EECS 361
Semester Review
Fall 2022

1) Classification of signals:
   a) Periodic
   b) Aperiodic
   c) Energy
   d) Power
   e) Continuous time
   f) Discrete time
   g) Deterministic
   h) Random (not considered in this class)
2) Phasor representation of sin & cos and complex numbers, magnitude and phase
3) Classification of Systems
   a) Linear/Nonlinear
   b) Time varying/Time invariant
   c) Causal/non-causal
   d) Continuous time/Discrete time
4) Special functions
   a) δ(t), δ[n],
   b) tri(t)
   c) u(t), u[n]
   d) rect(t)
5) Convolution
   a) Continuous time
   b) Discrete time
6) Impulse response of linear time invariant systems, h(t), h[n]
7) Step response of systems a(t), a[n]
8) Impulse response of cascaded linear time invariant systems, h(t), h[n].
9) Bounded input/Bounded output (BIBO) stability and constraints on h(t) and h[n] for stability
10) Causality and constraints on h(t), h[n] for causality
11) Fourier Series
    a) Complex exponential
    b) Sin/Cos
    c) Cos
12) Power-Parsaval's theorem for periodic signals
13) Spectral plots for periodic signals: one sided and double sided
14) Fourier Transform
15) Fourier Transform theorems and properties
16) Energy-Parsaval's theorem for aperiodic signals
17) Transfer Function of linear time invariant systems - H(f) and H(z)
18) Amplitude and phase response of linear time invariant systems, H(f) and H(z).
19) Transfer Function of cascaded linear time invariant systems - - H(f) and H(z)
20) Criteria for an ideal linear time invariant system, - H(f) - H(z), h(t)- h[n]

a) ILPF
b) IBPF

21) Distortion for linear time invariant systems
   a) Amplitude distortion
   b) Phase distortion
22) Signals and systems
   a) Finding system output given input and system description
   b) Bandwidth and its definitions
   c) Inverse time duration-bandwidth relationship
   d) Inverse rise time and bandwidth relationship
   e) If $B_h >> B_s$ then minimal distortion,
      where B_h=system bandwidth and B_s=signal bandwidth
23) Sampling theorem
   a) Minimum sample rate
   b) Spectrum of a sampled signal
   c) Aliasing
   d) Recovering x(t) from its sampled version $x_s(t)$
24) Discrete Fourier Transform (DFT)
   a) Definition
   b) DFT is the Fourier Series of a <u>periodic</u> sampled (discrete) signal
   c) Parameters of the DFT, frequency resolution=Δf, record length=L, number of points in the DFT=N. sample rate f_s, time resolution=Δt, baximum frequency (bandwidth)=B
      i. $\Delta f=1/L$
      ii. $N = L f_s = L/ \Delta t$
      iii. $B = (N/2)^* \Delta f$
   d) Conjugate symmetry of the DFT
   e) Picket fence effect
   f) Spectral leakage
      i. Windows
      ii. Main lobe width and Side lobe attenuation
   g) Circular convolution
   h) Using the DFT for linear convolution, role of padding with zeros
25) z-Transform and digital filters
   a) Definition
   b) Frequency response, $H(e^{j\Omega})$, is H(z) evaluated on the unit circle
   c) Properties of z-transforms
      i. Linearity
      ii. Delay
      iii. Convolution
   d) Poles and zeros for the general difference equation (ARMA system)
   e) All poles of H(z) must be inside the unit circle for the system to be stable
   f) Relationship of pole and zero locations to the frequency response

g) Analysis-given the general difference equation find:
   i. Location of poles and zeros
   ii. Frequency response, $H(e^{j\Omega})$
   iii. Or given H(z) find the ARMA (implementation) equation
   iv. Convert among, H(z), difference equations, $H(e^{j\Omega})$, pole/zero diagram

h) Design of digital filters
   i. Position the poles and zeros in the z-plane to get desired frequency response
   ii. Construct H(z)
   iii. Determine the coefficients for the difference equation from H(z).
   iv. Draw system implementations

i) Feedback & stability for discrete time systems