

# DIGITAL SIGNAL PROCESSING: COSC390

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## UNIT 2.1 OUTLINE

1. **Introduction:** Types of filters (reading: ch. 3, ch. 5)
  - Butterworth
  - Bessel
  - Chebyshev
2. LTI systems and their properties (reading: ch. 5)
3. Convolution (reading: ch. 7)
  - Implementation with FFT
  - Impulse and step response

Future lectures will cover:

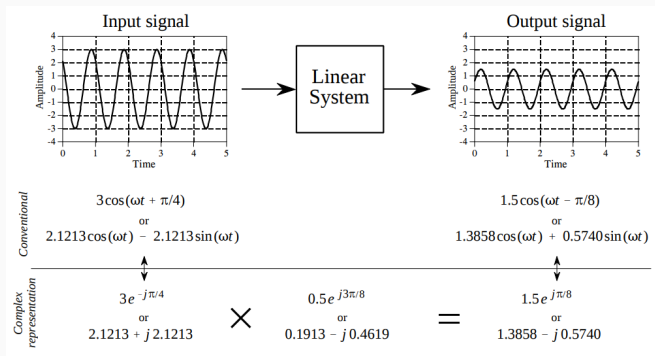
1. SNR of filtered signals: SNR
2. Common filter kernels (moving average, windows)
3. Recursive filters
4. FIR and IIR definitions

## INTRODUCTION: TYPES OF FILTERS

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# INTRODUCTION: TYPES OF FILTERS

Chapter 3 lists three types of anti-aliasing filters: Butterworth, Bessel, and Chebyshev. Filters are examples of linear, time-invariant (LTI) devices



**Figure 1:** A linear, time-invariant system has special properties encapsulated by the *convolution* operation.

# INTRODUCTION: TYPES OF FILTERS

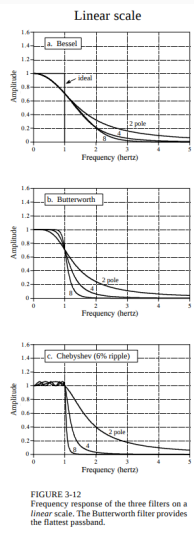
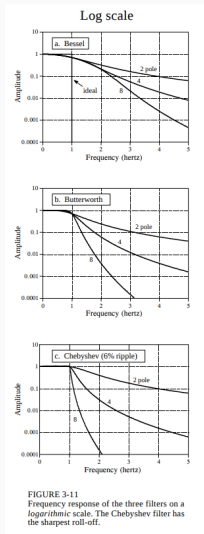


Figure 2: Comparison of transfer function magnitudes.

# INTRODUCTION: TYPES OF FILTERS

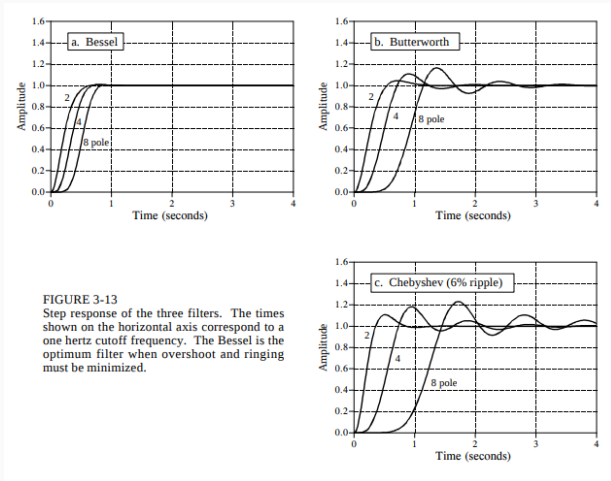


FIGURE 3-13  
Step response of the three filters. The times shown on the horizontal axis correspond to a one hertz cutoff frequency. The Bessel is the optimum filter when overshoot and ringing must be minimized.

Figure 3: Comparison of transfer function step responses.

