

DIGITAL SIGNAL PROCESSING

Chapter 5 FIR Filter Design



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FIR Filter design

- Aims
 - To explain type of filter, FIR filter specifications and FIR filter design steps.
- Expected Outcomes
 - At the end of this course, students should be able to design digital FIR filter based on the filter specifications and steps.



Definition of Filter

Filter is required in the digital signal processing to filter the raw input signals to the desired frequency and suppress noise in signal processing.

- Filter consists of Finite Impulse Response (FIR) and Infinite Impulse Response Filter (IIR).
- There are four type filter such as Low -pass, High-pass, Band-pass and Band -stop filter.



FIR Filter

It is a system where the output of the system only depend on the input signals.
The system only has zeros and no poles.
The system has no feedback.
The system always stable.
Example of the difference equation that can describe the system;

 \rightarrow y(n) = x(n) + x(n-2) - 2x(n+1)



FIR Filter Specifications

- □ The process of filter design begins with filter specifications which include the filter characteristics (Low-pass, high-pass, band-pass, band-stop filter), Filter Type (**FIR** or **IIR**), passband frequency, stopband frequency, transition width frequency, cut-off frequency, sampling frequency, filter order (N), stopband attenuation and passband ripple.
- The second step is to calculate filter frequency Response, H(e^{jω}).
- The third step is to find the filter coefficient or filter impulse response.
- The last step is to implement filter coefficient and choose appropriate filter structure for implementation.

FIR Filter Specifications

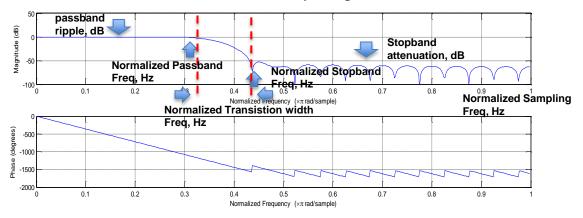
- Before designing the FIR Filter, the set of the specification must be defined.
- □ In order to do that, need to determine the cut-off frequency, $\omega_c (2\pi f_c)$ for the filter.
- □ The Frequency Response for the ideal Low-Pass Filter is: $H_d(e^{j\omega}) = e^{-j\omega}$, $|\omega| \le \omega_c$ 0, $\omega_c < |\omega| \le \pi$



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FIR Filter Specifications

□ The FIR Low-Pass Filter is illustrated by diagram below:



Amplitude Parameters;

- $\Box \qquad \delta_p = \text{ Passband Ripple, } R_p = -20 \log[(1 \delta_p)/(1 + \delta_p)] \text{ in dB}$
- **D** Peak ripple, $\alpha_p = -20\log[(1 \delta_p)]$ in dB
- $\Box \qquad \delta_s = \text{ Stopband Ripple/Attenuation,}$
- $\Box \qquad A_s = -20log[(\delta_s / 1 + \delta_p)] \text{ in dB}$
- \Box Minimum stopband attenuation, $\alpha_s = -20log[(\delta_s)]$ in dB

Frequency Parameters;

- $\Box \qquad Sampling frequency = F_s$
- \Box Passband frequency, $\omega_p = 2\pi f_p$, Normalized Passband frequency = ω_p / F_s
- \Box Stop-band frequency, $\omega_s = 2\pi f_s$, Normalized Passband frequency = ω_s / F_s
- $\Box \qquad \text{Transition width, } \Delta \omega = \omega_{s} \omega_{p}$
- $\Box \qquad \text{Cut-off frequency, } \omega_{c} = (\omega_{s} + \omega_{p}) / 2$



FIR Filter Coefficient (Impulse response)

The Impulse Response of the Filter is defined as: $h_{d}(n) = \frac{\sin(n - \alpha)\omega_{c}}{\pi(n - \alpha)} \text{ where } \alpha = N/2$

□ The FIR Filter Specification is defined as below;

$$\begin{split} 1 - \delta_{p} < \mid H(e^{j\omega}) \mid &\leq 1 + \delta_{p} \ , \ 0 < \mid \omega \mid \leq \omega_{p} \\ \mid H(e^{j\omega}) \mid &\leq \delta_{s} \ , \ \omega_{s} \leq \mid \omega \mid < \pi \end{split}$$



The Filter is designed by *windowing* the impulse response:

 $h(n) = h_d(n)w(n)$

➤ w(n) is a finite-length window that is equal to zero outside the interval of 0 ≤ n ≤ N



- Basically, there are 4 type of window :
- ➢ Rectangular
 - $w(n) = 1, \ 0 \le n \le N$
 - 0, elsewhere
- ≻Hanning
 - $w(n) = 0.5 0.5 \cos(2\pi n/N), 0 \le n \le N$
 - 0, elsewhere



➤ Hamming

 $w(n) = 0.54 - 0.46\cos(2\pi n/N), 0 \le n \le N$

0, elsewhere

> Blackman

w(n) = 0.42 - 0.5cos($2\pi n/N$) + 0.08cos($4\pi n/N$), 0 ≤ n ≤ N

0, elsewhere



 The relationship between the length of window, N and Filter Transition width is shown below:

$$N\Delta f = c$$

• c is a parameter of the window.



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- The window parameter, c is shown below:
 - 1. Rectangular $N\Delta f = 0.9$, $\alpha_s = -21 \text{ dB}$
 - 2. Hanning $N\Delta f = 3.1$, $\alpha_s = -44 \text{ dB}$
 - 3. Hamming $N\Delta f = 3.3$, $\alpha_s = -53 \text{ dB}$
 - 4. Blackman $N\Delta f = 5.5$, $\alpha_s = -74$ dB



FIR Filter Design by Window Method: Example

Design the FIR Filter to meet the following specification by using window method.

```
0.99 \le |H(e^{j\omega})| \le 1.01, \ 0 < |\omega| \le 0.19\pi
|H(e^{j\omega})| \le 0.01, \ 0.21\pi \le |\omega| \le \pi
```

Solution:

- 1. From the spec given, $\delta_s = 0.01$, thus $\alpha_s = -40 \text{ dB}$.
- 2. From the window parameter, the stopband attenuation is close to **Hanning** Window, thus it is preferable to use **Hanning** Window for this design.



