ESE 531: Digital Signal Processing

Lecture 10: February 15, 2022

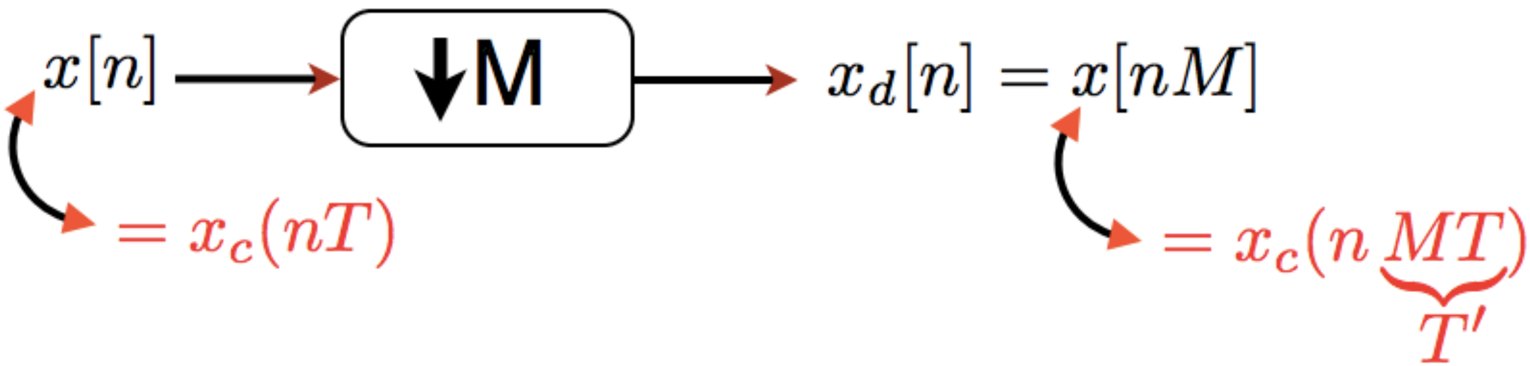
Non-Integer and Multi-rate Sampling

Lecture Outline

- Review: Downsampling/Upsampling
- Non-integer Resampling
- Multi-Rate Processing
  - Interchanging Operations

Downsampling

- Definition: Reducing the sampling rate by an integer number

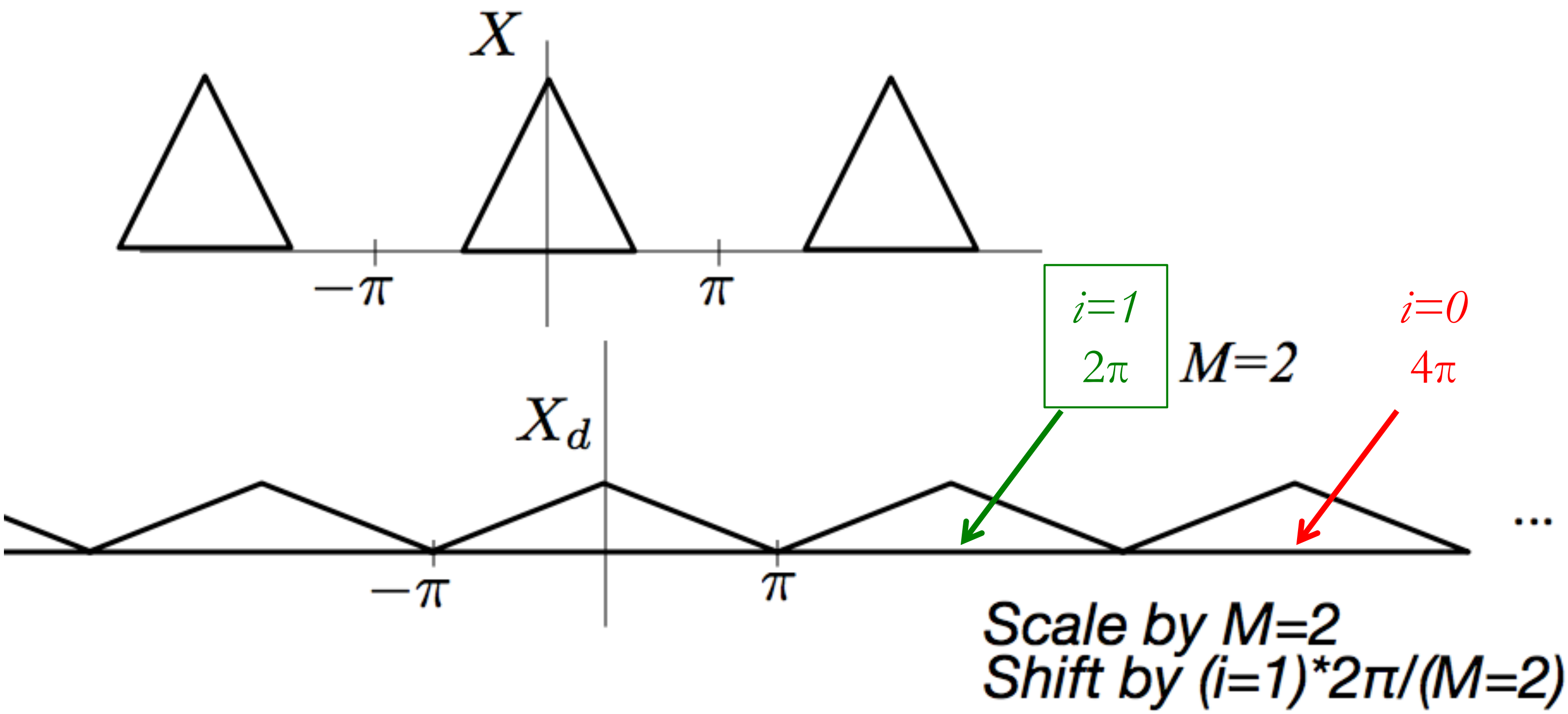


$$X_d(e^{j\omega}) = \frac{1}{M}\sum_{i=0}^{M-1} X\left(e^{j\left(\frac{\omega}{M} - \frac{2\pi}{M} i\right)}\right)$$

stretch by M — replicate

Example

$$X_d(e^{j\omega}) = \frac{1}{M}\sum_{i=0}^{M-1} X\left(e^{j\left(\frac{w}{M} - \frac{2\pi}{M}i\right)}\right)$$



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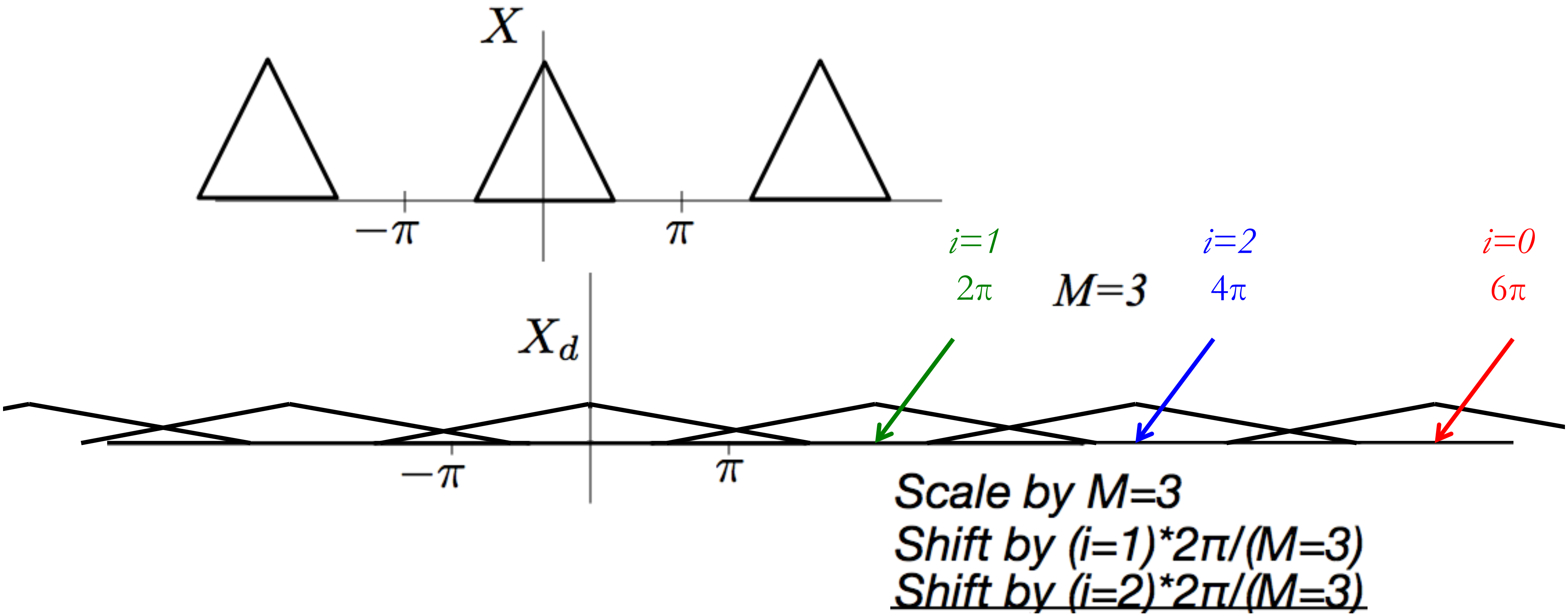
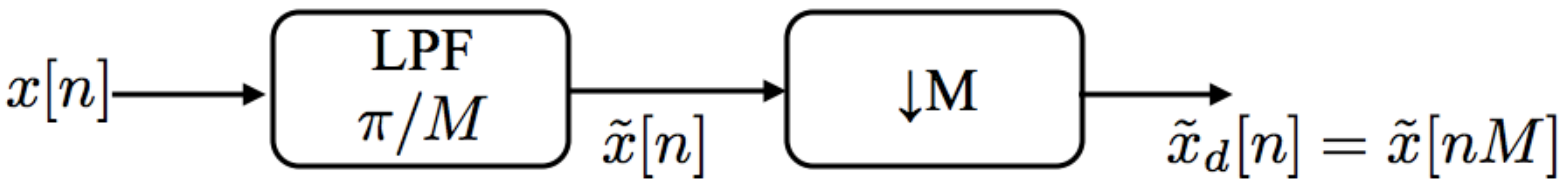
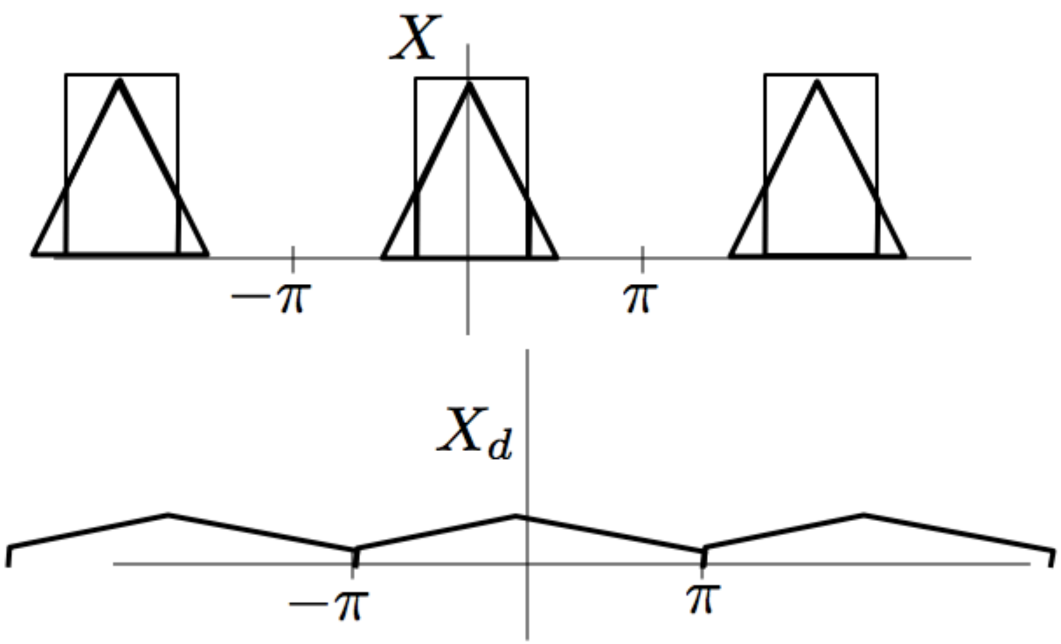


Example





M=3

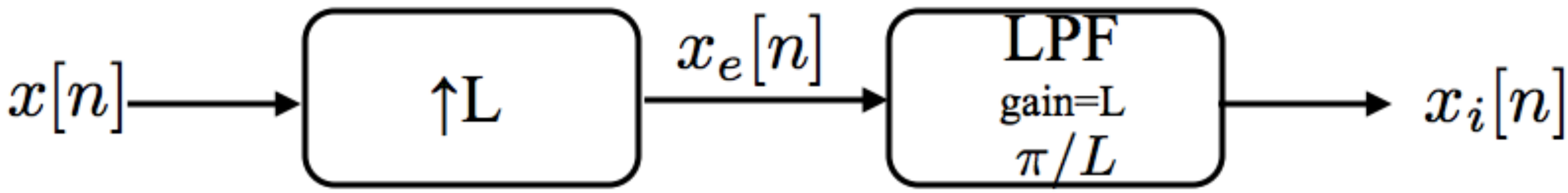
Upsampling

- Definition: Increasing the sampling rate by an integer number

$$x[n] = x_c(nT)$$

$$x_i[n] = x_c(nT') \quad \text{where } T' = \frac{T}{L} \qquad L \text{ integer}$$

Frequency Domain Interpretation

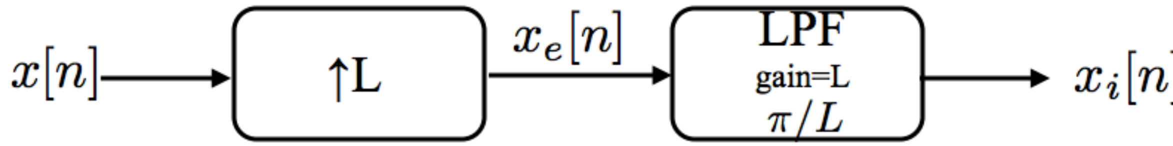


$$X_e(e^{j\omega}) = \sum_{n=-\infty}^{\infty} \underbrace{x_e[n]}_{\neq 0 \text{ only for } n=mL \text{ (integer m)}} e^{-j\omega n}$$