# INTRODUCTION: TYPES OF FILTERS

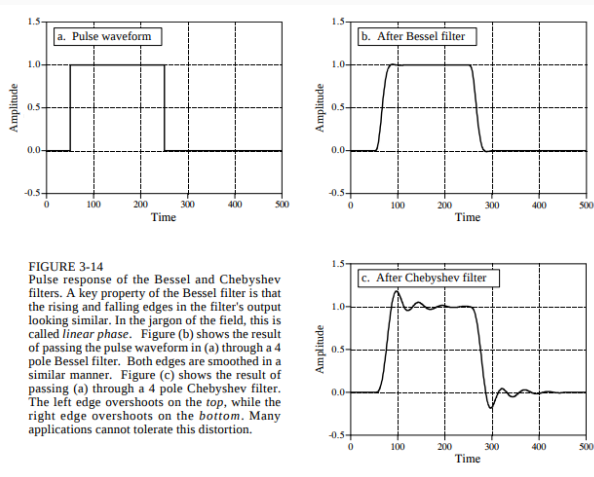


FIGURE 3-14

Pulse response of the Bessel and Chebyshev filters. A key property of the Bessel filter is that the rising and falling edges in the filter's output looking similar. In the jargon of the field, this is called *linear phase*. Figure (b) shows the result of passing the pulse waveform in (a) through a 4 pole Bessel filter. Both edges are smoothed in a similar manner. Figure (c) shows the result of passing (a) through a 4 pole Chebyshev filter. The left edge overshoots on the *top*, while the right edge overshoots on the *bottom*. Many applications cannot tolerate this distortion.

Figure 4: Comparison of transfer function pulse responses.

The single-pole Butterworth transfer functions are derived from the single-pole RC filter circuit:

$$H_{LP}(\omega) = \frac{\omega_0}{\omega_0 + j\omega} \quad (1)$$

$$H_{HP}(\omega) = \frac{\omega}{\omega - j\omega_0} \quad (2)$$

- What frequency causes a singularity in the transfer functions?
- What is the phase and group delay of this filter?



General expression for the transfer function of Butterworth filter (low-pass):

$$|H_{LP}(\omega)| = \frac{G_0}{\sqrt{1 + \left(\frac{j\omega}{\omega_0}\right)^{2n}}} \quad (3)$$

The integer  $n$  is the number of poles.  $G_0$  is the *gain*, and  $\omega_0$  is the corner or cutoff frequency.

- Can we plot the poles in the complex plane?
- What is the phase and group delay of this filter?

## INTRODUCTION: TYPES OF FILTERS

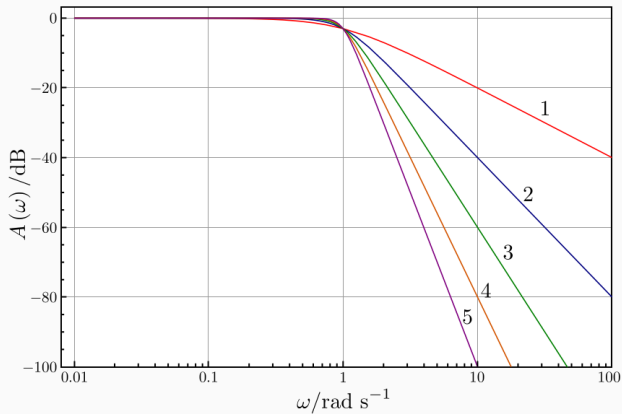


Figure 5: Gain of butterworth filters with  $n$  poles.

## INTRODUCTION: TYPES OF FILTERS

The n-th order low-pass Bessel filter transfer function is a ratio of reverse Bessel polynomials:

$$H(\omega) = \frac{\theta_n(0)}{\theta_n(j\omega/\omega_0)} \quad (4)$$

where the reverse Bessel polynomials  $\theta_n(x)$  are given by

$$\theta_n(x) = \sum_{k=0}^n \frac{(n+k)!}{(n-k)!k!} \frac{x^{n-k}}{2^k} \quad (5)$$

- What is  $\theta_3$ ?
- How do we turn this into a high-pass filter?
- What are the pole locations of the 3rd-order Bessel filter?

