

A High Performance & Efficiently Designed IIR Filter Using Graphical Virtual Tool (LabVIEW)

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Abstract: Designing of digital filters based on LABVIEW involves the concept of virtual instrumentation. In the recent years LABVIEW finds many applications in different fields for example industrial purpose like level sensor, temperature sensor pressure sensor etc. Virtual instruments are used in LABVIEW. Each VI has three parts a block diagram, a front panel and a connector panel. Connector panel is used to represent the VI in the block diagrams of other. Controls and indicators on the front panel allow an operator to input data into or extract data from a running virtual instrument. The front panel can also serve as a programmatic interface. A digital IIR filter system is developed using LABVIEW software. IIR filters possess certain properties which makes them preferred design choices in many situations over FIR filters. Following are the types of IIR filter Butterworth filters, Chebyshev filters, Inverse chebyshev filters, Elliptic filters. Virtual instrument reads the desired parameters of the filters entered by the user on the front panel and determines its magnitude response and filter coefficients.

Keywords: ADC, DAC, DSP, Design Approach, Virtual instrument, LABVIEW, IIR filter.

I. INTRODUCTION TO BASICS OF THE FILTERS

The function of a filter is to remove unwanted parts of signal or to extract some useful parts of signal, such as the components lying within a certain frequency range.



Fig.1 Block diagram of basic filter

Infinite impulse response is a property of signal processing systems. IIR filters have impulse response function that is non zero over an infinite length of time. Simplest example of analog IIR filter is an RC filter made up of single resistor (R) feeding in to a node shared with single capacitor (C). Impulse response of this filter is exponential characterized by RC time constant. The exponential function is asymptotic to a limit and thus never settles to a fixed value that's why response is considered infinite. Digital filters use digital processor to perform numerical calculations on sampled values of signal. Block diagram of digital filter is given below.

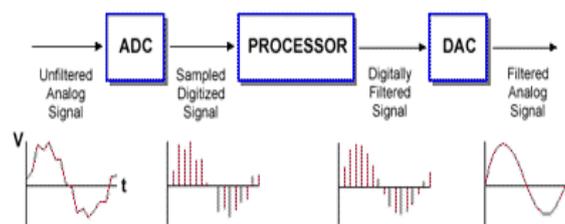


Fig.2 A block diagram of basic digital filter

Firstly the input signal must be sampled and digitized using an analog to digital converter. The result of converter is binary numbers, representing successive sampled values of the input signal, which are transferred to the processor and it performs numerical calculation on them. These calculations involve multiplying the input values by constants and adding the products together.

Following are the advantages of digital filters i.e.

- The principal advantage of digital filters is the flexibility available in their design.
- The ease of data storage is one of the main advantages of digital filters.
- Digital filters are programmable.
- Fast DSP processors can handle complex combinations of filters in parallel or cascade, making the hardware requirements relatively simple and compact.

II. FUNDAMENTALS OF THE OPERATION OF AN IDEAL FILTERS

Ideal filters pass specified frequency range while attenuate specified unwanted frequency range. Filters can be classified according to their frequency range characteristics. Following are the filter classifications based on frequency range a filter passes or attenuates.

- Low pass filters pass low frequencies and block high frequencies.
- High pass filters pass high frequencies and block low frequencies.
- Band pass filters pass a certain band of frequencies.
- Band stop filters attenuate a certain band of frequencies.

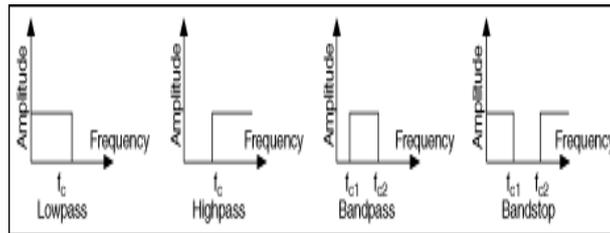


Fig. 3 Ideal frequency characteristics

The frequency points f_c , f_{c1} and f_{c2} specify the cut off frequencies for the different filters. An ideal filter has a gain of one (0 dB) in the passband so the amplitude of the signal neither increases nor decreases. Fig. 4 shows pass band and stop band for each filter type.

