Discrete-time filtering

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Resumen

1. Introduction to Discrete Filters
   - Classification of discrete filters
2. Filtering in time domain
   - Time domain parameters
   - Moving average filter
   - Noise Reduction vs. Step Response
   - Frequency Response
   - Leaky integrator filter
3. Filtering in frequency domain
   - Frequency domain parameters
   - Filters by windowing
   - Kaiser window filter
   - FIR filter design
   - FIR types
   - FIR structures

Filtering in different domains

- Filtering in time domain (signal restoration, smoothing, denoising).
  - Describes when something occurs and what the amplitude of the occurrence is.
  - Each sample contains information that is interpretable without reference to any other sample.
  - The step response describes how information represented in the time domain is being modified by the system.
  - Example: EKG signal, accelerometer, gyroscope.
- Filtering in frequency domain (signal separation).
  - The information is contained in the relationship between many points in the signal.
  - The frequency response shows how information represented in the frequency domain is being changed.
  - Example: digital filter for a hearing aid, equalizer.

It is not possible to optimize a filter for both domains. Good performance in the time domain results in poor performance in the frequency domain, and vice versa.

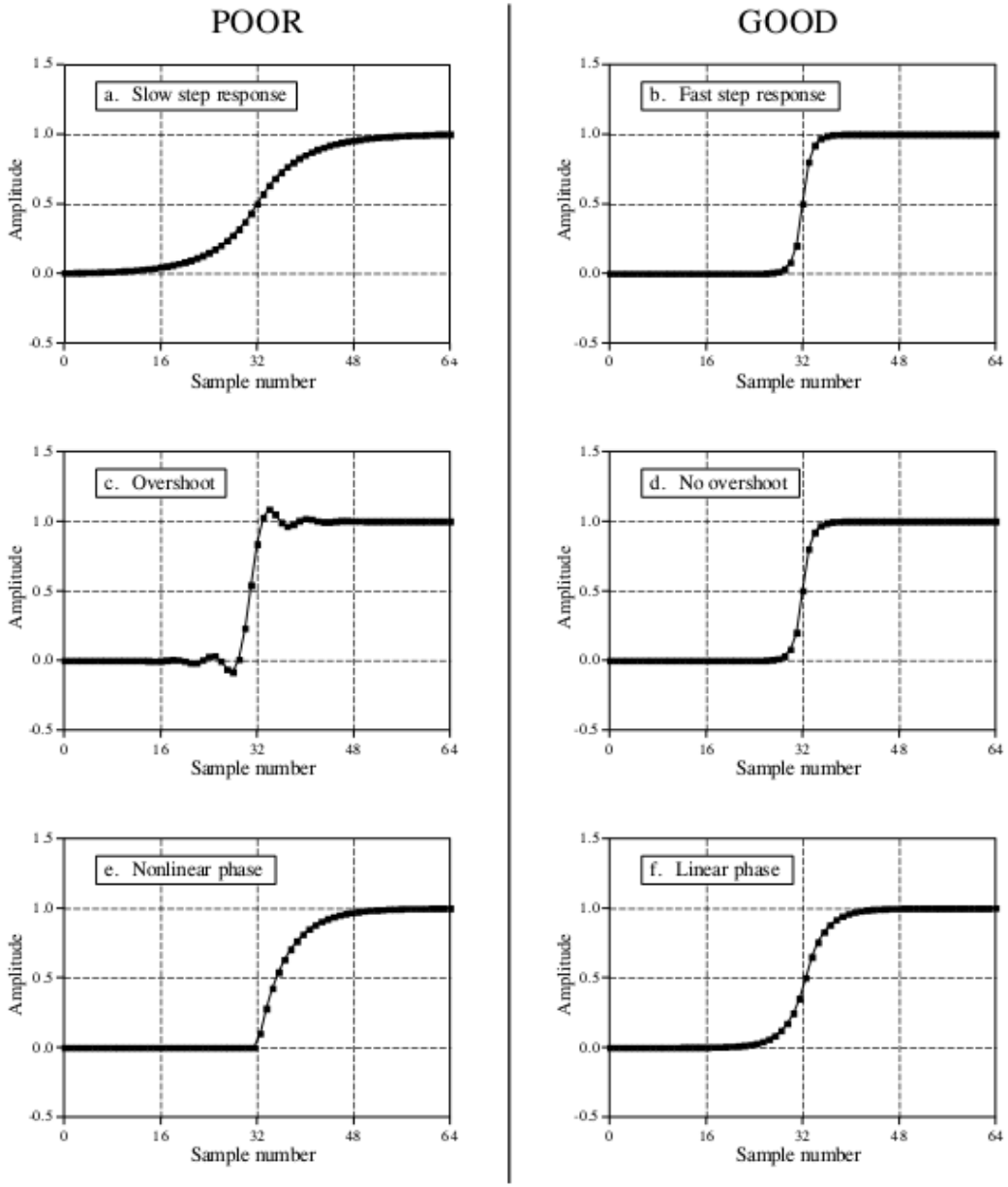
Classification of discrete filters

Table : Classification of discrete filters

	Finite impulse response (FIR)	Infinite impulse response (IIR)
Filtering in time domain	Moving average	Leaky Integrator
Filtering in frequency domain	Windowed Filters Equiripple Minimax	Bilinear z-transform

Time domain parameters, step response

- Risetime (between 10%∼90% amplitude).
- Overshoot.
- Linear phase.



Moving average filter

- The moving average filter is a convolution of the input signal with a rectangular pulse having an area of one.
- *Local average*.
- There is a delay of N/2 samples between input and output.

$$y[n] = x[n] * h[n] = \frac{1}{N} \sum_{k=0}^{N-1} x[n-k]\,, \tag{1}$$

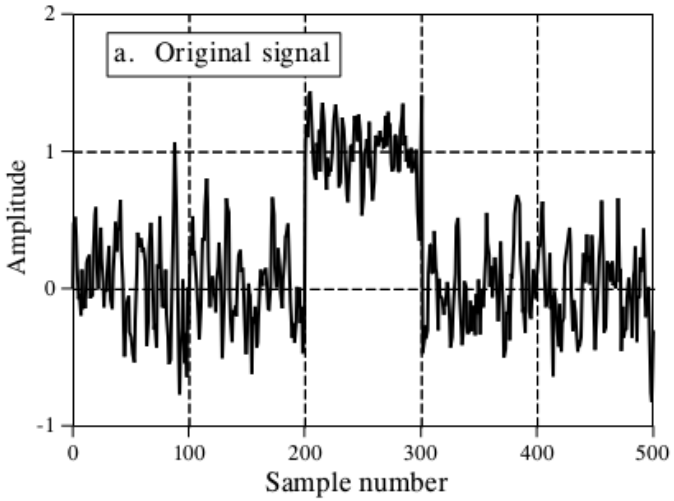
$$h[n] = \frac{1}{N} \sum_{k=0}^{N-1} \delta[n-k]\,, \tag{2}$$

$$SNR = 10 log_{10}(\sqrt{N})\,. \quad [3] \tag{3}$$

- It can be seen that the moving average filter is a FIR filter. Why?

Noise Reduction vs. Step Response

- It reduce random white noise while trying to keep the sharpest step response.



