**EE089**

**DIGITAL SIGNAL PROCESSING LABORATORY**

**Lab 7**

# TRANSFER FUNCTION AND DIGITAL FILTER DESIGN

**Full name:………………………………..**

**Student number:………………………...**

**Class:……………………………………..**

**Date:……………………………………...**

# I. OBJECTIVES

Students know how to write m-file in Matlab to illustrate the properties of Fourier transform, transfer function and the design of Resonator filter, Hamming window…

# II. REQUIRED EQUIPMENT

1. Computer
2. Matlab software

# III. PROCEDURE

**Problem 1:**

Consider a second order filter with the I/O equation:

$$y[n] = 1.5y[n-1] - 0.9y[n-2] + x[n] + 0.7x[n-1] + 0.6x[n-2]$$

a. From the I/O equation, calculate the transfer function H(z);

b. From the coefficients of this transfer function (numerator and denominator), write a Matlab program to compute the first 100 samples of the impulse response, using *filter* function.

c. Plot the pole/zero pattern of the impulse response. Compute and plot the magnitude (in dB) and phase (in degree) spectra of the filter, using the *fft*, *abs* and *angle* functions.

**Problem 2:**

Consider the following digital signal:

$$x(n) = 2\sin(2\pi f_1 nT_s) + \sin(2\pi f_2 nT_s) + 1.5\sin(2\pi f_3 nT_s)$$
$$n = 0, 1, \ldots, L-1$$
$$f_1 = 50Hz, f_2 = 60Hz, f_3 = 80Hz, f_s = 1/T_s = 1kHz$$

Write m-file to:

a. Generate the signal above for $L = 50$. Then, compute its 2048-points FFT and plot the magnitude spectrum $|X(f)|$ over $0 \le f \le 200Hz$

b. Calculate the minimum number of samples $L_{min}$ to achieve a sufficient frequency resolution. Given that the minimum number of samples $L_{min}$ is associated with the minimum difference in frequencies among the component frequencies of $x_n(n)$, by the formula:

$$L_{min} = \frac{f_s}{\Delta f_{min}}$$

Generate the corresponding signal and plot its magnitude spectrum.

c. Apply the Hamming window as the equation below to your signal (b) and plot its magnitude spectrum.

$$w_{\text{hamming}}(n) = \begin{cases} 0.54 - 0.46\cos\left(\dfrac{2\pi n}{L-1}\right) & \text{if } 0 \le n \le L-1 \\ 0 & \text{otherwise} \end{cases}$$

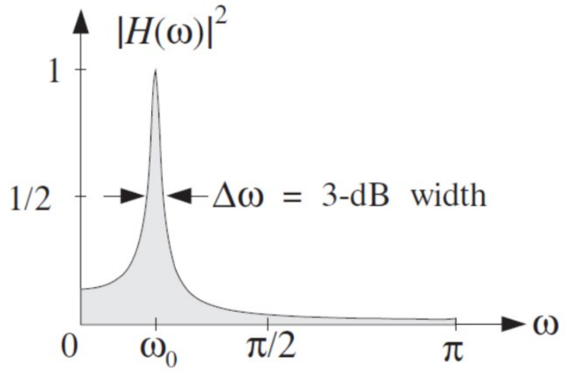
**Problem 3:**

Consider the transfer function of a normalized resonator filter:

$$H(z) = \frac{G}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

$$\begin{cases} G = (1-R)\sqrt{1 - 2R\cos(2w_0) + R^2} \\ a_1 = -2R\cos w_0 \\ a_2 = R^2 \end{cases}$$

with $$R = 1 - \frac{\Delta\omega}{2}$$




a. Express the filter coefficients (G, a1, a2) in terms of sampling frequency $f_s$, peak frequency $f_o$ and width $\Delta f$ .

b. Write a Matlab function ResonatorFilter the returns the filter coefficients using $f_s$, $f_o$ and $\Delta f$, as input parameters.

c. Test your function with $f_s = 30Mhz, f_o = 2Mhz, \Delta f = 0.5Mhz$ and plot the filter frequency response (magnetude spectrum in dB) over $0 \le f \ll 0 \le f \le 15Mhz$ .

**Problem 4:**

If $y(t)$ is a noisy cardiogram signal with random noise defined by

$$y(t) = x(t) + 0.1 * randn([1\ N])$$

where $x(t)$ is the cardiogram signal defined by function $ecg()$ in Matlab, write the MATLAB code to do as follows:

a) Use finite impulse response low-pass filter with order $n = 10$, cut-off frequency $\omega_{cut-off} = 15\ Hz$, and the sample rate 500Hz/second to filter the signal $y(t)$. Plot the filtered signal and original signal $y(t)$ on the same graph. Also, known that the samples taken are 500. (*Hint*: the low pass filter design can be performed by using the function $designfilt()$ in MATLAB code)