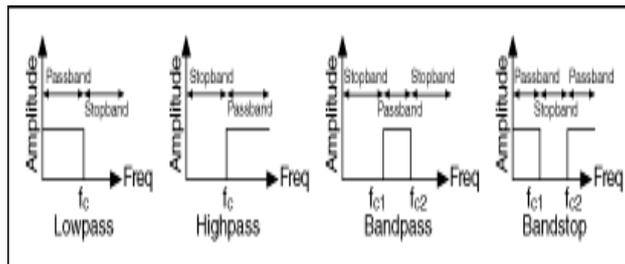


Fig.4 Pass band and stop band

III. PRACTICAL DESIGN OF A FILTER

In practical filters a transition band always exists between pass band and stop band. In this band the gain of filter changes gradually from one in the pass band to zero in the stop band

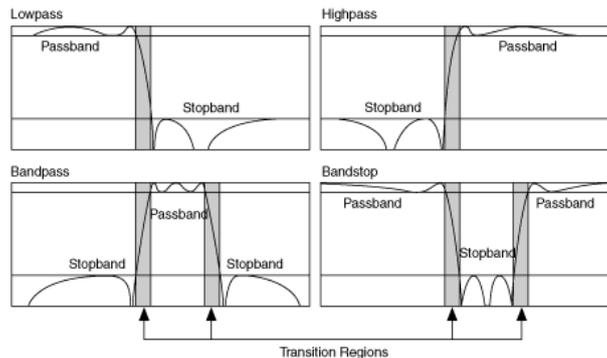


Fig.4 Response of non ideal filters

IV. DIGITAL FILTERS

Filters can be classified according to their impulse responses. There are mainly two types of digital filter i.e. Finite impulse response filters, which are also known as non recursive filters because they don't have feedback. These filters operate on current and past input values and Infinite impulse response filters, which are also known as recursive filters because they have feedback or recursive part of filter.

V. IMPULSE RESPONSE

An impulse is a short duration signal that goes from zero to a maximum value and back to zero again in a short time. The impulse response of a filter is the response of filter to an impulse and depends on the values upon which the filter operates. The Fourier transform of impulse response is frequency response of filter.

VI. BASICS OF A DIGITAL IIR FILTER

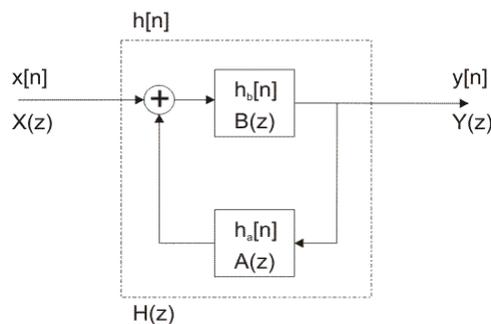


Fig.5 Block diagram of IIR filter

The design of digital filter depends on both values, past outputs and present input. If such a filter is subjected to an impulse then output of this filter need not necessarily become zero. The impulse response of such a filter can be infinite in duration. Such type of filter is called an infinite impulse response filter and indicates that the system is prone to feedback and instability. Following equation defines the direct form transfer function of an IIR filter

$$H(z) = \frac{b_0 + b_1z^{-1} + \dots + b_{N_b-1}z^{-(N_b-1)}}{1 + a_1z^{-1} + \dots + a_{N_a-1}z^{-(N_a-1)}} \quad (1)$$

Where a_n and b_n are reverse and forward coefficients of the IIR filter. It can be written in the form of general difference equation as follows

$$Y_i = \frac{1}{a_0} \left(\sum_{j=0}^{N_b-1} b_j X_{i-j} - \sum_{k=0}^{N_a-1} a_k Y_{i-k} \right) \quad (2)$$

Where

- b_j , set of forward coefficients,
- N_b , number of forward coefficients,
- a_k , set of reverse coefficients,
- N_a , number of reverse coefficients

Above equation describes a filter with an impulse response of theoretically infinite length for nonzero coefficients. In design of IIR filter with LABVIEW coefficient a_0 is 1. The IIR filters can be designed by any one of the following methods.

- Impulse invariance
- Bilinear transformation

1. Impulse invariance

Following are the steps of this method.

