Research Journal of Engineering Sciences E- ISSN 2278 – 9472
Vol. **5(4),** 26-29, April **(2016)** *Res. J. Engineering Sci.*

Digital Filter Design

M.S. Mandloi
Department of Electronics and Communication Engineering, Rewa Engineering College Rewa, 486002, M.P., India
mantar_singh2003@rediffmail.com

Available online at: www.isca.in, www.isca.me
Received 5th January 2016, revised 29th February 2016, accepted 14th April 2016

Abstract

Filters are useful for communications and signal processing. Filter is used for reduce noise,audio processing and video processing. Digital filters are generally used in communication systems. Digital filter performance is important as a good result. Design of digital filter satisfy necessary conditions. Digital filter provide good results.design of low pass, high pass, band pass and band stop filters through digital filter. Digitall filter is calculate filter coefficients for different structures of filter. Digital filter is determine filters specifications.

Keywords: FIR Filter, IIR Filter, Windowing.

Introduction

Digital filtering is a sequence of discrete data which remove noise. Digital filter change the sample rate, and/or modify the input information[1]. FIR filter have no feedback. Finite impulse response filter is more stable because zero phase shift provide. IIR filter has provide feedback. IIR filter has unstable because out of phase is provided.

FIR Filters

FIR filter is the moving average operation is given following equation[2,3]:

$$y(n) = \sum_{k=0}^{M} \mathrm{b_k} \mathrm{X}\ [\mathrm{n\text{-}k}] \qquad b_k = \frac{1}{M+1} \tag{1}$$

Where: X[n] is filter input at index n sample, Y[n] is filter output at index n sample, M is filter order, b_k is filter coefficients, Z Transform can define as

$$\mathrm{X}\ (\mathrm{z}) = \sum_{n=0}^{\infty} \mathrm{X}\ [\mathrm{n}] z^{-k} \tag{2}$$

Where[4]: X (z) is z-transform of x $[n]$, Z is complex variable; Output of z transform of filter is following:

$$y(z) = \sum_{k=0}^{M} \mathrm{b_k}\, z^{-k} \mathrm{X}\ (\mathrm{z}) \quad b_k = \frac{1}{M+1} \tag{3}$$

Transfer function of z domain is written as following:

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{k=0}^{M} \mathrm{b_k}\, z^{-k} \quad b_k = \frac{1}{M+1} \tag{4}$$

Transfer function of frequency domain can found following[4]:

$$\mathrm{Z} = e^{j2\pi f} \tag{5}$$

Where: j= $\sqrt{-1}$, this is writing as following:

$$\mathrm{H}\ (e^{j2\pi f}) = \frac{1}{M+1} \sum_{k=0}^{M} e^{-j2\pi f k} = \frac{1}{M+1} e^{-j\pi f M} \frac{\sin[\pi f(M+1)]}{\sin(\pi f)} \tag{6}$$

Moving average filter phase response and magnitude response which value M=7 determine through (H $e^{j2\pi f}$) shown in Figure-1.

