GUJARAT TECHNOLOGICAL UNIVERSITY

Digital Signal Processing
Subject Code: 3715203
Semester I

Type of course: Core

Prerequisite:

1. Digital signal processing.

Rationale:

Teaching and Examination Scheme:

Teaching Scheme			Credits	Examination Marks				Total Marks
				Theory Marks		Practical Marks		
L	T	P	C	ESE (E)	PA (M)	PA (V) ESE	PA (I)	
4	2	0	5	70	30	30	20	150

L- Lectures; T- Tutorial/Teacher Guided Student Activity; P- Practical; C- Credit; ESE- End Semester Examination; PA- Progressive Assessment;

Content:

Sr. No.	Content	Total Hrs	% Weightage
1	Signals and Signal processing, Classification of signals; Examples of typical signals; Signal applications; Causality, Stability. Discrete time signals; The sampling process; Characterization of linear time-invariant systems; Convolution and its properties; Difference equations	6	10
2	The Discrete Fourier Transform, The discrete-time Fourier transform; Discrete Fourier transform and its properties; Linear convolution; Fast Fourier transform; z-transform and inverse z-transform.	6	15
3	Linear Time-Invariant Discrete Systems in the Transform Domain, Finite dimensional discrete systems; Transfer function; Simple digital filters; Inverse systems; Linear phase filters; Chebyshev's theorem, Remez algorithm	6	15
4	Digital Processing of Continuous-Time Signals, Sampling of continuous-time signals; Paley-Wiener theorem; Nyquist frequency; Kotelnikov - Shenon's theorem; Analog low pass filter design; Design of analog high pass; Band pass and band shop filters; Analog - to - digital converter; Digital- to-Analog converter.	6	10
5	Digital Filter Structures, Block diagram representation; Basic finite and infinite impulse response (FIR & IIR); Digital filter structures; All pass filters; IIR tapped cascaded lattice structure; FIR cascaded lattice structure; Digital sine-cosine generator. Preliminary considerations; Bilinear transformation method of IIR filter design; Design of low pass; High pass; Band pass; Band	7	20

	shop IIR digital filters; Spectral transformations of IIR digital filters; FIR filter design based on Windowed Fourier series; Design of FIR digital filters with least- mean-square error; Digital IIR filters design; Analog filters - Bessel, Butterworth, Chebyshev, elliptic filters		

Reference Books:

1. Digital Signal Processing A Practical Guide for Engineers and Scientists by Steven Smith, Elsevier publication
2. Digital Signal Processing Fundamentals & Applications by Li Tan, Elsevier publication

Course Outcome:

- After learning the course the students should be able to:
- Understand about the basics of DSP and different types signal processing.
- Analyse about the discrete Fourier Transform, Linear Convolution and their application in DSP.
- Evaluate the finite dimensional discrete systems and chebyshev's algorithm.
- Understand about the sampling and Nyquist criteria for sampling along with the band pass and band stop filters.
- Understand about the different FIR and IIR filters in details with their applications.

List of Experiments: (with Open Ended Problems)

1. Design and Implement a Digital Butterworth Low Pass System with the following specifications:
   - Passband Frequency: 1000Hz
   - Passband Attenuation: 1db
   - Stopband Frequency: 12000Hz
   - Stopband Attenuation: 80db
   - Sampling Rate: 16000Hz
     a. Obtain and Plot the impulse response of the given digital low pass filter.
     b. Obtain and Plot the Magnitude and Phase responses of the given digital low pass filter.
     c. Verify the cut-off frequency of the filter from its Magnitude Response.
     d. Implement the Difference Equation by passing the samples of a sinusoidal signal of amplitude 0.9, frequency 1000Hz and number of cycles equal to 1000.
     e. Verify the output signal amplitude with the theoretical and expected amplitude.

