



## End Semester Examination

DEC 2021

Max. Marks: 60

Class: BE ELECTRONICS

Course Code: EXC-605

Name of the Course: Digital Signal Processing

Duration: 120 Min

Semester: VII

Branch: ELECTRONICS

Instruction:

- (1) All questions are compulsory.
- (2) Use of scientific calculator is allowed.
- (3) Draw neat diagram.
- (4) Assume suitable data if necessary with justification.

Q No.		Max. Marks	CO
Q.1	Attempt the following Short Answer Questions.		
	1) A professional audio equipment uses a sampling rate of 48 kHz, but consumer audio equipment uses a rate of 44.1 kHz. Therefore, to transfer music from a professional recording to a CD, what should be the sampling rate conversion factor and how this conversion can be achieved ?	2	C03
	2) Justify the need of window function in the design of Linear phase FIR filter? Name two window functions.	2	C02
	3) The block diagram of a three stage decimator which is used to reduce the sampling rate from 80 KHz. Assuming decimation factors of 4, 2 and 5, find the sampling rate at the output <div style="text-align: center;"> <math>x[n]</math> — <math>H_1[z]</math> — <math>D=4</math> — <math>H_2[z]</math> — <math>D=2</math> — <math>H_3[z]</math> — <math>D=5</math> — <math>y[n]</math>  <math>F_s=80 \text{ KHz}</math> <span style="float: right;"><math>F_s = ?</math></span> </div>	1	C03
	4) IIR filter cannot have Linear Phase characteristics. Justify.	2	C02
	5) Transfer function of analog LPF is given below. $H(s) = \frac{1}{s^2 + s + 1}$ Find the transfer function of Analog HPF with cutoff frequency = 5 rad/sec	2	C01
	6) What is Group Delay ? What is the significance of Group Delay in Linear Phase FIR Filter.	2	C02

	<p>7) Given <math>H_d(e^{jw}) = \begin{cases} e^{-3jw} &amp; 0 \leq w \leq \frac{\pi}{2} \\ 0 &amp; \frac{\pi}{2} &lt;  w  &lt; \pi \end{cases}</math></p> <p>What is the order of the FIR filter ?.</p>	2	C02
	<p>8) A Digital Butterworth HPF is required to meet the following Specifications</p> <p>Pass band ripple <math>\leq 1.5</math> dB  Stop band attenuation <math>\geq 30</math> dB  Pass band edge <math>= 5</math> KHz  Stop band edge <math>= 4</math> KHz  Sampling rate <math>= 20</math> KHz</p> <p>Find the order of the filter if BLT is used for filter design.</p>	2	C01
	<p>9) Read the following description of problem. What type of filter should be used to solve the problem of noise filtering?</p> <p>The recording of a heart beat (an <u>ECG</u>), may be corrupted by noise from the <u>AC mains</u>. The exact frequency of the power and its <u>harmonics</u> may vary from moment to moment.</p> <p>One way to remove the noise is to filter the signal with a <u>notch filter</u> at the mains frequency and its vicinity, but this could excessively degrade the quality of the ECG since the heart beat would also likely have frequency components in the rejected range.</p> <p>To circumvent this potential loss of information, ----- filter could be used. The filter would take input both from the patient and from the mains and would thus be able to track the actual frequency of the noise as it fluctuates and subtract the noise from the recording.</p>	1	C04
Q.2	<p>A) Derive one to one frequency transformation from Analog filter frequency to Digital filter frequency.</p> <p>Draw diagram of magnitude spectrum of ideal Band Stop Analog Filter and the corresponding magnitude spectrum of ideal Band Stop Digital Filter using frequency-To-Frequency Transformation Mapping.</p>	5	C01
Q.2	<p>B) Analog filter with transfer function <math>H(s)</math> is given below.</p> $H(s) = \frac{s + 0.1}{(s+0.1)^2 + 16}$ <p>Sampling Time <math>T = 0.1</math> Second</p> <p>Obtain the transfer function of a Digital IIR filter using bilinear transformation. Assume Sampling Time <math>T = 0.1</math> Second.</p>	5	C01

