INSTITUTE OF AERONAUTICAL ENGINEERING
(Autonomous)
Dundigal, Hyderabad - 500 043

QUESTION BANK

Department	**Computer Science and Engineering (AI and ML)**				
Course Title	**Image and Speech Processing**				
Course Code	ACAC05				
Program	B.Tech				
Semester	V				
Course Type	Core				
Regulation	UG-20				
Course Structure	Theory			Practical	
	Lecture	Tutorials	Credits	Laboratory	Credits
	3	-	3	-	-
Course Coordinator	Dr. Shaik Jakeer Hussain, Associate Professor				

COURSE OBJECTIVES:

The students will try to learn:

I	The fundamental concepts of digital image processing methods and techniques
II	The algorithms to solve image processing problems and meet design specifications for industry, medicine and defense applications.
III	Methods and digital systems for efficient quantization and coding of speech signals.
IV	The concepts of linear predictive analysis (LPC) for speech synthesis

COURSE OUTCOMES:

After successful completion of the course, students should be able to:

CO 1	**Make use of** image transform techniques for analyzing images in transformation domain for image pre-processing.	Understand
CO 2	**List** the lossy and lossless compression models for achieving image compression.	Analyze
CO 3	**Illustate** the difference between acoustic phonetics and articulatory phonetics for speech processing	Understand
CO 4	**Utilize** digital model designed by sampled speech signal for speech processing applications like speech recognition, speech synthesis and verification.	Apply

CO 5	**Analyze** methods to estimate pitch period to design vocoders,aritificial intelligence voice-controlled assistants like Alexa	Analyze
CO 6	**Apply** linear predictive coding for speech synthesis, compression and spectrographic displays	Apply

QUESTION BANK:

Q.No	QUESTION	Taxonomy	How does this subsume the level	CO's
		MODULE I		
	Digital Image Introduction and Transformation Techniques			
	PART A-PROBLEM SOLVING AND CRITICAL THINKING QUESTIONS			
1A	**QUESTION**	**Blooms Taxonomy Level**	**How Does This Subsume The Level**	**Course Outcome**
1 1	Derive the Haar transformation matrix for N = 8 and explain how it is constructed.	Apply	The learner to **Recall** the image transforms and **Understand** basis function and apply haar transformation for N=8 .	CO1
2	Determine the convolution and correlation between the following images. $\begin{bmatrix} 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 1 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 \end{bmatrix}$ $\begin{bmatrix} 1 & 2 & 3 \\ 4 & 5 & 6 \\ 7 & 8 & 9 \end{bmatrix}$	Apply	The learner to **Recall** the representation of digital image and **Understand** the properties of 2D FT and apply the convolution and correlation property on image.	CO 1

3	A common measure of transmission for digital data is baud rate defined by number of bits transmitted per second. Find how many minutes it would take to transmit a 2048*2048 images with 256 intensity levels using a 33.6k baud modem.	Apply	The learner to **Recall** the size of the image , and **Understand** grey levels, baud rate and apply for the calculation of time taken for the image .	CO 1
4	Compute the Haar transform of the N=2,and N=4 image	Apply	The learner to **Recall** the image transforms and **Understand** basis function and apply haar transformation for N=2 and 4 .	CO1
5	Obtain the slant transforms matrix For N=8.	Apply	The learner to **Recall** the representation of digital image and **Understand** basis function apply slant transforms matrix For N=8.	CO1
6	Compute the 2D DFT of the 4 × 4 grayscale image f(x, y) shown below. $(x, y) = \begin{bmatrix} 1 & 2 & 3 & 4 \\ 5 & 6 & 7 & 8 \\ 1 & 2 & 3 & 4 \\ 5 & 6 & 7 & 8 \end{bmatrix}$	Apply	The learner to **Recall** the representation of digital image and **Understand** the properties of 2D DFT and apply it on image coefficients f(x,y).	CO1
7	What is the Inverse 2D DFT of the transform coefficients F(k, l) given below. $f(k, l) = \begin{bmatrix} 16 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \end{bmatrix}$	Apply	The learner to **Recall** the representation of digital image and **Understand** the properties of 2D DFT and apply it on transform coefficients F (k,l).	CO1

8	Obtain the KL transform basis for the following matrix of samples $f(m,n) = \begin{bmatrix} 4 & -2 \\ -1 & 3 \end{bmatrix}$	Apply	The learner to **Recall** the representation of digital image and **Understand** the properties of 2D DFT and apply KL transform of the 2 x 2 image	CO1
9	Find hadamard forward and reverse transformation of given matrix $u = \begin{bmatrix} 1 & 2 & 1 & 3 \\ 2 & 3 & 4 & 1 \\ 3 & 4 & 2 & 1 \\ 4 & 1 & 2 & 3 \end{bmatrix}$	Apply	The learner to **Recall** the representation of digital image and **Understand** the properties of 2D DFT and apply hadamard forward and reverse transformation	CO1
10	Derive the DCT matrix for N = 4 and verify that it obeys the orthogonality property.	Analyze	The learner to **Recall** the representation of digital image and **Understand** the properties of 2D FT and apply DCT transformation and analyze the matrix for N=4 .	CO1
		PART B-LONG ANSWER QUESTIONS		
1	Explain any four basic relationships between pixels.	Understand	The learner to **Recall** the relationship between pixels and **Understand** the Neighbor of a pixels	CO 1
2	Demonstrate the components of digital image processing system and explain each block.	Understand	The learner to **Recall** the image coordinates and **Understand** elements of image processing system.	CO 1
3	Define digital image. Discuss how digital images are represented with neat diagrams.	Understand	The learner to **Recall** the representation of digital image and **Understand** the processing of digital image.	CO 1
4	Discuss sampling and quantization With necessary diagrams.	Understand	The learner to **Recall** the sampling and quantization techniques and **Understand** the conversion of analog image in to digital image.	CO 1

5	Illusttrate the effect of increasing sampling frequency and quantization levels on image with an example	Understand	The learner to **Recall** the sampling and quantization and **Understand** effect of increasing the sampling frequency and greylevels	CO 1
6	List and explain applications of image processing	Remember	——	CO 1
7	What is spatial resolutions? Discuss the effect on the image by reducing it.	Understand	The learner to **Recall** the sampling and quantization **Understand** effect of increasing the sampling frequency.	CO 1
8	Explain the concept of non-uniform sampling and quantization with examples.	Understand	The learner to **Recall** quantization and **Understand** the non uniform quantization.	CO 1
9	Discuss the most commonly used distance measures in image processing	Understand	The learner to **Recall** digital image, neighbours and **Understand** various distance measures.	CO 1
10	The image refers to a two dimensional light intensity function. Discuss in detail.	Understand	The learner to **Recall** the Gray levels and **Understand** the Gray level to binary conversion.	CO 1
11	Discuss the image acquisition using a single sensor, sensor strips and sensor arrays.	Understand	The learner to **Recall** image acquisition and **Understand** various sensors	CO 1
12	What is Hadamard transform? Explain in detail and Write its properties.	Understand	The learner to **Recall** nthe image transforms and **Understand** basis function of transform.	CO1
13	Explain about KL Transform and Write its properties .	Understand	The learner to **Recall** the image transforms and **Understand** basis function of KL transform	CO1
14	Define the following two properties of 2D-DFT: i) Convolution ii) Correlation	Understand	The learner to **Recall** the discrete fourier transform and **Understand** the properties of 2D DFT	CO1

15	Illustrate the following mathematical operations on digital images i) Array versus Matrix operations ii) Linear versus Nonlinear Operations	Understand	The learner to **Recall** the fundamental concept of images and **Understand** various mathematical operations on digital image.	CO 1
16	Describe the need of image transform? List out various transform used in image processing.	Understand	The learner to **Recall** the image transforms and **Understand** different transforms.	CO1
17	Explain the following terms: (i) Adjacency (ii) Connectivity (iii) Regions (iv) Boundaries	Understand	The learner to **Recall** the concept of pixels and **Understand** the relationship between pixels	CO 1
18	State the following two properties of 2D-DFT i) Translation ii) Rotation	Understand	The learner to **Recall** the discrete fourier transform and **Understand** the properties of 2D DFT	CO1
19	Derive the basis function for walsh transform	Understand	The learner to **Recall** the image transform and **Understand** the basis function of walsh.	CO1
20	Define 4,8-,m-adjacancy.Explain the lengths of shortest 4,8,m-paths between pixels with examples.	Understand	The learner to **Recall** the concept of pixels and **Understand** the relationship between pixels	CO 1
PART C-SHORT ANSWER QUESTIONS				
1	Define digital image processing	Remember	—–	CO 1
2	Write any two origins of image processing?	Remember	—–	CO 1
3	Mention different types of digital images.	Remember	—-	CO 1
4	Mention different bands in electromagnetic spectrum.	Remember	—-	CO 1
5	Which step is the objective of digital image processing?	Remember	—-	CO 1
6	Explain the hardware components of an image processing.	Understand	The learner to **Recall** the digital image and **Understand** the components of an image processing	CO 1
7	What is meant by Pixel?	Remember	—-	CO 1