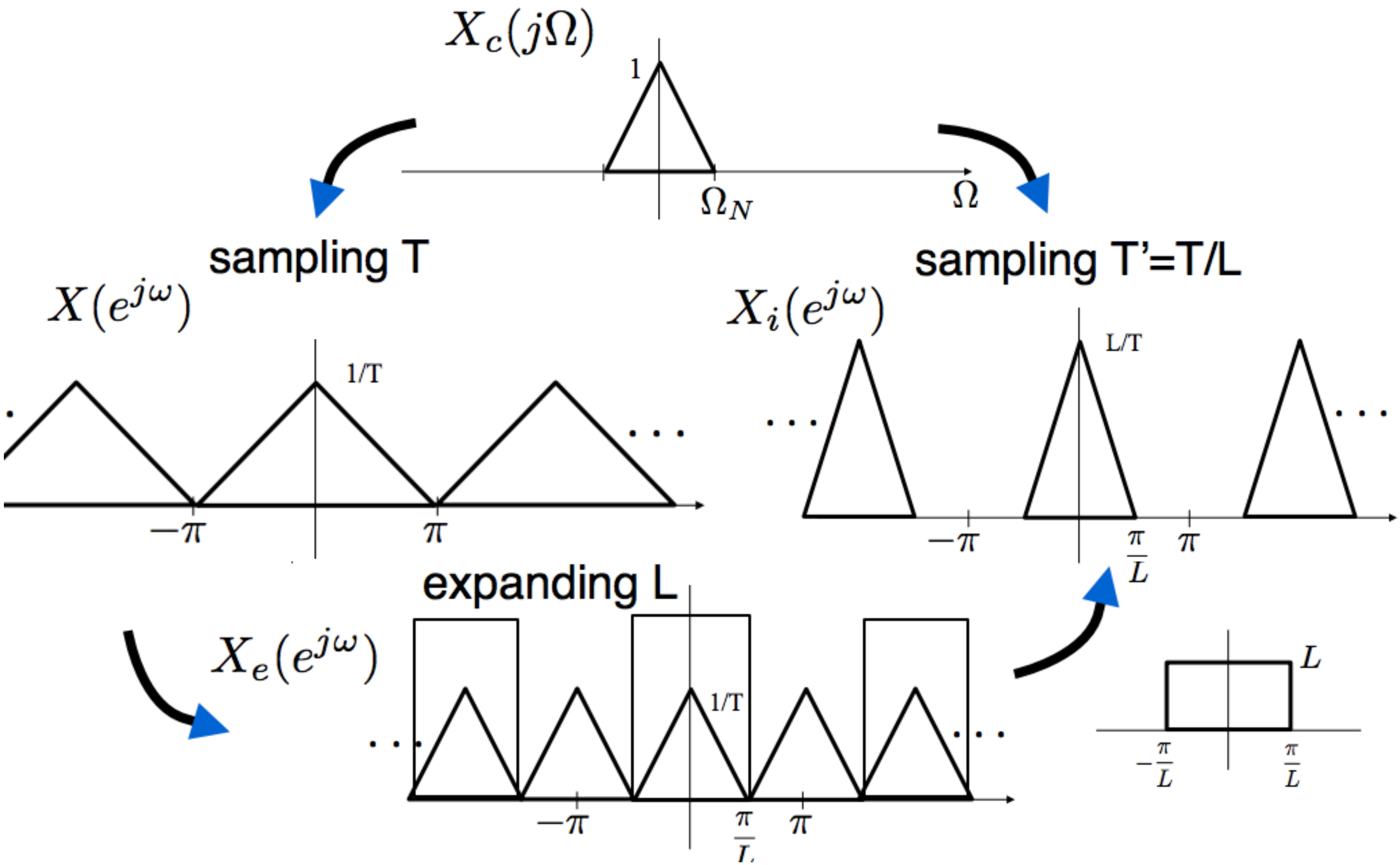
$$= \sum_{m=-\infty}^{\infty} \underbrace{x_e[mL]}_{=x[m]} e^{-j\omega mL} = \boxed{X(e^{j\omega L})}$$

Compress DTFT by a factor of L!

Example



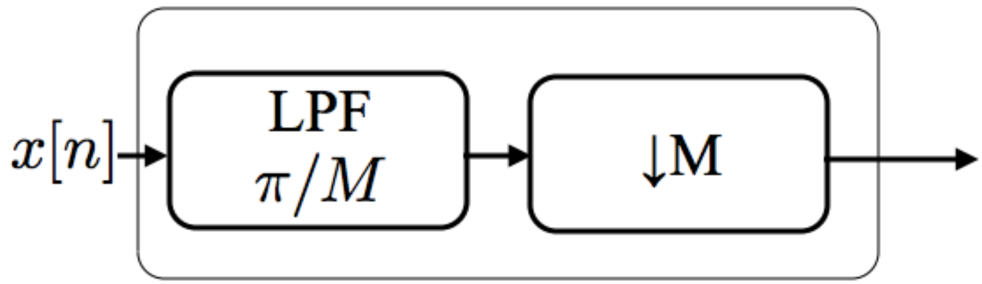
$x[n]$
↑L
$x_e[n]$
LPF
gain=L
π/L
$x_i[n]$



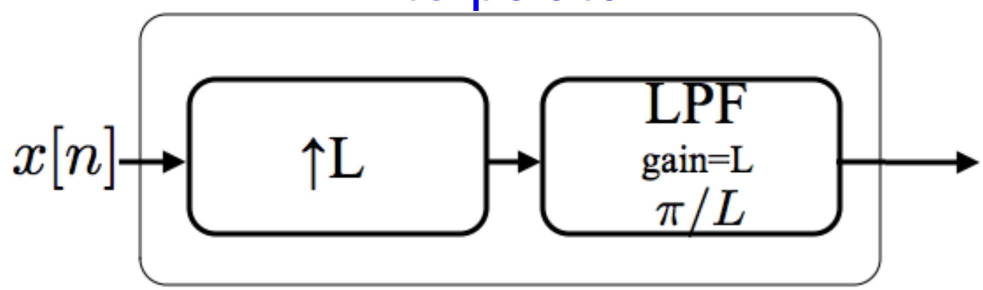
$X_c(j\Omega)$
1
Ω_N
Ω
sampling T
$X(e^{j\omega})$
1/T
$-\pi$
π
sampling T'=T/L
$X_i(e^{j\omega})$
L/T
$-\pi$
$\frac{\pi}{L}$
π
expanding L
$X_e(e^{j\omega})$
1/T
$-\pi$
$\frac{\pi}{L}$
π
L
$-\frac{\pi}{L}$
$\frac{\pi}{L}$

Interpolation and Decimation

decimator

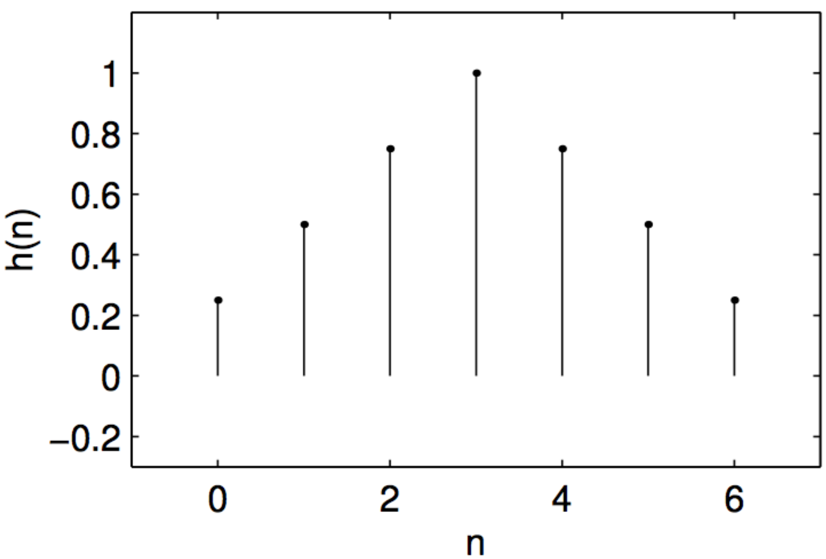


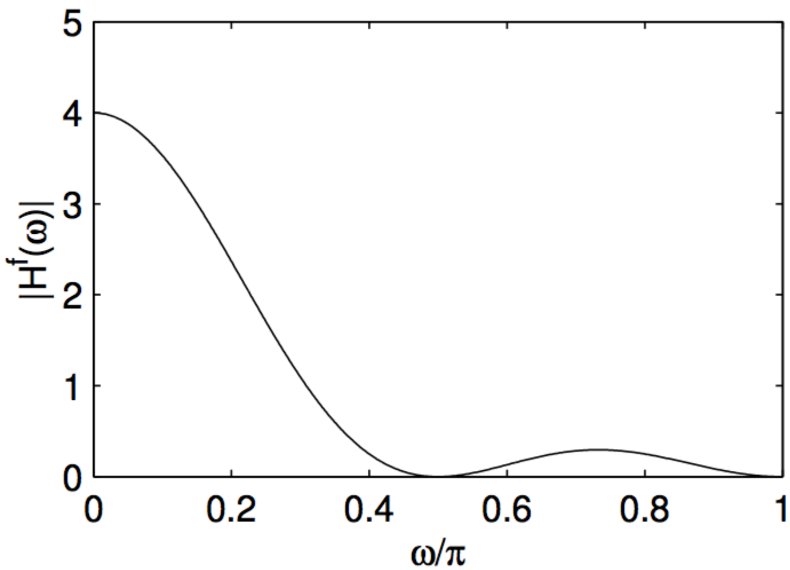
interpolator



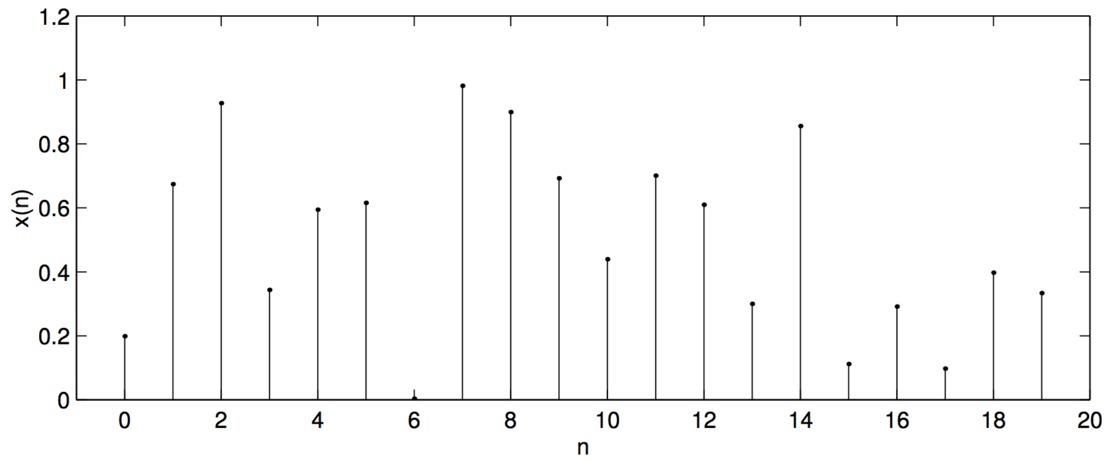
Interpolation Filter Example 1

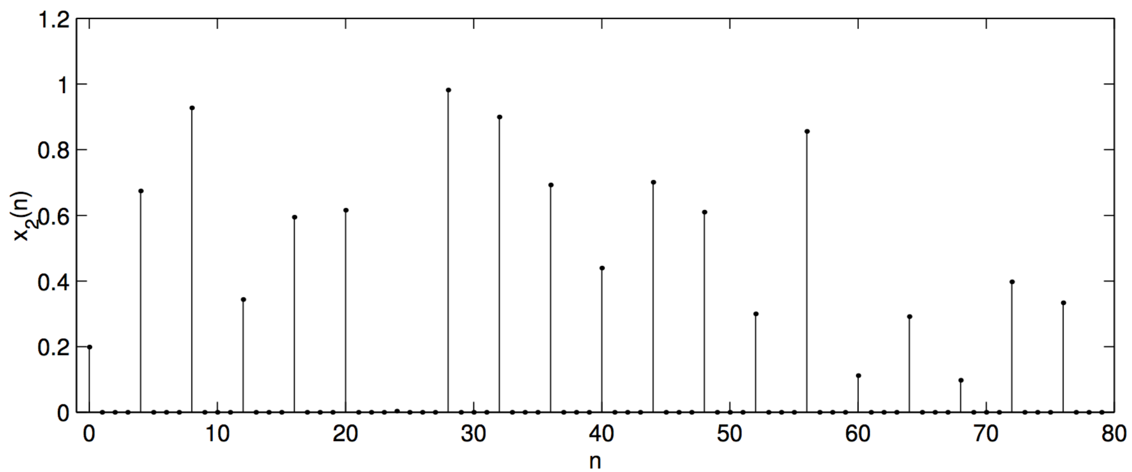
- This time we use a filter of length 7 with the effect of linear interpolation



