



# Non-Recursive Filters

FIR Filters

All Zero Filters

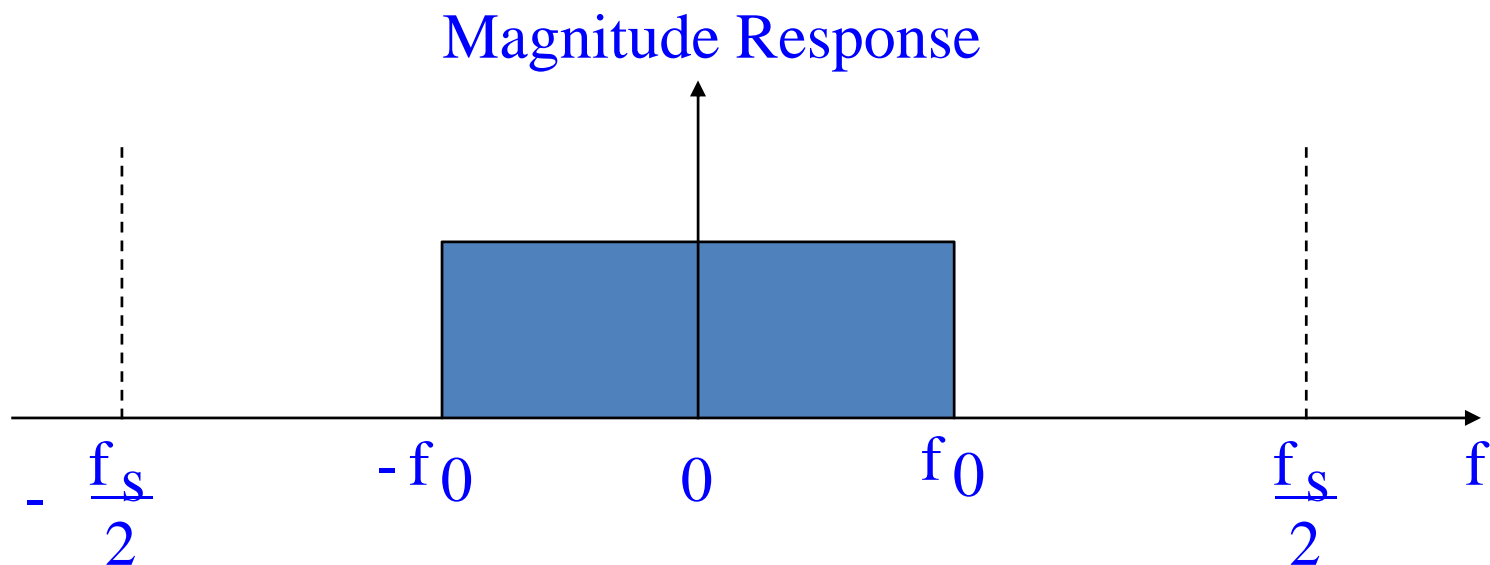
Moving Average Filters



## Filter Design

- Let us examine a low pass non-recursive filter with the following characteristics
  - $N = 9$  samples
  - $f_s = 2000$  Hz (sample freq)
  - $f_0 = 200$  Hz (cutoff freq)

Note: assume time is continuous for the moment





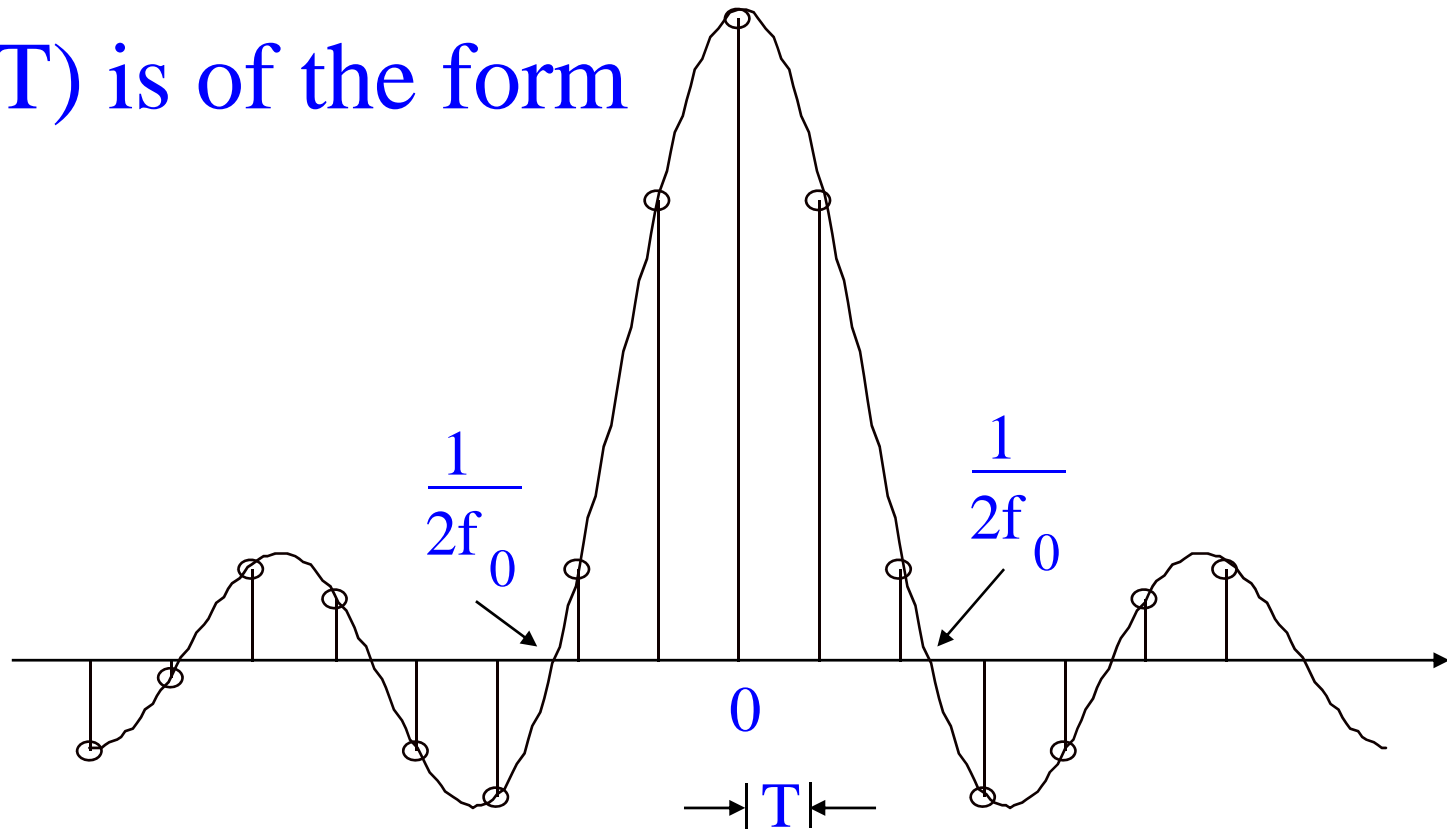
## Impulse Response

$$h(kt) = h(t) \Big|_{t=\frac{2\pi}{\omega_s}k} = \frac{\omega_0}{\pi} \frac{\sin \omega_0 t}{\omega_0 t} \Big|_{t=\frac{2\pi}{\omega_s}}$$
$$\frac{\omega_0}{\pi} \frac{\sin \frac{\omega_0}{\omega_s} 2\pi k}{\frac{\omega_0}{\omega_s} 2\pi k}$$



# Impulse Response

$h(kT)$  is of the form



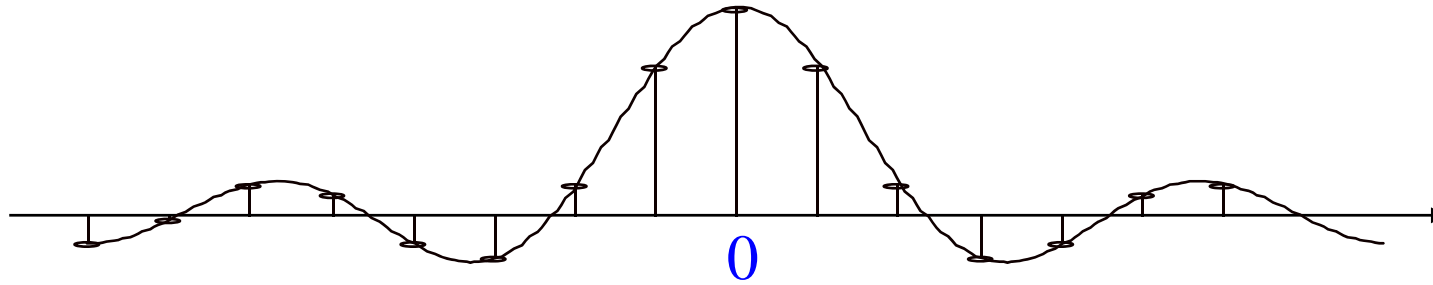


## Truncation

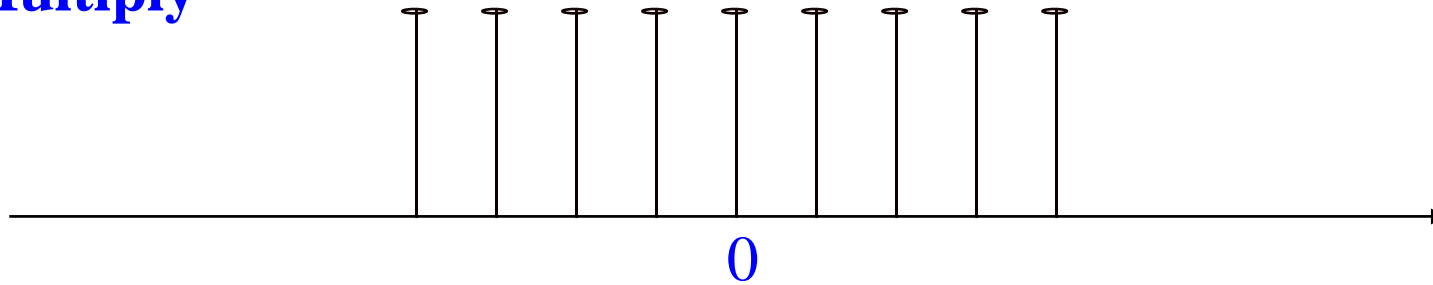
- To realise the frequency response we are required to use all of the infinite samples in the impulse response. But we are building a filter with a finite time response! We can do this by simply truncating the above series; we can set the coefficients past the fourth to zero by multiplying by a function which is 1.0 out to the fourth sample, and is zero elsewhere.
- Another word for truncation is **windowing**.