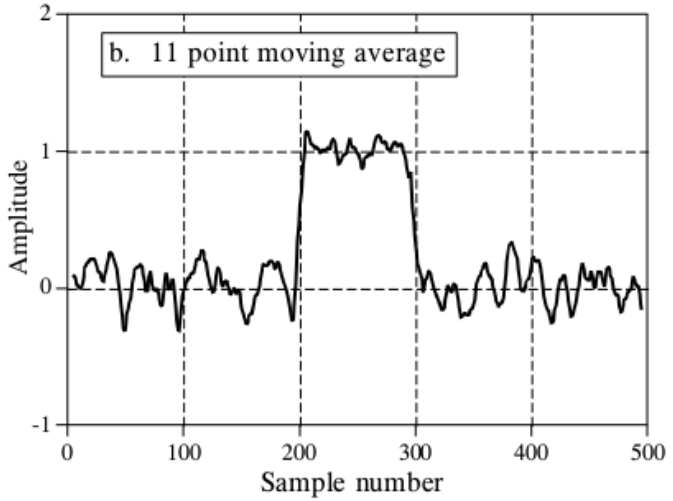
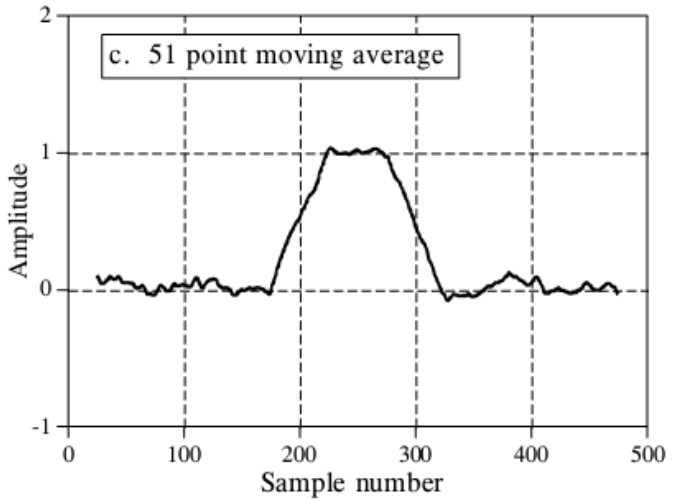
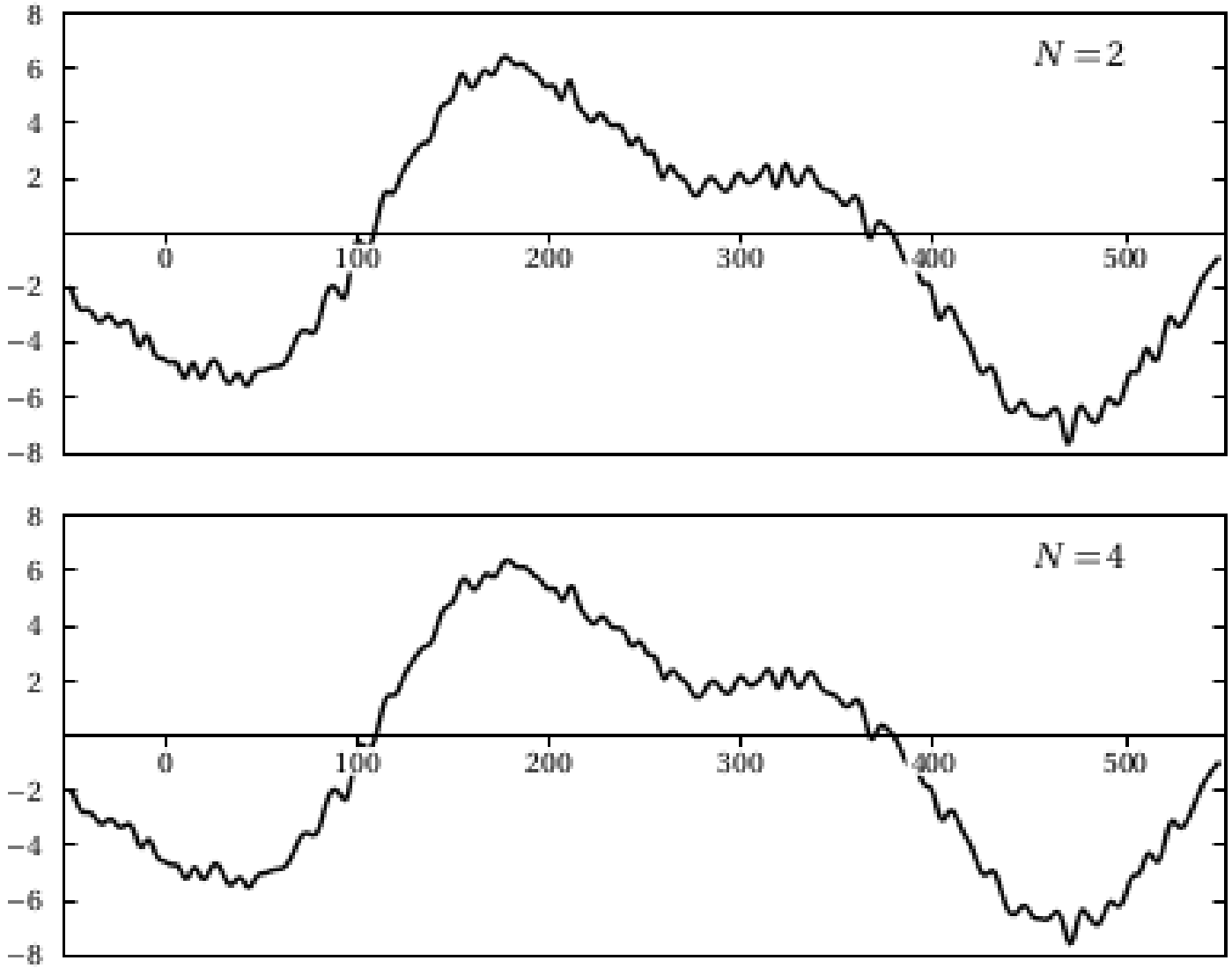


FIGURE 15-1
Example of a moving average filter. In (a), a rectangular pulse is buried in random noise. In (b) and (c), this signal is filtered with 11 and 51 point moving average filters, respectively. As the number of points in the filter increases, the noise becomes lower; however, the edges becoming less sharp. The moving average filter is the *optimal* solution for this problem, providing the lowest noise possible for a given edge sharpness.

