SCHOOL OF ENGINEERING
Faculty of Science and Engineering

MACQUARIE University

ELEC4844/8844 Practical Class – Week 2, 2024

Review of Signals & Systems

TASK 1

a) Consider a continuous-time signal of 440 Hz sine wave. It is sampled to obtain two discrete-time signals at f_{S1} = 1200 Hz and f_{S2} = 6000 Hz.
Draw the two sampled signals over a period of 20 ms, and compare.
Play the two sampled signals over a period of 2 s, using MATLAB sound function.

b) Generate the following two signals for comparison:

- 700-Hz cosine wave, sampled at 1500 Hz
- 800-Hz cosine wave, sampled at 1500 Hz

Plot the sampled signals (over 20 ms), and use MATLAB sound function to play the above signals (over 2 seconds).
Further, add a phase shift of π/2 to both signals. What do you find when comparing their plots as well as sounds?

c) Given a continuous-time signal:

$$x(t) = A\sin(2\pi f_0 t + \theta)$$

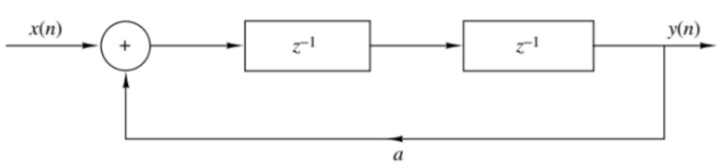
with A = 128, f_0 = 1200 Hz, and $\theta = \pi/4$. Draw the following on the same plot and compare:

1. use f_s = 12 kHz to sample $x(t)$ in the time range [0, 0.001];
2. use f_s = 42 kHz to sample $x(t)$ in the time range [0, 0.001];
3. convert the last sampled signal in (2) from float to 8-bit integer.

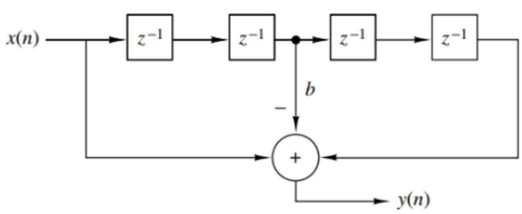
Zoom in to check the detail. Think about which step induces more error in this case, sampling or quantisation, and why.

TASK 2

Consider the following discrete-time LTI systems:

1) 

, in which a = -0.64

2) 

, in which b = 2

3) $y[n] + 1.8y[n-1] + 0.81y[n-2] = x[n] - 2x[n-1] + x[n-2]$

4) $y[n] - 1.8y[n-1] + 0.81y[n-2] = x[n] + 2x[n-1] + x[n-2]$

5) $h[n] = [0.25, 0.5, 0.25]$

6) $h[n] = [1, 0, -1]$

7) $H(z) = (0.3 + 0.3z^{-1})/(1 - 0.3z^{-1})$

8) $H(z) = 1/(1 + 0.1z^{-1} - 0.3z^{-2})$

For each system,

a) use MATLAB function zplane to plot the zeros and poles;

b) use MATLAB function impz to plot the impulse response;

c) use MATLAB function freqz to plot the frequency response in magnitude and phase;

d) generate input $x(n) = 0.9^n \sin(n)\, u(n)$, and use MATLAB function conv to compute the output;

e) generate normally distributed random noise with mean of 0.2 and standard deviation of 0.3 as input, and use MATLAB function filter to compute the output.

Note that, you are expected to find out how to use these functions effectively by inspecting their help documentations first.

TASK 3

Recall the definition formula of Fourier transform for a continuous-time signal $f(t)$

$$F(\omega) = \int_{-\infty}^{\infty} f(t)e^{-j\omega t}dt$$

To be able to compute the Fourier transform numerically using MATLAB, $f(t)$ can be uniformly sampled to obtain $f(mT_S)$, in which $T_S = 1/f_S$ is the sampling period. Then, approximately we have

$$F(\omega) = \sum_{m=-\infty}^{\infty} f(mT_S)e^{-j\omega mT_S}T_S$$

a) Apply the formula above to compute the spectrum of a rectangular pulse

$$f(t) = \begin{cases} 1, & -T/2 < t < T/2 \\ 0, & |t| \geq T/2 \end{cases}$$

Set $T = 1$ s, $f_S = 20$ Hz, and compute the spectrum of $f(t)$ for frequency from -10 Hz to 10 Hz with 0.1 Hz steps. Plot both real and imaginary part of the spectrum.

b) Change T to a different value (e.g. 3 s) while other parameters remain identical. Repeat the computation and plotting above to verify the scaling property of Fourier transform.

c) Let $g(t) = f(t)e^{j2\pi f_C t}$. Set $T = 1$ s, $f_C = 3$ Hz, and compute the spectrum of $g(t)$ from -10 Hz to 10 Hz with 0.1 Hz steps. Plot the result to verify the frequency-shifting property of Fourier transform.