

ESE 531: Digital Signal Processing

Lec 10: February 18, 2020

Non-Integer and Multi-rate Sampling

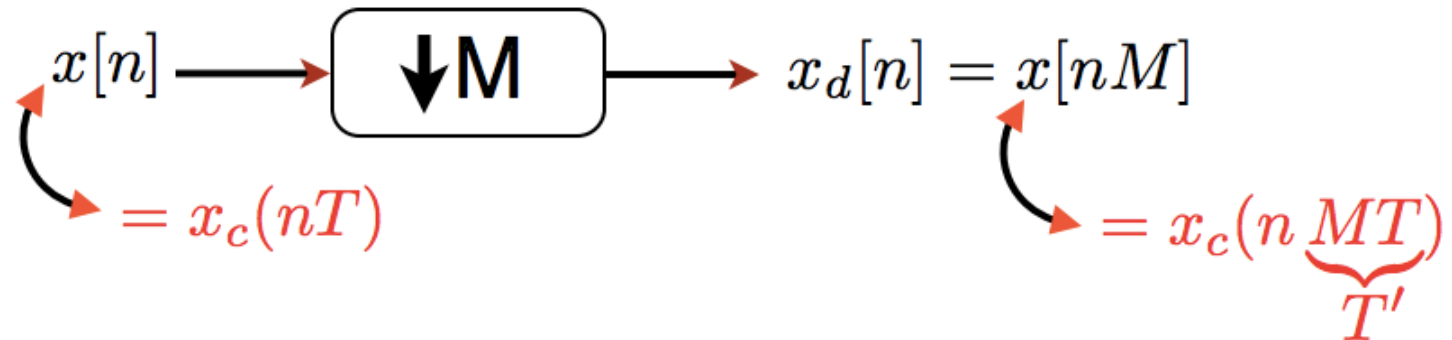


Lecture Outline

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

Downsampling

- Definition: Reducing the sampling rate by an integer number

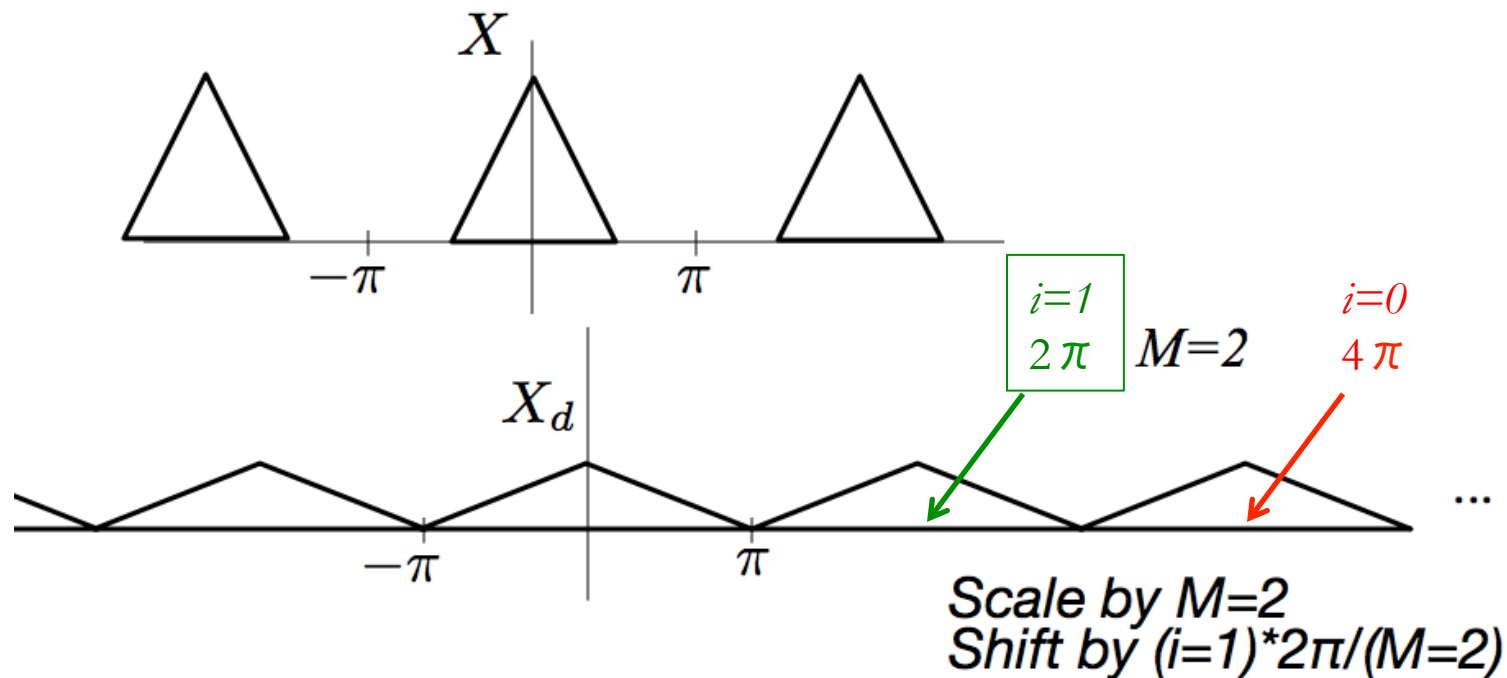


$$X_d(e^{j\omega}) = \frac{1}{M} \sum_{i=0}^{M-1} X\left(e^{j\left(\frac{\omega}{M} - \frac{2\pi}{M}i\right)}\right)$$

stretch by M replicate

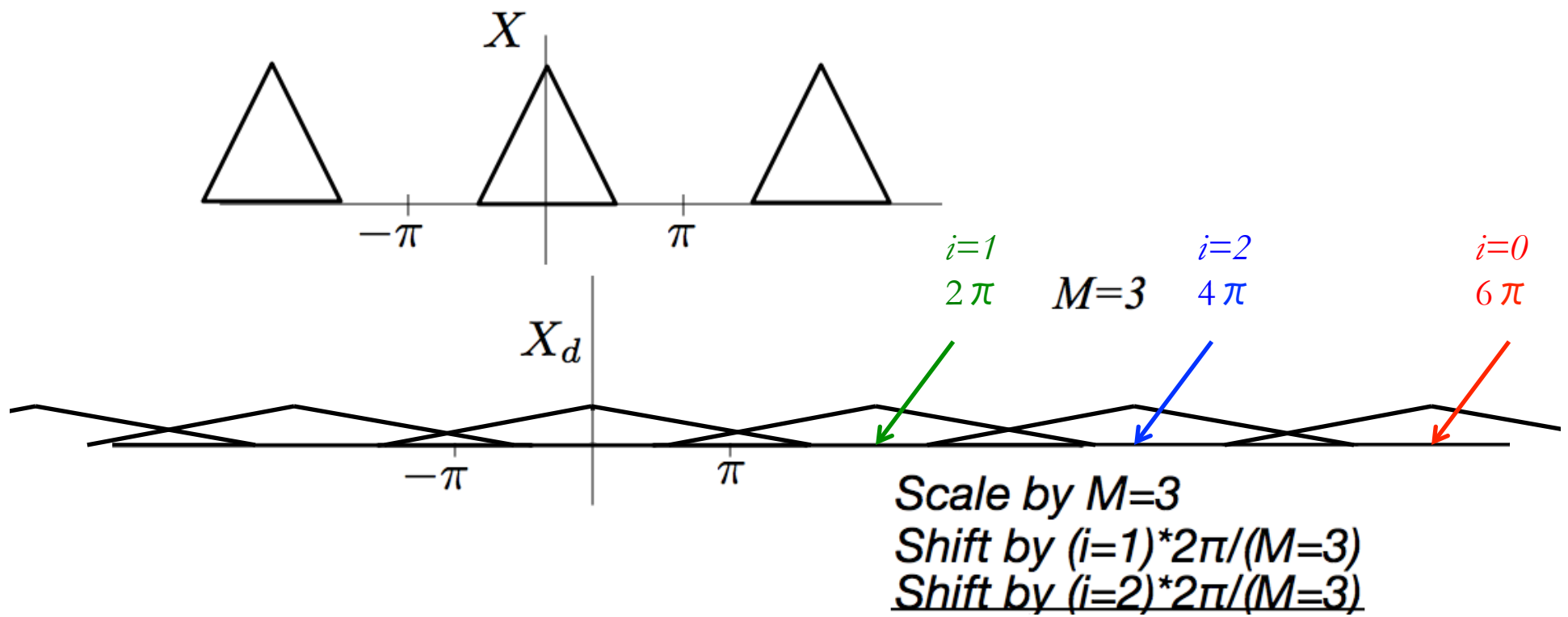
Example

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

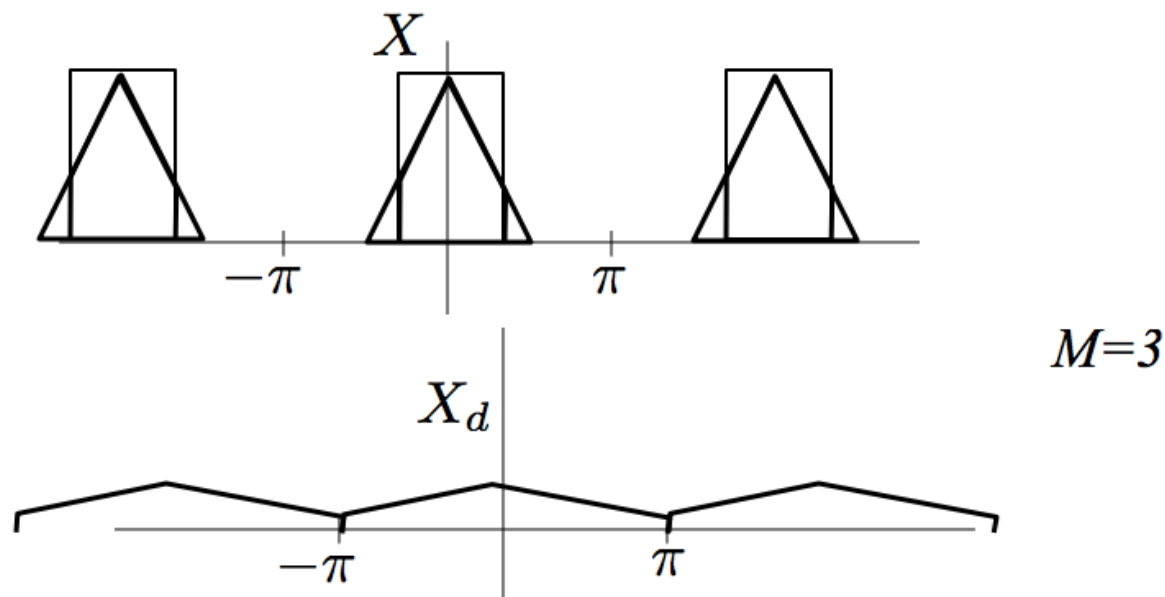
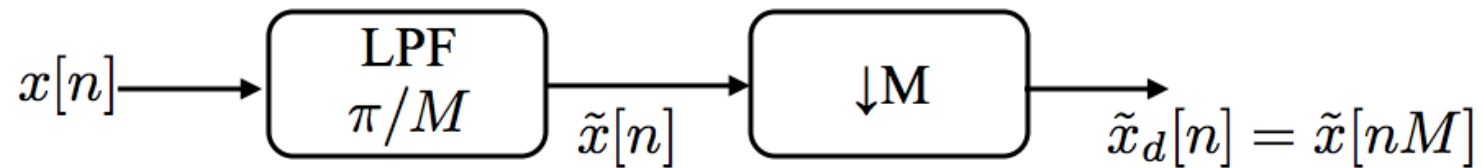


Example

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



Example





Upsampling

- Definition: Increasing the sampling rate by an integer number

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

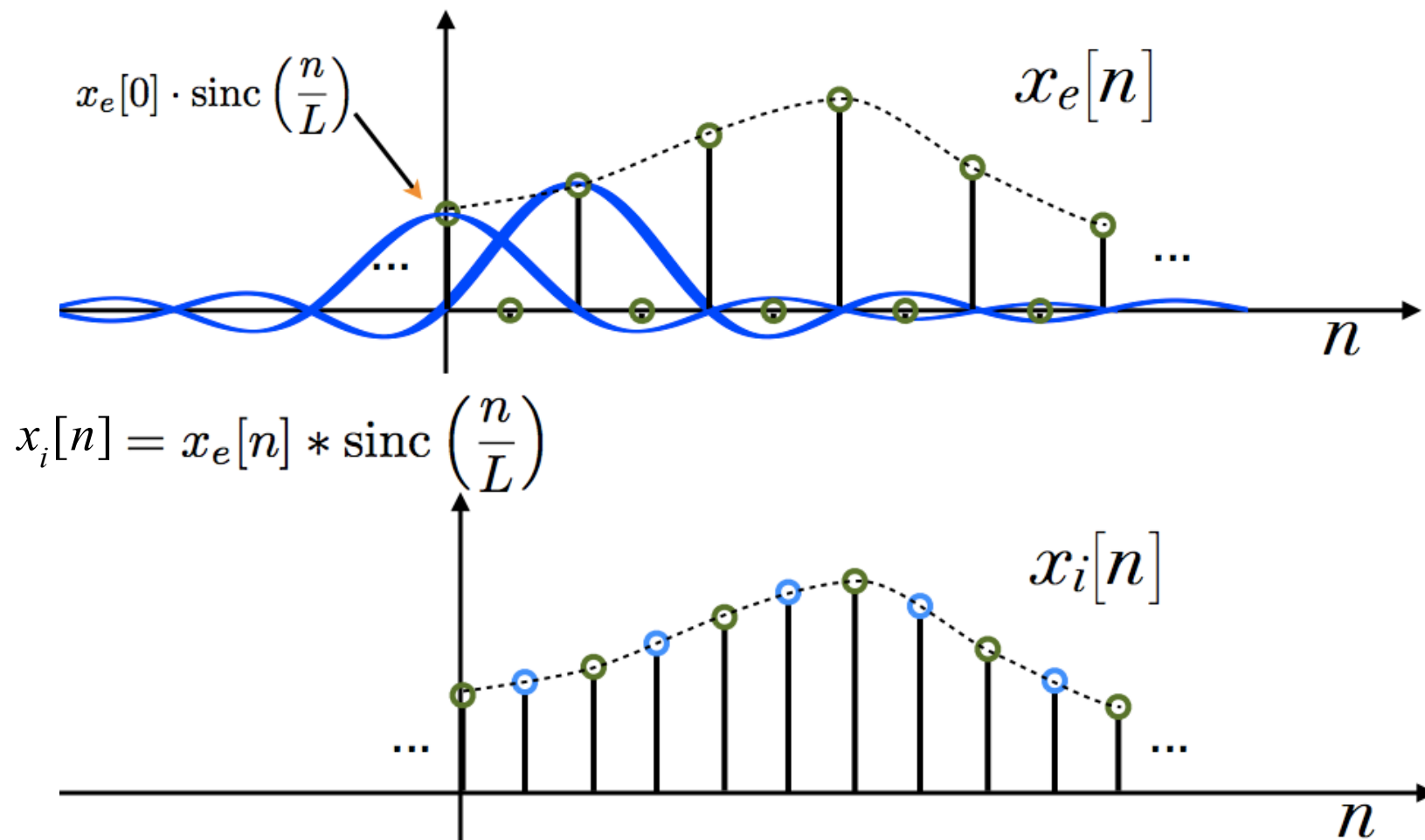
$$x_i[n] = x_c(nT') \quad \text{where } T' = \frac{T}{L} \quad L \text{ integer}$$

