IIR FILTERS (INFINITE IMPULSE RESPONSE FILTERS)

• IIR filters are implemented using the recursion

The General IIR Filter

The general form for the IIR filter is:

$$y(n) = \sum_{k=0}^{N} a_k x(n-k) - \sum_{j=1}^{M} b_j y(n-j)$$

= $a_0 x(n) + a_1 x(n-1) + a_2 x(n-2) + \dots + a_N x(n-N)$
- $b_1 y(n-1) - b_2 y(n-2) - \dots - b_M y(n-M)$

Recursion

$$y_n = x_n + y_n - 1$$

$$y_0 = x_0 + y_{-1}$$

$$y_1 = x_1 + y_0$$

$$\mathbf{y}_2 = \mathbf{x}_2 + \mathbf{y}_1$$

... and so on.

$$Y10 = x10 + y9$$

$$Y10 = x10 + x9 + x8 + x7 + x6 + x5 + x4 + x3 + x2 + x1 + x0$$

Advantages and disvantages IIR

- Advantages of recursive filter
 - Generally require a much lower order filter, they are faster than FIR filters
- Disadvantages
 - The feedback component will also feed back the noise from the original signal.

Some IIR Filter

Chebyshev

Butterworth

CHEBYSHEV FILTER

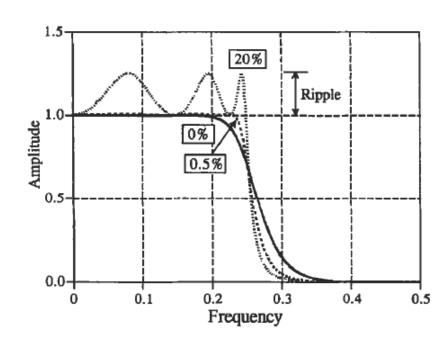
Chebyshev filters

 Are analog and digital filters that use a mathematical strategy for achieving a faster roll off by allowing ripple in the frequency response (they use Chebyshev polynomials).

Ripple vs. Roll-off

FIGURE 20-1

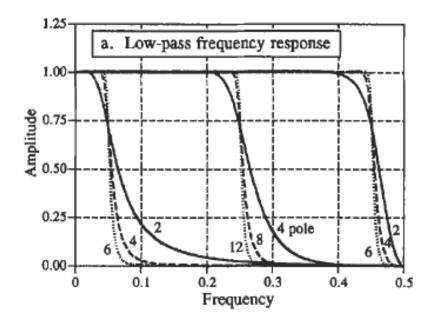
The Chebyshev response. Chebyshev filters achieve a faster roll-off by allowing ripple in the passband. When the ripple is set to 0%, it is called a maximally flat or Butterworth filter. Consider using a ripple of 0.5% in your designs; this passband unflatness is so small that it cannot be seen in this graph, but the roll-off is much faster than the Butterworth.

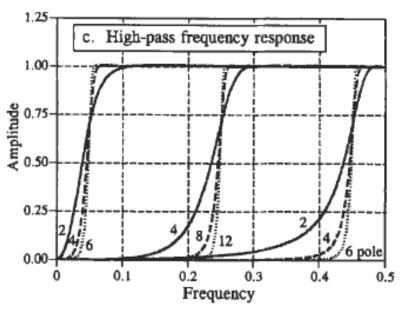


A ripple of 0.5% is often a good choice for digital filters.

