It appears, after reading a lot on this subject, there is really no concrete definition for any of these terms! In my research, it appears that Upsampling was probably the most common term used for those early DACs where, in an attempt to sound better, they raised the sample rate of the 16/44.1 digital signal during playback. Later on the term Sample Rate Conversion was applied to be more accurate, with sub-terms, Upsampling & Downsampling, with Resampling covering both, (i.e. SoX Resampler) to make it clear what the end result of the process would be.

My concern with using the term Upsampling to describe what we are doing today is that it came into use at a time when it was done during playback through a DAC or CD Player. We are now upping the sample rate using a PC, with most authorities agreeing that the PC software uses a much more complicated and probably more accurate algorithm to do the sample rate conversion. These are done for the same reason, to make life easier for the reconstruction filter. It is, however, descriptive of what we are doing!

So what do we call today's presentation? Who cares!!! People seem to use them interchangeably. Any of the terms below provided by Wikipedia is potentially correct, so take your choice! BTW, notice that even Wikipedia isn't too sure if the definitions provided below are complete and correct! It appears to me that the word we can use for best understanding is UPSAMPLING! (The winnah!)

John Morrison

Transcoding

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For other uses, see Transcode (disambiguation) and H.264/MPEG-4 AVC products and implementations#Transcoding.

Transcoding is the direct analog-to-analog or digital-to-digital conversion of one encoding to another,[1] such as for movie data files (e.g., PAL, SECAM, NTSC), audio files (e.g., MP3, WAV), or character encoding (e.g., UTF-8, ISO/IEC 8859). This is usually done in cases where a target device (or workflow) does not support the format or has limited storage capacity that mandates a reduced file size,[1] or to convert incompatible or obsolete data to a better-supported or modern format.

In the analog video world, transcoding can be performed just while files are being searched, as well as for presentation. For example, Cineon and DPX files have been widely used as a common format for digital cinema, but the data size of a two-hour movie is about 8 terabytes (TB).[1] That large size can increase the cost and difficulty of handling movie files. However, transcoding into a JPEG2000 lossless format has better compression performance than other lossless coding technologies, and in many cases, JPEG2000 can compress images to half-size.[1]

Transcoding is commonly a lossy process, introducing generation loss; however, transcoding can be lossless if the output is either losslessly compressed or uncompressed.[1] The process of transcoding into a lossy format introduces varying degrees of generation loss, while the transcoding from lossy to lossless or uncompressed is technically a lossless conversion because no information is lost, however the process is irreversible and is more correctly known as *destructive*.

In contrast to a conversion, the prefix "trans" emphasizes a conversion from a source to a different destination.[*citation needed*]

Upsampling

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Upsampling is interpolation, applied in the context of digital signal processing and sample rate conversion. When upsampling is performed on a sequence of samples of a continuous function or signal, it produces an approximation of the sequence that would have been obtained by sampling the signal at a higher rate (or density, as in the case of a photograph). For example, if compact disc audio is upsampled by a factor of 5/4, the resulting sample-rate increases from 44,100 Hz to 55,125 Hz.

Sample rate conversion

From Wikipedia, the free encyclopedia
(Redirected from Resampling (audio))
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Sample-rate conversion is the process of changing the sampling rate of a discrete signal to obtain a new discrete representation of the underlying continuous signal.[1] Application areas include image scaling,[2] and audio/visual systems, where different sampling-rates may be used for engineering, economic, or historical reasons.

For example, Compact Disc Digital Audio and Digital Audio Tape systems use different sampling rates, and American television, European television, and movies all use different frame rates. Sample rate conversion prevents changes in speed and pitch that would otherwise occur when transferring recorded material between such systems.

Within specific domains or for specific conversions, the following alternative terms for sample-rate conversion are also used: **sampling-frequency conversion**, **resampling**, **upsampling**, **downsampling**, **interpolation**, **decimation**, **upscaling**, **downscaling**. The term **multi-rate digital signal processing** is sometimes used to refer to systems that incorporate sample-rate conversion.

Audio[edit]

Audio on Compact Disc has a sampling rate of 44.1 kHz; to transfer it to Digital Audio Tape, which uses 48 kHz, method 1 above can be used with L=160, M=147 (since 48000/44100 = 160/147).[4] For the reverse conversion, the values of L and M are swapped. Per above, in both cases, the low-pass filter should be to 22.05 kHz.

See also[edit]

Sample rate conversion in multiple dimensions:

- Multivariate interpolation

Techniques and processing that may involve sample-rate conversion:

- Oversampling
- Transcoding