8	What are the different fields in which Digital Image Processing is used?	Remember	—-	CO 1
9	What is the need of image processing?	Remember	—-	CO 1
10	Explain connectivity and path in relationship between pixels.	Understand	The learner to **Recall** the digital image and **Understand** the relationship between pixels	CO 1
11	Discuss about 4,8,diagonal neighbours.	Understand	The learner to **Recall** the relationship between pixels and **Understand** the image connectivity	CO 1
12	Explain region and boundary in the image.	Understand	The learner to **Recall** the image connectivity and **Understand** the region and boundary of an image	CO 1
13	Write the changes in sizes of different resolution images?	Remember	——	CO 1
14	What is meant by illumination and reflectance in image function?	Remember	——	CO 1
15	Define histogram.	Remember	———	CO 1
16	List the different components in a simple Image formation model.	Remember	——	CO 1
17	Explain about sampling role in digitization process.	Understand	The learner to **Recall** the sampling theorem and **Understand** the digitization process	CO 1
18	Explain about quantization in digitization process.	Understand	The learner to **Recall** the sampling and quantization techniques and **Understand** the digitizing amplitude values	CO 1

19	Obtain histogram equalization for the following image segment of size 5 X 5? Write the inference on image segment before and after equalization 20 20 20 18 16 15 15 16 18 15 15 15 19 15 17 16 17 19 18 16 20 18 17 20 15	Analyze	The learner to **Recall** the operation of pixels and gray levels and **Understand** the histogram equalization and **apply** image segment of size 5 X 5 and **analyze** the result with original image	CO 1
20	Apply the steps involved in histogram equalization on the image. $\begin{bmatrix} 4 & 4 & 4 & 4 & 4 \\ 3 & 4 & 5 & 4 & 3 \\ 3 & 5 & 5 & 5 & 3 \\ 3 & 4 & 5 & 4 & 3 \\ 4 & 4 & 4 & 4 & 4 \end{bmatrix}$	Apply	The learner to **Recall** the operation of pixels and gray levels and **Understand** the histogram equalization and apply image segment of size 5 X 5 and analyze the result with original image	CO 1

MODULE II

Image Compression

PART-A PROBLEM SOLVING AND CRITICAL THINKING QUESTIONS

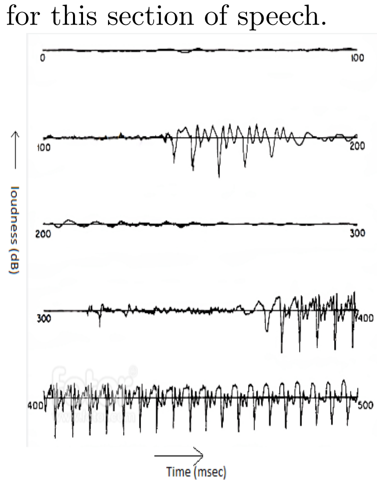
1	Determine the Huffman code procedure for the following data. SYMBOL PROBABILITY a1 0.1 a2 0.4 a3 0.06 a4 0.1 a5 0.04 a6 0.3	Analyze	The learner to **Recall** compression technique **Understand** the encoding process then **apply** Huffman code procedure and **analyze** fallowing symbol and probability values.	CO 2
2	A source emits letters from an alphabet A={a1,a2,a3,a4,a5} with probabilities P(a1)=0.2, P(a2) = 0.4 , P(a3) = 0.2 , P(a4) = 0.1 and P(a5) = 0.1.Find Huffman code for this source? Find the average length of the code and its redundancy	Analyze	The learner to **Recall** compression technique **Understand** the encoding process then **apply** Huffman code for given source and **analyze** average length of the code and its redundancy	CO 2

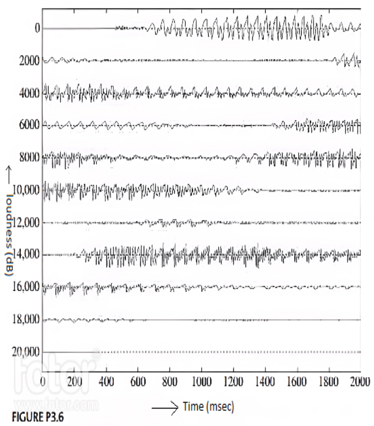
3	For the image shown below compute the compression ratio that can be achieved using Huffman coding. $$\begin{bmatrix} 3 & 3 & 3 & 2 \\ 2 & 3 & 3 & 3 \\ 3 & 2 & 2 & 2 \\ 2 & 1 & 1 & 0 \end{bmatrix}$$	Apply	The learner to **Recall** compression technique **Understand** the encoding process then **apply** uffman code for given image.	CO 2
4	A source emits three symbols A,B,C with a probability{0.5,0.25,0.25} respectively. Construct an arithmetic code to encode the word 'C A B'	Apply	The learner to **Recall** compression technique **Understands** the encoding process then **apply** arithmetic coding for given image.	CO 2
5	Encode the word a1,a2,a3,a4 using arithmetic coding and find tag for the given probabilities.a1=0.2, a2= 0.2,a3=0.4, a4=0.2	Apply	The learner to **Recall** the codeword and **Understand** the procedure for arithmetic coding and **apply** it to find the word length and code length	CO 2
6	Perform Huffman algorithm for the following intensity distribution, for a 64 x 64 image. Obtain the coding efficiency and compare with that of uniform length code. R0 = 1008, r1 = 320 , r2 = 456 , r3 = 686 ,r4 = 803 .r5 = 105 ,r6 =417 ,r7 = 301	Analyze	The learner to **Recall** the code efficiency and **Understand** the intensity distribution and **apply** Huffman algorithm and compare with that of uniform length code.	CO 2
7	Explain in detail about histogram processing.	Understand	The learner to **Recall** the probability of occurrence of gray levels and **Understand** the histogram processing.	CO 2

8	Obtain Huffman coding for the source symbols S={S0 ,S1, S2, S3,S4} and the corresponding probabilities P= {0.4 ,0.2,0.2,0.1,0.1}.	Understand	The learner to **Recall** the encoding techniques and **Understand** the uffman coding for source symbols and probabilities.	CO 2
9	Explain the steps in histogram equalization.	Understand	The learner to **Recall** the operation of pixels and gray levels and **Understand** the histogram equalization .	CO 2
10	Why JPEG is better than a Raw free? Summarize the merits and de-merits.	Understand	The learner to **Recall** the compression technique and **Understand** the JPEG is better than a Raw free.	CO 2
PART B-LONG ANSWER QUESTIONS				
1	Differentiate Lossless and Lossy compression.	Understand	The learner to **Recall** the techniques and **Understand** the redundancies in a digital image	CO 2
2	Demonstrate run length encoding with example.	Understand	The learner to **Recall** the techniques and **Understand** the compression models in digital image	CO 2
3	Explain the need for image compression .How run length encoding approach is used for compression? Is it lossy? Justify.	Understand	The learner to **Recall** the need for image compression and **Understand** the run length encoding	CO 2
4	Demonstrate arithmetic coding with example.	Understand	The learner to **Recall** the need for image compression and **Understand** the arithmetic Coding.	CO 2
5	Explain the average length of the code. Is Huffman code uniquely decodable? If so , Justify your answer.	Understand	The learner to **Recall** average length of the code and **Understand** uffman code.	CO 2
6	How an image is compressed using JPEG image compression standard?	Understand	the learner to **Recall** the compression standard and **Understand** the JPEG image.	CO 2
7	Explain in detail about the arithmetic coding	Understand	the learner to **Recall** the compression techniques and **Understand** the arithmetic coding in a digital image	CO 2

