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Research Paper

Designing of Low Pass IIR Digital Filter Using Butter Worth and Chebyshev Type I & Type II Window Technique.

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Abstract: This paper presents the design of low pass IIR digital filter using Butterworth and Chebyshev type I & chebyshev type II window technique at minimum order. In the design of frequency selective filters, the desired filter characteristics is specified in the frequency domain in terms of desired magnitude and phase response of the filter. In the filter design process, we have determined the coefficients of a causal IIR filter by using MATLAB simulation.

Keywords: Butterworth, Chebyshev, Digital Signal Processing, Filter, IIR, Low pass filter.

1. Introduction

Digital signal processing (DSP) is the process of analyzing and modifying a signal to optimize or improve its efficiency or performance. It involves applying various mathematical and computational algorithms to analog and digital signals to produce a signal that's of higher quality than the original signal. Digital signal processing (DSP) refers to various techniques for improving the accuracy and reliability of digital communications. The theory behind DSP is quite complex. ADSP circuit is able to differentiate between human-made signals, which are orderly, and noise, which is inherently chaotic. The field of signal processing is a very important field of study and one that makes possible various other fields such as communications. Speech recognition systems such as dictation software need to analyze and process signal data to identify individual words in a spoken sentence. DSP applications include audio and speech processing, sonar, radar and other sensor array processing, spectral density estimation, statistical signal processing, digital image processing, signal processing for telecommunications, control systems, biomedical engineering, and seismology, among others.

1.2. Advantages of digital filters

- No component ageing.
- High immune to noise.
- No problems of impedance matching.
- Operated under wide range of frequencies.
- Coefficients of digital filters can be altered to obtain desired characteristics.
- Multiple filtering is possible only in digital filters.

1.3. Disadvantages of digital filter

- Quantization error arises due to finite word length in the representation of signals & parameters

2. IIR filter

Infinite impulse response (IIR) is a property applying to many linear time-invariant systems. Common examples of linear time-invariant systems are most electronic and digital filters. In signal processing, a digital filter is a system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal. IIR filters are digital filters with infinite impulse response. Unlike FIR filters, they have the feedback (a recursive part of a filter) and are known as recursive digital filters therefore. In signal processing, a recursive filter is a type of filter which re-uses one or more of its outputs as an input.

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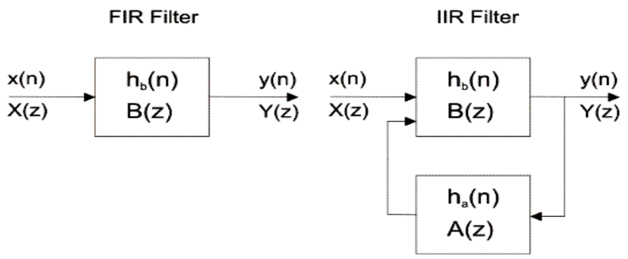

Figure 1: Block diagrams of FIR and IIR filters

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3. Butterworth Filter

The Butterworth filter is a type of signal processing filter designed to have a frequency response as flat as possible in the pass band. It is also referred to as a maximally flat magnitude filter**.** The frequency response of the Butterworth filter is maximally flat (i.e. has no ripples) in the pass band and rolls off towards zero in the stop band. When viewed on a logarithmic Bode plot, the response slopes off linearly towards negative infinity. A first-order filter's response rolls off at −6 dB per octave (−20 dB per decade) (all first-order low pass filters have the same normalized frequency response). A second-order filter decreases at −12 dB per octave, a third-order at −18 dB and so on. Butterworth filters have a monotonically changing magnitude function with ω, unlike other filter types that have non-monotonic ripple in the pass band and/or the stop band. Compared with a Chebyshev Type I/Type II filter or an elliptic filter, the Butterworth filter has a slower roll-off, and thus will require a higher order to implement a particular stop band specification, but Butterworth filters have a more linear phase response in the pass-band than Chebyshev Type I/Type II and elliptic filters can achieve. Digital implementations of Butterworth and other filters are often based on the bilinear transform method or the matched Z-transform method, two different methods to discretize an analog filter design. In the case of all-pole filters such as the Butterworth, the matched Z-transform method is equivalent to the impulse invariance method. For higher orders, digital filters are sensitive to quantization errors, so they are often calculated as cascaded biquad sections, plus one first-order or third-order section for odd orders.A transfer function of a third-order low-pass Butterworth filter design

$$\frac{V_o(s)}{V_i(s)} = \frac{R}{s^3(L_1C_2L_3) + s^2(L_1C_2R) + s(L_1 + L_3) + R}.$$

