

## COM303: Digital Signal Processing

### Lecture 12: Filter design

# Overview

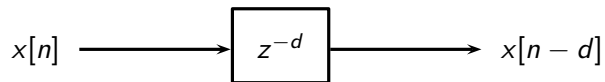
- ▶ two more ideal filters
- ▶ filter design: problem statement
- ▶ IIR design

a couple more ideal filters

# Overview

- ▶ the fractional delay
- ▶ the Hilbert filter

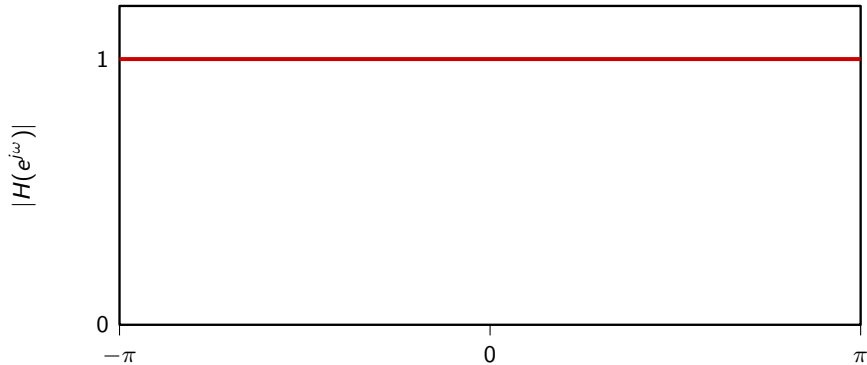
consider a simple delay...



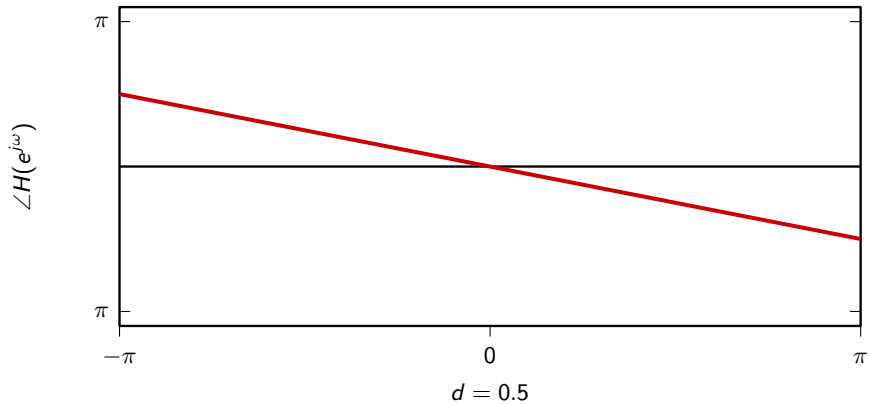
$$H(e^{j\omega}) = e^{-j\omega d} \quad d \in \mathbb{Z}$$

what happens if, in  $H(e^{j\omega})$  we use a non-integer  $d \in \mathbb{R}$ ?

## Fractional delay: magnitude response



## Fractional delay: phase response





## impulse response

$$\begin{aligned}h[n] &= \text{IDTFT} \left\{ e^{-j\omega d} \right\} \\&= \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{-j\omega d} e^{j\omega n} d\omega \\&= \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{j\omega(n-d)} d\omega \\&= \frac{1}{\pi(n-d)} \frac{e^{j\pi(n-d)} - e^{-j\pi(n-d)}}{2j} \\&= \frac{\sin \pi(n-d)}{\pi(n-d)} \\&= \text{sinc}(n-d)\end{aligned}$$

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## impulse response

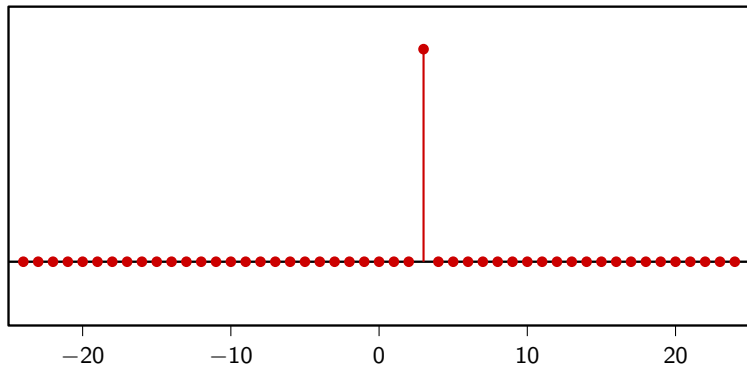
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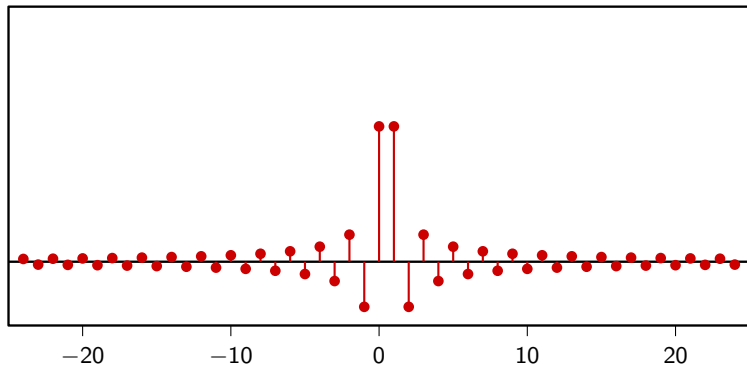
## Fractional delay: impulse response

$$d = 3$$



## Fractional delay: impulse response

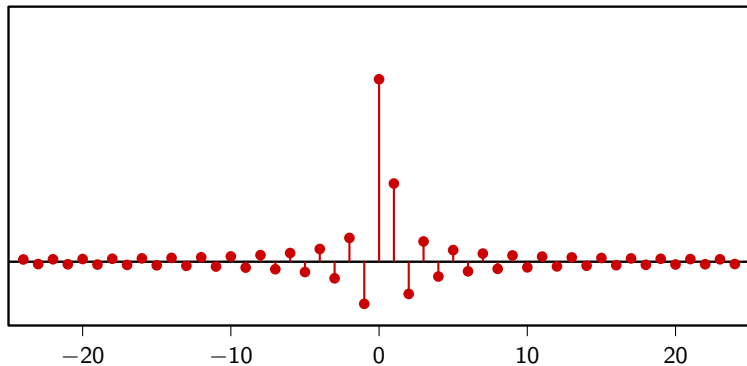
$$d = 0.5$$





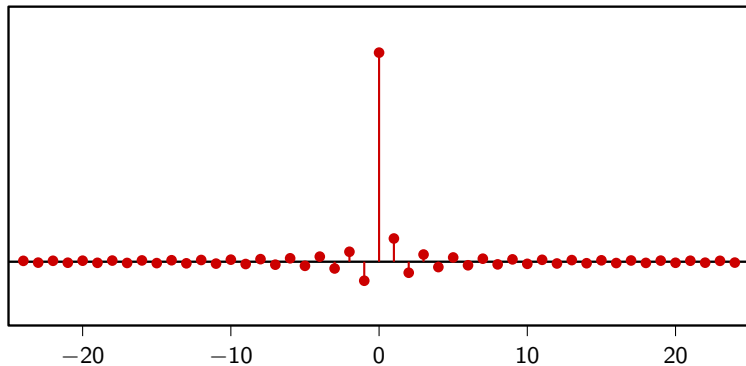
## Fractional delay: impulse response

$$d = 0.3$$



## Fractional delay: impulse response

$$d = 0.1$$

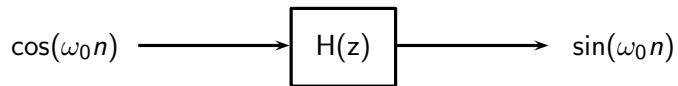


## fractional delay

- ▶ fractional delay computes “in-between” values for samples
- ▶ it is an ideal filter!
- ▶ often approximated with local interpolation
- ▶ all will be clear when we study the sampling theorem

the Hilbert filter

## a quirky machine



can we build such a thing?

