Designing finite impulse response digital filter

Design

To design a realizable and stable lollops digital filter with a finite impulse response; that is, FIR filter and to implement it using windowing method on Exiling thus reaching desired filter specifications. Motivation: Digital filters empower us to bring better performance and higher flexibility at the expense of computational complexity. They are emerging as the microprocessor components and are being used In virtually all spheres - from your cell phones to the cellular base stations.

FIR filtration is a key technology used for the interpretation of stable and linear phase frequency response. This project on FIR filters will give us an insight in a very useful and interesting research area. This will be an opportunity to imbibe and deem over fundamentals and learn about digital tools not covered In the basic curriculum. Scope of the Project: To understand the difference between digital and analog filters. To have profound study with theoretical and mathematical aspects fully transparent about FIR.

To enlist the various issues related to filter designing. To learn the hardware description language, Overlong. To comprehend the various methods of designing an FIR filter - especially the window method. To gauge the pros and cons of the different types of available windows. To learn to work on Exiling SIS Design Suite. To enhance skills by working on Spartan E or Vertex 5 kits. To develop skills on MUTUAL DES toolbox and filter designing tools.

To Implement the various sub-parts of a filter: DC Registers/Memory Time Delay Elements DACCA To design a filter on an FAGAN fulfilling all the chosen specifications. Literature Survey DES--(Digital Signal Processing)

Digital Signal Filter (FIR, IR) Digital Signal Processing Digital signal processing (DES) is concerned with the representation of discrete time signals by a sequence of numbers or symbols and the processing of these signals. Digital signal processing and analog signal processing are sub-fields of signal processing.

DES includes sub-fields like: audio and speech signal processing and , sonar and radar signal processing, sensor array processing, spectral estimation, statistical signal processing, , signal processing for communications, control of systems, biomedical signal processing, seismic data processing, etc. The goal of DES is usually to measure, filter and/or compress continuous real-world analog signals. The first step is usually to convert the signal from an analog to a digital form, by sampling it using an DC, which turns the analog signal into a stream of numbers.

However, often, the required output signal is another analog output signal, which requires a DACCA Even if this process is more complex than analog processing and has a discrete value range, the application of computational power to digital signal processing allows for many advantages over analog processing in many applications, such as error detection and correction in transmission as well as data compression. BLOCK DIAGRAM OF DES DES Tornados

In DES, engineers usually study digital signals in one of the following domains: time domain (one-dimensional signals), spatial domain (multidimensional signals), frequency domain, autocorrelation domain, and wavelet domains. They choose the domain in which to process a signal by making an informed guess (or by trying different possibilities) as to which domain best represents the essential characteristics of the signal. A sequence of samples from a measuring device produces a time or spatial domain representation, whereas a discrete Fourier transform produces the frequency domain information, that is, the frequency picture.

Autocorrelation is defined as the cross-correlation of the signal with itself Time Domain The most common processing approach in the time or space domain is enhancement of the input signal through a method called filtering. Generally consists of some linear transformation of a number of surrounding samples 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 " non-linear". Linear filters satisfy the superposition condition, I. . If an input is a sighted linear combination of different signals; the output is an equally weighted linear combination of the corresponding output signals. A " causal" filter uses only previous samples of the input or output signals; while a " non-causal" filter uses future input samples. A non-causal filter can usually be changed into a causal filter by adding a delay to it. A " time-invariant" filter has constant properties over time; other filters such as adaptive filter. Some filters are " stable", others are " unstable".

A stable filter produces an output that converges to a constant value with time, or remains bounded within a finite interval. An unstable filter can produce an output that grows without bounds, with bounded or even zero input. A " finite impulse response" filter uses only the input signals, while an " infinite impulse response" filter uses both the input signal and previous samples of the output signal. FIR filters are always stable, while IR filters may be unstable. Filters can be represented by block diagrams which can then be used to derive a sample processing algorithm to implement the filter using hardware instructions.

A filter may also be described as a difference equation, a collection of zeros and poles or, if it is an FIR filter, an impulse espouse or step response The output of a digital filter to any given input may be calculated by convolving the input signal with the impulse response. Frequency domain 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.

Often the Fourier transform is converted to the power spectrum, which is the magnitude of each frequency component squared. The most common purpose for analysis of signals 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. In addition to frequency information, phase information is often needed. This can be obtained from the Fourier transform. With some applications, how the phase varies with frequency can be a significant consideration.