Obtain $x_i[n]$ from $x[n]$ in two steps:

$$(1) \text{ Generate: } x_e[n] = \begin{cases} x[n/L] & n = 0, \pm L, \pm 2L, \dots \\ 0 & \text{otherwise} \end{cases}$$

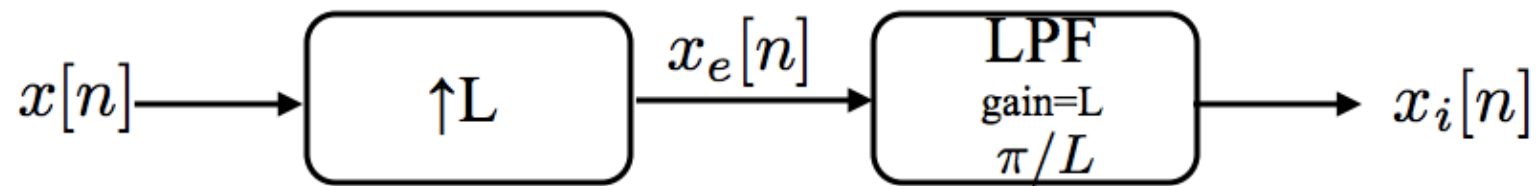
Upsampling

(2) Obtain $x_i[n]$ from $x_e[n]$ by bandlimited interpolation:



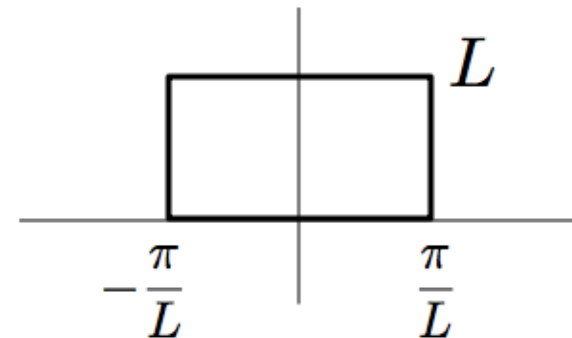
Frequency Domain Interpretation

$$x_i[n] = x_e[n] * \text{sinc}(n/L)$$

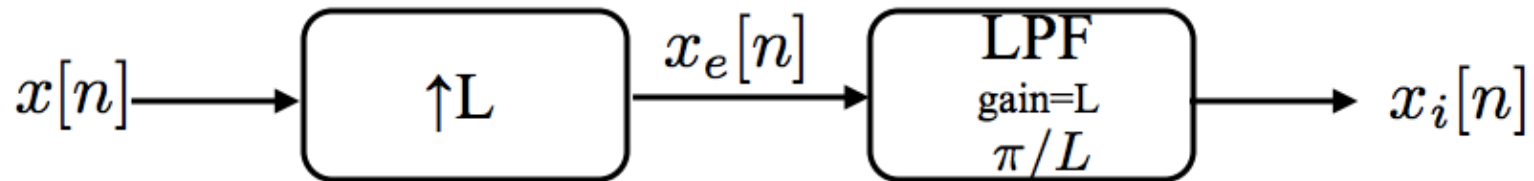


$\text{sinc}(n/L)$

DTFT $\Rightarrow$



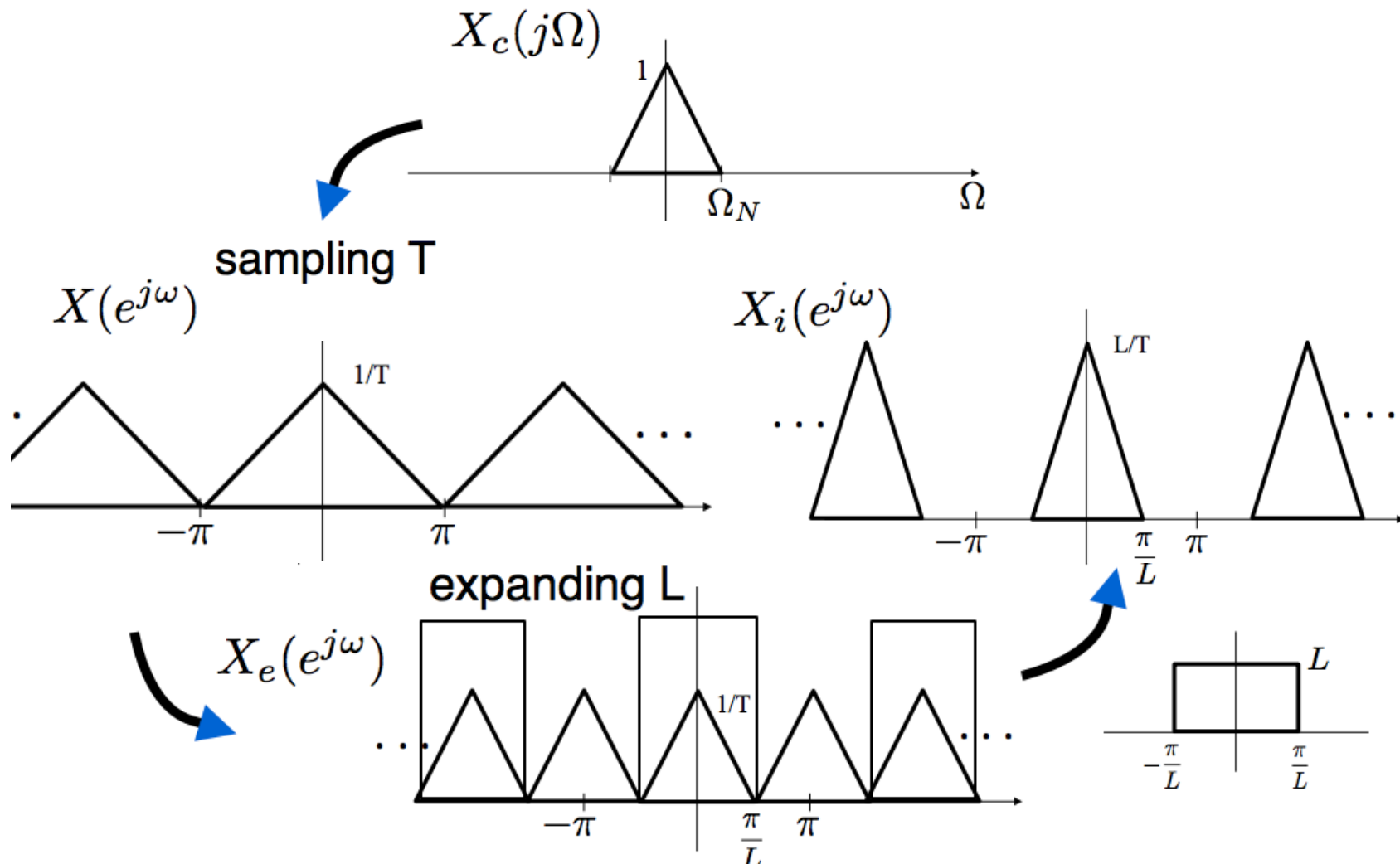
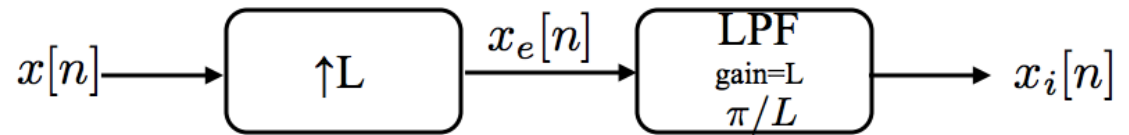
Frequency Domain Interpretation



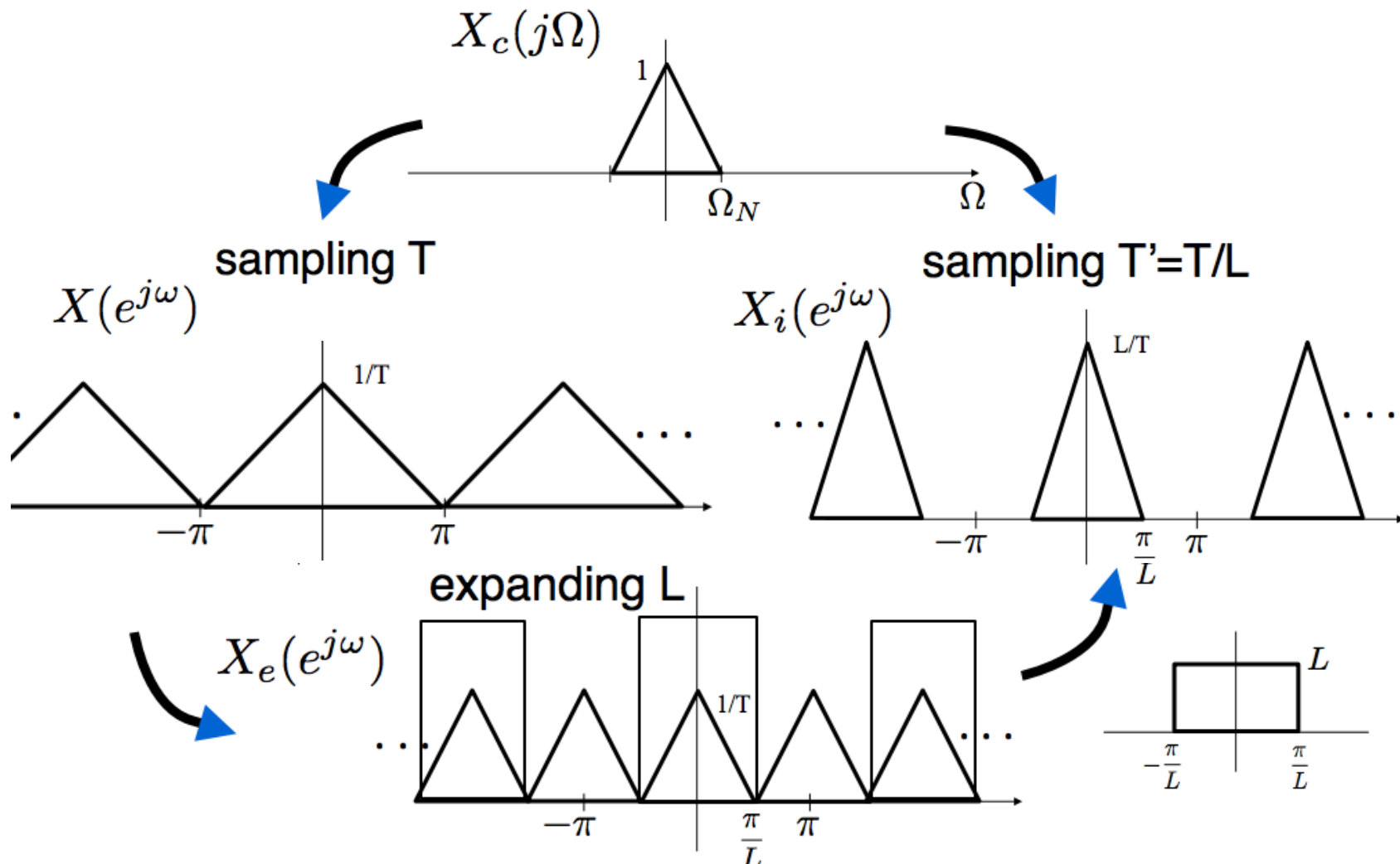
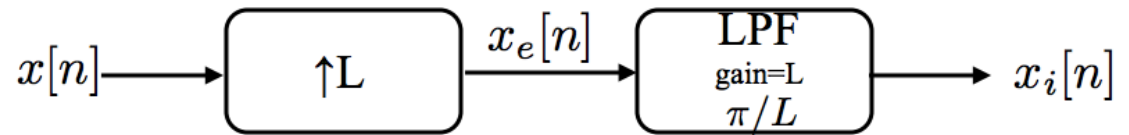
$$\begin{aligned}
 X_e(e^{j\omega}) &= \sum_{n=-\infty}^{\infty} \underbrace{x_e[n]}_{\substack{\neq 0 \text{ only for } n=mL \\ \text{(integer } m)}} e^{-j\omega n} \\
 &= \sum_{m=-\infty}^{\infty} \underbrace{x_e[mL]}_{=x[m]} e^{-j\omega mL} = X(e^{j\omega L})
 \end{aligned}$$

Compress DTFT by a factor of L!

Example



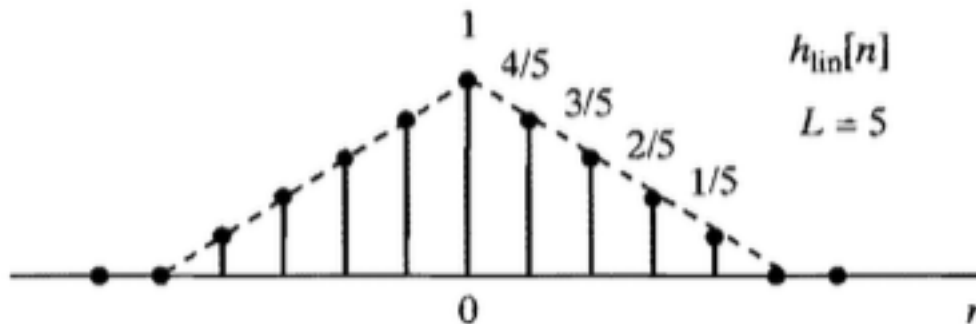
Example



Practical Interpolation

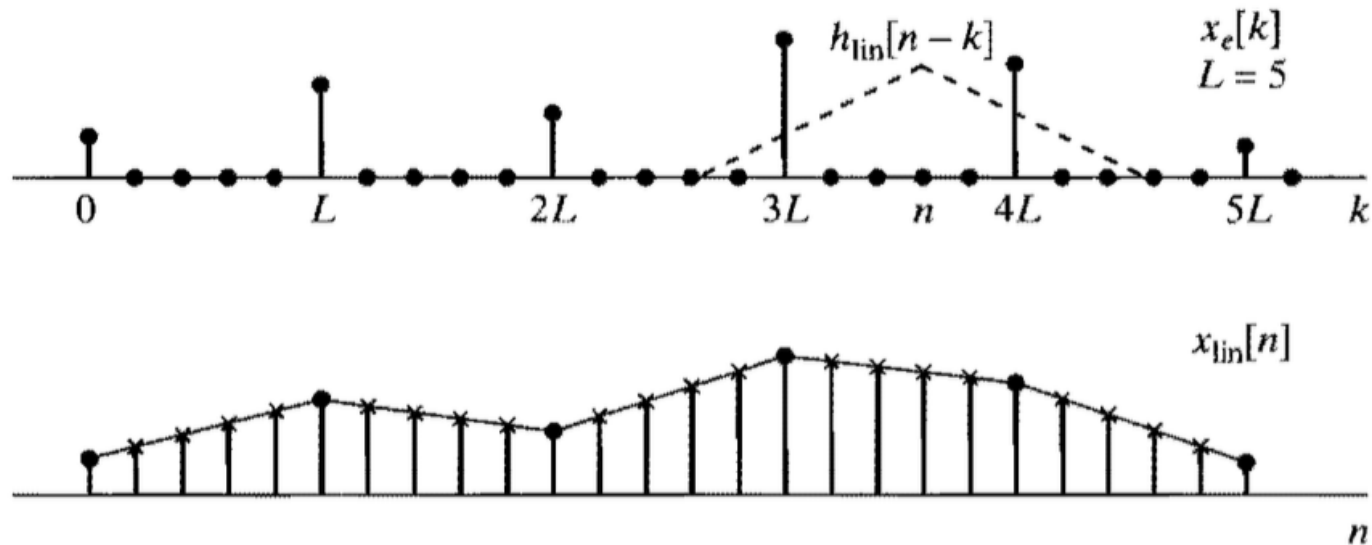
- ❑ Interpolate with simple, practical filters
 - Linear interpolation – samples between original samples fall on a straight line connecting the samples
 - Convolve with triangle instead of sinc