Chebyshev filter design

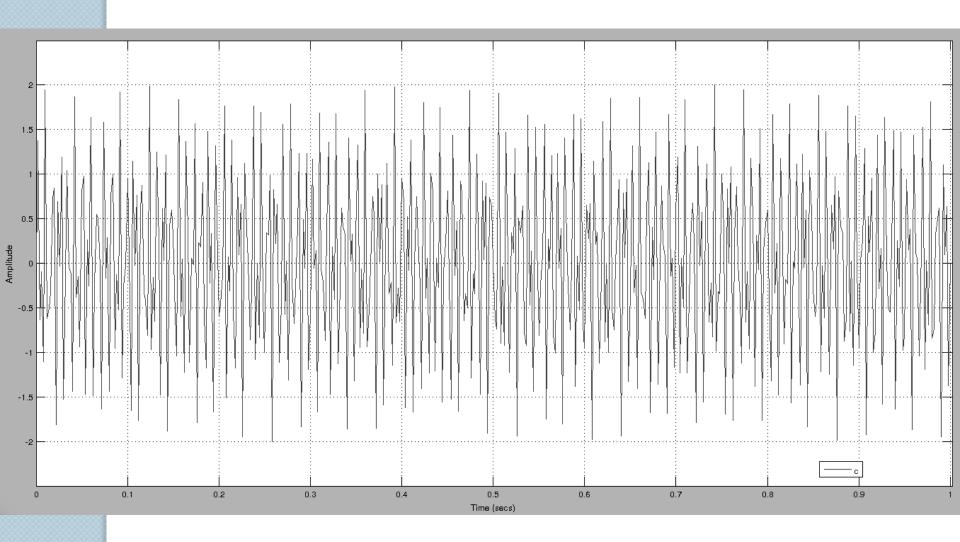
- High-pass or low-pass response
- Cutoff frequency
- The percent ripple in the passband
- Number of poles

