

Cairo University  
Faculty of Computers and Artificial Intelligence  
Digital Signals Processing  
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Assignment [10%] 2023

*Hint:* You can use any programming language for this assignment. It is preferred to use python language.

### Problem 1 [Fourier Transform]- 5pts

1. Generate a **1 sec** digital signal from your microphone *pronouncing your name* and save it as an uncompressed wav audio **“.wav”** file. You can use 'Audacity' or any other external software to generate the wav audio file.
2. Apply all-the-points FFT and plot its magnitude part. [2 pts for FFT code + 1 pts for the plot]
3. Apply the inverse FFT. Compare it to the original audio by listening to both. [1 pts for inverse FFT code + 1 pt for the generation of original audio back correctly]

### Problem 2 [Convolution and LTI system] – 5pts

Write a program that calculates  $y[n] = x[n] * h[n]$  (this is not simple multiplication; it is the convolution between  $x$  and  $h$ ), where  $y[n]$  is 598 samples,  $x[n]$  is 500 samples, and  $h[n]$  is 99 samples.

1. Generate an impulse response,  $h[n]$ , according to the algorithm below. This filter kernel is called a **"low-pass windowed-sinc"** filter. When convolved with an input signal, this filter passes sinusoids that have fewer than 25 cycles in 500 samples, and blocks sinusoids with a higher frequency. Make a plot of this signal. [1 pts for the plot of  $h(n)$ ]

for  $i = 0$  to 98

$$h[i] = 0.31752 * \sin(0.314159 * (i - 49.00001)) / (i - 49.00001)$$
$$h[i] = h[i] * (0.54 - 0.46 * \cos(0.0641114 * i))$$

2. Test your program by convolving  $h[n]$  with the signal described below. [1pts for the convolution code]

What should the output of your program be in response to this signal?  
Why? [1pts for the convolution output and explanation]

$x[n] = 1$  for  $n = 0$ ,  $x[n] = 0$  otherwise

3. Generate a more complicated test signal,  $x[n]$ , that consists of two sinusoids added together. The first sinusoid will have an amplitude of 1, and make 6 complete cycles in the 500 samples. The second sinusoid will have an amplitude of 0.5 and make 44 complete cycles in the 500 samples. Make a plot of this signal. [1pts for the convolution output]

4. Test your convolution program by filtering the signal you created in 3, with the filter kernel you created in 2. Plot this signal. Has the filter passed the lower frequency signal, while blocking the higher frequency signal? Comment on the results of the convolution. [1pts for the comment]