The magnitude of the frequency response (gain) *G*(ω) is given by

$$G(\omega) = |H(j\omega)| = \frac{1}{\sqrt{1+\omega^6}},$$

Obtained from

$$G^2(\omega) = |H(j\omega)|^2 = H(j\omega)H^*(j\omega) = \frac{1}{1+\omega^6},$$

And the phase is given by

$$\Phi(\omega) = \arg(H(j\omega)).$$

The gain of an *n*-order Butterworth low pass filter is given in terms of the transfer function *H(s)* as

$$G^2(\omega) = |H(j\omega)|^2 = \frac{{G_0}^2}{1+\left(\frac{j\omega}{j\omega_c}\right)^{2n}}$$

Where

- n = order of filter
- ω_c = cut off frequency (approximately the -3dB frequency)

It can be seen that as *n* approaches infinity, the gain becomes a rectangle function and frequencies below ω_c will be passed with gain G_0, while frequencies above ω_c will be suppressed. For smaller values of *n*, the cutoff will be less sharp.

3.1. Properties of the Butterworth filter are

- Monotonic amplitude response in both pass band and stop band
- Quick roll-off around the cut off frequency, which improves with increasing order
- Considerable overshoot and ringing in step response, which worsens with increasing order
- Slightly non-linear phase response
- Group delay largely frequency-dependent

4. Chebishiv Filter

Chebyshev filters are analog or digital filters having a steeper roll-off and more pass band ripple (type I) or stop band ripple (type II) than Butterworth filters. Chebyshev filters have the property that they minimize the error between the idealized and the actual filter characteristic over the range of the filter but with ripples in the pass band. This type of filter is named after Pafnuty Chebyshev because its mathematical characteristics are derived from Chebyshev polynomials. Because of the pass band ripple inherent in Chebyshev filters, the ones that have a smoother response in the pass band but a more irregular response in the stop band are preferred for some applications.

4.1. Type 1 Chebyshev filter

Type I Chebyshev filters are the most common types of Chebyshev filters. The gain (or amplitude) response, $G_n(W)$, as a function of angular frequency W of the *n*th-order low-pass filter is equal to the absolute value of the transfer function $H_n(S)$ evaluated at s=JW:

$$G_n(\omega) = |H_n(j\omega)| = \frac{1}{\sqrt{1+\varepsilon^2 T_n^2\left(\frac{\omega}{\omega_0}\right)}}$$

Where ε is the ripple factor, ω_0 is the cut-off frequency and T_n is a Chebyshev polynomial of the n th order. The pass band exhibits equiripple behaviour, with the ripple determined by the ripple factor. In the pass band, the Chebyshev polynomial alternates between -1 and 1 so the filter gain alternate between maxima at $G = 1$ and minima at

$$G = 1/\sqrt{1+\varepsilon^2}.$$

The ripple factor ε is thus related to the passband ripple δ in decibels by:

$$\varepsilon = \sqrt{10^{0.1\delta} - 1}.$$

At the cutoff frequency ω_0 the gain again has the value $1/\sqrt{1+\varepsilon^2}$

But continues to drop into the stop band as the frequency increases. This behaviour is shown in the diagram on the right. The common practice of defining the cut off frequency at −3 dB is usually not applied to Chebyshev filters; instead the cut off is taken as the point at which the gain falls to the value of the ripple for the final time.

The 3 dB frequency ω_H is related to ω_0 by:

$$\omega_H = \omega_0 \cosh\left(\frac{1}{n}\cosh^{-1}\frac{1}{\varepsilon}\right)$$

The order of a Chebyshev filter is equal to the number of reactive components (for example, inductors) needed to realize the filter using analog electronics. An even steeper roll-off can be obtained if ripple is allowed in the stop band, by allowing zeroes on the $j\omega$ -axis in the complex plane. However, this results in less suppression in the stopband. The result is called an elliptic filter, also known as Cauer filter.

