it is a direct multiple of the original digital signal. For an upsampler to make a 96kHz digital signal from a 44.1kHz signal, it will have to perform awkward mathematical operations to obtain a 96kHz signal. (96kHz / 44.1KHz equals 2.1768707…). This results in a less accurate output from the digital filter, with everything else following (i.e. digital-to-analog conversion and analog filtering) also being less accurate. As well, exactly like oversampling, the artificially higher sampling frequency created by an upsampler doesn’t increase the actual frequency response of the system, but simply increases the lower limit of the frequencies that need to be eliminated.

What does all this really mean?

You can hear the differences between the various types of digital filters, regardless of the marketing names used. Over the past two decades, we have witnessed vast improvements in both digital filters and DACs. It is demonstrably true and there does exist real progress. However, the “latest and greatest” upsampling method is not necessarily better than the classic oversampling method. In fact, and most probably, these latest methods actually deteriorate sound quality if the conversion takes the sampling rate to a frequency that is not a direct integer multiple of the original sampling rate, being 44.1 kHz for audio CD.

Simaudio MOON CD Players

Often we’re asked why we don’t include some type of upsampler in our **MOON SuperNova, MOON Eclipse**, **MOON Eclipse LE**, **MOON Nova**, **MOON Nova LE** and **MOON Equinox** CD Players. The answer is simple: All of these CD Players use a digital filter that performs eight times (8x) oversampling with a digital output signal at 24-bits of resolution. In numerical terms this represents 24-bit/352.8kHz digital signal resolution, which is *significantly higher* than that of either a 24-bit/96kHz or 24-bit/192kHz upsampler. What would be the point of using a lower resolution technique? Furthermore, the 352.8kHz digital signal does not suffer from any error and mathematical truncation resulting from the use of a non-integer multiple sample rate conversion frequencies like 96kHz and 192kHz. Using this same reasoning and pushing the envelope even further, the **MOON Andromeda** employs a digital filter that performs sixteen times (16x) oversampling yielding an astounding 24-bit/705.6kHz level of resolution.