### HKN ECE 310 Final Review Session

Whomst: Arjun, Eric, Bex

### Logistics + Shameless Plug

Slides and Recording will be uploaded here:
 Student Services (illinois.edu)
 hkn.illinois.edu/services

#### Services:

**YouTube channel (NEW!):** HKN has a YouTube channel with videos for difficult or cool concepts from undergraduate courses! Subscribe and leave a like to keep up to date on our content. Link to channel **here**.

**One-on-one tutoring:** HKN offers one-on-one tutoring for all core ECE classes. A list of tutors with the classes they specialize in can be found here: **Tutors List** 

Office Hours: A newer service HKN offers is Office Hours! We have Office Hours for all the core ECE classes, and more. Our Office Hours hosts have a variety of specializations from Quantum Computing to Destiny 2. The Office Hours Schedule can be found here: Office Hours Schedule

**Pre-exam Panic Pacifiers:** HKN will have students available to answer your questions right before (~an hour before) your exam for all core 100, 200, and 300-level ECE classes for both EE and CompE! Come with any last-minute questions or get some last-minute tips and tricks. The schedule, along with the schedule for all review sessions this semester, can be found here: **Review Session/Pre-exam Panic Pacifier Schedule** 

**SLIDES** 



### Topics

- Circular Convolution
- Upsampling & Downsampling
- Windowing
- FFT
- Filter Design
- Generalized/linear phase
- Practical A/D and D/A conversion

### Circular Convolution Exercise 1

Compute the linear AND circular convolution  $y[n] = x[n] *_4 h[n]$  where

$${x[n]}^{3}_{n=0} = {2, 4, 6, 8} \& h[n] = {-1, -1, 1}$$

Table Method: h[n] rows and x[n] columns  $\rightarrow y[n] = \{-4, 2, -8, -10\}$ 

<b>l</b> =	0	1	2	3	y[n]
×[ <b>l</b> ] =	2	4	6	8	
h(< 0 - <b>l</b> > <sub>4</sub> )	-1	0	1	-1	(-1)*2+6-8 = -4
h(< 1 - \( \ell_{>_4} \)	-1	-1	0	1	-2-4+8 = 2
h(< 2 - <b>l</b> > <sub>4</sub> )	1	-1	-1	0	2-4-6 = -8
h(< 3 - \( \ell_{>_4} \)	0	1	-1	-1	4-6-8 = -10

Linear result:  $y[n] = \{2, 2, 0, -2, -14, -8\}$ 

### Upsampling

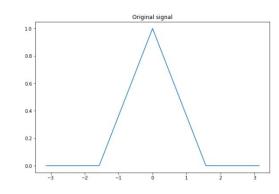
- In the DFT, we are adding zeros between every value we already have
  - If we upsample by U, we want there to be U times as many signal entries, so we add U-1 0s in between
- Example
  - o Signal: [0, 1, 2, 3]
  - Upsampled by 3: [0, 0, 0, 1, 0, 0, 2, 0, 0, 3, 0, 0]
- DTFT Effects:

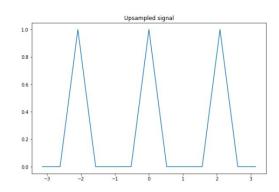
$$y[n] = \begin{cases} x \left[ \frac{n}{U} \right], & n = \pm U, 2U, \\ 0, & \text{else} \end{cases}$$

$$Y_d(\omega) = \sum_{n = -\infty}^{\infty} y[n] e^{-j\omega n}$$

$$= \sum_{n = -\infty}^{\infty} x[n] e^{-j\omega nU}$$

$$Y_d(\omega) = X_d(U\omega).$$

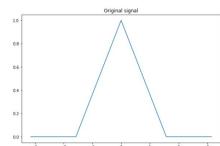


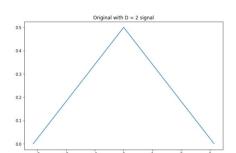


### Downsampling

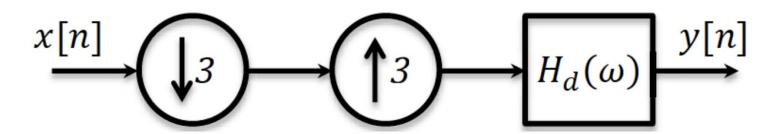
- The opposite of upsampling (wild)  $x_d[n] = x[Dn]$  with downsampling factor D
  - Effectively, keep only every D values from input signal
- Example
  - o Signal: [0, 1, 2, 3, 4, 5, 6, 7, 8, 9]
  - o Downsampled by 3: [0, 3, 6, 9]
- DTFT Effects:
  - Expands DTFT (susceptible to aliasing !!!)
    - Solution to aliasing: low pass filter first

$$Y_d(\omega) = \frac{1}{D} \sum_{k=0}^{D-1} X_d \left( \frac{\omega - 2\pi k}{D} \right).$$





