Study of Digital Filters used In Signal Processing

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ABSTRACT

In the digital revolution, the uses of digital signals have increased significantly. Digital filters are very important part of DSP. In fact, there is an extraordinary performance in one of the key reasons that DSP has become so popular in various fields.

Typical requirement is considered for comparing the filters, which should have a specific frequency function. The filter should have a specific impulse response. They should be stable, localized and computational complexity implemented in particular hardware or software.

Digital signal processing allows the inexpensive construction of wide variety of filters. This means that any requirement on the frequency function is a requirement on the impulse response. However, in certain applications some filters are used which have explicit impulse response. But which are better than other?

In this paper certain numbers of filters are studied and compared to show which filter is better for processing and analyzing the signal.

Categories and Subject Descriptors

1.5.4 [Applications] Signal Processing

1.4.4 Filtering

General Terms

Design.

Keywords

Digital filters, Error function, Frequency response, Impulse response, Frequency function, Signal separation, Signal restoration, Domain, IIR, FIR.

1 INTRODUCTION

Digital filters are used in computers for communication purposes, cell phones, music and video players, personal video recorders and digital cameras, etc. Digital filters are very important part of digital signal processing. In fact, their extra ordinary performance is one of the key reasons that DSP has become so popular. Digital filters are so good that the performance of the filter is frequently ignored. Filters have two uses these are signal separation and signal restoration. Signal separation is needed when a signal has been contaminated with interferences, noise or other signals. Signal restoration is used when a signal has been distorted in some way. These problems can be attached with either analog or digital filters. Which is better? Analog filter are cheap, fast and have a large dynamic range in both amplitude and frequency. While digital filters, in comparison are vastly superior in the level of performance that can be achieved. The entire transition occurs within only one hertz's. Digital filters can achieve thousands of times better performance than analog filter.

In DSP filters input and output signals are in the time domain. This is because sampling at regular intervals of time usually creates signals. But this is not only way of sampling can takes place. The second most common way of sampling is at the equal intervals in space. Many other domains are possible however; time and space is by far the most common. The time domain in DSP refers to samples taken over time.

Every linear filter has an impulse response, a step response and a frequency response. Each of these responses contains complete information about the filter, but in a different form. All three of these representations are important, because they describe how the filter will react under different circumstances. The most straightforward way to implement a digital filter is by convolving the input signal with the digital signal filters for impulse response. There is also another way to make digital filters, called recursion. When a filter is implemented by convolution, Weighting samples in the input and adding them together calculate each sample in the output. It is also used in many applications such as control system, signal conditioning in real time measurement system and target tracking.

2 TIME DOMAIN

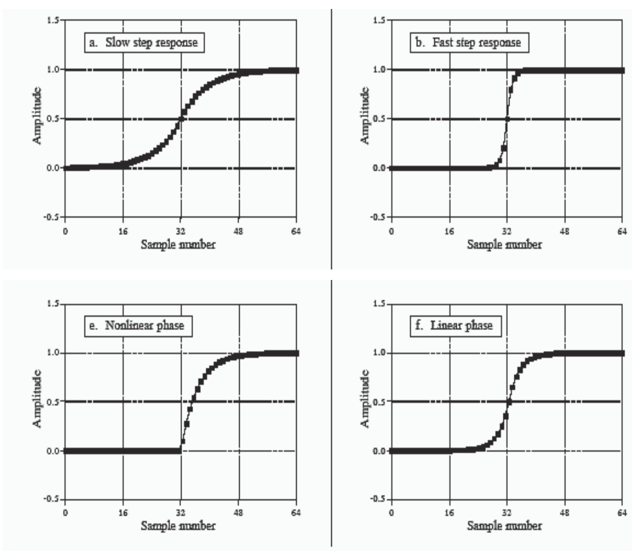
Digital filters play an important role for processing of sound. This sound is produced by a computer or by musical instrument like a violin. Digital filter is used to convert analog signal in to digital signal. Digital filters can also be used to eliminate noise from a sound as to make a sound continues. It also used to remove undesired part of a signal. However, there is an affect that digital filters have on the sound itself. While filtering in the time domain, a filter alters the attack / rise and decay regions of the sound envelope. An envelope of the sound is "the shape of amplitude variation during the modulation". In time domain sampling of the signal is collecting equal discrete part of the analog signal, and quantization is assigning a number to each amplitude of the sampled signal.