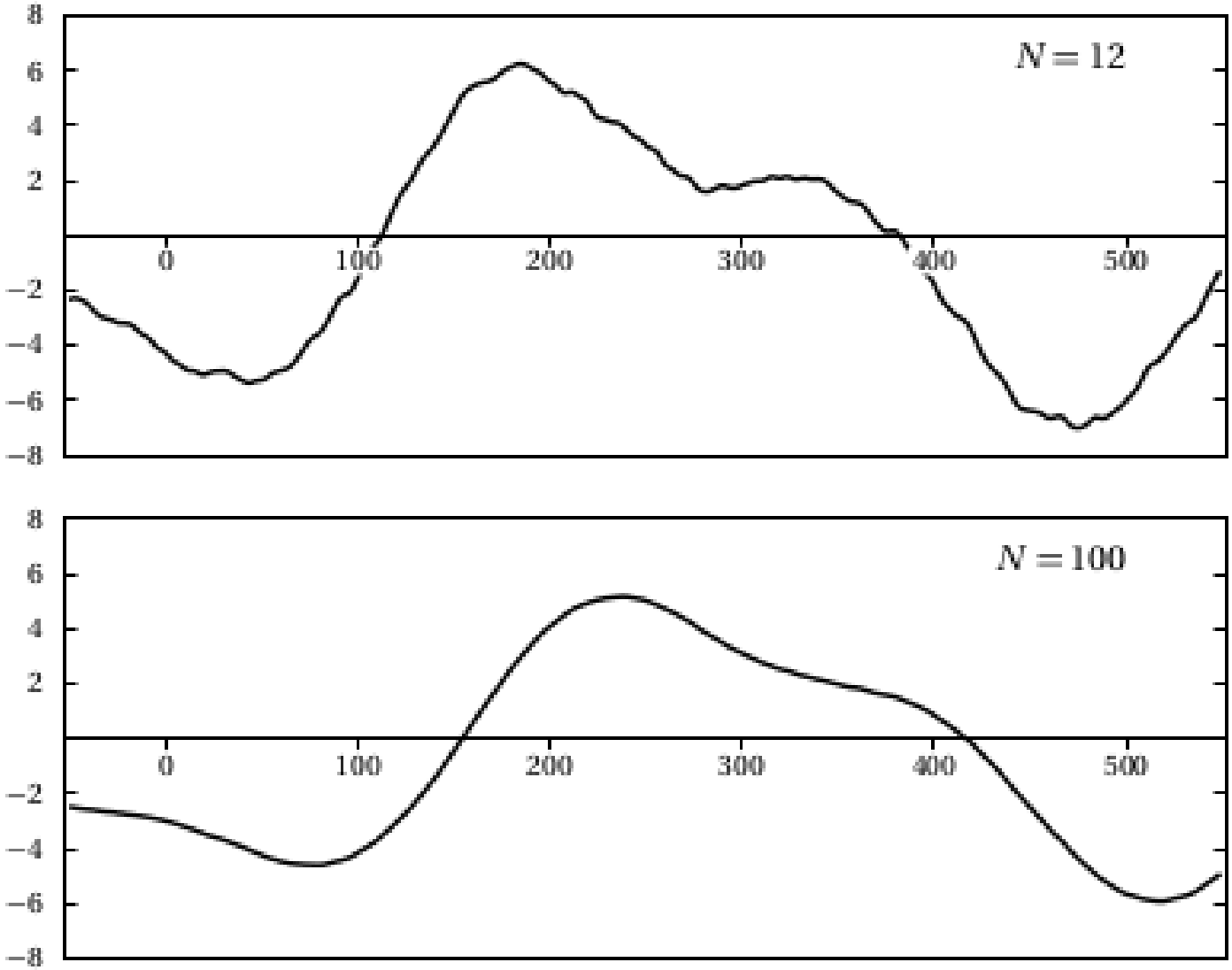


Noise Reduction

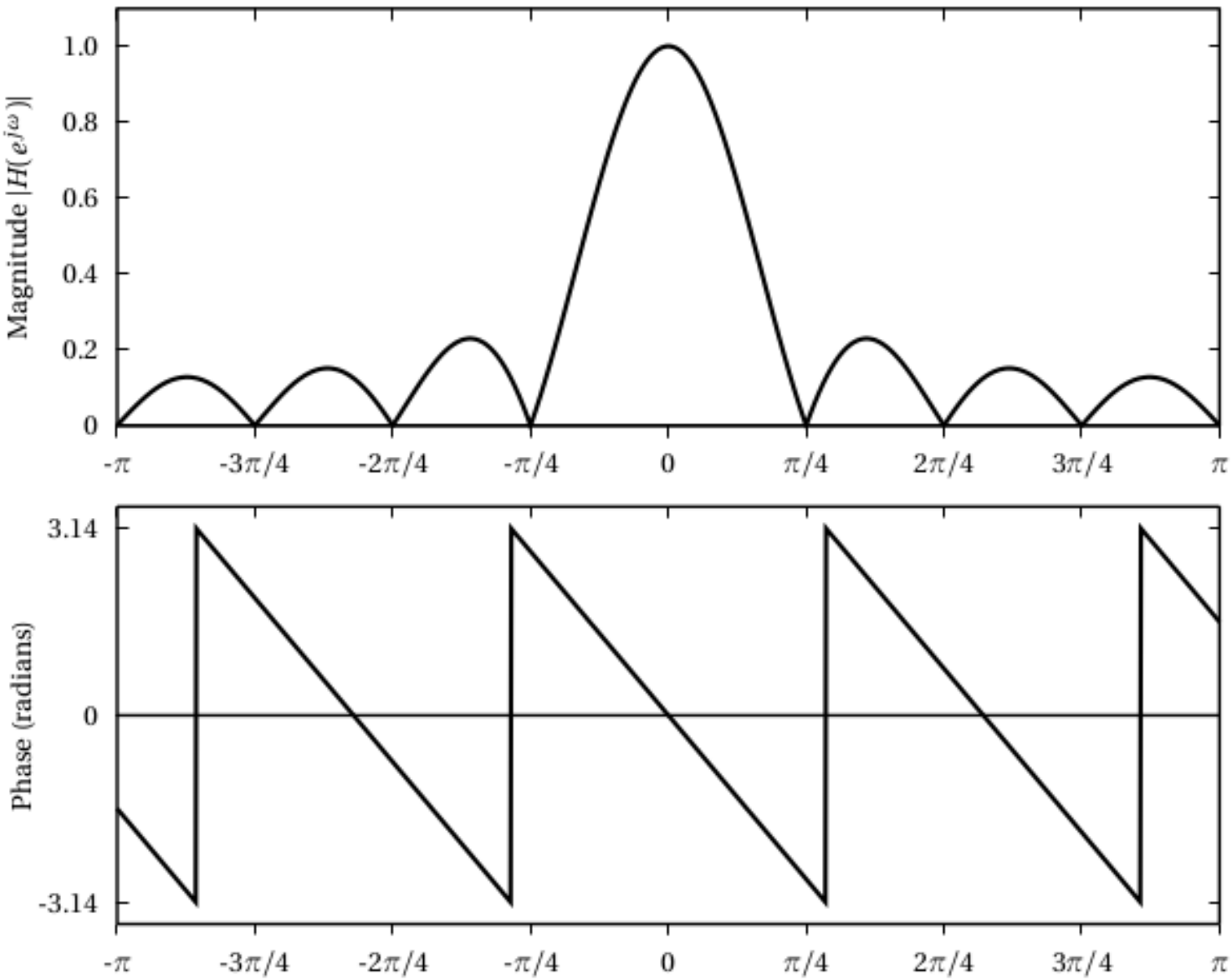


Noise Reduction

- Note how the signal is delayed as N grows.



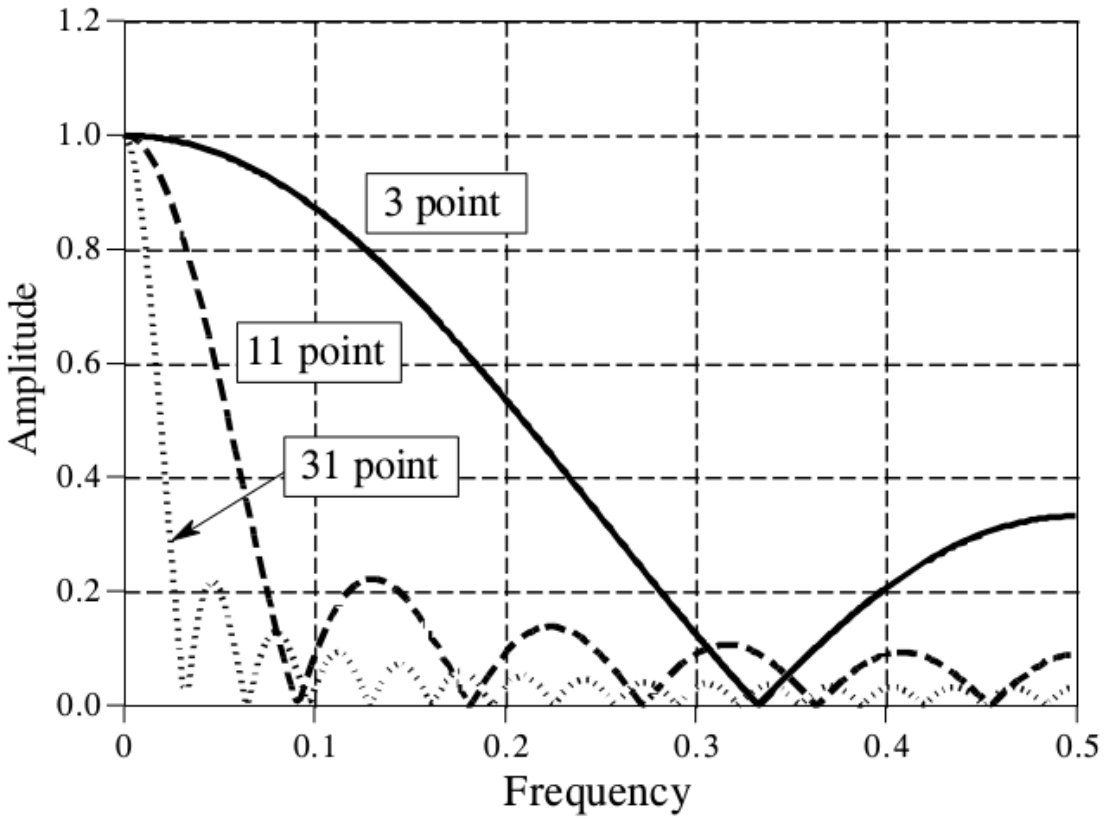
Frequency Response



Frequency Response

- The moving average filter cannot separate one band of frequencies from another.
- In short, the moving average is a good *smoothing filter* (the action in the time domain), but a bad low-pass filter (the action in the frequency domain).

$$H[e^{j\omega}] = \frac{1}{N}\frac{\sin(\omega N/2)}{\sin(\omega/2)}e^{-j\frac{N-1}{2}\omega}. \tag{4}$$



Leaky integrator filter

- The moving average filter is a convolution of the input signal with a rectangular pulse having an area of one.
- *Local average*.
- There is a delay of N/2 samples between input and output.