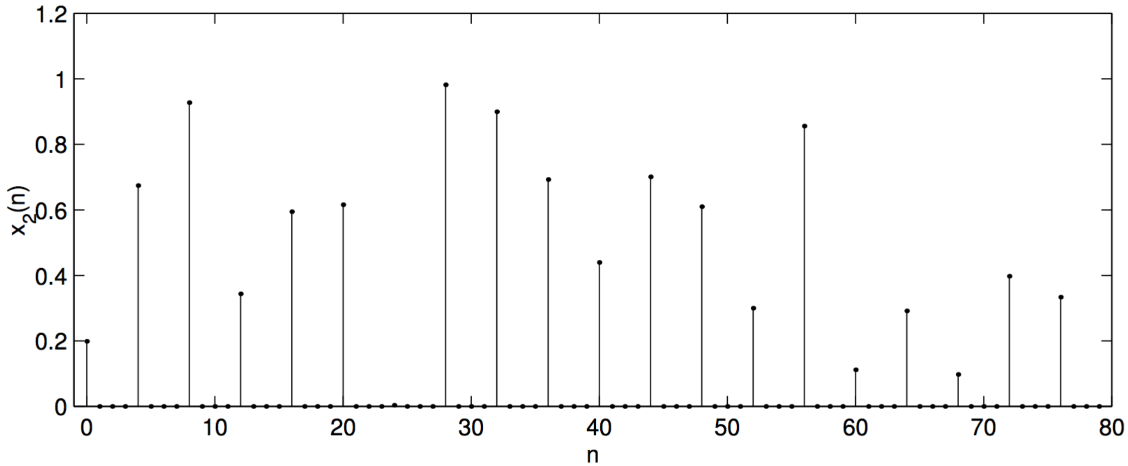


Interpolation Filter Example 1

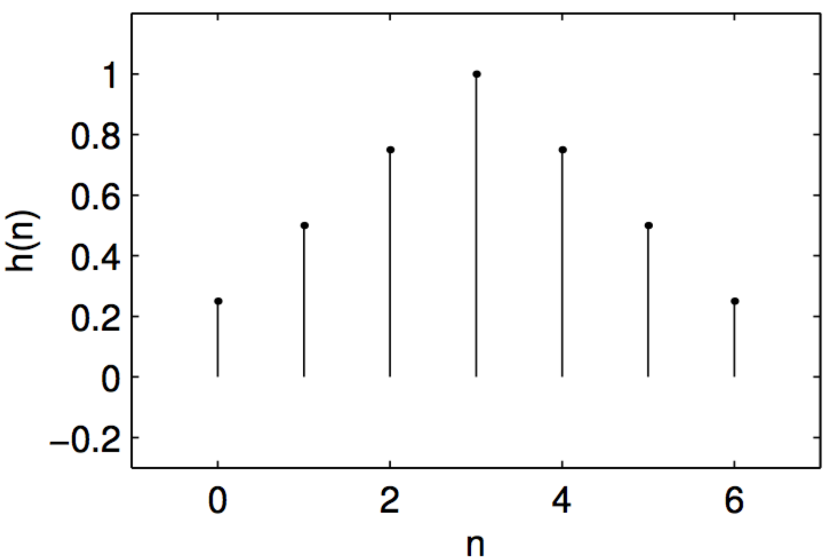


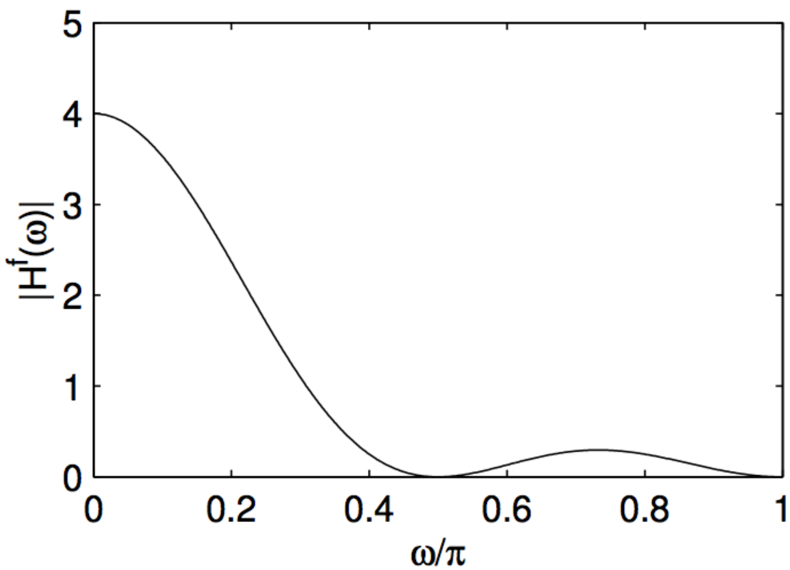


Interpolation Filter Example 1

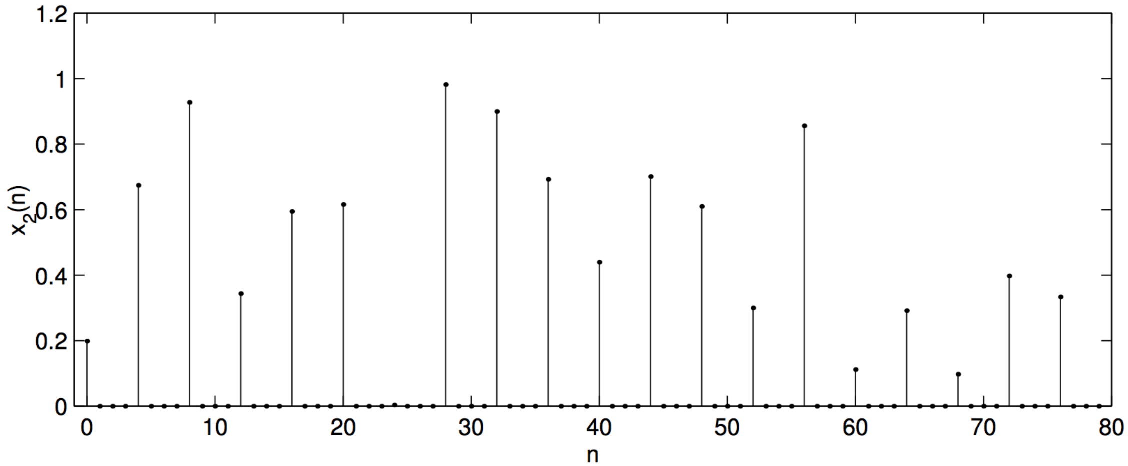


*
(convolve)



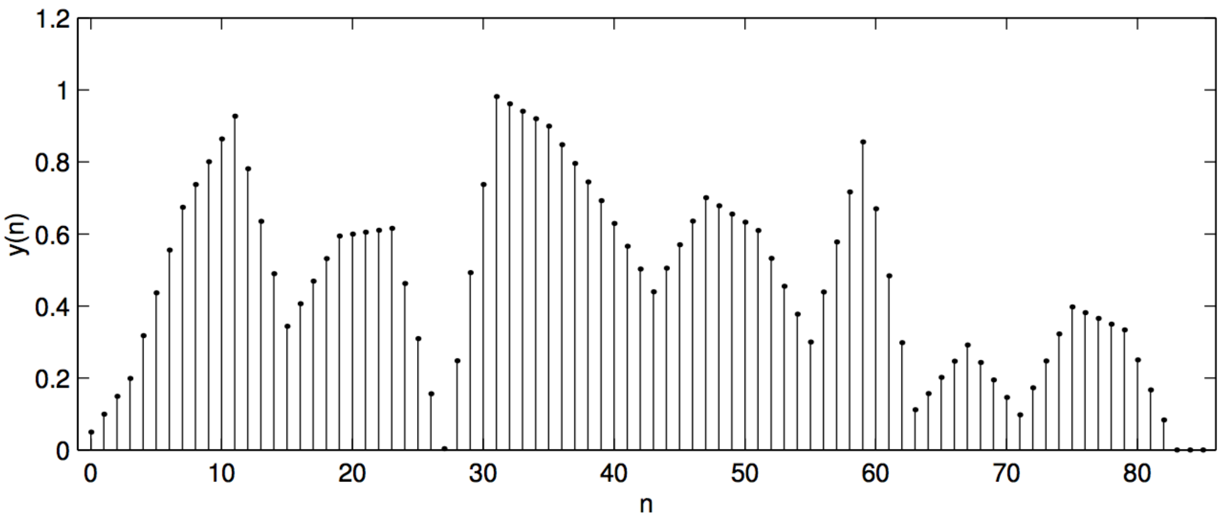


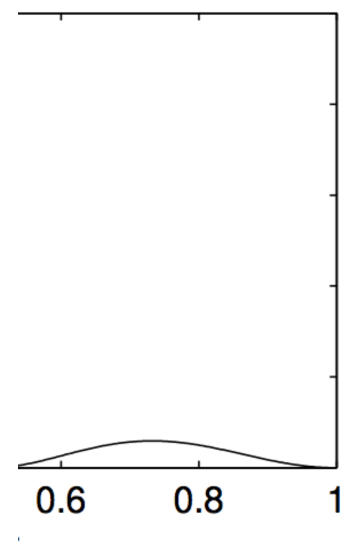
Interpolation Filter Example 1



*
(convolve)

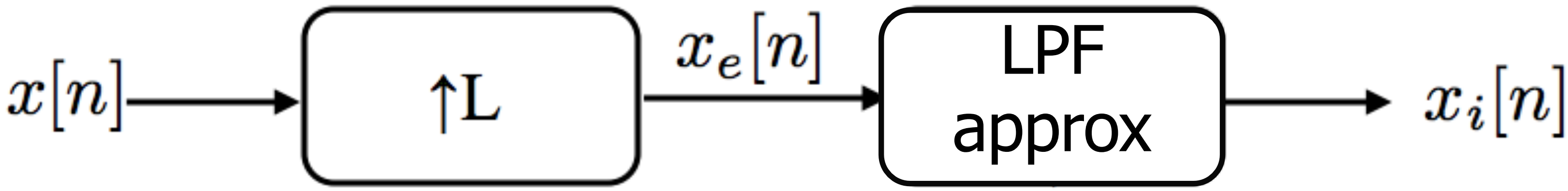
h(n)



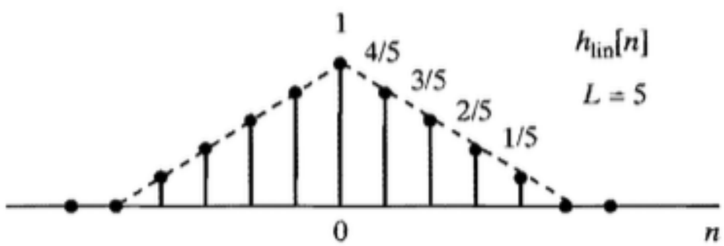


Linear Interpolation -- Frequency Domain

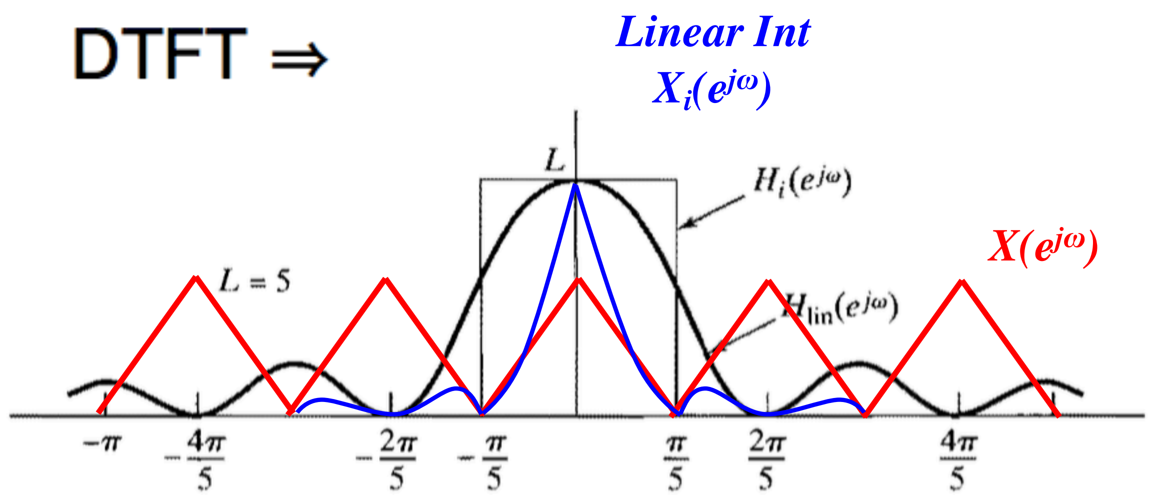
$$x_i[n] = x_e[n] * h_{lin}[n]$$



$$h_{\text{lin}}[n] = \begin{cases} 1 - |n|/L, & |n| \le L, \\ 0, & \text{otherwise}, \end{cases}$$

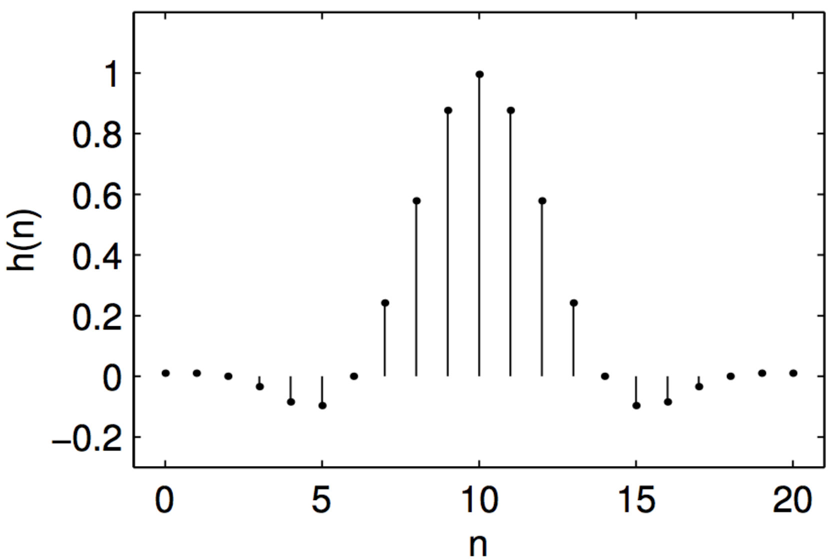


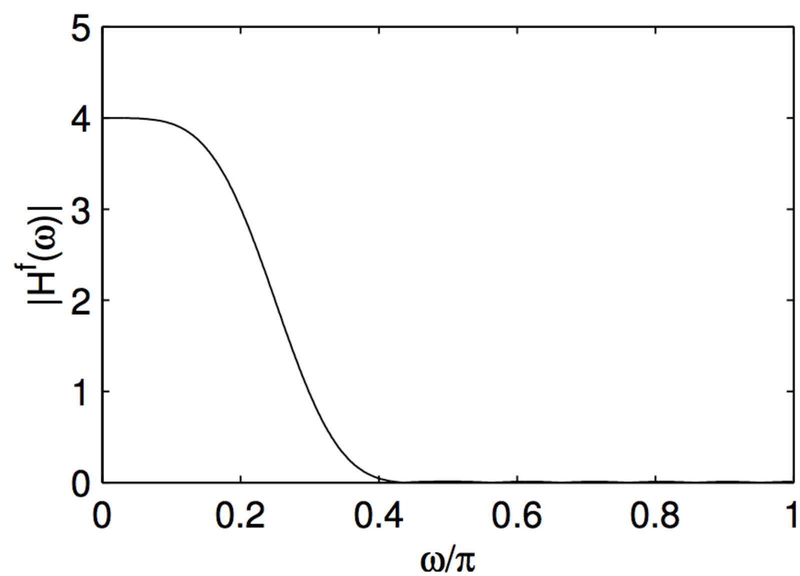
DTFT ⇒



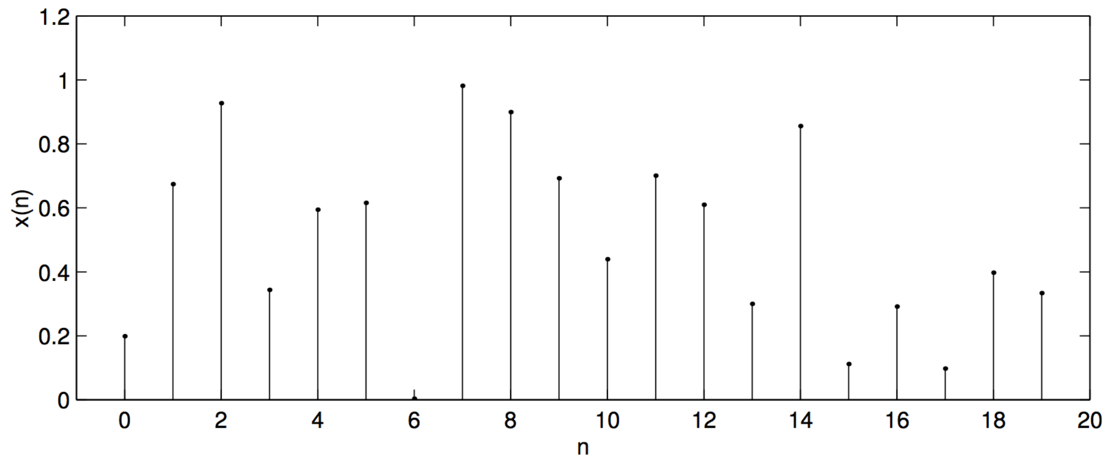
Interpolation Filter Example 2

- In this example, we interpolate a signal x(n) by a factor of 4.
- We use a linear phase Type I FIR lowpass filter of length 21.

