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Now that I have done a little bit of research into the noise shaping and oversampling properties of converter design, I thought it would be appropriate to investigate the use of filters in digital applications. Through understanding filters, I hope to understand more of the processes by which analog is converted to digital and vice-versa, particularly the problems that can occur in such a scenario. Most of the literature concerning filters is rather mathematically intensive, but I endeavored to get the most out of the articles anyway.

The most important distinction to make when discussing filters is their impulse response. Impulse response is related to how long the filter acts upon the input signal. Traditional analog filters use an infinite impulse response design, meaning that the filter's output continues (much like a decaying echo). This infinite response is achieved by using a method just like that of noise shaping—the recursive network. As the signal is passed through the filter, parts of the output are fed back and mixed with any other incoming input. Infinite impulse filters affect the frequency response of an input signal without regard to phase considerations. Technically speaking, IIR filters are not phase linear. How this non-linearity translates to the real world is by annoying ringing or phase shifts in the original signal. No matter how well engineered an analog filter is, therefore, it will always be of the IIR type and thus always have a hint of phase distortion. Usually this phase distortion can be moved outside of the audible band, which is especially useful in digital audio applications.

The alternative to IIR filtering is a filter design that has a finite response. With FIR filters, the filter itself only acts upon the incoming signal once; there is no recursiveness to the function. Digital filters can be designed as FIR models, but it is very expensive to do so. The advantage of making an FIR filter is that phase shifts are smoothly allocated according to frequency equaling a continuous time delay across the spectrum. In real world terms, this means that frequency responses can be tailored without any artifacts. Obviously, such a system has greater application to audio than the imperfect analog designs.

The methodolgy of designing a digital FIR filter is a more complicated issue than the mere understandings of its workings. To understand FIR filters, one has to grasp the concept of transforms. These transforms are what I am most interested in at the present. Right now, I have just brushed the surface of transform pairs—transforms which have a duality between time and frequency. For example, a DC current (horizontal line in the time domain) corresponds to a signal frequency (0 Hz) in the frequency domain (represented by a vertical line). Conversely, a vertical line in the time domain (one sample of give amplitude) corresponds to a horizontal line in the frequency domain (one given amplitude and an infinite amount of represented frequencies). The interchangeability of the two transform graphs is their duality. The most common transform pair when talking about filters is the (sin x)/x to rectangle mapping. Hopefully, some homemade graphs will help elucidate this duality for me.

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