## Upsampling and Downsampling in DFT Exercise

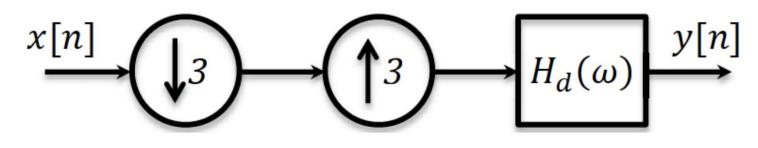


 $H_{d}(w)$  is ideal LPF with cutoff  $2\pi/5$ 

A. 
$$x[n]=cos(0.2\pi n)$$
 y[n]=?

B. 
$$x[n]=cos(0.6\pi n)$$
 y[n]=?

## Upsampling and Downsampling in DFT Exercise



$$y[n] = \frac{1}{3}\cos(0.2\pi n) = \frac{1}{3}x[n]$$
$$y[n] = \frac{1}{3}\cos(\frac{\pi}{15}n)$$

## Upsampling and Downsampling in DTFT Exercise

$$X_d(\omega) = \begin{cases} 1 - \frac{3|\omega|}{\pi}, & |\omega| \le \frac{\pi}{3} \\ 0, & \frac{\pi}{3} < |\omega| \le \pi \end{cases}$$

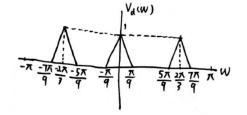
Given  $V_d(w)$  is  $X_d(w)$  upsampled by L, sketch  $V_d(w)$  if L=3

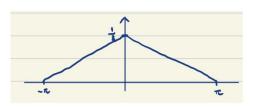
Given  $W_d(w)$  is  $X_d(w)$  downsampled by D, sketch  $V_d(w)$  if D=3

### Upsampling and Downsampling in DTFT Exercise

$$X_d(\omega) = \begin{cases} 1 - \frac{3|\omega|}{\pi}, & |\omega| \le \frac{\pi}{3} \\ 0, & \frac{\pi}{3} < |\omega| \le \pi \end{cases}$$

Given  $V_d(w)$  is  $X_d(w)$  upsampled by L, sketch  $V_d(w)$  if L=3





Given  $W_d(w)$  is  $X_d(w)$  downsampled by D, sketch  $V_d(w)$  if D=3

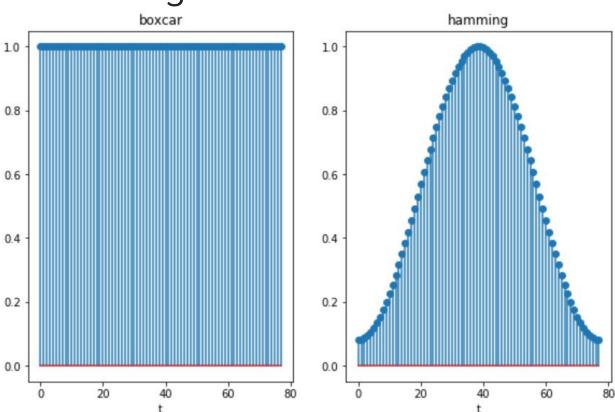
### Windowing

- Recall that the DFT implies infinite periodic extension of our signal.
- This extension can lead to artifacts known as "spectral leakage"
- Window functions help with these artifacts
  - Rectangular window
  - · Hamming window
  - · Hanning window
  - · Kaiser window
- Windowing is just multiplication in the time domain