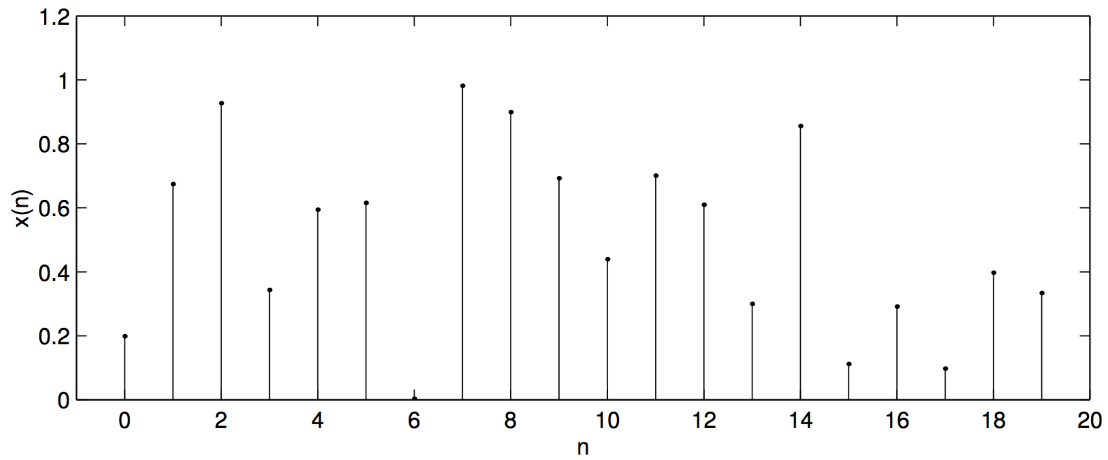


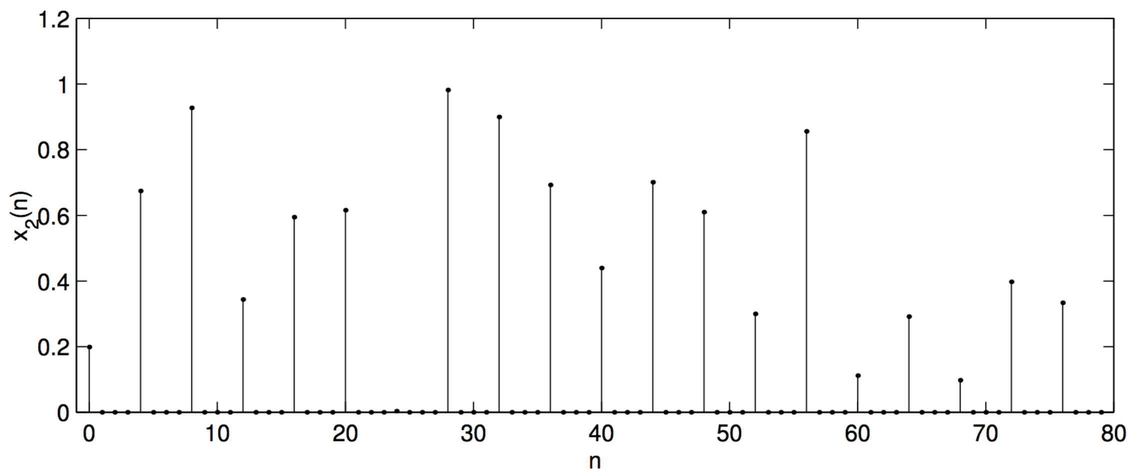


Interpolation Filter Example 2

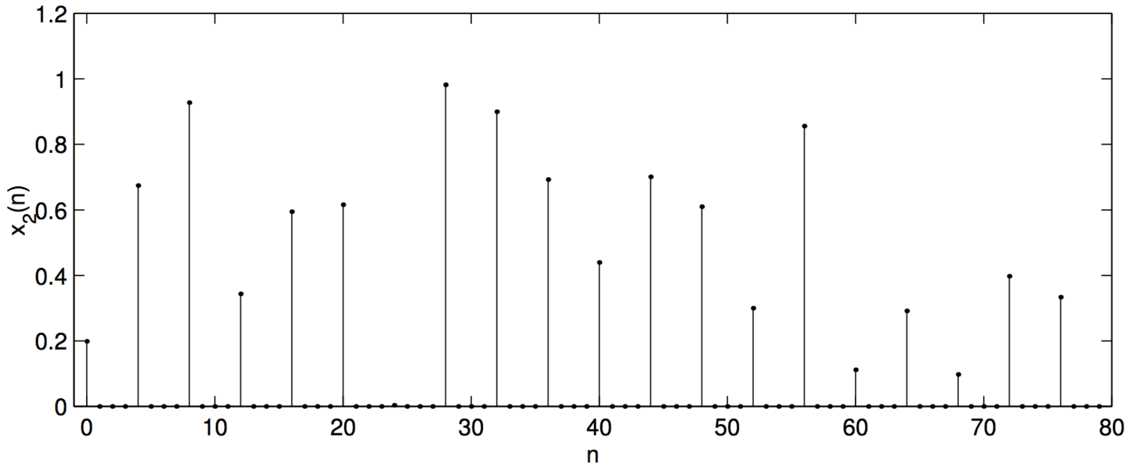


Interpolation Filter Example 2

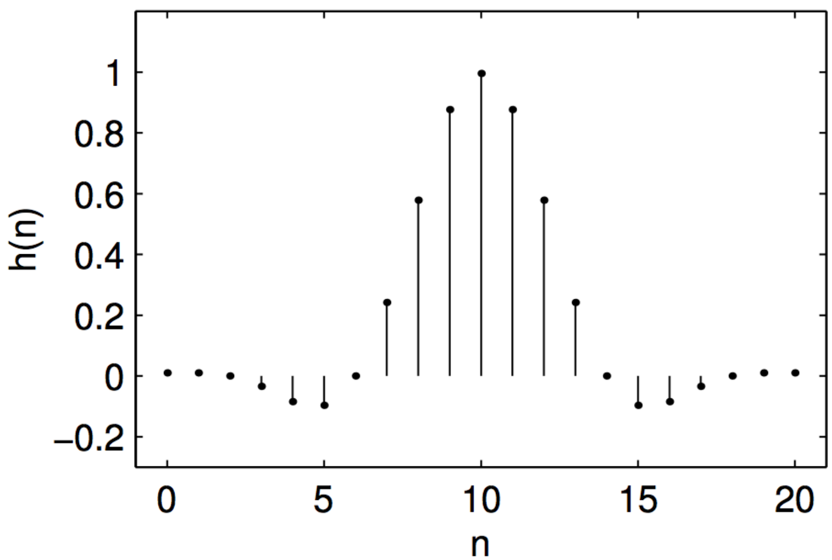


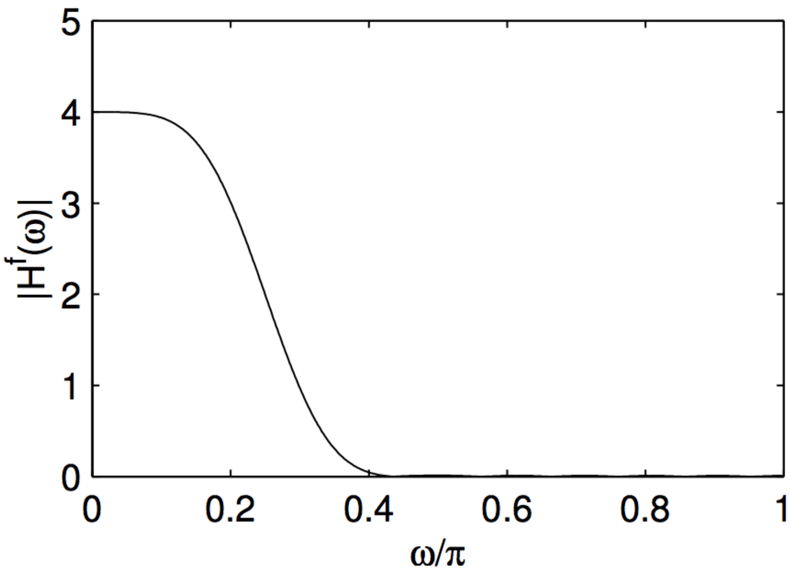


Interpolation Filter Example 2



*
(convolve)





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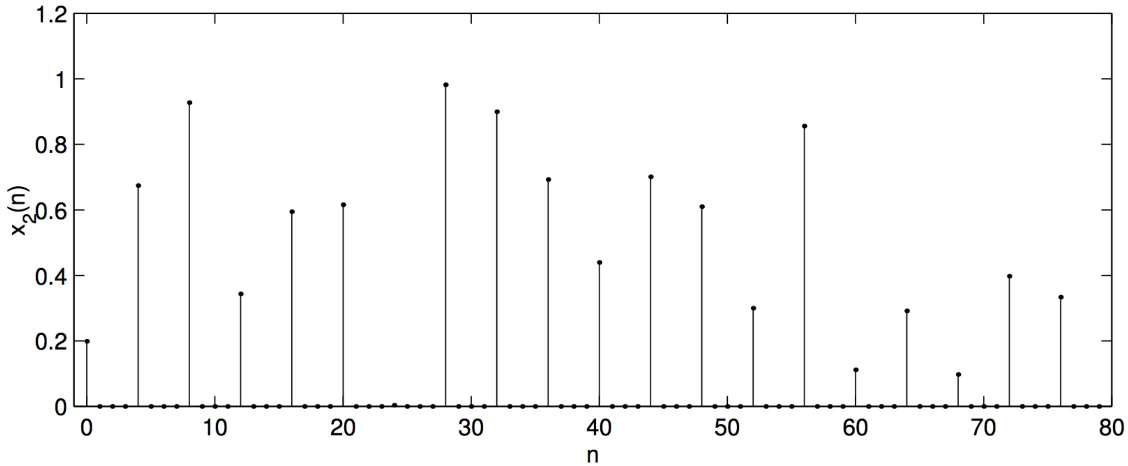

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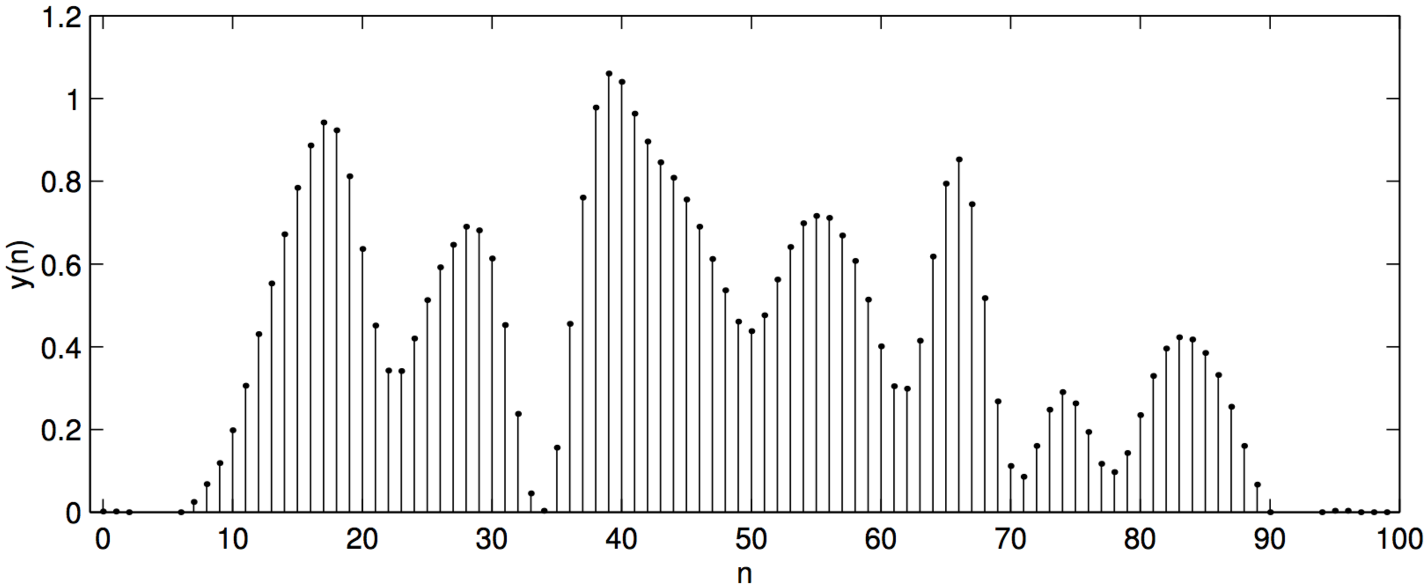