Filtering, particularly in non-real time work can also be achieved by converting to the frequency domain, applying the filter and then converting back to the time domain. The main applications of DES are:- audio signal processing audio compression digital image processing video compression beech processing speech recognition digital communications RADAR SONAR seismology biomedicine. DIGITAL FILTERS In signal processing, the function of a filter is to remove unwanted parts of the signal. There are two main kinds of filters, analog and digital. We prefer digital filter over analog because of the following : 1) A digital filter is programmable.) Digital filters are easily designed, tested and implemented. 3) The characteristics of analog filter circuits are subject to drift and are dependent on temperature. Digital filters do not suffer from these problems, and so are extremely stable with respect OTOH to time and temperature. 4) Digital filters can handle low frequency signals. There are a few terms used to describe the behavior and performance of Digital FIR filter including the following: Filter Coefficients-The set of constants, also called tap weights, used to multiply against delayed sample values.

For an FIR filter, the filter coefficients are, by definition, the impulse response of the filter. Impulse Response-A filter's time domain output sequence when the input is an impulse. An impulse is a single unity-valued sample followed and preceded by zero-valued samples. For an FIR filter the impulse response of a FIR filter is the set of filter coefficients. Tap-The number of FIR taps, typically N, tells us a couple things about the filter. Most importantly, it tells us the amount of memory needed, the number of calculations required, and the amount of " filtering" that it can do.

Basically, the more taps in a filter results in better stop band attenuation (less of the part we want filtered out), less rippling (less variations in the pass band), and steeper roll off (a shorter transition between the pass band and the stop band). Multiply-Accumulate (MAC)-Len the context of FIR Filters, a " MAC" s the operation of multiplying a coefficient by the corresponding delayed data sample and accumulating the result. There is usually one MAC per tap. APPROACH 1 . Window Design methods 2. Frequency sampling method 3.

Weighted Least Square design 4. Parks McClellan Method Our Approach:- Window design method 1. Kaiser Window 2. Rectangular Window 3. Hahn window 4. Hamming window 5. Bartlett Hahn Kaiser Window This mathematical function basically implements the Kaiser window: Kaiser window has variable shape. This implies that in the above equation, by changing the value of alpha we can control the relationship between main lobe width ND side lobe height. This is one of the main reasons behind its widespread popularity.

Kaiser window has the following frequency response (shown in figure on next page):- Fig: Comparison between time domain representation of Kaiser Window for different values of pi*alpha (blue for 2. 5 and green for 10). Fig: Low pass filter of length 73 designed using Kaiser Window. Fig: Plot of frequency response of low pass filters: blue has pi*alpha = 0. 2, green has pi*alpha= 4 As can be seen from the graph, at higher values of pi*alpha, though the stop band attenuation is higher, the mainline width has also increased. In the practical world, there is always a trade-off between these two filter characteristics.

In mathematics, Bessel functions, first defined by the mathematician Daniel Bernoulli and generalized by Frederica Bessel, are canonical solutions y(x) of Vessel's Given below is plot of Bessel function of first kind: The Bessel function has a power series expansion which results in general computational complexity. Hamming Window: The " raised cosine" with these particular coefficients was proposed by Richard W. Hamming. The window is optimized to minimize the maximum (nearest) side lobe, paving it a height of about one-fifth that of the Hahn window, a raised cosine with simpler coefficients.

Coos Hyperbolic Window: We have used a new class of adjustable windows based on the cosine hyperbolic function. The proposed window has been derived in the same way as the Kaiser window, but it has computational cost advantage due to having no power series expansion in its time domain representation. After the spectrum design for the proposed function is obtained through MUTUAL, comparisons with the Kaiser window in terms of various spectral characteristics are performed. Simulation results show hat the proposed window performs better sideline roll-off ratio than the Kaiser window for the same window length and normalized mainline width.

Moreover, the paper presents the application of the proposed window in finite impulse response (FIR) filter design. The results show that the filters designed by the proposed window provide better far end Stoppard attenuation than the filters designed by the Kaiser window. The formula for this proposed window is Fig : Comparison between Bessel function (red) and Cosine hyperbolic function (green) Fig: Comparison between Coos Hyperbolic (blue) and Kaiser (green) window for a filter f length 20. As can be seen, the main lobe is narrower and the side lobe ripple ratio is lower in the coos hyperbolic window.