Roll No											

PRESIDENCY UNIVERSITY
BENGALURU

SCHOOL OF ENGINEERING
END TERM EXAMINATION - MAY/JUNE 2024

Semester : Semester VIII - 2020
Course Code : ECE3028
Course Name : Speech Signal Processing
Program : B.Tech. Electronics and Communication Engineering

Date : Jun 3, 2024
Time : 01:00 PM - 04:00 PM
Max Marks : 100
Weightage : 50%

Instructions:
(i) Read all questions carefully and answer accordingly.
(ii) Question paper consists of 3 parts.
(iii) Scientific and non-programmable calculator are permitted.
(iv) Do not write any information on the question paper other than Roll Number.

PART A

ANSWER ANY THREE QUESTIONS **(3 Q X 5 M = 15 M)**

1. Formant synthesis models the vocal tract as a digital filter with resonators and anti-resonators. Describe briefly about auditory mechanism used by human brain to perceive speech or sound.

(CO1) [Knowledge]

2. Human speech production is complex, involving cognitive planning, muscular actions, and sound generation. Give a brief about applications of speech signal processing with appropriate block diagram.

(CO1) [Knowledge]

3. Noise is mixed with the speech during the generation of the speech or may be in transmitting the speech over a noisy channel or at the receiver end. Describe briefly about how speech and silence is discriminated using short time energy and zero crossings.

(CO2) [Knowledge]

4. The time-varying spectral characteristics of the speech signal can be displayed two-dimensionally, which is known as **spectrogram**. How signals are represented in Fourier domain and explain the significance of Fourier domain representation.

(CO3) [Knowledge]

5. Filters are used to pass the desired frequency signals and rejects undesired frequency signal. Determine the complex cepstrum of the decaying exponential sequence, $a^n u(n),\ |a| < 1$.

(CO4) [Knowledge]

PART B

ANSWER ANY TWO QUESTIONS **(2 Q X 20 M = 40 M)**

6. The overlap-add (OLA) methodis an example for Fourier transform implementation of the STFT. Why short time analysis is essential in speech signal processing? Explain it with suitable diagrams.

(CO2) [Comprehension]

7. The most typical format for storing sound signals is the wav-file format. Explain short time Fourier analysis using DFT interpretation.

(CO3) [Comprehension]

8. The affine projection algorithm (APA) is one of the most widely used algorithms for speech processing application due to its improved performance in terms of low steady-state error and fast convergence speed. Describe homomorphic systems for convolution.

(CO4) [Comprehension]

PART C

ANSWER ANY THREE QUESTIONS **(3 Q X 15 M = 45 M)**

9. Speech Coding is the process of transforming a speech signal into a representation for efficient transmission and storage of speech. Write about the speech production mechanism for voiced, unvoiced sounds and explain its classification.

(CO1) [Application]

10. The acoustic waveform impinges on the ear (the basilar membrane) and is spectrally analyzed by an equivalent filter bank of the ear. Explain the short time energy and magnitude in speech processing with necessary equations.

(CO2) [Application]

11. The STFT is a mathematical and computational tool for representing the speech signal. Explain filter bank summation method for short time analysis with suitable diagrams.

(CO3) [Application]

12. “Quefrency” is the independent variable of the cepstrum. Explain cepstrum distance measures.

(CO4) [Application]