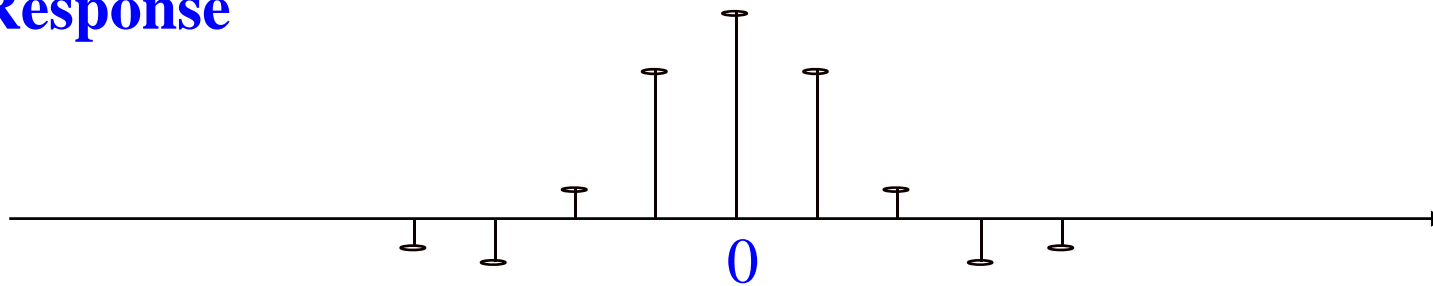
# Impulse Response



**Multiply**



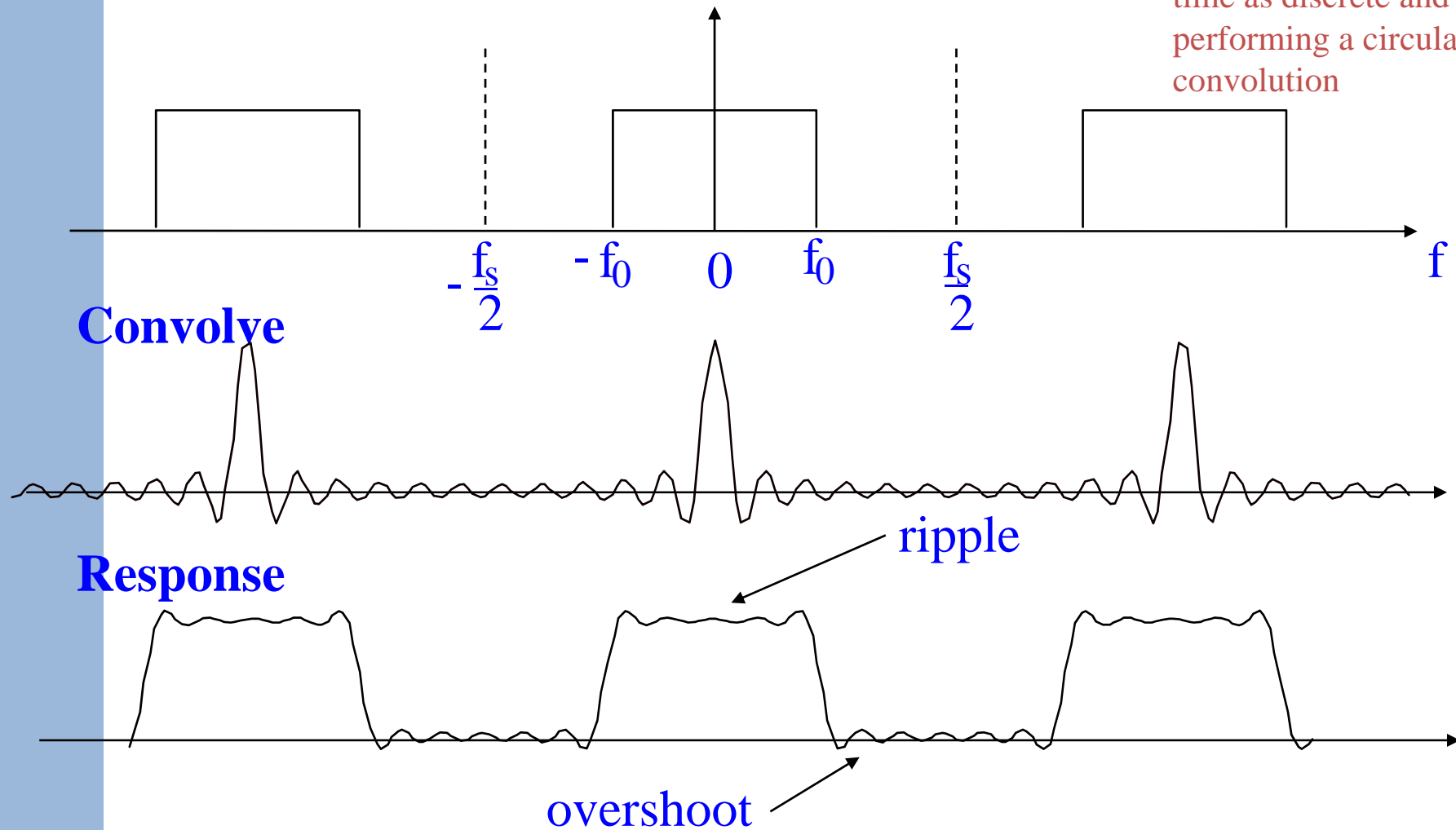
**Response**





# Frequency Response

Note: In this case we are treating time as discrete and performing a circular convolution





## Comments

- The overshoot and ripple in the frequency domain is the price we pay for truncating the impulse response.
- It is the result of selecting the Fourier coefficients to minimise the mean squared error between the function and its Fourier series approximation.
- We, of course, recognise it as **Gibb's** phenomenon.
- The poor behaviour in the neighbourhood of the discontinuity is the result of convolving with a "bumpy" function.

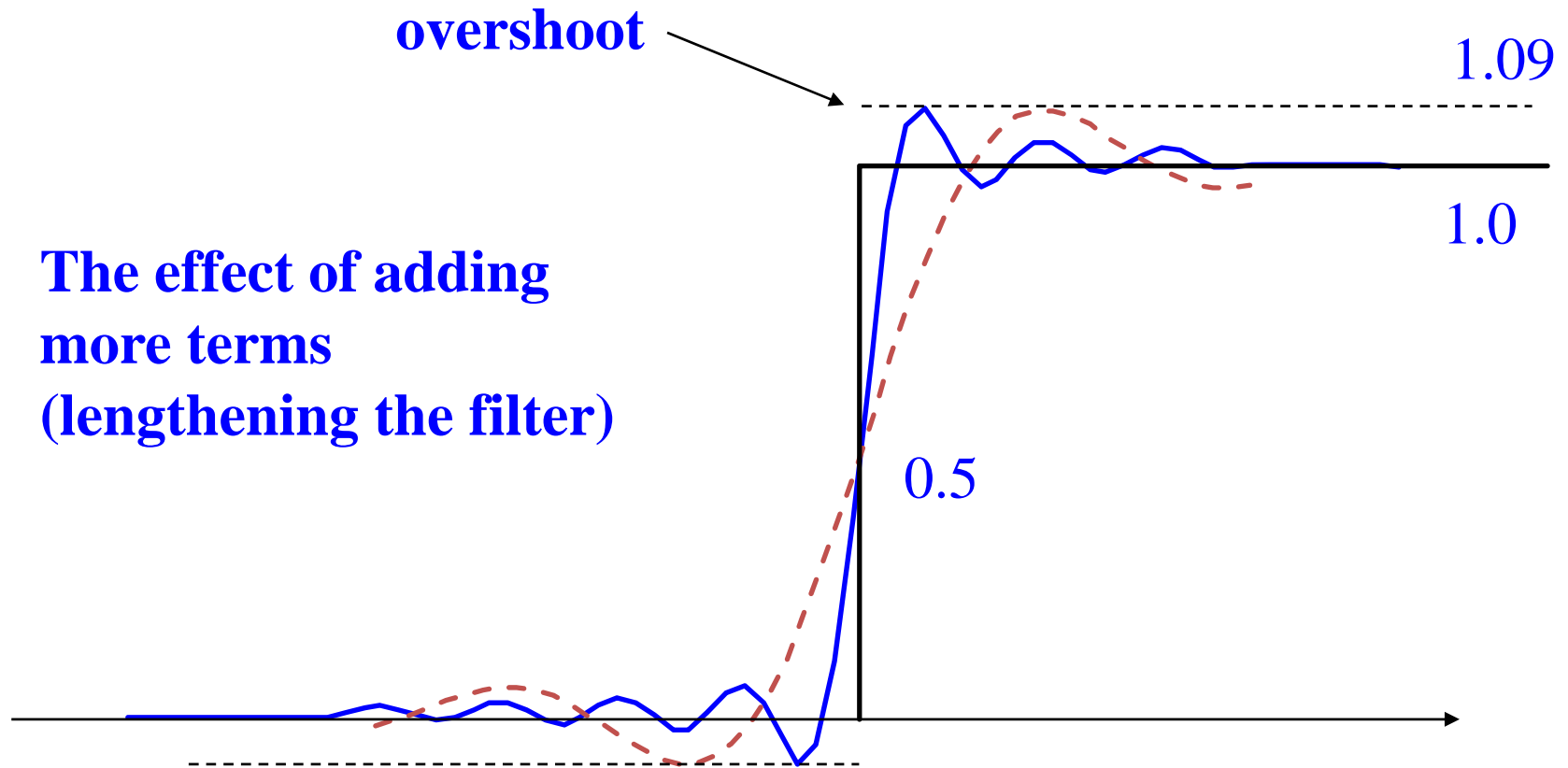




- Gibb's phenomena is not due to the fact that only a finite number of terms have been employed in the reconstruction!
- The magnitude of the peak ripple is invariant to the number of terms in the series.
- Adding more terms simply moves the ripple towards the discontinuity; it does not change its structure.



