

Digital Signal Processing and Filter Design using Scilab

Iman Mukherjee

Department of Electrical Engineering, IIT Bombay

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Outline

- 1 Basic signal processing tools
 - Discrete Fourier Transform
 - Fast Fourier Transform
 - Convolution
 - Plotting
 - Group Delay
 - Aliasing
- 2 Filter Design
 - Non-Recipe Based
 - Recipe Based
 - An Example Application

DFT

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- Faster way - $\text{fft} \dots$

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- $x = \text{fft}(a, -1, \text{dim}, \text{incr})$ - multidimensional fft
- $\text{fftshift}(\text{abs}(y))$ - rearranges the fft output, moving the zero frequency to the center of the spectrum

Exercise 1

The convol Command

- With the convol command

The convol Command

- With the convol command
- Without the convol command (multiplying in the frequency domain)

Exercise 2

Bode and Pole-Zero Plots

- Demo Pole-Zero Plot

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- Demo Pole-Zero Plot
- Demo Bode Plot

Group Delay

Rate of change of Phase w.r.t Frequency

$$\tau_g = \frac{d\phi}{d\omega}$$

Where,

τ_g is the Group Delay

ϕ is for phase delay

ω is for frequency

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- Scilab commands to remember -
- $t = \text{soundsec}(n [,rate])$ - generates n sampled seconds of time parameter
- $v = \text{linspace}(x1,x2 [,n])$ - linearly spaced vector
- $\text{mtlb_hold}(flag)$

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- `ss2tf`, `tf2ss`, `dscr` - State-Space $j\text{-}\omega$ Transfer Function, Discretizing Continuous Systems
- `r = repfreq(Sys,frq)` - Frequency response
- `playsnd(data)`
- `filtered_output = flts(input,filter)`

Analog filter family prototypes

- Butterworth

Analog filter family prototypes

- Butterworth
- Chebyshev

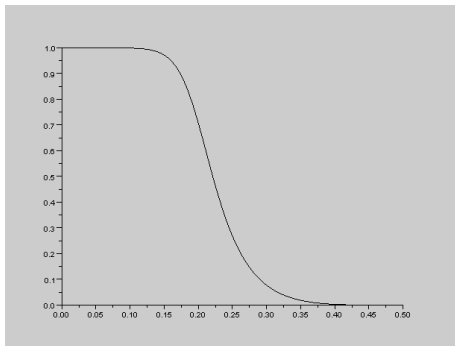
Analog filter family prototypes

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- Chebyshev
- Inverse Chebyshev

Analog filter family prototypes

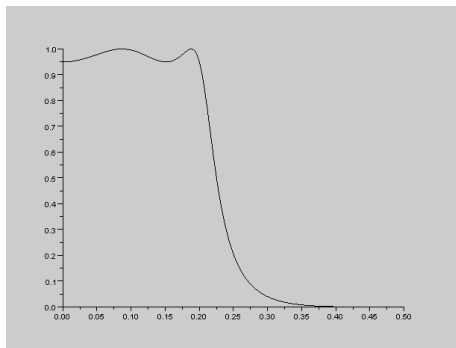
- Butterworth
- Chebyshev
- Inverse Chebyshev
- Elliptic/Chauer

Butterworth



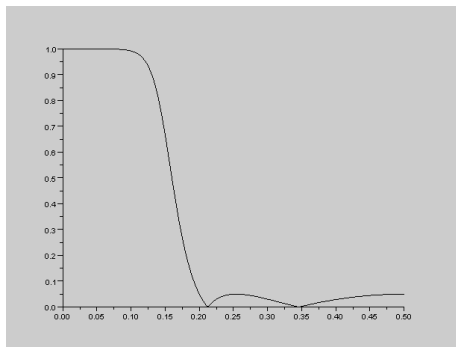
- Passband: Monotonic
- Stopband: Monotonic
- No ripples
- Wide Transition, slow Roll-off

Chebyshev



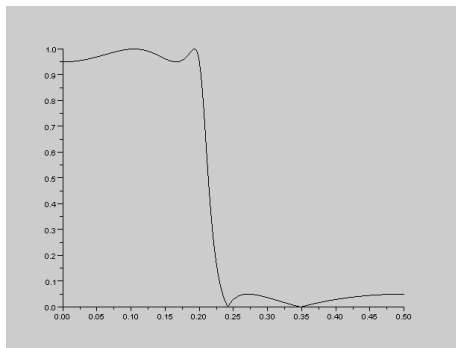
- Passband: Equiripple
- Stopband: Monotonic
- Only ripples in Passband
- Lesser Transition width, slow Roll-off at high frequencies

Inverse Chebyshev



- Passband: Monotonic
- Stopband: Equiripple
- Only ripples in Stopband
- Lesser Transition width, slow Roll-off at low frequencies

Elliptic



- Passband: Equiripple
- Stopband: Equiripple
- Ripples in both - Passband and Stopband
- Least Transition, sharp and fast Roll-off

An example

Steps :

- Decide over the filter specifications (type, PB and SB cut-offs, ripples etc.)

Let us design a Low-Pass Chebyshev Filter

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- Implement Analog Transfer function

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- Decide over the filter specifications (type, PB and SB cut-offs, ripples etc.)
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- Perform bilinear transformation

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- Calculate the filter coefficients
- Implement Analog Transfer function
- Perform bilinear transformation
- Convert to the needed filter type

Let us design a Low-Pass Chebyshev Filter

Scilab Commands in Filter Design

- iir - $[hz]=iir(n,ftype,fdesign,frq,delta)$
- eqiir -
 $[cells, fact, zzeros, zpoles]=eqiir(ftype, approx, om, deltap, deltas)$
- eqfir - $[hn]=eqfir(nf, bedge, des, wate)$
- wfir - $[wft, wfm, fr]=wfir(ftype, forder, cfreq, wtype, fpar)$
- analpf - $[hs, pols, zers, gain]=analpf(n, fdesign, rp, omega)$
- trans - $hzt=trans(pd, zd, gd, tr_type, frq)$

Exercise 3

Audio effects

- Flanging - A sound file seems like riding on a wave
- Echo - Delayed and added back
- Equalizer - Different Frequency Bands
 - Low
 - Mid
 - High

Thank you !

-  Sanjit K. Mitra, “Digital Signal Processing, A computer based approach”, *Tata McGraw-Hill Edition 1998.*
-  Steven W. Smith, “<http://www.dspguide.com/> ”
-  Carey Bunks, Francois Delebecque, Georges Le Vey and Serge Steer, “Scilab Group INRIA Meta2 Project/ENPC Cergrene ”, *Signal Processing with Scilab.*