$$y[n] = x[n] * h[n] = \frac{1}{M} \sum_{k=0}^{M-1} x[n-k]\,, \tag{5}$$

$$y[n] = \frac{1-M}{M}\, y_{M-1}[n-1] + \frac{1}{M}\, x[n-k]\,. \tag{6}$$

Doing,

$$\lambda = \frac{1-M}{M}\,, \tag{7}$$

Then,

$$y[n] = \lambda\, y_{M-1}[n-1] + (1-\lambda)\, x[n-k]\,. \tag{8}$$

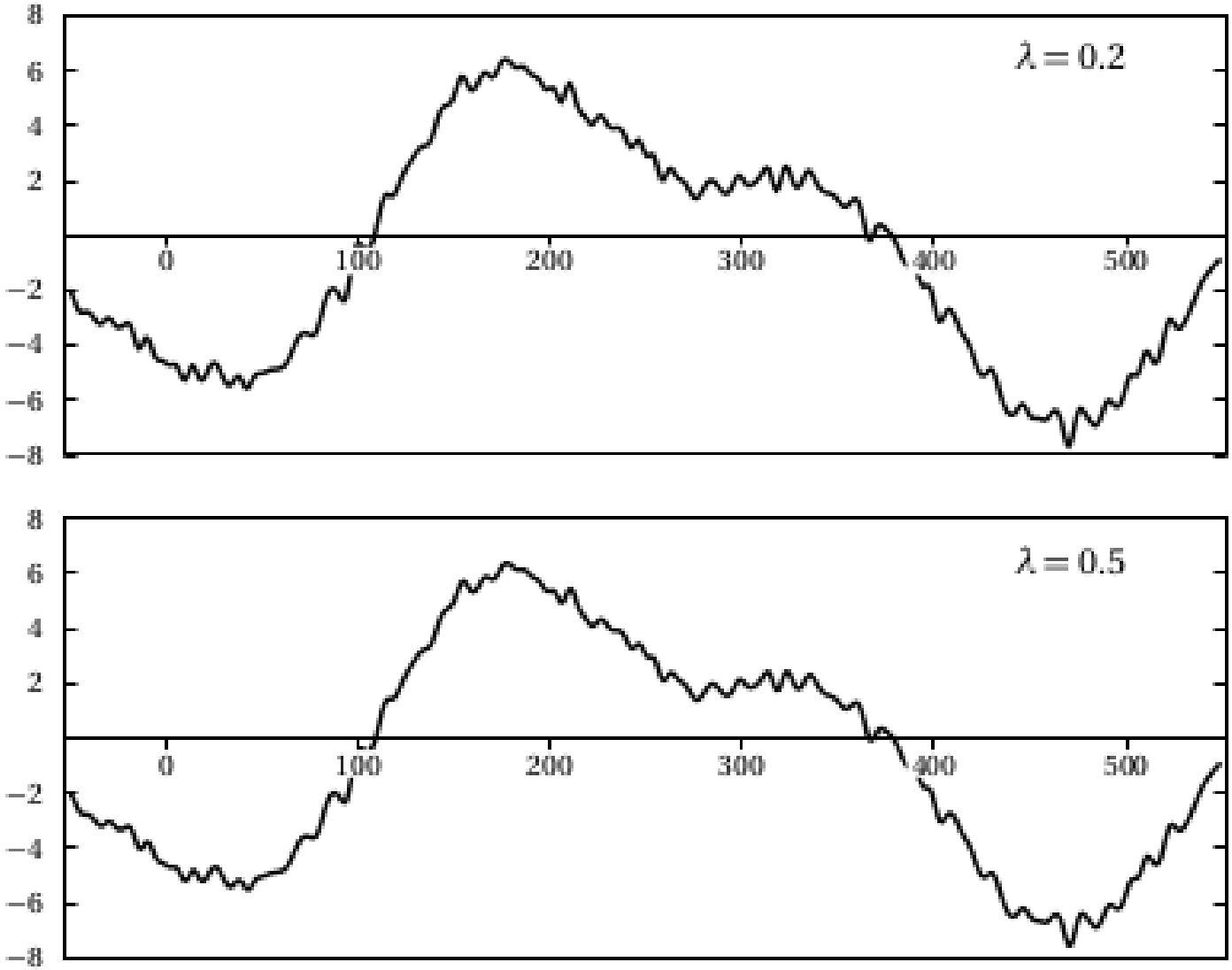
- It can be seen that the leaky integrator filter is an IIR filter. Why?

Leaky integrator filter

$$y[n] = \lambda\, y_{M-1}[n-1] + (1-\lambda)\, x[n-k]\,.$$

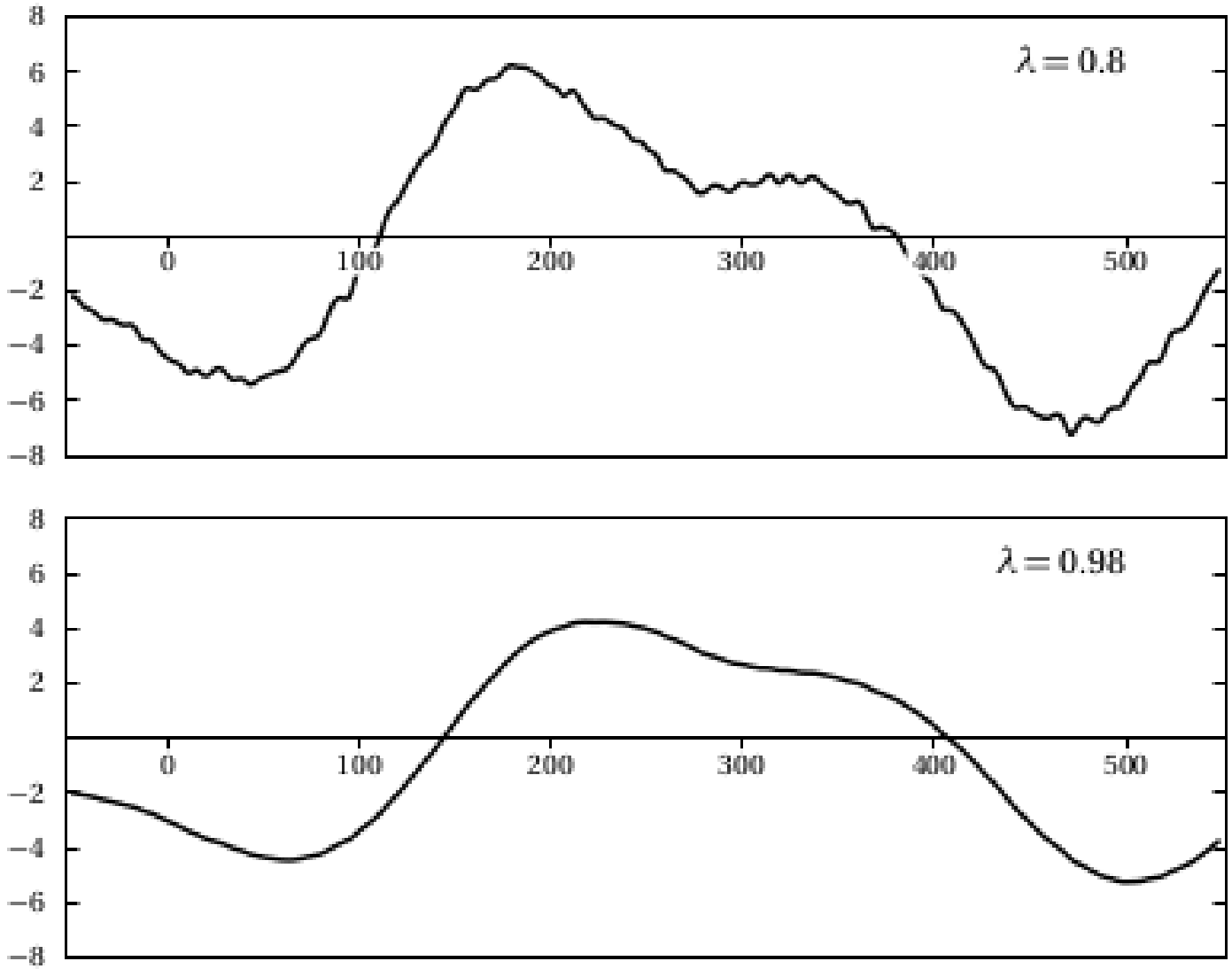
- No longer a convolution sum; instead, an instance of a *constant coefficient difference equation*.
- We need to set the initial conditions.
- The value of λ (which is the pole of the system) determines the smoothing power of the filter.