- Decide upon the desired frequency response
- Design an appropriate analog filter
- Calculate the impulse response of this analog filter
- Sample the impulse response of this analog filter
- Use the result as the filter coefficients

This method seems simple but it is complicated by all the problems inherent in dealing with sampled data systems. Particularly this method is subject to problems of aliasing and frequency resolution. IIR filters are very sensitive to quantization errors. It is a feature peculiar to digital systems. Its effects are nonlinear and signal dependent. In order to prevent severe distortion due to the band limiting this method is restricted to the design of lowpass and bandpass filters.

2. Bilinear transformation

This method overcomes the effect of aliasing that is caused due to analog frequency response containing components at or beyond the Nyquist frequency. This is also called frequency wrapping because this is a method of compressing the infinite, straight analog frequency axis to a finite one long enough to wrap around the unit circle once only.

VII. TYPES OF IIR FILTER

- Butterworth filters
- Chebyshev filters
- Inverse chebyshev filters
- Elliptic filters

The IIR filter designs differ in the sharpness of the transition between the pass band and stop band, where they exhibit various characteristics.

1. Butterworth filters

The frequency response of this filter has no ripples in the passband and the stopband therefore it is called maximally flat filter. Butterworth filter uses a Taylor series approximation to the ideal at both $\omega=0$ and $\omega=\infty$.

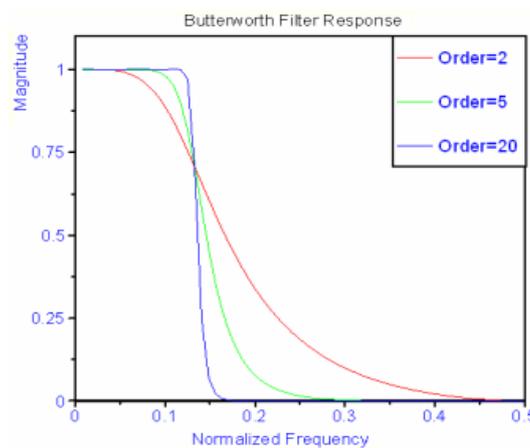


Fig.6 Response of Butterworth filter

We can see from above figure as order of filter increases response of filter closer to the ideal response. Butterworth filters have following characteristics:

- Smooth response at all frequencies.
- Monotonic decrease from the specified cut-off frequencies.
- Maximal flatness, with the ideal response of unity in the pass band and zero in the stop band.
- 3dB down frequency, that corresponds to the specified cut-off frequencies.

The transfer function of Butterworth filter is as follows

$$B(\omega) = \frac{1}{\left[1 + \left(\frac{\omega}{\omega_0}\right)^{2n}\right]^{\frac{1}{2}}} \tag{3}$$

where n is the order of filter.

2. Chebyshev filters

This filter uses a chebyshev approximation across the passband and a Taylor series at $\omega=\infty$. Characteristics of this filter is as follows

- Minimization of peak error in the passband.
- Equiripple magnitude response in the passband.
- Monotonically decreasing magnitude response in the stopband.
- Sharper rolloff than Butterworth filters.

Chebyshev filter can achieve sharper transition between passband and stopband with lower order filter than Butterworth filter. This results in smaller absolute errors and faster execution speeds. Frequency response of this filter is given by

$$|H(\Omega)|^2 = \left(1 + \varepsilon^2 T_N^2\left(\frac{\Omega}{\Omega_p}\right)\right)^{-1} \tag{4}$$

where ε is a parameter of the filter related to ripple present in the passband and $T_N(x)$ is the Nth order Chebyshev polynomial.

$$\begin{aligned} T_N &= \cos(N \cos^{-1}x) & |x| \leq 1 \\ &= \cosh(N \cosh^{-1}x) & |x| \geq 1 \end{aligned} \tag{5}$$

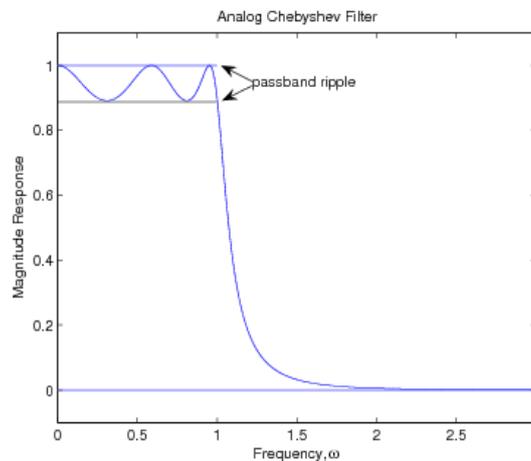


Fig.7 Response of Chebyshev filter

3. Inverse Chebyshev filters

Inverse Chebyshev filter uses a Taylor series approximation at $\omega=0$ and a Chebyshev across the stopband. This filter differ from chebyshev filter in following ways

- These filters minimize peak error in stopband instead of passband.
- These filters have an equiripple magnitude response in the stopband instead of passband.
- These filters have a monotonically decreasing magnitude response in the passband instead of stopband.