2.1 Time domain parameter

Information represented in the time domain describes when something occurs and what the amplitude of the occurrence is. It may not be obvious why the step response is of such concerns in time domain filters.

You may be wondering why the impulse response isn't the important parameter. The step response is useful in time domain analysis because it matches the way human view and the information contained in the signal. The first thing you will do is dividing the signal into regions of similar characteristics. Some of the regions may be smooth, other may have pure amplitude peaks, and other may be noisy. This segmentation is accomplished by identifying the points that separate the signals. The step function is the purest way of representing a division between two dissimilar regions. The step response is important because it describes how the filter is modifying the dividing lines. The step response parameters that are important in filter design are shown in fig.1.

Figure 1. Step response parameters.



To distinguish events in a signal, the duration of the step response must be shorter than the spacing of the events. The step response should be as fast as possible. The most common way to specify the rise time is to quote the number of samples between 10 % and 90 % amplitude levels. The most common processing approach in the time or space domain is enhancement of the input signal through a method called filtering. The filtering generally consist of some transformation of a number of surrounding around the current sample of the input or output signal. There are various ways to characterize filters. For example, a linear filter is a linear transformation of input samples, other filters are nonlinear. Linear filters satisfy the super position condition, if an input is a weighted linear combination of different signals, the output is an equally weighted linear combination of the corresponding output signals.

A causal filter uses only previous sample of the input or output signal, while a noncausal filter uses future input samples. A noncausal filter can usually be changed in to a causal filter by adding a delay to it.

A time invariant filter has constant properties over time, other filters such as adaptive filters change in time.

Some filters are stable, other are unstable. A stable filter produces an output that converges to constant value with time, or remains bounded within a finite interval. An unstable filter produces output, which diverges.

A finite impulse response (FIR) filter uses only input signal, while an infinite impulse response (IIR) filter uses both the input signal and previous sample of the output signal. FIR filter are always stable, IIR filters may be unstable.

3 RECURSIVE FILTERS

In signal processing, a recursive filter is a type of filter, which reuses one or more of its output as an input. This feedback typically results in unending impulse response. However, recursive filter does not always have an infinite impulse response. Recursive filters are an efficient way of achieving long impulse response, without having to perform a long convolution. They execute very rapidly, but have less performance and flexibility than other digital filters.

3.1 Finite impulse response (FIR)

A finite impulse response is a type of digital filter. The impulse response, the filters response to a Kronecker delta input, is finite because it settles to zero in a finite number of sample intervals.

This is in contrast to infinite impulse response filters, which have internal feedback and may continue to respond indefinitely. This type of filter expresses each output sample as weighted sum of the last N inputs, where N is the order of the filter. Since they do not use feedback, they are inherently stable.

If the coefficient is symmetrical, then such a filter is in linear phase, so it delays signals of all frequencies equally. This is important in many applications. It is also straightforward to avoid overflow in FIR filter.

The main disadvantage is that they may require significantly more processing and memory resources than IIR variants. FIR filters are generally easier to design than IIR filters.

3.1.1 PROPERTIES OF FIR.

A FIR filter has number of useful properties, which some times make it preferable to an infinite impulse response filter. FIR filters:

Are inherently stable.
Require no feedback.
They can be designed to be linear phase.

3.2 Infinite impulse response (IIR)

IIR is a property of signal processing system. Systems with that property are known as IIR systems. They have an impulse response function, which is non-zero over an infinite length of time. This is in contrast to finite impulse response filters, which have fixed duration impulse responses. This type of filter is the digital counter part to analog filters. Such a filter contains internal state; output and the next internal state are determined by a linear combination of the previous inputs and outputs.

