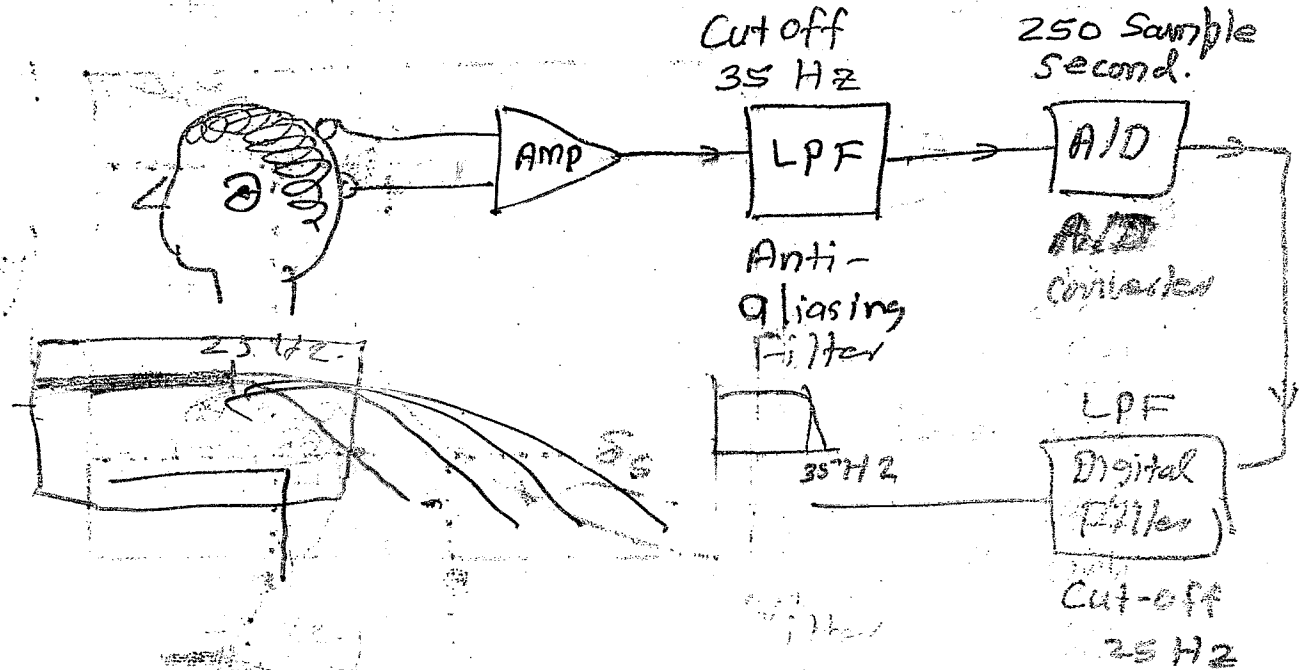


# Example 8.14 Digital Butterworth Filter

\* EEG Signals  $\sim$  0-25 Hz



\* Design a Digital Butterworth Filter with cutoff 25 Hz.

\* General Analogue prototype (Continuous Time)  
 Unity Gain CT Butterworth Filter of order N

$$|H(\omega)|^2 = \frac{1}{1 + \left(\frac{\omega}{\omega_c}\right)^{2N}}$$

1) In order to design the digital filter starting from an analog (CT) prototype, we need to know N and  $\omega_c$ .

2) The digital filter has to be specified in the digital domain. That is, filter

① In the given problem:

$$f_s = \text{Sampling Frequency} = 250 \text{ Hz}$$

Desired  $\omega_p$  }  $\omega = \Omega T$

$$\omega_p = \Omega_p \cdot T = (2\pi \times 25) \frac{1}{250}$$

$$H(\omega) = \dots = 0.2\pi$$

② Pick a value for  $\omega_s$

$$\text{Let } \omega_s = 0.3\pi$$

③ Choose  $\delta_p$  and  $\delta_s$

$$\text{Let } \delta_p = -1 \text{ dB}$$

$$\delta_s = \dots \text{ dB}$$

④ Now the digital specifications are complete.

