Midterm Examination – EE092IU

Date: Nov 5th, 2019

Duration: Unlimited

SUBJECT: DIGITAL SIGNAL PROCESSING-EE092IU

Instructions: Ten A4 pages of notes are allowed in the exam. Answer 5 of 6 given questions

Question 1:

By definition, the first and second of Fibonacci numbers are 0 and 1 (e.g. $x(0) = 0$ and $x(1) = 1$), and each subsequent number is the sum of the previous two.

a) Write the first ten values of the Fibonacci sequence

b) Express and sketch the sequence in (a) versus the Delta function (Impulse)

c) Assumed the signal $x[n]$ in (a) is the input of a system with the impulse response $h[n] = \{-1,2,0,1\}$. Using the convolution table to calculate the output signal $y[n] = x[n] * h[n]$

d) Repeat the question (c) by using the 4-samples-block- Over Add Block algorithm?

Question 2:

A Causal discrete time LTI system is described by

$$y[n] - \frac{3}{4}y[n-1] + \frac{1}{8}y[n-2] = x[n]$$

Where $x[n]$and $y[n]$ are the input and output of the system, respectively

a) Determine the frequency response $H(\omega)$ of the system.

b) Find the impulse response $h[n]$of the system

c) Realize the block diagram of the system

Hint: Use the delay property of the Fourier Transform and $H(\omega) = Y(\omega)/X(\omega)$

Question 3:

Consider the following sound wave, where t is in milliseconds

$$x(t) = \sin(10\pi t) + \sin(90\pi t) + 2\sin(40\pi t)\cos(20\pi t)$$

This signal is prefilterd by an analog antialiasing prefilter $H(f)$ and then sampled at an audio rate of 40 kHz. The resulting samples are immediately reconstructed using an ideal reconstructor. Determine the output $y_a(t)$ of the reconstructor in the following cases and compare it with the audible part of $x(t)$:

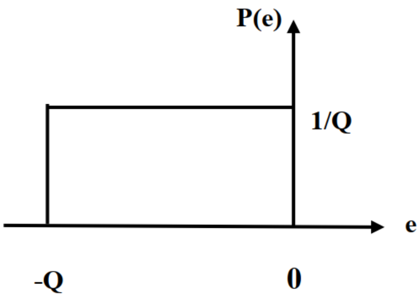
a) When there is no prefilter, that is, $H(f) = 1$.

b) When $H(f)$ is an ideal prefilter with cut off of 20 kHz.

c) When $H(f)$ is a practical prefilter that has a flat passband up to 20 kHz and attenuates at a rate of 48 dB/octave beyond 20 kHz (You may ignore the effects of the phase response of the filter.)

Question 4:

Find the mean and noise power of the quantization if the quantized value x_Q is obtained by truncation of x instead of rounding, show that the truncation error $e = x_Q - x$ will be in the interval $-Q < e \leq 0$. Assume a uniform probability density $p(e)$ over this interval, that is



$$p(e) = \begin{cases} \dfrac{1}{Q}, & -Q < e \leq 0 \\ 0, & \text{otherwise} \end{cases}$$

Question 5:

The Impulse response $h[n]$ of a filter is non-zero over the index range of n be [3,6]. The input signal $x[n]$ to this filter is non-zero over the index range of n be [10,20].

Consider the direct and LTI forms of convolution

$$y[n] = \sum_m h[m]x[n-m] = \sum_m x[m]h[n-m]$$

a) Determine the overall index range n for the output $y[n]$. For each n, determine the corresponding summation range over m, for both the direct and LTI forms.

b) Assume $h[n] = 1$ and $x[n] = 1$ over their respective index ranges. Calculate and sketch the output $y[n]$ using the direct form of the Convolution. Identify (with an explanation) the input on/off transient and steady state parts of $y[n]$.

Question 6:

Determine whether the discrete time systems described by the following I/O equation are linear and/or time-invariant

a) $y[n] = 3x[n] + 5$
b) $y[n] = x^2[n-1] + x[2n]$
c) $y[n] = e^{x[n]}$

Good luck!