Noise Reduction

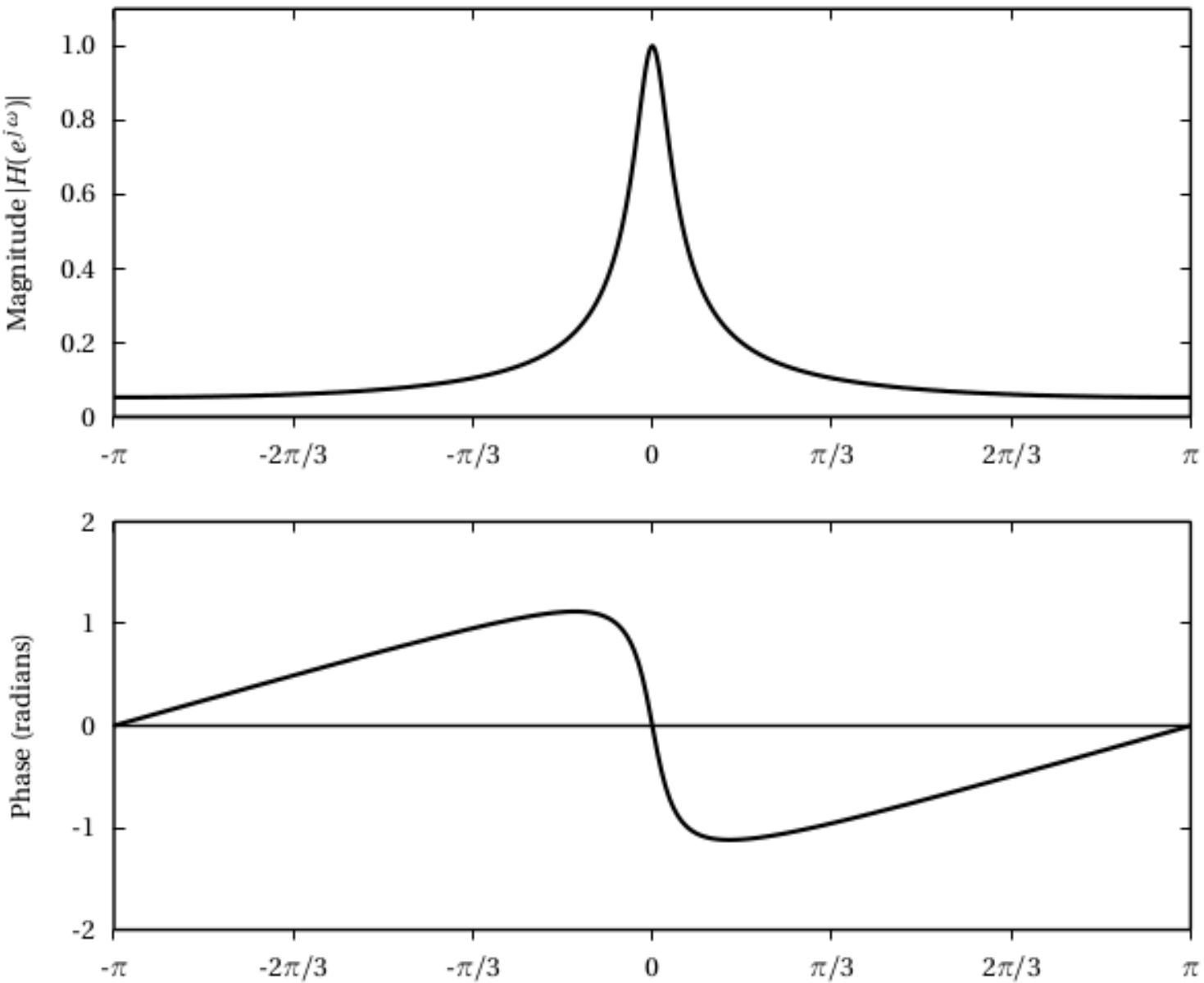


Noise Reduction

- Note how the signal is delayed as λ grows.

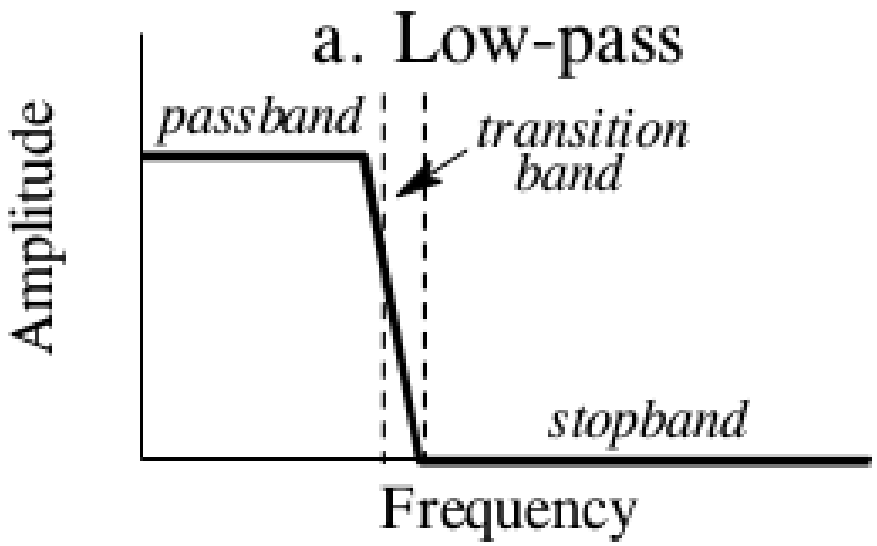


Frequency Response

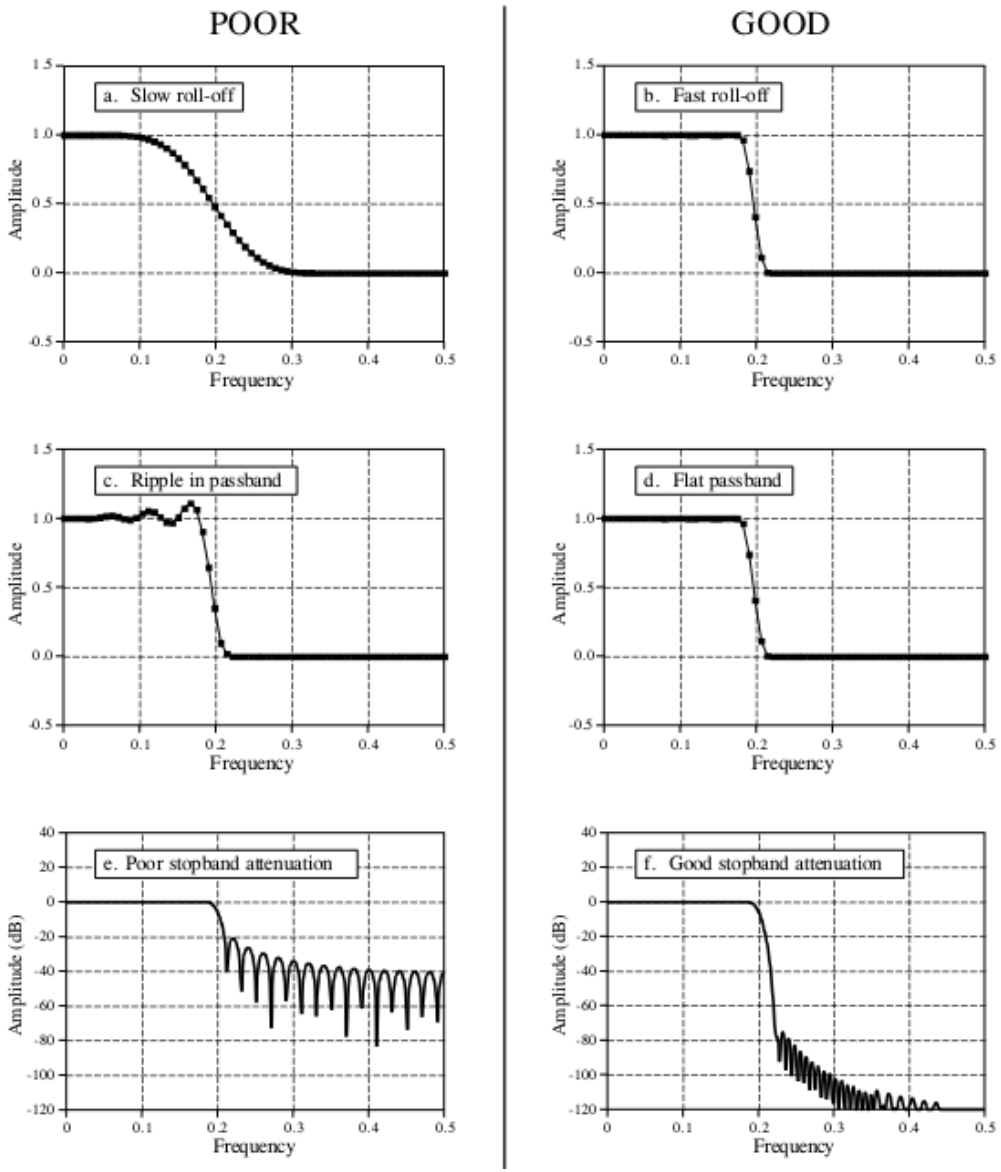


Frequency domain parameters

- Passband.
- Stopband.
- Transition band (fast roll-off).
- Passband ripple.
- Stopband ripple.