in the frequency domain

$$H(e^{j\omega})[\tilde{\delta}(\omega - \omega_0) + \tilde{\delta}(\omega + \omega_0)] = -j[\tilde{\delta}(\omega - \omega_0) - \tilde{\delta}(\omega + \omega_0)]$$

we can derive two values:

$$\begin{cases} H(e^{j\omega_0}) &= -j \\ H(e^{-j\omega_0}) &= +j \end{cases}$$

in the frequency domain

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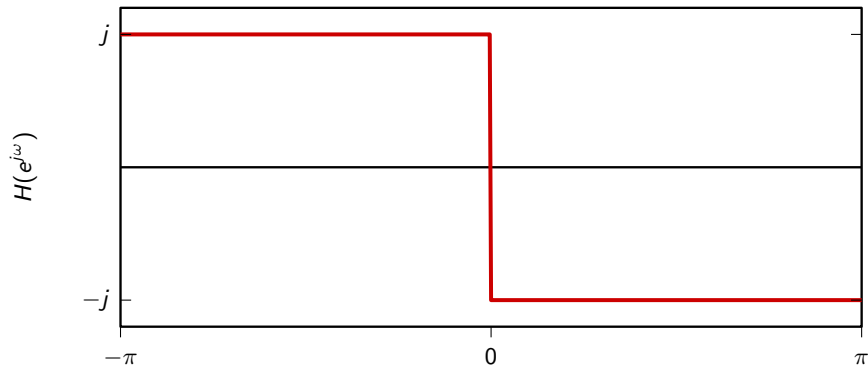
in the frequency domain

for the machine to work at all frequencies:

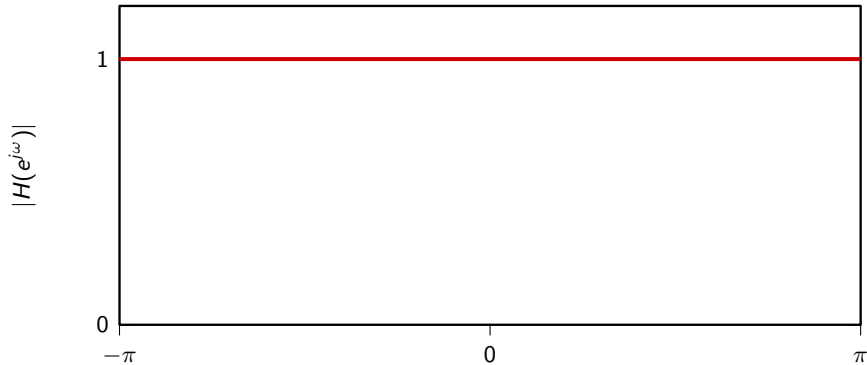
$$H(e^{j\omega_0}) = \begin{cases} -j & \text{for } 0 \leq \omega < \pi \\ +j & \text{for } -\pi \leq \omega < 0 \end{cases} \quad (2\pi\text{-periodic})$$



# Hilbert filter



## Hilbert filter is an allpass



## impulse response

$$\begin{aligned}h[n] &= \frac{1}{2\pi} \int_{-\pi}^0 j e^{j\omega n} d\omega + \frac{1}{2\pi} \int_0^{\pi} -j e^{j\omega n} d\omega \\&= \frac{1}{2\pi n} [1 - e^{-j\pi n} - e^{-j\pi n} + 1] \\&= \begin{cases} \frac{2}{\pi n} & n \text{ odd} \\ 0 & n \text{ even} \end{cases}\end{aligned}$$

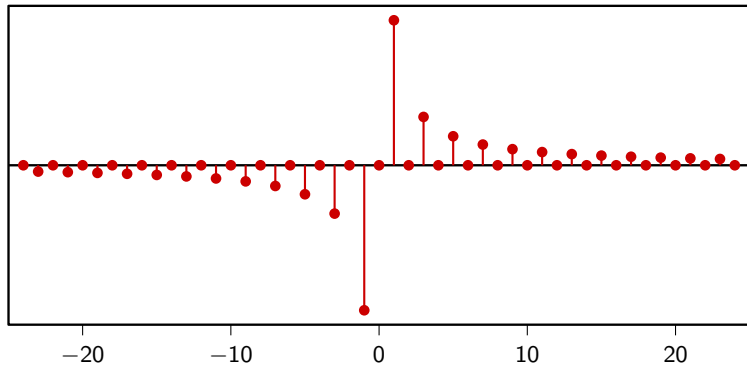
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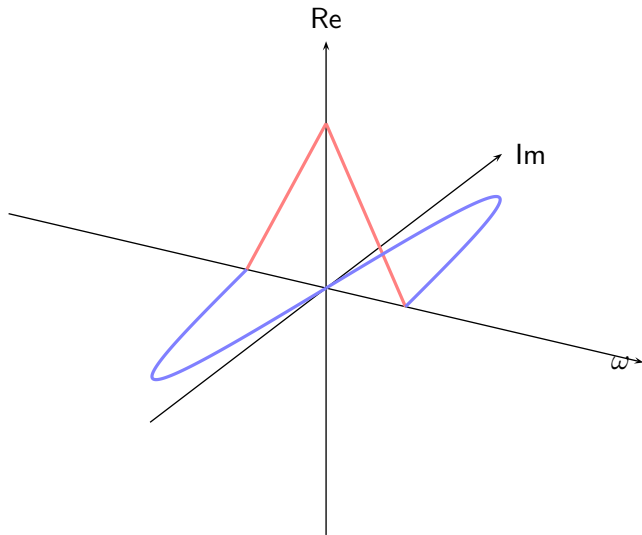
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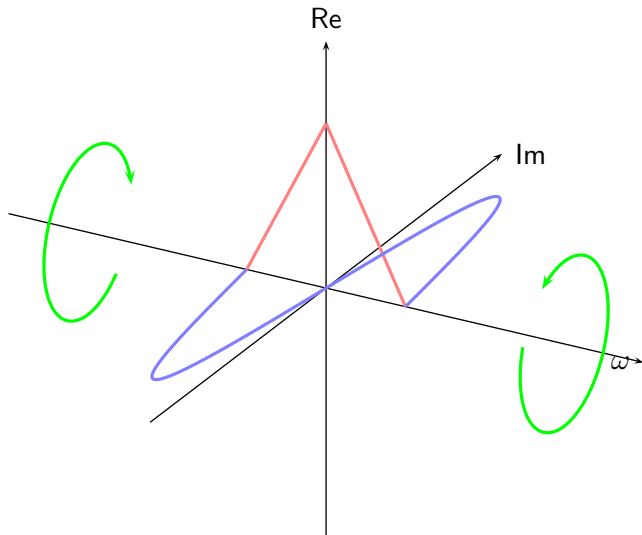
## Hilbert filter



what does the Hilbert filter do?

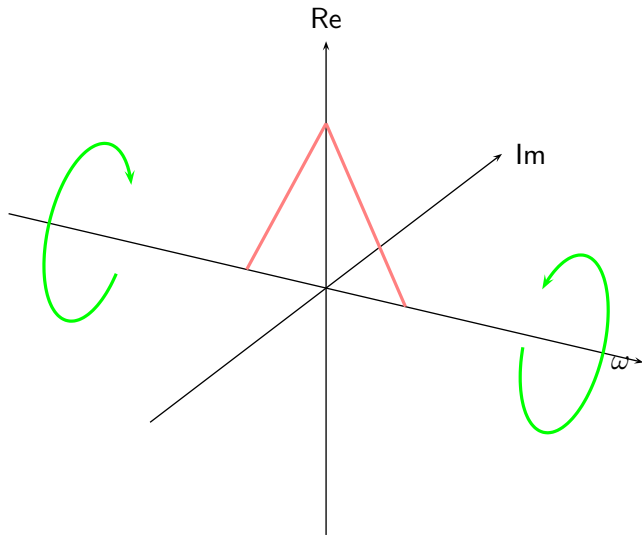


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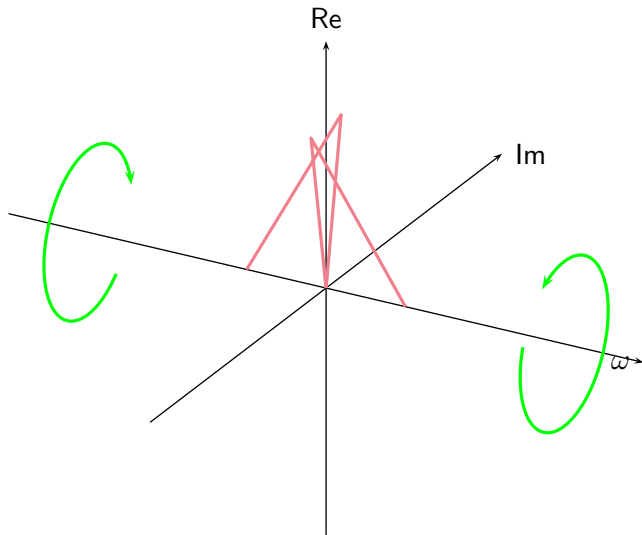




