Administrative stuff

- Matlab4 & Matlab5 grades will be posted tonight
- Brief discussion of Matlab6

- Reminder of assignments for rest of term:
  - HW due next Monday
  - Matlab6 due next Monday
  - Exam 3 next Wednesday
  - Matlab7 due Dec 20

EE-125: Digital Signal Processing

Lecture: IIR wrapup, and Sampling rate conversion

Professor Tracey

Outline

- Brief wrap-up of IIRs from last time
- Changing the sampling rate (P&M 6.5, more detail in Chapter 11)
  - Motivation
  - Reducing sampling rate by an integer (downsampling)
  - Increasing sampling rate by an integer (upsampling)
  - Changing rate by a ratio of integers

Wrapping up IIR

- Let’s briefly review different commonly used IIR filter types

- I’ll have a very easy multiple choice question about this on exam 3

Common IIR Filter Types (P&M 10.3.4) (KNOW THIS FOR EXAM!!!)

Type	Characteristic
Butterworth	Monotonic, smooth response; *maximally flat* at F=0, Infinity (N-1 zero derivatives)
Chebyshev Type I	Minimized absolute difference in **passband**; passband ripple, monotonic in stopband
Chebyshev Type II	Minimized absolute difference in **stopband**; via stopband ripple, monotonic in passband
Elliptical	Lowest filter order for given transition band; equiripple in both stop and pass band; optimized for magnitude (phase may be distorted)
Bessel	Linear phase within passband; at expense of magnitude (rolloff is slower than others)