## INTRODUCTION: TYPES OF FILTERS

The n-th order low-pass Chebyshev filter transfer function is

$$|H(\omega)| = \frac{1}{\sqrt{1 + \epsilon^2 T_n^2(\omega/\omega_0)}} \quad (6)$$

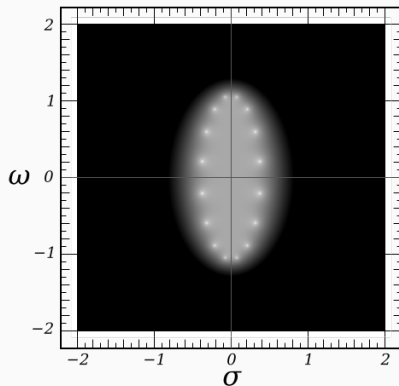
where the Chebyshev polynomials  $T_n(x)$  are given by

$$T_n(x) = \cos(n \cos^{-1}(x)) \quad |x| < 1 \quad (7)$$

$$T_n(x) = \cosh(n \cosh^{-1}(x)) \quad x \geq 1 \quad (8)$$

$$T_n(x) = (-1)^n \cosh(n \cosh^{-1}(-x)) \quad x \leq -1 \quad (9)$$

- Can we plot  $T_2(x)$  in Octave?
- Pole locations are interesting (next slide).



**Figure 6:** Eight-pole Chebyshev filter in the complex plane. The poles form an ellipse, due to the trigonometric nature of the definition of Chebyshev polynomials.

## INTRODUCTION: TYPES OF FILTERS

In the octave signal package, we can access the transfer functions of these filters:

```
pkg load signal;  
[b1,a1] = butter(n,omega); %(e.g. include "high")  
[b2,a2] = besself(n,omega);  
[b3,a3] = cheby1(n,rp,omega); %rp pass-band ripple  
x = (...); %data  
y = filter(b1,a1,x);
```

Use **help** function on these for more information. The **filter** function is using the pole-zero information stored in the coefficients *a* and *b* to apply the transfer function to the data (more later).

## INTRODUCTION: TYPES OF FILTERS, OCTAVE PROGRAMMING EXAMPLE

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## INTRODUCTION: TYPES OF FILTERS

If you have the **signal** package, you can specify a n-th order butterworth filter with **butter** as above. Otherwise, you can write a small function using Eq. 3 for the digital response.

1. Create a white-noise sample (**randn**) of  $10^4$  samples. You can also specify a time vector of the same size.
2. Plot the spectrum of this noise using **fft**<sup>1</sup>).
3. **Filter** the noise with a 2nd-order low-pass butterworth filter, at a cutoff frequency of  $0.2 f_s$ . Either use the **filter** function and **fft**, or multiply in the Fourier domain.
4. **Filter** the noise with a 2nd-order high-pass butterworth filter, at a cutoff frequency of  $0.4 f_s$ .
5. Now do (3) then (4), and plot the spectrum. Repeat for (4) then (3). Do you see the same spectrum?

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<sup>1</sup>See the **Aliasing.m** script in Unit 1

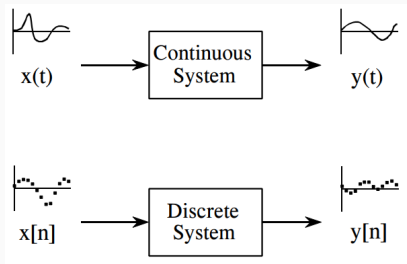


## LTI SYSTEMS

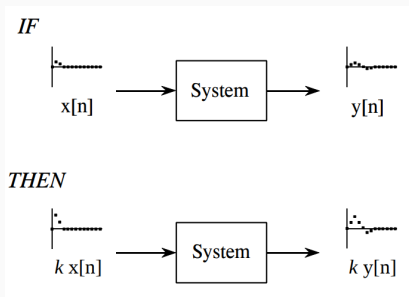
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**Filters are LTI systems.** Let's review the properties of LTI systems:

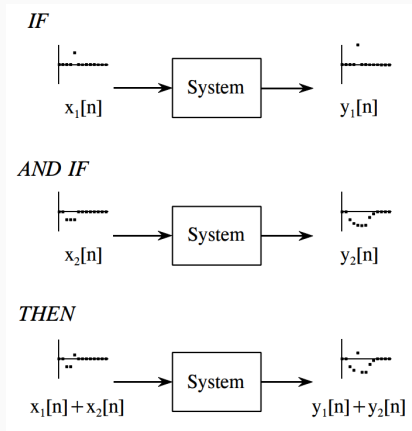
1. Continuous vs. discrete
2. Scaling property
3. Distributive property
4. Time-invariance (also causality)
5. Commutative property
6. Combination of properties



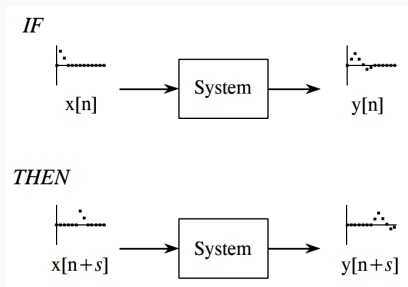
**Figure 7:** A continuous LTI system response to continuous data, and a discrete LTI system response to discrete data. All properties should hold both cases.