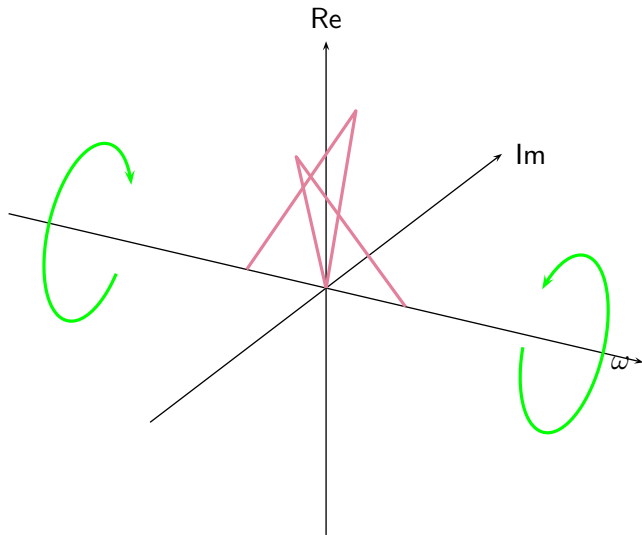
## effect of the Hilbert filter (real part)



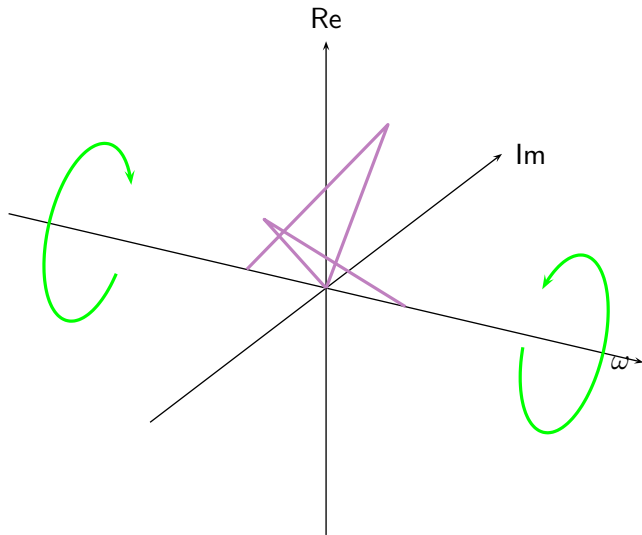
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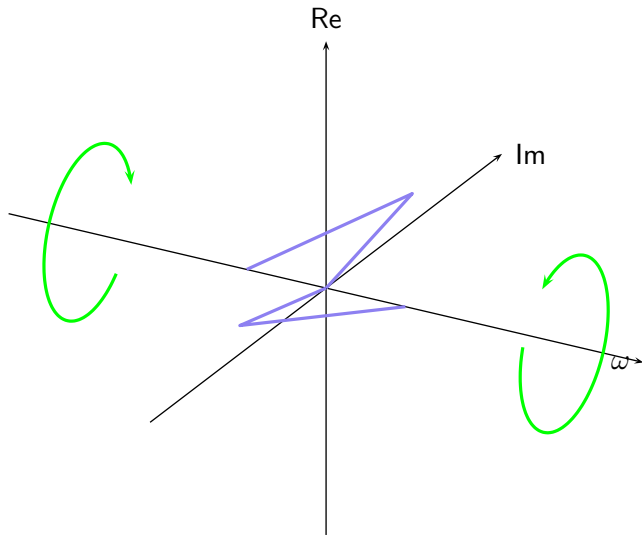
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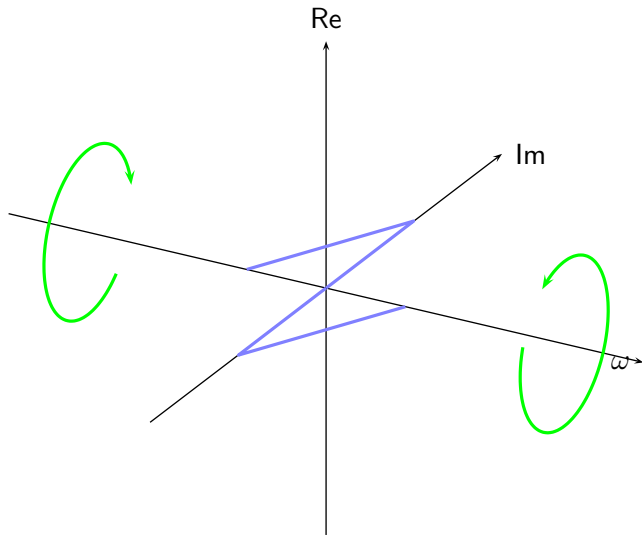
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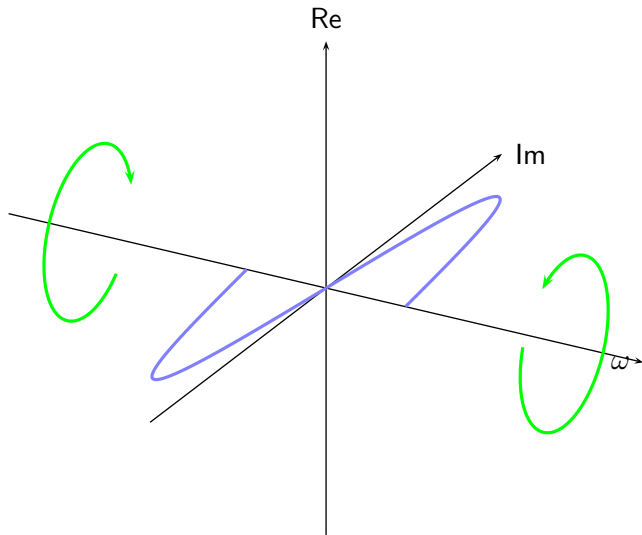
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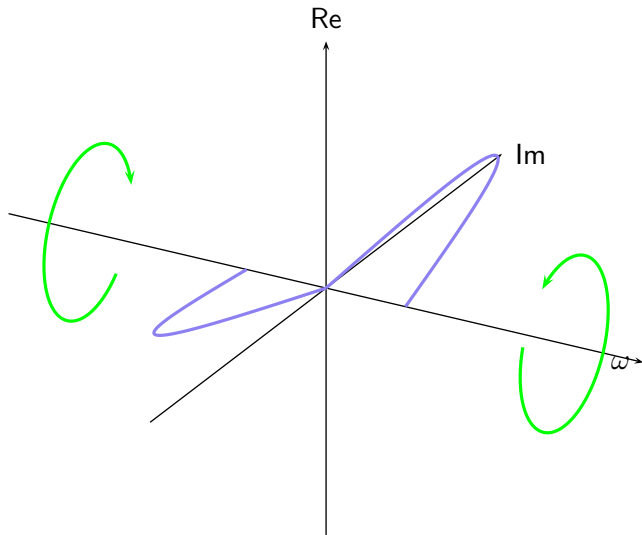
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## effect of the Hilbert filter (imaginary part)