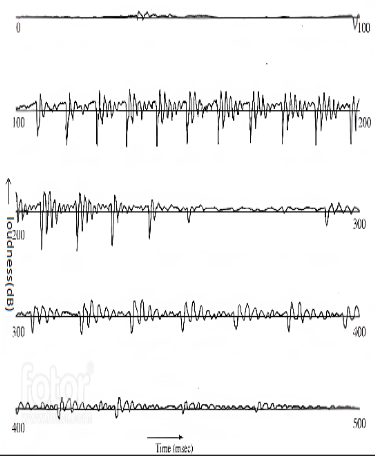
8	Describe run length encoding with examples	Understand	the learner to **Recall** the compression techniques and **Understand** the run length encoding in a digital image	CO 2
9	What is mean by bit plane slicing and write the applications of it	Understand	the learner to **Recall** the image to take number of bits and **Understand** bit plane slicing	CO 2
10	List out and explain in detail about the image compression	Understand	——	CO 2
11	Relate the JPEG compression standard and the steps involved in JPEG compression	Understand	——	CO 2
12	Which type of method to generating variable length codes with an example.	Understand	the learner to **Recall** the compression techniques and **Understand** the variable length encoding in a digital image	CO 2
13	Show arithmetic encoding process with an example.	Understand	the learner to **Recall** the FT and prepare algorithms and **Understand** the arithmetic encoding	CO 2
14	Why LZW coding and what is the need for image relate with an example.	Understand	The learner to **Recall** the compression techniques and **Understand** the LZW coding compression in a digital image	CO 2
15	Relate the JPEG compression standard and the steps involved in JPEG compression.	Understand	The learner to **Recall** the compression techniques and **Understand** the JPEG compression in a digital image	CO 2
16	Select and match the Redundancies and their removal methods with examples	Understand	The learner to **Recall** the compression techniques and **Understand** the removal methods in a digital image	CO 2
17	Demonstrate with example source encoder and decoder	Understand	The learner to **Recall** the FT and prepare algorithms and **Understand** the coding for binary arithmetic process for source encoder and decoder.	CO 2

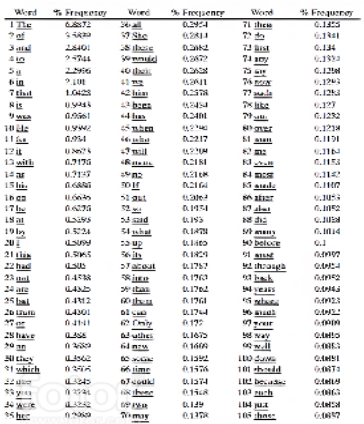
18	Compare error free compression and lossy compression	Understand	The learner to **Recall** the FT and prepare algorithms and **Understand** to image intensity levels for error free compression	CO 2
19	Relate JPEG 2000 standard with merits and demerits	Understand	The learner to **Recall** the compression techniques and **Understand** the JPEG compression in a digital image	CO 2
20	Draw a transform coding system and relate with image compression.	Understand	The learner to **Recall** the compression techniques and **Understand** the transform coding compression in a digital image	CO 2
PART C-SHORT ANSWER QUESTIONS				
1	What is image compression?	Remember	——-	CO 2
2	What is the need for Compression?	Remember	——-	CO 2
3	What are the types of redundancy?	Remember	——-	CO 2
4	Define coding redundancy.	Remember	——-	CO 2
5	Define interpixel redundancy.	Remember	——-	CO 2
6	Define Psychovisual redundancy.	Remember	——-	CO 2
7	What is run length coding?	Remember	——-	CO 2
8	Define compression ratio.	Remember	——-	CO 2
9	Define source encoder.	Remember	——-	CO 2
10	Define channel encoder.	Remember	——-	CO 2
11	What are the operations performed by error free compression?	Remember	——-	CO 2
12	What is Variable Length Coding?	Remember	——-	CO 2
13	What is JPEG?	Remember	——-	CO 2
14	What are the coding systems in JPEG?	Remember	——-	CO 2
15	What are the basic steps in JPEG?	Remember	——-	CO 2

16	State whether the following huffman code 0,01, 10,011 for the given symbols a1,a2,a3,a4 is uniquely decodable or not.	Remember	——-	CO 2
17	Match the Lossless compression–coding with example.	Remember	——-	CO 2
18	Explain the Data Compression and Data Redundancy.	Understand	The learner to **Recall** the FT and prepare algorithms and **Understand** to image intensity levels for data compression.	CO 2
19	Draw and relate block diagram of Lossy Compression.	Remember	——-	CO 2
MODULE III				
Fundamentals of Human Speech Production				
PART A-PROBLEM SOLVING AND CRITICAL THINKING QUESTIONS				
1	The waveform plot shows a 500 msec section(100msec/line) of speech waveform. (a) Indicate the regions of voiced speech, unvoiced speech and silence (b)For the voiced regions estimate the pitch period on a period by period basis and plot the pitch period vs time for this section of speech. 	Apply	The leaners will try to **recall** the speech signal representation and **understand** voiced, unvoiced, silence,stop parts in speech signal waveform.	CO 3

2	Segment the waveform in Figure below into regions of voiced speech (V), regions of unvoiced speech (U), and regions of silence (or background signal) (S). The waveform corresponds to the sentence “Good friends are hard to find.”  FIGURE P3.6 Waveform for the sentence “Good friends are hard to find.”	Apply	The leaners will try to **recall** the speech signal representation and and **understand** voiced, unvoiced, silence,stop parts in speech signal waveform.	CO 3
3	Identify the sounds (from the ARPAbet alphabet) of all the words in the following pair of sentences: 1. She eats some Mexican nuts. 2. Where roads stop providing good driving. What can you say about the sound sequence at word boundaries for these two sentences	Analyze	The learner will **recall** the concept of phonetics in english language and **understand** the difference in vowels, semivowels,diphthongs and **apply** it to the given sentences to analyze how the speech signal waveform is generated	CO 3
4	Identify the place of articulation and the manner of articulation for all the sounds in the sentence: “I enjoy the simple life.” Show your results as a sequential table of place and manner features, over time (i.e., versus the sounds in the utterance).	Understand	The learner will recall the concept of phonetics in english language and apply it forthe given sentences	CO 4

5	Note that the reflection coefficients for the junction of two lossless acoustic tubes of areas A_k and A_{k+1} can be written as either $$r_k = \frac{\frac{A_{k+1}}{A_k} - 1}{\frac{A_{k+1}}{A_k} + 1},$$ $$r_k = \frac{1 - \frac{A_k}{A_{k+1}}}{1 + \frac{A_k}{A_{k+1}}}.$$ (a) Show that since both A_k and A_{k+1} are positive, $-1 <= r_k <= 1$. (b) Show that if $0 < A_k < \infty$ and $0 < A_{k+1} < \infty$, then $-1 < r_k < 1$	Analyze	The learners will try to recal the lossless tube boundary conditions and analyze the conditions	CO 4
6	Draw the digital network diagram for the parallel form implementation of discrete time vocal tract model transfer function,V(z) for M = 3	Understand	The leaner will recall the concept of vocal tract model transfer function and draw the signal flow graph	CO 4

7	The waveform plot of Figure below is for the word "cattle." Note that each line of the plot corresponds to 100 msec of the signal. (a) Indicate the boundaries between the phonemes; i.e., give the times corresponding to the boundaries /K/ /AE/ /T/ /AX/ /L/. (b) Indicate the point where the fundamental frequency of voiced speech is (i) the highest and (ii) the lowest. What are the approximate pitch frequencies at these points? 	Apply	The leaners will try to recall the pitch frequecy and speech signal concepts and apply the same for the given example	CO 3

<table>
<tr>
<td>8</td>
<td>Figure below shows the most common 105 words in the Brown corpus of American English words from a corpus of more than a million words. What general statement(s) can you make about the most commonly occurring words in this list? What percentage of these top 105 words have more than one syllable