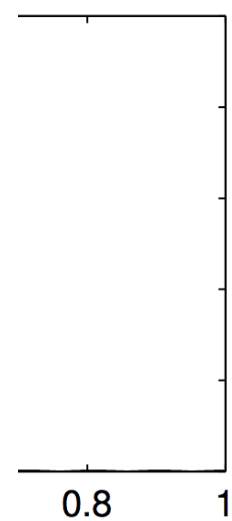

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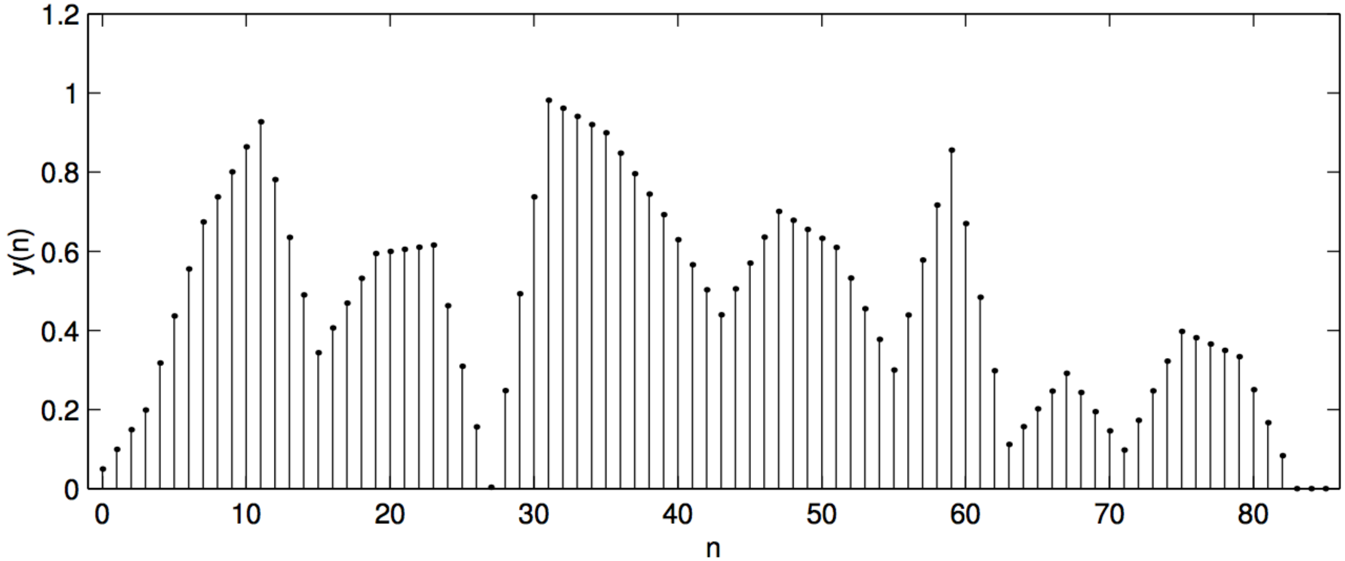

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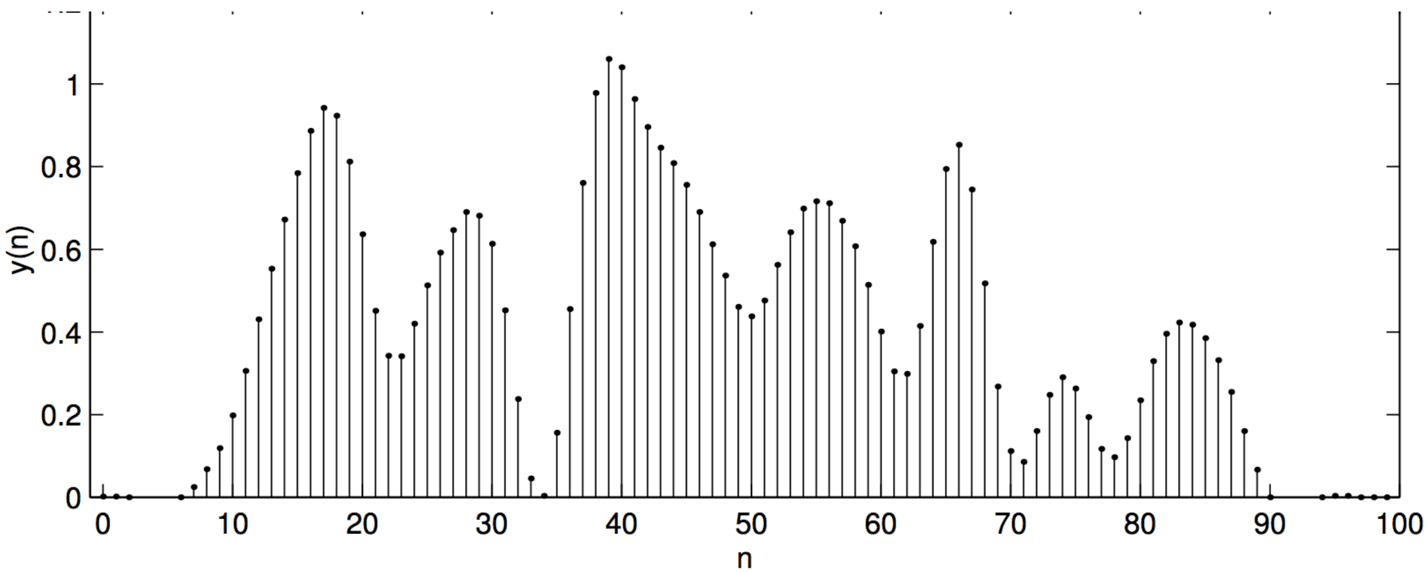
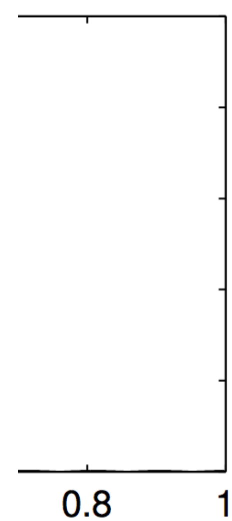
ɔnvolve)





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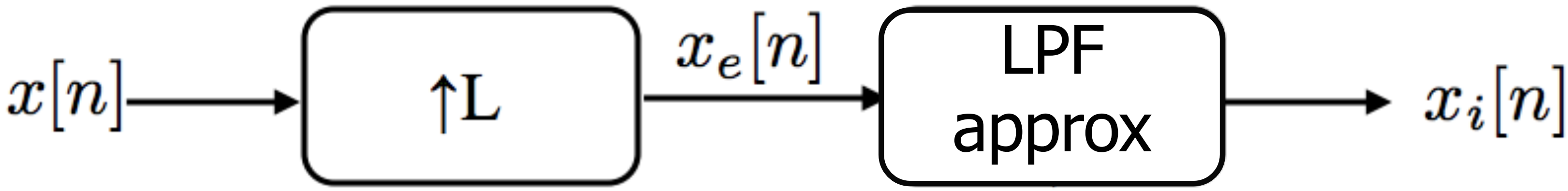

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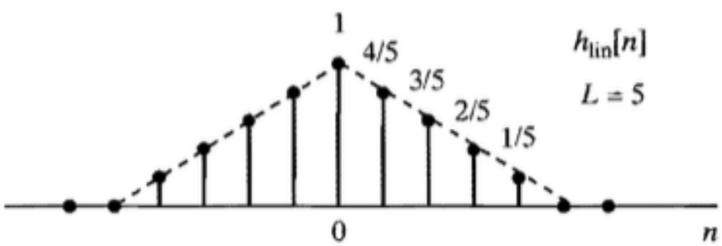

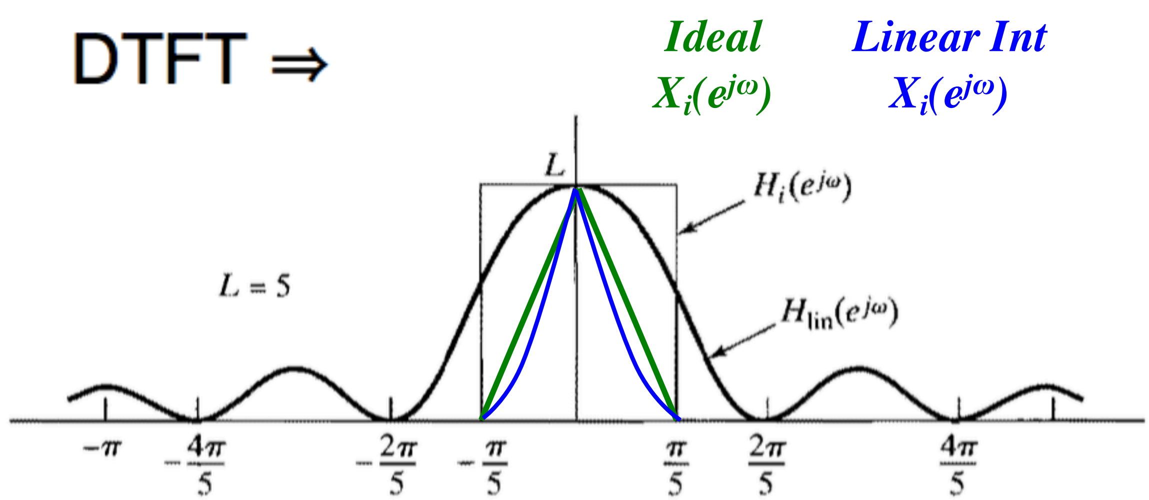

Interpolation Filter Example 3

- When interpolating a signal x(n), the interpolation filter h(n) will in general change the samples of x(n) in addition to filling in the zeros.
- Can a filter be designed so as to preserve the original samples x(n)?

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- To be precise, if $y(n) = h(n) * [\uparrow 2]\, x(n)$ then can we design h(n) so that $y(2n) = x(n)$?

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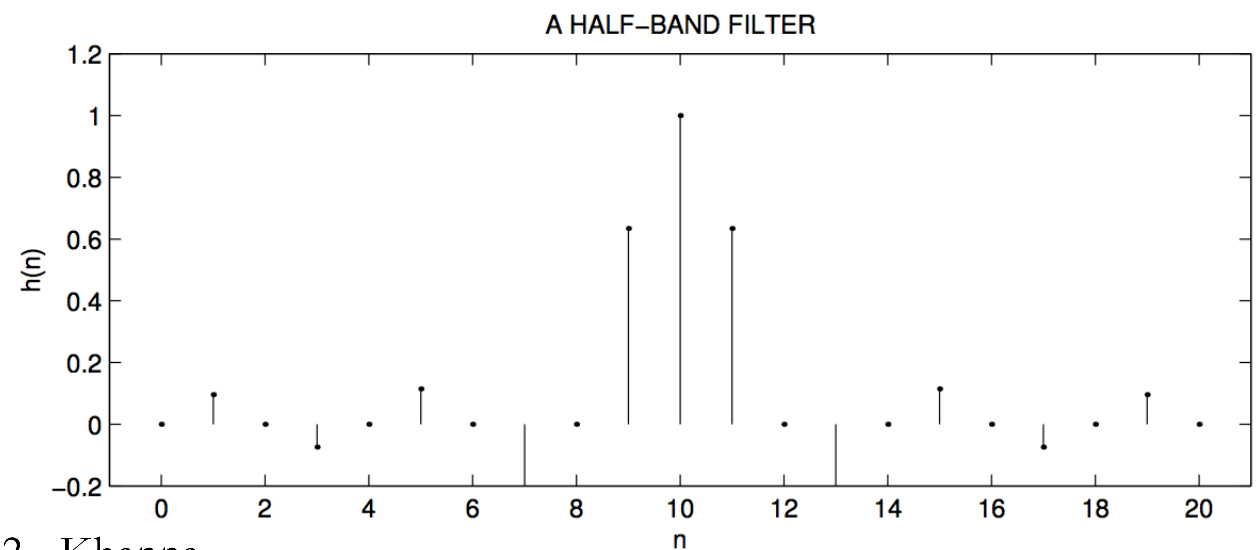
- To be precise, if $y(n) = h(n) * [\uparrow 2]\, x(n)$ then can we design h(n) so that $y(2n) = x(n)$?
  - Or more generally, so that $y(2n + n_o) = x(n)$?

Interpolation Filter Example 3

- When interpolating by a factor of 2, if h(n) is a half-band filter, then it will not change the samples x(n).
- A n_o-centered half-band filter h(n) is a filter that satisfies:

$$h(n) = \begin{cases} 1, & \text{for } n = n_o \\ 0, & \text{for } n = n_o \pm 2, 4, 6, \ldots \end{cases}$$

- That means, every second value of h(n) is zero, except for one such value, as shown in the figure.



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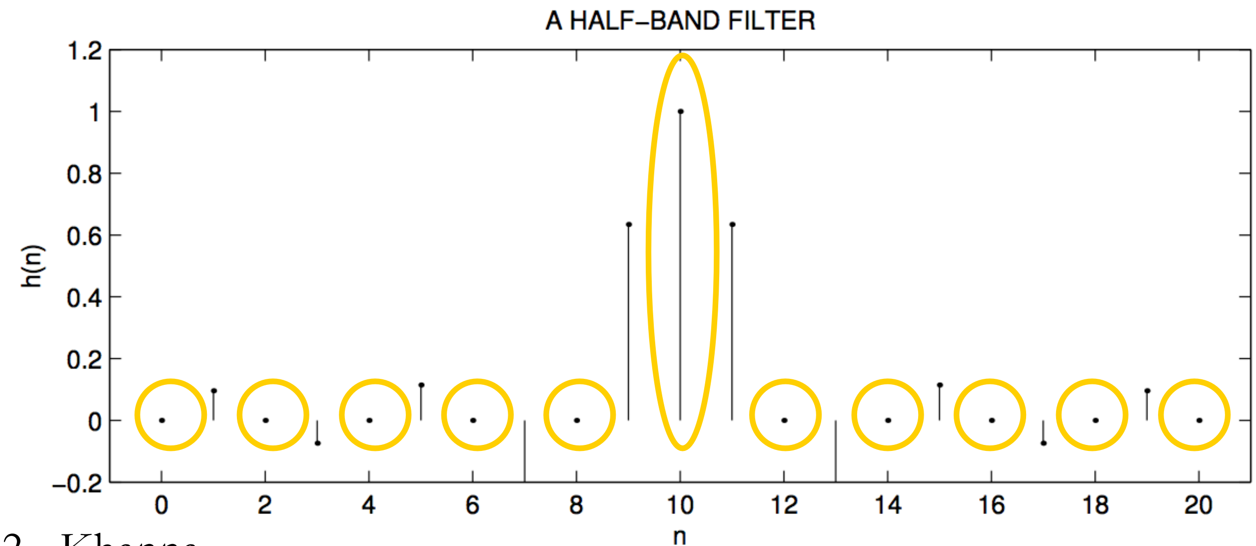