Butterworth

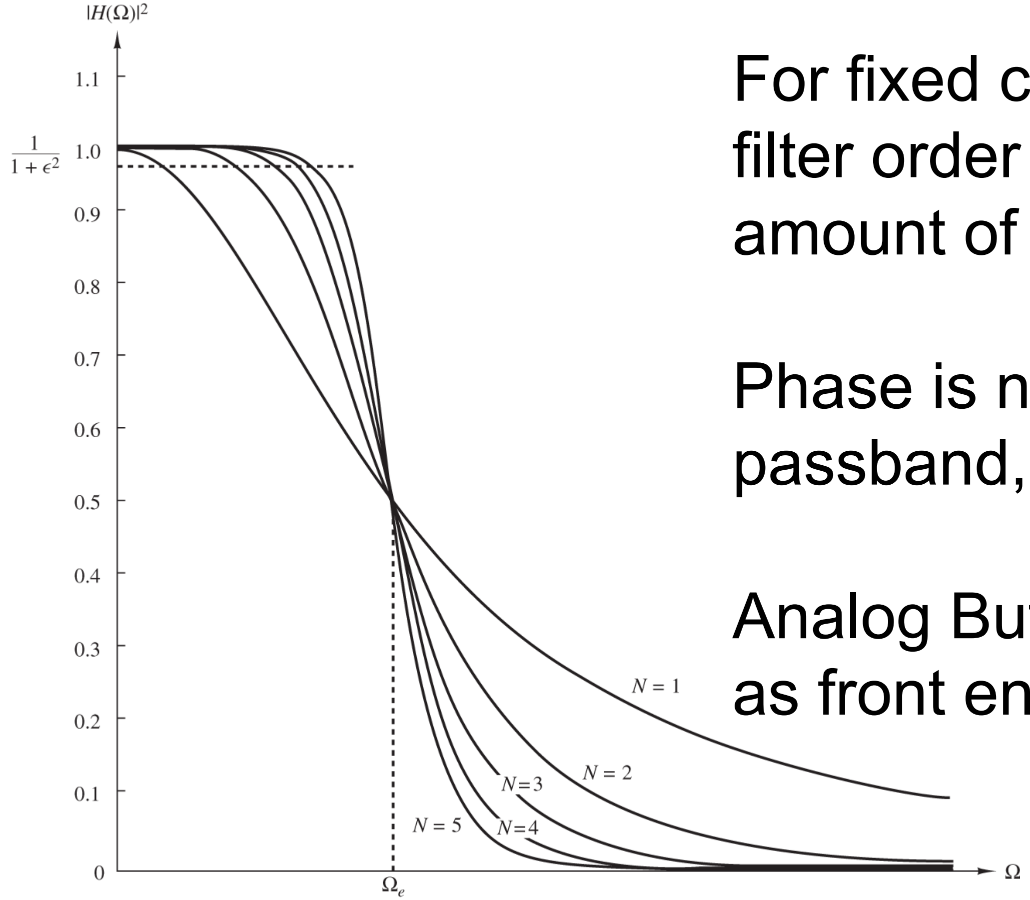


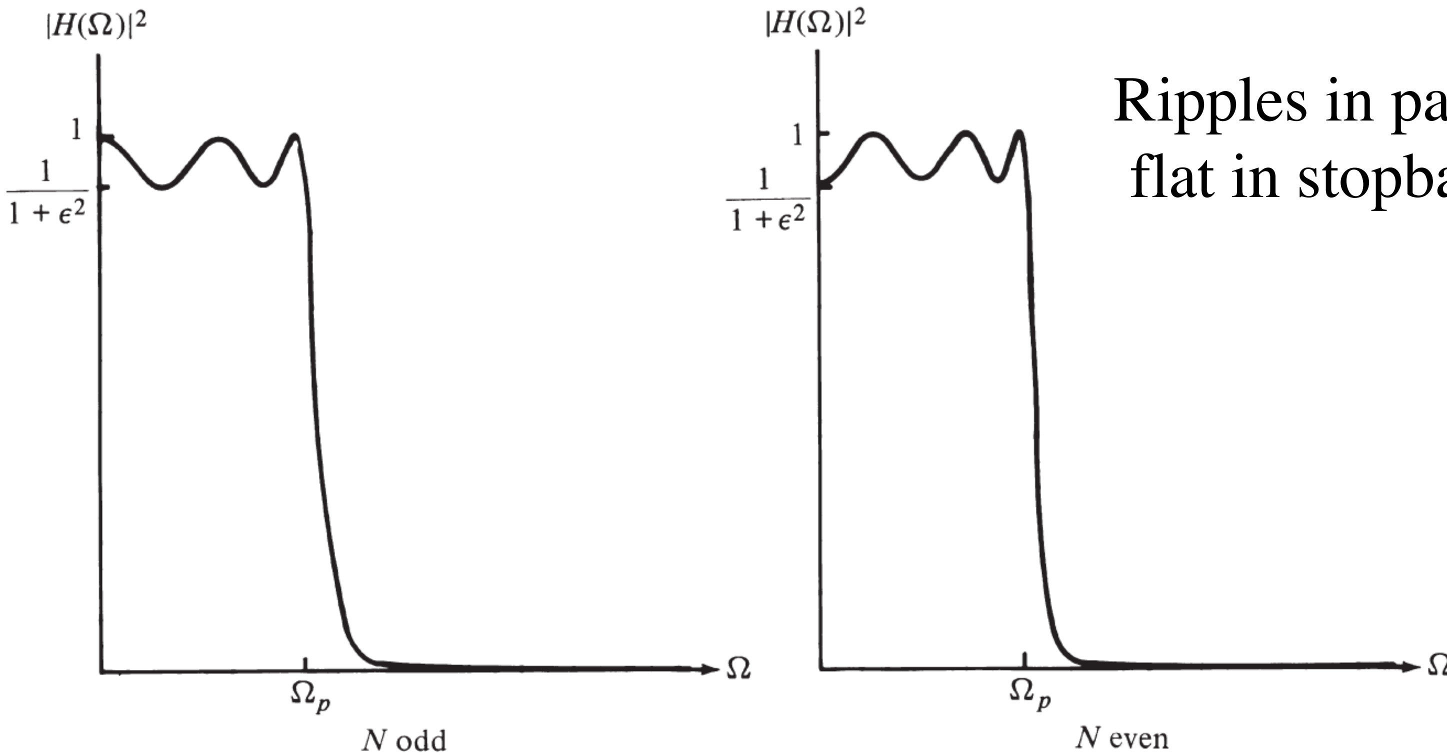
Figure 10.3.10 Frequency response of Butterworth filters.