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



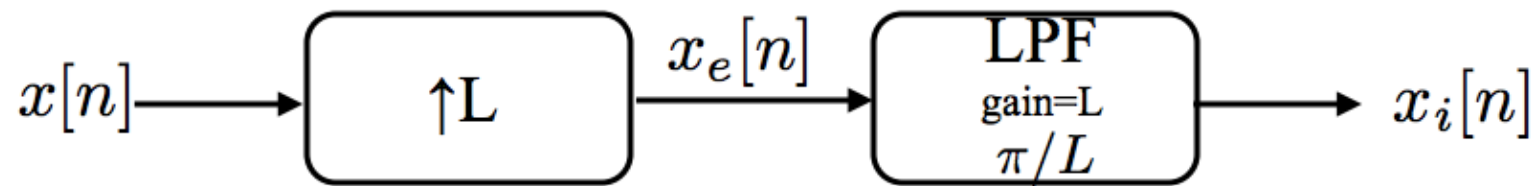
Practical Interpolation

- ❑ Interpolate with simple, practical filters
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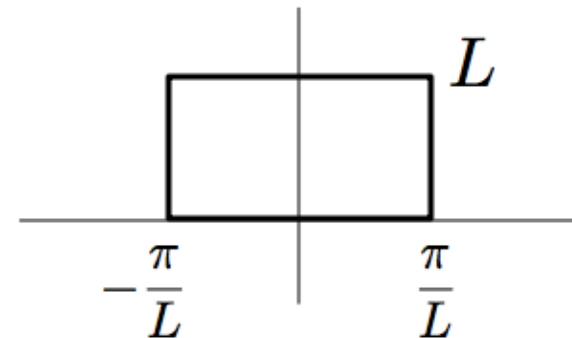
Frequency Domain Interpretation

$$x_i[n] = x_e[n] * \text{sinc}(n/L)$$



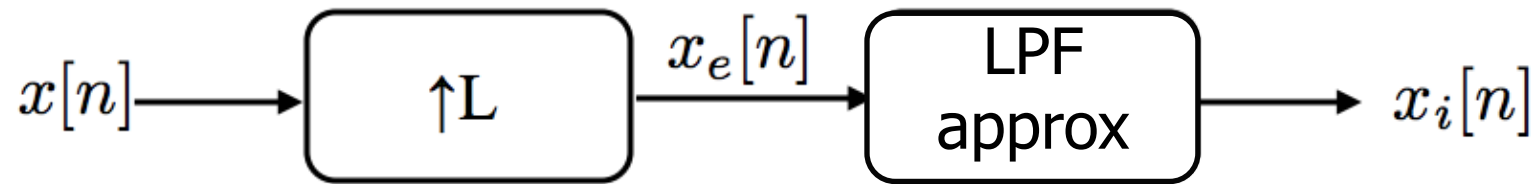
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DTFT $\Rightarrow$

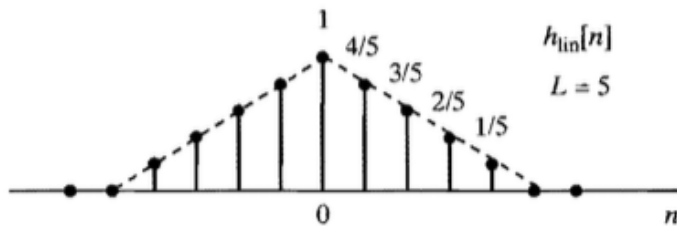


Linear Interpolation -- Frequency Domain

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

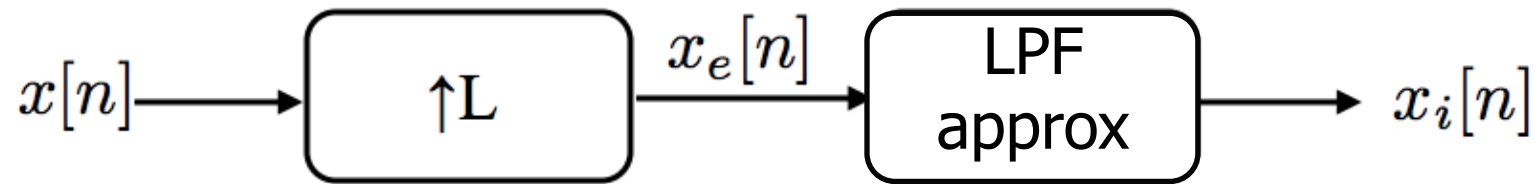


$$h_{lin}[n] = \begin{cases} 1 - |n|/L, & |n| \leq L, \\ 0, & \text{otherwise,} \end{cases}$$

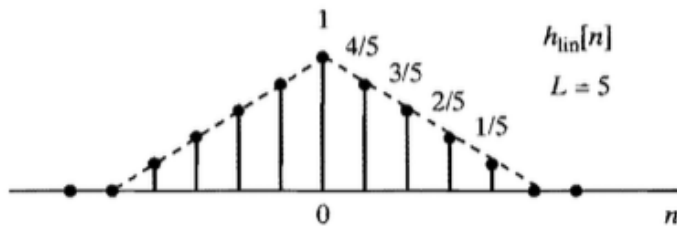


Linear Interpolation -- Frequency Domain

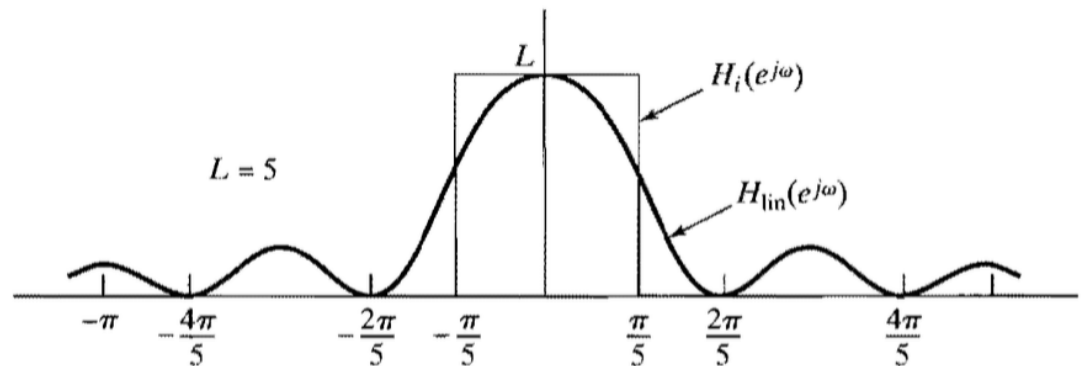
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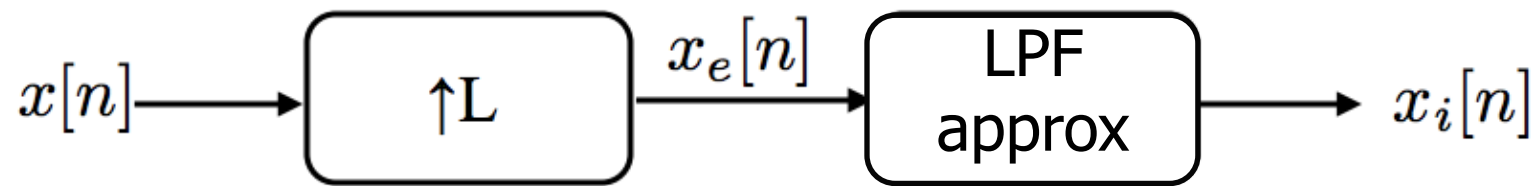


DTFT $\Rightarrow$

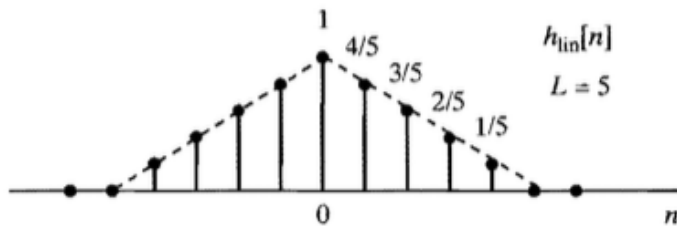


Linear Interpolation -- Frequency Domain

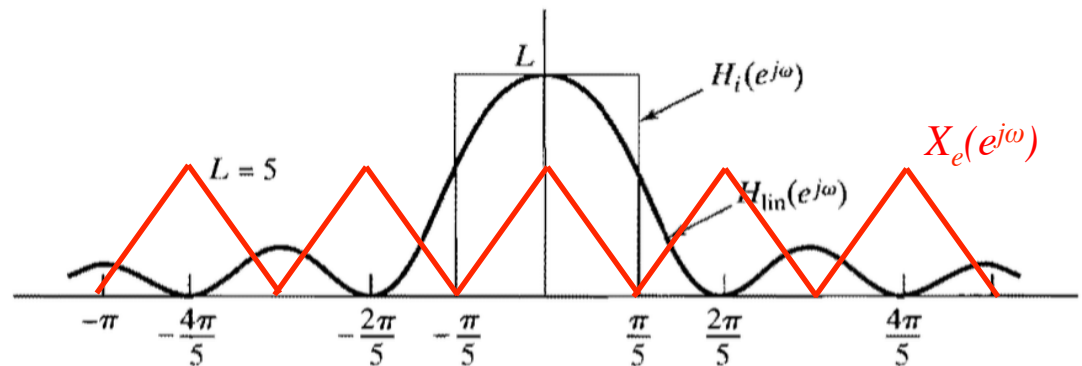
$$x_i[n] = x_e[n] * h_{lin}[n]$$



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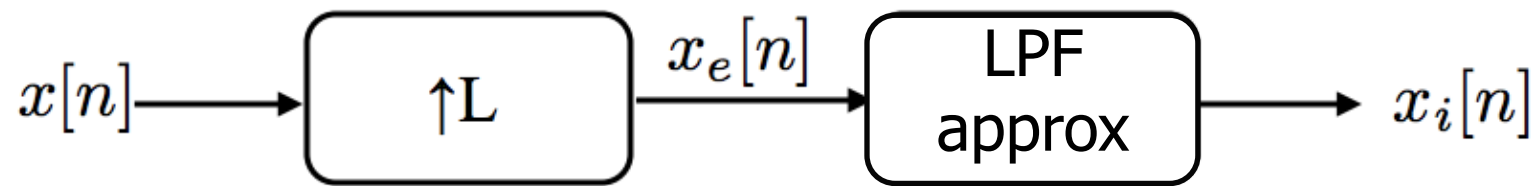


DTFT $\Rightarrow$

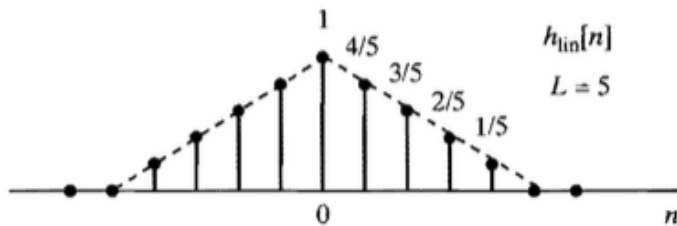


Linear Interpolation -- Frequency Domain

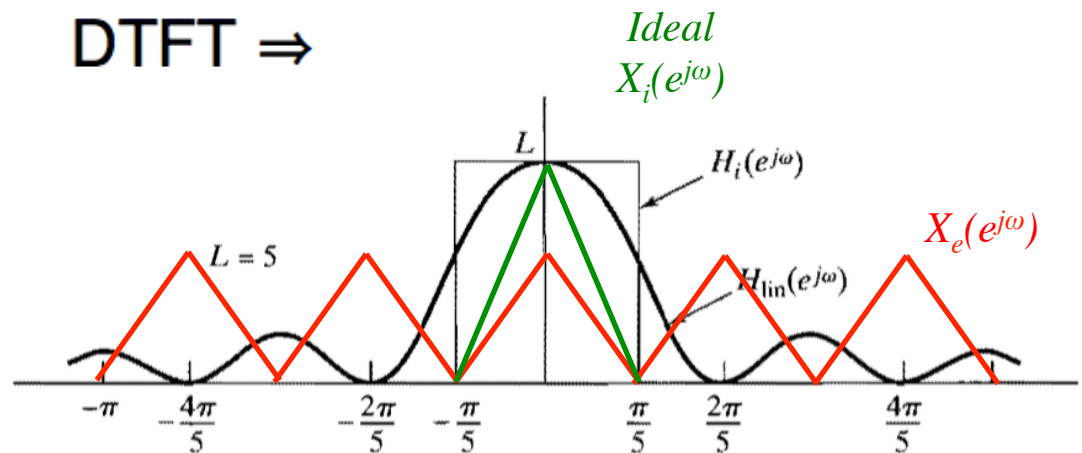
$$x_i[n] = x_e[n] * h_{lin}[n]$$



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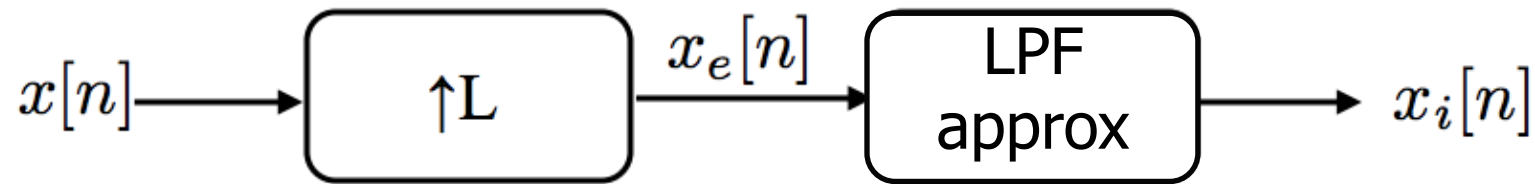


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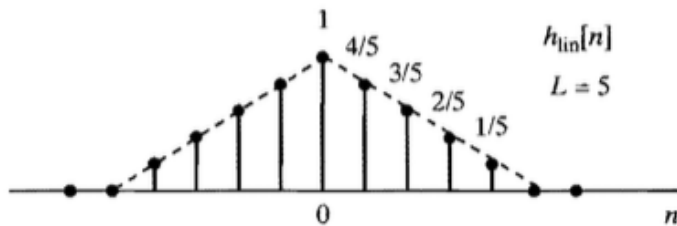