2. Design and Implement a Digital Chebyshev Low Pass System with the following specifications:
   - Passband Frequency: 1000Hz
   - Passband Attenuation: 1db
   - Stopband Frequency: 12000Hz
   - Stopband Attenuation: 80db
   - Sampling Rate: 16000Hz
     a. Obtain and Plot the impulse response of the given digital low pass filter.

b. Obtain and Plot the Magnitude and Phase responses of the given digital low pass filter.
c. Verify the cut-off frequency of the filter from its Magnitude Response.
d. Implement the Difference Equation by passing the samples of a sinusoidal signal of amplitude 0.9, frequency 1000Hz and number of cycles equal to 1000.
e. Verify the output signal amplitude with the theoretical and expected amplitude.

3. Design and Implement an FIR system of order 127 and using Hamming Window samples and cut-off frequency 2000Hz and sampling rate fs = 32000.
    a. Obtain and Plot the impulse response of the given digital low pass filter.
    b. Obtain and Plot the Magnitude and Phase responses of the given digital low pass filter.
    c. Verify the cut-off frequency of the filter from its Magnitude Response.
    d. Implement the Difference Equation by passing the samples of a sinusoidal signal of amplitude 0.9, frequency 1000Hz and number of cycles equal to 1000.
    e. Verify the output signal amplitude with the theoretical and expected amplitude.

4. Calculate and plot Cn = (2*A*tau/T)*sinc(2*n*f0*tau) with A = 10, T = 1 milli sec and tau = 0.1 milli sec on n*f0 axis to depict 5 zero crossing points on both the negative and positive frequency axes.
5. Repeat the above with T = 10000 milli sec and compare the following obtained in both the cases:
    - First Zero Crossing Frequency
    - Total Average Power, Partial and Percentage average powers contributed by the complex exponentials distributed between positive and negative first zero crossing frequencies.
    - Inter Sample Distance in the frequency domain.

6. Implement DFT and IDFT functions using the related mathematical relations as the basis.
    a. Find the DFT samples of X(k) of a rectangular pulse of amplitude 10 and having its pulse width equal to 2 msecs.
    b. Plot the magnitude and phase parts of X(k) along the frequency axis ‘f’.
    c. Find the average power of the above signal using time domain and frequency domain signal samples
    d. Find the DFT samples of X(k) of an exponentially decaying pulse given by alpha*exp(-alpha*t)*u(t).
    e. Plot the magnitude and phase parts of X(k) along the frequency axis ‘f’.
    f. Find the average power of the above signal using time domain and frequency domain signal samples

g. Find the DFT samples of X(k) of a triangular pulse of maximum amplitude 100 and pulse width 4 msec and centered around origin.

h. Plot the magnitude and phase parts of X(k) along the frequency axis 'f'.

i. Find the average power of the above signal using time domain and frequency domain signal samples

7. Calculate and plot Cn = $(\alpha * f_0)/(\alpha + j*2*\pi*n*f_0)$ with α = 10000, f0 = 100Hz on n*f0 axis to depict 100 frequency components on both the negative and positive frequency axes.
   a. Identify the first Half Power frequency.
   b. Calculate and display the Total Average Power of a of the signal $x_p(t)$.
   c. Calculate the partial amount of average power contributed by the complex exponentials between the Half-Power Frequency points on the frequency axis.
   d. Find the percentage of average power contributed by the same complex exponentials.
      a. Repeat the above exercise with α= 10000.

8. Find and plot the impulse response of the discrete time system equivalent to RC system.
   a. Apply DFT/FFT for the obtained impulse response to get H(k) and plot the magnitude and Phase responses of the system across frequency values.
   b. Identify the half-power frequencies from the obtained plot.
   c. Implement the difference equation of the system by passing a sinusoidal signal of frequency f0 and amplitude 10.

9. Implement Goertzel Function on MATLAB and using the function implement a DTMF Signal Detection System on MATLAB.

10. Assuming an ideal and symmetrical frequency response of a low pass filter with the following specifications given by
   - $H(w) = 1$; for $0 \leq w \leq \pi/2$ and
   - $H(w) = 0$; for $\pi/2 < w \leq \pi$
      a. Design a FIR system with the order given by 128 (Even number) using Frequency Sampling Method.