<table>
<tr><th>Word</th><th>% Frequency</th><th>Word</th><th>% Frequency</th><th>Word</th><th>% Frequency</th></tr>
<tr><td>1 The</td><td>6.8872</td><td>36 all</td><td>0.2954</td><td>71 then</td><td>0.1355</td></tr>
<tr><td>2 of</td><td>3.5839</td><td>37 She</td><td>0.2814</td><td>72 do</td><td>0.1341</td></tr>
<tr><td>3 and</td><td>2.8401</td><td>38 there</td><td>0.2682</td><td>73 first</td><td>0.134</td></tr>
<tr><td>4 to</td><td>2.5744</td><td>39 would</td><td>0.2672</td><td>74 any</td><td>0.1333</td></tr>
<tr><td>5 a</td><td>2.2996</td><td>40 their</td><td>0.2628</td><td>75 my</td><td>0.1298</td></tr>
<tr><td>6 in</td><td>2.101</td><td>41 we</td><td>0.2611</td><td>76 now</td><td>0.1293</td></tr>
<tr><td>7 that</td><td>1.0428</td><td>42 him</td><td>0.2578</td><td>77 such</td><td>0.1283</td></tr>
<tr><td>8 is</td><td>0.9943</td><td>43 been</td><td>0.2454</td><td>78 like</td><td>0.127</td></tr>
<tr><td>9 was</td><td>0.9661</td><td>44 has</td><td>0.2401</td><td>79 our</td><td>0.1232</td></tr>
<tr><td>10 He</td><td>0.9392</td><td>45 when</td><td>0.2298</td><td>80 over</td><td>0.1218</td></tr>
<tr><td>11 for</td><td>0.934</td><td>46 who</td><td>0.2217</td><td>81 man</td><td>0.1191</td></tr>
<tr><td>12 it</td><td>0.8623</td><td>47 will</td><td>0.2209</td><td>82 me</td><td>0.1164</td></tr>
<tr><td>13 with</td><td>0.7176</td><td>48 more</td><td>0.2181</td><td>83 even</td><td>0.1153</td></tr>
<tr><td>14 as</td><td>0.7137</td><td>49 no</td><td>0.2168</td><td>84 most</td><td>0.1142</td></tr>
<tr><td>15 his</td><td>0.6886</td><td>50 If</td><td>0.2164</td><td>85 made</td><td>0.1107</td></tr>
<tr><td>16 on</td><td>0.6636</td><td>51 out</td><td>0.2063</td><td>86 after</td><td>0.1053</td></tr>
<tr><td>17 be</td><td>0.6276</td><td>52 so</td><td>0.1954</td><td>87 also</td><td>0.1052</td></tr>
<tr><td>18 at</td><td>0.5393</td><td>53 said</td><td>0.193</td><td>88 did</td><td>0.1038</td></tr>
<tr><td>19 by</td><td>0.5224</td><td>54 what</td><td>0.1878</td><td>89 many</td><td>0.1014</td></tr>
<tr><td>20 I</td><td>0.5099</td><td>55 up</td><td>0.1865</td><td>90 before</td><td>0.1</td></tr>
<tr><td>21 this</td><td>0.5065</td><td>56 its</td><td>0.1829</td><td>91 must</td><td>0.0997</td></tr>
<tr><td>22 had</td><td>0.505</td><td>57 about</td><td>0.1757</td><td>92 through</td><td>0.0954</td></tr>
<tr><td>23 not</td><td>0.4538</td><td>58 into</td><td>0.1763</td><td>93 back</td><td>0.0952</td></tr>
<tr><td>24 are</td><td>0.4325</td><td>59 than</td><td>0.1762</td><td>94 years</td><td>0.0945</td></tr>
<tr><td>25 but</td><td>0.4312</td><td>60 them</td><td>0.1761</td><td>95 where</td><td>0.0923</td></tr>
<tr><td>26 from</td><td>0.4301</td><td>61 can</td><td>0.1744</td><td>96 much</td><td>0.0922</td></tr>
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<tr><td>35 her</td><td>0.2969</td><td>70 may</td><td>0.1378</td><td>105 those</td><td>0.0857</td></tr>
</table>
FIGURE P2.14
List of 105 most common words in Brown corpus of American English</td>
<td>Apply</td>
<td>The learners will recal the concept of phonetics and apply it to the given example</td>
<td>CO 3</td>
</tr>
<tr>
<td>9</td>
<td>How many consonant pairs can occur within a single word? Name them. What rule do they obey in terms of place and manner of articulation. How many consonant triplets can occur within a single word? Name them. What rule do they obey in terms of place and manner of articulation.</td>
<td>Undertand</td>
<td>The leaners will try to recall the concept of consonants and non consonants in english language</td>
<td>CO 3</td>
</tr>
</table>

10	In the analysis of the acoustic tube model of the human vocal tract, we have made many assumptions to simplify the solutions. These include the following: • The tube is lossless; • The tube is rigid; i.e., its cross-sectional area does not change with time; • The tube is constructed as a concatenation of a number of uniform sections, each with a different length and cross-sectional area; • The tube's total length is fixed. . State the effects of each of these assumptions on the properties of the resulting system (volume velocity) transfer function of the vocal tract; i.e., had any of the assumptions been eliminated, how would the vocal tract transfer function have changed.	Understand	The learners will try to recall the transfer function of vocal tract and then consider the effects of the conditions given	CO 4
PART B-LONG ANSWER QUESTIONS				
1	Demostrate and explain the human speech production system with a schematic view of human vocal tract.	Understand	The learner recall the human speech production and understand the mechanism of speech production	CO 3
2	Classify the sound produced by human speech and illustrate the continuant sounds using waveforms	Analyze	The leaners will recall the different sound produced by human speech and understand the continuant sound produced with waveforms	CO 3

3	Compare (1) Diphothongs (2) Semivowels (3) Stops (4) Affricates with the help of waveforms and explain it	Analyze	The leaners will try to recall the sound waves produced and analyze the diphothongs, semivowels,sops,affricatives with waveforms	CO 3
4	Describe nasal consonants and fricatives using examples and waveforms	Understand	The learners wil try to recall the sound waves produced and describe the nasal consonants and fricatives with waveforms	CO 3
5	Examine the various methods for speech production and explain in detail the acoustic theory of speech production	Apply	The learners will understand the speech production in human and apply the acoustic theory of speech production for sound production	CO 3
6	Identify and explain with a schematic the speech production mechanism	Understand	The learners will try to recall the mechanism for speech production	CO 3
7	What is a phoneme? Differentiate between vowels, semivowels, diphtongs and consonants using waveforms	Understand	The learners will try to recall phonemes of english language and then try to differentiate between vowels,semivowels,diphthong and consonants	CO 3
8	Distinguish between voice and unvoiced speech signal based on its significance in speech processing	Analyze	The learners will try to recall the speech waveforms and understand the difference between voiced and unvoiced signals to apply for speech processing and analyze its performance	CO 3
9	What are the speech properties and illustrate the properties of speech in waveform?	Understand	The learners will try to recall the speech signal waveforms and illustrate the properties in speech waveform.	CO 3
10	Show the formant frequencies and vocal tract configuration schematic for vowel and diphthongs and explain the sound production	Understand	The leaners will try to recall the formant frequencies of speech signal and try to understand vocal tract configuration to produce vowels and diphthongs	CO 3

11	Draw and explain the discrete time model and radiation model of vocal tract for speech production in detail	Understand	The leaners will try to understand the discrete time model and radiation model of vocal trct for producing speech signals in human	CO 4
12	List out the different losses existing in vocal tract and discuss their effects of them in speech production.	Analyze	The leaners will try to understand the discrete time model of vocal tract and recall the losses existing in it to analyze the effects on speech production	CO 4
13	Demostrate the sound production in vocal tract using schematic of vocal system and vocal cord model	Understand	The leaners will try to recall the schematic of vocal cord and vocal cord model for sound production	CO 4
14	Give an account of wave propagation in lossless tube with equations and schematic	Analyze	The learners will try to recall the lossless tube schematics and apply the wave propagations equations to analyze the sound propagation	CO 4
15	Comment upon the boundary conditions at lips and glottis for lossless tube vocal tract model	Analyze	The learners will try to recall and understand the schematic of lossless tube of vocal tract and analyze the boundary conditions	CO 4
16	Derive the transfer function of lossless tube model using z-transforms	Apply	The learners will try to recall the lossless tube model and understand the z-transforms to derive the transfer function for lossless tube model	CO 4
17	What are the distinctive features of sounds and draw the chart depicting breakdown of consonants of english language.	Apply	The learners will try to recall the distinctive features of sound and consonants of english language to understand the breakdown process of consonants based on articulation.	CO 3
18	Draw the schematic vocal tract configurations for production of vowels /a/,/i/ and /u/. How it affects the formants frequencies?	Understand	The learners will try to recall the schematic of vocal tract for producing vowels and understand the effects of vocal tract shape on formant frequencies	CO 3