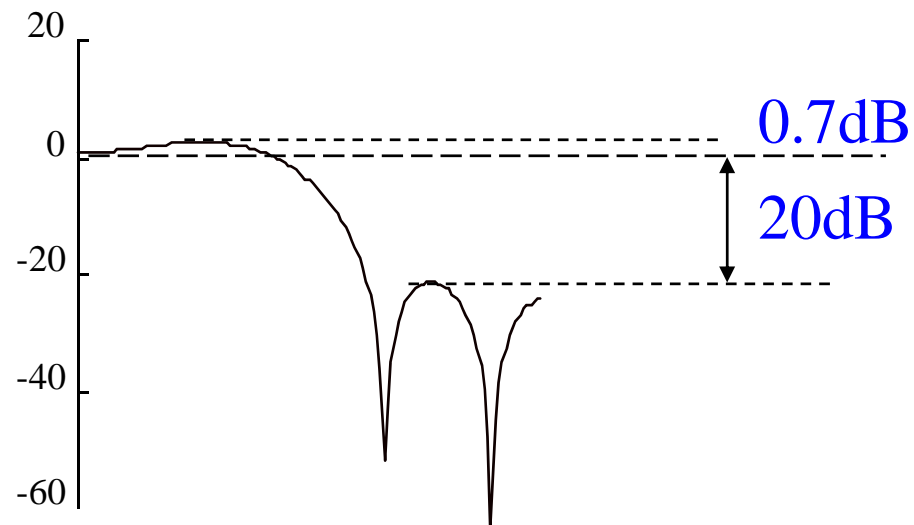
# Gibb's Effect





## Comments

- Note that ripple in the neighbourhood of 10% overshoot in the passband represents only 0.7 dB, but that 10% ripple in the stop band represents 20 dB.





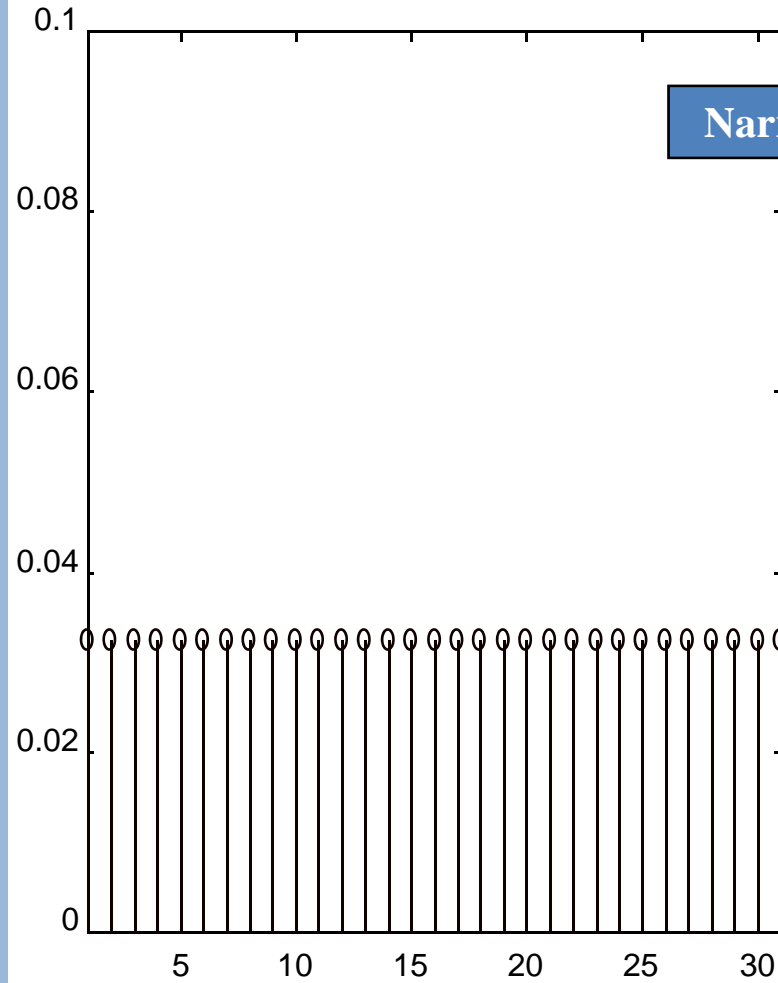
- We would probably find this filter inadequate to meet filtering requirements in the stop band.
- How do we improve performance?
- We have commented that the poor ripple characteristics are due to the bumpy function used in the convolution.
- To reduce the ripple, we need a truncation function whose Fourier Transform is not as bumpy (i.e., smoother)
- The bumpy behaviour of the transform is related to the discontinuity of the truncation function.



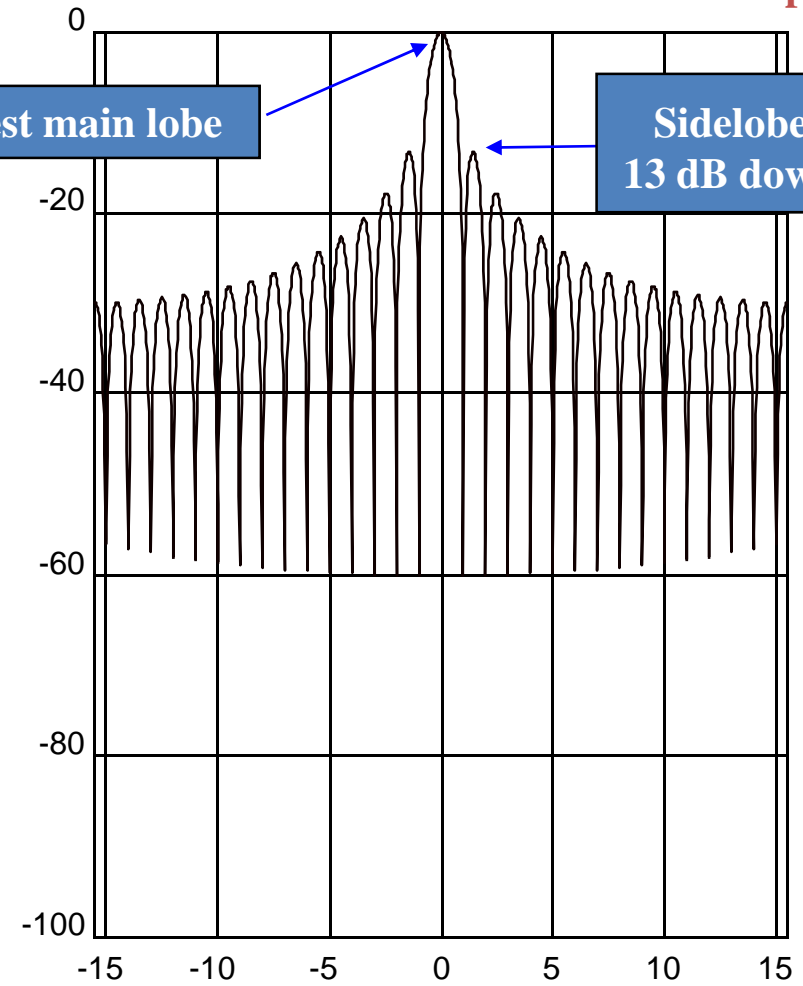
- The study of smooth truncation functions classically referred to as windows is a very important, but not well-understood segment of DSP.
- Misconceptions abound!
- What follows is an assortment of common window functions followed by a series of important observations.