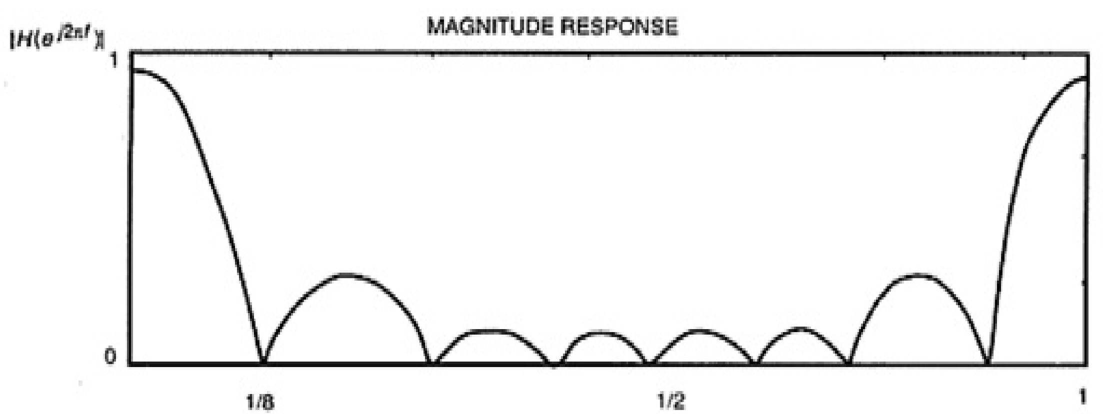

Figure-1
Magnitude response of simple MAF (moving average filter) with M=7

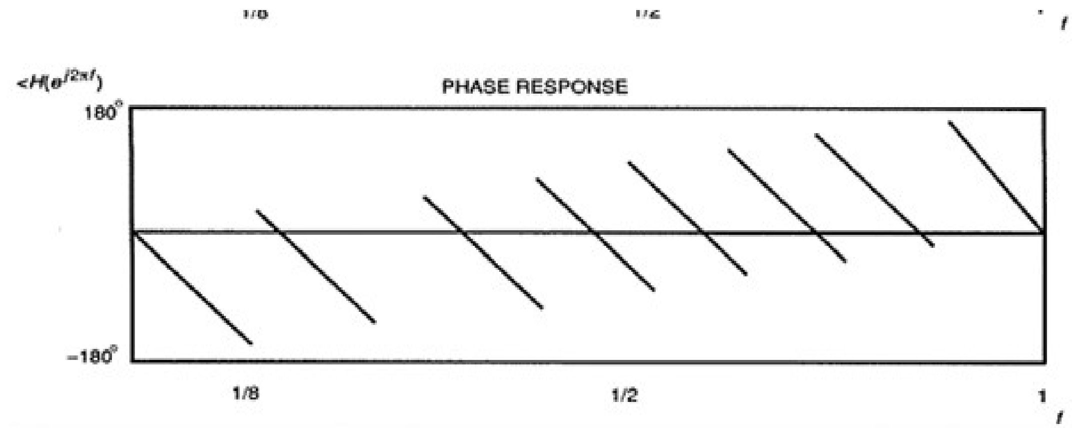

Figure-2
Phase response of simple MAF (moving average filter) with M=7

Filter Design Technique

There are three techniques of filter design: **Window method:** The window method calculates ideal impulse response and multiplying a window function to improving filters frequency response.

Frequency sampling method: This sampling technique is filter specification in frequency domain technique and samples can determine inverse transform of weighting coefficients.

Optimal approximations: Optical approximations provide good results with the optical approximation method.

Application

Digital filter has implemented following properties: i. Noise Reduced. ii. Remove interferences, iii. Ultimate realizations, iv. Remove DC signals

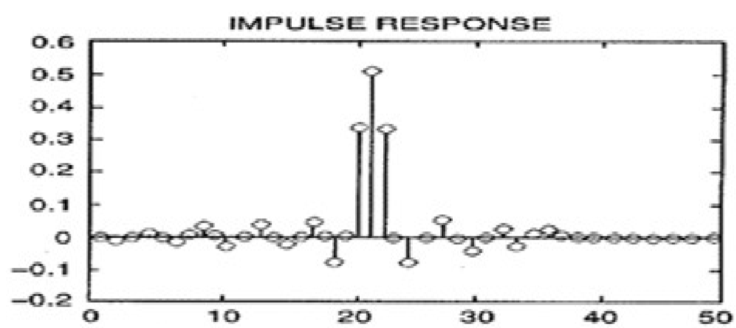

Figure-3
Impulse response of optimal fourteen order semi band finite impulse response filter

