Digital Impulse Response Filter

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Introduction

A digital filter is defined as a “mathematical algorithm implemented in hardware and /or software that operates on a digital input signal to produce a digital output signal for filtering purposes”(ucsb.edu). In addition, some of the advantages of the digital filters over analog filters are listed as; linear phase response, less environmental effects on filters, and can operate on lower frequencies(ucsb.edu). Furthermore, a digital filter can be categorized into two general classes: Finite Impulse Response (FIR), and Infinite Impulse response(IIR). The main idea in the design of different digital filters is to implement the concept of delays (affect the presence/lack-of distortion in a signal), along with gain of a signal to get the desired effect of a signal in frequency domain. Impulse response is defined as the output of a filter when the input is an impulse of amplitude 1 at time equals 0. Finite Impulse response, as the name implies, has a finite impulse response duration while IIR filters have an infinite impulse response duration(hwe.design). The FIR filters solely rely on the input signals and do not use previous outputs values, as opposed to the IIR filters depending on both the inputs signals and previous output values(feedback structure). As an effect, when the input impulse goes away, the filter output response remains because of the feedback structure that provided the filter with previous outputs.

Methodology

For this project, the IIR filter was chosen over the FIR because although it can be unstable if not designed correctly, the IIR filter boasts some advantages over the FIR filter that is preferred for this project. Some of which includes: - Achieving sharp bandpass characteristics with a lower order filter (2nd, 4th or 6th), compared to that of FIR filter which can be much higher (40 -200). - This leads to less computations and less resources used, which is vital for FPGA or microcontroller implementation. The project specifically aims to design a passband filter with a passband frequency of 300Hz to 3.4kHz frequency, with a sampling rate of 8kHz and a sample duration of 128ms. Designing a digital filter can be done in different ways, as long as the design ensures a good performance, stability and implementation efficiency of the filtering process. This project implemented a Direct Form II structure, along with a second-order Butterworth filter to shape the frequency response of the filter. According to the Stanford University journal, The Direct-II structure is regarded as a two-pole filter followed by a two-zero filter. It is defined to be canonical with respect to delay, meaning that the delay elements associated with the two-poles and two-zeros section are shared. In addition, it is worth mentioning that in a Direct-II structure implementation, zeros and poles can be subjected to round-off errors in the coefficients a1 and b1, especially in higher order filters. The Butterworth coefficients serve as the analog prototype, which the filter is built with. The Butterworth filter is an analogous filter design “that produces the best output response with no ripple in the passband while covering uniform gain inside the passband”. In this design, the ‘transition band’ or roll-off is defined as how quickly the filter cuts off frequencies outside the passband, and this depends on the order of filter implemented. This project implements a 2nd order filter which offers a smooth but relatively long transition band. This wasn’t exactly a major concern for the project as the main goal was to show the filter’s implementation and results. This can be improved in future works.