For fixed cutoff frequency, filter order N determines amount of attenuation

Phase is not linear in passband, but is not too bad..

Analog Butterworths often used as front end in biomed systems

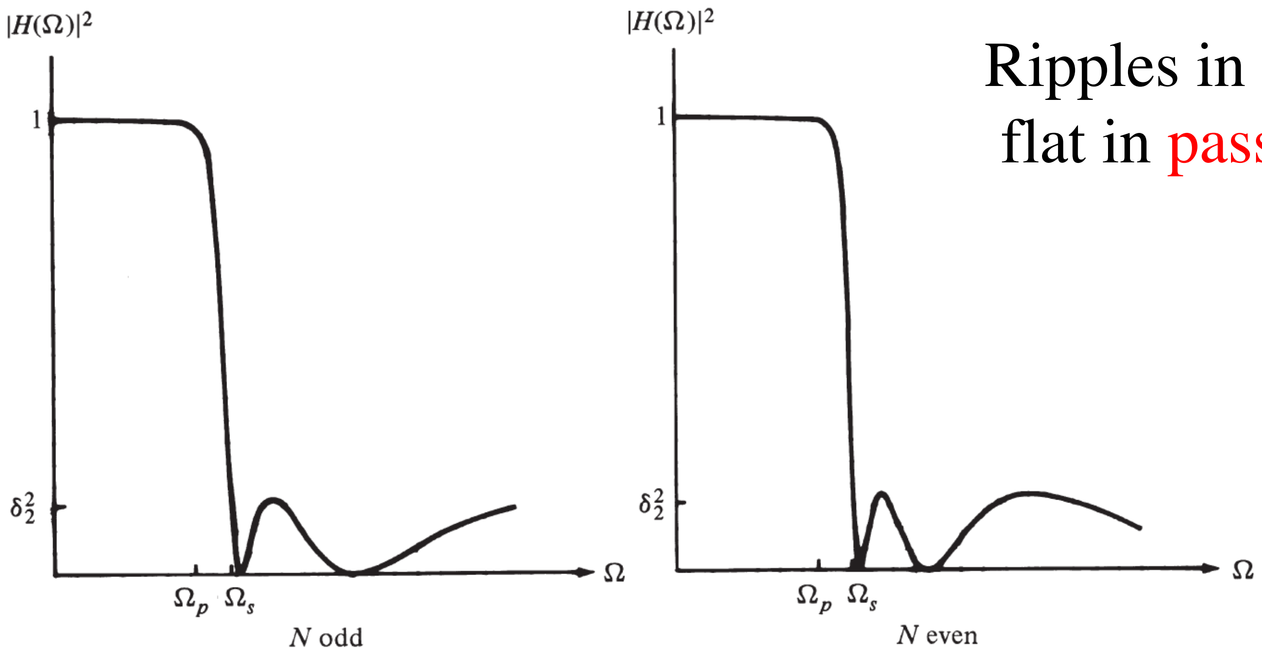
Chebyshev, type I – poles only



Ripples in passband, flat in stopband

Figure 10.3.11 Type I Chebyshev filter characteristic.

Chebyshev, type II – poles and zeros



Ripples in stopband, flat in passband

Figure 10.3.13 Type II Chebyshev filters.

