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Digital Audio Processing
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Well this week was a busy week for me, but I managed to continue my forays into the world of digital audio nonetheless. The main focus of my research has been on the topics of oversampling and noise shaping. Although I understand the basic principles of these concepts, I thought that they deserved a closer look and a deeper insight since they are such fundamental operations to digital audio.

One of the main considerations for oversampling is the factor of cost. In any digital-to-analog or analog-to-digital converter, some sort of analog filtering of the signal is necessary. In an ideal converter, this filter has an infinitely sharp cutoff point (the brick wall effect) and no phase distortion or ripple of neighboring frequencies (ripple is a pattern of unwanted changes in amplitude). Unfortunately, such ideal converters are impossible to construct considering the limitations of resistance values and capacitor tolerances. At best, analog filters can only closely approximate the ideal filter design. Even if these analog parts could be constructed to exacting specifications, time itself causes capacitors to drift and resistors to change their values. In either scenario, well designed filters that are less prone to drift are extremely expensive. On the other hand, oversampling techniques, using less steep analog filters, are more cost-effective. It is not that oversampling causes a sound quality that is unapproachable using purely analog filter design, but that it allows a sound quality that is unattainable at the same cost of an analog filter design. Fellow Cornellian M.W. Hauser addresses this concept when he states, "Any electrical performance achievable with oversampling is achievable without it, although not necessarily at the same cost or in the same implementation technology."

While the concept of oversampling in the analog-to-digital phase seems understandable to most people, some technology students still have problems understanding the reasons for oversampling on the digital-to-analog side of things. In fact, just yesterday while I was in my Digitally Controlled Music Systems class, our professor was explaining the workings of oversampling technique and admitted, when asked by a fellow student, to not knowing the reason for oversampling on the D/A side of things. It is truly a simple process. Firstly, one must accept that the D/A converter outputs a "staircase" type wave. The steps in this staircase are basically square waves at the half the sampling frequency. Since square waves contain not only the fundamental frequency but continuous odd partials above that frequency, the resulting analog signal now contains frequency information above the Nyquist frequency. In other words, if the sampling frequency is 44.1 kHz, the highest fundamental on the analog staircase will be 22.05 kHz, but the output analog signal will contain frequencies infinitely higher. Some ignorant students may ask why these higher frequencies are necessary to be deleted from the output analog signal since we cannot hear that high anyway. The simple answer is that at ultra high frequencies, amplifiers and speakers become non-linear. In other words, the added stress of trying to reproduce these ultra high frequencies by amplifiers and speakers causes them, one could say, to reproduce the audible frequencies less well. An analogy could be made to person A saying something at a regular speed while person B is including the same information as person A but also some extra information which causes person B to talk really fast (same information+extra information). Person B (the analog of the amplifier trying to reproduce ultra high frequencies) is thus harder to comprehend because he has a lot of unnecessary words and information which are obscuring the basic information. Filters are thus necessary after D/A conversion to get rid of this extra information. If extra samples are thus added to the signal (through interpolation or whatever method), the sampling frequency (and thus the fundamental frequency of the staircase steps) is therefore raised. A less steep analog filter can then be used to easily filter out the unnecessary high frequency information while preserving the integrity of the information.

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