Normalised Rectangular Window



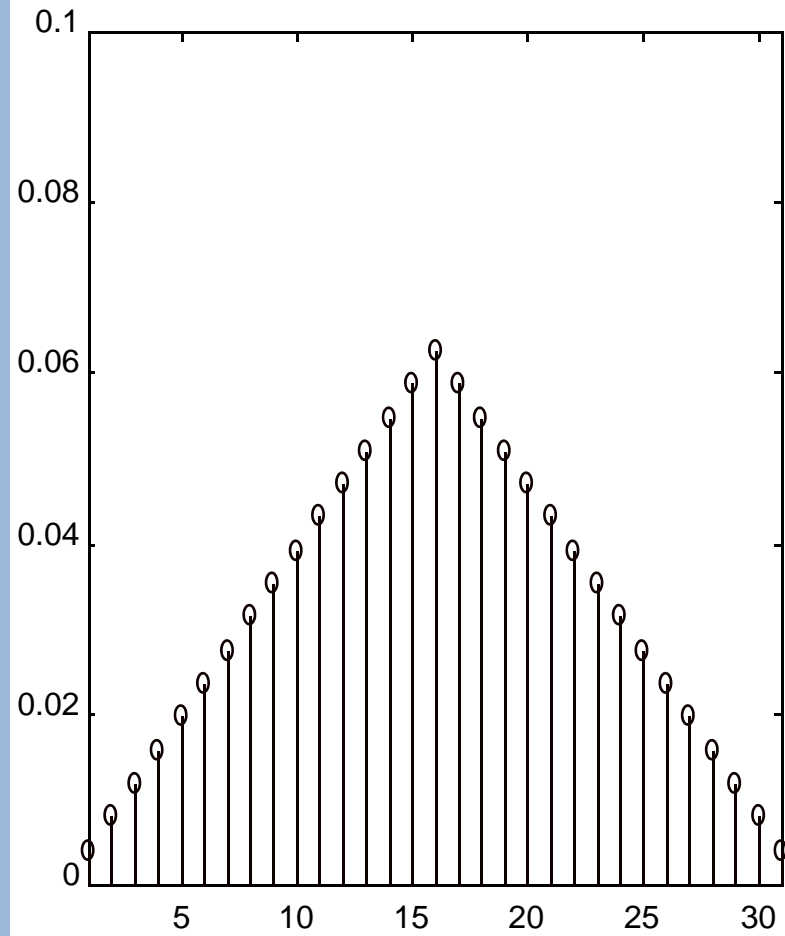
Spectrum of Rectangular Window



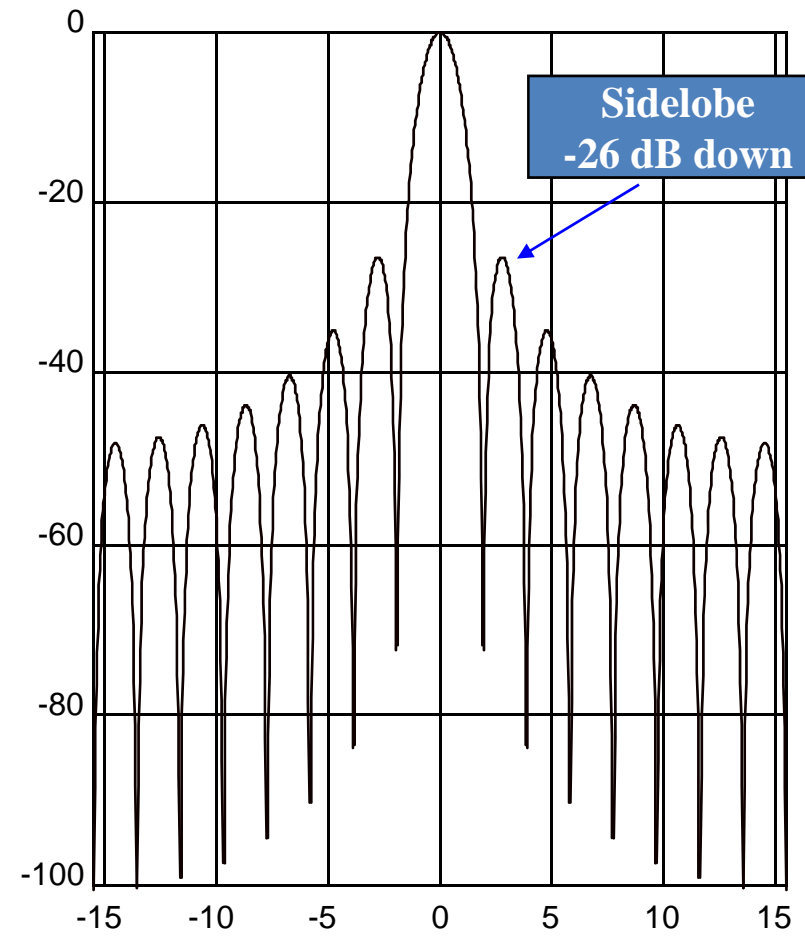
winplot



Normalised Barlett Window

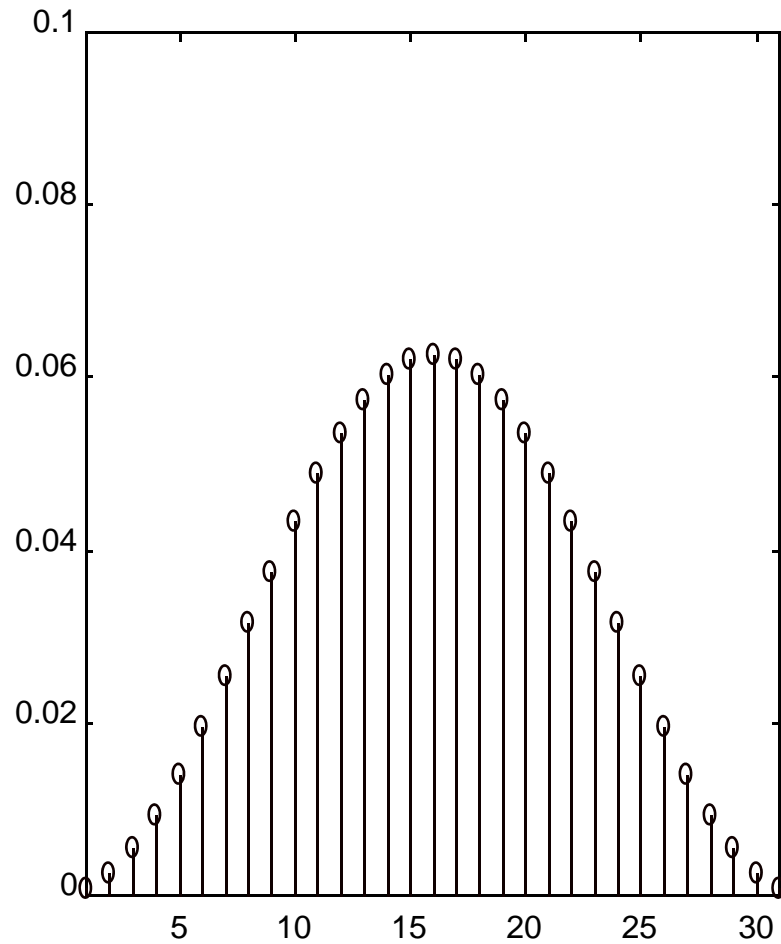


Spectrum of Barlett Window

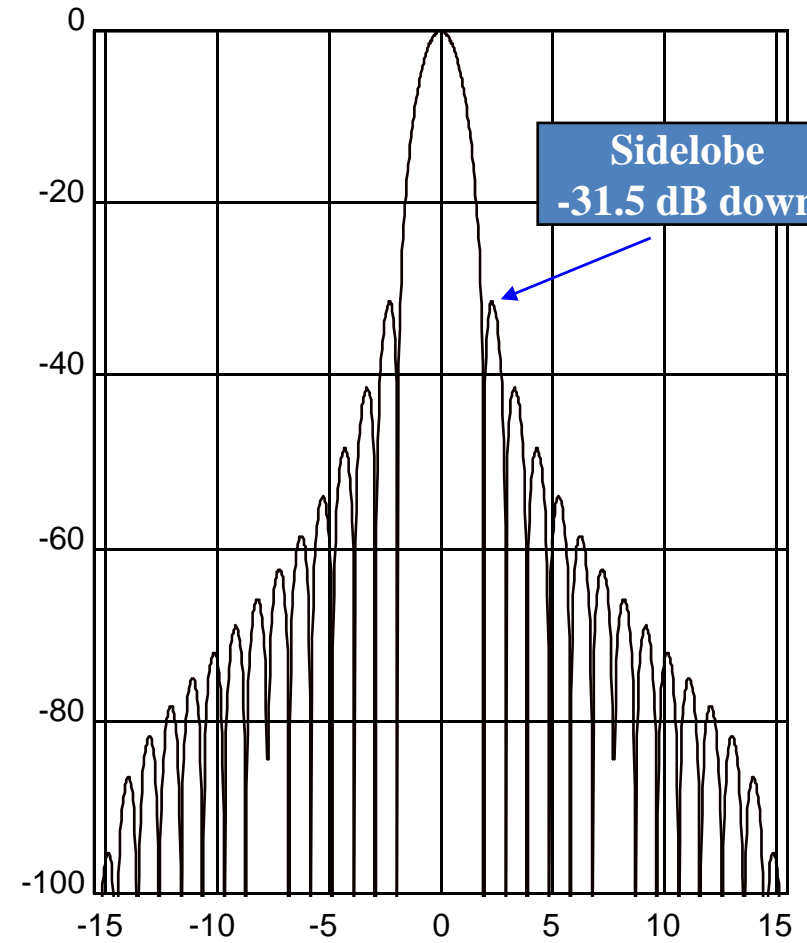