Linear Interpolation -- Frequency Domain

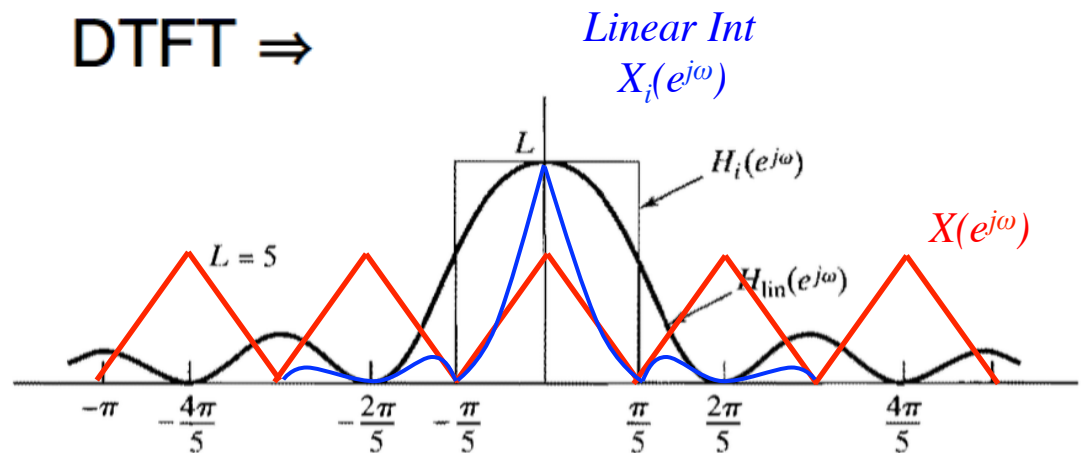
$$x_i[n] = x_e[n] * h_{lin}[n]$$



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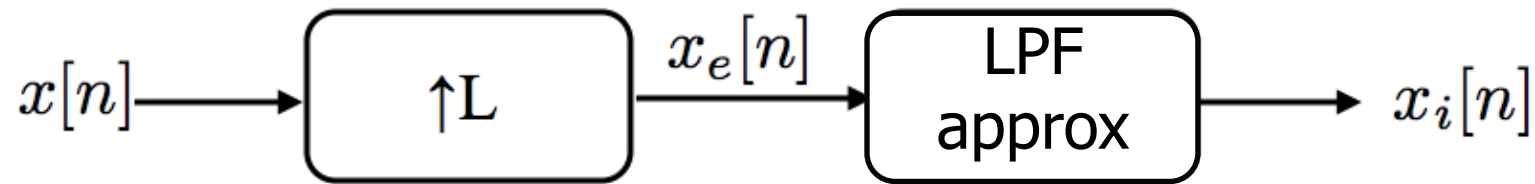


DTFT $\Rightarrow$

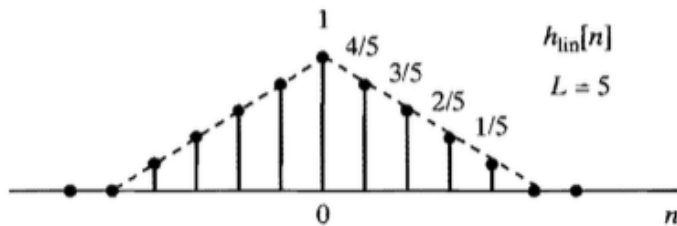


Linear Interpolation -- Frequency Domain

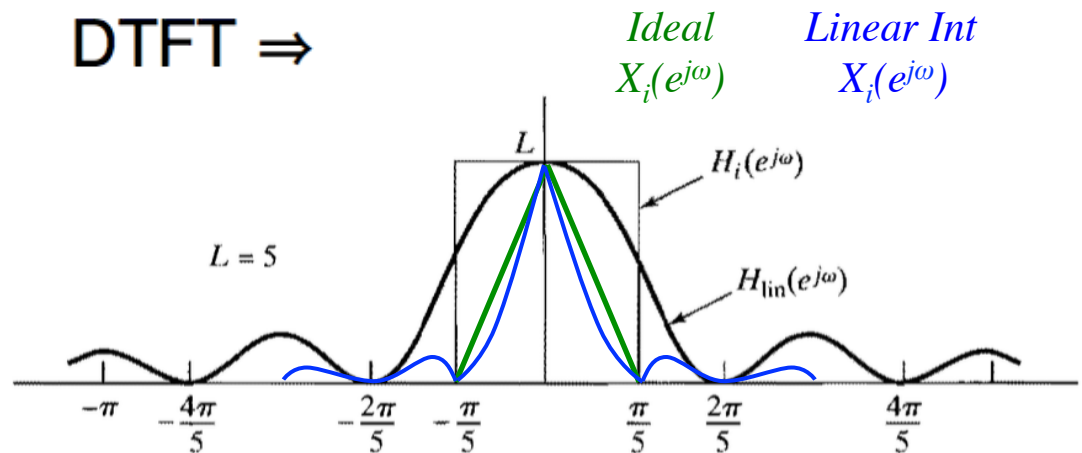
$$x_i[n] = x_e[n] * h_{lin}[n]$$



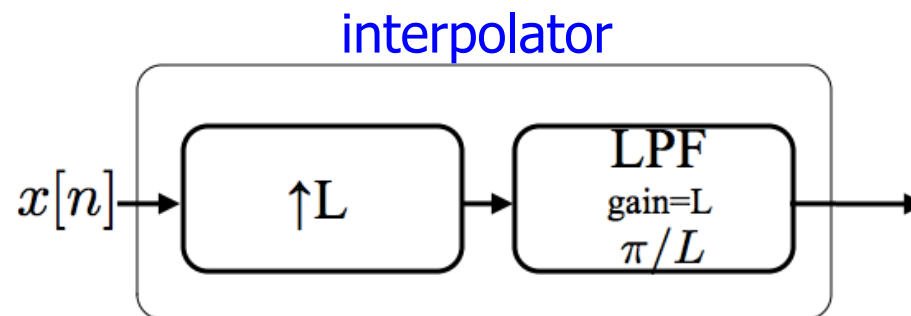
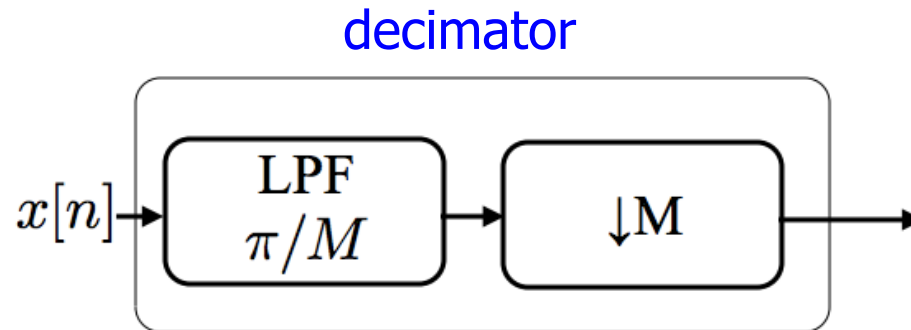
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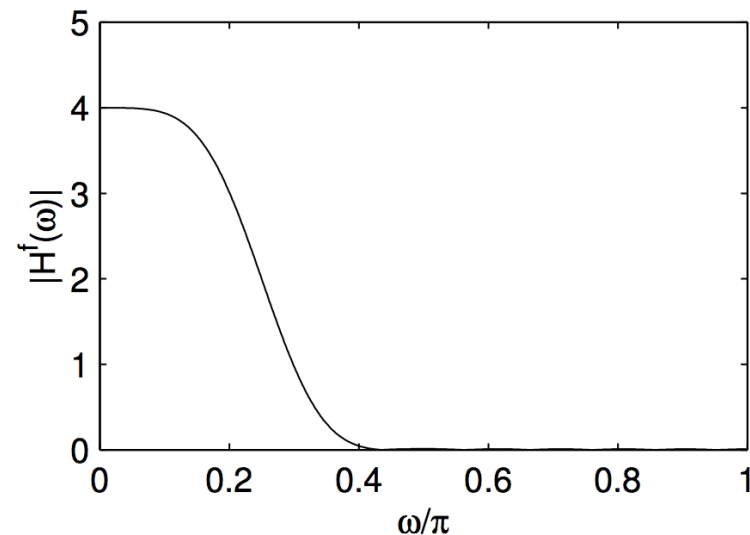
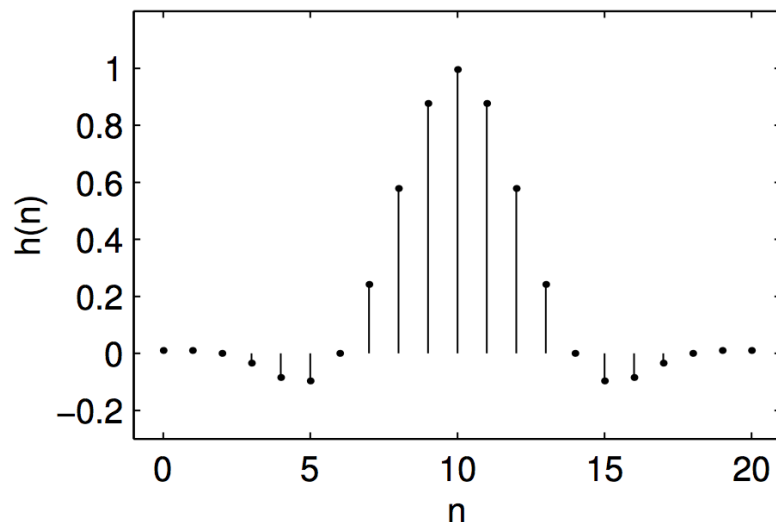


Interpolation and Decimation



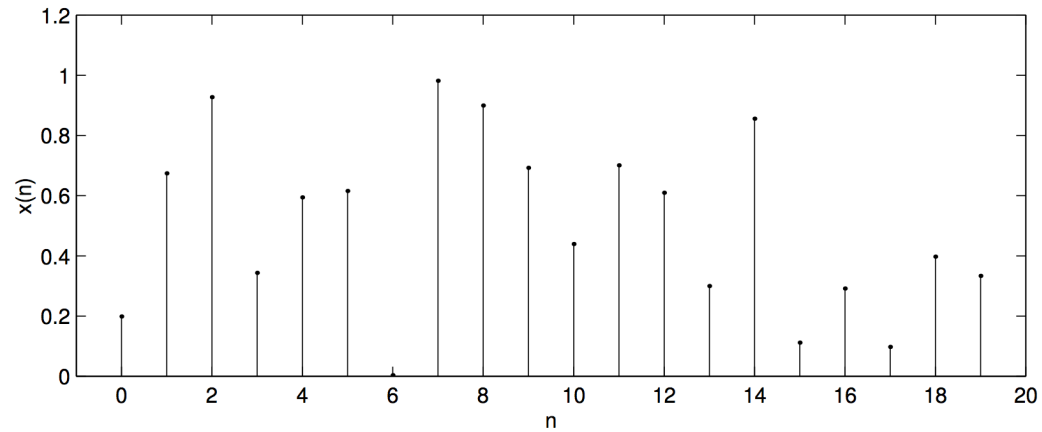
Interpolation Filter Example 1

- ❑ 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.



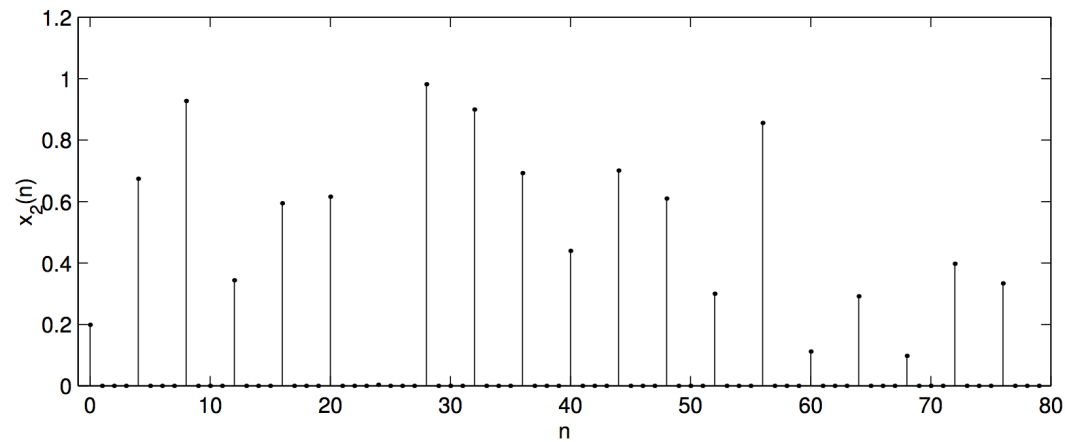
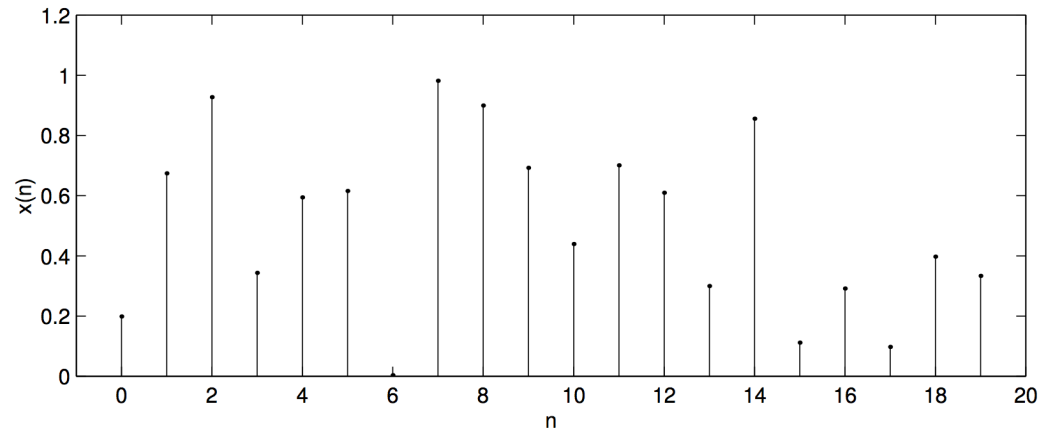


Interpolation Filter Example 1

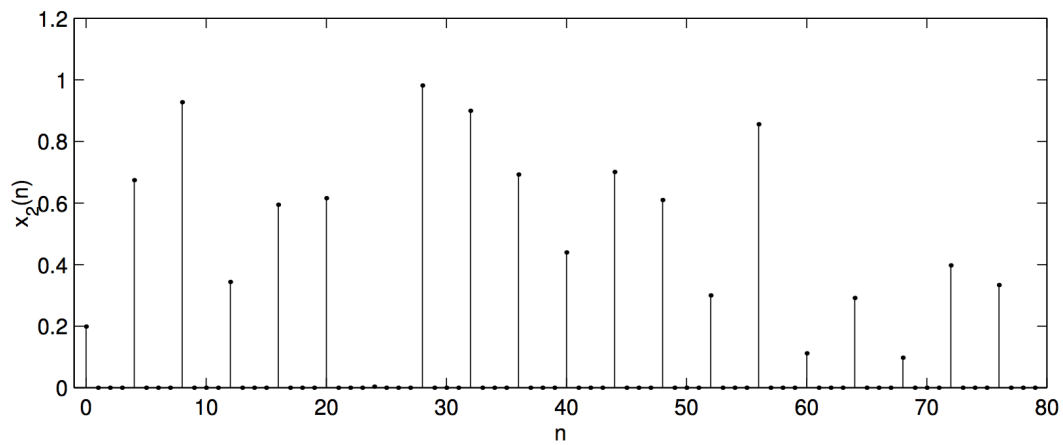




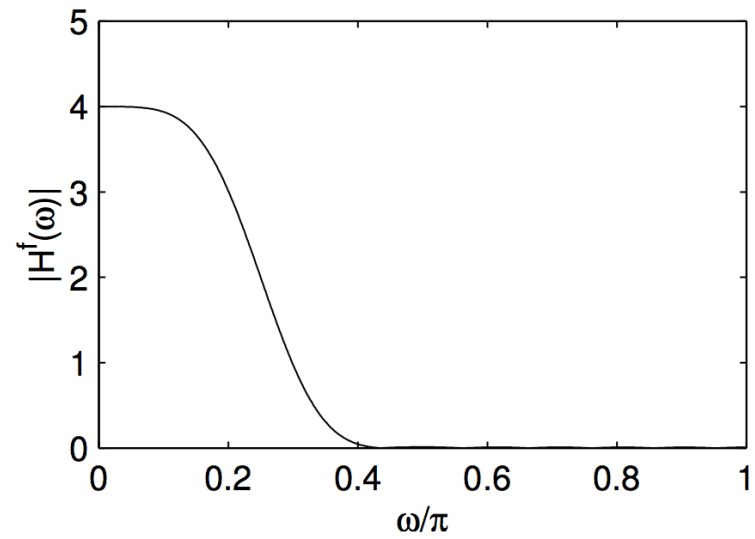
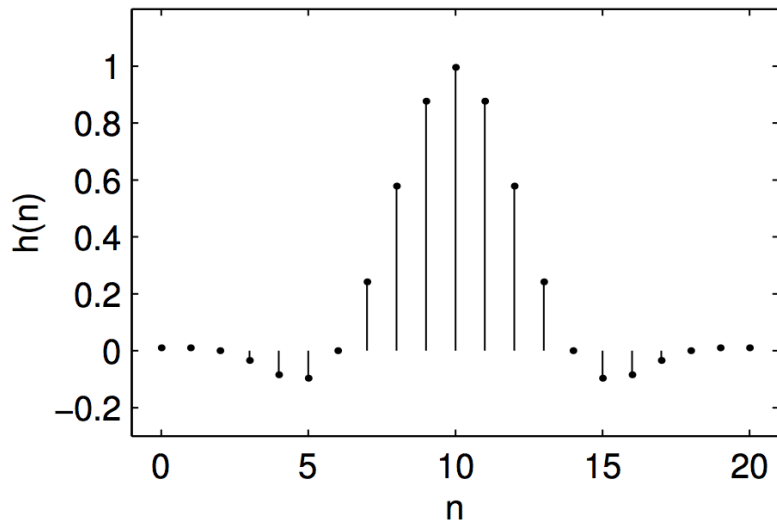
Interpolation Filter Example 1