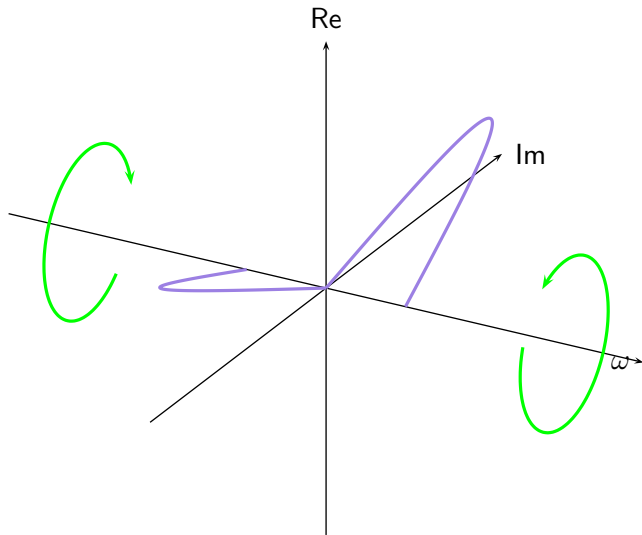


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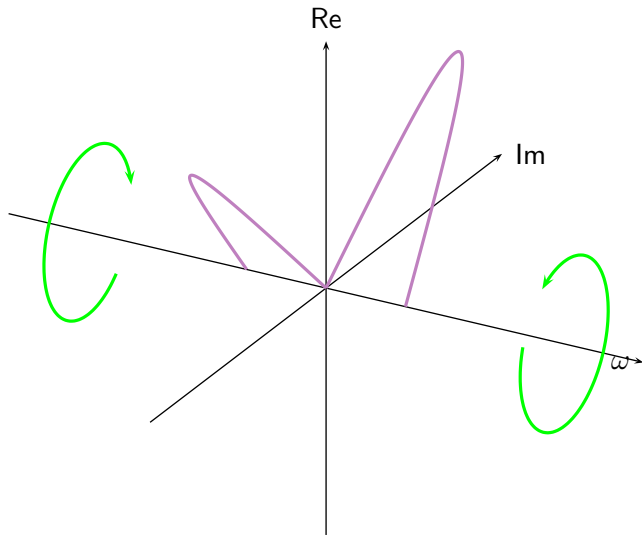




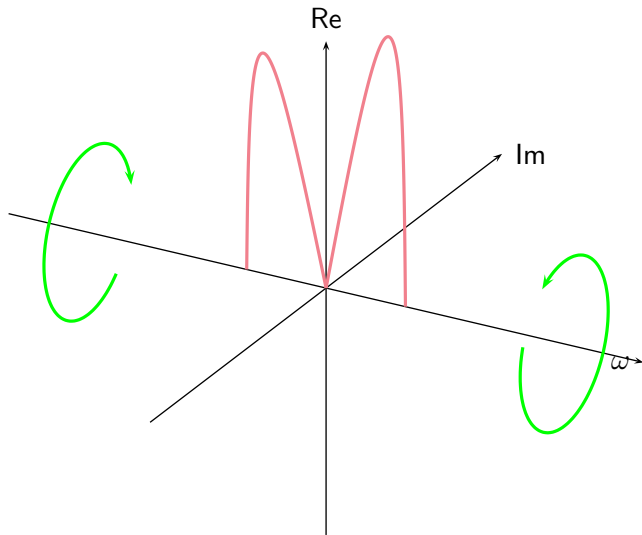
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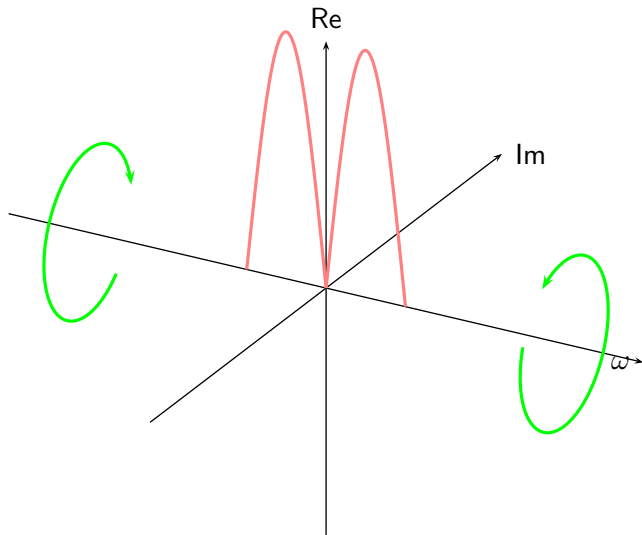
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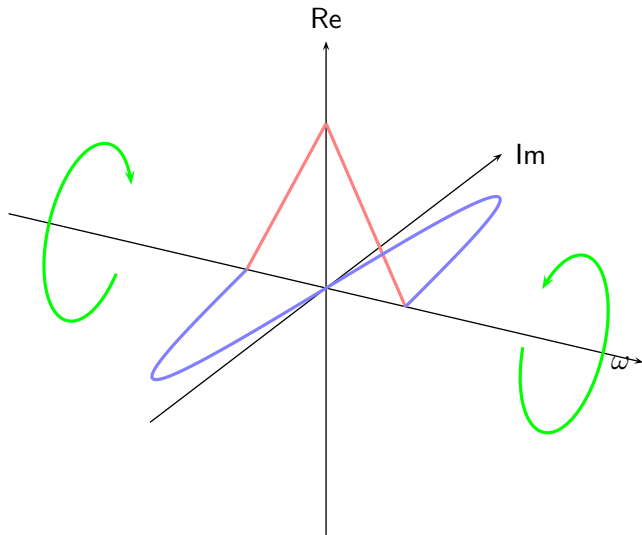
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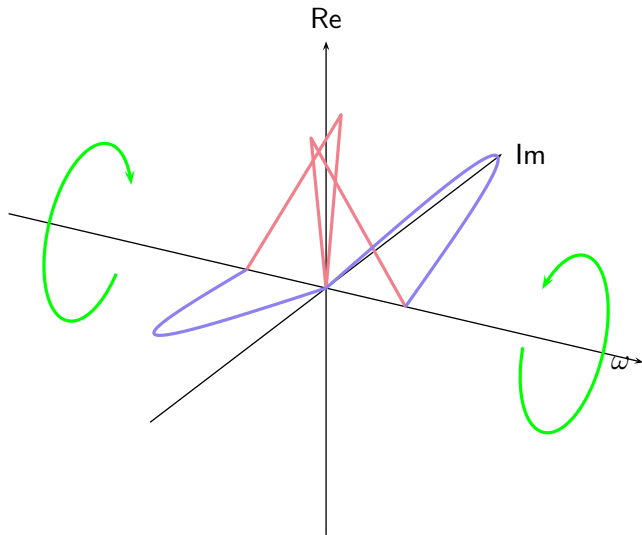
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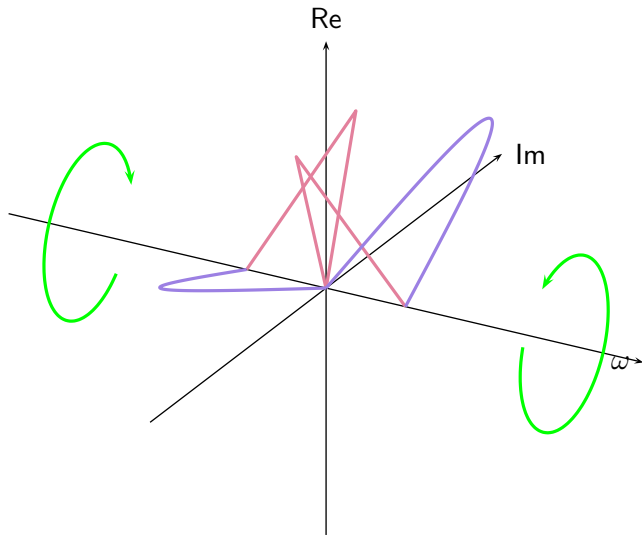
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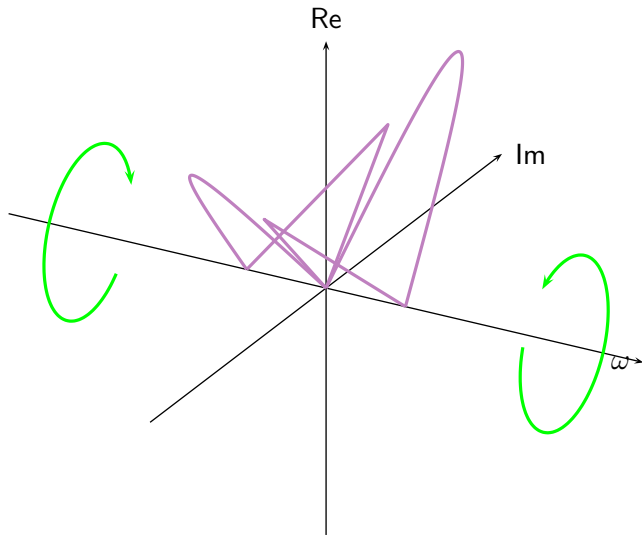
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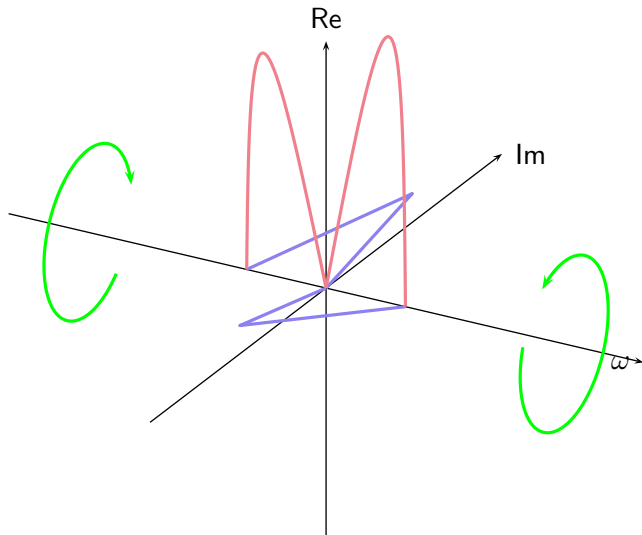