Normalised Hann Window



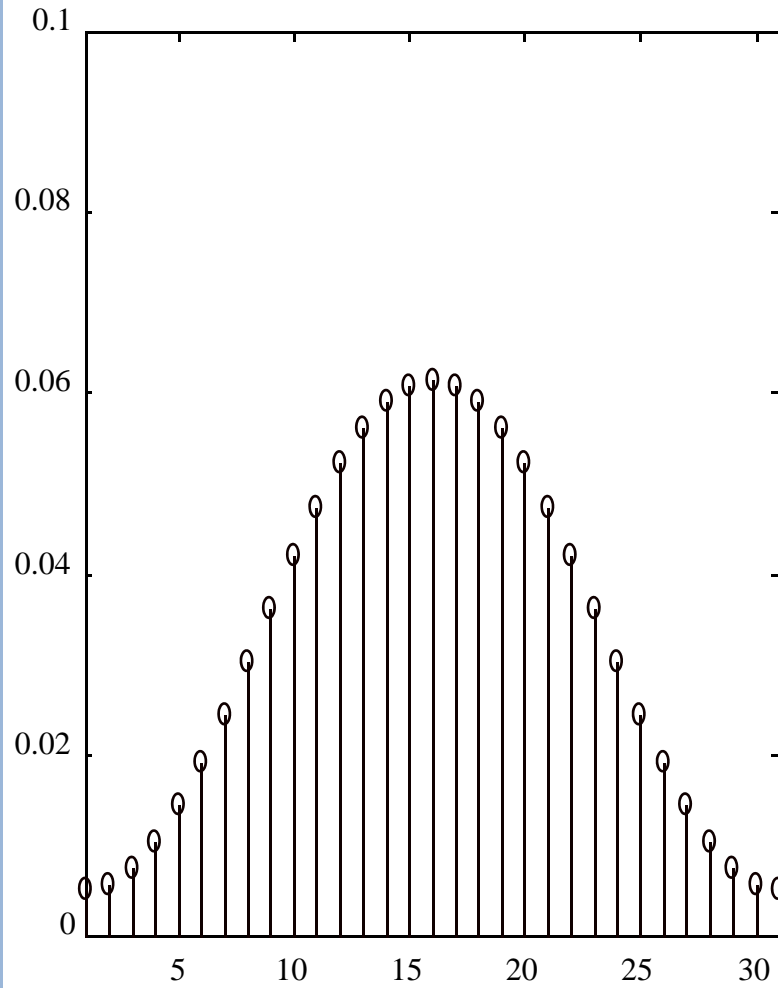
Spectrum of Hann Window



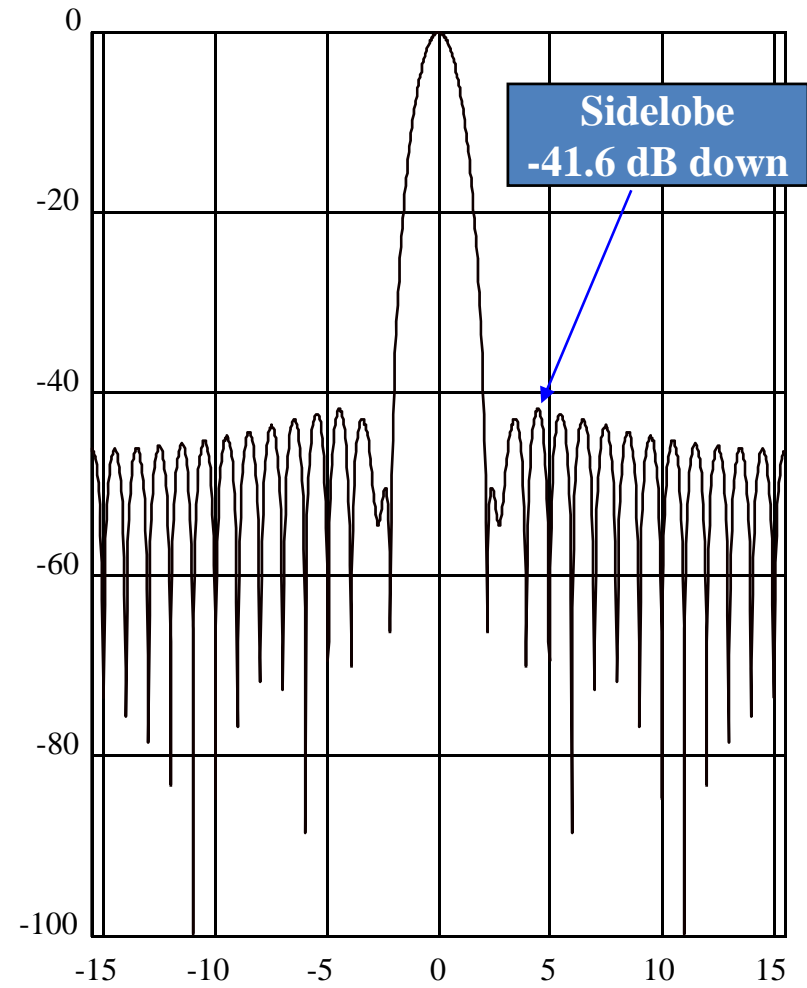




Normalised Hamming Window

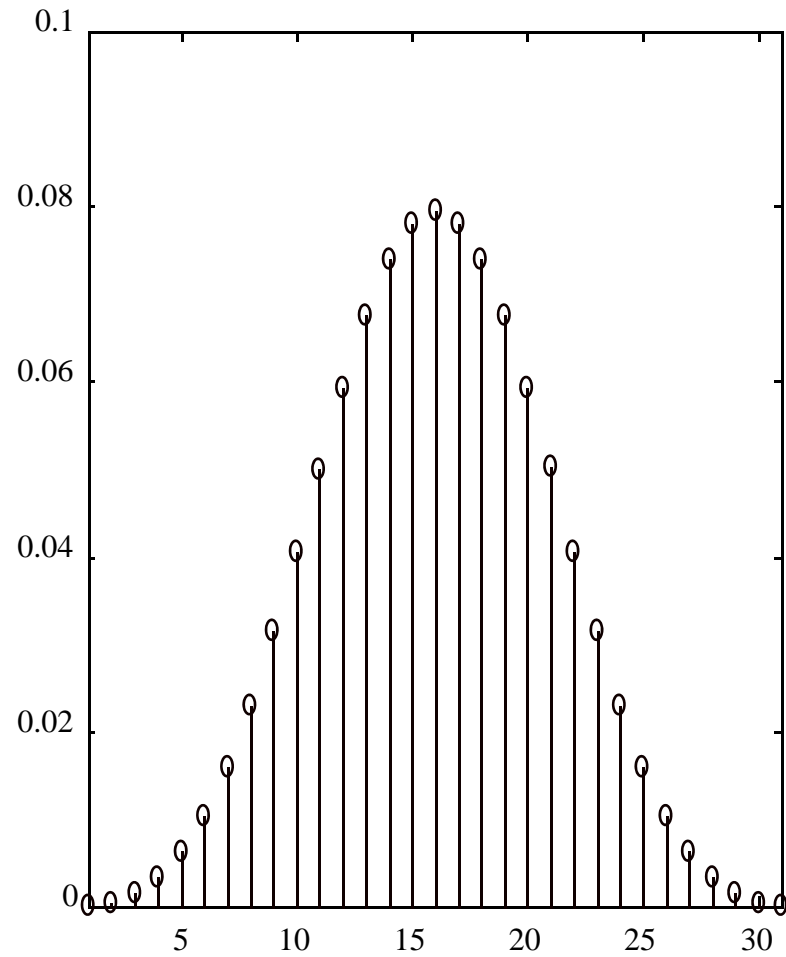


Spectrum of Hamming Window

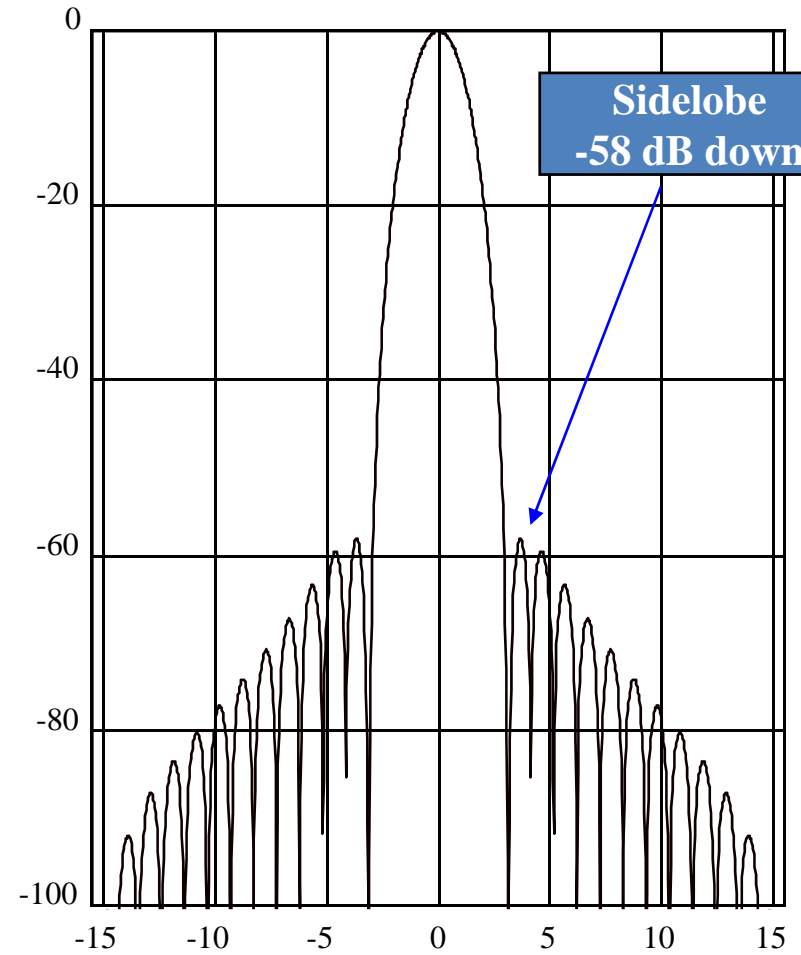




Normalised Blackman Window