4.2. Type II Chebyshev filter

Also known as inverse Chebyshev filters, the Type II Chebyshev filter type is less common because it does not roll off as fast as Type I, and requires more components. It has no ripple in the pass band, but does have equiripple in the stop band. The gain is:

$$G_n(\omega,\omega_0) = \frac{1}{\sqrt{1+\frac{1}{\varepsilon^2 T_n^2(\omega_0/\omega)}}}.$$

In the stop band, the Chebyshev polynomial oscillates between -1 and 1 so that the gain will oscillate between zero and

$$\frac{1}{\sqrt{1+\frac{1}{\varepsilon^2}}}$$

And the smallest frequency at which this maximum is attained is the cut off frequency . The parameter ε is thus related to the stop band attenuation γ in decibels by:

$$\varepsilon = \frac{1}{\sqrt{10^{0.1\gamma} - 1}}.$$

As with most analog filters, the Chebyshev may be converted to a digital (discrete-time) recursive form via the bilinear transform. However, as digital filters have a finite bandwidth, the response shape of the transformed Chebyshev is warped. Alternatively, the Matched Z-transform method may be used, which does not wrap with response.

5. Simulation results

The filters are designed and simulated in MATLAB R17.Simulation results are obtained for IIR Butterworth, Chebyshev type 1 and chebyshev type II using the modified analog to digital mapping technique. Coefficients are essential for designing the filter. Hence coefficients are computed apart from simulation results. From simulation results it is observed that Chebyshev type II has better magnitude response than Butterworth and Chebyshev type I window technique.