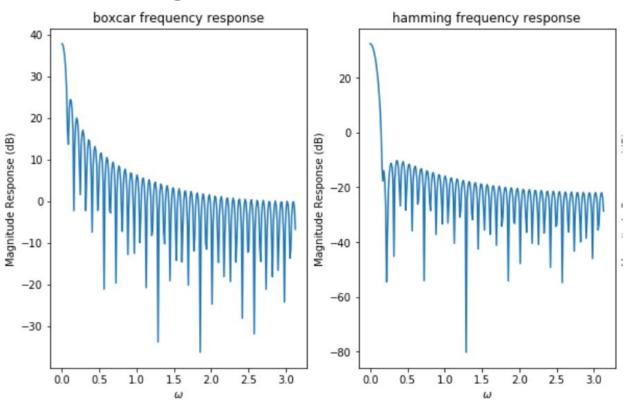
$$x_w = x[n]w[n]$$

- We care about the main lobe width and side lobe attenuation of these windows.
  - In particular, know the tradeoffs between the rectangular and Hamming windows Hamming has more side lobe attenuation but wider main lobe

### Windowing

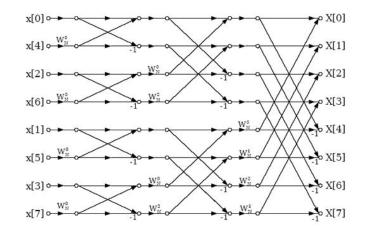


### Windowing



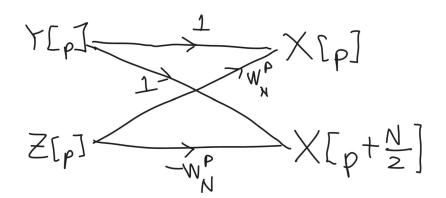
### Fast Fourier Transform

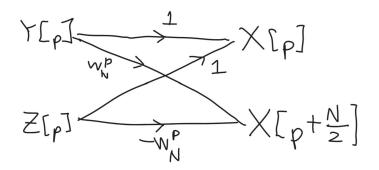
- · Class of "divide-and-conquer" algorithms to compute the DFT efficiently.
- Naïve DFT takes  $O(n^2)$  operations, FFT takes  $O(n \log n)$ .
- Decimation in Time vs. Decimation in Frequency
- · Butterfly diagrams



### **FFT Butterflies**

$$W_N = e^{-j\frac{2\pi}{N}}$$



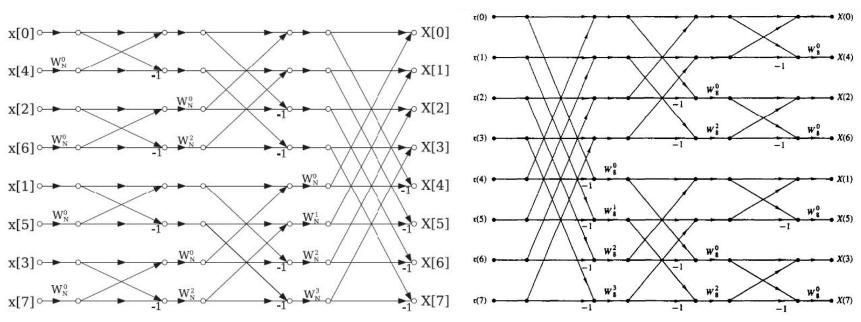


Decimation in Time

Decimation in Frequency

Slightly different!

### **FFT Butterflies**



#### **Decimation in Time**

The time points are "decimated" to do the first FFTs.

Figure 6.11 N = 8-point decimation-in-frequency FFT algorithmn.

#### **Decimation in Frequency**

The frequency points are "decimated" at the end.

### Fast Linear Convolution via FFT

- Convolution in the time domain requires  $O(n^2)$  operations.
- By convolution theorem, perhaps we can do better in the frequency domain?
- Don't forget multiplication in DFT domain is *circular* convolution in time.
- To avoid aliasing, we adopt the following procedure
- Given signal *x* and filter *h* of lengths *N* and *L*, respectively:
- 1. Zero-pad x and h to length N + L 1
- 2. Take their FFTs
- 3. Multiply in frequency domain
- 4. Take the inverse FFT

This procedure takes  $O(n \log n)$  operations.

Suppose you want to convolve two real-number signals of length 7000 and length 1100.

How many real-number multiplications does this take using classic convolution?

How many real-number multiplications does this take using using FFT?