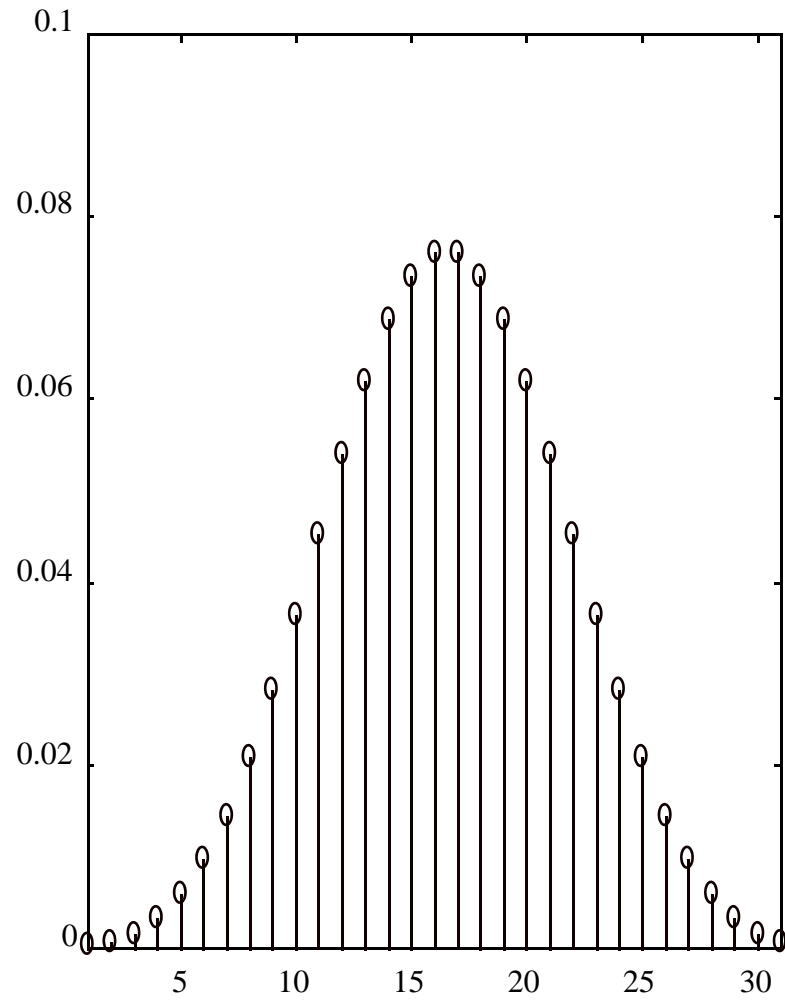


Spectrum of Blackman Window

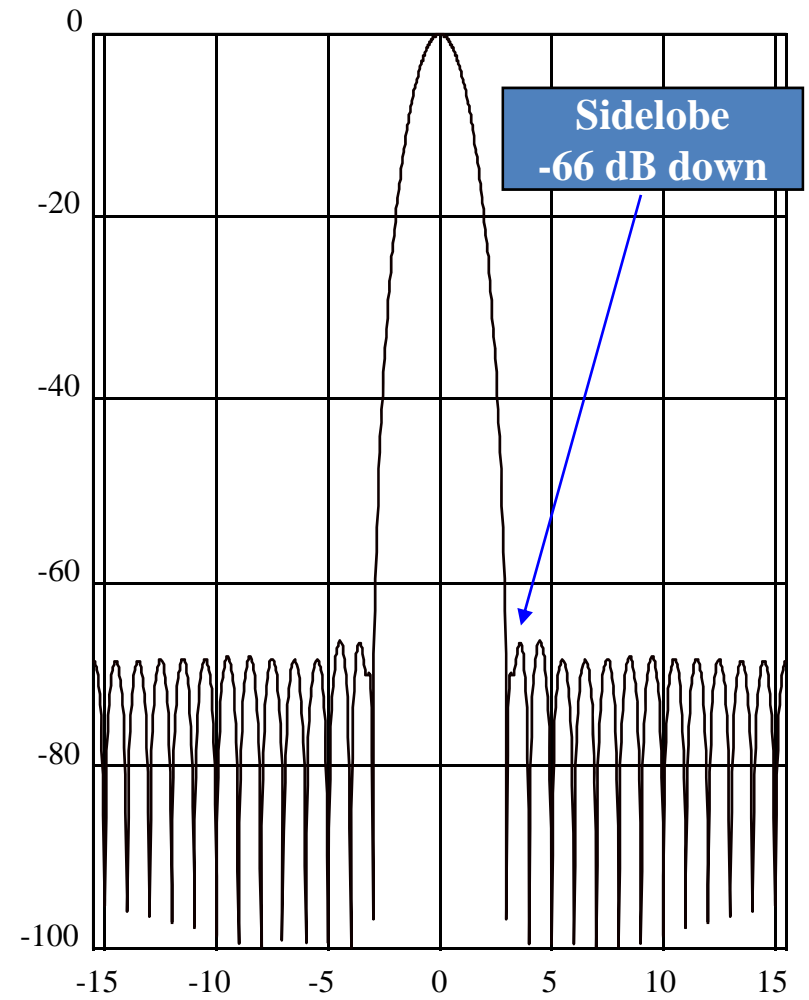




Normalised 3-Term Blackman-harris Window

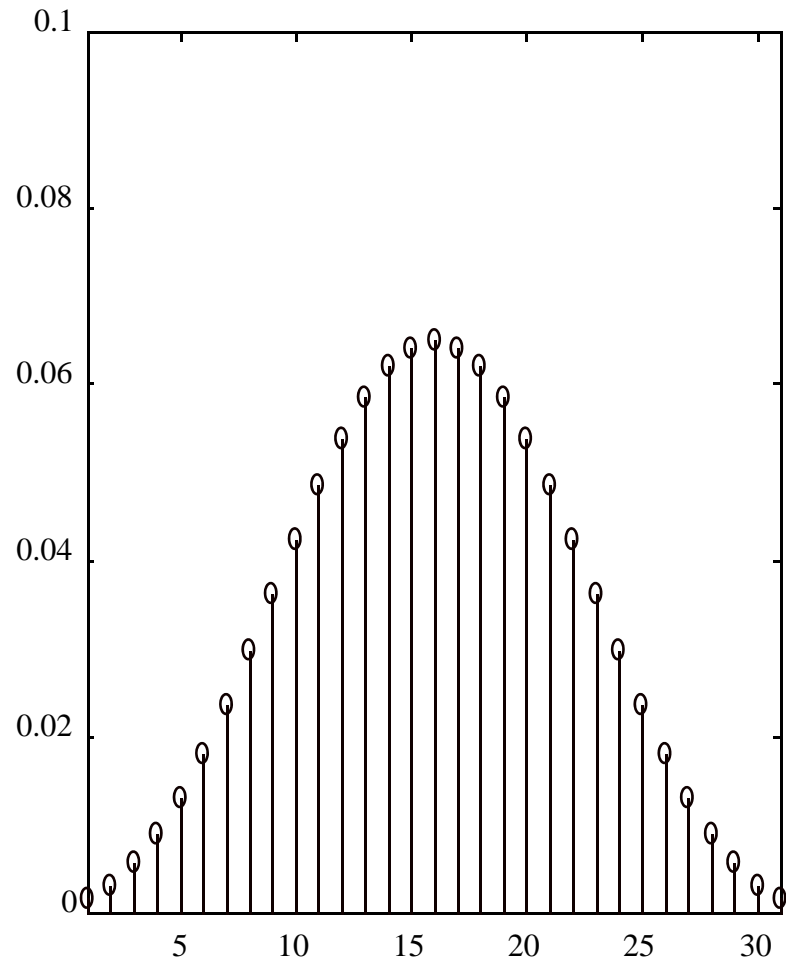


Spectrum of 3-Term Blackman-harris Window

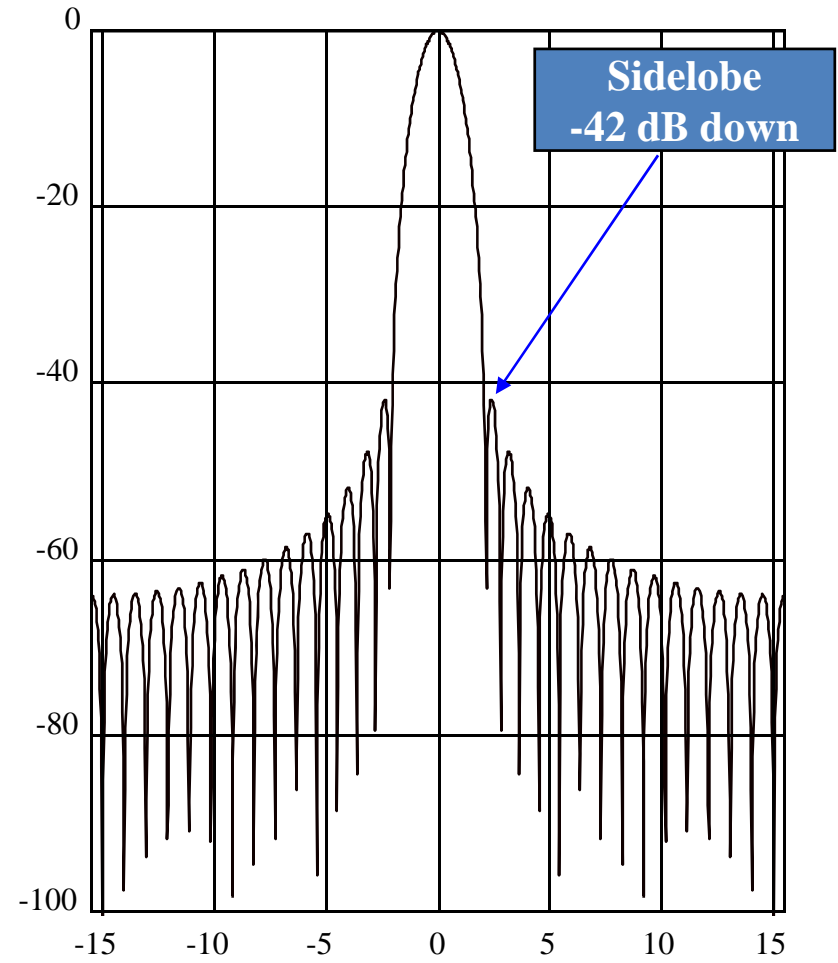




Normalised Kaiser Window (Beta=5.65)

