Digital Signal Processing

Spring Semester 2022

Digital Systems, Part 1

Last time's learning objectives

- Explain how the Discrete/Fast Fourier Transform (DFT/FFT) differs from the DTFT
 - \checkmark The DFT is the DTFT computed at discrete values of ω
 - √ The FFT is a fast algorithm for computing the DFT
- Compute the DFT/FFT on paper
 - ✓ Compute the DTFT, then replace ω with k (where k = 0, 1, 2, ..., N-1; N=DFT length)
- Compute the DFT/FFT in Matlab (and interpret the results)
 - ✓ Use the fft() function

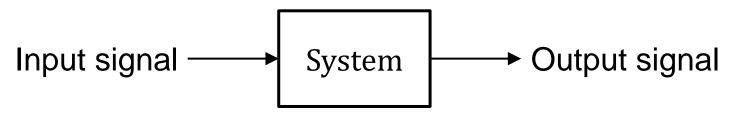
Today's learning objectives

From today's lecture, you should be able to...

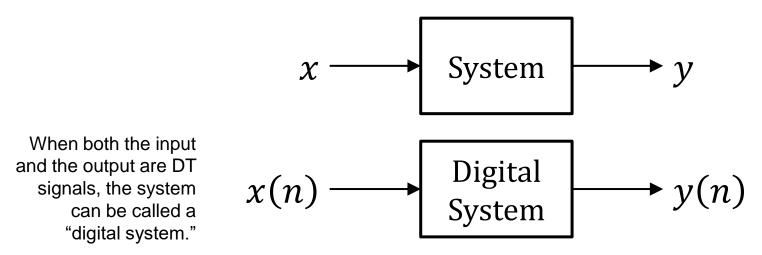
- Explain the terms "digital system" and "digital filter"
- List the ways to characterize a filter
- List the ways to apply a filter

Digital systems and filters

In signal processing, a **system** takes an input signal and yields an ouput signal:



By convention, the **input** is denoted by x and the **output** is denoted by y:



Digital systems and filters

A digital **filter** is a digital **system** designed to:

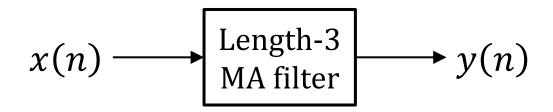
- Remove unwanted frequencies (e.g., noise)
- Attenuate (reduce) certain frequencies
- Amplify (increase) certain frequencies

Linear and Time-Invariant

In this class, we will focus on LTI DT systems.



Example: Length-3 moving average filter



Input/output equation (called a difference equation):

$$y(n) = \frac{1}{3} (x(n+1) + x(n) + x(n-1))$$



- What's the filter's impulse response?
- Is the filter memoryless?
- Is the filter causal?
- What type of filter is it?
 - Is it a lowpass filter? A highpass filter? A bandpass filter?
 - What frequencies does it attenuate?
- Does the filter induce phase distortion?

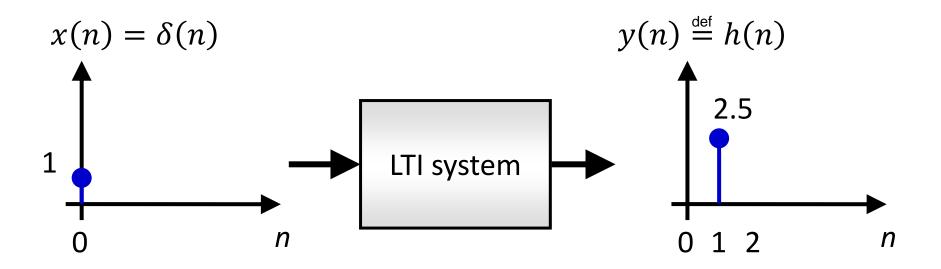
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Impulse response of LTI systems

LTI systems (filters) can be characterized by how they change an impulse:

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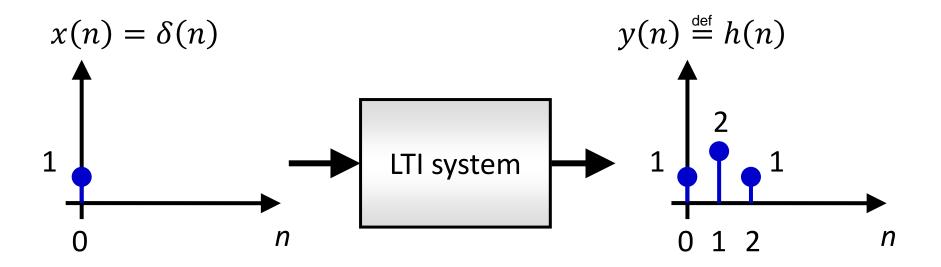
Example of getting impulse response



Thus, the system's **impulse response** is...

$$h(n) = 2.5\delta(n-1)$$

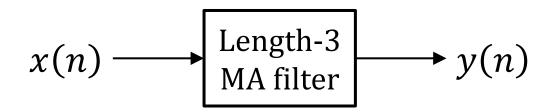
Example of getting impulse response



Thus, the system's impulse response is...

$$h(n) = \delta(n) + 2\delta(n-1) + \delta(n-2)$$

Impulse response of MA filter



Input/output equation (called a difference equation):

$$y(n) = \frac{1}{3} (x(n+1) + x(n) + x(n-1))$$

The system's **impulse response** is...

$$h(n) = \frac{1}{3} \left(\delta(n+1) + \delta(n) + \delta(n-1) \right)$$

Just replace x(n) with $\delta(n)$ in the I/O equation!

