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Reduction of Group Delay in Butterworth Low pass Filter Using Delay Equalization Filter

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Abstract

The present paper proposes a new technique towards reduction of the group delay in Butterworth Low passfilter to get the desired magnitude response .For designing this Butterworth Low passfilter the approach used is Delay Equalization Filter known as all pass filter .The results in this paper has been achieved with the help of MATLAB and SIMULINK Filter design Toolbox.

Keywords-, all pass Filter, Butterworth Low pass Filter,MATLAB and Simulink Filter design Toolbox.

Introduction

The day by day overwhelming advancements in the fabrication of microchips and their application for the design of efficient digital system led to the emergence of new discipline that has come to be known as digital signal processing (DSP). Through the use of DSP sophisticated communication system has evolved, the INTERNET has emerged, astronomical signals can be distilled into valuable information about the cosmos, seismic signals can analyzed to determine the stability of volcano or strength of earthquake etc. Digital signal processing is the processing of digitized discrete time sampled signals. Processing is done by general-purpose computers or by digital circuits such as ASICs, field-programmable gate arrays or specialized digital signal processors (DSP chips). Typical arithmetical operations include fixed-point and floating-point, real-valued and complex-valued, multiplication and addition. Other typical operations supported by the hardware are circular buffers and look-up tables. Examples of algorithms are the Fast Fourier transform (FFT), finite impulse response (FIR) filter, Infinite impulse response (IIR) filter, andadaptive filters.

Filters-Filters are frequency selective circuits that allow a certain band of frequency to pass while attenuating the other frequencies .Filters are classified as Analog Filter and Digital Filter. The digital filter isone of the most powerful tools of DSP which consist of software andhardware. TheInput and output signals in the digital filter are in discrete form whereas the input and output are in continuous form in Analog filter. Further the filters are classified which depends on usage of Frequency like as Low pass filter, High pass Filter , Band pass Filter, Band reject Filter, Tunable filter, Passive Filter, Active Filter etc.

Low pass filter:-A low-pass filter is an electronic filter that passes low-frequencysignals and attenuates (reduces the amplitude of) signals with frequencies higher than the cutoff frequency.

High pass filter:- A high-pass filter (HPF) is an electronic filter that passes high-frequencysignals but attenuates (reduces the amplitude of) signals with frequencies lower than the cutoff frequency

Band pass Filter:-A band-pass filter is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range.

Band Reject filter:- In signal processing, a band-stop filter or band-rejection filter is a filter that passes most frequencies unaltered, but attenuates those in a specific range to very low levels. It is the opposite of a band-pass filter. A notch filter is a band-stop filter with a narrow stop band (high Q factor).

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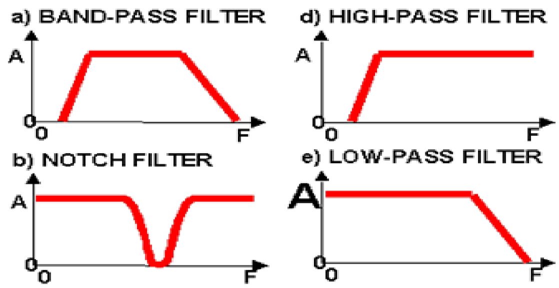

Fig 1 Response of Various types of filter.

Delay Equalization Filter- theother term to all pass filter in MATLAB is iirgrpdelay. In signal processing the all pass filter is a filter which passes all its frequencies with equally in gain but changes its phase relationship between various frequencies. It does this by varying its phase shift as a function of frequency.The all pass filter is used for delay equalization, or to compensate the undesired phase shifts or to convert the unstable filters to stable filters.

Group Delay-Group Delay is a measurement of time or in other words it is that amount of time required for signal to propagate through device.In this paper the concerned is not only with the filter's delay but however also with each frequency component of the signal that experience the same delay so that there phase or shape with each other is maintained.