In theory, the impulse response of such a filter never dies out completely, hence the name IIR, though in practice, this is not true given the finite resolution of computer arithmetic. IIR filters normally require less computing resources than an FIR filter of similar performance. However due to feedback, high order IIR filters may have problems which instability, arithmetic overflow and limit cycles and require careful design to avoid such pitfalls. Additionally, since the phase shift is inherently a nonlinear function of frequency, the time delay through such a filter is a frequency dependant, which can be a problem in many situations. Second order IIR filters are often called 'biquads' and a common implementation of higher order filters is to called cascade biquads.

If an IIR filter implemented, first an analog filter is designed and then it is converted to digital by applying discretization techniques such as bilinear transform or impulse invariance. For ex. Chebyshev filter Butter worth filter and the Bessel filter.

3.2.1 PROPERTIES OF IIR.

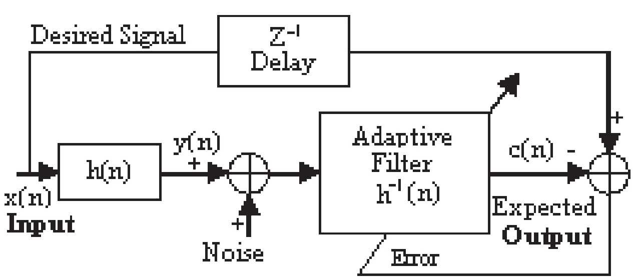
An IIR filter has number of useful properties, which some times make it preferable to finite impulse response filter. IIR filters:

- Are more efficient.
- Have greater processing flexibility.
- Have vast improvement in latency characteristics.

4 ADAPTIVE FILTER

The principal disconsolation is to find a method to calculate an inverse filter using available information y (n) and h(n) or only y(n). The main characteristics of this kind of filters are its ability to adopt its behavior by dynamically changing its internal parameters according to a certain set of learning exercise. There are many algorithms that perform these adapting process, the least means acquire algorithm is one of the most popular. It minimizes the means square error between the output signal and a desired signal as shown in fig.2.

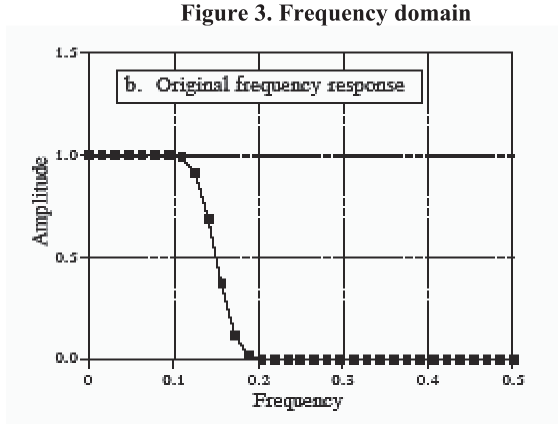
Figure 2. Adaptive Filter.



5 FREQUENCY DOMAIN

It is a term used to describe the analysis of mathematical function or signal with respect to frequency. Signals are converted from time or space domain to the frequency domain usually through the Fourier transform. The Fourier transform converts the signal information to a magnitude and phase component of each frequency. The Fourier transform is converted into the power spectrum, which is the magnitude of each frequency component squared. The most common purpose for analysis of signal in the frequency domain is analysis of signal properties. The engineer can study the spectrum to determine which frequencies are present in the input signal and which are missing.

Figure 3. Frequency domain



There are some commonly used domain transformations for ex. The spectrum converts a signal to the frequency domain through Fourier transform, which takes the logarithm, then applies another Fourier transform. This emphasizes the frequency component with smaller magnitude while retaining the order of magnitude of frequency components. The frequency domain shows how much the signal lies within each given frequency band over a range of frequencies. The frequently domain representation can also include information on the phase shift that must be applied to each sinusoid in order to be able to recombine the frequency components to recover the original time signal. A spectrum analyzer is the tool commonly used to visualize real worlds signals in the frequency domain.