19	Depict the second vs first formant frequencies for vowels of english language and explain the vowel triangle.	Understand	The learners will try to recall the articulation for vowels and depict the formant frequencies using vowel triangle	CO 3
20	Examine the effect of nasal coupling on acoustic theory of speech production using nasal tube mathmatical model	Analyze	The learners will try to recall the acoustic theory to understand speech production and apply on nasal tube mathmatical model and analyze the effect of nasal coupling	CO 4
PART C-SHORT ANSWER QUESTIONS				
1	Write a short note on semivowels and nasal with artriculation diagrams	Understand	The learners will try to recal the phonetics of english language and understand semivowels and nasal phonetics	CO 3
2	Explain the differnce between uniform and non uniform quantization. Also illustrate which is more advantageous in speech processing with example.	Apply	The learners will try recall the quantization and try to differentiate between uniform and non uniform quantization and apply it to speech signal	CO 3
3	How do you divide the speech produced by human and state their differences.	Analyze	The leaners recall vocal tract system of human to o understand how speech is produced and apply the phonetics theory to analyze speech waveforms to differentiate between phonetics of the language	CO 3
4	What are the Affricates in english language	Understand	The learners will try to recal the phoetics in language and understand affricates in speech waveform	CO 3
5	Describe the phoneme classes of english language using examples and waveforms	Apply	The learners recall the english language and understand the different classes of phonetics and apply it on english sentences and analze the speech waveform	CO 3

6	Briefly explain the effects existing in realistic acoustic models for speech production	Understand	The learners recall the acoustic models and understand the effects or losses existing in the real acoustic models.	CO 3
7	Explain in detail the human hearing hearing mechanism with suitable diagrams	Understand	The learners will try to recall the organs in ear involved in hearing and understand the role of each part in recognizing sound	CO 3
8	Draw a labelled signal flow graph for lossless tube model of order 3 using delays corressponding to half of sampling period.	Apply	The learners will recal the lossless tube model and understand the signal flow graph and apply it on lossless tube model to determine transfer function	CO 4
9	Illustrate the vocal tract and nasal tract using diagram and explain it for speech production.	Apply	The learners will try to recall the vocal and nasal tract models and understand how speech is produced using these models	CO 4
10	Explain the discrete-time model for speech production	Understand	The learners recall the discrete time model and understand how speech is produced by the model	CO 4
11	Demostrate the boundary condition at glottis in lossless tube with digital delay blocks and equations	Analyze	The learners recal the lossless tube model to understand the boundary conditions at glottis and apply the digital signal processing techniques to analyze the boundary conditions	CO 4
12	Discuss with example how sampling and quantization process is applied to speech signals	Apply	The learners recall the sampling theorem to understand qunatization steps involved to apply on speech signal	CO 4
13	Discuss the relation between various speech parameters.	Understand	The learners recall the speech waveform and understand the parameters involved in speech production to find the realtion between them	CO 3

14	Distinguish between the vocal tract models, radiation model and excitation models	Analyze	The learners recall the z-transforms to understand the transfer function derivation and apply on the digital models of sampled speech sounds to analyze the speech waveform produced	CO 4
15	List the organs involved in speech production. Specify the role each of these organs in the speech production process	Analyze	The leaners will recall the organs in human vocal tract and understand the role of each organ in speech production to analyze the speech signals	CO 3
16	Give examples for the following speech sounds (a) Fricatives (b) Stops (c) Nasals and explain how these sounds can be differentiates	Apply	The learners recall the phonetics to understand their difference using waveforms and apply to the examples	CO 3
17	Comment on the relation between the transfer function of vocal tract and digital filters	Apply	The learners recall the digital filter to understand digital filter frequency response and comment on the relation between them	CO 4
18	Derive the transfer function of uniform lossless tube using z-transforms	Apply	The learners recall the uniform lossless tube to understand the system interpretation of uniform vocal tract and apply z-transforms to derive the transfer function	CO 4
19	Illustrate the boundary condition at lips in lossless tube with digital delay blocks and equations	Analyze	The learners recal the lossless tube model to understand the boundary conditions at lips and apply the digital signal processing techniques to analyze the boundary conditions	CO 4
20	Demostrate the wavepropagation of sound in concatenated lossless tube using signal flow graph and derive the reflection coefficient	Analyze	The learners recall the lossless tube to understand the wave propagation and apply digital signal processing techniques to analyze signal flow graph to derive relection coefficiet	CO 4

MODULE IV				
Time domain models for speech processing				
PART A-PROBLEM SOLVING AND CRITICAL THINKING QUESTIONS				
1	The rectangular window is defined as $w_R[n] = \begin{cases} 1 & 0 \leq n \leq L-1 \\ 0 & \text{otherwise.} \end{cases}$ The Hamming window is defined as $w_H[n] = \begin{cases} 0.54 - 0.46\cos[2\pi n/(L-1)] & 0 \leq n \leq L-1 \\ 0 & \text{otherwise.} \end{cases}$ (a) Show that the Fourier transform of the rectangular window is $W_R(e^{j\omega}) = \frac{\sin(\omega L/2)}{\sin(\omega/2)} e^{-j\omega(L-1)/2}.$ (b)Express $w_H[n]$ in terms of $w_R[n]$ and thereby obtain an expression for $W_H[e^{j\omega}]$ in terms of $W_R[e^{j\omega}]$.	Apply	The learners recall fourier transform of signals and understand the window transfer function to apply on fourier transform coefficients	CO 5
2	The short-time zero-crossing rate was defined $Z_{\hat{n}} = \frac{1}{2L} \sum_{m=\hat{n}-L+1}^{\hat{n}} \lvert \text{sgn}(x[m]) - \text{sgn}(x[m-1]) \rvert.$ Show that Zn^ can be expressed as $Z_{\hat{n}} = Z_{\hat{n}-1} + \frac{1}{2L} \{ \lvert \text{sgn}(x[\hat{n}]) - \text{sgn}(x[\hat{n}-1]) \rvert - \lvert \text{sgn}(x[\hat{n}-L]) - \text{sgn}(x[\hat{n}-L-1]) \rvert \}.$	Apply	The learner recal the z-transforms to understand autocorrelation and apply to short time zero crossing schematic to calculate the transfer function	CO 5

3	The short-time average magnitude difference function (AMDF) of the signal x (n) is defned as $\gamma_n(k) = \frac{1}{N}\sum_{m=0}^{N-1}\lvert x(n+m)-x(n+m-k)\rvert$ (a) Using the inequality $\frac{1}{N}\sum_{m=0}^{N-1}\lvert x(m)\rvert \leqslant \left(\frac{1}{N}\sum_{m=0}^{N-1}\lvert x(m)\rvert^2\right)^{1/2}$ show that $\gamma_n(k) \leqslant \left[2(\hat{R}_n(0)-\hat{R}_n(k))\right]^{1/2}$	Apply	The learners recall z-transforms and understand the AMDF for speech signal and apply it to calculate short-time average autocorrelation of a speech signa;	CO 5
4	Consider the signal $x(n) = \cos\omega_0 n \quad -\infty < n < \infty$ (a) Find the long-time autocorrelation function, $\phi(k)$, for x (n) (b) Sketch $\phi(k)$ as a function of k (c) Find and sketch the long-time autocorrelation function of the signal $y(n) = 1 \text{ if } x(n) \geqslant 0$; $= 0 \text{ if } x(n) < 0$	Apply	The learners recall the autocorrellation function to understand long time and short time autocorrelation definitions and apply it to sketch the frequency response	CO 5
5	Give the equation for average short-time average zero-crossing rate and show that $Z_n = Z_{n-1} + \frac{1}{2N}\left\{\lvert sgn[x(n)]-sgn[x(n-1)]\rvert - \lvert sgn[x(n-N)]-sgn[x(n-N-1)]\rvert\right\}$	Analyze	The learners recall the concept of z-tranforms to understand autocorrelation to apply on speech signal to analyze the short time autocorrelation frequency response	CO 5
6	Consider the periodic impulse train calculate the autocorrelation function, $R_n(k)$ using rectangular window. (a) How would the result change if the window is Hamming window (b) Find and sketch the modified short time autocorrelation function.	Analyze	The learners recall z-transform to undestand windowing function and autocorellationm to apply on impulse train function to analyze the effect of window length on frequency response of autocorrellation.	CO 5

