ASSIGNMENT Two

Digital Filter Structures, Filter Design and DSP application in Communications

1. (a)Describe the Frequency sampling structure of FIR filters

   (b)Design a bandstop filter using the frequency sampling method technique in MATLAB. The specifications are

   Lower stopband edge: 0.3π

   Upper stopband edge: 0.6 π

   Lower passband edge: 0.4π

   Upper passband edge: 0.5 π

   A_s=50dB, R_p=0.5dB

   Plot the impulse response and magnitude response (in dB) of the designed filter.

[GROUP 1]

2. (a)Describe the Lattice structure of IIR filter

   (b)Consider a causal IIR system with a system function

$$H(z) = \frac{1 + 2z^{-1} + 3z^{-2} + 2z^{-3}}{1 + 0.9z^{-1} - 0.8z^{-2} + 0.5z^{-3}}$$

   (a)Determine the equivalent Lattice Structure

   (b)Check if it is stable

[GROUP 2]

3. (a)Describe DUAL-TONE MULTIFREQUENCY (DTMF) SIGNALS

   (b) DTMF tone generation and detection:

Implement the DTMF tone generation and detection in MATLAB with the following specifications:

a. Input keys: 1, 2, 3, 4, 5, 6, 7, 8, 9, *, 0, #, A, B, C, D (key frequencies are given in Figure 1).

b. Sampling frequency is 8,000 Hz.

c. Program will respond to each input key with its DTMF tone and display the detected key.

[GROUP 3]

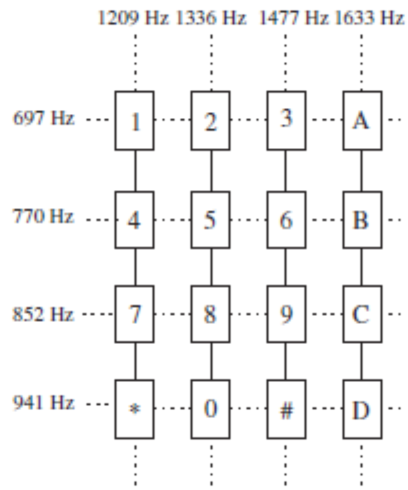

Figure 1: DTMF Key Frequencies

4. An audio playback application is described in Figure 2.

   Due to the interference environment, the audio is corrupted by 15 different periodic interfer- ences. The DSP engineer uses an FIR adaptive filter to remove such interferences as shown in Figure 2.

   a. What is the minimum number of filter coefficients?
   b. Set up the LMS algorithm for the adaptive filter using the number of taps obtained in (a)

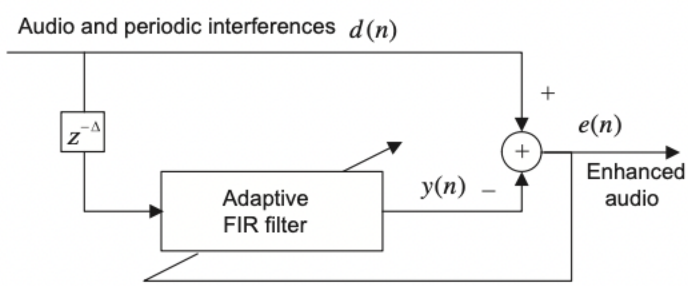

Figure 2: Interference Cancellation

[GROUP 4]

5. Describe and implement using MATLAB the following Applications of the DSP in Communication

   (a)PULSE-CODE MODULATION and DIFFERENTIAL PCM (DPCM)
   (b)DIFFERENTIAL PCM (DPCM) and ADAPTIVE PCM AND DPCM (ADPCM)

[GROUP 5]

6. (a) Design a 7-tap highpass FIR filter whose cutoff frequency is 2,500 Hz using the Hanning window function. Assume that the sampling frequency is 8,000 Hz. List the FIR filter coefficients and plot the frequency responses for each design.