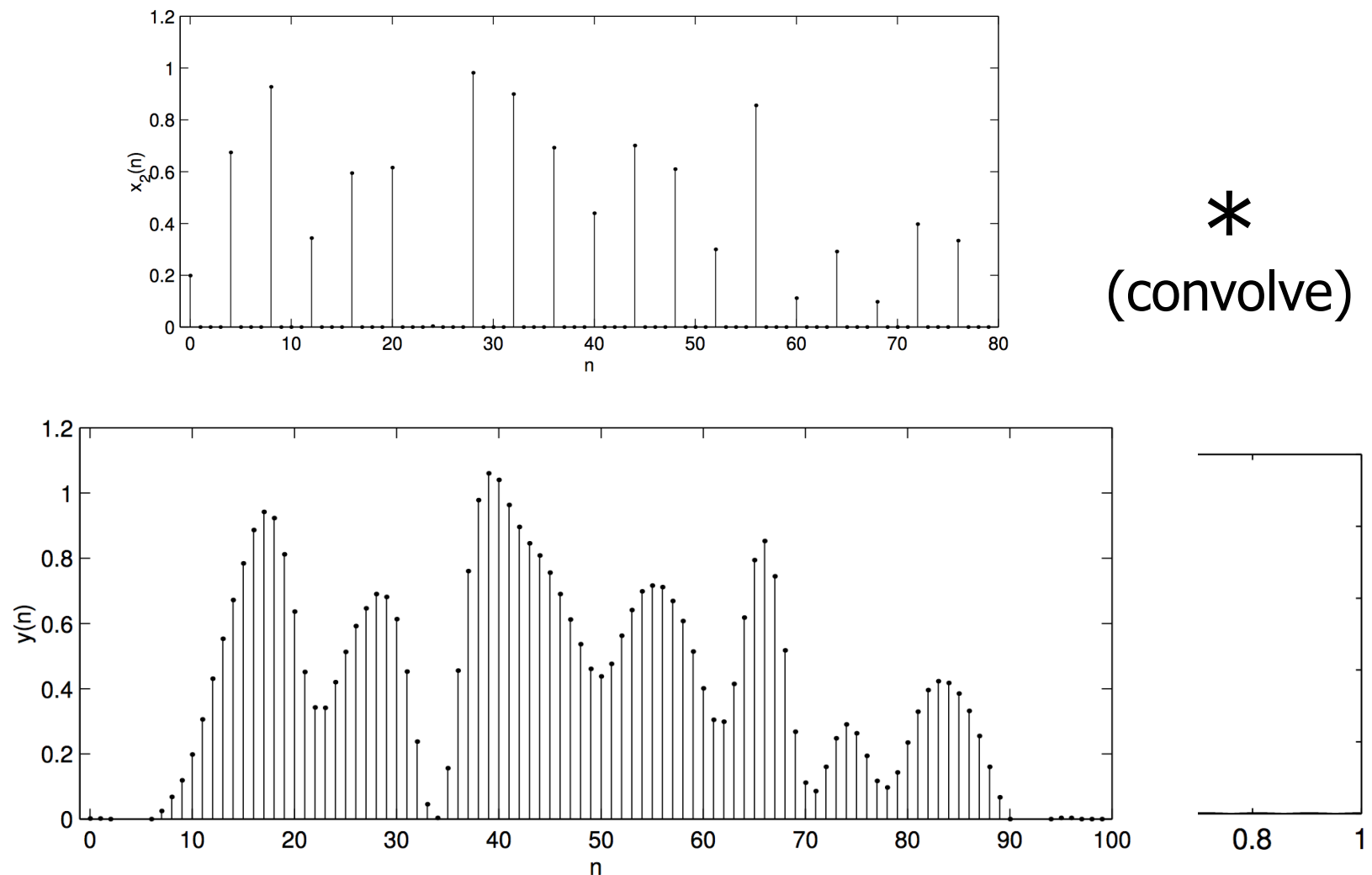
Interpolation Filter Example 1



$*$
(convolve)

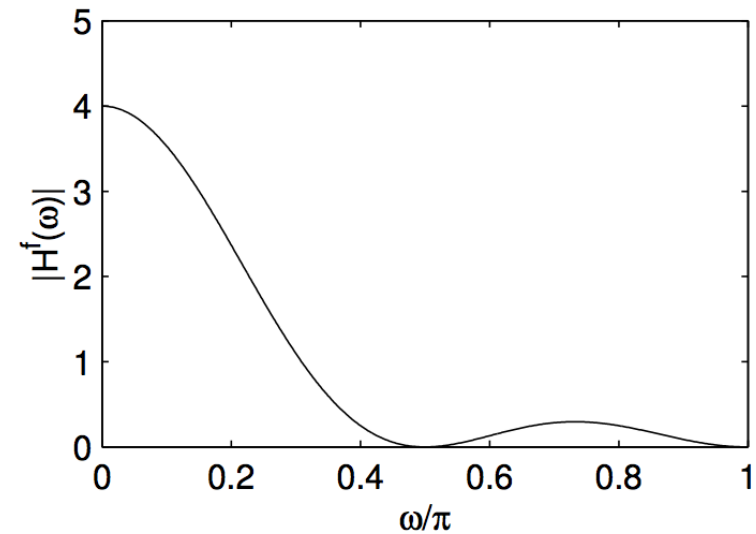
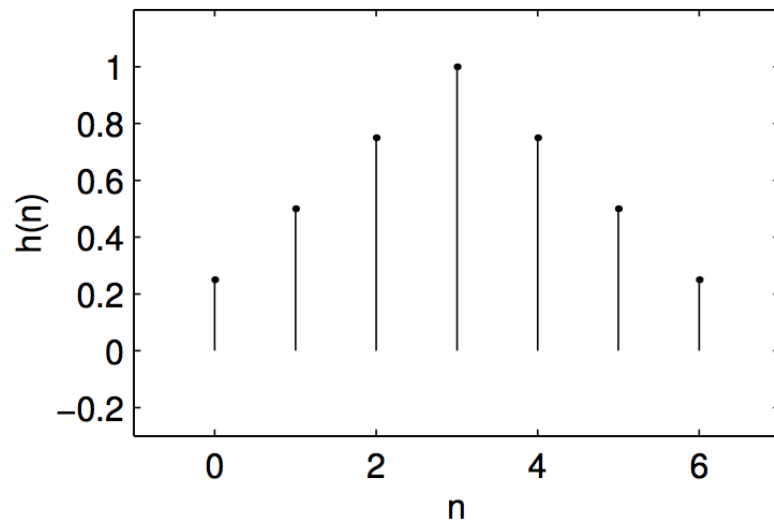


Interpolation Filter Example 1



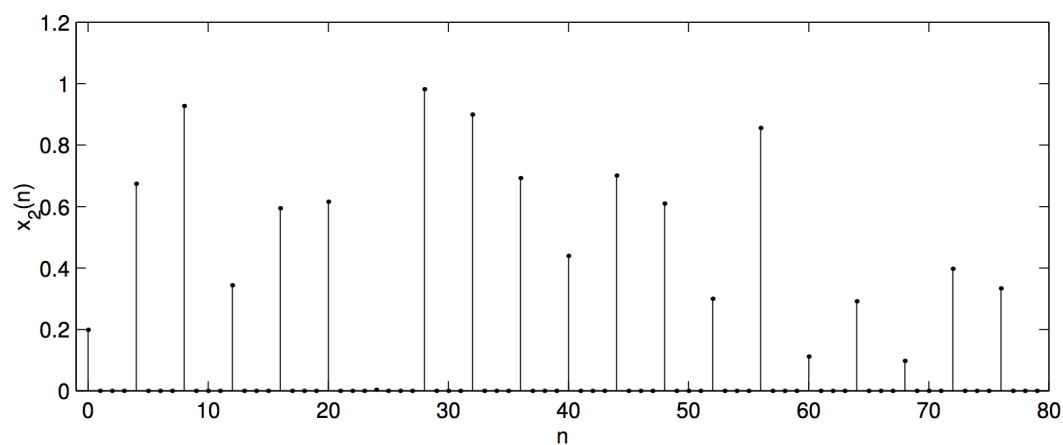
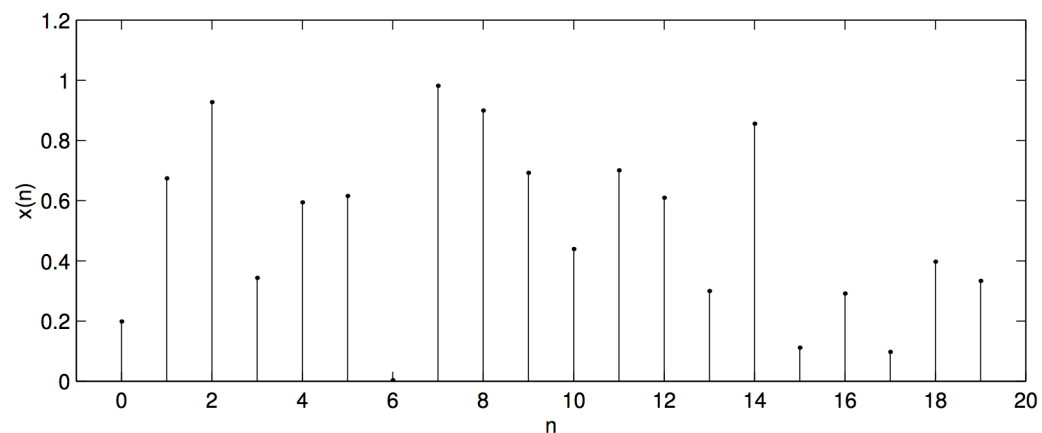
Interpolation Filter Example 2

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

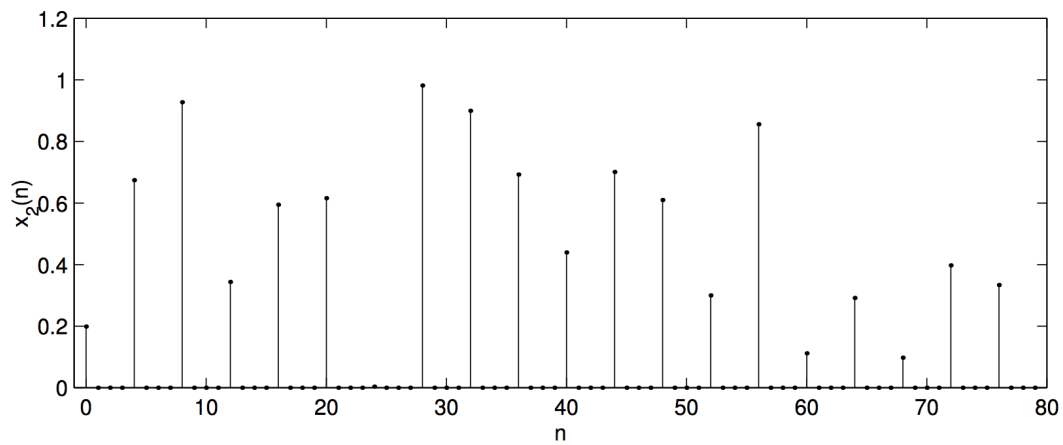




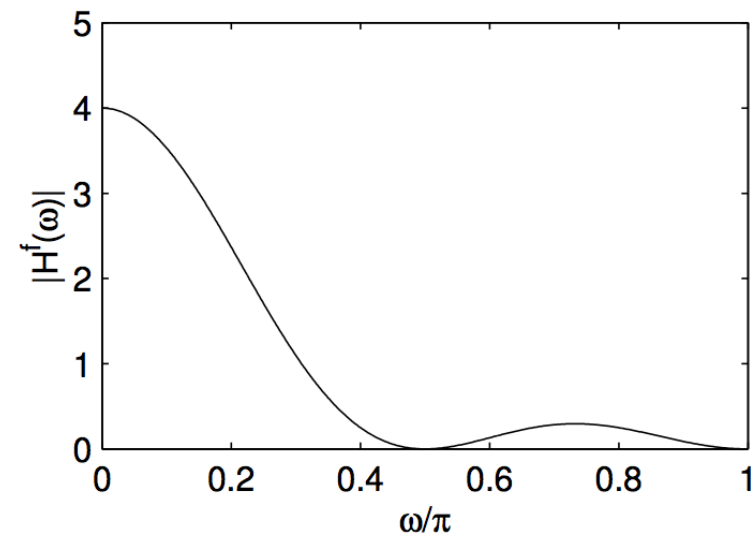
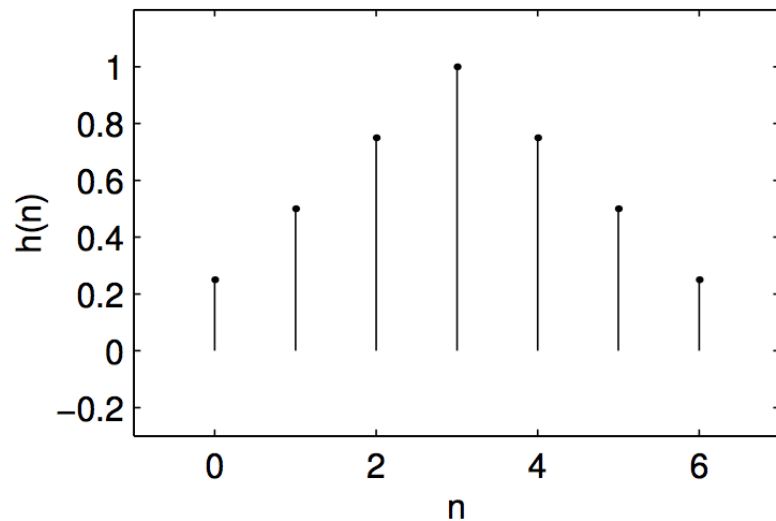
Interpolation Filter Example 2