7	Assume there are seven independent pitch detectors, each having a probability p of correctly estimating pitch period, and probability $1 - p$ of incorrectly estimating pitch period. The decision logic is to combine the seven pitch estimates in such a way that an overall error is made only if four or more of the individual pitch detectors make an error. Derive an explicit expression for the probability of error of the parallel processing pitch detector in terms of p. [Hint: Consider the result of each pitch detector a Bernoulli trial with probability $(1 - p)$ of making an error and probability p of no error.]	Analyze	The learners recall the pitch period and understand the autocorrelation method for period estimation to apply windowing function on sampled speech signal and analyze the filtered output	CO 5
8	A proposed pitch detector consists of a bank of digital bandpass filters with lower cutoff frequencies given as $F_k = 2^{k-1}F_1$, k = 1, 2, . . . , M, and upper cutoff frequencies given as $F_{k+1} = 2^k F_1$, k = 1, 2, . . . , M. Show that if input is periodic with fundamental frequency F_0 such that $F_k < F_0 < F_{k+1}$,then the filter outputs of bands 1 to k-1 will have little energy,the output of band k will contain the fundamental frequency, and bands k + 1 to M will contain one or more harmonics. .	Apply	The learners recall filter banks to understand the windowing of sampled speech signal and apply short time autocorrelation to identify the fundamental frequnecies in the filtered output	CO 5

9	Assume there are three independent pitch detectors, each having a probability p of correctly estimating pitch period, and probability $1-p$ of incorrectly estimating pitch period. The decision logic is to combine the seven pitch estimates in such a way that an overall error is made only if four or more of the individual pitch detectors make an error.Derive an explicit expression for the probability of error of the parallel processing pitch detector in terms of p. [Hint: Consider the result of each pitch detector a Bernoulli trial with probability $(1-p)$ of making an error and probability p of no error.] (b) For what value of p is the overall error probability less than 0.05?	Analyze	The learners recall the pitch period and understand the autocorrelation method for period estimation to apply windowing function on sampled speech signal and analyze the filtered output probability of error	CO 5
10	A proposed pitch detector consists of a bank of digital bandpass filters with lower cutoff frequencies given as $F_k = 2^{k-1}F_1$, k = 1, 2,3..M and upper cutoff frequencies given as $F_{k+1} = 2^k F_1$, k = 1, 2,...M. If input is periodic with fundamental frequency F_0 such that $F_k < F_0 < F_{k+1}$,then (a)Determine F1 and M such that this method would work for pitch frequencies from 50 to 800 Hz	Analyze	The learners recall filter banks to understand the windowing of sampled speech signal and apply short time autocorrelation to analyze the possible fundamental pitch frequencies detected using the pitch detector	CO 5

PART B-LONG ANSWER QUESTIONS				
1	State and explain (i) Short-time energy (ii) Short-time average magnitude and (iii) short-time zero crossing rate. How do you distinguish voiced and unvoiced segments in speech waveform using these parameters	Analyze	The learners recall the fourier transforms to understand the autocorellation function and apply it on sampled speech signal to analyze the frequnecy response and calculate speech parameters	CO 5
2	Give an overview of pitch estimation by FFT analysis of speech signal	Apply	The learners recall the FFT analysis to understand the sampled speech signal to identify the pitch period	CO 5
3	With the help of neat block diagram, explain the working of clipping auto correlator. What are the advantages of using 3 level clipper.	Apply	The learners recall the autocorellation and understand the working of clipping auto correlator to identify the effect of order of clipper and state its advantages	CO 5
4	Explain how short-time energy and short-time magnitude can be used to distinguish voice, unvoiced and silence regions in speech signal	Understand	The learners recall speech signal waveforma and understand short time energy and short time magnitude methods to differentiate between voiced, unvoiced and silenced parts of speech	CO 5
5	Examine how speech vs silence discrimination using short time energy and zero crossing with neat schematic of signal flow	Analyze	The learners recall the sampling to understand sampling of speech signal and apply short time energy and zero crossing to analyze the speech and silence	CO 5
6	Demostrate the pitch period estimation using parallel processing approach	Understand	The learners recall the pitch period and understand parallel processing approach to calculate pitch period	CO 5
7	Outline linear filtering interpretation of short-time spectrum analysis with suitable block diagram	Understand	The learners recall the linear filtering and understand the interpretation of linear filtering of short-time spectrum.	CO 5

8	With a neat diagram, explain non-linear smoother for estimation of parameters in speech processing. Justify the need for delays in non-linear smoother.	Understand	The learners recal the non-linear smoother and understand the role of delay block in non-linear smoother for estimating the speech processing parameters	CO 5
9	Define (1) short time energy (2) zero-crossing rate (3) average magnitude function (4) average magnitude difference function in the speech processing and list out their metrits and demerits	Analyze	The learners recall sampling of speech waveforms to understand pitch period and apply short time energy, zero crossing rate, average magnitude function to analyze the merits and demerits in speech processing	CO 5
10	Explain how to distinguish voiced and silence, using energy and zero crossings.	Understand	The learners recal voiced and silence parts and understand the energy and zero crossing analysing methods to distinguish between them	CO 5
11	Explain the linear filter operation of a short-time spectrum analysis with related equations and· block diagram. Also discuss about magnitude of the short time spectrum using both low pass filters and band pass filter	Understand	The learners recal the short time spectrum and understand the linear filter operation on speech waveform using low pass and band pass filters	CO 5
12	From the basic equation for auto-correlation function of a discrete-time deterministic signal ϕ(k),derive the equation for short-time auto-correlation function R_n(K). Draw the related block diagram so as to obtain R_n(K) from the sequence X(n).	Analyze	The learners recall the sampling and understand sampling of speech signal and apply autocorrelation to analyze the coefficients of short time autocorrelation for speech waveform	CO 5
13	What is AMDF? How will you use AMDF for pitch measurement? What is the difference in the computation of pitch using AMDF and autocorrelation method? Which is computationally efficient and why?	Analyze	The learners recall the pitch period and understand average magnitude difference function working (AMDF) to apply for calculating pitch period and analyze the computation efficiency of AMDF and autocorrelation methods	CO 5

14	Explain the different functions used in time domain analysis of speech signals.	Understand	The learners recall FFT and DFT and understand the time domain analysis of speech signal	CO 5
15	Illustrate the different techniques of pitch estimation	Understand	The learners recal pitch estimation and understand the different techniques for deriving the pitch period from speech waveform	CO 5
16	What is the role of window function in speech signal processing? Why a window size of 20-30 is normally preferred for short-time speech signal processing	Apply	The learners recall sampling of speech to understand short time speech signal and apply windowing function on speech signal to identify the speech segment	CO 5
17	Give the equation for estimation of short-time autocorrelation function of speech. How pitch can be estimated from ACF?	Apply	The learners recal the pitch period to understand the short time autocorrelation function and apply it for calculating pitch period	CO 5
18	What is the minimum sample rate needed for speech signal? How bit rate can be reduced using DPCM	Apply	The learners recall the sampling to understand qunatization and apply digital modulation on the quantized speech signal	CO 5
19	Why pitch tracking is needed?What happens to pitch during unvoiced speech segments?	Apply	The learners recall the pitch period to understand the need for pitch period calculation in speech processing and apply it on speech waveform to identify unvoiced speech segments	CO 5
20	Illustrate the short time average zero crossing rate with neat diagram	Understand	The learners recall the short time autocorrelation and understand the average zero crossing rate for calculating pitch perid	CO 5

PART C-SHORT ANSWERS QUESTIONS				
1	Why do we consider short time representation of speech signals? What do you mean by windowing? Explain the concept of short-time speech processing with suitable general block diagram	Apply	The learners recal the sampling of speech signal and understand short-time autocorrelation and apply windowing function to calculated pitch period	CO 5
2	Explain pitch period estimation using short-time correlation	Understand	The learners recal the short-time correlation and understand this method for calculating pitch period estimation	CO 5
3	Discuss with related equations (a) Short time energy (b) Short time zero crossing rate	Analyze	The learners recal the DFT to understand the autocorrelation and apply on speech signal to identify the pitch period to analyze speech signal waveform	CO 5
4	Examine short time autocorrelation functions with necessary waveforms	Analyze	The learners recal the DFT to understand autocorrelation and apply on sampled speech signal and analyze the short time autocorrelation function waveforms	CO 5
5	Illustrate the short time average zero crossing rate with neat diagram	Understand	The learners recall the short time autocorrelation and understand the average zero crossing rate for calculating pitch perid	CO 5
6	Explain DPCM for speech signal	Apply	The learners recal the sampling and understand the speech sampling and apply the DPCM for quantizing the sampled speech signal	CO 5
7	How will you get narrow band spectrogram? What are its merits and demerits	Understand	The learners recall the spectrogram and narrow band spectrogram to understand the merits and demerits of narrow band spectrogram	CO 5
8	Explain a speech analyisis-synthesis system	Understand	The learners recall the DFT and understand the speech analysis and synthesis sytem by block diagram	CO 5