   (b)Design a 5-tap bandpass FIR filter with lower and upper cutoff frequencies of 2,500 Hz and 3,000 Hz, respectively, using the following window Blackman function. Assume a sampling frequency of 8,000 Hz. List the FIR filter coefficients and plot the frequency responses for each design. **[GROUP 6]**

7. (a)Describe how LINEAR PREDICTIVE CODING (LPC) OF SPEECH is applied in communication

   (b)Design a Butterworth digital lowpass filter to satisfy the specifications:

   passband edge: 0.4π, $Rp = 0.5$ dB

   stopband edge: 0.6π, $As = 50$ dB

   Use the impulse invariance method with $T = 2$. Determine the system function in the rational form, and plot the log-magnitude response in dB. **[GROUP 7]**

8. (a)Describe how DELTA MODULATION (DM) is applied in Digital communication

   (b)Consider the following Laplace transfer function

   $$\boxed{H(s) = \frac{s + 0.1}{(s + 0.1)^2 + 9}}$$

   a. Determine the H(z) using the bilinear transformation method if the sampling rate fs=10Hz.
   b. Plot the magnitude response |H(f)| and phase response φ(f)with respect to H(s) for the frequency range from 0 to f_s/2Hz.

[GROUP 8]

9. (a)Consider an FIR lattice filter with coefficient $K_1 = 0.65, K_2 = -0.34, and\ K_3 = 0.8$.

   (i) Find its impulse response by tracing a unit impulse input through the lattice structure
   (ii) Draw the equivalent direct-form structure
   (b)

   Consider an FIR filter with system function

   $$H(z) = 1 + 2.88z^{-1} + 3.4048z^{-2} + 1.74z^{-3} + 0.4z^{-4}$$

   Sketch the direct form and lattice realizations of the filter and determine in detail the corresponding input–output equations. Is the system minimum phase? **[GROUP 10]**

10. F

Compute the eight-point DFT of the sequence

$$x(n) = \left\{\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0\right\}$$

using the in-place radix-2 decimation-in-time and radix-2 decimation-in-frequency algorithms. Follow exactly the corresponding signal flow graphs and keep track of all the intermediate quantities by putting them on the diagrams.

[GROUP 11]

11. Design a Butterworth digital lowpass filter to satisfy the specifications:

passband edge: 0.4π, $Rp = 0.5$ dB
stopband edge: 0.6π, $As = 50$ dB

Use the bilinear transformation technique and the bilinear function, determine the system function in the rational form, and plot the log-magnitude response in dB.

12. Design a 11-tap lowpass FIR filter whose cutoff frequency is 1,600 Hz using the hanning window function. Assume that the sampling frequency is 8,000 Hz. List the FIR filter coefficients and plot the frequency responses for each design.

13. The normalized lowpass filter with a cutoff frequency of 1rad/sec is given as

$$H(s) = \frac{1}{s+1}$$

Use the Bilinear Transformation to design the corresponding digital IIR lowpass filter with cutoff frequency of 15Hz and a sampling rate of 90Hz. Determine the system function in the rational form, and plot the log-magnitude response in dB.

14. Given a sequence X(n) for $0 \le n \le 4$, whrer $x(0) = 1, x(1) = 2, x(2) = 8, x(3) = 3\ and\ x(4) = 4$,

    (a) Evaluate its DFT X(k) using the decimation-in-frequency FFT method
    (b) Determine the number of complex multiplications
    (c) Evaluate its DFT X(k) using the decimation-in-time FFT method

15. Given the system modeling as an application of adaptive filters as seen in Figure 2 and using a single-weight adaptive filter $y = wx(n)$ to perform the system-modeling task,

    (a) Set the LMS algorithm to implement the adaptive filter assuming that initially $w = 0 \quad and \quad \mu = 0.5$;
    (b) Perform adaptive filtering to obtain $y(0), y(1), y(2) \quad and \quad y(3)$, given

$$d(0) = 1,\ d(1) = 2,\ d(2) = -2,\ d(3) = 2,$$

$$x(0) = 0.5,\ x(1) = 1,\ x(2) = -1,\ x(3) = 1$$

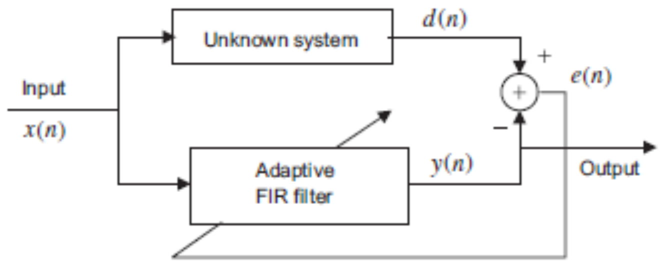

Figure 3: System Modelling