### Classic Convolution

Partial overlap at the beginning: (1 + 2 + ... + 1099) = (1099)(1100)/2

Partial overlap at the end: (1099 + 1098 + ... + 2 + 1) = (1099)(1100)/2

Full overlap in the middle: (1100)(7000 - 1100 + 1)

Total: 7,700,000 multiplications.

### **FFT**

Zero-pad both signals to length 8192. (0)

2x FFT.

- Multiplication by 1 is free.
- Each left-side of the butterfly is multiplied by one complex number.
- This is done log\_2 times, one for each butterfly phase.
- Total: (2 \* 8192 \* log\_2(8192) \* 4)

Multiply the signals together (8192 \* 4)

1x IFFT: (1 \* 8192 \* log\_2(8192) \* 4)

Total: 1,310,720 multiplications.

### Some Hard Numbers...

Convolving length 7000 with length 1100:

Classic convolution: 7,700,000 multiply operations

• FFT: 1,310,720 multiply operations

That's 83% faster.

### So what:

 A modern-day computer can do around 1e9 operations per second.

- 7,700,000 operations = 7.7 milliseconds.
- 1,310,720 operations = 1.3 milliseconds.

Ok that doesn't really do anything.

### Fine. Some Hard(er) Numbers...

The average MP3 file seems to be about 4MB.

Convolving length 4e6 with length 4e6 (apparently to add reverb or filter or something):

- Classic convolution: 1.6e13 multiply operations
  - About 4.5 hours.

- FFT: 1.12e9 multiply operations
  - About a second.

Ok that's pretty good.

### FFT: The Most Important Algorithm of the Modern World

- Fast large-integer and polynomial multiplication
- Solve partial differential equations
- Filtering algorithms
- Digital recording, sampling, and pitch correction
- JPEG and MP3
- Used in 5G, LTE, Wi-Fi, and other communication systems

These are ALL made viable by FFT. Without it, these would be so slow...

### Trivia Time!

Who was the first to come up with the FFT?

Answer: Gauss. Yes, that Gauss. And in 1805.

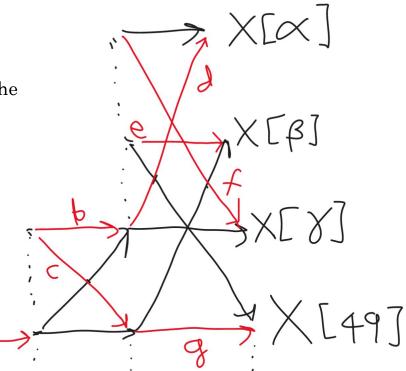
Yes. This predates Fourier.

And he didn't even publish it:(

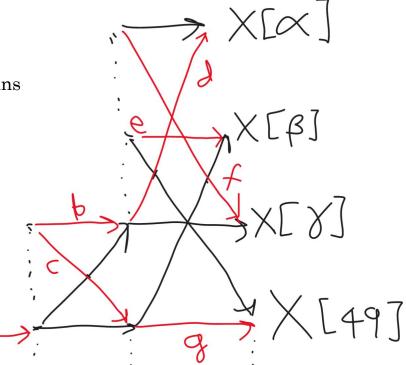
What is the first step when drawing the entire FFT diagram of the following signal?

$$x[n] = \{1, 2, 3\}$$

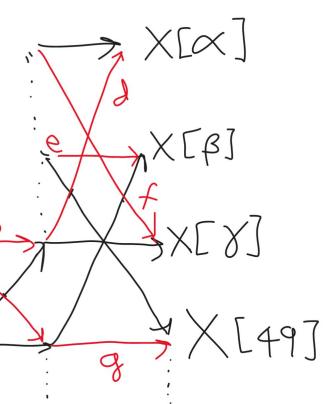
This is part of a 64-bit decimation in time radix-2 FFT. Find the value of the letters and greek symbols.



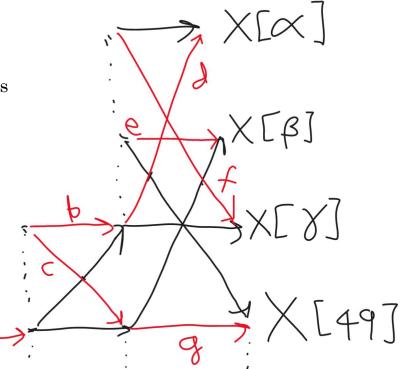
We know X[beta] and X[49] are connected by the final FFT. This means beta + 64/2 = 49 = beta + 32. This implies **beta = 17**.