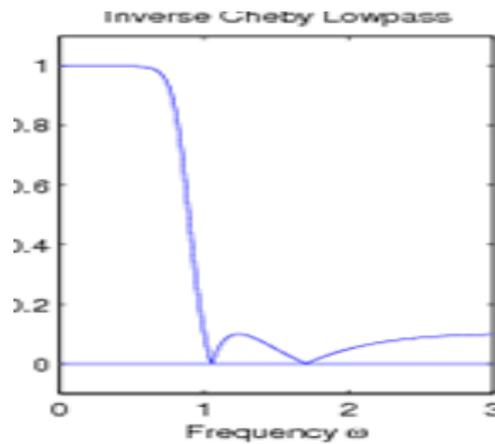


Fig.8 Response of Inverse Chebyshev filter

4. Elliptic filters

The elliptic function filter uses a Chebyshev approximation across both the passband and stopband. Elliptic filters have following characteristics

- Minimization of peak error in the passband and the stopband.
- Equiripples in the passband and stopband

Elliptic filters provide the sharpest transition between the passband and stopband, which accounts for their widespread use. Transfer function of Elliptic filters is given by

$$|H(\Omega)|^2 = (1 + \epsilon^2 U_N(\frac{\Omega}{\Omega_c}))^{-1} \tag{6}$$

where $U_N(x)$ is the Jacobian elliptic function of order N and ϵ is a constant related to passband ripple.

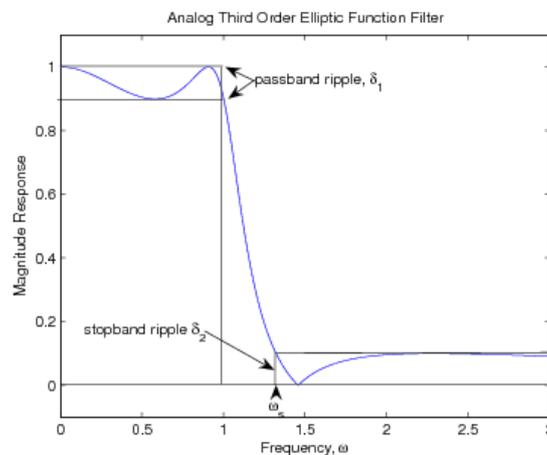


Fig.9 Response of Elliptic filter

VIII. SIMULATION AND DISCUSSION

IIR filters are used for applications where linear characteristics are not of concern. IIR filter is better for lower order tapping. IIR filters must have at least one pole. It is a recursive filter means it has feedback. IIR filters may be unstable depending on the location of poles where as FIR filter is always stable. Pole-Zero plots is an important tool. It can be used to determine stability.

We can distinguish from pole-zero plot whether the filter is low pass, high pass, band pass or band stop. Low pass filters have poles closer to the origin than zeros. They may not have zeros at all. High pass

filters will have the zeros close the origin and will probably have at least one on the origin. Band pass filters will have zeros close to the origin and some poles farther away. The transfer function of 3rd order elliptic low pass filter is given by

$$H(z) = \frac{0.1335z^3 + 0.056z^2 + 0.056z + 0.1335}{z^3 - 1.507z^2 + 1.2646z - 0.3786} \quad (7)$$

If both poles and zeros of a transfer function are all inside or on the unit circle of the z-plane, the filter is called minimum phase. Frequency responses of different IIR filters are shown in following figures under following parameters

- Lower cutoff frequency-2K
- Upper cutoff frequency-3.80K
- Passband ripple-0.03
- Stopband attenuation-90.00

