PART-1-

** Introduction - (DSP) 3

- Classification

- DTS / CTS

** Aliasing effect. - 1

** Linear or non linear

** Sampling 2

** Sinusodial signal

- LTI system

* ** - Correlation.

- AS → DS / DS → AS (Recover)

PART-B

(I) Z - transformation / inverse Z- transformation

(II) filter -

(III) DFT.

(IV) FFT / convolution / window.

PART- A (Details)

(I) Define - Signal, DSP with example.

- why DSP needed.

- Block Diagram of DSP and introduce the difference components of the system.

- introduce different parts of DSP.

- advantage and disadvantage of DSP.

(II) Signal - classification of signals.

- discrite time signal

- Continouse time signal.

- what in noice. & signal to noice (SNR)

- what is mean by SNR = 0 dB. (zero)

- What is Nayquist rate + math.

*** - Find discrete time signal after sampling.

DTS

(III) ~~DTS~~ — Unit Ramp sequence & exponential sequence of DTS.

* — Show that any signal can be decomposed into an even and odd component.

* — Find D.Time. signal.

— reconstruct analog signal by ideal interpolation.

- analysis of discrete time → linear time invariant.

- Discrete time system described by different equation.

- implement of DTS.

— Correlation & convolution of DTS.

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(IV) Sampling — Define sampling + classification.

— down sampling.

— Determine sampling rate.

— find minimum sampling frequency by avoiding Aliasing effect.

— Sampling processes of analog signal.

→ find discrete time signal.

→ why need to change the sampling rate of a signal.

* Discrete time system.

(V) Discrete time sinusoidal signal —

— Define (D.T.S.S) —

— write express of (DTSS) & find Highest rate of oscillation.

** — Show DTSS is periodic if its frequency is a rational number.

** — Prove that highest rate of a oscillation in a ~~discrete~~ DTSS is attended when ($\omega = \pi$ or $\omega = -\pi$)

VI Aliasing effect:

- Define Aliasing effect.
- Show the effect with example.
- Why do we need antialising filter in DSP.

- What is Aliased frequency and when it occurs.

- find Aliased frequency.

- Determine whether the system is linear or non-linear.

- Determine X(w) of a signal.

- Determine if the following system are time invarient or time varint.

- Define causual and non-causul System with example.

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** - Determine power and energy of the unit step sequence.

** - Describe A → D signal conversion process
D → A " " (Recover) process.

* Quantization -

— Define quantization.

- What is quantization process.

- Define quantization ~~step~~ step size and quantization ~~step~~ error with example.

** - if number of quantization levels L = ____, then how many bits per sample will be required.

TOBI DFT

* LTI system.

— Define LTI system.

— Discuss the response of LTI system to arbitrary inputs.

— impulse response of LTI system.

** Correlation

— Define correlation.

— Define (normalized auto-correlation & cross correlation).

— Describe cross-correlation function.

→ Application of cross-correlation.

** — Determine cross-correlation.

— Determine cross-correlation.

— Define correlation between two signal: write down its significiant.

PART-B

Z-Transformation

- introduction of Z-transformation /inverse.
  - Defination.
  - Z-transformation and ROC of infinity duration sequence.
  - Properties of Z-transformation.
  - " " inverse - Z- transformation.
  - one sided - Z-transformation.
  - why Z-transformation.
    - → Z-transformation Vs furier transformation.
    - → Determine (Z /inverse Z) transformation.

ROC –

- Define ROC.
- Properties of ROC.
- find ROC.
- importance of ROC.
- Scaling property.
- transform to recover sequence.
- which approach for invert Z-transf –

Set-2

Discrete furier transformation

- Define furier series.
- importance of furier transformation.
- Dft leakage problem & reason. & explain problem with example.
- Define furier serice of CTS.
- furier integral.
- Properties of Contious time (F.T).
- Derive/define DFT euation.
- DFT Resculation.
- DFT classification.
- Descrive IDFT.
- Properties of DFT.
- why it is called DFT pain.
- $\boxed{\text{find DFT}}$
- DFT vs FFT.

- find DFT by DIF Algorithm

- find IDFT by DIF Algorithm.

- Define Decimation in-time algorithm.

** - Draw flow graph of a two point DFT for a Decimation - in - time decomposition.

** - Relationship between the magnitude of a signal before and after DFT with example.

** - why do we need to convert a time domain signal into frequency domain.

- Perform and Calculate DFT on wave. where → (sin value. 10 HZ sampling 40 HZ). & plot frequency spectrum for this wave.

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FFT

- Represent furier series.

- explain Harmonic furier series.

- furier and inverse furier integral.

** → need of furier transformation.

→ Convolution properties of furier transformation.

→ Determine furier transformation.

→ Overlap add and over-lap save methods uses.

Find four point of sequence $x(n) = U(n)$

Draw amplitude, magnitude, phase and power spectrum

- Define zero stuffing technique & its effect in signal analysis.

- Danielsons lanczos lemma

- N point = 512. How many computation needed for DFT & FFT. also calculate speed factor.

convolution

F.F.T

- Define F.f.T
- why ffT is needed.

*→ DIT for ffT

→ Calculate DFT by (DIF / DTF).

→ Twiddle factor. & its effect.

- use of twiddle factor on DFT.
- Compute DFT By DIF algorithm.

→ Bit reversal techinique.

**→ Draw structure of FFT implementation for a 4 Bit DFT.

- Radix-2 FFT algorithm.
- 4 input Butterfly Diagram for FFT.
- How we get benifit from the redundacy and symmetry of the (twiddle factor)?

window

- what is window function. & application.
- frequency domain characterestic of Rec window and Hamming window.
- Zero padding techinque.
- ✓ stable & unstable system.
- ✓ window process. Hamming and rectangular window on DFT output.

- DTF resolution.

- effect of Zero stuffing fancur on DFT

** - use of window to solve the leakage effect.

- Show the response of Humming and Rectangular window in frequency domain (FD) and describe the finding from the specturms.

Convolution:

- ~~Importance~~ of Convolution in Dsp.
- ~~State~~ and prove Convolution theorem.

** - Compute Convolution of signal.

- Brandwith of signal.
- How can we extract a particular brand of a signal
- Convolation properties of furier transformation.
- Process of computing convolution between the signals and impulse response of the system.
- explain convolution sum of signals.
- stable and unstable system.

*** - find Convolution sum of ______

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Filters:

- Design of FIR and IIR filters.
- what is IIR/FIR filters.
- explain FIR/IIR filter with equation and diagram.
- Define adaptive filter.
- Block Diagram and explain the function of adaptive filter.
- Analog filter and Digital filter.

realize the secound order digital filter $y(n) = \ldots$

- Discuss LMS filters with nessary figure and equation.

→ obtain impulse response $H(z)$ of a digital filter corresponding to an analog filter —

Find sampling rate by using impulse invariance method.

→ find sampling rate by using impulse invariance method.

- ~~FIR V IIR~~
- Application of Digital filter in DSP.
- why analog ~~filter~~ signal are generally low pass filtered before they are convert to digital form.
- Properties of FIR filter.
- How to design digital filter from analog filter.
- linear phase low pass filter.
- Properties of linear low pass filter.
- Determine plot the associated phase.
- why ripping occurs in pass-band and stop band.