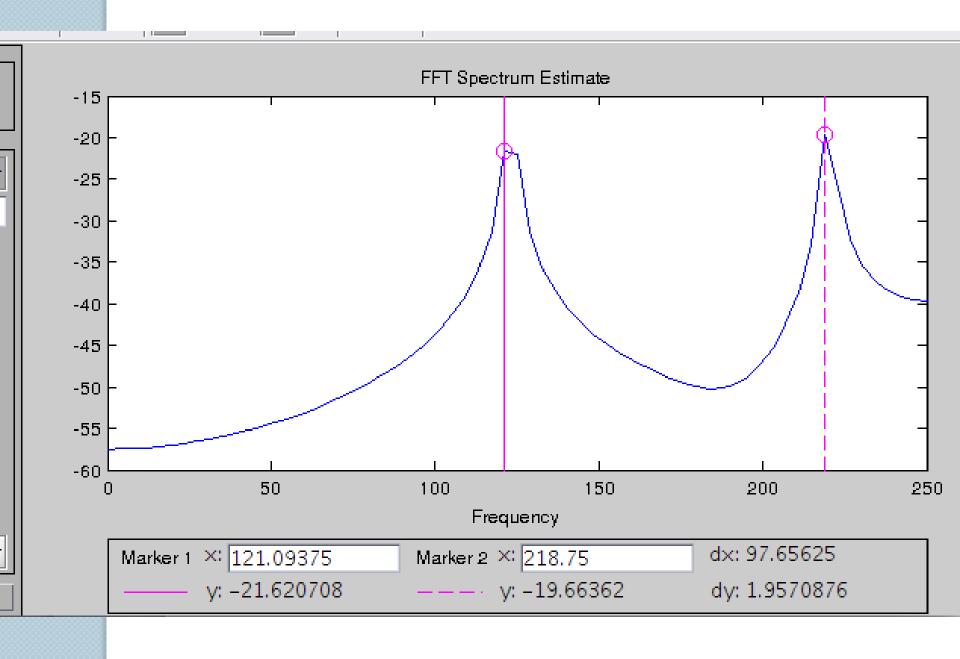


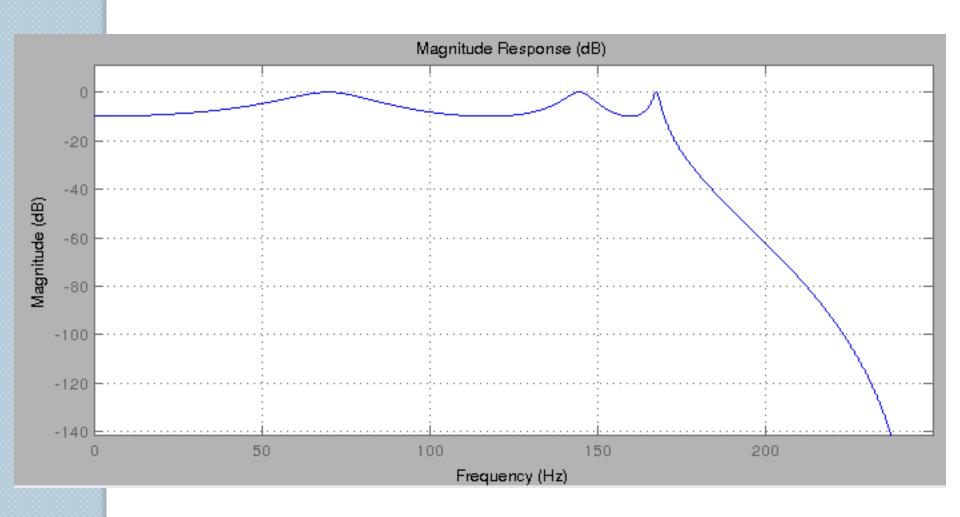


Exercise

```
sampleRate = 500;
   fin = 1;
   t = 0:(1/sampleRate):1;
   a = \sin(2*pi*123*t);
   b = \sin(2*pi*220*t);
   c = a + b;
   %plot(t, alphaRythm, '-', t, betaRythm, '-');
   plot(t, c, '-')
   length(a)
   %importar los datos a sptool con una velocidad de muestreo de 500
   %----diseño del filtro----
   %frec de corte
   frec_corte = 170
   %m□×ima frec a leer
   freq_max = sampleRate/2;
   %frec normalizada
   frec_corte_norm = frec_corte/freq_max;
   %orden del filtro
   orden = 6:
   %rizo del bandpass (decibeles)
   bpr = 10:
   %coeficientes del filtro
   [b a] = cheby1(orden,bpr,frec_corte_norm)
   %realizar filtro
   filtrada = filter(b, a, c);
b =
   0.0228 0.1367 0.3418 0.4557 0.3418 0.1367 0.0228
a =
   1.0000 0.4048 1.4812 -0.0858 1.1157 0.0543 0.6409
```

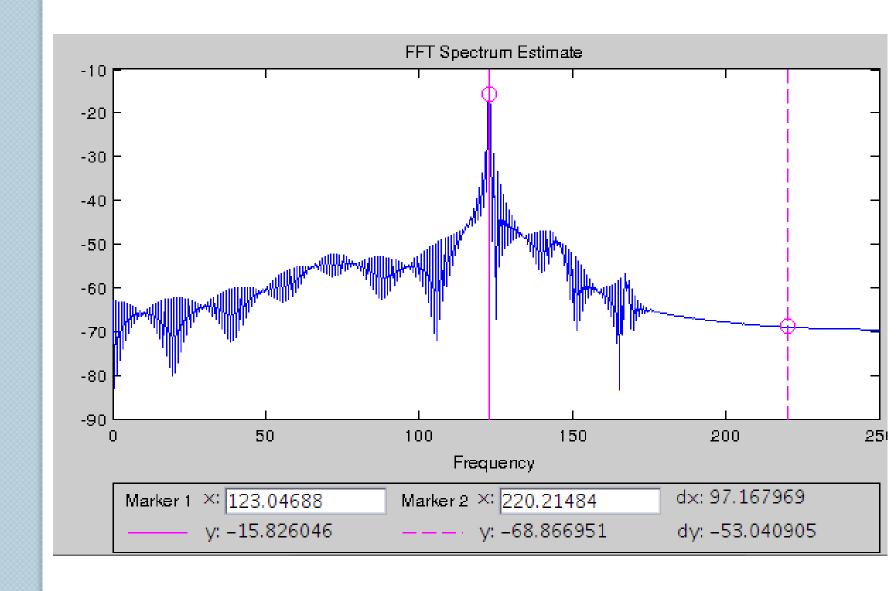






Coeff

```
Numerator:
0.022784060585876057813958794895370374434
 . 13.67043.63515256333005964961557765491307
0.341760908788140860270488019523327238858
0.455681211717521128523600282278493978083
 .341760908788140860270488019523327238858
  136704363515256333005964961557765491307
0.022784060585876057813958794895370374434
Denominator:
 0.404825131846837660276605674880556762218
   481179241824444936526106175733730196953
   0857689643218016994907770822464954108
   115717219196490139765387539227958768606
 0.054267529355076227215448625429417006671
 0.640949493211831144989787389931734651327
```



" «... ONE TOOL TO RULE THEM ALL»

fdatool