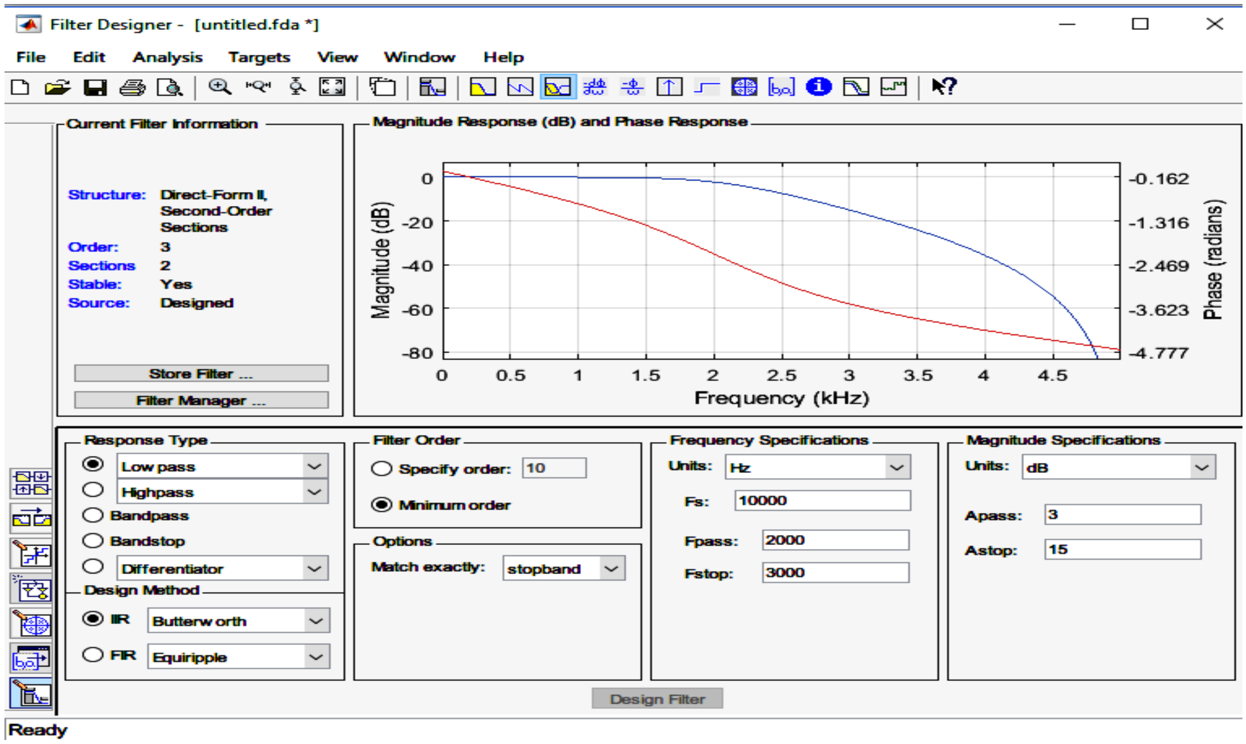

Figure -1: Magnitude and Phase response of Butterworth window technique

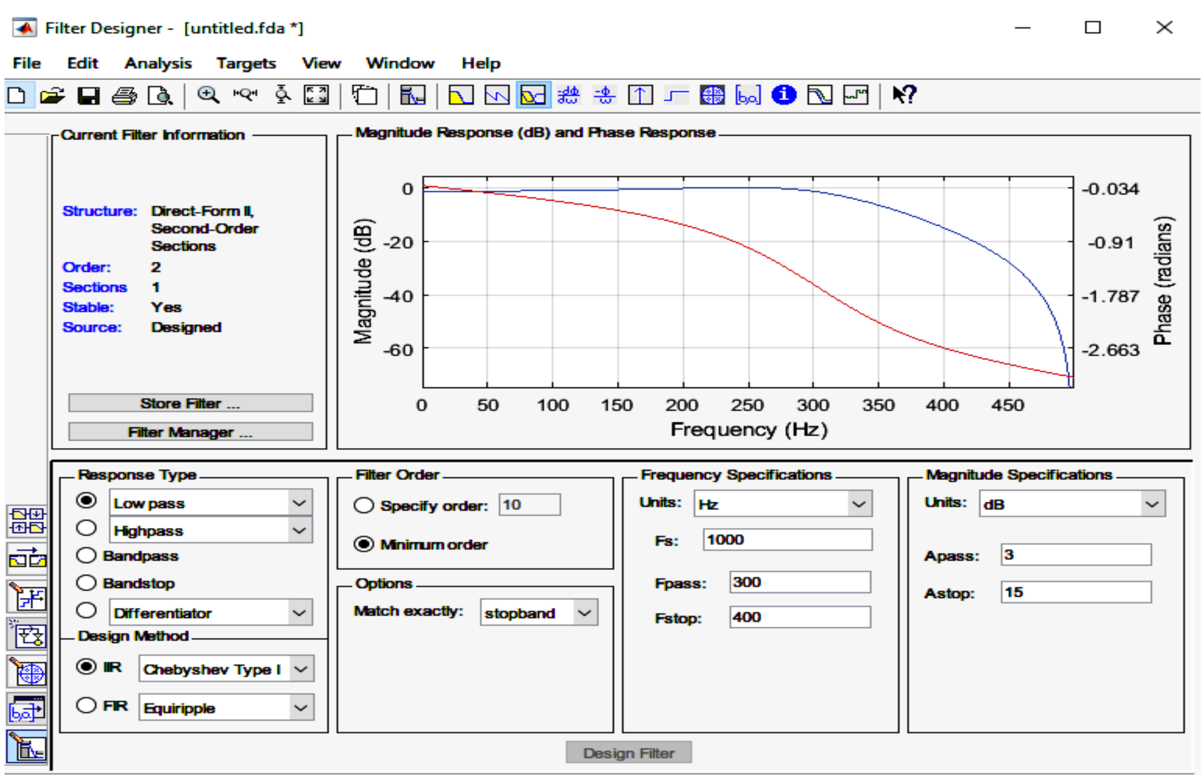

Figure -2: Magnitude and Phase response of Chebyshev type-1 window technique

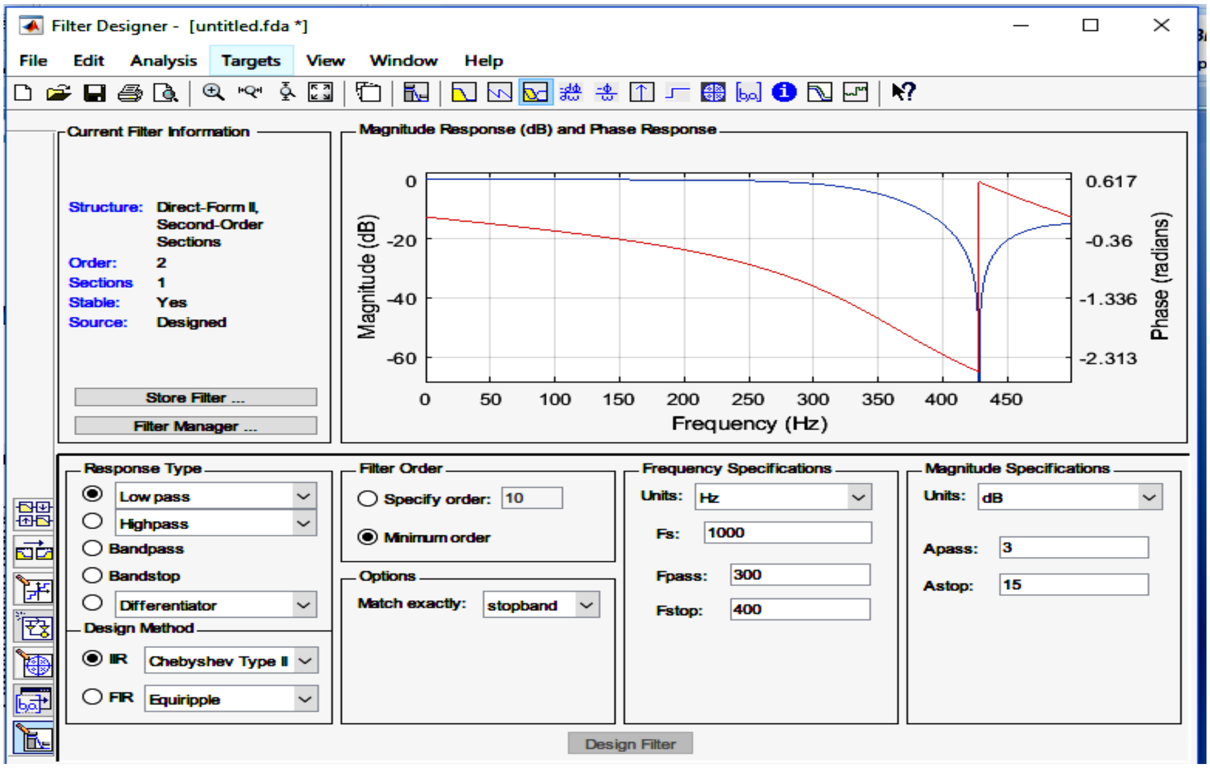

Figure -3: Magnitude and Phase response of Chebyshev type-I1 window technique

Table-1: Filter Coefficients

	Numerator			Denominator			Gain
Butter worth	1	2	1	1	-o.3309	0.3470	0.2540
Chebyshev-I	1	2	1	1	0.3923	0.3630	0.4388
Chebyshev-II	1	1.79944	1	1	0.7047	0.2997	0.5275

6. Conclusion

It can design the filter quickly and effectively by MATLAB, characteristics of filters can be compared to achieve the optimum design. The higher filter order number, the better filtering effect, but the more computation time occupied, so in meet index requirements of the situation should try to reduce filter order number N. IIR digital filter can use lower order number to meet the same design requirements compare with FIR digital filter. Design filter on MATLAB is used in digital signal processing, and has wide application and development prospect.

References

1. L.Fesquet. IIR Digital Filtering of Non-uniformly Sampled Signals via State Representation. Signal Processing, 90(10):2811-2821, Oct 2010.
2. Huang Meng, Tang Lin, Zhen Yu and Zhanz Ze. Optimized FIR Filter Design Based on Self adaptive Genetic Algorithm. Modern Electronics Technique, Feb 2010.
3. Jong-Sik Lim. Design of Low-Pass Filters Using Defected Ground Structure. IEEE Transactions on Microwave Theory & Techniques, Aug 2005.
4. Wang Yang, Zeng and Yi-cheng. A complexity reduction approach for FIR notch filter design. Microcomputer Information, Feb 2010.
5. M.D.Lutovac,D.V.Tosic and B.L.Evans. Filter Design for Signal Processing. Prentice-Hall, 2008.