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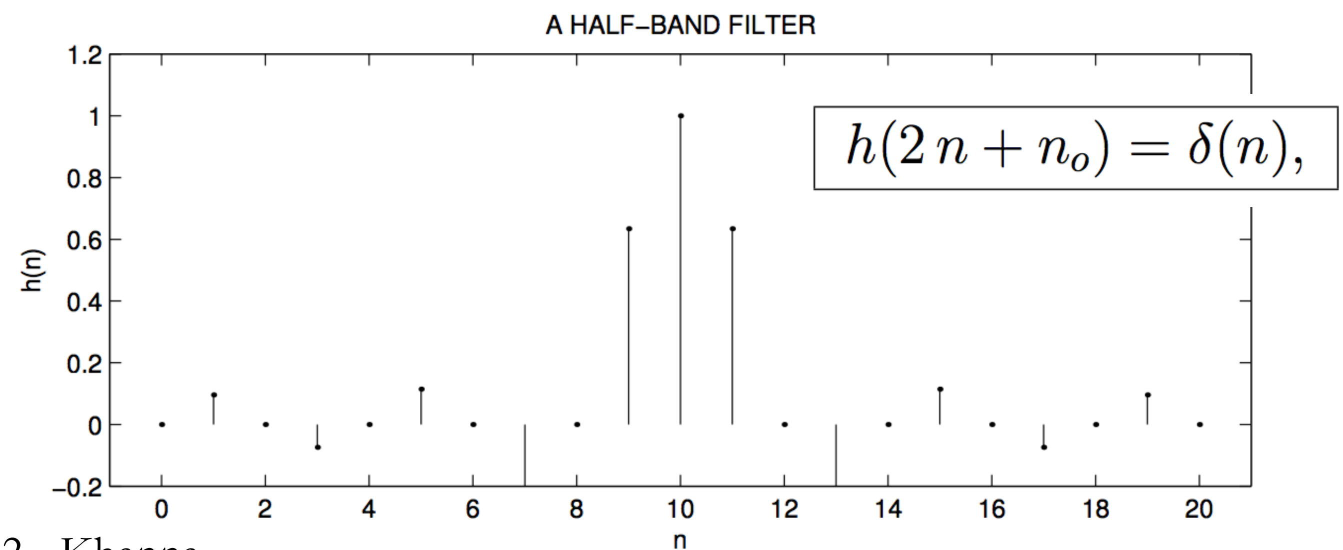

Interpolation Filter Example 4

- When interpolating a signal x(n) by a factor L, the original samples of x(n) are preserved if h(n) is a Nyquist-L filter.
- A Nyquist-L filter simply generalizes the notion of the halfband filter to $L > 2$.

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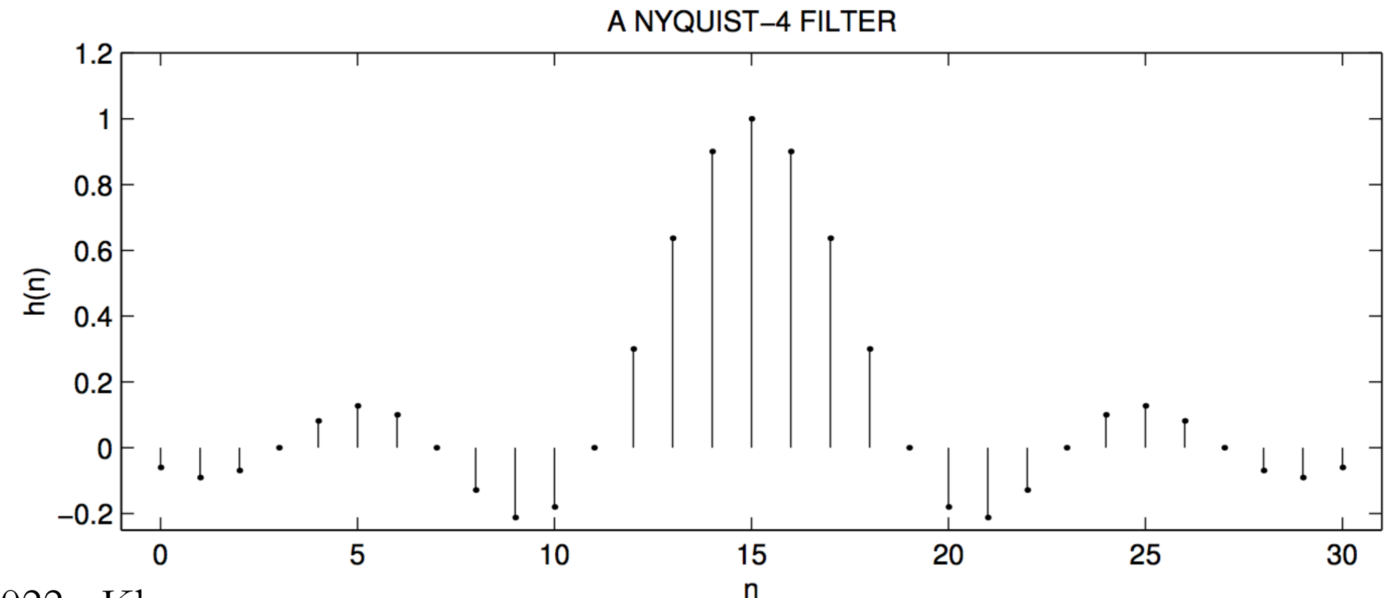
- A (0-centered) Nyquist-L filter h(n) is one for which

$$h(L\,n) = \delta(n).$$



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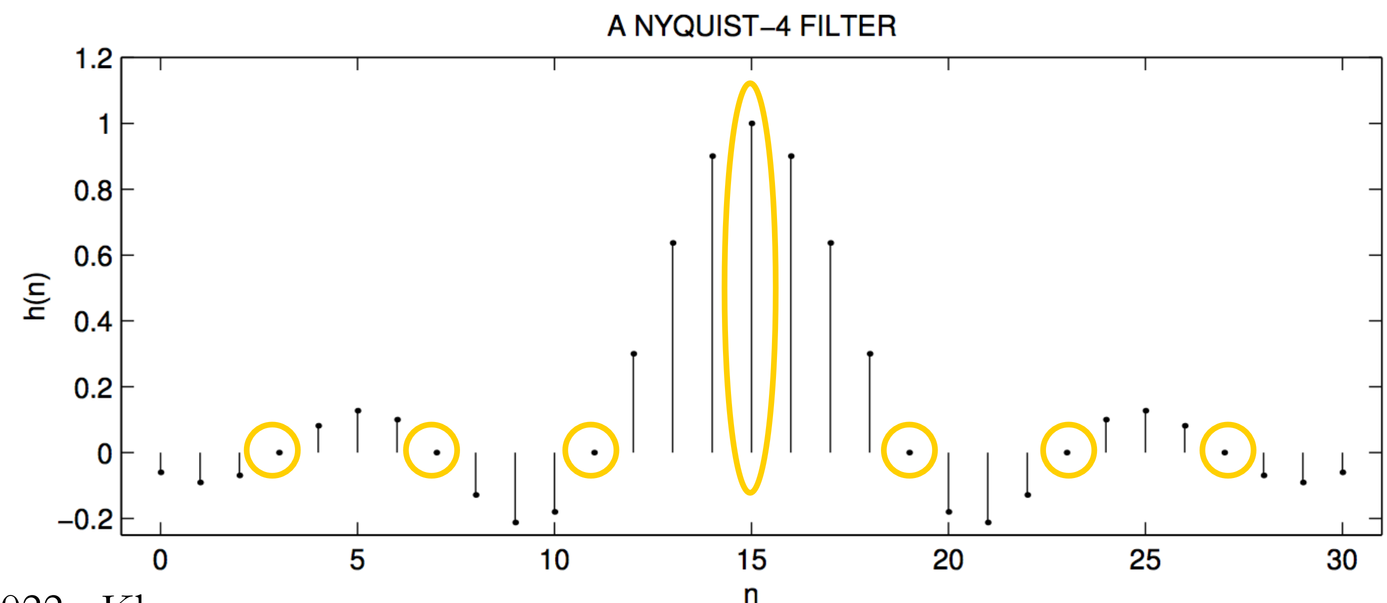


Non-integer Resampling

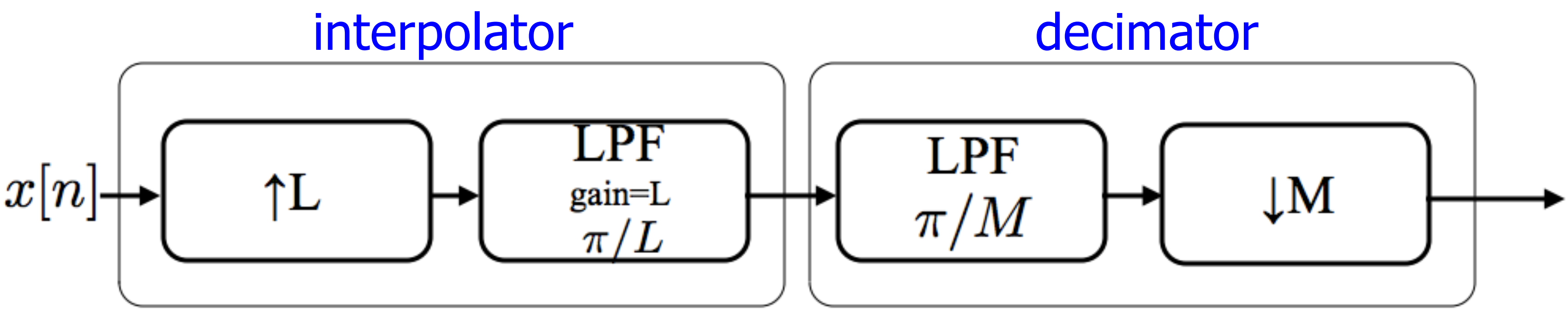
Penn

Non-integer Resampling

- $T' = TM/L$

Non-integer Resampling

- T'=TM/L
  - Upsample by L, then downsample by M



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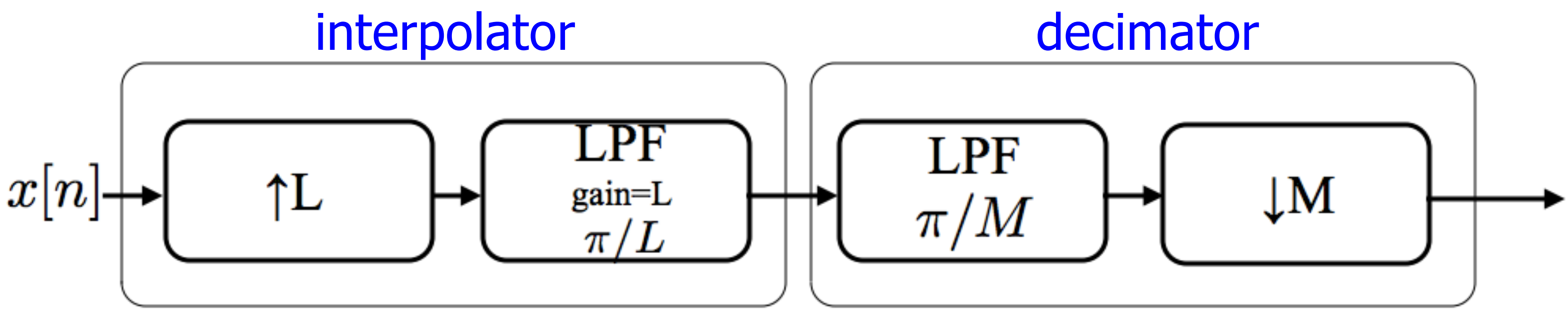
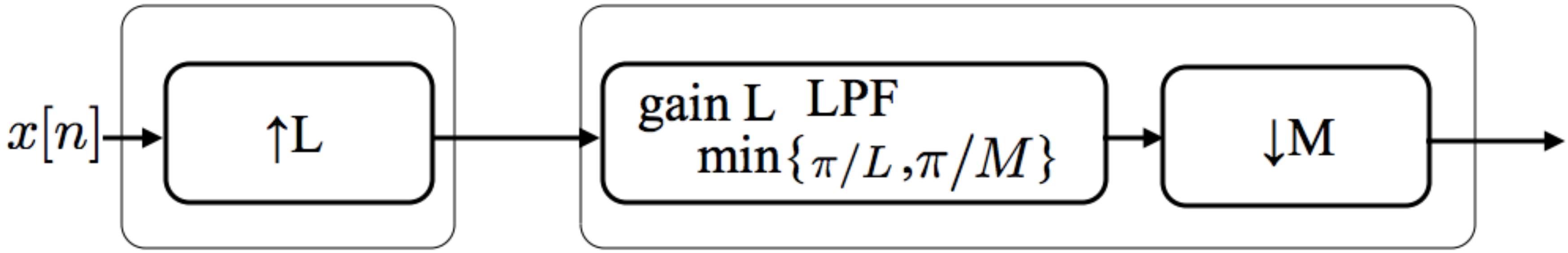
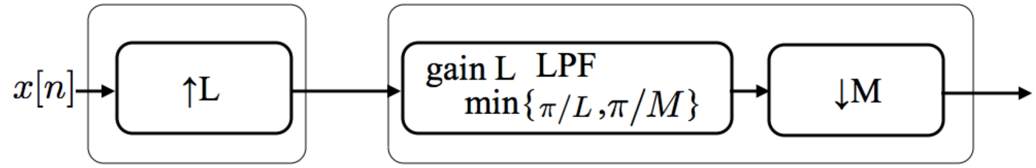