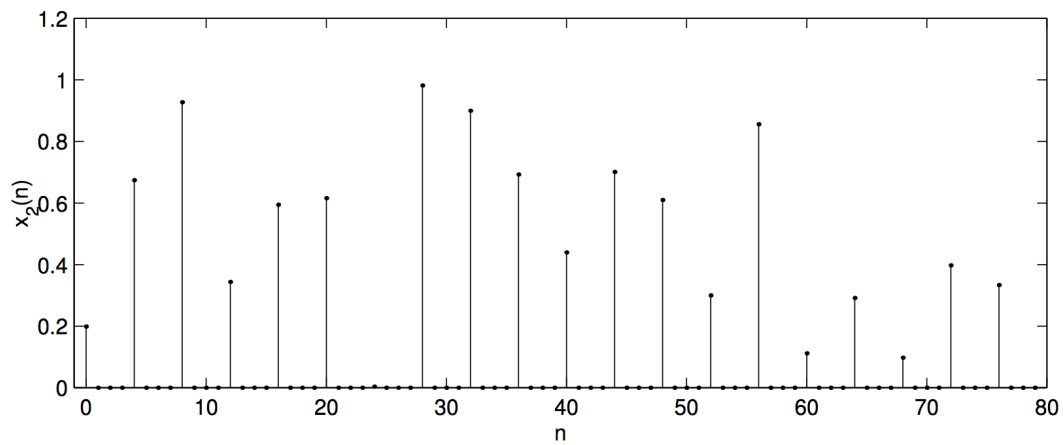
Interpolation Filter Example 2



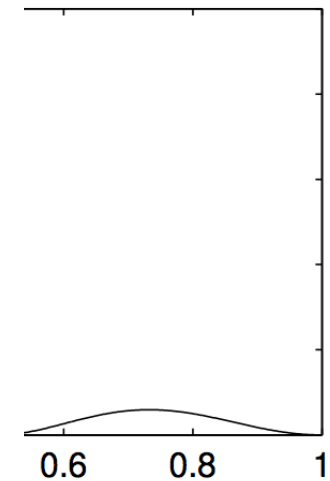
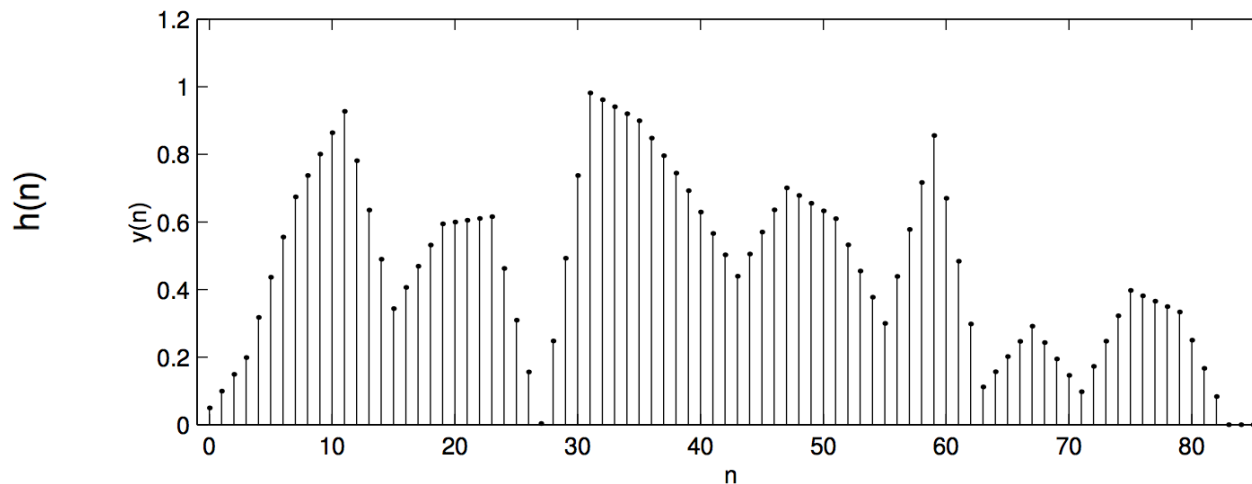
$*$
(convolve)



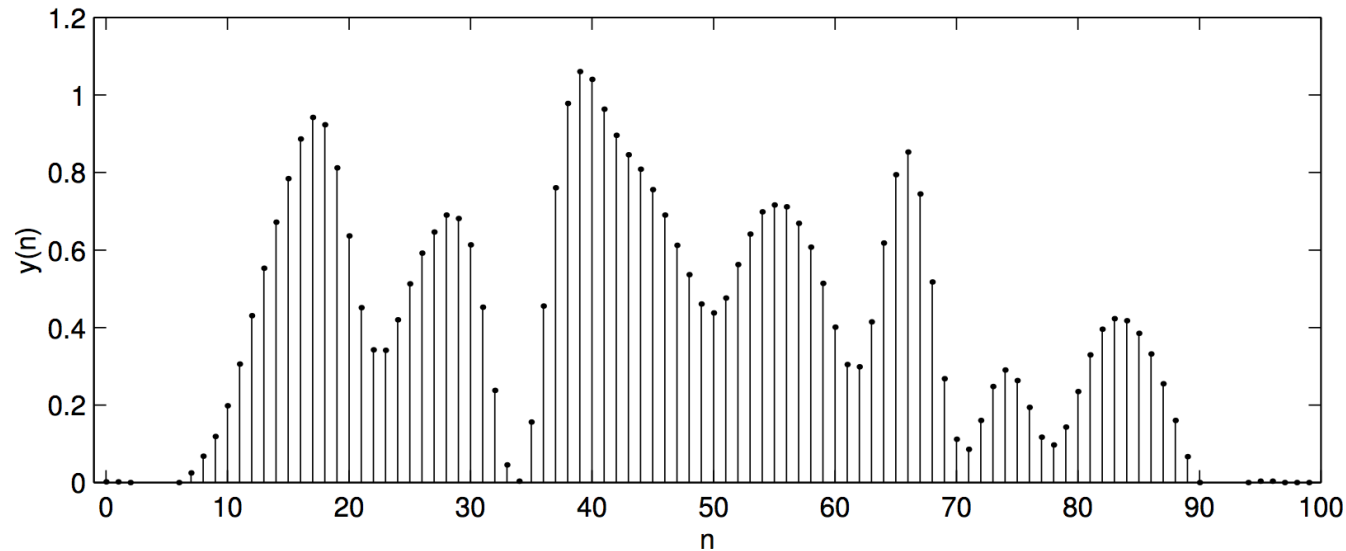
Interpolation Filter Example 2



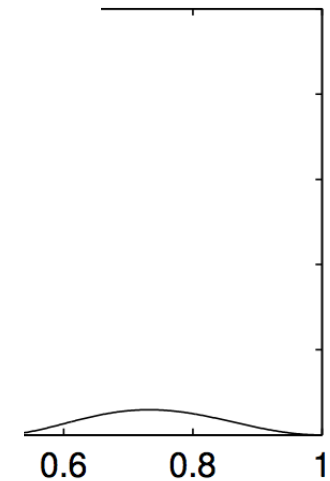
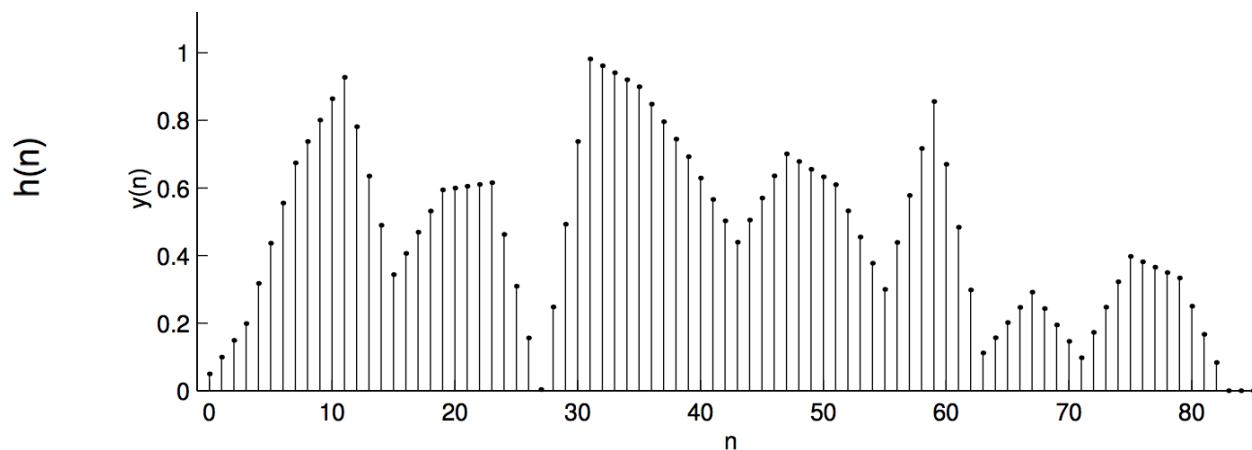
$*$
(convolve)



Interpolation Filter Example 2



$*$
nvolve)





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 $x(n)$?



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



Interpolation Filter Example 3

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- ❑ Can a filter be designed so as to preserve the original samples $x(n)$?
- ❑ 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 $y(2n + n_o) = x(n)$?

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

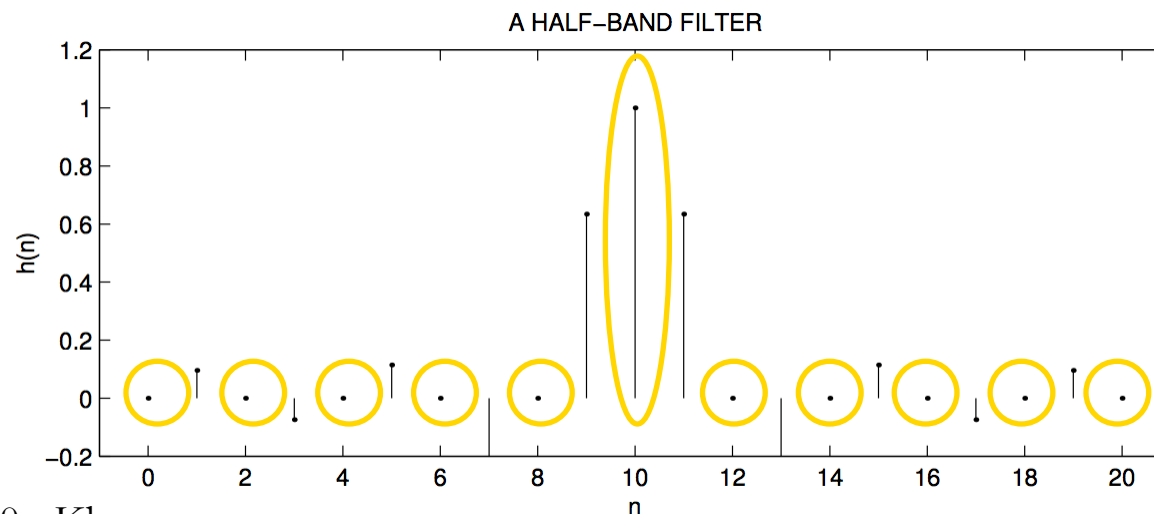
-
- A HALF-BAND FILTER
- Stem plot showing the impulse response $h(n)$ versus n . The horizontal axis n ranges from 0 to 20, and the vertical axis $h(n)$ ranges from -0.2 to 1.2. The plot shows a symmetric impulse response centered at $n=10$, with a peak value of 1.0 at $n=10$. The values at $n=9$ and $n=11$ are approximately 0.64, at $n=8$ and $n=12$ are approximately 0.0, at $n=7$ and $n=13$ are approximately -0.05, at $n=6$ and $n=14$ are approximately 0.12, at $n=5$ and $n=15$ are approximately 0.0, at $n=4$ and $n=16$ are approximately 0.0, at $n=3$ and $n=17$ are approximately -0.05, at $n=2$ and $n=18$ are approximately 0.0, at $n=1$ and $n=19$ are approximately 0.1, and at $n=0$ and $n=20$ are approximately 0.0.

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:

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

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



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

-
- A HALF-BAND FILTER
- $h(2n + n_o) = \delta(n)$
- The plot shows the impulse response $h(n)$ for a half-band filter. The x-axis represents n from 0 to 20, and the y-axis represents $h(n)$ from -0.2 to 1.2. The response is symmetric around $n=10$, where it reaches its maximum value of 1.0. The side lobes decay in amplitude as n moves away from 10.
- | n | $h(n)$ |
|-----|--------|
| 0 | 0.00 |
| 1 | 0.05 |
| 2 | 0.00 |
| 3 | -0.05 |
| 4 | 0.00 |
| 5 | 0.10 |
| 6 | 0.00 |
| 7 | -0.10 |
| 8 | 0.00 |
| 9 | 0.65 |
| 10 | 1.00 |
| 11 | 0.65 |
| 12 | 0.00 |
| 13 | -0.10 |
| 14 | 0.00 |
| 15 | 0.10 |
| 16 | 0.00 |
| 17 | -0.05 |
| 18 | 0.00 |
| 19 | 0.05 |
| 20 | 0.00 |



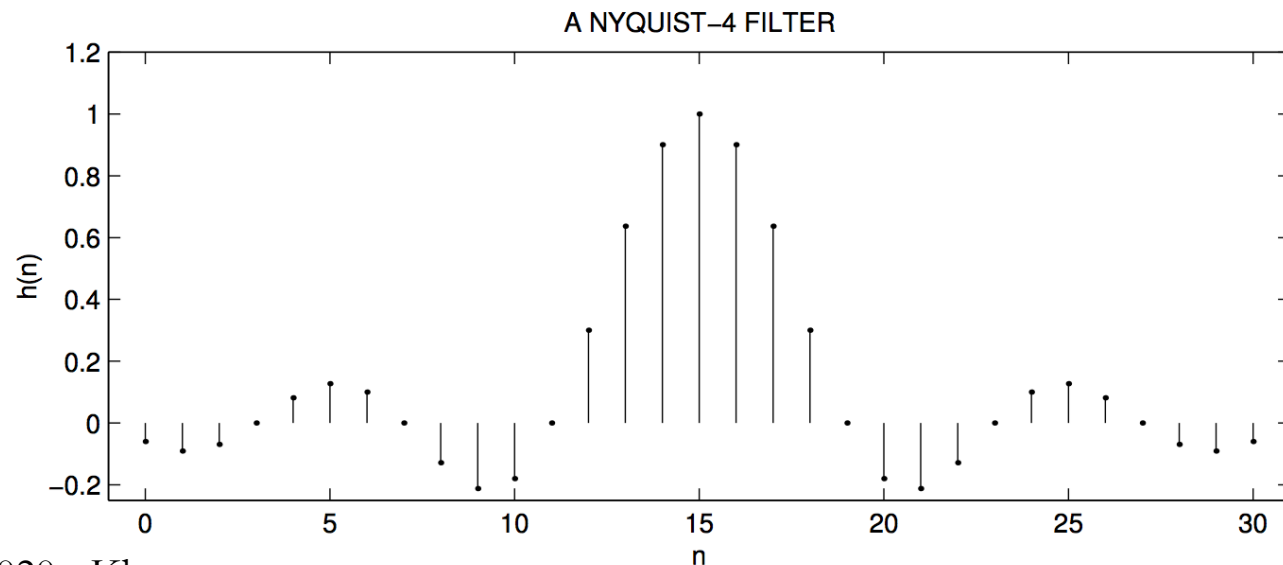
Interpolation Filter Example 4

- ❑ 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 $L > 2$.