⑤ Translate the specs. to analog domain.

$$\omega_p \longrightarrow \Omega_p = \frac{\omega_p}{T} = \frac{0.20\pi}{(1/250)} = 157 \text{ rad/s}$$

$$\omega_s \longrightarrow \Omega_s = \frac{\omega_s}{T} = \frac{0.3\pi}{(1/250)} = 235.6 \text{ rad/s}$$

• When  $\Omega = \Omega_p$ , gain = -1 dB

$$20 \log_{10} |H(\Omega_p)| = -1$$

$$20 \log_{10} |H(\Omega_p)|^2 = -0.4 \Rightarrow \frac{1}{1 + \left(\frac{\Omega_p}{\Omega_c}\right)^{2N}} = 10^{-0.2}$$

$$\frac{1}{1 + \left(\frac{\Omega_p}{\Omega_c}\right)^{2N}} = 10^{-0.2} \quad (1)$$

Substituting  $H(s)$  in the standard form

• When  $\Omega = \Omega_s$ , gain = -15 dB

\* Sampling rate = 1000 / second

$$\frac{1}{1 + \left(\frac{\Omega_s}{\Omega_c}\right)^{2N}} = 10^{-1.5} \quad (2)$$

P2. Solution:

$$\Rightarrow N = 5.88 \rightarrow \text{Set } N = 6$$

(Filter order known)

Using  $N = 6$

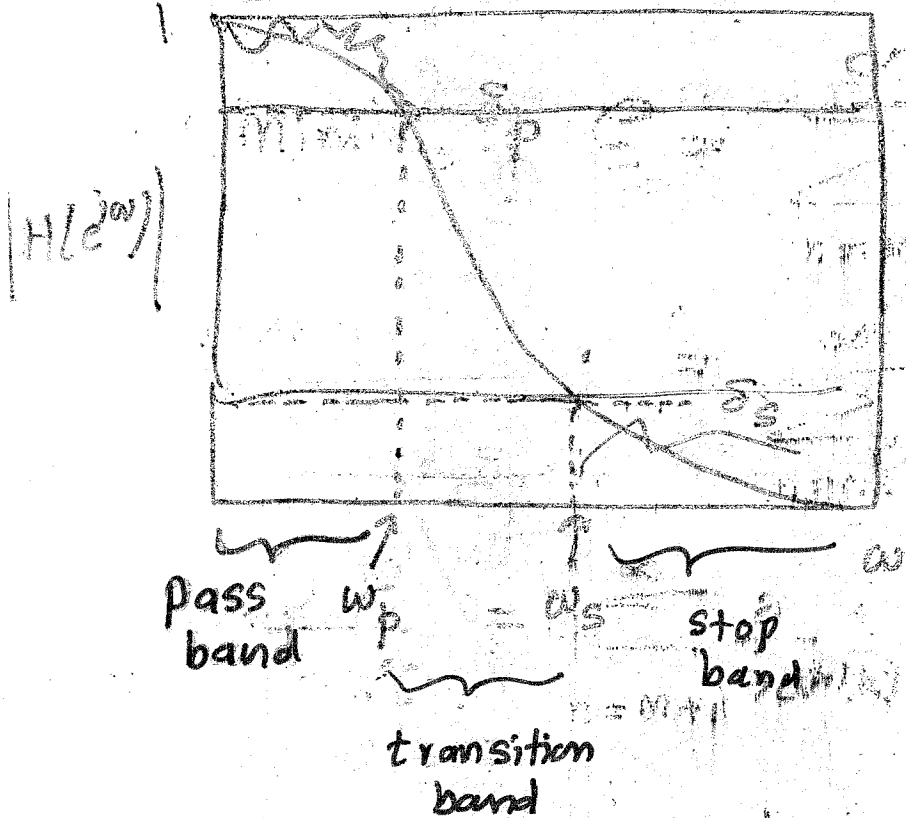
$$\Rightarrow \Omega_c = 176.2 \text{ rad/s } (28.0 \text{ Hz})$$

Reactive  $\Omega$  and  $\Omega_c$  before

$$\text{Cutoff } \omega_c = \frac{2\pi \times 28.0}{2\pi} \text{ in Hz}$$

# Filter Design Specifications in the Digital Domain

Domain



$\omega_p, \omega_s$  : Edge Frequencies of the transition band

$\omega_p$  : Edge frequency of passband

$\omega_s$  : Edge frequency of stopband

$\delta_p$  : minimum - passband gain

$\delta_s$  : maximum stopband gain

Analog Butterworth Filter with, Order = 6.

$$H(s) = \frac{1}{(s^2 + 0.5176\Omega_c s + \Omega_c^2)(s^2 + 1.0442\Omega_c s + \Omega_c^2)(s^2 + 1.9317\Omega_c s + \Omega_c^2)}$$

Impulse Invariance Design:

