جامعة بيرزيت

BIRZEIT UNIVERSITY

Electrical and computer engineering

Digital Signal Processing (DSP)

Assignment No (2)

Submission deadline: **Saturday 26/11/2016** (no later than 11:55 PM) only through Moodle (itc.birzeit.edu)

Question 1:

(a) Using MATLAB, determine the factored form of the following z-transforms:

(i)

$$G1(z) = \frac{2Z^4 - 5Z^3 + 13.48Z^2 - 7.78Z + 9}{4Z^4 + 7.2Z^3 + 20Z^2 - 0.8Z + 8}$$

(ii)

$$G1(z) = \frac{5Z^4 + 3.5Z^3 + 21.58Z^2 - 4.6Z + 18}{5Z^4 + 15.5Z^3 + 31.7Z^2 + 22.52Z + 4.8}$$

And show their pole-zero plots. Determine all possible ROCs of each of the above z-transforms, and describe of their inverse z-transforms (left-sided, right-sided, or two-sided sequences) associated with each of the ROCs.

(b) Use M-file *residues* to determine the z-transform as a ratio of two polynomials in Z^{-1}

From each of the partial-fraction expansions listed below:

(i) $X1(Z) = 3 - \frac{4}{5+Z^{-1}} - \frac{7}{6-Z^{-1}}, |Z| > 0.2$

(ii) $X2(Z) = \frac{-4}{(4+2Z^{-1})^2} + \frac{6}{4+2Z^{-1}} + \frac{5}{1+0.64Z^{-2}}, |Z| > 0.8$

(c) A causal stable LTI system is characterized by an impulse response $h(n) = 1.2\delta(n) + 0.5(-0.5)^n u(n) - 0.6(0.2)^n u(n)$. Use MATLAB to determine the impulse response hi(n) of its inverse system, which is causal and stable.

Question 2:

(a) Apply one of the methods you have learnt to compute discrete-time convolution of the following two finite duration signals:

h(n) ={ 0.5, 1,0.5}, and x(n) = {5,0,2,0,-1,0,1,0, 2,0,4,0,6,0}, noting that every other sample of x(n) is zero.

Suggest a potential DSP application of this particular convolution?

(b) use h(n) to 'recover' samples of an audio signal in Matlab. Use the following steps:

- WAVREAD() – read in an audio file of your choice (see help of wavread() Matlab function)

- Set every other sample to zero to produce an array like x(n) in part (a). call this x(n).

- CONV() – convolve x(n) with h(n) (given in part a above). Let the result of convolution be y(n).

- Plot (stem or plot) a 20 samples section of y(n) together with the same 20 samples section of x(n) and compare them.

Submit the plot and Matlab code and any other observation. For two or more adjacent plots in the same figure see help of matlab. For two or more plots in the same axes, the code structure is stem(n,x,n,y) –see HELP STEM or HELP PLOT.

Don't forget to listen to signal after each processing step, using:

```
-   SOUND( x_original, Fs)    %original audio signal
-   SOUND(x,Fs)               % alternate samples zero
-   SOUND(y,Fs)               % convolution result
```

….. and report what you hear.

Question 3:

A simple reverberation ("reverb") effect can be produced using an FIR filter that has a number of non-zero coefficients spread out at large intervals. The sample below employs 3 coefficients, the first being a unit impulse that reproduces the input signal with zero delay. The next coefficient reproduces a delayed and attenuated "echo" of the input signal, and so on.

```
[x,Fs,nbits] = wavread('cntry.wav');  %use any audio file you like

Sound(x,Fs)    %listen to original audio file

h = zeros(1,4000);

h(1) = 1;

h(1000) = ¼;

h(2000) = 1/16;

y = conv(x,h);
```

```
y=y/max(y);   %normalize filter output

sound(y,Fs);
```

Starting with this code, investigate the result of:

- Adding more filter coefficients (i.e. more echos)
- Varying more interval (i.e. delay) between echo's.
- Varying strength of echo's (i.e. values of filter coefficients)

The values of the 2nd, 3ed, 4th, … coefficients relates to the magnitude of echos and how rapidly echos decay, depends on the reflectivity of the walls in the room/concern hall. The delay interval depends on the size of the room and the round-trip time of the sound.

Briefly summarize your final design and any observations.