Problem formulation

In this paper the consideration is mainly on the frequency response error especially on group delay response of Butterworth low pass filter case.The overall delay remains an important consideration in practical applications, thusto reduce the group delay and get the desired frequency response is the main objective of this paper and exploring the results upto approximation of ideal response.

Results formulation

The Butterworth filter is a one of the type of signal processing filter which is designed to have a flat frequency response as possible in pass band. It is also known as maximally flat magnitude filter. Butterworth showed that a low pass filter can be designed whose cutoff frequency when normalized to 1 rad/second, whose frequency response was

$$G(\omega)=\sqrt{\frac{1}{1+\omega^{2n}}}$$

Where w is angular frequency (radians/second)

N is number of poles which is equal to number of reactive elements in passive filter.
Butterworth dealt only in even orders of filters.
The gain of n th order Butterworth low pass filter is given in terms of transfer function H(s) is

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

Where

N is order of filter
Wc=cutoff frequency
Go=DC gain

Thr transfer function is written as

$$H(s)=\frac{G_0}{\prod_{k=1}^{n}\frac{(s-s_k)}{w_c}}$$

Where $polesares_k = w_c e^{\frac{j(2k+n-1)\pi}{2n}}$ k=1,2,….n.

The Magnitude response is the absolute value of a filter's complex frequency response. The phase response is the angle component of a filter's frequency response .The magnitude an phase response functions possess symmetry properties .from the definition of z transform the complex function of real variable ω can be expressed as

$$H(e^{j\omega}) =\sum_{n=-inf}^{inf} h(n)e^{-j\omega n}$$

$$=\sum_{n=-inf}^{inf} h(n)\cos(\omega n) - j\sum_{n=-inf}^{inf} h(n)\sin(\omega n)$$

$$=H_R(e^{j\omega}) + jH_I(e^{j\omega})$$

$$=|H(e^{j\omega})|e^{j\varphi(\omega)}$$

Where

$|H(e^{j\omega})|$ -the quantity is called the magnitude spectrum and it is

$$|H(e^{j\omega})|=\sqrt{{H_R}^2(e^{j\omega}) + {H_I}^2(e^{j\omega})}$$

$\varphi(\omega)$ - the quantity is called the phase spectrum

$$\varphi(\omega) = tan^{-1}[\frac{H_I(e^{j\omega})}{H_R(e^{j\omega})}]$$

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Also it can be expressed as

$$\varphi(\omega) = \frac{1}{2j} ln\left\{\frac{H(z)}{H(z^{-1})}\right\} | z = e^{j\omega}$$

Truncate the terms from infinity upto finite N terms with length (2N+1) ,exploring symmetry property ,thus came to conclusion that magnitude and phase depends on even and odd symmetric parts of frequency response.

$\log[\mathrm{H}(e^{j\omega})] = \mathrm{h_n}(0)+2\sum_{n=1}^{N} \mathrm{h(n)} \cos(\omega n)$

$$\varphi(\omega) = -2\sum_{n=1}^{N} \mathrm{h(n)} \sin(\omega n)$$

Group delay response as [1]

$$\tau(w) = -\frac{d\varphi(w)}{dw} = 2\sum_{n=1}^{N} n.h(n)\, cos(\omega n)$$

All pass filter –The z transform of all pass filter is

$\mathrm{H(z)}=\frac{z^{-1}-\overline{z_0}}{1-z_0 z^{-1}}$

Where z_o is the complex pole zero pair.Complex pole-zero pairs in all-pass filters help control the frequency where phase shifts occur.

With the help of MATLAB and Signal processing Toolbox the Group Delay of Low pass filter is calculated as

1) Determine the sampling frequency (fs),passband frequency(fp) in radian/sec,stopband frequency (fs) radian/sec,passband attenuation (Rp) in decibels ,stopband attenuation in decibels.(Rs)
2) By using the MATLAB commands for designing butterworth filter
[n w]=buttord(fp,fs,amx,amin)
And find the coffiencients of filter
[b a]=butter(n,w)
3) The group delay can be find by
[gd,w]=grpdelay(b,a,512).