Method 2

1. Find all poles of  $H(s)$ .

\* Substitute  $\Omega_c$  in  $H(s)$

\* Get  $H(s)$  in the partial fraction form

\*  $s \rightarrow e^{sT}$

\* Recombine numerator/denominator

Bilinear Transform Technique:

\* Pre-warping:

$$\Omega_s = \frac{2}{T} \tan\left(\frac{\omega_s}{2}\right)$$

$$\Omega_p = \frac{2}{T} \tan\left(\frac{\omega_p}{2}\right)$$

Determine  $N$  and  $\Omega_c$  as before

Substitute  $s = \frac{2}{T} \frac{z-1}{z+1}$  in  $H(s)$

$$e(n) = \begin{cases} x(n) + \sum_{l=1}^N a(l)x(n-l) - b(n), & 0 \leq n \leq M \\ x(n) + \sum_{l=1}^N a(l)x(n-l), & n \geq M+1 \end{cases}$$

$$\text{Minimize: } \mathcal{E} = \sum_{n=M+1}^{\infty} |e(n)|^2$$

$$= \sum_{n=M+1}^{\infty} e^*(n)e(n)$$

$$\frac{\partial \mathcal{E}}{\partial a^*(k)} = \sum_{n=M+1}^{\infty} \frac{\partial}{\partial a^*(k)} (e^*(n)e(n)) = 0$$

$$1 \leq k \leq N$$

We know, for  $n \geq M+1$

$$e(n) = x(n) + \sum_{l=1}^N a(l)x(n-l)$$

$$e^*(n) = x^*(n) + \sum_{l=1}^N a^*(l)x^*(n-l)$$

$a(k)$  appears only once, when  $l=k$

$$\sum_{n=M+1}^{\infty} \frac{\partial}{\partial a^*(k)} (e^*(n)e(n)) = \sum_{n=M+1}^{\infty} e(n) \cdot x^*(n-k)$$

## Solving for $b[k]$

$$e[n] = x[n] + \sum_{l=1}^N a[l] \cdot x[n-l] - b[n]$$

Set  $e[k] = 0$ , for  $0 \leq k \leq M$

$$b[k] = x[k] + \sum_{l=1}^N a[l] \cdot x[k-l]$$

Solve for  $b[k]$

\* Less sensitive to noise than Padé

\*  $M, N$ : Not known "a-priori"

\* A method to design a filter

|| A method to model a signal

A method to estimate the power

1. Spectral analysis in a discrete-time signal

2. The signal  $x[n]$  is processed by a filter

# FIR Filter

Desired  $\{h[n]\}$  specified at discrete points  
 $h[n] = \begin{cases} \text{specified} & 0 \leq n \leq M-1 \\ 0 & n > M-1 \end{cases}$

\*  $M$  is  $M$  - shift (Causal Filter)  
 $H(z) = \sum_{k=0}^{M-1} h[k] z^{-k}$  FIR filter.

$$h[n] = \sum_{k=0}^{M-1-n} h[k] z^{-k}$$

## Desired Response

$$= h[0] + h[1]z^{-1} + \dots + h[M-1]z^{-(M-1)}$$

$$= z^{-(M-1)} (h[0]z^{M-1} + h[1]z^{M-2} + \dots + h[M-1])$$

zeros  $M-1$

$M-1$  Poles at  $z=0$

$H(e^{j\omega}) = h_d(\omega)$   $M=3$

Symmetric  $M-1$   $h[n] = h[M-1-n]$   
 Assume  $M$  is odd  $h[n] = h[M-1-n]$

$$h[M-1-n] = h[n]$$

$$h[n] = \frac{1}{M}$$



# FIR Filter Design by Frequency Sampling

- \* Desired <sup>response</sup> freq. specified at  $M$  different points.
- \* Use the  $M$ -specified values to get the  $M$ -point ( $M$ -tap) FIR filter.  
 $h[n], 0 \leq n \leq M-1$ .

## Desired Response

$$H_d(e^{j\omega}), \quad \omega = \omega_k = \frac{2\pi}{M} k$$

$$0 \leq k \leq \frac{M-1}{2} \quad (M\text{-odd})$$

$$0 \leq k \leq \frac{M}{2} - 1 \quad (M\text{-even})$$

$$H(e^{j\omega_k}) \triangleq H_d(e^{j\omega_k})$$

$$= \sum_{n=0}^{M-1} h[n] \cdot e^{-j\frac{2\pi}{M} k \cdot n}$$

$$h[n] = \frac{1}{M} \sum_{k=0}^{M-1} H_d(e^{j\omega_k}) e^{j\frac{2\pi}{M} k \cdot n}$$