Elliptical filters – optimally fast transition, ripple in both stop and pass bands

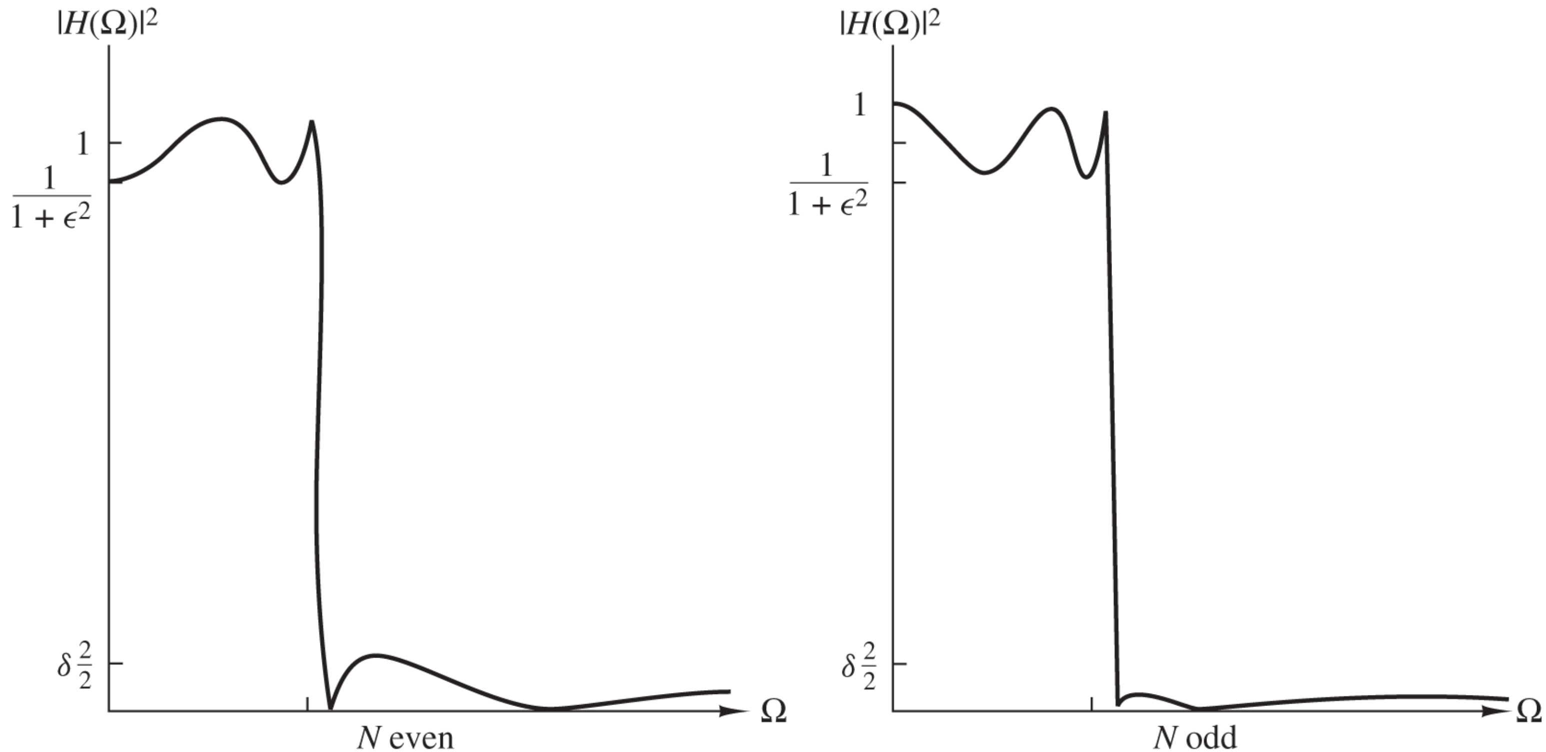


Figure 10.3.14 Magnitude-squared frequency characteristics of elliptic filters.