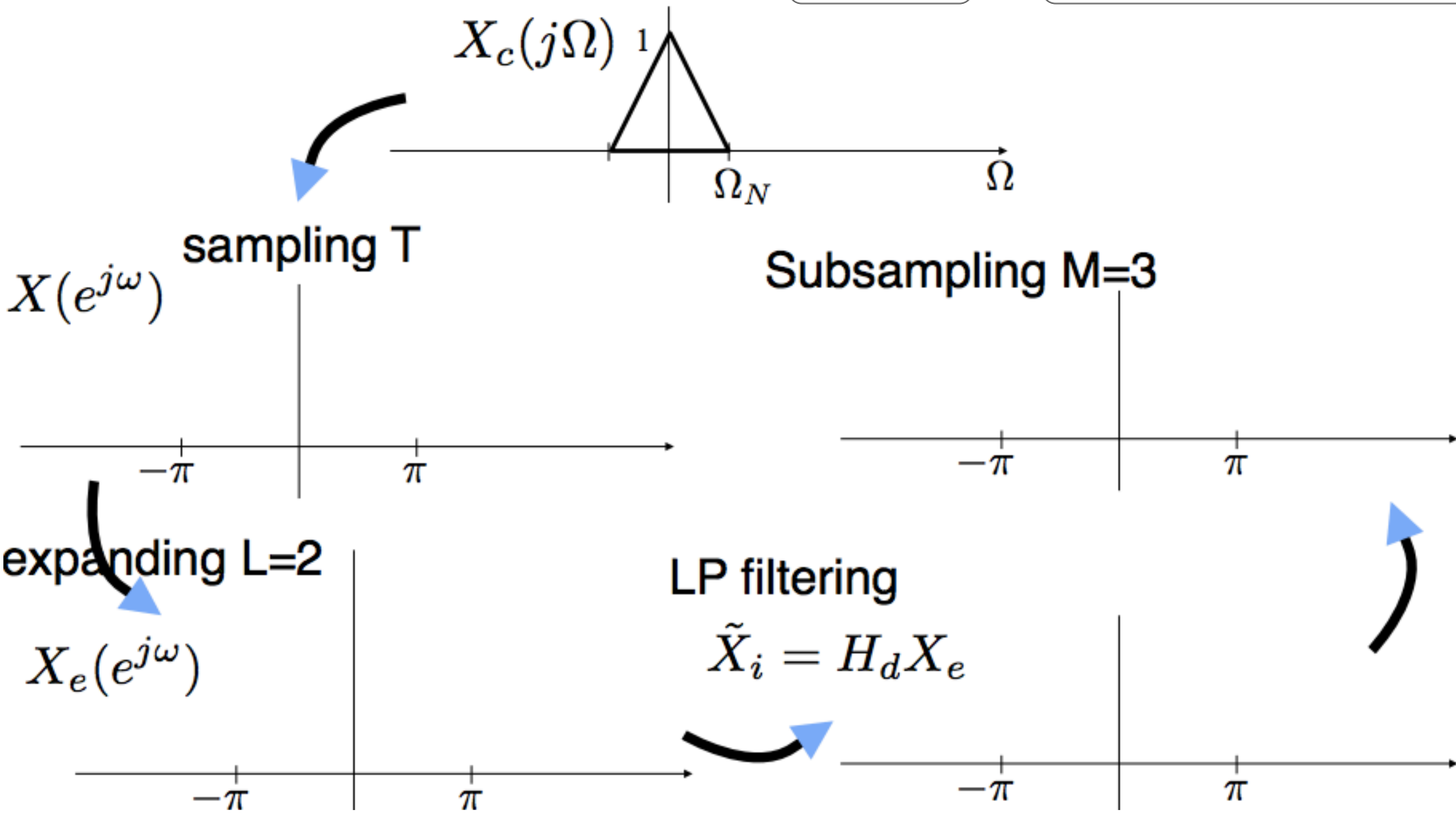
Or,



Example

- T'=3/2T → L=2, M=3





Example

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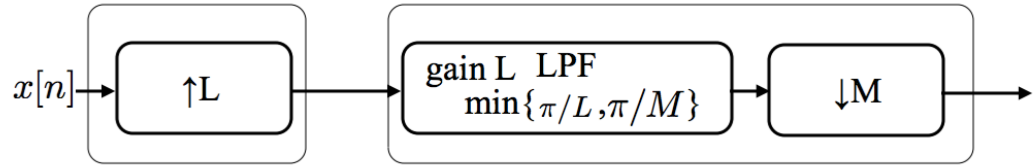
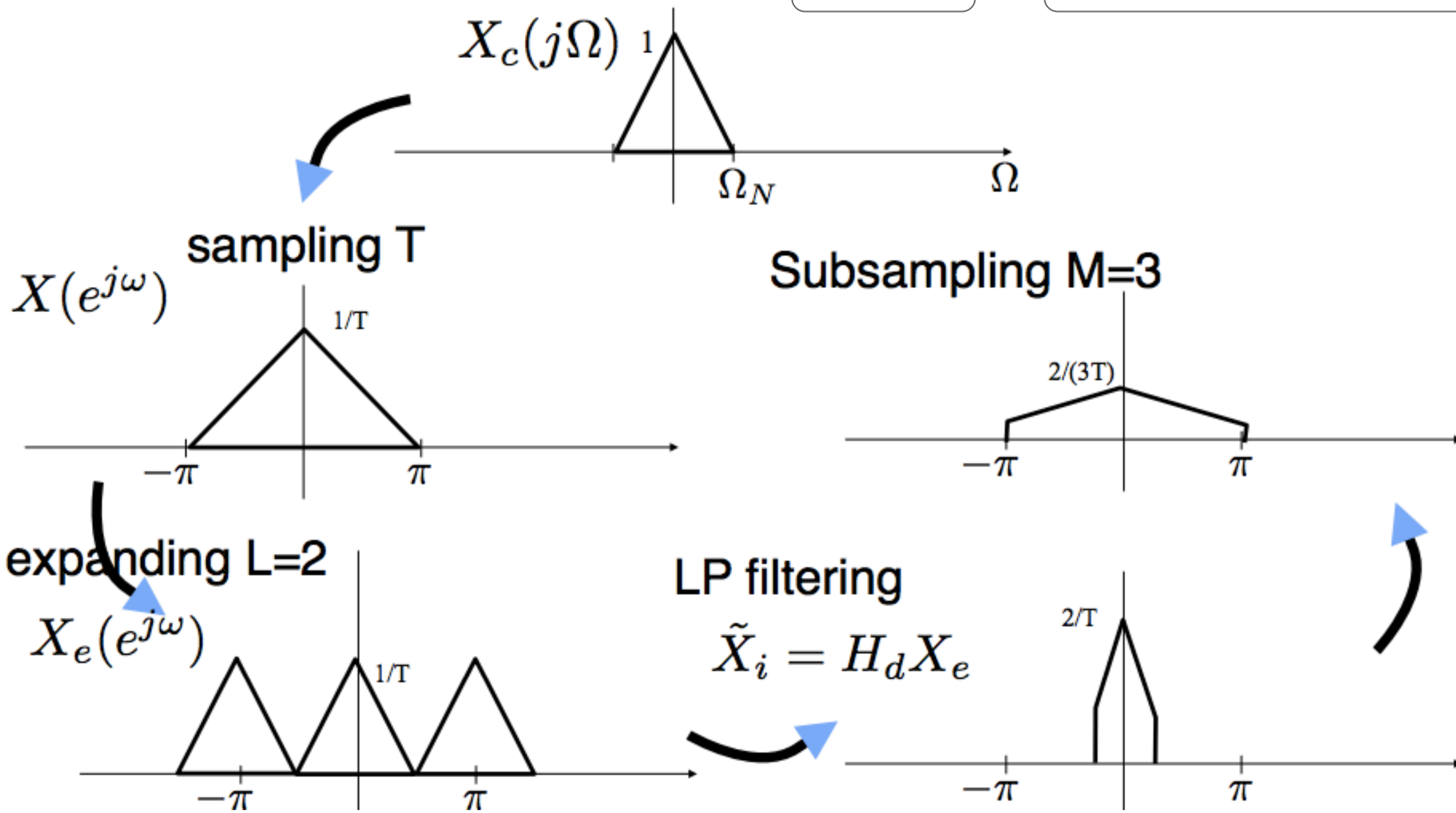
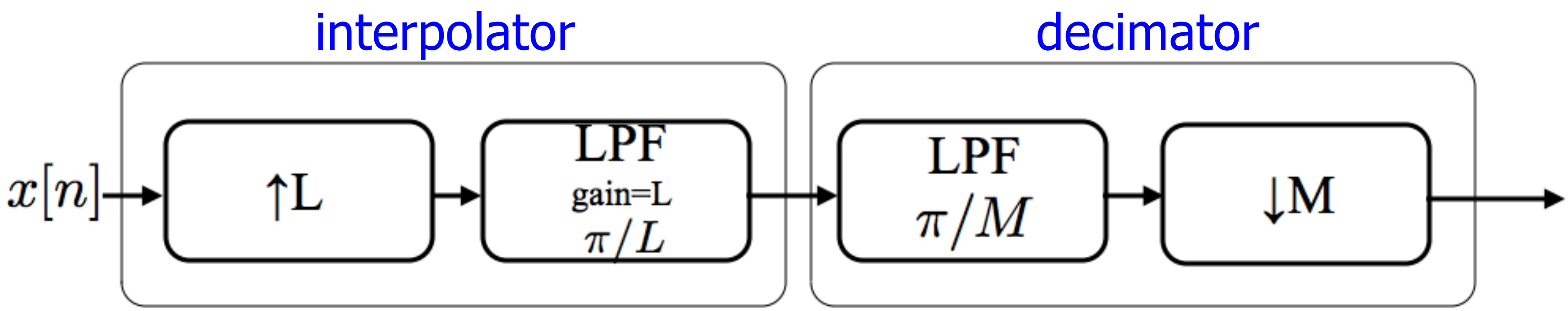




Non-integer Sampling

- T'=TM/L
  - Downsample by M, then upsample by L?



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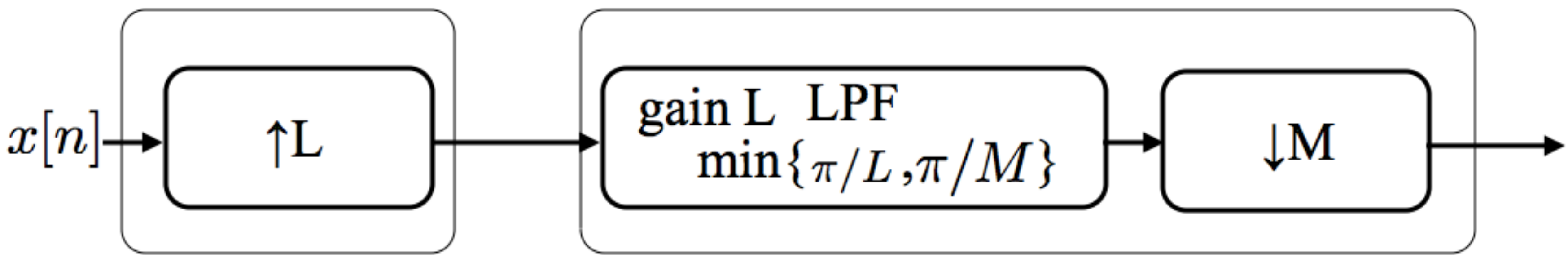


Example

- What if we want to resample by 1.01T?

Example

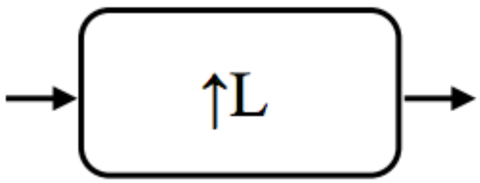
- What if we want to resample by 1.01T?
  - Upsample by L=100
  - Filter $\pi/101$
  - Downsample by M=101

Example

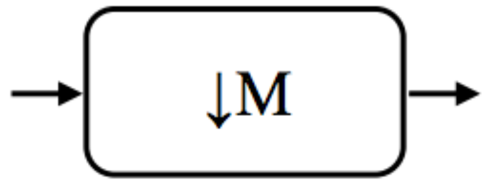
- What if we want to resample by 1.01T?
  - Upsample by L=100
  - Filter $\pi/101$ ($$$$$)
  - Downsample by M=101
- Fortunately there are ways around it!
  - Called multi-rate signal processing
  - Uses compressors, expanders and filtering

Interchanging Operations

LTI?



"expander"



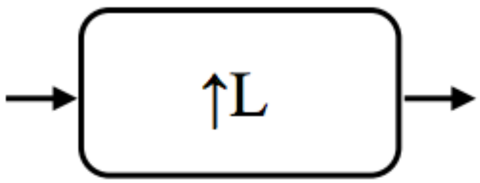
"compressor"

<u>not LTI!</u>

Upsampling
-expanding in time
-compressing in frequency

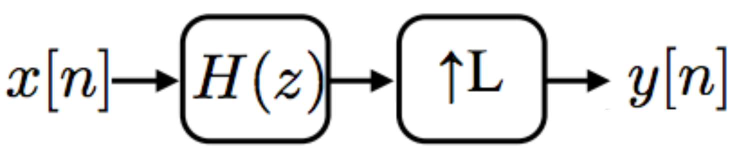
Downsampling
-compressing in time
-expanding in frequency

Interchanging Operations - Expander

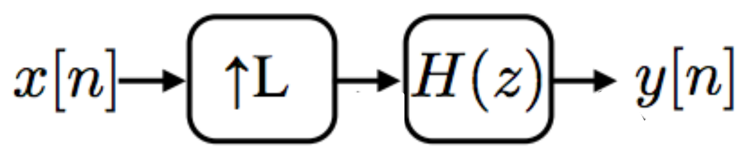


“expander”

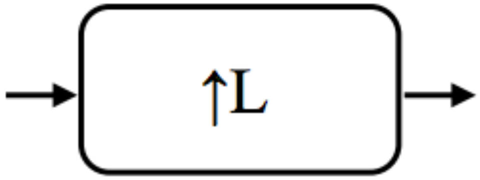
Upsampling
-expanding in time
-compressing in frequency



?