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 $L > 2$.
- ❑ A (0-centered) Nyquist- L filter $h(n)$ is one for which

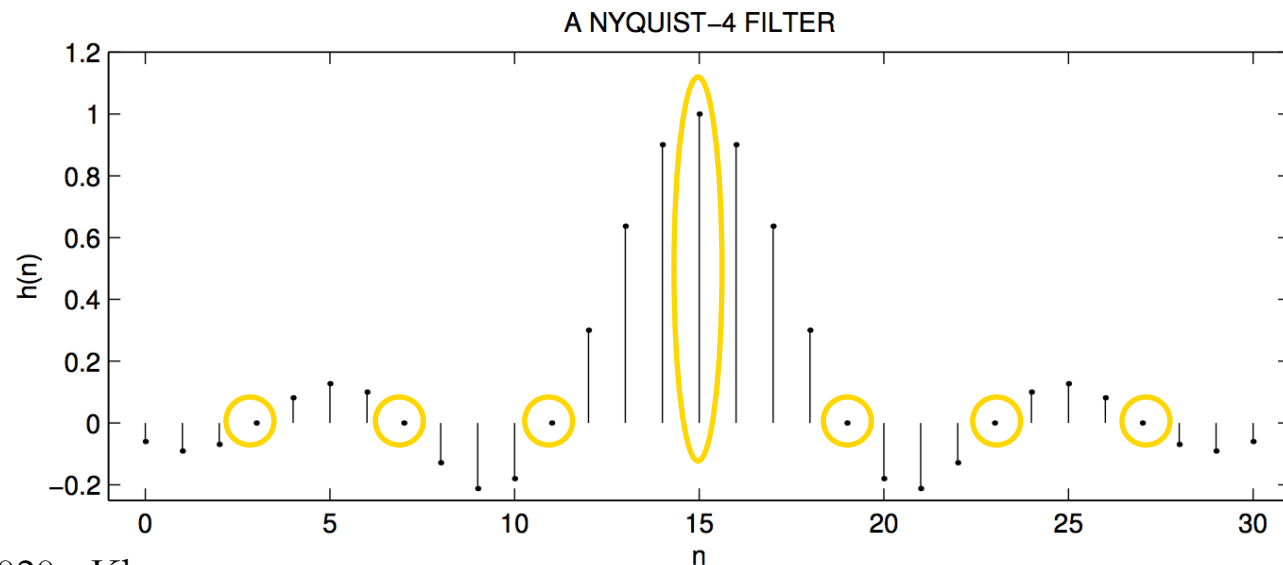
$$h(Ln) = \delta(n).$$



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 $L > 2$.
- ❑ A (0-centered) Nyquist- L filter $h(n)$ is one for which

$$h(Ln) = \delta(n).$$



Non-integer Resampling





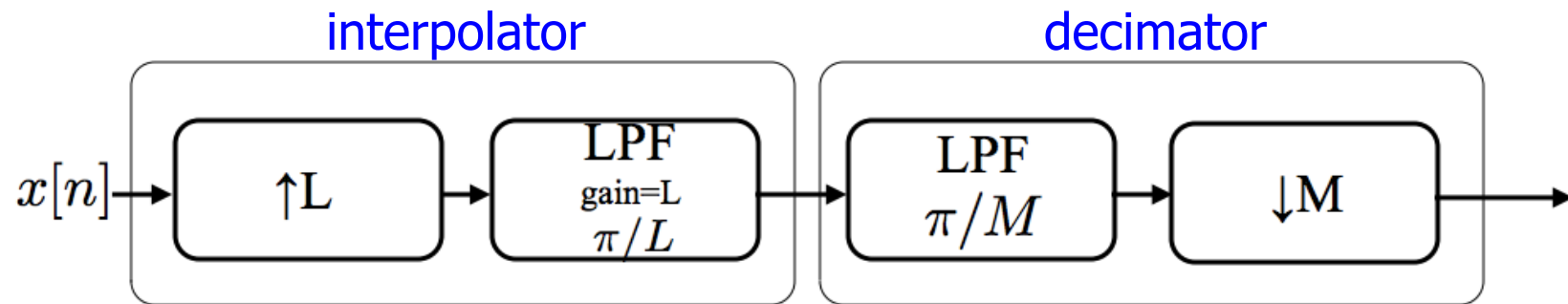
Non-integer Resampling

□ $T' = TM/L$

Non-integer Resampling

□ $T' = TM/L$

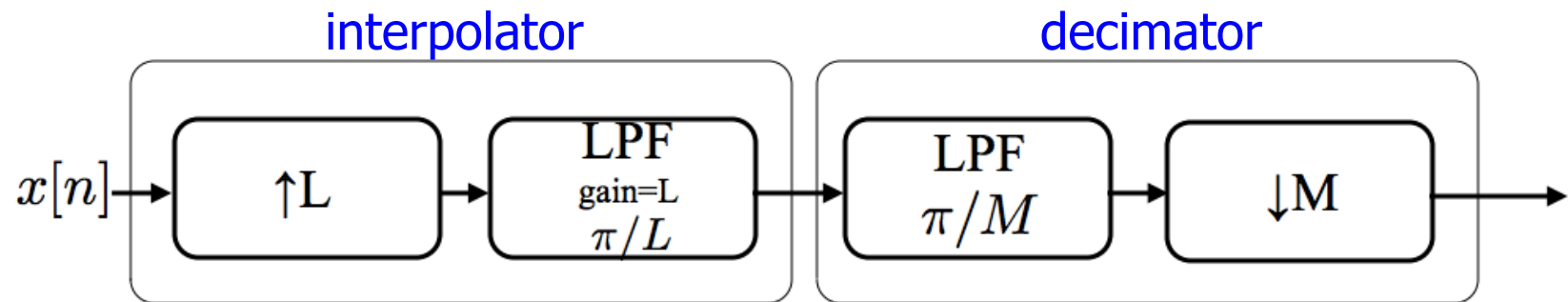
- Upsample by L , then downsample by M



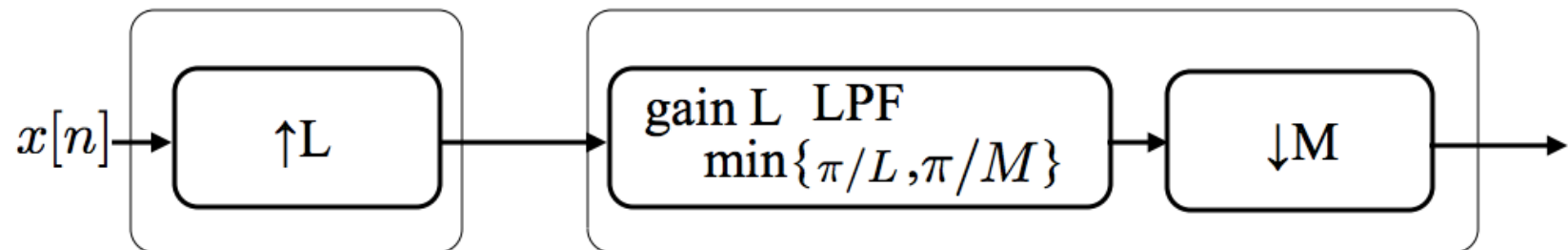
Non-integer Resampling

□ $T' = TM/L$

- Upsample by L , then downsample by M

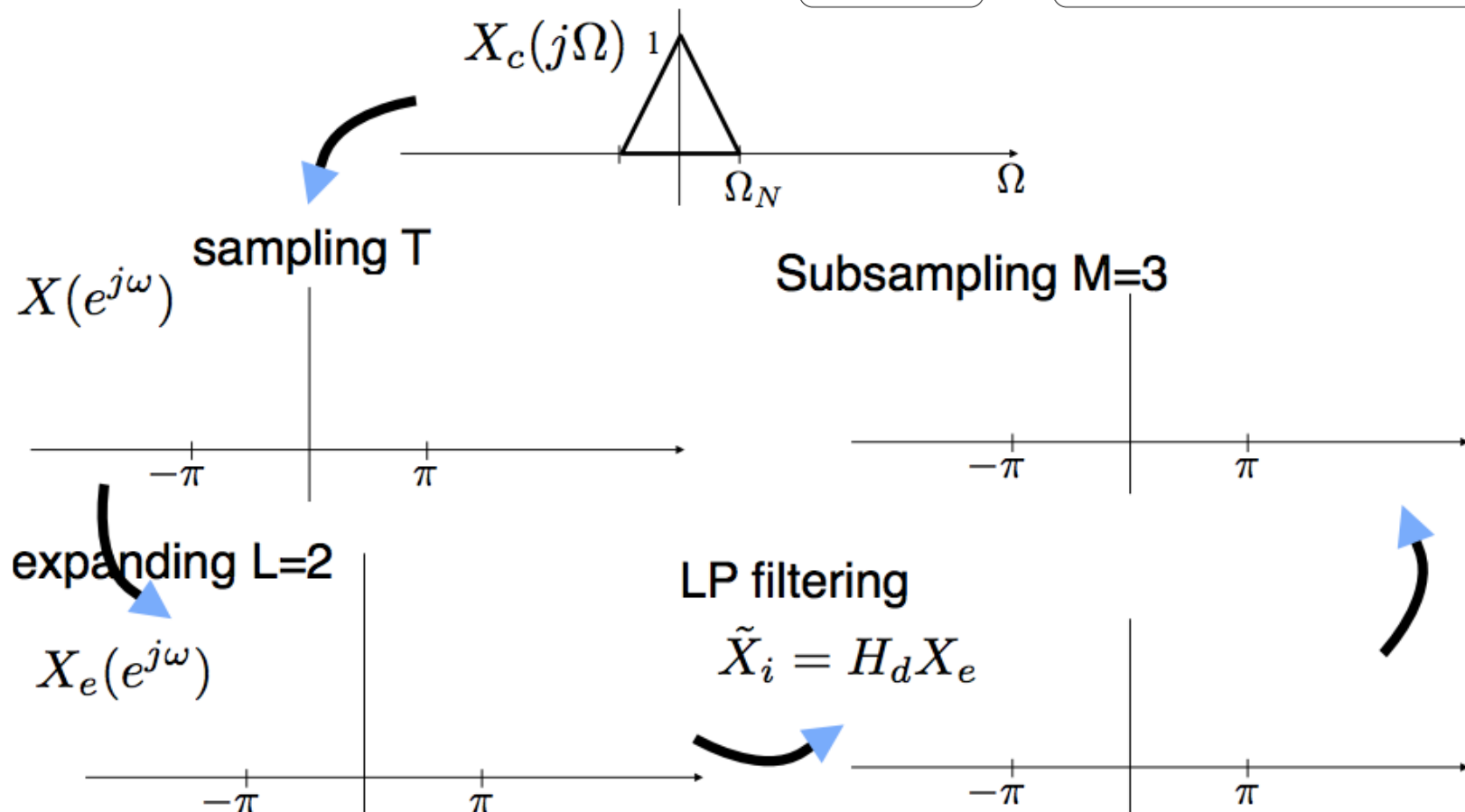
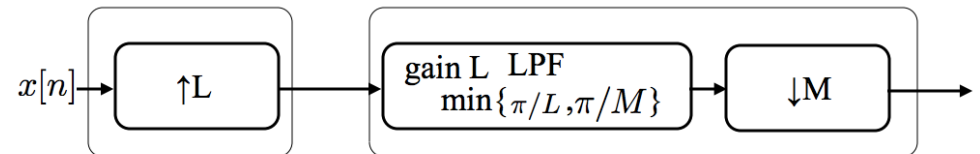


Or,



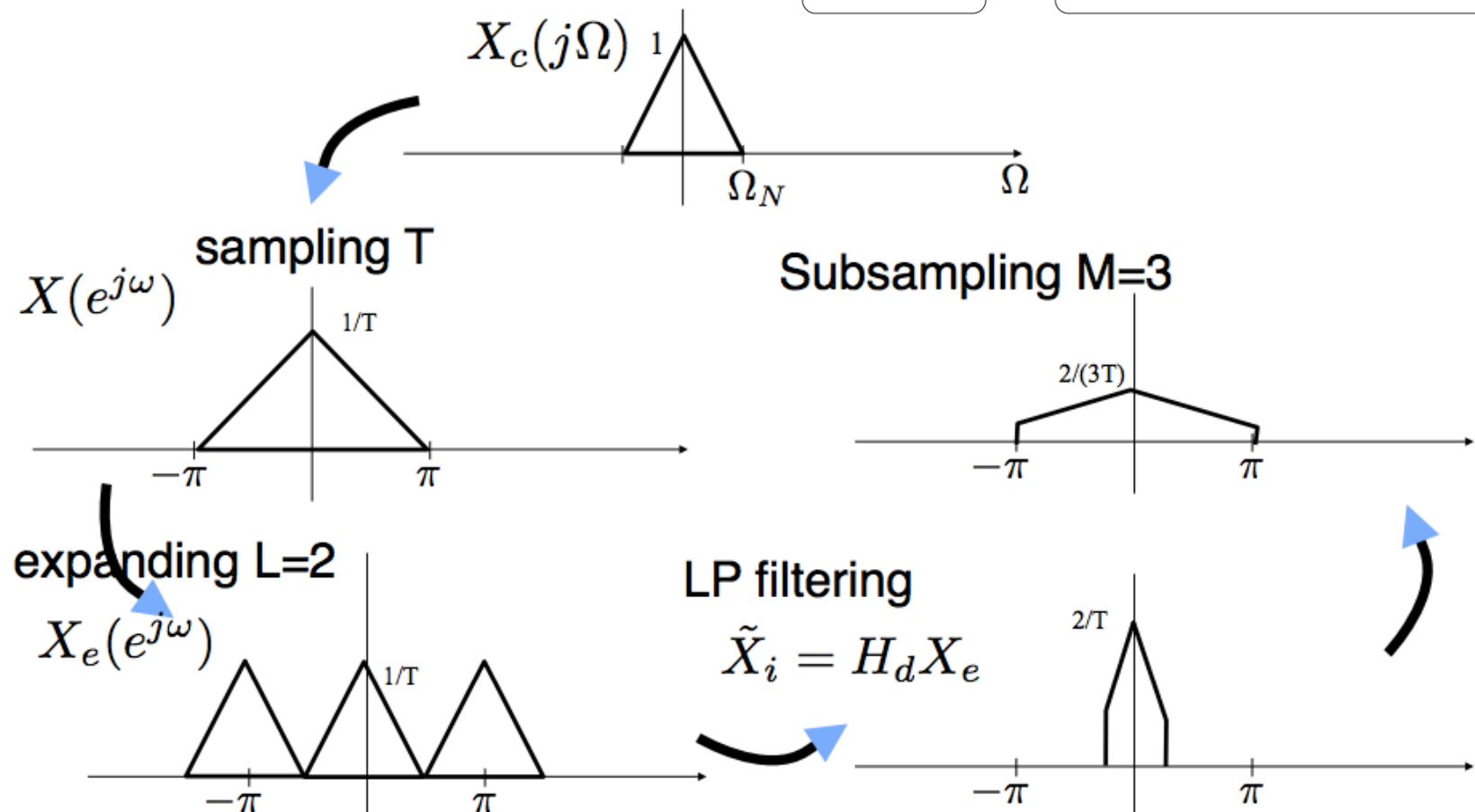
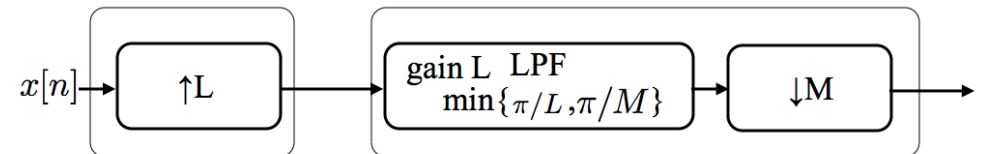
Example