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Bessel – linear phase in passband

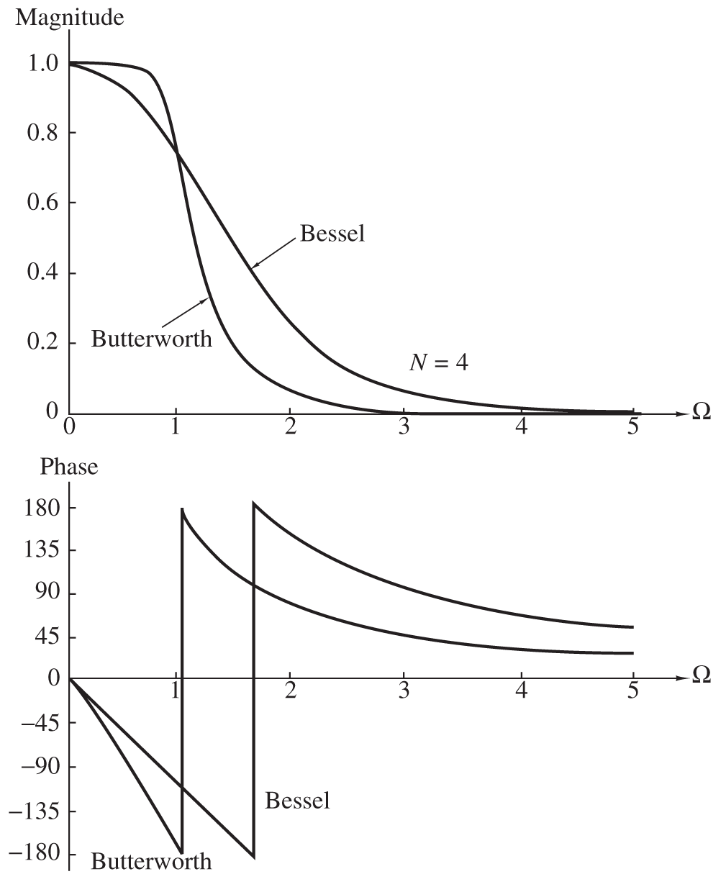

Figure 10.3.15 Magnitude and phase responses of Bessel and Butterworth filters of order $N = 4$.

Cost of linear phase:
Slower rolloff in amplitude

Interesting if you are a lot about phase but can't do filtfilt 'trick'

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Some final points on IIR…

- Mapping above let us translate existing analog designs into digital filters. What if we want to start from scratch?
- If we know the desired impulse response h(n), we can come up with a pole/zero (or all-pole model that fits it).
    - Example: Prony's method models data as sum of damped exponentials. Can be useful in modeling data (tires, ocean modes)
- If we have a desired frequency response, we can use the Yule-Walker method. Comes up with IIR that matches desired; may not be optimal
- An interesting article on the history and stability issues in IIR: "The rise and fall of recursive digital filters," Charles M. Rader, IEEE Signal Processing magazine, 2006

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Frequency domain look at downsampling (DS): no prefiltering / no aliasing

Analog (CT) signal

Ω= 2 pi F, F in Hz

Analog (CT) signal x pulse train (CTFT)