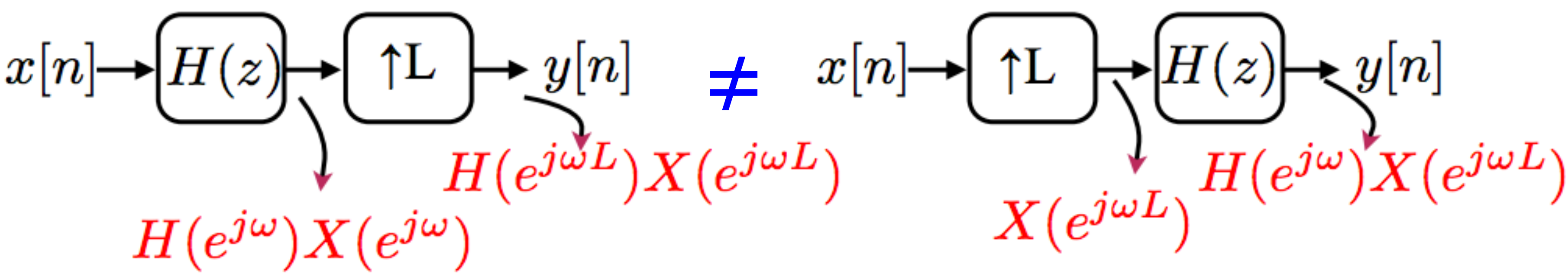


Interchanging Operations - Expander

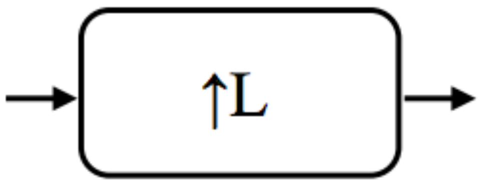


"expander"

Upsampling
-expanding in time
-compressing in frequency



Interchanging Operations - Expander



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Upsampling
-expanding in time
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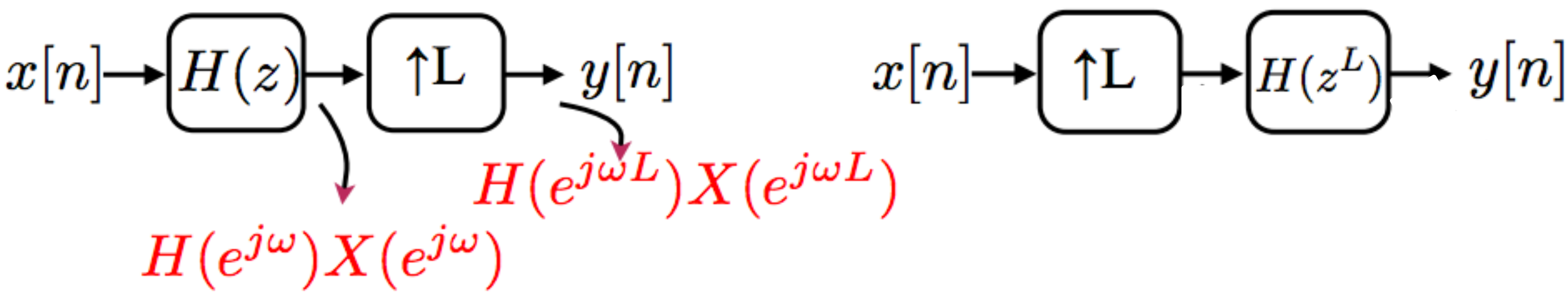
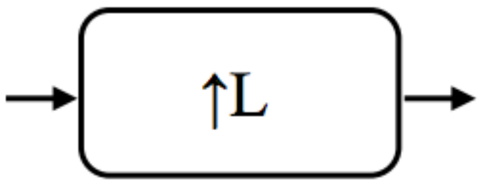


Interchanging Operations - Expander

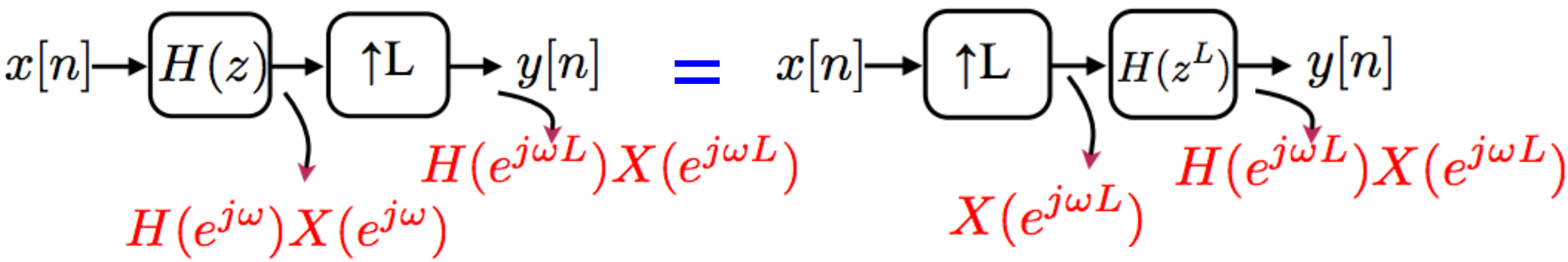


"expander"

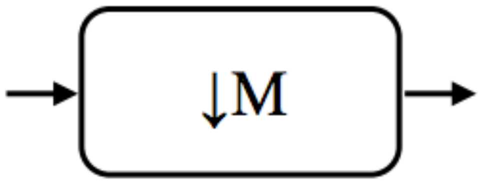
Upsampling

-expanding in time

-compressing in frequency



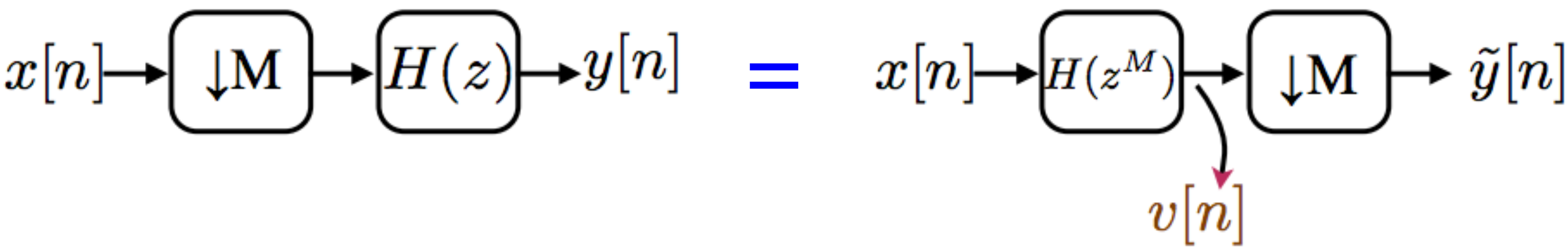
Interchanging Operations - Compressor



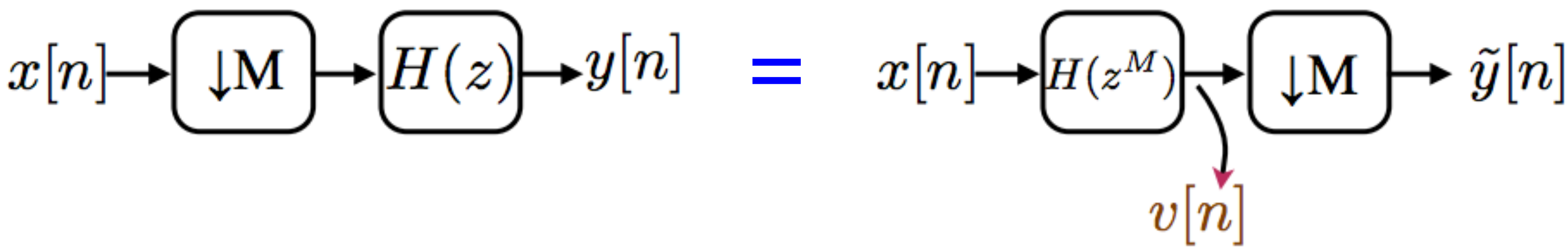
not LTI!

"compressor"

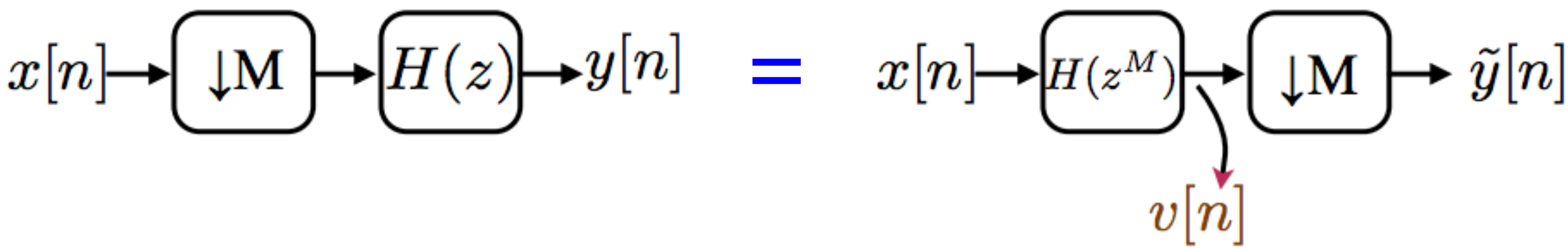
Downsampling
-compressing in time
-expanding in frequency



Interchanging Operations - Compressor



Interchanging Operations - Compressor

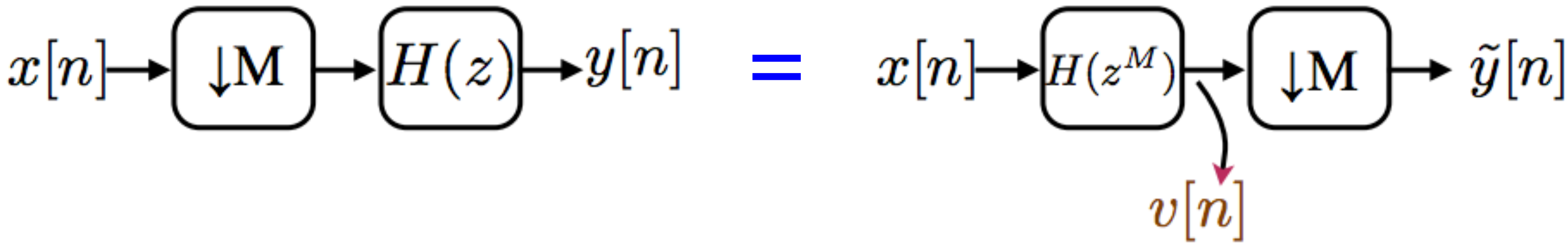


$$Y(e^{j\omega}) = H(e^{j\omega})\left(\frac{1}{M}\sum_{i=0}^{M-1} X\left(e^{j\left(\frac{\omega}{M}-\frac{2\pi i}{M}\right)}\right)\right)$$

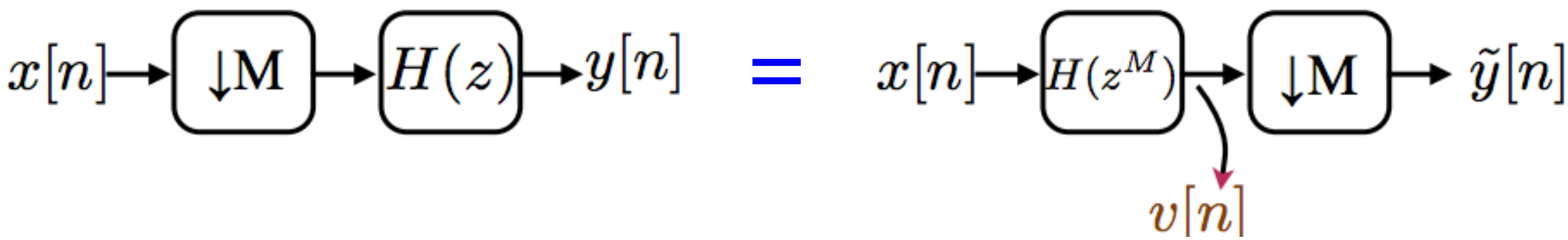
$$= \frac{1}{M}\sum_{i=0}^{M-1} \underbrace{H\left(e^{j(\omega-2\pi i)}\right)}_{H(e^{j\omega})} X\left(e^{j\left(\frac{\omega}{M}-\frac{2\pi i}{M}\right)}\right)$$

$$= \frac{1}{M}\sum_{i=0}^{M-1} H\left(e^{jM\left(\frac{\omega}{M}-\frac{2\pi i}{M}\right)}\right) X\left(e^{j\left(\frac{\omega}{M}-\frac{2\pi i}{M}\right)}\right)$$

Interchanging Operations - Compressor



Interchanging Operations - Compressor



$$Y(e^{j\omega}) = \frac{1}{M}\sum_{i=0}^{M-1} H\left(e^{jM(\frac{\omega}{M}-\frac{2\pi i}{M})}\right) X\left(e^{j(\frac{\omega}{M}-\frac{2\pi i}{M})}\right)$$

$$V(e^{j\omega}) = H(e^{j\omega M})X(e^{j\omega})$$

$$\tilde{Y}(e^{j\omega}) = \frac{1}{M}\sum_{i=0}^{M-1} V\left(e^{j(\frac{\omega}{M}-\frac{2\pi i}{M})}\right)$$

=

Interchanging Operations - Summary

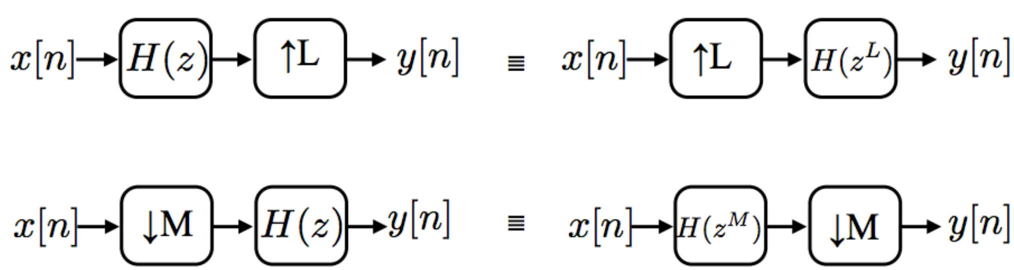
Filter and expander | Expander and expanded filter*

Compressor and filter | Expanded filter* and compressor

*Expanded filter = expanded impulse response, compressed freq response

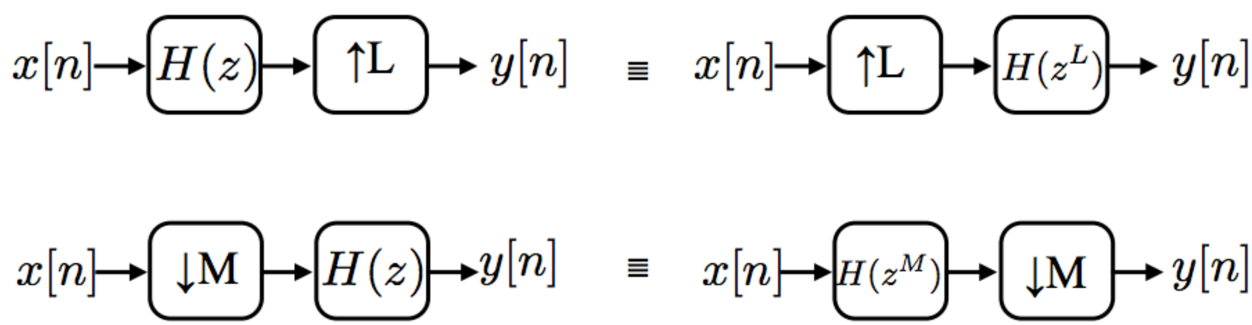
Multi-Rate Signal Processing

- What if we want to resample by 1.01T?
  - Expand by L=100
  - Filter π/101 ($$$$$)
  - Compress by M=101



Big Ideas

- Non-integer Resampling
- Multi-Rate Processing
  - Interchanging Operations



Admin

- HW 4 due Sunday

- Tania Friday office hours shifted to Saturday
  - Same time
  - Same Link on Piazza