

# EE482: Digital Signal Processing Applications

## Quiz 01 Review

# Outline

- Quiz Logistics
- Chapter 2 – DSP Fundamentals
- Chapter 3 – FIR Design

# Quiz Logistics

- Covers Ch 1-3
- Allowed a single double-sided sheet of notes
- You are expected to know how to solve problems by hand and using Matlab commands
- Calculators are allowed
  - But not expected to be required

# DSP Fundamentals

- Basic signals
  - Delta, sinusoids
    - Know the relationships between frequency representations
- Systems
  - Block diagram representation
  - Linearity and time invariance
  - Convolution
- Z-transform ROC
  - stability, causality,
  - Convolution
- Frequency response (DTFT)
  - Existence from z-transform
  - Magnitude and phase response
- Discrete Fourier Transform
  - Relationship with DTFT
  - Frequency resolution
  - Fast Fourier transform (`fft.m`)
- Fixed-Point Issues
  - Number format
  - Quantization errors
    - signal, coefficients
  - Arithmetic errors
    - Roundoff, overflow

# FIR Filters

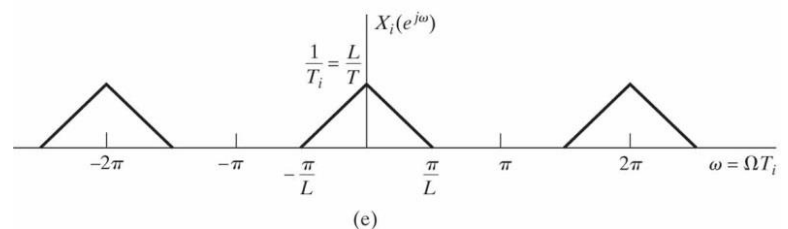
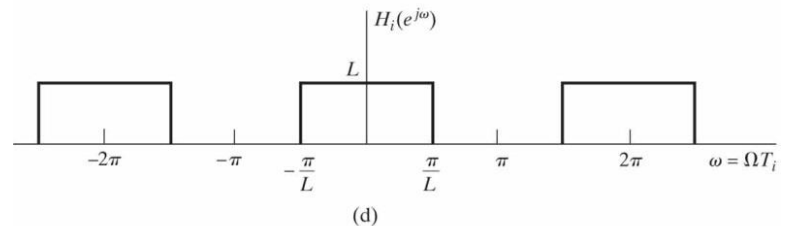
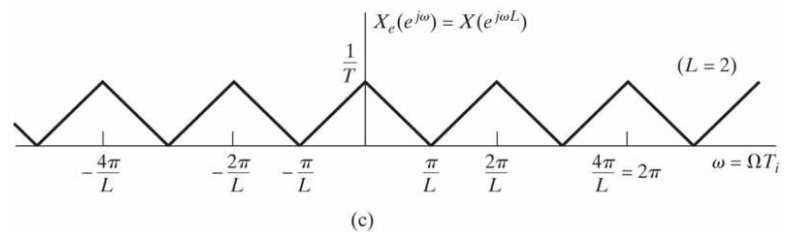
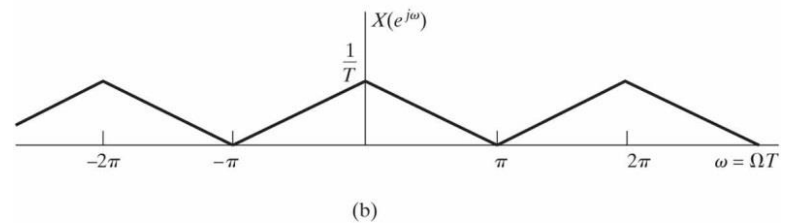
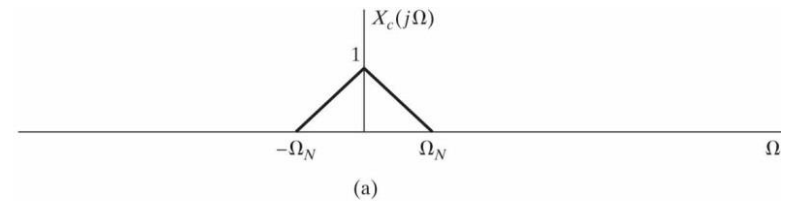
- Advantages of FIR design
- Filter types
  - Lowpass, highpass, bandpass, bandstop
- Filter specifications
  - Graphical and with equations
- Linear phase filters
  - What are they and why does it matter
- FIR design process
  - Determine desired system  $H_d(z)$ 
    - Compute impulse response  $h_d(n)$
  - Select window  $w(n)$ 
    - Length  $L$
  - Window impulse response
    - $h(n) = w(n)h_d(n)$
    - Be sure to shift for causality and truncate to length  $L$

# Windowing

- Why do windowing?
  - Gibbs phenomenon
- Trade-off between mainlobe width and sidelobe height
  - Mainlobe – transition band
  - Sidelobe amplitude – ripple
- Frequency domain convolution for smearing (smoothing)
- Window design
  - How to select appropriate window
    - Table 9.2 in FIR lecture
  - Solve for minimum window length

# Upsampling

- Increase sampling rate
  - Zero insertion
- No need to worry about aliasing
- Need a interpolation LP filter to generate “smooth” signal
  - Interp filter needs to have magnitude equal to the increase factor  $L$
- Squish spectrum in by  $L$



# Downsampling

- Decrease sampling rate
  - Drop samples
- Need to worry about aliasing
  - Design LP filter to prevent aliasing
- Expand (stretch) each  $2\pi$  spectrum copy from the center
  - “Pull edges”

