S15:DSAA: Final

No queries will be entertained during the exam. If there are errors or ambiguities, make appropriate assumptions, state them and proceed.

180 Minutes **180 Points**

Part A: Briefly answer in the space provided [$5 \times 14 = 70$]

1. A discrete time system is specified by $y[n] = 0.5 \cdot x[n] + 0.2 \cdot x[n-1] + 0.3 \cdot x[n-2]$. Given the input, $x[n] = \delta[n]$, find and plot $y[n]$.

2. 2D DFT is "separable". This makes the 2D DFT computation efficient. By writing the expression for 2D DFT, demonstrate this.

3. Find $y[n] = x[n] * h[n]$ by finding convolution of $x[n] = \{\underline{8}, 4, 2, 1\}$ with $h[n] = \{\underline{1}, 1, 1, 1\}$

4. What are the basis functions/vectors used in DFT? Show that these functions are orthogonal.

5. List three properties of DFT that you like.

6. Your input is a song/music as a digital sample sequence (mono) denoted by $x[n]$. Write matlab code for creating real time filters (with small finite delay) for two output channels $y_{left}[n]$ and $y_{right}[n]$ such that left channel has more low frequency and right has more high frequency. (hint: use simple low pass and high pass filters and emphasize these pass pands. It may be easier to implement in time domain.)

7. You are given an image with a face in the middle of the image. Explain how will you create a motion blurr (special effect) equivalent to camera "zooming in" fast onto the face. Write pseudocode/steps.

8. Consider the following system:

$$y[n] = 3y[n-1] - 2y[n-2] + x[n] + x[n-1]$$

Write a simple matlab code to compute the impulse response of the system in the last question. If the impulse response is infinite in length, output only the first 100 values. $(y[-1] = y[-2] = 0)$

9. When do you prefer to use mean filter and median filters respectively in image filtering? What type of noise they remove? Show examples of 1D images where (a) mean filter is superior than median filter (b) median filter is superior to mean filter.

10. Consider a message generated out of 8 symbols (a,b,c,d,e,f,g,h). A naive encoding will require 3 bits per symbol. If we know the probabilities $P(a) = 0.2; P(b) = 0.1; P(c) = 0.06; p(d) = 0.04; p(e) = 0.4; p(f) = 0.08; p(g) = 0.07; P(h) = 0.05$. we can build a variable length Huffman code that requires lesser (than 3) number of bits. Show the Huffman codes. What will be the gain (compression ratio) with such an encoding?

11. In the context of error correcting Hamming codes, consider the following H matrix.

0.6
0.15
.75
0.05

100 → a = 20
b = 10
c = 6
d = 4
e − 40
f − 8
g − 7
h − 5

70

$$H = \begin{bmatrix} 1 & 1 & 1 & 0 & 1 & 0 & 0 \\ 1 & 1 & 0 & 1 & 0 & 1 & 0 \\ 1 & 0 & 1 & 1 & 0 & 0 & 1 \end{bmatrix}$$

Check whether the following codes have errors; and if there are, assuming there can be at max only one error, recover the original code. (a) 1111111 (b) 0000000 (c) 1010101 (d) 0001111

12. Expand (in the contect of DSAA) FM, ASK, CDMA, GSM, Codec, GPS, DCT,EM, BPS, FFT

13. Show pictorially how Amplitude Modulation modifies the carrier wave. Which one is higher in frequency - message or carrier? Why?

14. Consider the description of a system known as ENCODER in the following paragraph. Based on the description, draw a block diagram for it and write down mathematical relations for the intermediate signals generated in the system:

    The system has a block called PREDICTOR. As succesive samples of the discrete time input signal $f(n)$ are introduced to the encoder, the predictor generates the anticipated value of each sample denoted by $f_1(n)$, based on k past samples. The output of the predictor is then rounded off to the nearest integer by a block called NEAREST INTEGER, denoted by $f_2(n)$, which is used to generate the error signal $e(n)$, which is then encoded by a SYMBOL ENCODER to generate the output.

Part B: Answer in the space provided with necessary technical details. [$10 \times 8 = 80$]

1. Write the encoding and decoding scheme for LZW. Demonstrate how LZW compression can be carried out on the following sequence. Show the dynamically created entries in the table.

   ababababaaaaa

   What is the compression ratio? Note that the original codes were 8 bit (ASCII)and the new codes are 9 bit (with an extended symbol table).

2. Prove that convolution is commutative. i.e., $A * B$ is same as $B * A$.

3. If you know the DFT of $x[n]$, how would you compute the DFT of $x[n - n_0]$. Derive. Also explain how the DFT of the derivative of $x[n]$ (i.e., $x[n] - x[n-1]$) can be computed.

4. With a block diagram explain the overall JPEG compression algorithm. Where is RLC used? Why RLC was chosen for this purpose?

5. Write the expression for DFT. Derive the FFT algorithm by showing the redundant operations/computations. Analyze the complexity. (no need to draw the signal flow graphs)

6. Given two images which are translated versions (in x and y) of each other. How do we estimate the translations? Suggest a solution based on Phase correlation.

7. Why DCT is used in many compression algorithms, and not DFT? Give a list of reasons (based on the reading material provided).

8. Consider a linear orthogonal transformation $y = Ax$. Write DFT in this form. Is $A^{-1} = A^T$ in the case of DFT? What about DCT?