9	What are formants? What is the reason for using a long window in time domain for accurate formant estimation	Apply	The learners recal the formants for speech waveforms and understand the windowing operation and apply it on speech waveform for formant estimation	CO 5
10	What is sampling? What is quantization? How signal to quantization noise vary with number of bits?How bit rate can be reduced using ADPCM	Analyze	The learners recall sampling and understand quantization to apply on the sampled speech signal to analyze the effect of digital modulation schemes like ADPCM on number of bits and bit rate	CO 5
11	Explain the method to estimating pitch and formants from the short-time magnitude spectrum of vowel segment	Understand	The learners recal pitch and formants and understand the short time magnitude spectrum of vowel segment	CO 5
12	How to suggest a suitable window length in time domain for accurate time localization of events in a speech signal	Analyze	The learners recalll the autocorrelation to understand short time autocorrelation and apply windowing function to analyze the time localizatioon of speech waveforms	CO 5
13	Explain the changes in short-time spectral envelope for a vowel consonant, vowel syllable/apa/	Apply	The learners recall consonants of english language to understand the vowels waveforms and apply short time spectrum for vowels	CO 5
14	How can you estimate pitch from short time magnitude spectrum of a voiced speech segment	Apply	The learners recal the pitch period and understand short time magnitude function and apply on speech waveform to identify the pitch period of speech	CO 5
15	Analyze the design of filter banks in SpeechCoding?	Analyze	The learners recal the autocorrelation to understand the short time autocorrelation and apply the overlap addition method to analyze the filter bank	CO 5

16	Show that STFT is invertible	Understand	The learners recall the short time fourier analysis and understand its properties to show that STFT is invertible.	CO 5
17	Explain how short time enrgy and short time magnitude can be used to distinguish voiced,unvoiced and silence regions of a speech signal	Apply	The learners recall speech waveform to understand voiced, unvoiced and silence parts and apply short time energy and short time magnitude to differentiate between the speech waveforms parts	CO 5
18	Distinguish between wide-band spectrogram and narrow band spectrogram	Analyze	The learners recal the speech waveform parameters to understand spectrogram and apply short time autocorrelation to analyze wide-band and narrow band spectrograms	CO 5
19	Discuss the approximate signal processing model suggested for human auditory perception	Understand	The learner recall the signal processing concepts to understand the signal processing model for human auditory perception	CO 5
20	Define spectrogram. How will you extract the formant structure from spectrogram	Apply	The learners recall the formant of english language to understand the spectrogram and apply short time autocorrelation to identify the formant from spectrogram	CO 5

MODULE V				
Short time fourier analysis and Linear predictive coding				
PART A-PROBLEM SOLVING AND CRITICAL THINKING QUESTIONS				
1	A speech signal is sampled at a rate of 10,000 samples/sec (i.e., Fs = 10,000). A Hamming window of length L samples is used to compute the STFT of the speech signal. The STFT is sampled in time with period R, and in frequency at N = 1024 frequencies. (a) It can be shown that the main lobe of the Hamming window has a symmetric full width of approximately $8\pi/L$. How should L be chosen if we want the full width of the main lobe to correspond to approximately 200 Hz analog frequency? (b)What is the spacing (in Hz) between sample points in the frequency dimension?	Apply	The learners recall the windowing function to understand the filtered speech signal and apply short time fourier transforms to identify the length of filter order	CO 6
2	A filter bank with N filters has the following specifications: 1. the center frequencies of the bands are ω_k; 2. the bands are symmetric around $\omega = \pi$, i.e., $\omega_k = 2\pi - \omega_{N-k}$, $P_k = P^*_{N-K}$, ω_k[n] = ω_{N-k}[n]; 3. a channel exists for ω_k=0. For both N even and N odd: (a) sketch the locations of the N filter bands; (b) derive an expression for the composite impulse response of the filter bank in terms of ω_k[n], ω_k, P_k, and N.	Apply	The learners recall the fourier transform to understand working of filter bank and apply short time fourier transforms to identify the frequency band of sampled speech signal	CO 6

3	A speech signal, s[n], is predicted using the predictor: $$\tilde{s}[n] = \sum_{k=k_0}^{k_1} \alpha_k s[n-k], \quad 1 \le k_0 < k_1 \le p,$$ where p is the usual order of linear predictive analysis for speech. Determine the optimum predictor that minimizes the mean-squared prediction error.	Analyze	The learners recall linear predictive polynomial and understand the solution methods like Cholesky Decomposition,Levinson–Durbin Algorithm and apply short time fourier to analyze the prediction error	CO 6
4	A sampled speech signal is predicted using the predictor: $$\tilde{s}[n] = \beta s[n - n_0], \quad 2 \le n_0 \le p.$$ where p is the normal predictor order for speech. Find the optimum values of β and E_n that minimize the mean-squared prediction error using both the autocorrelation and covariance formulations	Apply	The learners recall the prediction error to understand the linear predictive coding analysis and apply the short time autocorrelator and covariance function to minimize the mean squared prediction error	CO 6
5	Determine the second-order linear prediction inverse filter, A(z), for which the two LSF are 666.67 Hz and 2000 Hz, when Fs = 8000 samples/sec. (Note: You may want to use the relationship $cos(\pi/6) = \sqrt{3/2}$ in your solution.)	Apply	The learners recall the filters to understand the linear prediction inver filter and apply the linear predictive coding analysis to identify the second order linear filter frequency response	CO 6

6	A causal LTI system has system function: $$H(z) = \frac{1 - 4z^{-1}}{1 - 0.25z^{-1} - 0.75z^{-2} - 0.875z^{-3}}.$$ (a) Use the Levinson-Durbin recursion to determine whether or not the system is stable. (b) Is the system minimum phase?	Analyze	The learners recall the causal LTI system to understand the linear predictive coding analysis and apply Levison-Durbin solution method on predictive polynomial to calculate phase of the system	CO 6
7	A speech signal was sampled with a sampling rate of Fs = 8000 samples/sec. A 300-sample segment was selected from a vowel sound and multiplied by a Hamming window. From this signal a set of linear prediction error filters $$A^{(i)}(z) = 1 - \sum_{k=1}^{i} \alpha_k^{(i)} z^{-k},$$ with orders ranging from i = 1 to i = 11 was computed using the autocorrelation method (a) Determine the z-transform $A^{(4)}(z)$ of the 4th-order prediction error filter. Draw and label the flow graph of the direct form implementation of this system. (b) Determine the set of k-parameters k1, k2, k3, k4 for the 4th-order prediction error lattice filter. Draw and label the flow graph of the lattice implementation of this system.	Apply	The learners recall the prediction error signal to understand the prediction error filter design and apply z-transforms to identify the lattice impelementation of the filter	CO 6

8	A speech signal was sampled with a sampling rate of Fs = 8000 samples/sec. A 300-sample segment was selected from a vowel sound and multiplied by a Hamming window. From this signal a set of linear prediction error filters $$A^{(i)}(z) = 1 - \sum_{k=1}^{i} \alpha_k^{(i)} z^{-k},$$ with orders ranging from i = 1 to i = 11 was computed using the autocorrelation method (a) Estimate the first three formant frequencies (in Hz) for this segment of speech. Which of the first three formant resonances has the smallest bandwidth? How is this determined?	Analyze	The learners recall the prediction error filter to understand the windowing function and apply linear prediction error transfer function to analyze the formant frequencies	CO 6