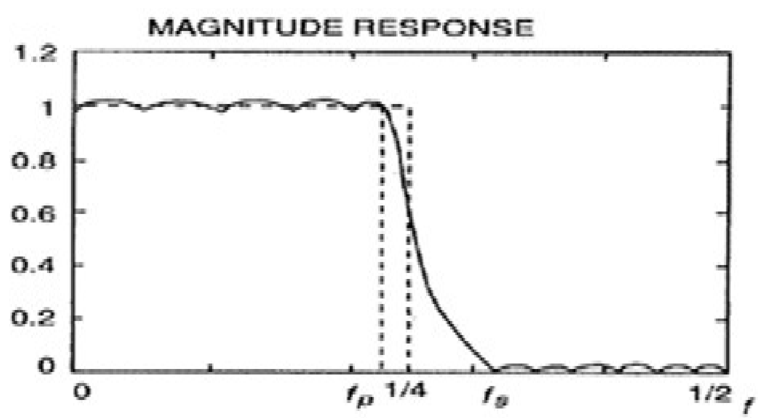

Figure-4
Magnitude response of optimal fourteen order semi band finite impulse response filter

These Characteristics use in FIR filter applications for telecommunication. FIR filter is stable.

Finite Word length Effects

Digital filter is implemented by finite numbers and arithmetic. Digital filter result filter coefficients and its input and output signal in discrete form. Effect of finite word length is following:

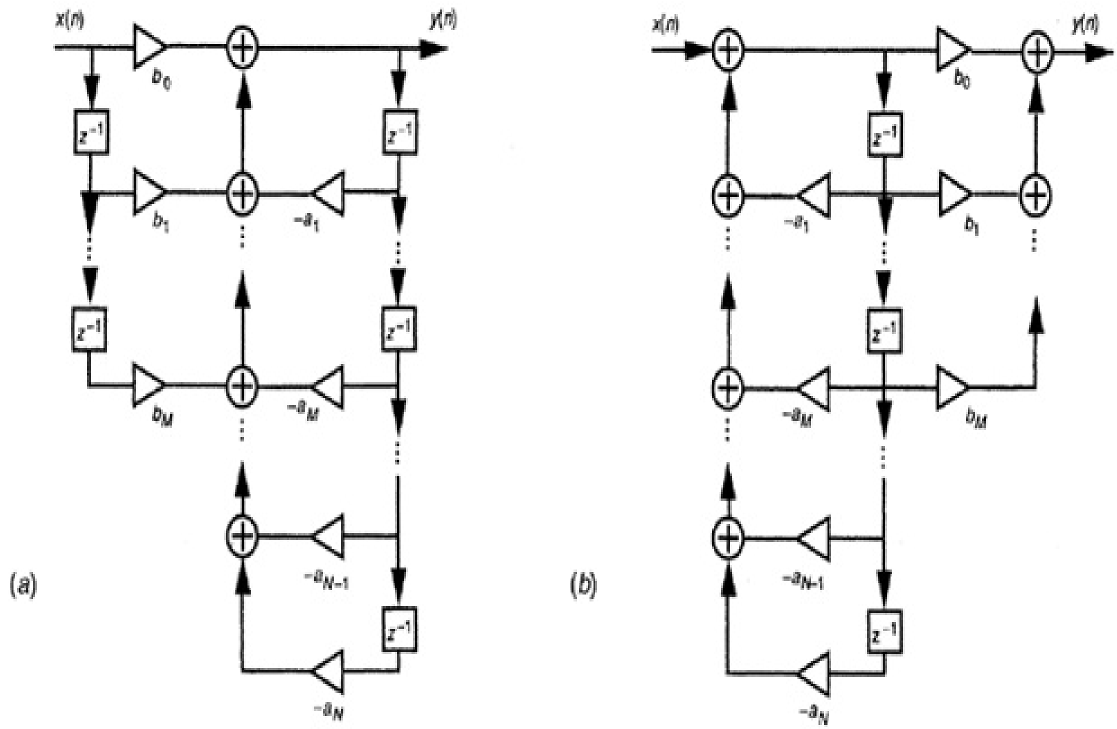

Figure-5
Realization of IIR filter (Direct from): (a) direct form I[5], (b) direct form II[5]