Frequency response parameters, cont'd



Strategy of filtering by windowing

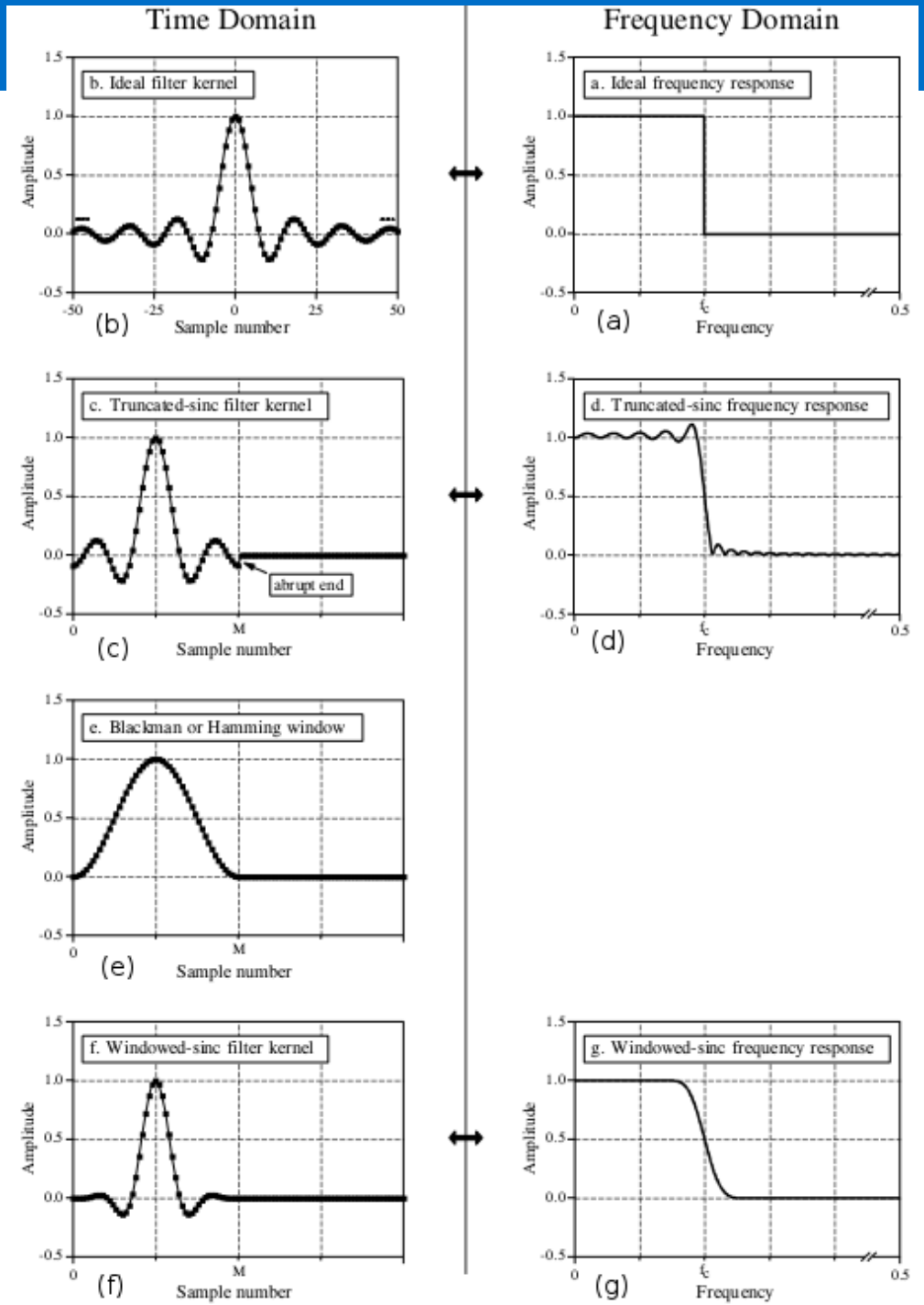
- Taking the Inverse Fourier Transform of an ideal frequency response (a) produces an ideal sinc filter kernel (b, impulse response).

$$hs[i] = \frac{\sin(2\pi f_C i)}{i\pi}$$

- Truncated-sinc (c) and frequency response (d). Discontinuity is significant no matter how long M is made (Gibbs phenomenon).

$$h[i] = hs[i] \cdot w[i]$$

- Multiplying the truncated-sinc (c) by the Blackman window (e) results in the windowed-sinc filter kernel (f) with frequency response (g).



Windows

Name of window function $w(n)$	Mathematical definition
Rectangular	1
Hanning	$0.5 - 0.5\cos\left[\frac{2\pi n}{N-1}\right]$
Hamming	$0.54 - 0.46\cos\left[\frac{2\pi n}{N-1}\right]$
Blackman	$0.42 - 0.5\cos\left[\frac{2\pi n}{N-1}\right] + 0.08\cos\left[\frac{2\pi n}{N-1}\right]$
Kaiser	$\frac{I_0\left[\beta\sqrt{1-\left(\frac{\lvert 2n-N+1\rvert}{N-1}\right)^2}\right]}{-I_o(\beta)}$ Where, $I_0(x) = \sum_{k=0}^{\infty}\left(\frac{x^k}{2^k\, k!}\right)^2$

Windows in time domain

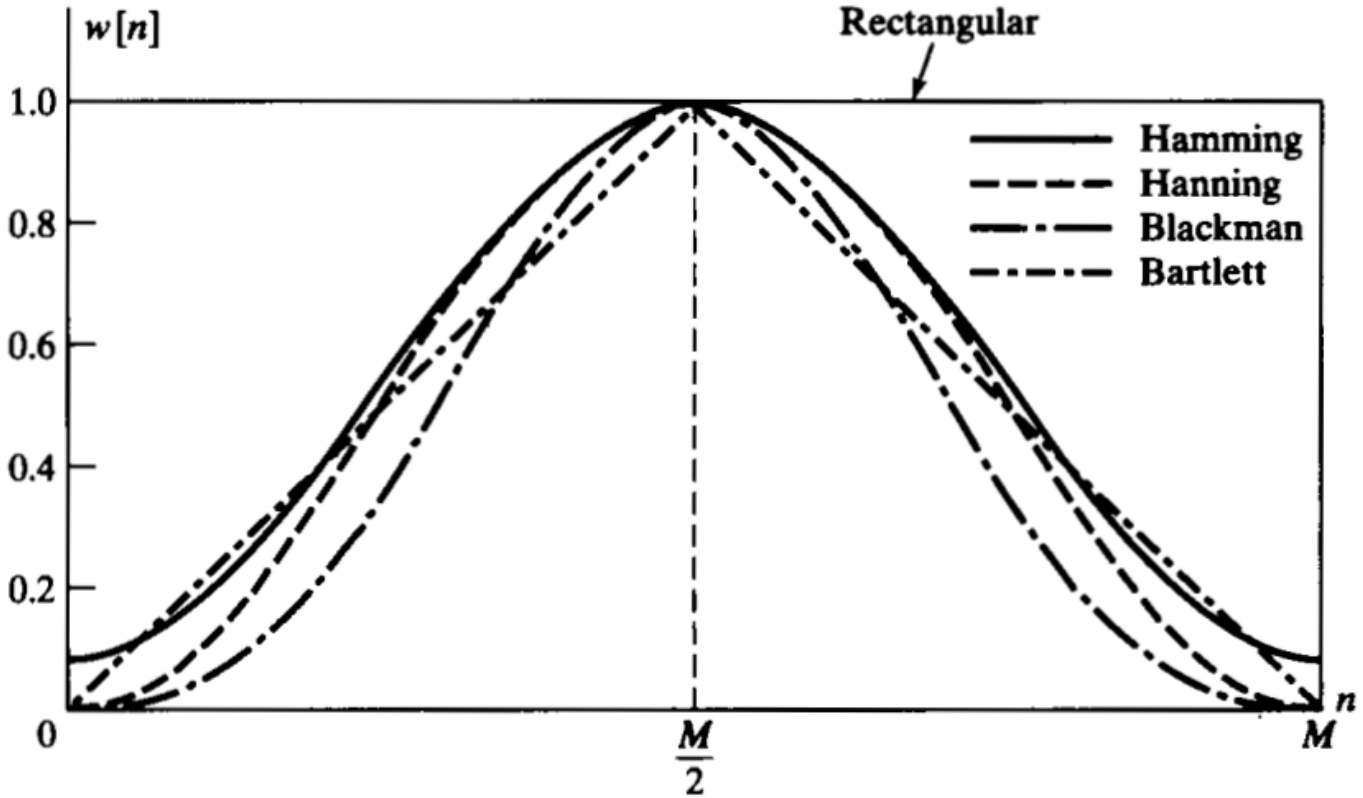


Figure 7.21 Commonly used windows.