9	A speech signal was sampled with a sampling rate of Fs = 8000 samples/sec. A 300-sample segment was selected from a vowel sound and multiplied by a Hamming window. From this signal a set of linear prediction error filters $$A^{(i)}(z) = 1 - \sum_{k=1}^{i} \alpha_k^{(i)} z^{-k},$$ with orders ranging from i = 1 to i = 11 was computed using the autocorrelation method (a) The minimum mean-squared prediction error for the 2nd-order predictor is E(2) = 0.5803. What is the minimum mean-squared prediction error E(4) for the 4th-order predictor? What is the total energy R[0] of the windowed signal s[n]? What is the value of the autocorrelation function R[1]?	Apply	The learners recall short time fourier analysis to understand the linear predictive polynomial and apply autocorrleaton function to determine the prediction error	CO 6
10	Autocorrelation segement corresponding to a speech segment is equal to R(0)=1,R(1)=0.9, R(2)=0.84 and R(3) = 0.78. Compute the linear predictive coefficient segments for speech segments for filter orders p=2, and p=3 using recursie method.	Apply	The learners recall the fourier transforma to understand the short time autocorrelation function and apply it on linear predictor polynomial to calculate the LPC coefficients	CO 6
PART B-LONG ANSWER QUESTIONS				
1	Summarize the Linear predictive coding analysis and its uses in speech signal processing applications.	Understand	The learners recall the linear predictive coding and understand their application in speech production,systhesis and recognition	CO 6

2	Outline the short-time fourier analysis for speech processing	Understand	The learners recall the fourier transform and understand short time fourier concepts to apply on sampled speech signal	CO 6
3	Interpret short time fourier transform using DTFT for sampled speech waveforms	Analyze	The learners recall fourier transform to understand short time fourier transform for applying on sampled speech signal to analyze effect for windowing	CO 6
4	Distinguish the short time fourier transform interpretation using DFT and DTFT	Analyze	The learners recall the DFT and DTFT to understand short time fourier transform for applying on sampled speech signal to analyze the effect of windowing on speech signal	CO 6
5	Explain the effect of window on resolution of DTFT speech signal	Understand	The learners recall the DTFT to understand effect of windowing on sampled signal	CO 6
6	Demostrate the relation between short time autocorrelation function and short time fourier transform	Understand	The learners recall the autocorrelation function and understand the relation between short time fourier transform and autocorrelation functions	CO 6
7	Inspect the linear filtering intepretation of short time fourier transform	Analyze	The learners recall fourier transforms to understand the short time fourier transform and apply linear filtering to analyze the linear filtering intepretation of short time fourier transform	CO 6
8	Interpret openlap addition method for short time fourier synthesis of speech	Understand	The learners recall the openlap addition method and understand the interpretation of short time fourier synthesis using openlap addition method	CO 6

9	Examine filter bank summation method for short time fourier synthesis	Analyze	The learners recal sampling of speech signal to understand short time fourier analysis and apply filter bank summation for analyzing sampling of speech signal in frequency dimension	CO 6
10	Outline the basic principles of linear predictive analysis with equations and LPC model systems	Understand	The learners recal the LPC model and understand the principles involed in developing LPC model with equations	CO 6
11	Compare the autocorrelation and covaraince methods for linear predictive analysis	Analyze	The learners recal the LPC model to understand speech segment and apply autocorrelation and covaraince methods for analyzing the limits of speech segment	CO 6
12	Discuss the Cholesky Decomposition Solution for LPC Analysis	Understand	The learners recall the LPC analysis for speech processing and understand the cholesky decomposition method for obtaining predictor coefficients	CO 6
13	Express Levinson–Durbin algorithm for calculating LPC preditor coefficients	Apply	The learners recal linear predictive analysis to understand LPC predictor equation and apply Levisonn-Durbin algorithm to calculate the LPC predictor coefficients	CO 6
14	Summarize the properties of LPC polynomial	Understand	The learners recal the LPC polynomial and understand the polynomial properties	CO 6
15	With linear predictive analysis, explain the lossless tube model	Apply	The learners try to recall the LPC analysis and apply it the lossless tube to derive the transfer function	CO 6
16	List out the methods for linear predictive analysis and explain them with required equations	Apply	The leaners will try to recall the methods for linear predictive anaysis and try to implement them to calculate linear predicitive coefficients	CO 6

17	Define prediction error signal. Calculate the normalized mean squared error	Apply	The learners recal the LPC polynomial to understand the prediction error and apply to calculate to the normalized mean squared error for autocorrelation method, covariance methods	CO 6
18	Express the mean squared prediction error in frequency domain using DTFT	Understand	The learners recal the DTFT and understand the mean squared prediction error interpretations in frequency domain	CO 6
19	Examine the effect of model order p on autocorrelation	Analyze	The learners recal the autocorrelation to understand the short time fourier analysis and apply linear predicitve analysis for analyzing the windowing filter order p	CO 6
20	Differentiate between the LPC solution methods for calculating the predictor polynomial coefficients	Analyze	The leaners recall the methods for linear predictive anaysis and try to implement them to calculate linear predicitive coefficients and understand the predictor polynomial and analyze the sampled speech signal	CO 6
PART C-SHORT ANSWER QUESTIONS				
1	Explain basic principles of linear predictive analysis.	Understand	The learners will try to recall the linear predictive analysis	CO 6
2	Explain an LPC based synthesizer using a block schematic. What is the assumption used in the synthesis?	Understand	The learners recall linera predictive analysis to understand working of frequnecy domain coder for speech and audio	CO 6
3	List out the differenece between covariance LPC and autocorrelation LPC and state their their relative advantages	Apply	The learners recal LPC covariance and autocorrelation to understand linear predictive analysis and apply it illustrate their advantages and disadvantages	CO 6
4	Explain the covariance method for LPC analysis	Understand	The learners recall the LPC analysis and understand the covaraince method for predictor polynomial analysis	CO 6

5	State and explain applications of LPC to speech processing	Understand	The learners recal linear predictive analysis and and understand the predictor polynomial for speech processing	CO 6
6	Write short note on applications of LPC parameters	Apply	The learners will try to recall the linear predictive coding (LPC) and understand the LPC parameters to apply in speech processing	CO 6
7	Gie an overview on channel vocoders design using digital filters and discrete fourier transforms	Undestand	The learners recall the DFT and understand digtal filters for desiging channel vocoders	CO 6
8	Explain linear predictive vocoder	Understand	The learners will try to recal the vocoder and explain it by linear predictive analysis	CO 6
9	What is spectrogram? What are their types?Explain its significance and applications in speech processing	Understand	The learnes recall the speech waveforms and understand how speech waveform is represented using spectrogram visually	CO 6
10	Derive the p^{th} order linear predictor system function	Apply	The learners recal z-transform to understand the linear predictor system and apply to derive the LPC polynomial and prediction error	CO 6
11	Develop the spectrogram by linear predictive analysis	Apply	The learners recall the short time fourier analysis to understand the linear predictive analysis of speech signal and apply it to develop spectrogram	CO 6
12	Differentiate spectrum obtained by short time fourier and linear predictive analysis	Analyze	The learners recall the spectrogram to understand the spectrum of speech signal by applying short time fourier analysis and linear predictive analysis and analyze their differences	CO 6

13	Give an overview on selective linear prediction	Understand	The learners recal the linear predictive spectrum and understand the method to obtain spectrum of selective band of frequencies	CO 6
14	Summarize the lattice methods for linear predictive analysis	Understand	The learners recall the linear predicitve analysis and understand the lattice method for calculating the predictor polynomial coefficients	CO 6
15	Differentiate between lattice method, covaraince method and autocorrelation method	Analyze	The learners recall the linear predictive polynomial to understand the linear predictive analysis for speech processing and apply LPC solution analysis methods to analyze the predictor polynomial coefficients	CO 6
16	Compare the computational requirements of the linear predictive analysis solution methods	Analyze	The learners recall the linear predictive analysis to understand the predictor polynomial and apply the solution methods to calculate predictor coefficients and analyze their computational requirements	CO 6
17	Give an overview of normalized mean squared error	Understand	The learners recall the linear predictive analysis and understand the prediction error and normalized mean squared error	CO 6
18	Elaborate the filter bank systhesis using FFT	Apply	The learners recall FFT to understand filter bank systhesis by applying FIR and IIR filters	CO 6
19	Explain the relation between overlap addition method of systhesis and filter bank synthesis	Understand	The learners recall overlap addition method and filter bank synthesis methods and understand the relation between them	CO 6

20	Discuss the effects of fixed and time-varying modifications of the short-time spectrum on the reconstructing the original signal	Analyze	The learners recall the short time fourier analysis to understan the short time spectrum and analyze the resultant speech signal after applying sampling both on time and frequency	CO 6

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HOD CSE(AI & ML)