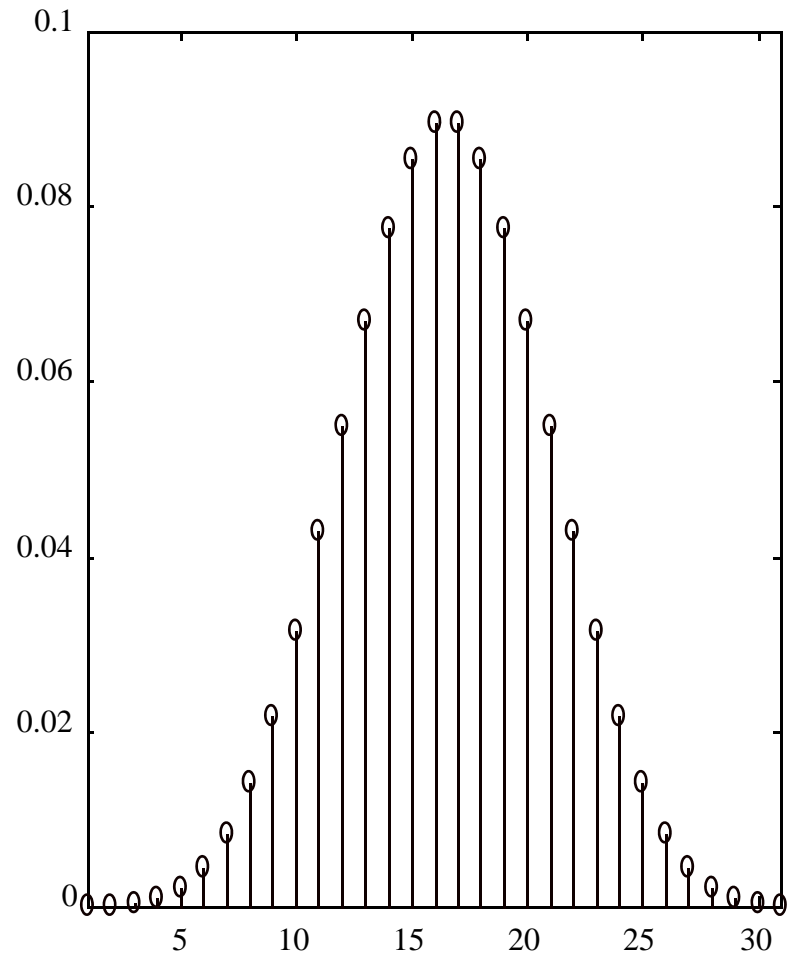


Spectrum of Kaiser Window (Beta=5.65)

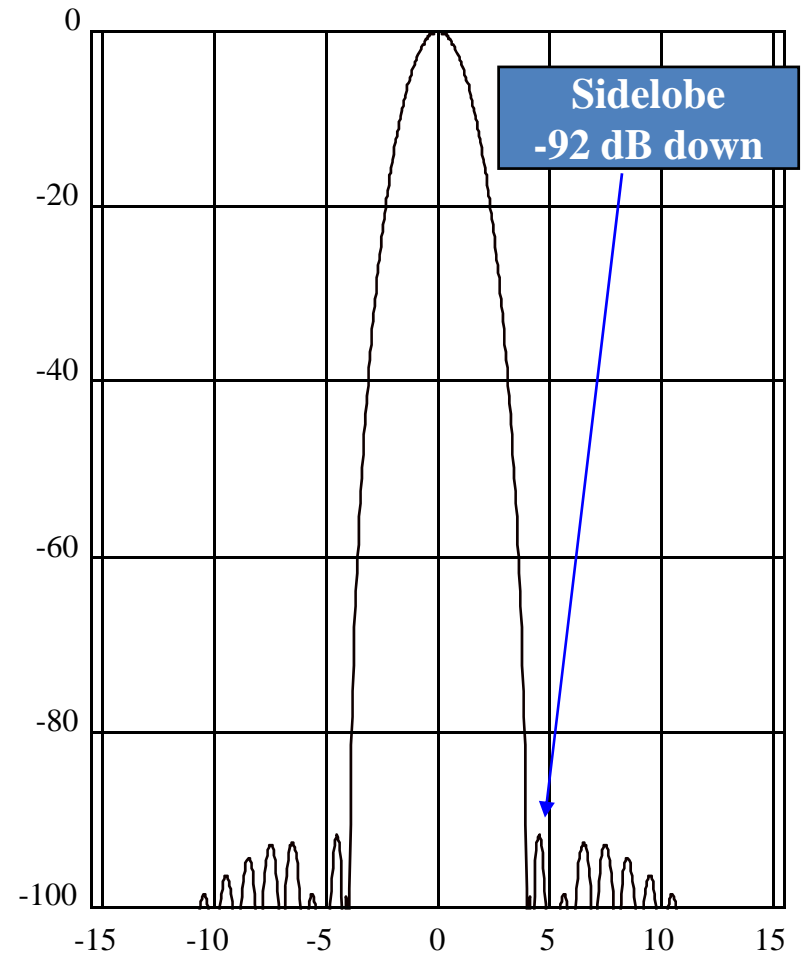




Normalised 4-Term Blackman harris Window



Spectrum of 4-Term Blackman harris Window



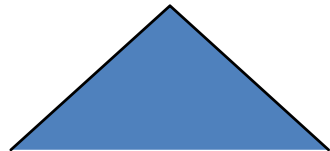
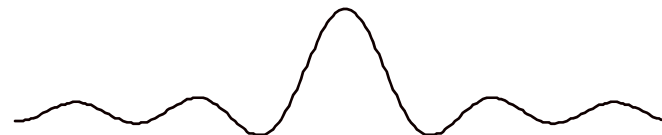


# Windows

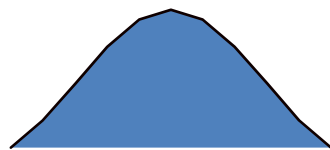
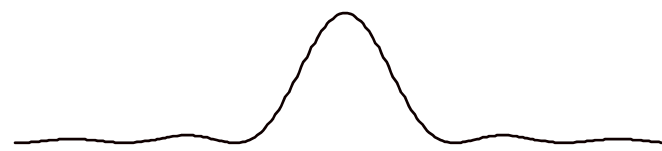
- Windows with “smoother” behaviour in the time-domain are “smoother” in the frequency domain.



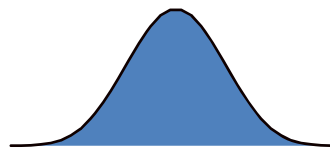
boxcar



Bartlett



Hann



bkharris



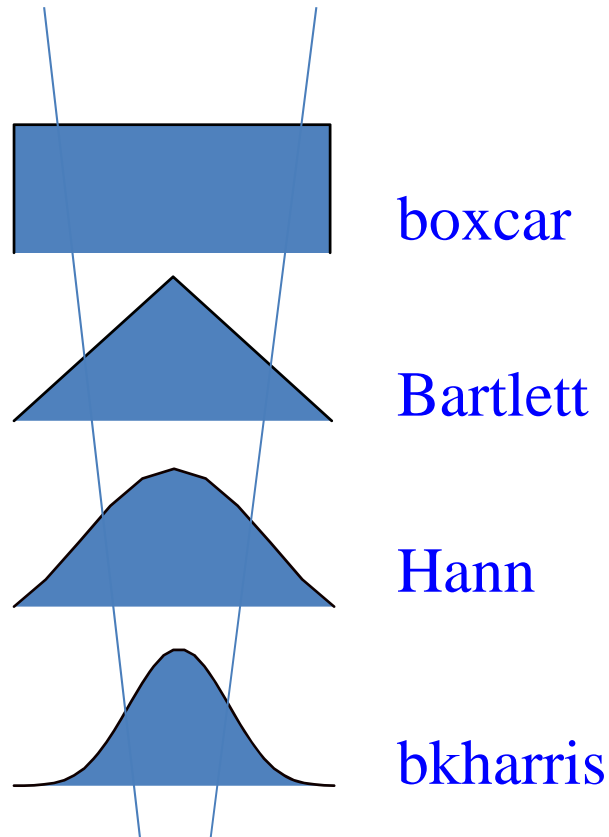
**smoother**





## Windows

- Windows that are “smoother” in the time-domain tend to have narrower time duration (e.g., smaller rms time-widths).



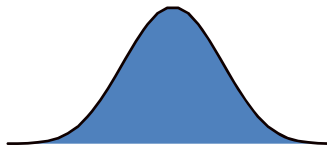
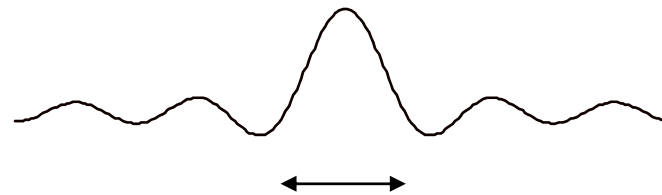


# Windows

- Windows that are “smoother” in the time domain tend to have wider bandwidths (e.g., greater rms bandwidths)