11. Calculate and plot Cn=(2*A*tau/T)*sinc(2*n*f0*tau) with A = 10, f0 = 1000Hz and tau = 0.1* T on n*f0 axis to depict 5 zero crossing points on both the negative and positive frequency axes.
    a. Identify the first zero crossing frequency.
    b. Calculate and display the Total Average Power of a Rectangular Pulse Train.
    c. Calculate the partial amount of average power contributed by the complex exponentials between the first zero crossing points on the frequency axis.
    d. Find the percentage of average power contributed by the same complex exponentials.

Repeat the above exercise with tau = 0.05*T.

12. Given three sinusoids with the following amplitudes and phases

$$x_1(t) = 5\cos(2 \cdot \pi \cdot 500 \cdot t)$$
$$x_2(t) = 5\cos(2 \cdot \pi \cdot 1200 \cdot t + 0.25 \cdot \pi)$$
$$x_3(t) = 5\cos(2 \cdot \pi \cdot 1800 \cdot t + 0.5 \cdot \pi)$$

    a. Create a MATLAB program to sample each sinusoid and generate a sum of three sinusoids, that is x(n) = x1(n)+x2(n)+x3(n) using a sampling rate of 8000Hz and plot the sum x(n) over a range of time that will exhibit approximately 0.1 second.
    b. Use the user defined dft() function and MATLAB built-in function fft() to compute the DFT coefficients and plot and examine the examine the spectrum of the signal.

13. Using the user defined dft() function obtain the spectral coefficients of
    - Rectangular Window
    - Hamming Window
    - Hanning Window
    - Blackman Window
        a. Assume window length of 255 and plot the magnitude and phase spectra of all the windows individually and compare them

14. Using MATLAB, design a fourth-order digital lowpass Chebyshev filter with cut-off frequency of 1.5kHz and a 0.5dB ripple at a sampling rate of 8,000Hz.

- Determine the transfer function and difference equation
- Plot the Magnitude and Phase Response of the filter.

15. Design a second order digital bandstop Butterworth filter with a centre frequency of 1.8kHz and a bandwidth of 200Hz and a passband ripple of 3dB at a sampling rate of 8,000Hz.
    a. Determine the transfer function and difference equation of the system.
    b. Use MATLAB to plot the magnitude and phase frequency responses.

16. Given a DSP system with a sampling rate set up to be 8,000Hz, develop a 800Hz single-tone generator using a digital IIR filter by completing the following steps
    - Determine the digital IIR filter transfer function
    - Determine the DSP equation
    - Plot the tone thus generated for a duration of 0.01 sec

17. Given x(0) = 1, x(1) = 1, x(2) = 0, x(3) = -1, use the Goertzel algorithm to compute the following DFT coefficients and their amplitude spectra
    - X(0)
    - X(1)
    - $|X(0)|^2$
    - $|X(1)|^2$

18. Design a 31-tap lowpass & highpass FIR filters whose cut-off frequency is 2,500Hz using the following window functions. Assume that the sampling rate is 8kHz.
    - Hamming Window
    - Hanning Window
    - Blackman Window

19. Design a 41-tap bandpass FIR filter with the lower and upper cot-off frequencies being 2,500Hz and 3,000Hz respectively, using the following window functions. Assume a sampling rate of 8kHz.
    - Hanning Window
    - Hamming Window
    - Blackman Window
        a. List the FIR filter coefficients and plot the frequency responses for each design.

20. Given the difference equation with the input-output relationship of a certain initially relaxed DSP system (all initial conditions are zero)

y(n)-0.4 y(n-1)+0.29 y(n-2) = x(n) + 0.5 x(n-1)

- Find the impulse response h(n) and frequency response H(w) of the above filter (using MATLAB)
- Find the step response of the system (using MATLAB)

21. Write a MATLAB program to read speech data from a PCM formatted speech signal and for passing it through Pre-Emphasis and De-Emphasis systems.

Also write the output signal from the de-Emphasis System into an output PCM formatted signal.

List of Open Source Software/learning website:

1. **Matlab**