FIR Filter Design by Window Method: Example 1

- 3. Now, calculate the Transition Band or Width, $\Delta \mathbf{f}$: From the given spec, $\Delta \boldsymbol{\omega} = \omega_s - \omega_p = 0.02\pi$, Thus, $\Delta \mathbf{f} = \Delta \boldsymbol{\omega} / 2 = 0.01$
- 4. By using Hanning Window, $N\Delta f = 3.1$

Thus, N = 3.1 / Δf = 3.1 / 0.01 = 310

5. Next, determine the Cut-Off Frequency, $\omega_{\rm c}$

 $\omega_{\rm c} = \omega_{\rm s} + \omega_{\rm p} / 2 = 0.21\pi + 0.19\pi = 0.40\pi / 2 = 0.2\pi$

6. Calculate the delay, α

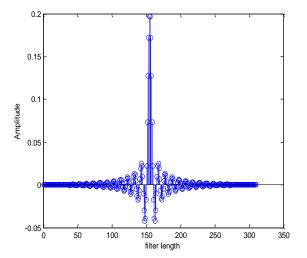
 $\alpha = N/2 = 310 / 2 = 155$

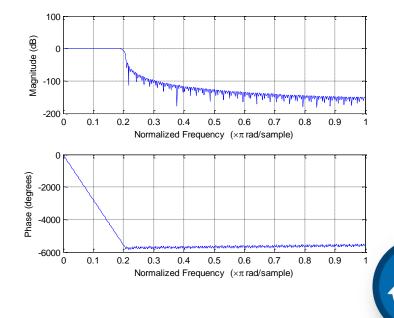
7. Finally, the Impulse Response of the FIR Filter that meet the spec is:

 $h_d(n) = sin [0.2\pi(n-155)]/(n-155)\pi$

FIR Filter Design by Window Method: Example

- A plot of FIR filter coefficient using Window method based on example 1;
- FIR Filter Coefficients, FIR Filter response





Design the Bandpass Filter to meet the following specification:

- passband frequency = 900-1100 Hz
- passband ripple = < 0.87 dB
- stopband attenuation > 30 dB
- sampling frequency = 15 kHz
- Transition frequency = 450 Hz
- Use Optimal Method to find suitable Filter Coefficients.



Solution:

1. Normalized all the frequencies by dividing the passband and stopband frequencies with sampling frequency.

450 -> 450/15000 = 0.03 900 -> 900/15000 = 0.06 1100 -> 1100/15000 = 0.073 1550 -> 1550/15000 = 0.1033 7500 -> 7500/15000 = 0.5

2. Obtain the Passband Ripple & Stopband Attenuation,

$$\delta_{p} = 0.10535, \delta_{s} = 0.031623$$



The value of N is determined by:

N =
$$-20\log_{10}(\sqrt{\delta_{p}\delta_{s}}) - 13 + 1$$

14.6 Δ f/Fs

Thus,
$$N = -20log_{10}(0.057719) - 13 + 1$$

14.6 (0.03)

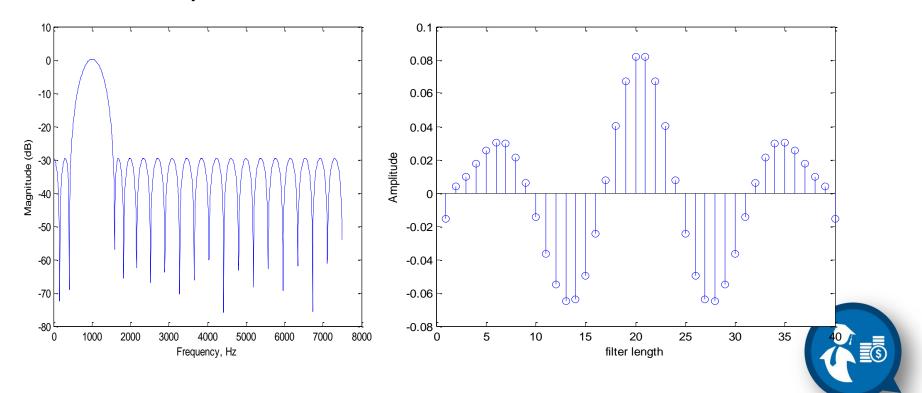
- The suitable Filter length, N = 28
- The Filter Coefficient can be determined by: h_d(n) = sin [0.015π(n-14)]/(n-14)π, 0 ≤ n ≤ N-1



Filter coefficients

Now, plot the filter coefficients based using Optimal method based on example 2;

□ Filter response;



FIR Filter Design by Optimal Method

- Optimal Method provides an easy and efficient way of computing FIR Filter Coefficients.
- For most application, optimal method will yield filters with a good amplitude response characteristics for reasonable values of N.



FIR FILTER DESIGN



To design FIR filter based on the filter specifications To improve filter design based on To identify type filter specifications of filter and filter and filter order or specifications length 3 To convert filter To calculate filter specifications coefficients using into the desired window or optimal filter method Communitising Technology

Conclusion

- Able to understand the type and characteristics of FIR filter design.
- Able to design the FIR filter using window and optimal method.
- Able to calculate the filter coefficients or filter impulse response from the filter specifications.





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