# SIO 207A: Fundamentals of Digital Signal Processing Class 13

Florian Meyer

Scripps Institution of Oceanography Electrical and Computer Engineering Department University of California San Diego

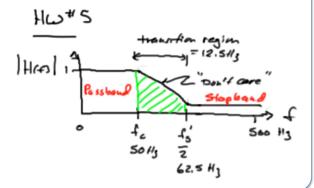




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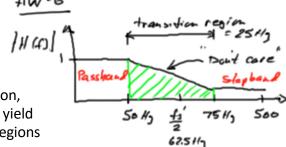
### Filter Design – Homework 5 & 6

- In Homework 5, transition width specified as  $50 {\rm Hz} \to 62.5 {\rm Hz}$  to protect against aliasing (all frequencies above  $f_{\rm s}'=62.5 {\rm Hz}$  will alias into region below  $62.5 {\rm Hz}$ )
- However, since the transition region is a "don't care" region, might as well allow aliasing of signals above  $62.5 \rm{Hz}$  into this region



#### Filter Design – Homework 5 & 6

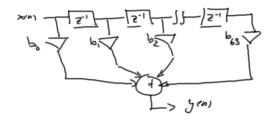
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- However, since the transition region is a "don't care" region, might as well allow aliasing of signals above  $62.5 \rm Hz$  into this region
- Thus expanding the transition region to  $50 Hz \to 75 Hz$  will not result in aliasing into the region we care about  $(0 Hz \to 50 Hz)$



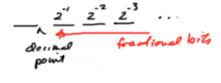
 The benefit is that the wider the transition region, the easier it is for the filter design algorithm to yield smaller ripples in the passband and stopband regions

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## Filter Design – Homework 6

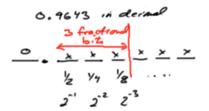


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- Normalize impulse response h[n] so that largest coefficients is 1
- Multiply coefficients by  $2^b$  , i.e.,  $b=3,2^3=8$
- Round to nearest integer part
- Multiply by  $2^{-b}$

• Look at the characteristics of quantized impulse response



## Frequency Sampling Approach to FIR Filter Design

 Let us assume we design an ideal filter in discrete frequency domain by setting all samples in stopband to 0

$$H(e^{j\omega}) = e^{-j\omega\frac{N-1}{2}} \sum_{k=0}^{N-1} \tilde{H}(k) \; e^{j\frac{2\pi}{N}k\frac{N-1}{2}} \frac{\sin\left[N\frac{\omega-\frac{2\pi}{N}k}{2}\right]}{\sin\left[\frac{\omega-\frac{2\pi}{N}k}{2}\right]}$$
 
$$\tilde{h}[n] = \mathrm{IFFT}\Big(\tilde{H}(k)\Big) \qquad \qquad \int_{\mathrm{samples of desired frequency response}}^{\mathrm{N-1}} \mathrm{d}k \, dk$$

impulse response with N samples period

see also Section 7.4 in Oppenheim & Schafer, 1999: "Optimal Approximations of FIR Filters"

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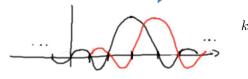
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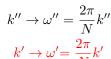
impulse response with N samples period

FIR filter response h[n] is equal to the first N samples of  $\tilde{h}[n]$  and zero otherwise

→ multiplication of h[n] with rectangular window



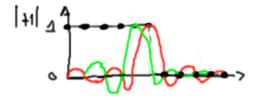
frequency response

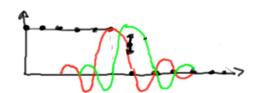


# Frequency Sampling Approach to FIR Filter Design

• Instead of just using samples of the ideal response, include 2-3 transition regions samples to mitigate the high sidelobes

Continuous Frequency Response Synthesis of "Ideal" Discrete Filter Continuous Frequency Response Synthesis: Equiripple Design



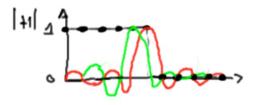


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Continuous Frequency Response Synthesis of "Ideal" Discrete Filter Continuous Frequency Response Synthesis: Equiripple Design



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Adjust value of transition sample to minimize stopband response levels

see also Section 7.4.3 in *Oppenheim & Schafer, 1999*: "The Parks-McClellan algorithm" (Equiripple design)