Windows in frequency domain

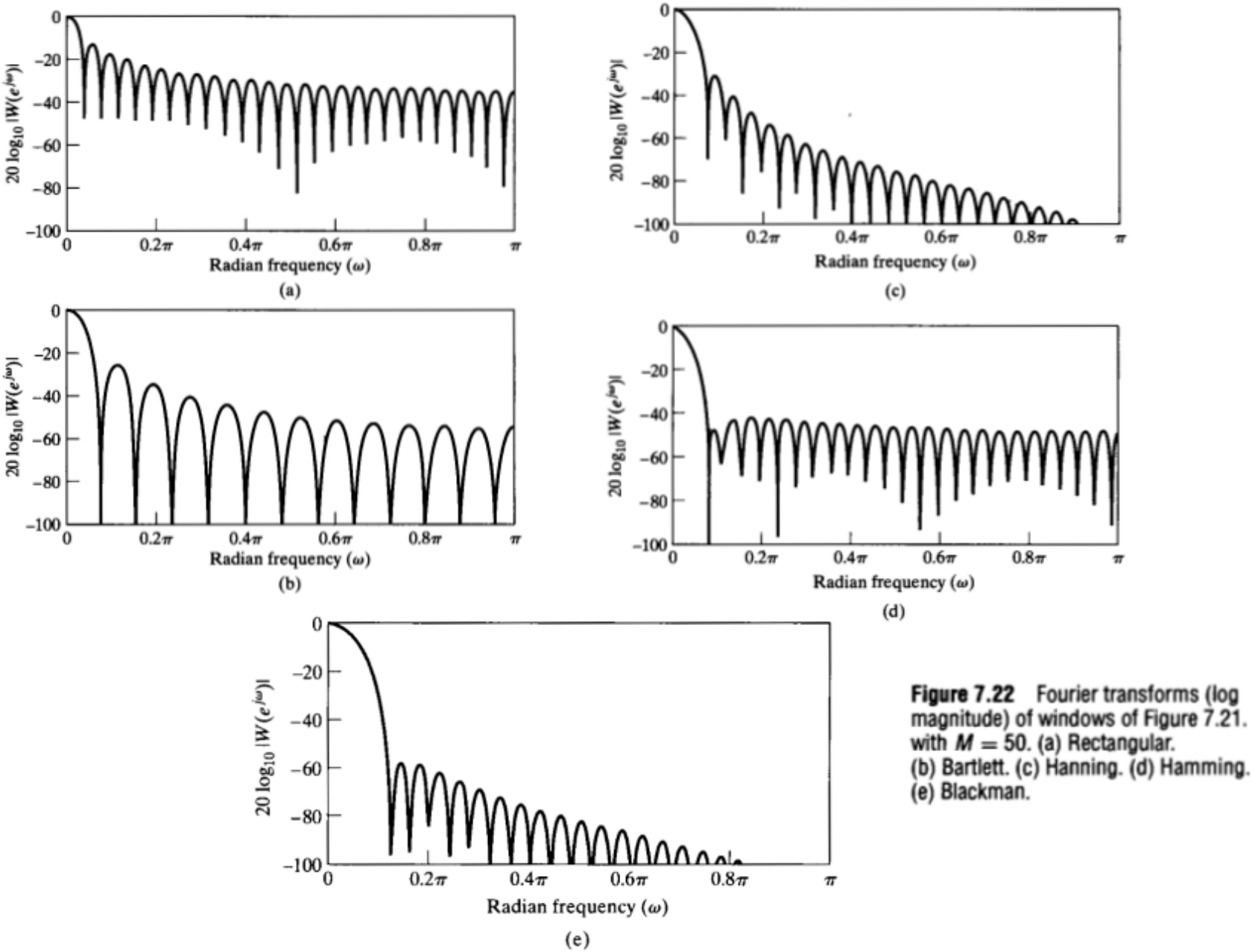
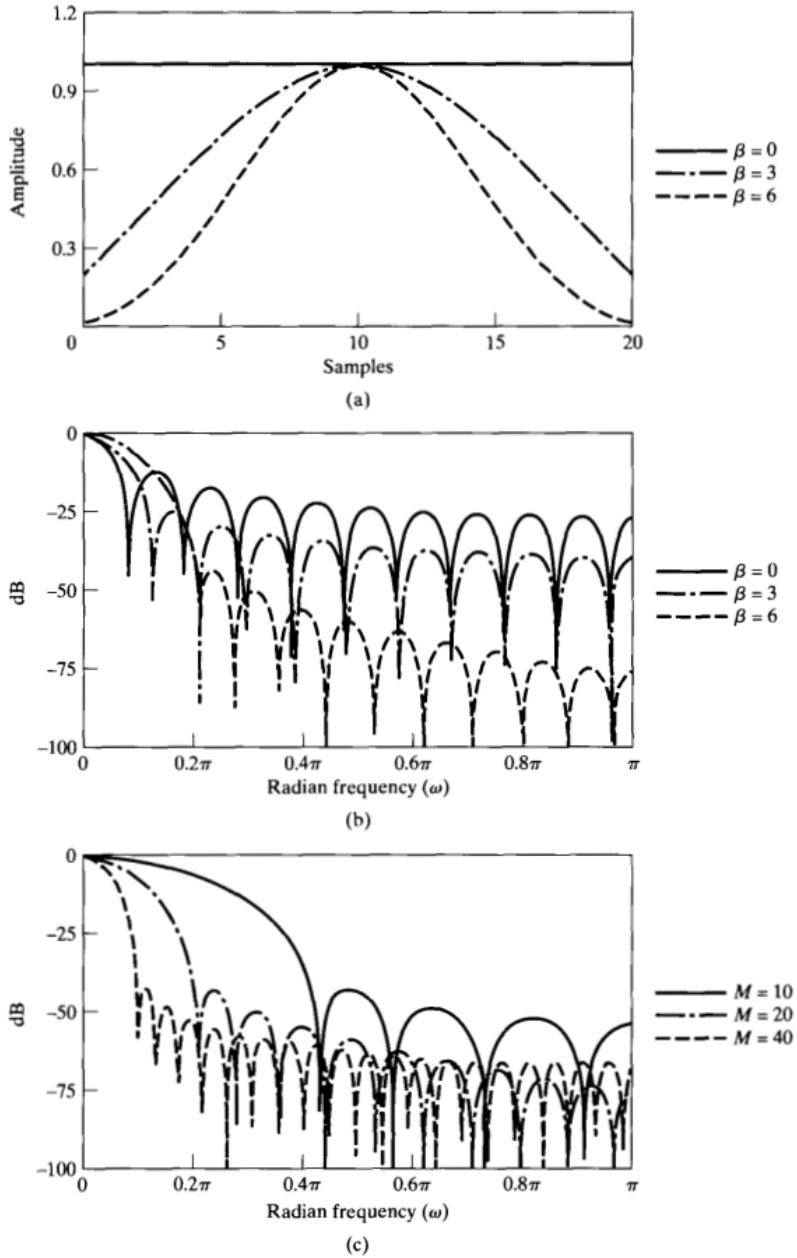


Figure 7.22 Fourier transforms (log magnitude) of windows of Figure 7.21. with $M = 50$. (a) Rectangular. (b) Bartlett. (c) Hanning. (d) Hamming. (e) Blackman.

Kaiser window filter

- The Kaiser window has two parameters:
  - Lenght, M+1.
  - Shape parameter, β.
- Trade-off between side-lobe amplitude and main-lobe width.

