Indian Institute of Engineering Science and Technology, Shibpur
Department of Electronics & Tele-Communication Engineering
Digital Signal Processing Lab

6th Semester

Sheet –4

IIR Filter Design

1) Design an IIR low-pass filter with 3-dB cut-off frequency at $\omega_c = 0.45\pi$ using a single stage realization and a cascade of four first-order low pass filters and compare their gain responses.
2) Write a MATLAB program to design a digital Butterworth low-pass filter using impulse invariance method. Determine the order of the analog prototype filter for this purpose. The input data required for your program are sampling frequency (F_S), pass-band edge frequency (F_p), stop-band edge frequency (F_s), maximum pass-band ripple (δ_p) and minimum stop-band attenuation (δ_s). Plot the gain response of the designed filter for the flowing inputs: $F_S = 80KHz, F_p = 4KHz, F_s = 20KHz, \delta_p = 0.5dB,\ \delta_s = 45dB$. You may use the M-file *impinvar* of MATLAB.
3) Repeat the above problem using bilinear transformation method. You may specifically use the M-file *bilinear* of MATLAB.

Remarks- make a report with a brief description on the programming used and attach the program you have practiced.

Signature of Teacher in charge

Bengal Engineering and Science University, Shibpur
Department of Electronics & Telecommunication Engineering
Digital Signal Processing Lab
6^{h} Semester
Sheet –3

FIR Filter Design

1) Design a low-pass FIR filter of length 21 and 41 respectively with a cutoff frequency of 2 KHz using the following window functions. Assume the sampling frequency is 8 KHz.

   Window function:
   a. Rectangular window function
   b. Hamming window function
   c. Hanning window function
   d. Blackman window function

2) Design 21-length and 41-length band-pass FIR filter with lower and upper cutoff frequency at 2.5 KHz and 3 KHz respectively using the following window functions. Assume the sampling frequency is 8 KHz.

   Window function:
   a. Rectangular window function
   b. Hamming window function
   c. Hanning window function
   d. Blackman window function

3) Use the frequency sampling method to design a linear phase low-pass FIR filter of length 21 and 41 respectively. Let the cutoff frequency be 2 KHz and assume a sampling frequency of 8KHz. List FIR filter coefficients and plot the frequency responses.

4) Use the frequency sampling method to design a linear phase band-pass FIR filter of length 21 and 41 respectively. Let the upper and lower cutoff frequency be 2.5 KHz and 3 KHz respectively and assume a sampling frequency of 8KHz. List FIR filter coefficients and plot the frequency responses.

Bengal Engineering and Science University, Shibpur
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Sheet –2

Discrete time signals and systems in the transform domain

1) Compute an N-point DFT of the following sequences and plot its magnitude and phase spectrum.

$$x[n] = \begin{cases} A & \text{for } n = 0,1,2,\ldots\ldots,M-1 \\ 0 & \text{otherwise} \end{cases}$$

; where $M = 10$ and (i) $N = 10$ (ii) $N = 100$ (iii) $N = 256$.
Plot the magnitude spectrum of DTFT of $x[n]$ and compare the plots for different lengths.

2) Write a program to plot the magnitude and phase response of discrete-time system characterized by its impulse response:

$$h[n] = \begin{cases} 0.5 & \text{for } n = 0 \\ \dfrac{\sin 0.5\pi n}{\pi n} & \text{otherwise} \end{cases}$$

(i) $n = -8:7$
(ii) $n = -16:15$
(iii) $n = -64:63$

3) Evaluate and plot the spectrum of your own voice signal.

4) Write a program to implement linear convolution via DFT-based approach, and compare your results using direct linear convolution.

5) Write a program to realize linear convolution between a small and a long discrete-time sequence using overlap-add method. (Do not use 'fftfilter' function of MATLAB)

Remarks- make a report with a brief description on the programming used and attach the program you have practiced.

Signature of Teacher in charge

Bengal Engineering and Science University, Shibpur
Department of Electronics & Telecommunication Engineering
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Sheet –1

Discrete time signals and systems

1) Write a MATLAB program to generate a sinusoidal sequence $x[n] = A\cos(\omega_0 n + \varphi)$, and plot the sequence using the 'stem' function. The input data specified by the user are the desired length L, amplitude A, the angular frequency ω_0 and the phase φ where $0 < \omega_0 < \pi$ and $0 < \varphi < 2\pi$ with a sampling rate of 20 KHz.

2) A discrete-time system is represented by the following input output relation:
$$y[n] = \frac{1}{M}\sum_{k=0}^{M-1} x[n-k]$$
This is an example of *M-point* moving average filter. Such a system is often used in smoothing random variations in data. Consider for example a signal corrupted by a noise whose minimum and maximum values are -0.5 and 0.5 respectively, i.e.
$$x[n] = s[n] + d[n]$$
Original uncorrupted signal is given by,
$$s[n] = 2[n(0.9)^n]$$
Investigate the effect of signal smoothing by a moving average filter of length 5, 7 and 9. Does the filtered signal improve with an increase in the filter-length? Is there any effect of the filter-length on the delay between the smoothed output and the noisy input?

3) Write a MATLAB program implementing the discrete-time system given by following input output relation,
$$y[n] = 0.5(y[n-1] + \frac{x[n]}{y[n-1]})$$
Show that the output y[n] of this system; for an input $x[n] = \alpha\mu[n]$ with $y[-1] = 1$; converges to $\sqrt{\alpha}$ as $n \rightarrow \infty$ where, α is a positive number.

4) Plot the given input speech file in MATLAB and write a program to implement a quantizer for the given speech file. Plot the variation of signal to quantization noise ratio (SNR_q) against variation of number of *bits/sample*.

Remarks-Make a report with a brief description on the programming used and attach the program you have practiced with result.

Signature of Teacher in charge