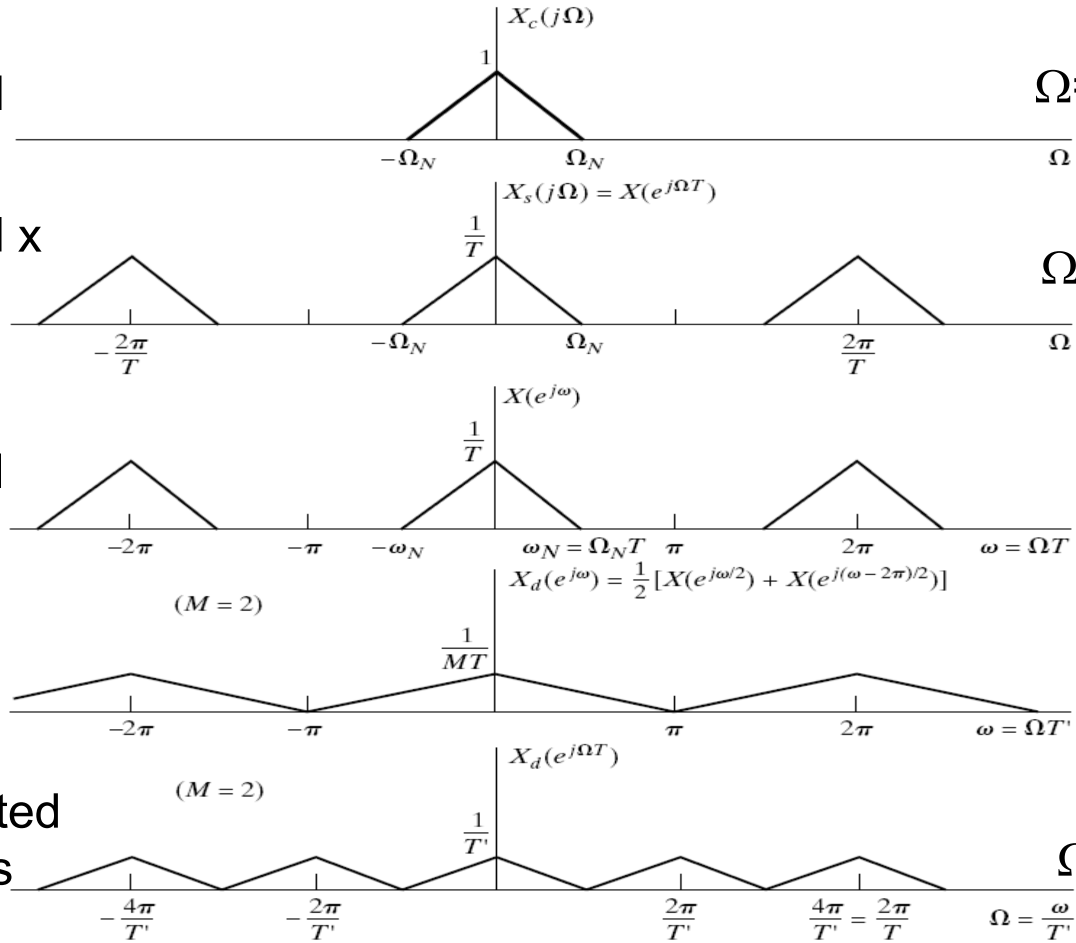
Ω= 2 pi F, F in Hz

Sampled DT signal

DS, no prefilter

Equivalent converted back to continuous frequency

Ω= 2 pi F, F in Hz



Frequency domain look at downsampling (DS): with (possible)aliasing

Analog (CT) signal

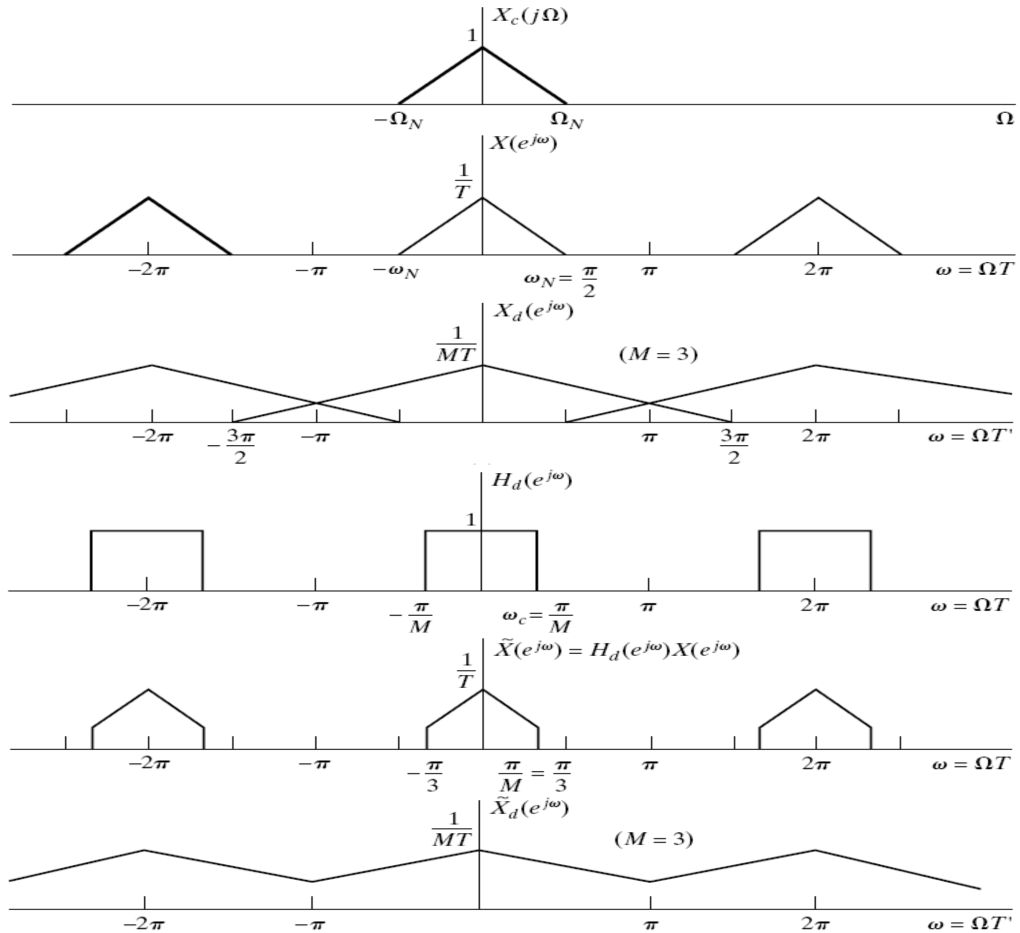
Sampled DT signal

DS, no prefilter