We know X[gamma] and X[49] are connected by the second-to-last FFT. This means gamma + 64/4 = 49 = gamma + 16. This implies **gamma = 33**.

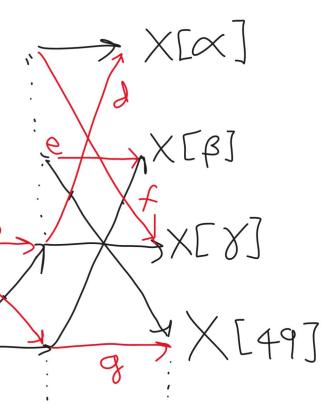


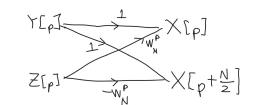
We know X[gamma] and X[alpha] are connected by the last FFT. This means alpha + 64/2 = gamma = 33 = alpha + 32. This implies **alpha = 1**.



FFT input indices are the bit reversal of their output index. Thus, delta is the bit-reversal of 49. This implies, **delta = 35**.

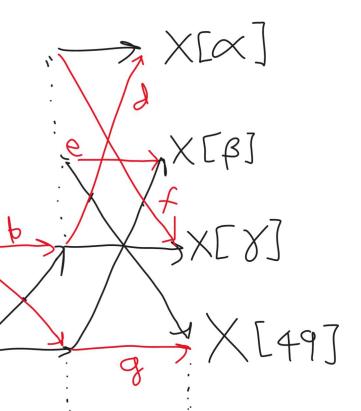
Not convinced? Draw out the full FFT if you wish.

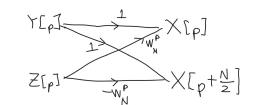




FFT input indices are the bit reversal of their output index. Thus, delta is the bit-reversal of 49. This implies, **delta = 35**.

Not convinced? Draw out the full FFT if you wish.





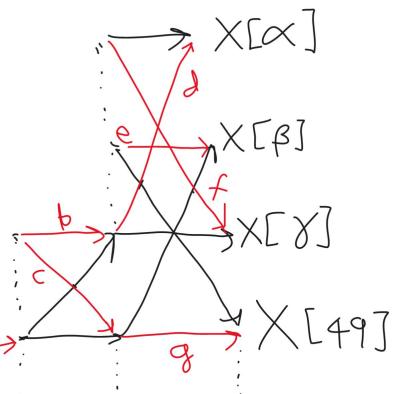
Let's do the easier letters first.

e = 1: Top part of butterfly.

 $\mathbf{b} = \mathbf{1}$ : Top part of butterfly.

f = 1: Down-diagonal of butterfly.

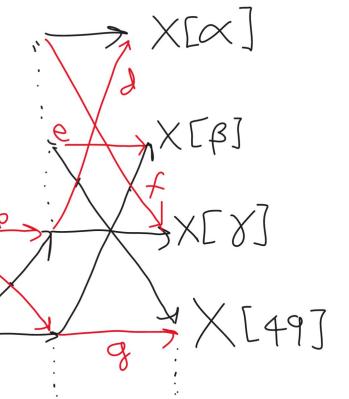
c = 1: Down-diagonal of butterfly.



d is the up-diagonal of the very first butterfly in the last FFT (the 64-length FFT), since alpha = 1.

Thus, use N = 64, p = 1.

$$d = W_{64}^1 = e^{-j\frac{2\pi}{64}1}$$



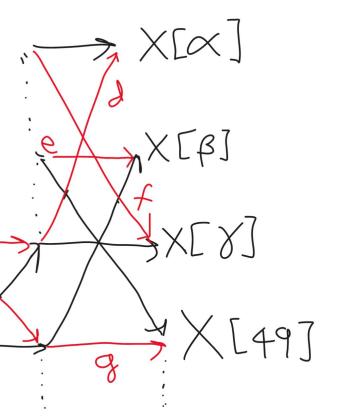
# $\begin{array}{c} 1 \\ 1 \\ 2 \\ 2 \\ p \end{array} \begin{array}{c} 1 \\ 2 \\ 2 \\ 2 \end{array} \begin{array}{c} 1 \\ 2 \end{array} \begin{array}{c} 1 \\ 2 \\ 2 \end{array} \begin{array}{c} 1 \\ 2$

### FFT Practice

g is the bottom arrow of the 17th butterfly in the last FFT (the 64-length FFT), since beta = 17.