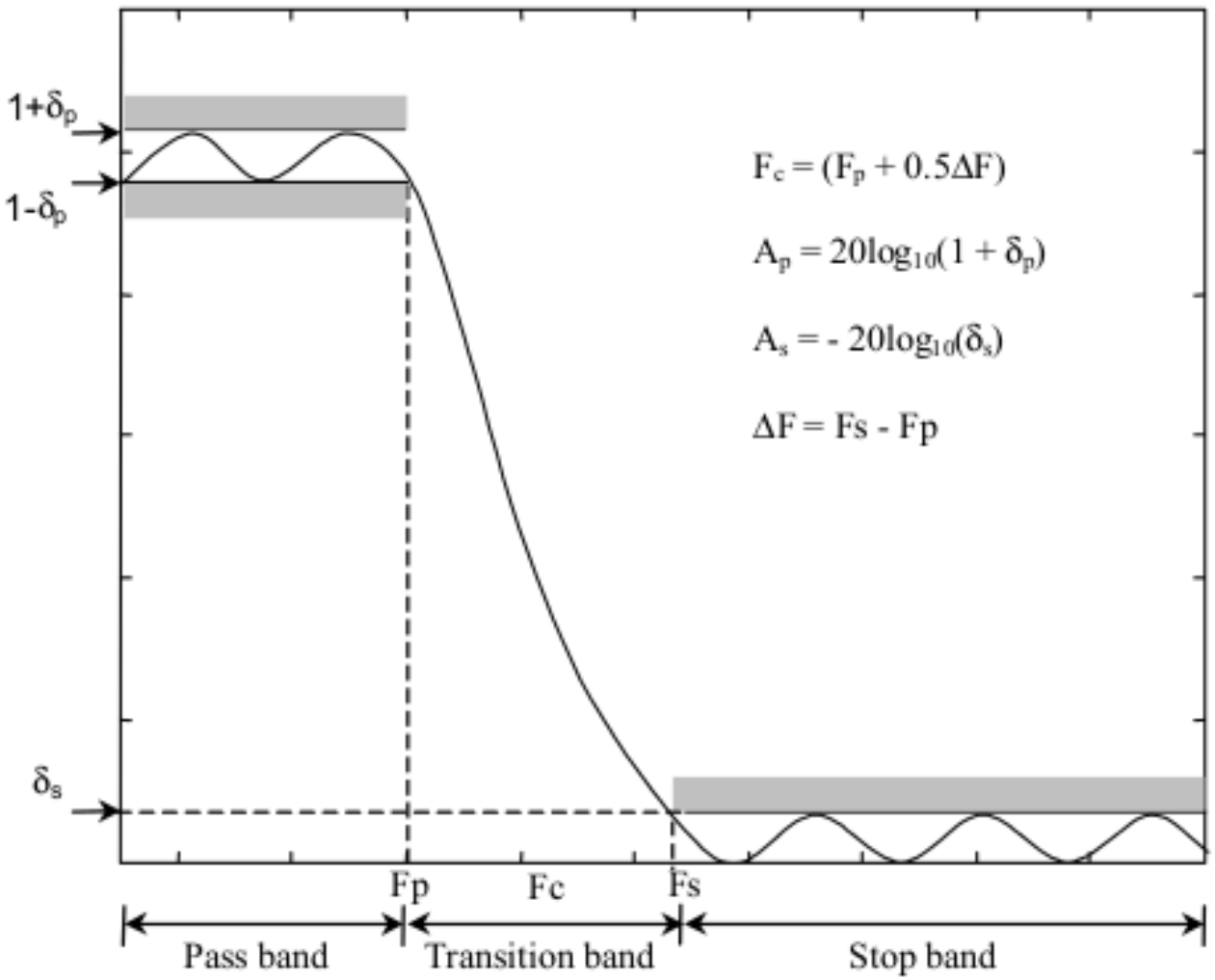


Kaiser window filter, cont'd

TABLE 7.1 COMPARISON OF COMMONLY USED WINDOWS

Type of Window	Peak Side-Lobe Amplitude (Relative)	Approximate Width of Main Lobe	Peak Approximation Error, $20\log_{10}\delta$ (dB)	Equivalent Kaiser Window, β	Transition Width of Equivalent Kaiser Window
Rectangular	−13	$4\pi/(M+1)$	−21	0	$1.81\pi/M$
Bartlett	−25	$8\pi/M$	−25	1.33	$2.37\pi/M$
Hanning	−31	$8\pi/M$	−44	3.86	$5.01\pi/M$
Hamming	−41	$8\pi/M$	−53	4.86	$6.27\pi/M$
Blackman	−57	$12\pi/M$	−74	7.04	$9.19\pi/M$

FIR filter design



FIR filter design, cont'd

Name of window function $w[n]$	Transition width ΔF in (Hz), (normalised)	Pass-band ripple A_p in (dB)	Ripple δ_p, δ_s	Side-lobe level in (dB)	Stop-band attenuation A_s in (dB)
Rectangular	0.9/N	0.741	0.089	-13	21
Hanning	3.1/N	0.0546	0.063	-31	44
Hamming	3.3/N	0.0194	0.0022	-41	53
Blackman	5.5/N	0.0017	0.000196	-57	74
Kaiser β=4.54	2.93/N	0.0274			50
β=5.65	3.63/N	0.00867			60
β=6.76	4.32/N	0.00275			70
β=8.96	5.71/N	0.000275			90

Example of FIR design

A FIR low-pass filter is required to have the following specifications:

1. Pass-band edge frequency $f_p = 2$ kHz
2. Transition band $\Delta f = 200$ Hz
3. Pass-band ripple $A_p = 0.1$dB
4. Minimum stop-band attenuation $A_s = 50$dB
5. Sampling frequency of $f_s = 10$ kHz

Example of FIR design, cont'd

Pass-band ripple, $A_p = 20\log_{10}(1 + \delta_p)$

$$\delta_p = \log_{10}^{-1}\left[\frac{0.1}{20}\right] - 1 = 0.0116$$

Minimum stop-band attenuation $A_s = -20\log_{10}(\delta_s)$

$$\delta_s = \log_{10}^{-1}\left[\frac{-50}{20}\right] = 0.00316$$

The normalised pass-band edge frequency

$$F_p = f_p / f_s = \frac{2x10^3}{10x10^3} = 0.2$$

The normalised transition width

$$\Delta F = \Delta f / f_s = \frac{200}{10x10^3} = 0.02$$

$$N = \frac{3.3}{0.02} = 165$$

FIR types

- M-tap FIR filter with impulse response $h[n]$ $n = 0, 1, \ldots, M-1$.
- We are interested in filters whose impulse response is symmetric or antisymmetric around the midpoint.
- The delay D introduced by these filters is equal to $(M-1)/2$ samples.
- D is an integer delay if M is odd.
- D is a fractional delay if M is even.

Type	Nb. of Taps	Symmetry	Delay	Phase
Type I	odd	symmetric	integer	$\alpha = 0$
Type II	even	symmetric	fractional	$\alpha = 0$
Type III	odd	antisymmetric	integer	$\alpha = \pi/2$
Type IV	even	antisymmetric	fractional	$\alpha = \pi/2$

Direct FIR form structures

$$H(z) = b_0 + b_1 z^{-1} + ... + b_{M-1} z^{M-1}$$

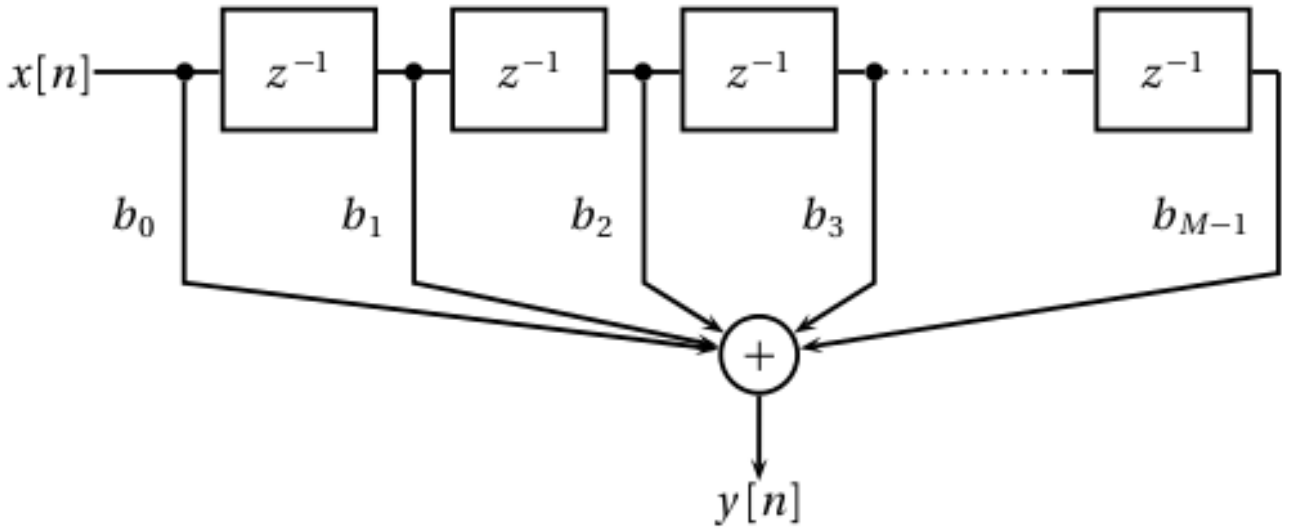


Figure 7.22 Direct FIR implementation.

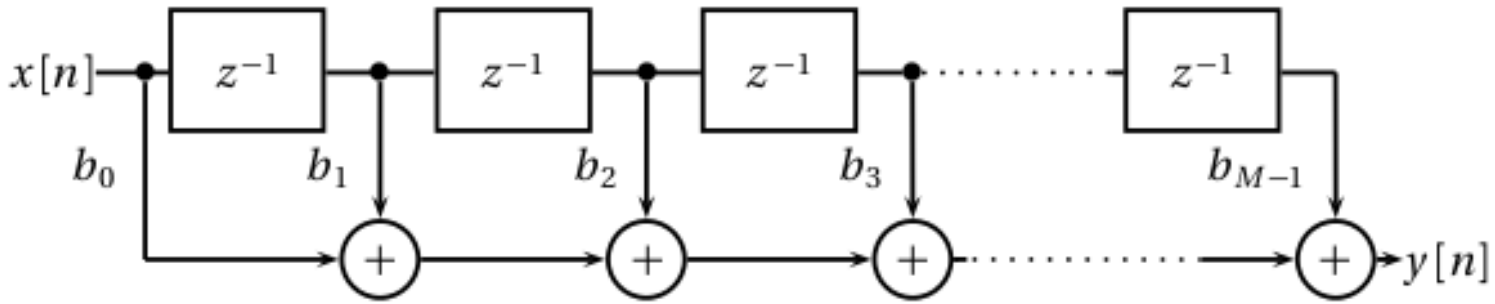


Figure 7.23 Transversal FIR implementation.

Bibliography

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3. Steven W. Smith, The Scientist and Engineer's Guide to Digital Signal Processing. Chapters 14, 15, and 16. www.dspguide.com
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