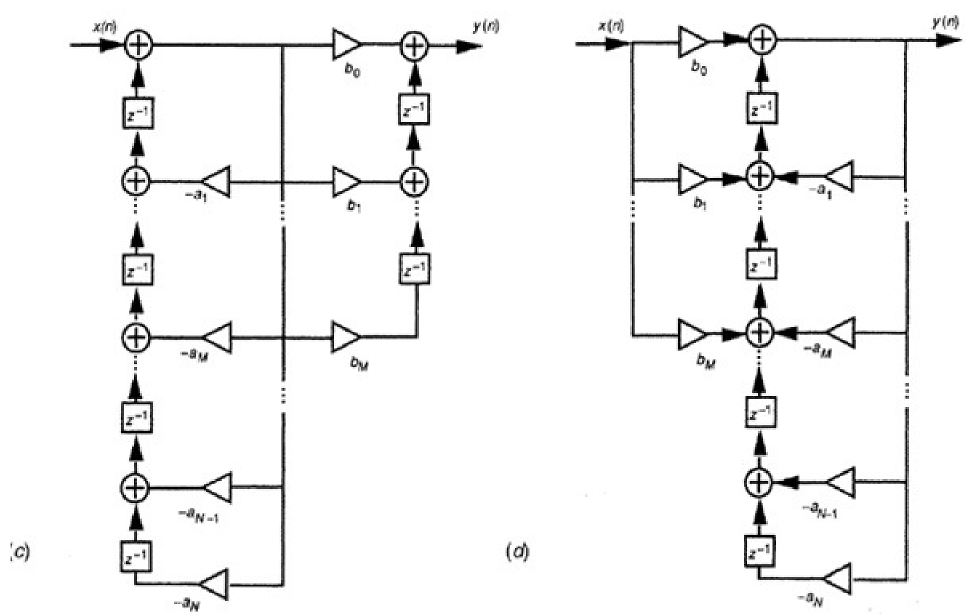

Figure-6
Realization of IIR filter (Direct from): (c) transposed direct form I[5], (d) transposed direct form II[5]

Discretization (quantization): As a result, the actual filter response differs slightly from the ideal response. This deterministic frequency response error is referred to as coefficient quantization error[6].

The use of finite precision arithmetic makes it necessary to quantize filter calculations by truncations. Quantization of the filter calculations also renders the filter slightly nonlinear. For large signals this nonlinearity is negligible, and round off noise is the major concern. With fixed-point arithmetic it is possible for filter calculations to overflow. The term overflow oscillation refers to a high-level oscillation that can exist in an otherwise stable filter because of the nonlinearity associated with the overflow of internal filter calculations. Another term for this is an adder overflow limit cycle.

IIR Filter: This filter also called recursive. IIR filter is always unstable. IIR filter is a non linear phase[7].
This filter transfer function can write as following:

$$H(z) = \frac{\sum_{i=0}^{P} b_{iz^{-i}}}{1+\sum_{j=1}^{Q} a_{jz^{-j}}} \qquad (7)$$

Where: b_i Does the coefficient of filter, Q is the feedback, a_i are filter coefficients (feedback).

Design using an analog prototype filter, in which an analog filter is designed to meet the (analog) specification and the analog filter transfer function, is transformed into a digital system function.

Design using digital frequency transformation, which assumes that a given digital low-pass filters, is available, and the desired digital filter is then obtained from the digital low-pass filter by a digital frequency transformation.

Conclusion

Digital filter design is useful for analysis of methods finding how will effect of filter spectrum. Digital filter is providing linear phase response. Digital filter is also provides transient and impulse response. Digital filter is also provides minimum and maximum phase response. Digital filter used for represents of signal and spectra through complex numbers. Digital filter some problem solve by MATLAB environment.

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