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More properties of DT systems

Length-3 MA filter's difference equation:

$$y(n) = \frac{1}{3} (x(n+1) + x(n) + x(n-1))$$

→ Not memoryless (i.e., requires storing previous/future samples in order to compute current output)

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More properties of DT systems

Length-3 MA filter's difference equation:

$$y(n) = \frac{1}{3} (x(n+1) + x(n) + x(n-1))$$

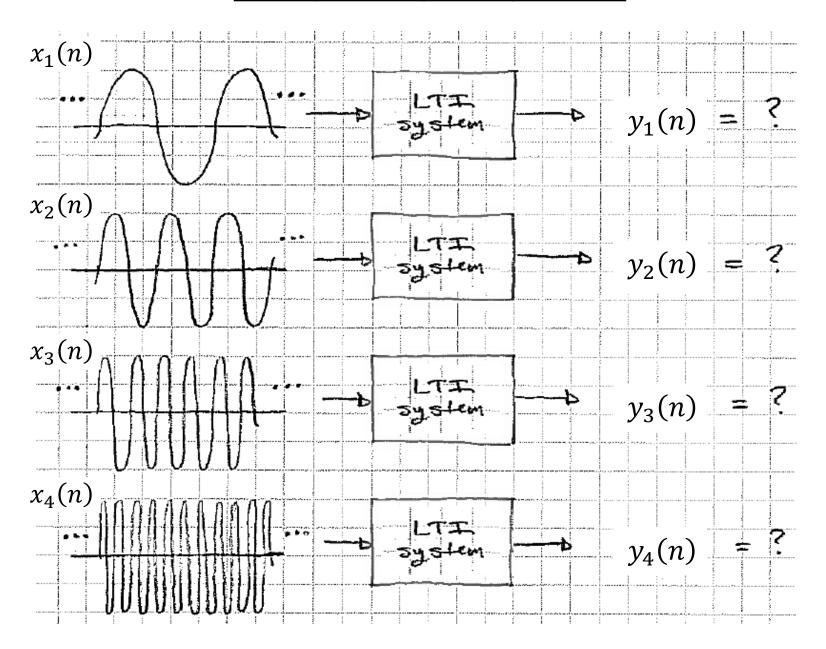
→ **Not causal** (i.e., requires storing future samples in order to compute current output)

More properties of DT systems

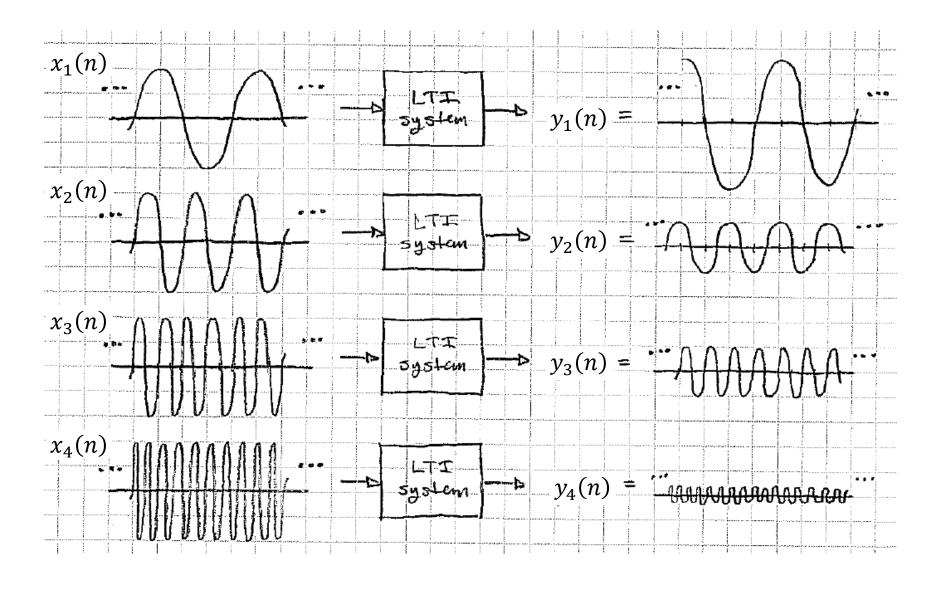
A system is memoryless if Memory (ess: the output at time 11 depends only on the input at time 11. A system is causal iff the output at time a depends only on past inputs (i.e., inputs at time instants $\leq n$ Memoryless => causal Note: Real-time systems must be causal Note:

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Frequency response



Frequency response



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Examples of linear-phase filters (no phase distortion) (Trick: They have symmetric impulse responses)