□ $T' = 3/2T \rightarrow L=2, M=3$



Example

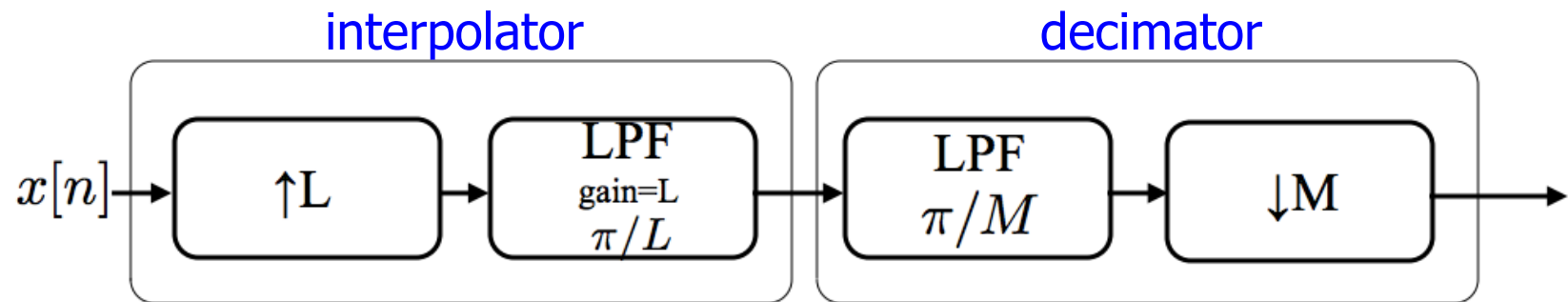
□ $T' = 3/2T \rightarrow L=2, M=3$



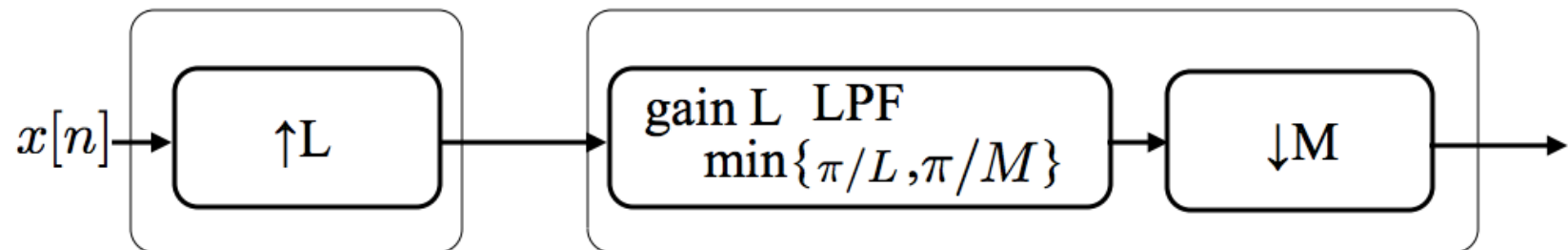
Non-integer Sampling

□ $T' = TM/L$

- Downsample by M , then upsample by L ?



Or,





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 $M=101$

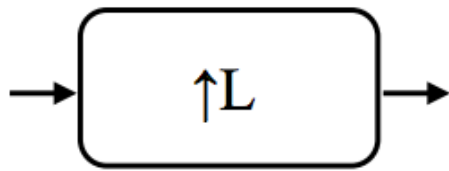


Example

- ❑ 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



“expander”

Upsampling

- expanding in time
- compressing in frequency



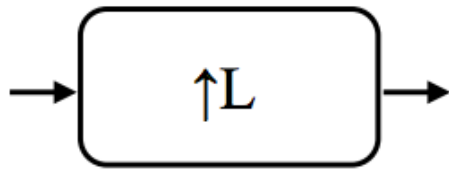
“compressor”

Downsampling

- compressing in time
- expanding in frequency

not LTI!

Interchanging Operations - Expander



“expander”

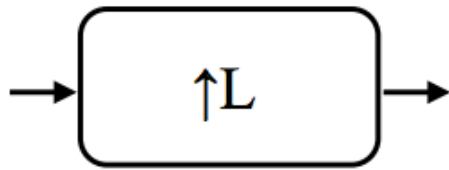
Upsampling

-expanding in time

-compressing in frequency



Interchanging Operations - Expander

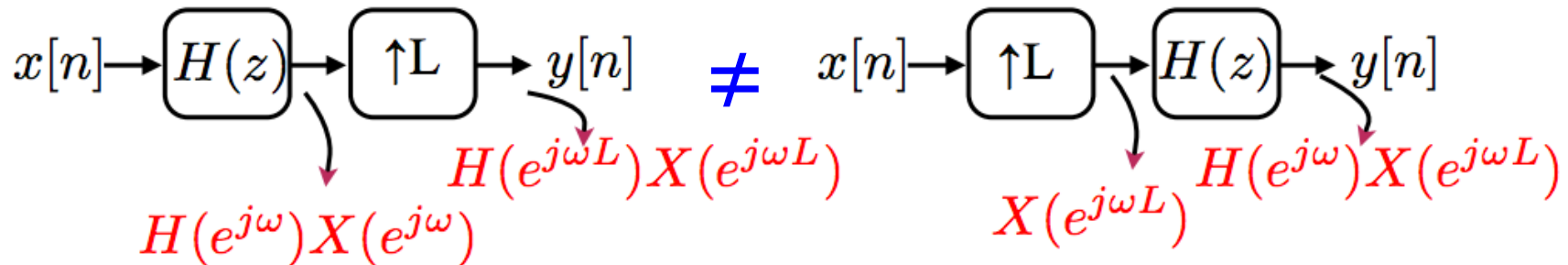


“expander”

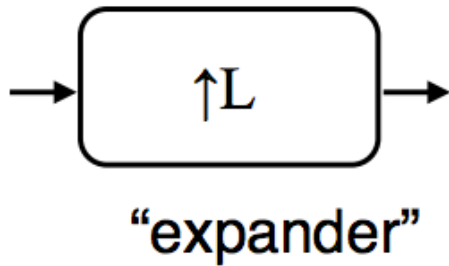
Upsampling

-expanding in time

-compressing in frequency

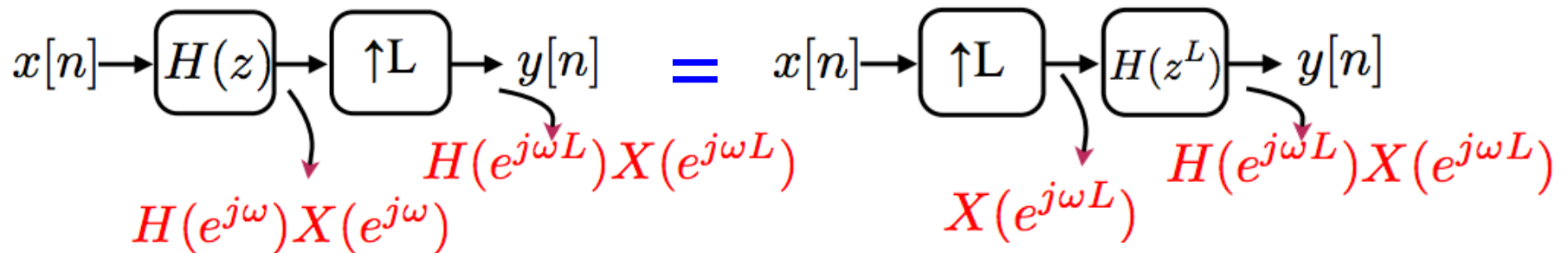


Interchanging Operations - Expander

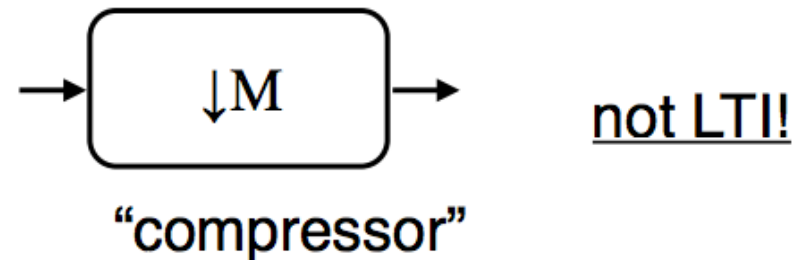


Upsampling

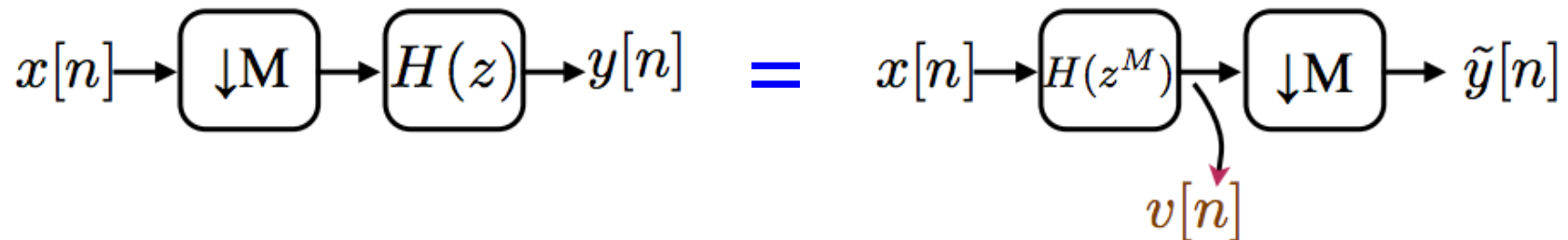
- expanding in time
- compressing in frequency



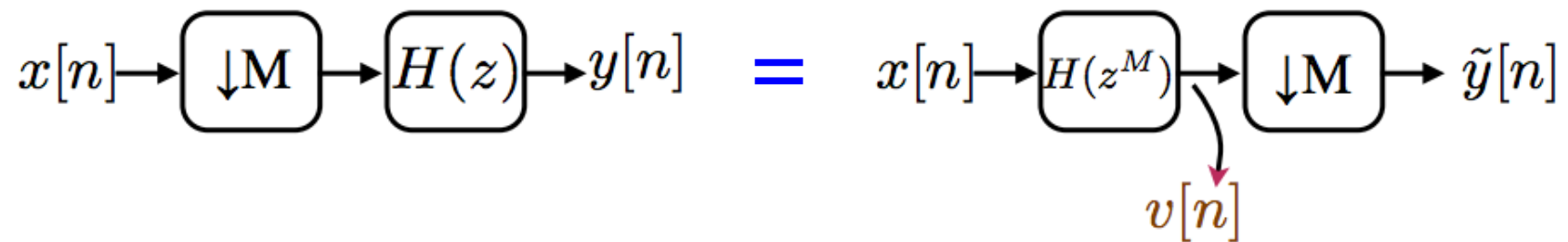
Interchanging Operations - Compressor



Downsampling
-compressing in time
-expanding in frequency

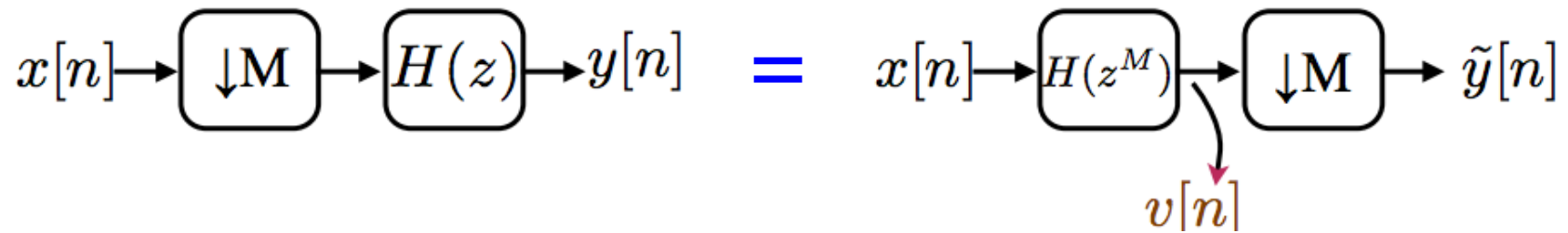


Interchanging Operations - Compressor



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

Interchanging Operations - Compressor



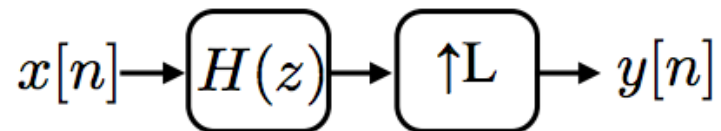
$$Y(e^{j\omega}) = \frac{1}{M} \sum_{i=0}^{M-1} H\left(e^{jM\left(\frac{\omega}{M} - \frac{2\pi i}{M}\right)}\right) X\left(e^{j\left(\frac{\omega}{M} - \frac{2\pi i}{M}\right)}\right)$$

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

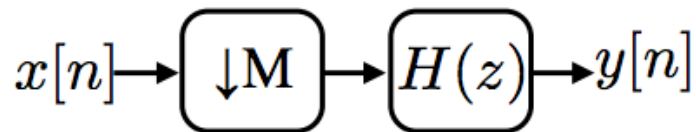
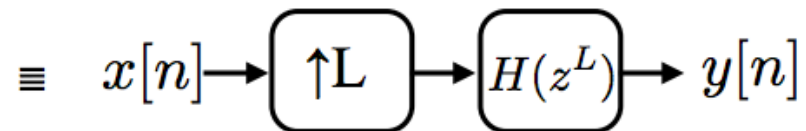
$$\tilde{Y}(e^{j\omega}) = \frac{1}{M} \sum_{i=0}^{M-1} V\left(e^{j\left(\frac{\omega}{M} - \frac{2\pi i}{M}\right)}\right)$$

Interchanging Operations - Summary

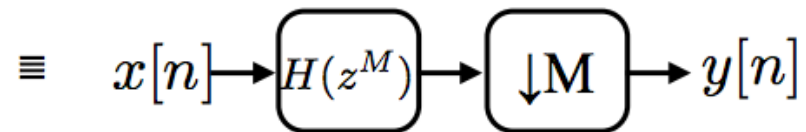
Filter and expander



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 $\pi/101$ (\$\$\$\$\$)
 - Compress by $M=101$

- ❑ Fortunately there are ways around it!
 - Called multi-rate
 - Uses compressors, expanders and filtering

Big Ideas

- ❑ Downsampling/Upsampling
- ❑ Practical Interpolation
- ❑ Non-integer Resampling
- ❑ Multi-Rate Processing
 - Interchanging Operations

$$x[n] \rightarrow H(z) \rightarrow \uparrow L \rightarrow y[n] \quad \equiv \quad x[n] \rightarrow \uparrow L \rightarrow H(z^L) \rightarrow y[n]$$

$$x[n] \rightarrow \downarrow M \rightarrow H(z) \rightarrow y[n] \quad \equiv \quad x[n] \rightarrow H(z^M) \rightarrow \downarrow M \rightarrow y[n]$$



Admin

- ❑ HW 4 due Sunday