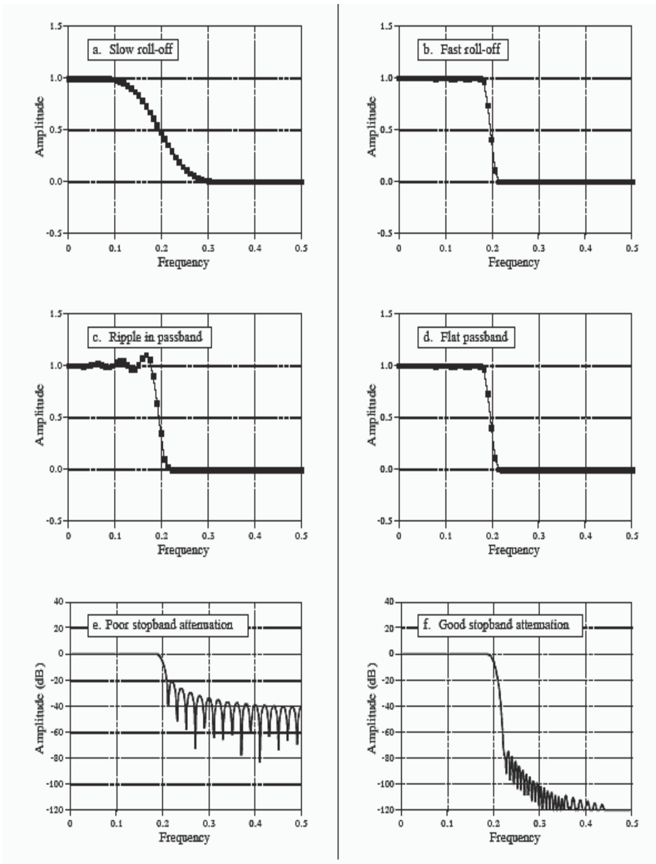
5.1 Frequency domain parameter

The frequency domain filter is to allow some frequencies to pass unaltered, while completely blocking other frequencies. The pass band refers to those frequencies that are passed, while the stock band contains those frequencies that are blocked. The transition band is between a fast roll-off it means that the transition band is very narrow. The division between the pass band and transition band is called the cutoff frequency.

In analog filter design, the cutoff frequency is usually defined to be where the amplitude is reduced to 0.707. Digital filters are less standardized and it is common to see 99%, 90%, 70.7% and 50% amplitude levels defined to be the cutoff frequency. Following figure shows that parameters measure how well a filter performs in the frequency domain. To separate closely spaced frequency, the filter must have a fast roll-off, as illustrated in figure a and b. For the pass band frequencies to move through the filter unaltered, there must be no pass band ripple as shown in fig. c and d. Lastly, to block the stop band frequencies, it is necessary to have a good stock band attenuation as shown in figure e and f. For example butter worth filters have a magnitude frequency response that is monotonic in the Pass band and in the Stop band. To match given constraints, this kind of filter usually requires higher order than Chebyshev or Elliptic filters. Chebyshev filters have a magnitude frequency response that either pass ripple in the pass band and is monotonic in the stop band for Type I Chebyshev filters or has ripple in the stop band is monotonic in the pass band for Type II Chebyshev filters.

For this reason, we considered a Type I Chebyshev filter; Elliptic filters have a magnitude frequency response that had rippled in the stop band and in the pass band.

Figure 4. Frequency response curves.