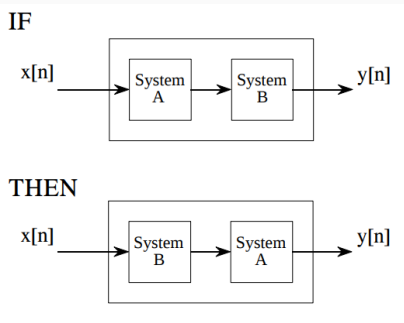
**Figure 8:** Scaling: If the data is scaled by a real constant, the output should be scaled by a real constant.



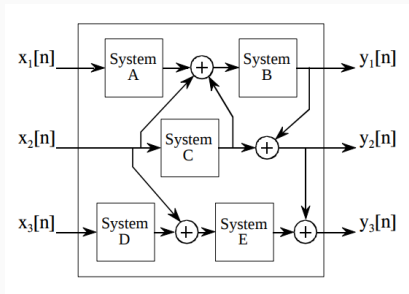
**Figure 9:** Distributive property: the LTI system should respond to each signal separately.



**Figure 10:** Time-invariance: the LTI system should respond when the signal arrives, independent of global time. Also, the filter should not respond *before* the signal arrives (causality).



**Figure 11:** Commutative property: the LTI system should not depend on the existence of previous systems.



**Figure 12:** Combination of properties: an LTI system may be built from a combination of LTI systems.

Suppose  $A(x)$ ,  $B(x)$ ,  $C(x)$ ,  $D(x)$ , and  $E(x)$  represent operators of LTI systems. What is the formula for a)  $y_1$ , b)  $y_2$  and c)  $y_3$ ? What is the the Fourier transform of the simplest output?



The **convolution** operator has all of the necessary properties of the LTI system operator. The convolution of data streams  $x_1$  and  $x_2$  can be implemented with the following basic algorithm:

```
x_1 = (...); %data1
x_2 = (...); %data2
y_1 = fft(x_1);
y_2 = fft(x_2);
Z_1 = y_1.*y_2;
result = real(ifft(Z_1));
```

Or, in one line:

```
result = real(ifft(fft(x_1).*fft(x_2))));
```

# LTI SYSTEMS: PROGRAMMING WITH OCTAVE

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Obtain the **impulse response** of the first-order low-pass butterworth filter. The impulse response is given by the identity:

$$x[n] \circ \delta[n] = x[n] \quad (10)$$

Recall the transfer function  $h(\omega)$  for the first-order butterworth filter. Take the inverse Fourier transform of this function, to obtain the impulse response  $h(t)$ . What are the units of  $h(t)$ ?

1. Plot this as a function of time in Octave.
2. Now define the delta function vector (impulse vector):

```
d = zeros(size(t));  
d(1) = 1;
```

3. Convolve  $d$  with  $h(t)$ . What do you see?

The impulse response of a filter is the function with which the signal is convolved:

$$x[n] \circ f[n] = y[n] \quad (11)$$

1. Plot this as a function of time in Octave.
2. Now define the delta function vector (impulse vector):

```
d = zeros(size(t));  
d(1) = 1;
```

3. Convolve  $d$  with  $h(t)$ . What do you see?

You can also use the built-in Octave function **conv** for convolution, but pay attention to sizes:

```
x_1 = (...); %data
x_2 = (...); %data
y = conv(x_1,x_2);
size(y)
size(x_1)
size(x_2)
```

Special case of **conv**:

```
y = conv(x_1,x_2,"same"); %default; "full"
size(y)
size(x_1)
size(x_2)
```

Convolve a sine-wave of a given frequency with the response of a first-order low-pass butterworth filter, and plot the result. Does the amplitude make sense? *Hint: don't forget factors of  $\Delta t$ !*

## SPECIAL TOPIC: IMPULSE RESPONSE OF RF ANTENNAS

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A paper on RF antenna response for UHE neutrino research:  
<https://doi.org/10.1016/j.astropartphys.2014.09.002>