Lecture 9: Sample Rate Conversion

Introduction

It is common to see varying sampling rates when dealing with audio signals and performing audio signal processing. Therefore, it is sometimes necessary to perform sample rate conversion. This lecture provided us with information about commonly used sample rates along with the most important methods for performing sample rate conversion. I will give a brief overview of some of the methods and provide an implementation example of sample rate reduction, i.e. downsampling by using lowpass filter implemented in the 2nd diary along with decimation.

Common Sampling Rates

Probably the most well-known sampling rate in audio is 44.1 kHz as it is the standard for consumer audio (e.g. CDs). A higher sampling rate of 48 kHz is the sample rate used in professional audio, i.e. in recording studios, DVD, movies (as it is compatible with common video standards), etc.
Lower sampling rates include 32 kHz, 24 kHz, 22.05 kHz and 11.025 kHz. 32 kHz is known as digital radio sample rate as it is the standard for BBC and EBU, 24 kHz is used for European digital radio and the last two sampling rates of 22.05 kHz and 11.025 kHz can often be encountered in PC sound effects and computer game sounds. Lastly, another common and fairly low sampling rate is 8 kHz also referred to as "toll quality" and it is mainly used in GSM phones.

Sampling Rate Conversion

Sampling rate conversion involves increase or decrease in the sampling rate. When the sampling rate is increased there is no change to signal bandwidth and the signal is simply oversampled. However, when the sample rate is decreased bandwidth is reduced in order to avoid aliasing.
Sampling rate conversion consists of two basic operations know as decimation and interpolation. Decimation is used when the sampling rate is lowered and as a result, we're getting rid of some samples. It is also sometimes referred to as downsampling or subsampling. Interpolation, on the other hand, is used when the sampling rate is increased. Therefore, there's a need for "inventing" new samples between the original ones which is then performed by interpolating between the samples to approximate the new sample values. It is also referred to as upsampling or sampling rate expansion.
One of the techniques to perform downsampling is to pass the original signal through a lowpass filter and retain every n-th sample afterwards. This results in a reduced sampling rate, while the lowpass filter simultaneously prevents aliasing.
Upsampling by interpolation can be performed by inserting $L - 1$ zeros between each sample and perform smoothing by passing the signal through a lowpass filter. Here the lowpass filter suppresses image frequencies. It is also important to note that in this case the signal needs to be amplified by factor L.

Conclusion

All in all sampling rate conversion is an important aspect of audio signal processing as there is very often a need to perform synchronization of various sampling rates, especially when working with music or various different devices. Therefore, it is important to know adequate techniques to perform the conversion efficiently while retaining as much information as possible.