Ideal prefilter shape

Effect of prefilter (before DS)

DS, after prefilter



Tufts

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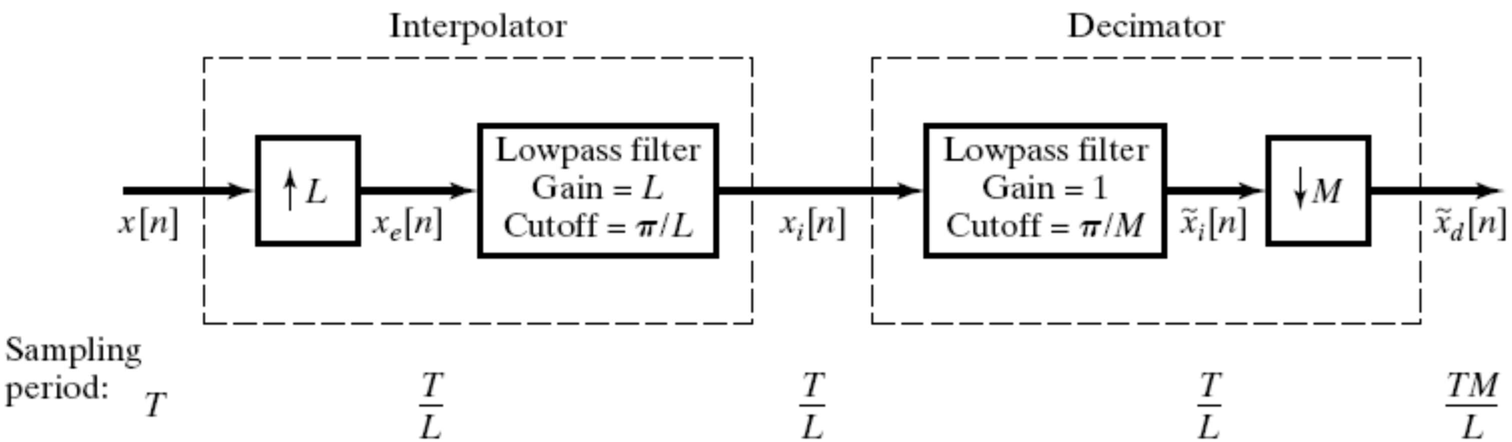
Changing Sampling rate by fraction (L/M) – matlab ‘resample’ function

We could go up by L, down by M…..



But easier is to combine the filters