Thus, use N = 64, p = 17.

$$g = -W_{64}^{17} = e^{-j\frac{2\pi}{64}17}$$



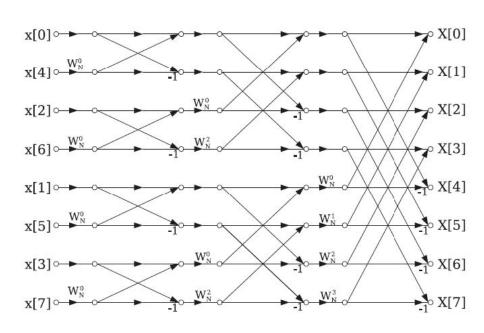
### A reminder...

In the last FFT, we're doing 8-length butterflies.

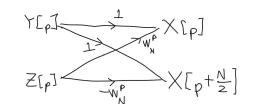
In the second-to-last FFT, we're doing 4-length butterflies.

In the third-to-last FFT, we're doing 2-length butterflies.

"Final elements" 0 and 1 are grouped into one butterfly. 2 and 3 are grouped. 4 and 5 are grouped. 6 and 7 are grouped.



Decimation in time for length-8 signal



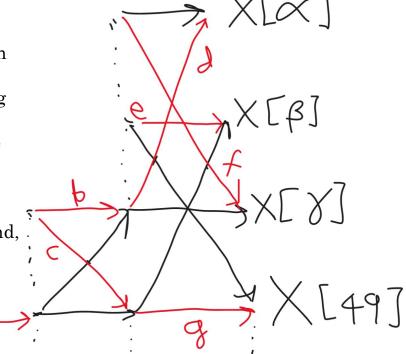
Analogously,

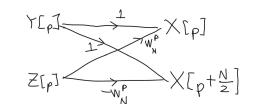
In the last FFT, we're doing 64-length butterflies.

In the second-to-last FFT, we're doing 32-length butterflies.

In the third-to-last FFT with a, we're doing 16-length butterflies.

So, elements 0-15 are in the first butterfly, elements 16-31 in the second, 32-47 in the third, and 48-63 in the fourth.



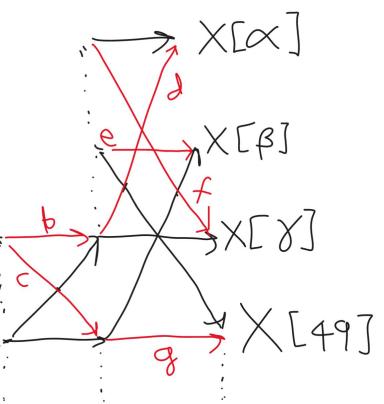


Elements 48-63 in the fourth butterfly.

49 is in the upper-half of this 16-length butterfly.

Therefore, a is actually the top arrow in the butterfly.

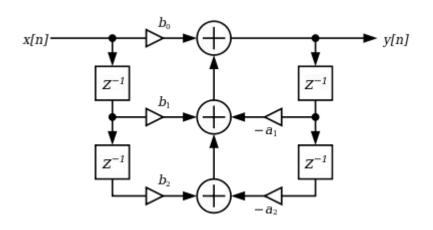
a = 1.

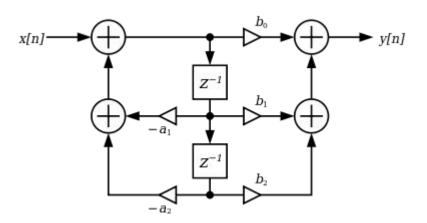


### Filter Design - IIR

Direct Form 1

Direct form 2





$$y[n] + a_1 y[n-1] + a_2 y[n-2] = b_0 x[n] + b_1 x[n-1] + b_2 x[n-2]$$
$$y[n] = -a_1 y[n-1] - a_2 y[n-2] + b_0 x[n] + b_1 x[n-1] + b_2 x[n-2]$$