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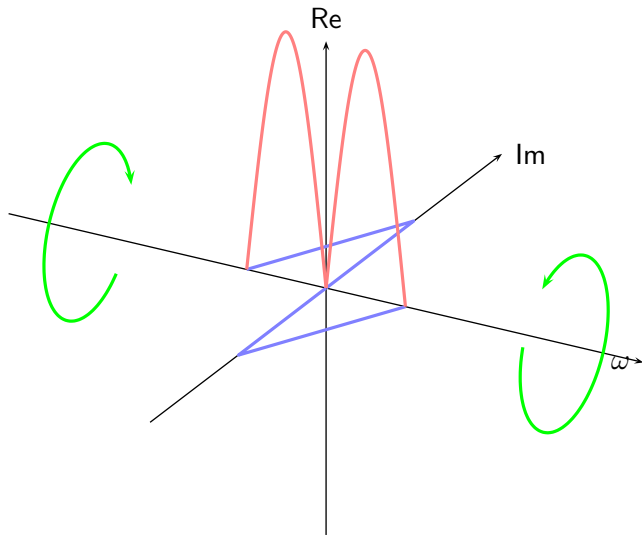




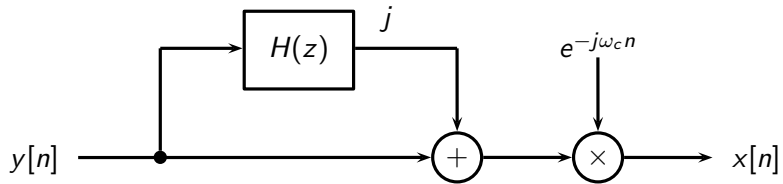
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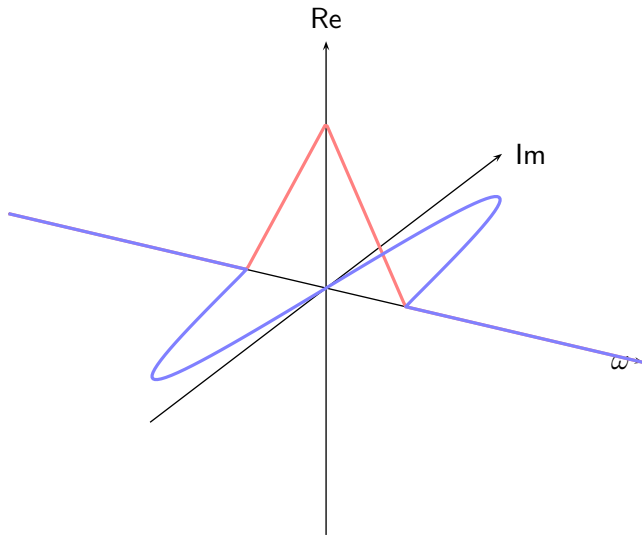