Q.2	<p><b>C)</b> We want to design a Discrete Time Low Pass Filter for a bandlimited voice signal with input frequency range 20 Hz to 3.4 KHz using BLT Method.</p> <p>The specifications are:  Passband Frequency = 4 kHz, with 0.8 dB ripple;  Stopband Frequency = 4.5 kHz, with 50 dB attenuation;</p> <p>NOTE :</p> <p>Assume <math>F_s = 2 \times (\text{Sum of digits of UCID Number}) \text{ KHz}</math></p> <p>If your UCID Number is <b><u>2 0 1 5 1 2 0 0 0 2</u></b></p> <p>Then <math>F_s = 2 \times (2+0+1+5+1+2+0+0+0+2) = 2 \times (13) = 26 \text{ KHz}</math></p> <p>Determine the following :</p> <ol style="list-style-type: none"> <li>1. Digital filter Pass band and Stop band frequencies,</li> <li>2. Corresponding Analog filter Pass band, Stop band frequencies.</li> <li>3. Cutoff frequency of the Digital filter.</li> </ol> <p>-----</p> <p style="text-align: center;"><b>OR</b></p> <p>-----</p> <p>A) Design a second order Butterworth Low Pass filter using Impulse Invariant technique. The cut off frequency required to be 40 Hz.</p> <p>Assume <math>F_s = 10 \times (\text{Sum of digits of UCID Number}) \text{ Hz}</math>.</p> <p>If your UCID Number is <b><u>2 0 1 5 1 2 0 0 0 2</u></b></p> <p>Then <math>F_s = 10 \times (2+0+1+5+1+2+0+0+0+2) = 10 \times (13) = 130 \text{ Hz}</math></p>	5	C01
Q.3	<p>A) Design 8<sup>th</sup> order Linear Phase Low Pass FIR filter with <math>\omega_c = \frac{\pi}{2}</math> using Rectangular window function.</p>	5	C02
Q.3	<p>B) Show the efficient Polyphase Realization of Decimator for <math>D=2</math> using 6<sup>th</sup> order causal FIR filter having <math>h[n] = \{ 0.1, 0.2, 0.3, 0.4, 0.5, 0.6, 0.7 \}</math>.</p> <p>-----</p> <p style="text-align: center;"><b>OR</b></p> <p>-----</p> <p>B) Draw Linear Phase Realization diagram of 6<sup>th</sup> order causal FIR filter having <math>h[n] = \{ 0.1, 0.2, 0.3, 0.4, 0.3, 0.2, 0.1 \}</math>.</p>	5	C03
Q.3	<p>C) Design 6<sup>th</sup> order Linear Phase High Pass FIR filter with cut off frequency <math>\omega_c = \frac{\pi}{2}</math> radian using Frequency Sampling Method.</p>	5	C02

Q.4	A) Derive an expression to obtain optimum filter coefficients of Wiener Filter.	5	C04
Q.4	<p>B) The sampling rate conversion can be as shown in the figure below.</p> <p>Note :</p> <p>If your UCID Number is <b>2015120089</b></p> <p>Then <math>L = 20</math> (First and second digit of UCID)</p> <p>And <math>M = L + 9 = 29</math> (L + Last digit of UCID)</p> <ol style="list-style-type: none"> <li>1) What will be the cut-off frequency of filter with transfer function <math>H(z)</math> ?</li> <li>2) If Sampling frequency of input Signal <math>x[n]</math> is <math>F_x</math> Hz what is the sampling frequency of signal <math>v[n]</math>, <math>w[n]</math> and <math>y[n]</math> ?</li> </ol>	5	C03
Q.4	<p>C) It is required to design a real time system for digital audio password Verification of the user using Mean Square Error Criteria.</p> <p>Input Specification: Audio Signal (i.e. Speech Signal) Assume that Audio Signal of user is already captured.</p> <p>State the following :</p> <ol style="list-style-type: none"> <li>(a) Framework/block diagram of the DSP system. Justify the need of each process.</li> <li>(b) Algorithms/Flowchart.</li> <li>(c) Explain the methodology to address the problem.</li> </ol>	5	C05