boxcar



bkharris

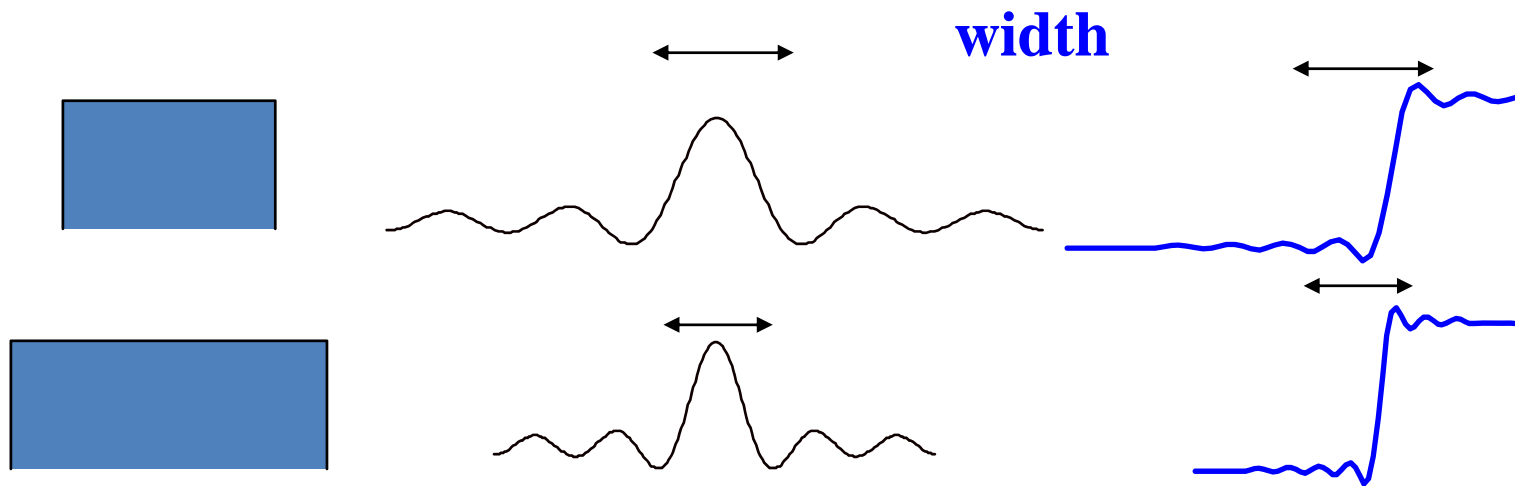






## Windows

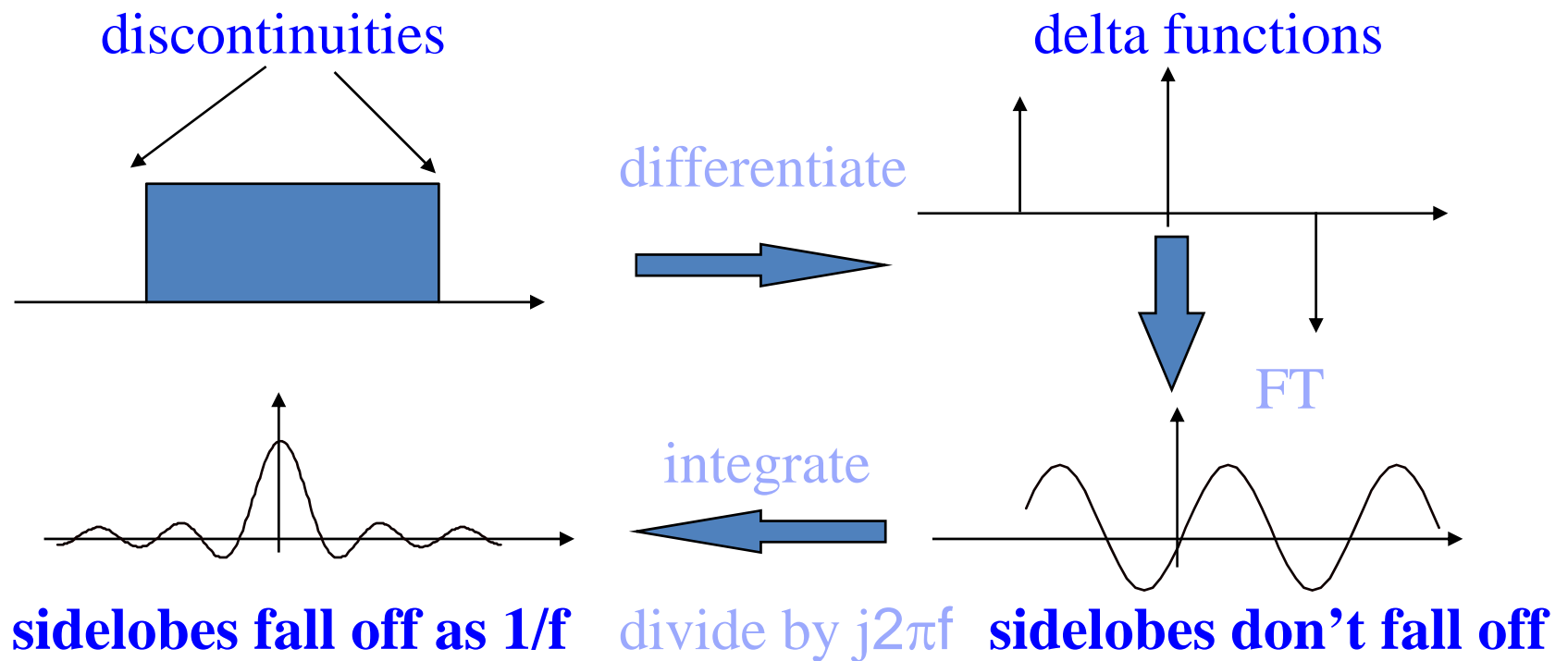
- When convolving with a wider function, the resultant function will have wider transitions (smaller slope).





## Smoother Windows

- One mathematical interpretation of "smoother" is continuous derivatives.





# Smoother Windows

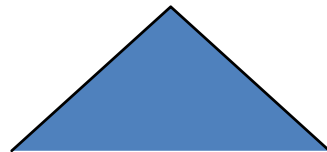
- So a delta functions in the  $n$ th derivative implies<sup>n</sup> that sidelobes fall off as  $1/f$  . **sidelobes fall off as**



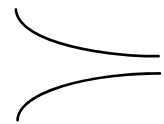
boxcar



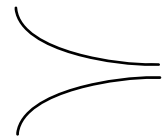
$$\frac{1}{f}$$



Bartlett



$$\frac{1}{f^2}$$



$$\frac{1}{f^3}$$

windows get narrower

**Need continuous derivatives**



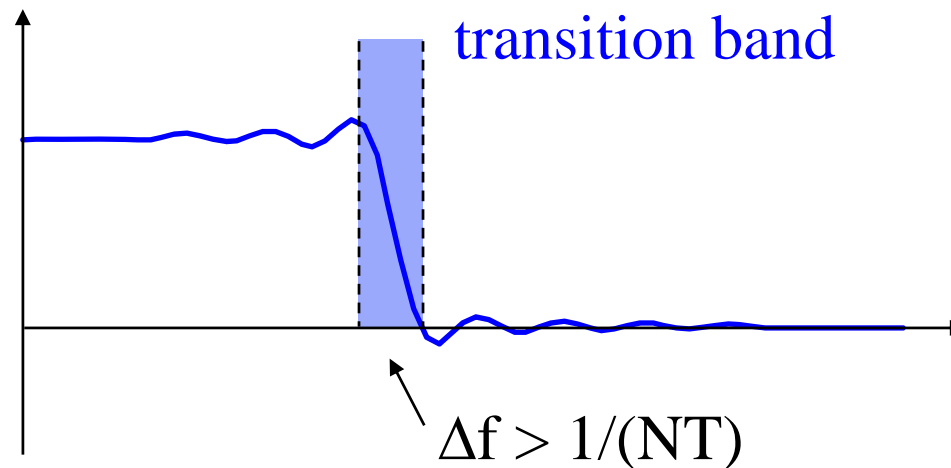
## Sidelobes

- Don't want constant sidelobes.
- Why?
- When we downsample all the noise adds up.
- Always use windows which have some sidelobe fall-off . ( About 3dB per octave is sufficient)



## Transition Width

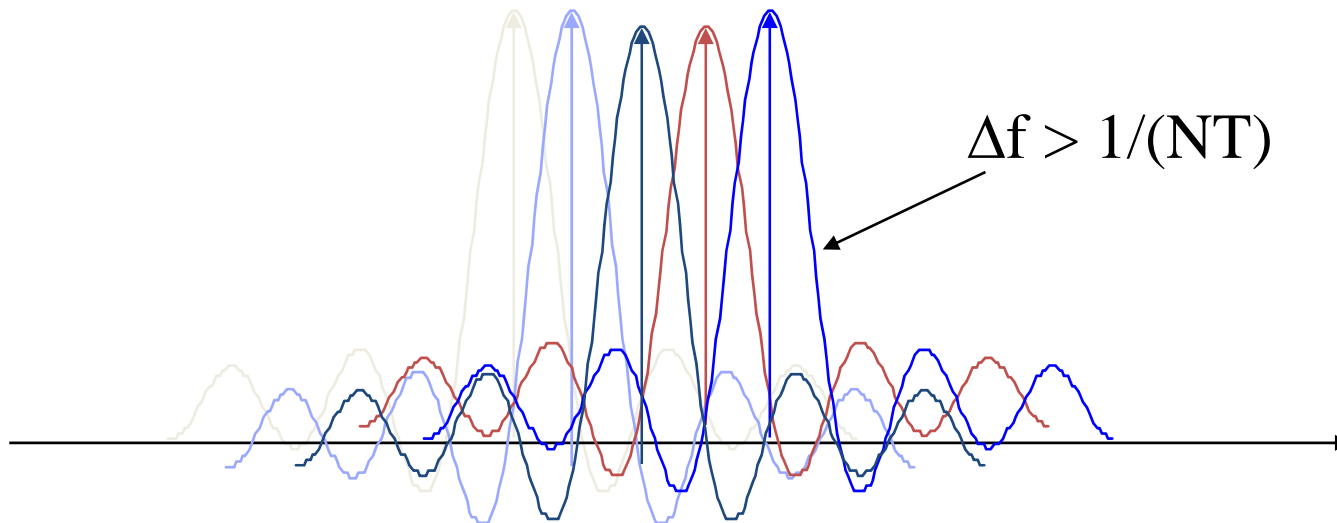
- The boxcar window gives the fastest rolloff in the transition band but has large sidelobes in the stop band.
- We trade rolloff for sidelobe suppression by using more sophisticated window functions.





## Another View

- If we use a boxcar function, the rolloff in the transition zone is determined by the final sinc function and this is the fastest possible rolloff that can be achieved.





## Optimal Windows

- Dolph-Chebyshev
  - Minimum main lobe width for lowest sidelobe level
- Kaiser-Bessel
  - Window is a good approximation to finite prolate spheroidal functions; minimum time bandwidth product
  - Family of windows with different sidelobe levels
- Blackman-harris
  - Minimum sidelobe for a fixed number of spectral lines