## Hilbert demodulation



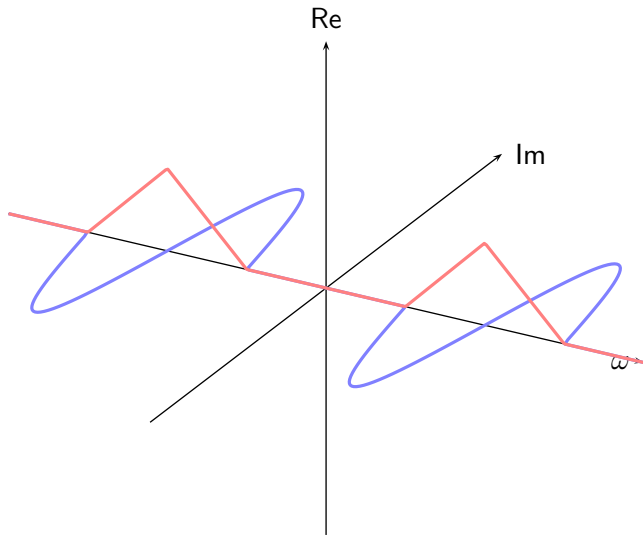
# Hilbert demodulation

$x[n]$



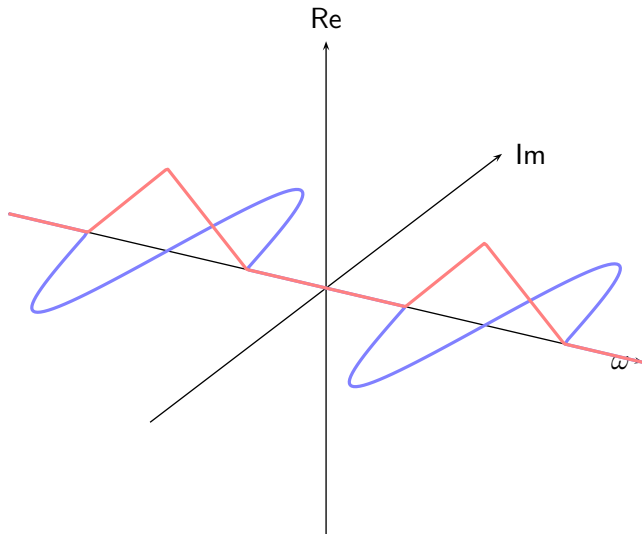
# Hilbert demodulation

$$x[n] \cos(\omega_0 n) = y[n]$$



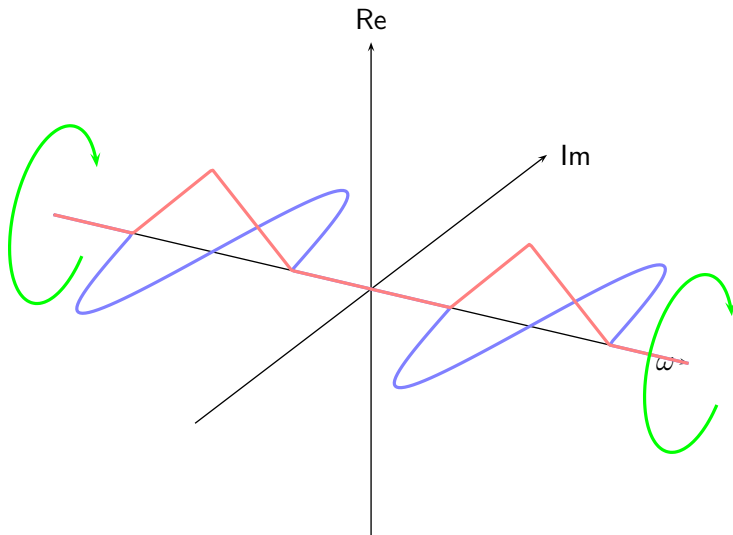
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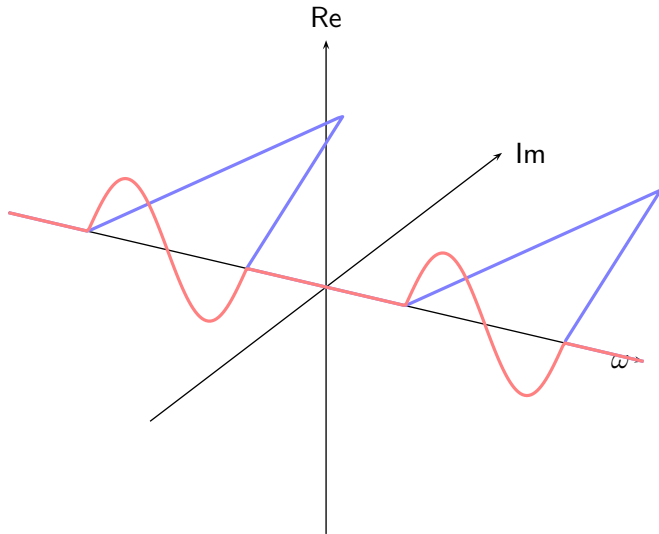
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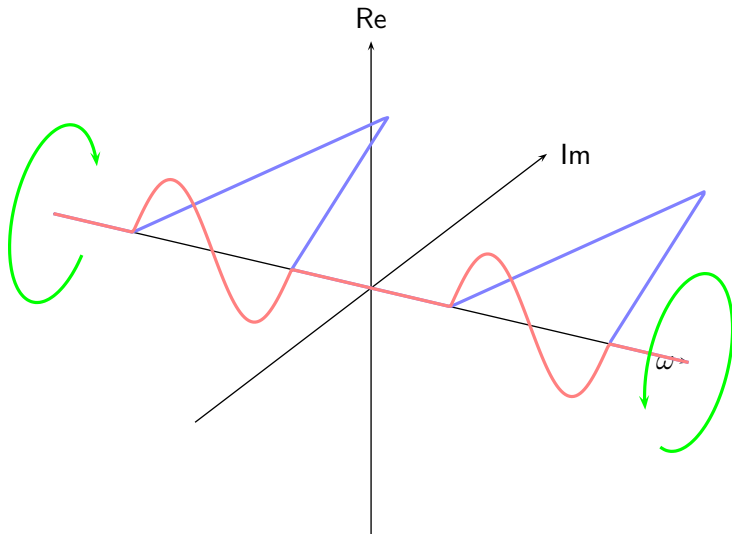
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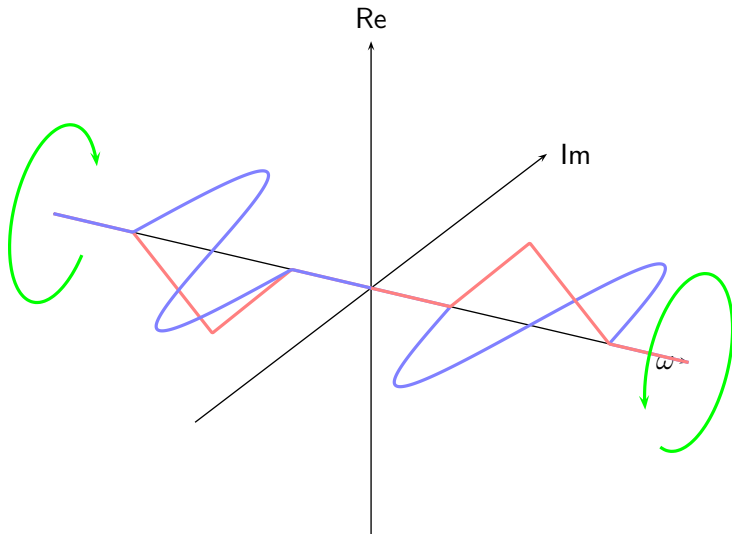
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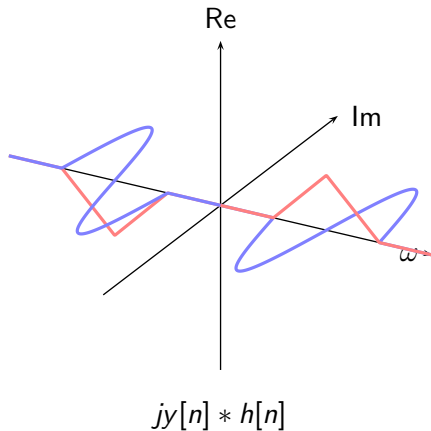
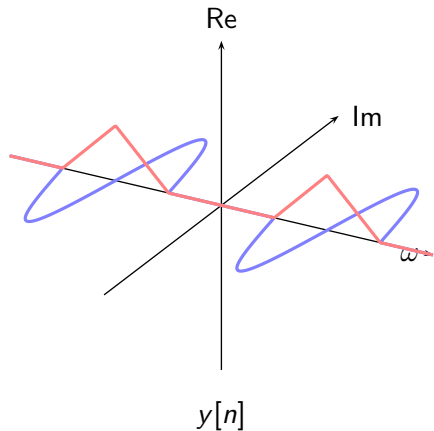


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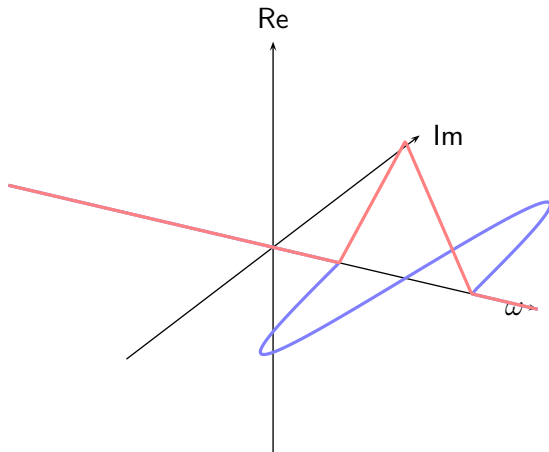
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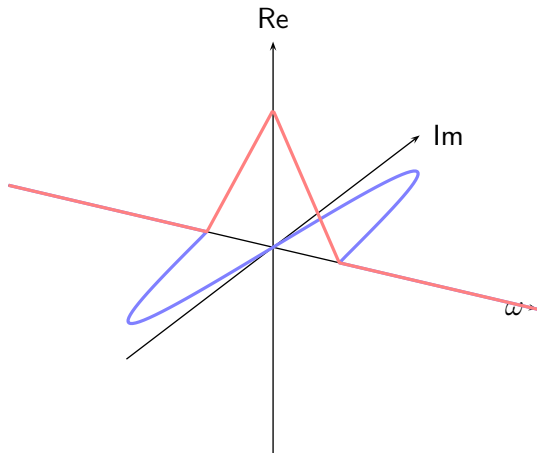
## Hilbert demodulation: $jy[n] * h[n] + y[n]$



Hilbert demodulation:  $jy[n] * h[n] + y[n] = x[n]e^{j\omega_0 n}$



Hilbert demodulation:  $(jy[n] * h[n] + y[n])e^{-j\omega_0 n}$



filter design

# The filter design problem

You are given a set of requirements:

- ▶ frequency response: passband(s) and stopband(s)
- ▶ phase: overall delay, linearity
- ▶ some limit on computational resources and/or numerical precision

You must determine  $N$ ,  $M$ ,  $a_k$ 's and  $b_k$ 's in

$$H(z) = \frac{b_0 + b_1 z^{-1} + \dots + b_{M-1} z^{-(M-1)}}{a_0 + a_1 z^{-1} + \dots + a_{N-1} z^{-(N-1)}}$$

in order to best fulfill the requirements

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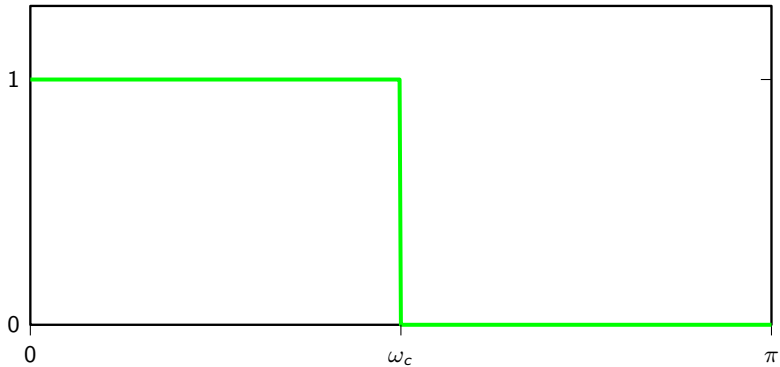
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## Example: lowpass specs