5.2 Magnitude and phase of frequency domain

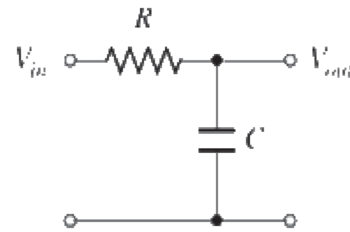
Using the Laplace, z-transform or Fourier transform, the frequency spectrum is complex, describing the magnitude and phase of signal, or of the response of a system, as a function of frequency. The phase information is possible to simplify the information in a frequency domain. A spectrum analyzer is a device that displays the spectrum. The power spectral density is a frequency domain that can be applied to a large class of signals that are neither periodic nor square- integral. Power spectral density of a signal needs only to be the output of a wide sense stationary random process.

6 LOW-PASS, HIGH-PASS, BAND-PASS AND BAND-REJECT FILTERS

6.1 Low-Pass filter

A low-Pass filter is a filter that passes low frequency signals but attenuates signal with frequencies higher than the cutoff frequency. The actual amount of attenuation for each frequency varies from filter to filter. It is some times called treble cut filter. It is used in audio applications.

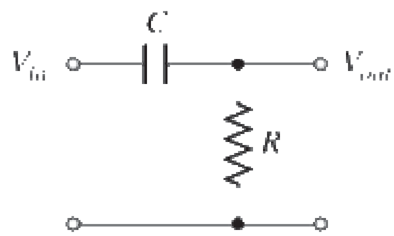
Figure 5. Low-Pass filter



6.2 High-pass filter

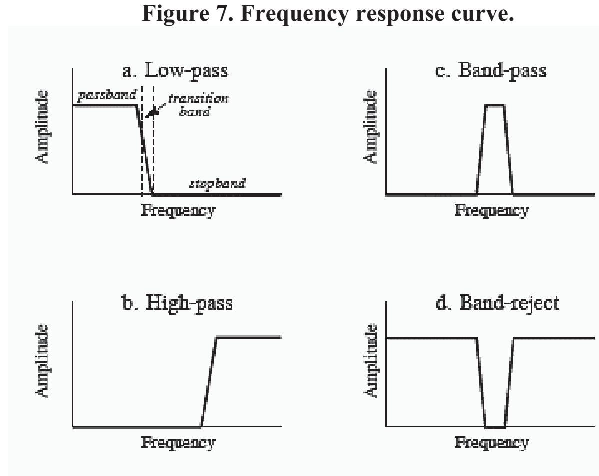
It is a filter that passes high frequencies well, but attenuates frequencies lower than the cutoff frequencies. The actual amount of attenuation for each frequency varies from filter to filter. It is some times called a low-cut filter. A band pass and band reject filters is a combination of high pass and low pass filter.

Figure 6. High-Pass filter



Frequency domain filters are generally used to pass certain frequency, while blocking the other frequency.

Figure 7. Frequency response curve.



7 CONCLUSION

The advantage of digital filters is that their response is repeatable. Phase distortions in audible can be easily achieved by IIR filters. IIR filters offer the processing capabilities to minimize pre-echo and time dispersion while maintaining acceptable cost and improved filter latency characteristics. IIR filters are computationally more efficient, which allows for greater processing flexibility. Digital filters can easily realize performance characteristics far beyond what are practically implementable with analog filters. It is not difficult, for ex. To create 1000 Hz's low-pass filter which can achieve near perfect transmission of 999 Hz's input while entirely blocking a 1001 Hz's signal. Practically analog filters can not discriminate between such closely spaced signals. Adaptive filter minimizes the means square error between the output signal and a desired signal. Frequency domain filters attain much better signal to noise ratio. Digital filters are noiseless for mathematical operations at each intermediate step in the transform. The frequency component exceeding half the sampling rate of the filter. High pass and low pass filters are used in digital image processing to perform transformation the spatial frequency domain. The effect of low pass filter can be simulated on a computer by analyzing its behavior in the time domain and then discretizing the model. The effect of low pass filter can be integrated and zero order hold can be analyzed. The passband filter is used for antialiasing process.

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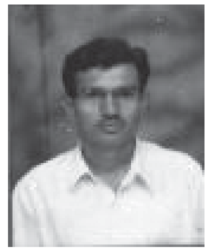
Author Biographies



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