### Filter Design - FIR

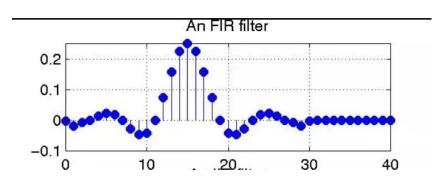
$x[n]$ $z^{-1}$ $z^{-1}$
$\sqrt[4]{b_{\scriptscriptstyle 0}}$ $\sqrt[4]{b_{\scriptscriptstyle 1}}$ $\sqrt[4]{b_{\scriptscriptstyle 2}}$ $\sqrt[4]{b_{\scriptscriptstyle N}}$
$\Sigma \longrightarrow \Sigma \longrightarrow \Sigma \longrightarrow y[n]$

Filter type	Symmetry	Length	$H_d(0)$	$H_d(\pi)$	Possible canonical filter types
Type-I	Even	Odd	May be non-zero	May be non-zero	LP, HP, BP, BS
Type-II	Even	Even	May be non-zero	Always zero	LP, BP
Type-III	Odd	Odd	Always zero	Always zero	BP
Type-IV	Odd	Even	Always zero	May be non-zero	BP, HP

$$H(\omega) = A(\omega)e^{j\Psi(\omega)}. \longrightarrow H(\omega) = |A(\omega)|e^{j(\Psi(\omega) + \angle A(\omega))}.$$

$$|H(\omega)| = |A(\omega)|$$

$$\angle H(\omega) = \Psi(\omega) + \angle A(\omega).$$



Go over Corey's notes, they're very good

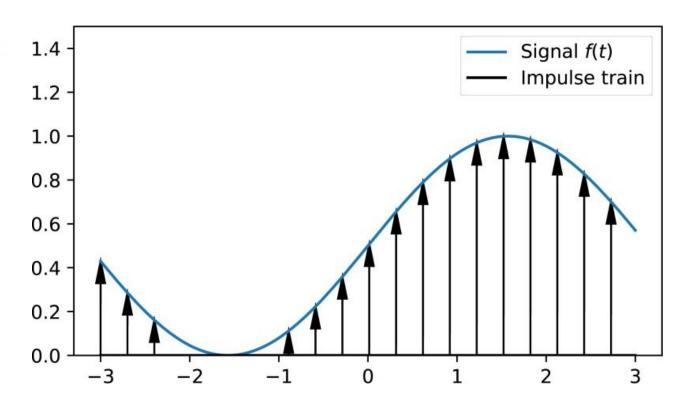
### GLP vs LP

Linear phase : 
$$\angle H(\omega) = -\alpha \omega$$

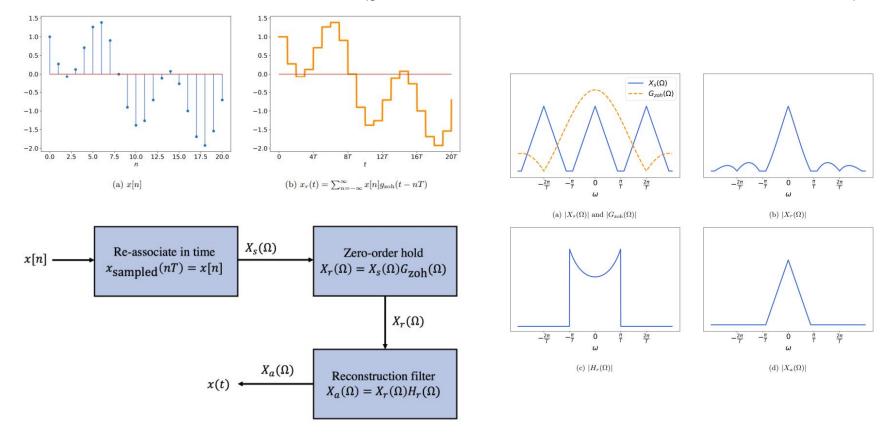
Generalized linear phase :  $\angle H(\omega) = -\alpha\omega + \beta(\omega)$ .

### Practical sampling

$$x_{\text{sampled}}(t) = x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT).$$



### Practical reconstruction (yes I stole this from course notes)



### The End

- Thanks for attending!
- Good luck studying!