# Practical limitations

- ▶ passband/stopband transitions cannot be infinitely sharp  
⇒ use *transition bands*
- ▶ magnitude response cannot be constant over an interval  
⇒ specify *magnitude tolerances over bands*
- ▶ in general:
  - smaller transition bands  $\Rightarrow$  higher filter order
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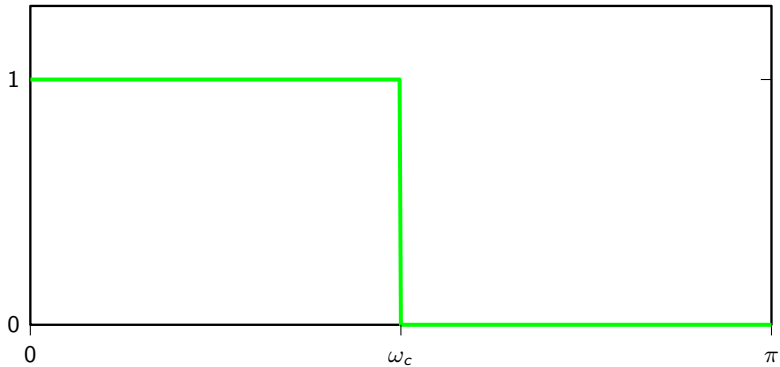
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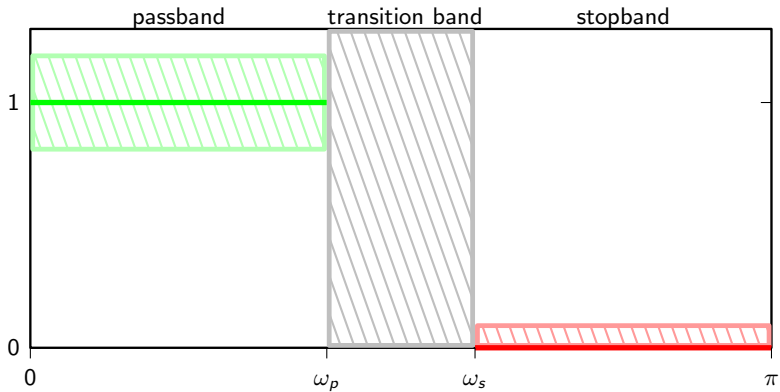
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## Example: lowpass specs



## Realistic specs





## Why we can't have a “vertical” transition

$$H(z) = B(z)/A(z) \text{ is } C^\infty$$

## Why we can't have a flat response

$$H(z) = B(z)/A(z), \quad \text{with } A \text{ and } B \text{ polynomials}$$

$$\begin{aligned} H(e^{j\omega}) = c \text{ over an interval} &\Rightarrow B(z) - cA(z) = 0 \text{ over an interval} \\ &\Rightarrow B(z) - cA(z) \text{ has an infinite number of roots} \\ &\Rightarrow B(z) - cA(z) = 0 \text{ for all values of } z \\ &\Rightarrow H(e^{j\omega}) = c \text{ over the entire } [-\pi, \pi] \text{ interval.} \end{aligned}$$

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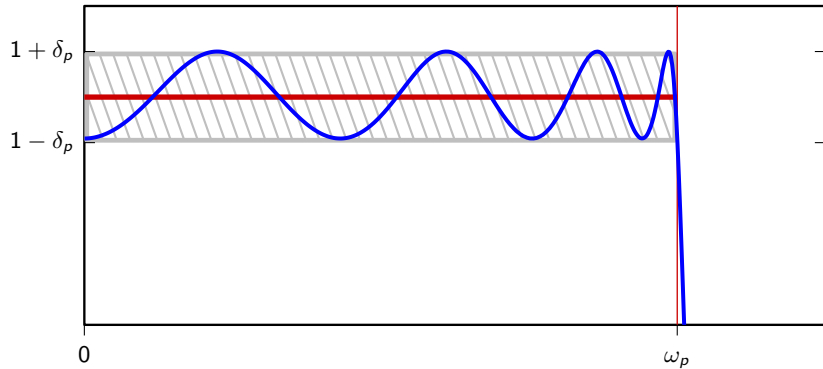
$$\begin{aligned} H(e^{j\omega}) = c \text{ over an interval} &\Rightarrow B(z) - cA(z) = 0 \text{ over an interval} \\ &\Rightarrow B(z) - cA(z) \text{ has an infinite number of roots} \\ &\Rightarrow B(z) - cA(z) = 0 \text{ for all values of } z \\ &\Rightarrow H(e^{j\omega}) = c \text{ over the entire } [-\pi, \pi] \text{ interval.} \end{aligned}$$

## Why we can't have a flat response

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## Important case: equiripple error



# The big questions

- ▶ IIR or FIR?
- ▶ how to determine the coefficients?
- ▶ how to evaluate the performance?



## IIRs: pros and cons

### Pros:

- ▶ computationally efficient
- ▶ strong attenuation easy
- ▶ good for audio

### Cons:

- ▶ stability issues
- ▶ difficult to design for arbitrary response
- ▶ nonlinear phase

# FIRs: pros and cons

## Pros:

- ▶ always stable
- ▶ optimal design techniques exist
- ▶ can be designed with linear phase

## Cons:

- ▶ computationally much more expensive
- ▶ may “sound” harsh

# The design methods

- ▶ finding  $N$ ,  $M$ ,  $a_k$ 's and  $b_k$ 's from specs is a hard nonlinear problem
- ▶ established methods:
  - IIR: conversion of analog design
  - FIR: optimal minimax filter design

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IIR design

## IIR: conversion of analog design

Filter design was an established art long before digital processing appeared

- ▶ lots of nice analog filters exist
- ▶ methods exist to “translate” the analog design into a rational transfer function
- ▶ most numerical packages (Matlab, etc.) provide ready-made routines
- ▶ design involves specifying some parameters and testing that the specs are fulfilled



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# Butterworth lowpass

Magnitude response:

- ▶ maximally flat
- ▶ monotonic over  $[0, \pi]$

Design parameters:

- ▶ order  $N$  ( $N$  poles and  $N$  zeros)
- ▶ cutoff frequency

Design test criterion:

- ▶ width of transition band
- ▶ passband error

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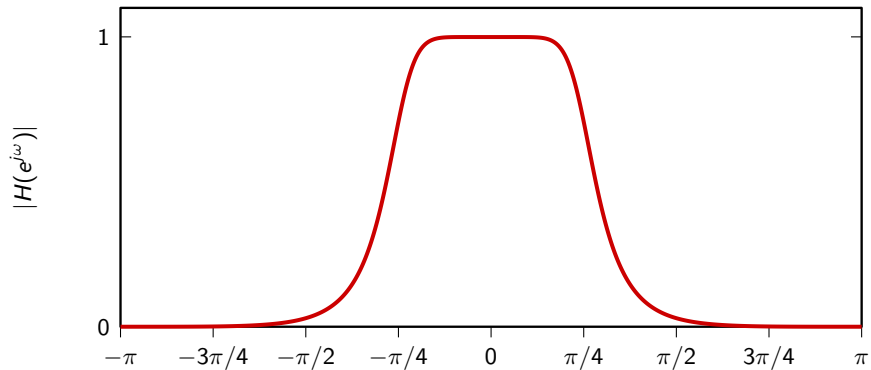
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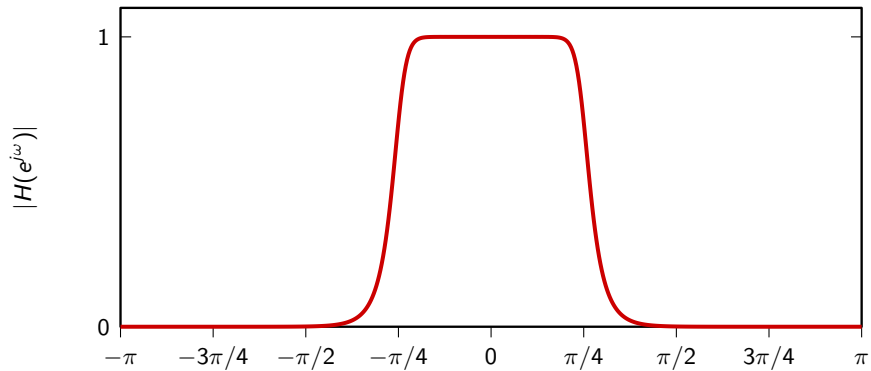
## Butterworth lowpass example

$$N = 4, \omega_c = \pi/4$$



## Butterworth lowpass example

$$N = 8, \omega_c = \pi/4$$



# Chebyshev lowpass

Magnitude response:

- ▶ equiripple in passband
- ▶ monotonic in stopband

Design parameters:

- ▶ order  $N$  ( $N$  poles and  $N$  zeros)
- ▶ passband max error
- ▶ cutoff frequency

Design test criterion:

- ▶ width of transition band
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# Chebyshev lowpass

Magnitude response:

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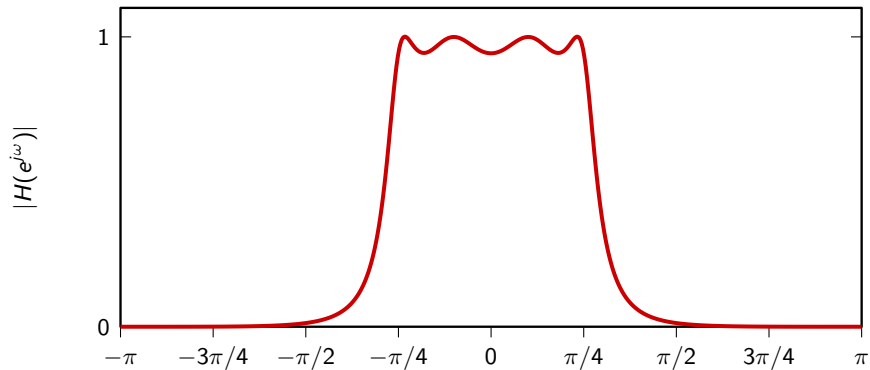
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Design test criterion:

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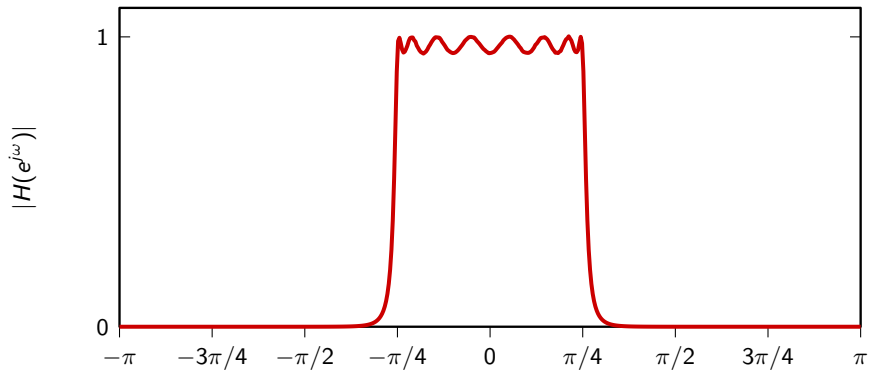
## Chebyshev lowpass example

$$N = 4, \omega_c = \pi/4, e_{\max} = 12\%$$



## Chebyshev lowpass example

$$N = 8, \omega_c = \pi/4, e_{\max} = 12\%$$



# Elliptic lowpass

Magnitude response:

- ▶ equiripple in passband and stopband

Design parameters:

- ▶ order  $N$
- ▶ cutoff frequency
- ▶ passband max error
- ▶ stopband min attenuation

Design test criterion:

- ▶ width of transition band

# Elliptic lowpass

Magnitude response:

- ▶ equiripple in passband and stopband

Design parameters:

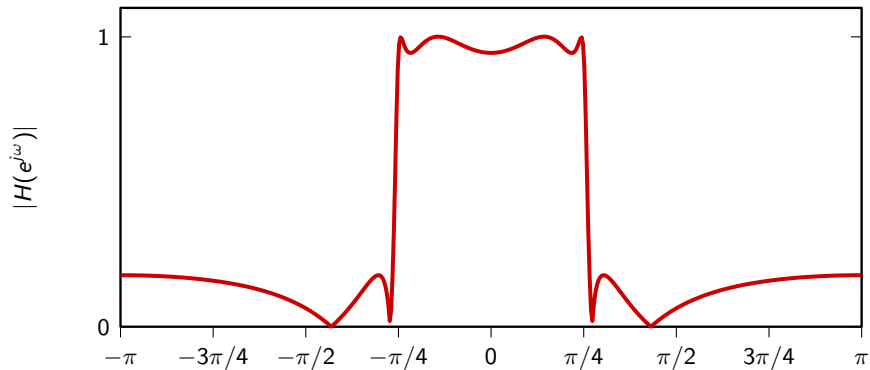
- ▶ order  $N$
- ▶ cutoff frequency
- ▶ passband max error
- ▶ stopband min attenuation

Design test criterion:

- ▶ width of transition band

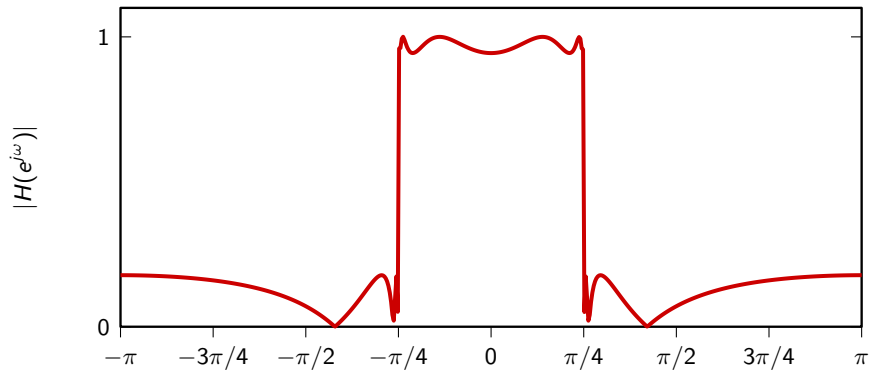
## Elliptic lowpass example

$$N = 4, \omega_c = \pi/4, e_{\max} = 12\%, \text{att}_{\min} = 0.03$$



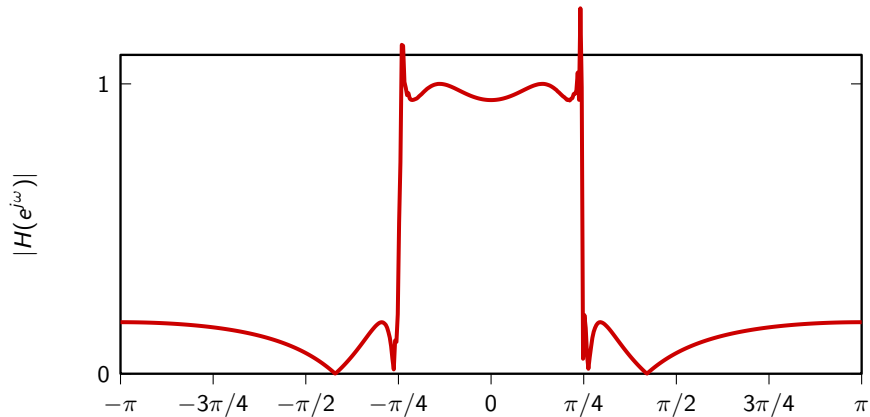
## Elliptic lowpass example

$$N = 6, \omega_c = \pi/4, e_{\max} = 12\%, \text{att}_{\min} = 0.03$$



## Elliptic lowpass example

$$N = 8, \omega_c = \pi/4, e_{\max} = 12\%, \text{att}_{\min} = 0.03$$





# Magnitude response in decibels

- ▶ filter max passband magnitude  $G$
- ▶ filter attenuation expressed in decibels as:

$$A_{\text{dB}} = 20 \log_{10}(|H(e^{j\omega})|/G)$$

- ▶ useful to compare attenuations between filters

## 4-th order lowpass comparison