The results are following Graphs

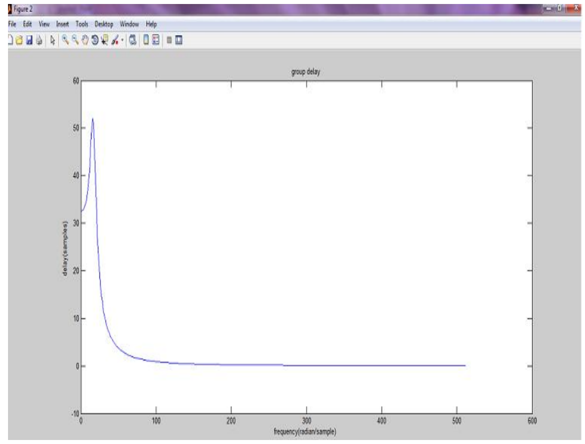

Fig 1 Group Delay of Butterworth Low pass filter.

As it can be seen from fig that the Group Delay is between 30 samples to 50 samples .This Group Delay has been reduced by the delay equalization filter.

The MATLAB has defined a new command known as iirgrpdelay ,through the group delay can be reduced.With the help of MATLAB and Signalprocessing Toolbox the Group Delay of Butterworth Low pass filter has been reduced which is described as is

1) By using the MATLAB commands for designing butterworth filter
[n w]=buttord(fp,fs,amx,amin)
And find the coffiencients of filter
[b a]=butter(n,w)
2) Find the zeros ,poles,gain of filter by MATLAB commands
[z,p,k]=butter(n,w)
3) Convert these z,p,k to sos (second order matrix) and design the all pass filter by iirgrpdelay MATLAB command ,then Calculate the desired group delay ,then design the filter by filter's object

Following is the graph obtained

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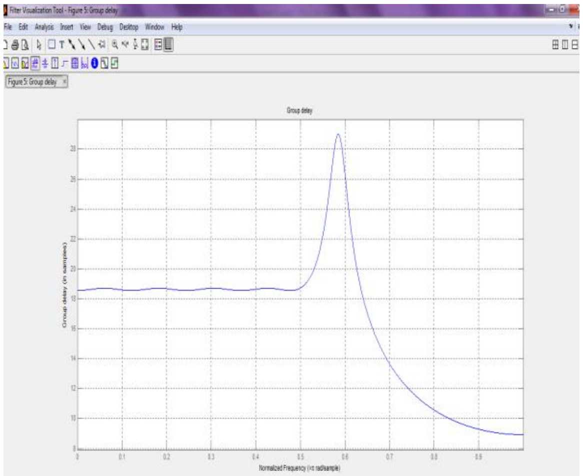

Fig 2 .a.The reduced Group Delay at Rp=1; Rs=40;fp=0.5;fs= 0.6

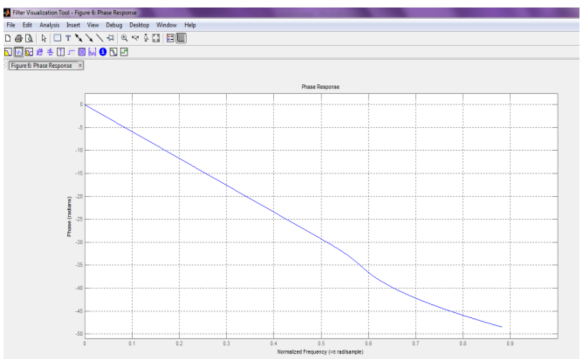

Fig 2.b. The phase response of Butterworth Low pass filter

Results –It can be observed that earlier the Group Delay is between 30 samples to 50samplesshown in fig 1, but after cascading the butterworth low pass filter with the all pass filter the group delay has been reduced upto 18 samples shown in fig 2.a and linearity shown in fig 2.b.

Conclusions

The Group Delay has been reduced with cascading of Butterworth low pass filter with the delay equalization filter. The linearity of the filter is improved as well as the stability is maintained .The group delay should be kept minimum so that the propagation time for signal is improved.

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