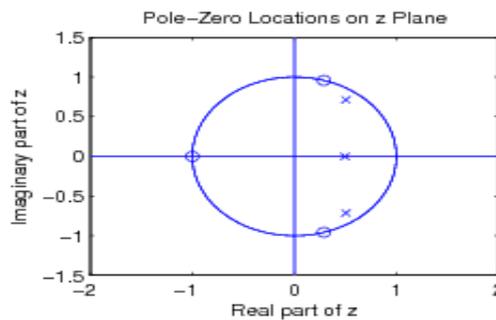


Fig.10 Pole-zero plot of 3rd order Elliptic filter

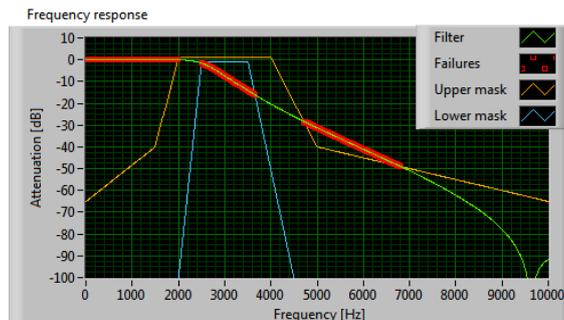


Fig.11 Frequency response of 4th order Butterworth filter

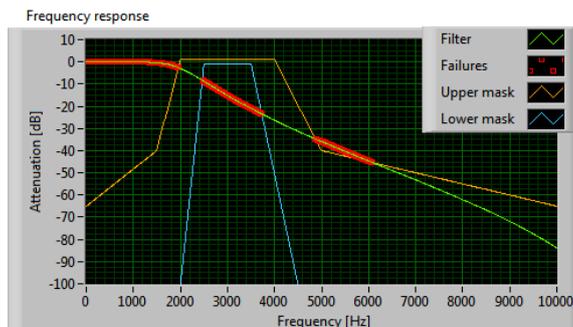


Fig.12 Frequency response of 4th order Chebyshev filter

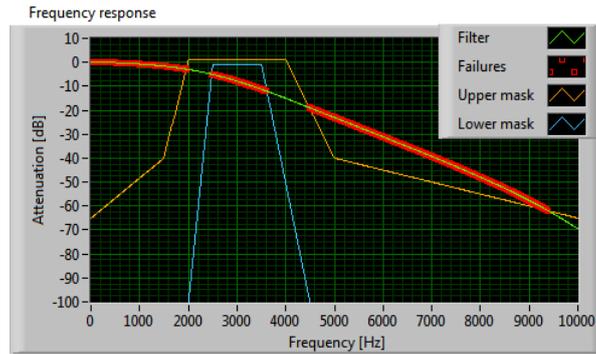


Fig.13 Frequency response of 4th order Inverse Chebyshev filter

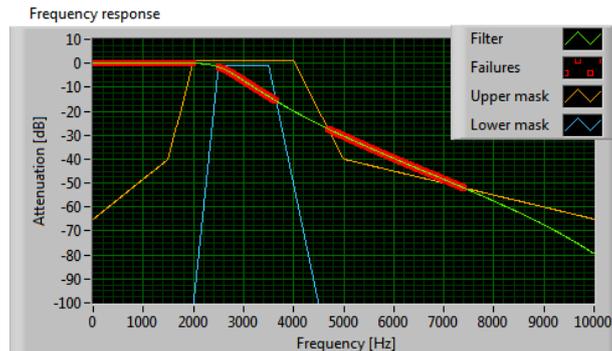


Fig.14 Frequency response of 4th order Elliptic filter

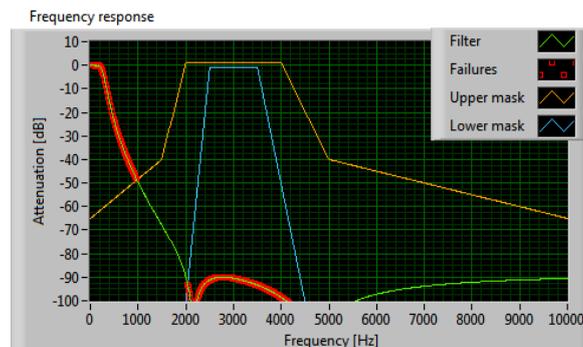


Fig.15 Frequency response of 4th order Bessel filter

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