CHAPTER - 5

SUMMARY AND CONCLUSIONS

Speech synthesis was the driving force behind initial attempts to process signal digitally. Because of the fast advances in micro-electronics technology, DSP has found numerous applications. Any scientific or technological problem requiring real time computation and control capabilities can only be tackled by DSP. The advantages of DSP techniques are, high precision, controlled accumulation of noise, stability and simple system hardware. The application potential of DSP has increased further because of availability of general as well as dedicated DSP processors. The areas where DSP techniques have been employed, are controls, graphics, voice/speech, instrumentation, telecommunications, industrial, medical, military etc.

The need of filtering is felt in quite a few fields in electronics in order to remove the unwanted frequency components including noise. However, the analog filters had some drawbacks and would not be successfully used in different applications. With the advent of DSPs, the digital filters would be used in the fields of robotics, intelligent sensors, seismic signal processing, medical signal processing etc.. Digital filters are preferred because they are software controlled, whose performance is not affected by the drift in the component values due to environmental conditions or tolerances. Adaptive filters can also be designed, which can change their performance themselves with change in input conditions. They can be used for very low frequency applications. The problem of high cost has eased to a large extent due to advancement in VLSI techniques. The challenging fields in digital filters are development of software with high speed algorithms, which will be user-friendly and menu-driven, design of hardware systems such as ADC, DAC, interface with high speed components, designing of adaptive filters and to apply these filters to other systems for control.

In this work it was proposed to design, implement and test digital filters. Butterworth filter was selected. The filter is straight forward in pole zero configuration of filter transfer function in S-plane; all the poles lie on the circle centred at the origin of the S-plane and all zeros lie at infinity. The magnitude response of Butterworth filters is also smooth decreasing monotonically with frequency and does not have undershoot and overshoot in pass band and in stopband, i.e., response is free from ripple. In this work low pass, high pass, band pass and band stop Butterworth 1st order filters are designed, simulated and implemented on microprocessor based system.

A digital filter is an algorithm or numerical procedure which converts a sequence of given numbers into another sequence that possesses some more desirable properties such as less noise or distortion. The digital filters find applications in many fields such as robotics, radar signal processing, pattern recognition, speech signature etc.. The digital filter structure determined directly from either difference equation or its system function is called direct form-I. The structure using smaller number of delays than direct form-I is called direct form-II; it is also called the cononical structure which is memory-efficient. Most commonly used techniques for designing of analog filters are Butterworth, Chebyshev and elliptic. Impulse invariance and bilinear transformation methods are used to translate S-plane singularities of analog filter in z-plane. A digital filter is then implemented from these z-plane singularities. Impulse invariance method maps singularities in the s-plane into equivalent singularities in the z-plane. The digital filter design process involves determination of transfer function and the coefficient values which satisfy given specifications in frequency domain. Let Y(s) denote the Laplace transform of the output y(t) of a continuous time (CT) filter, when subject to an input x(t). If Y(s) consists of only simple poles, then,

$$Y(s) = \sum_{i=1}^{M} \frac{A_i \, x(s)}{s+s_i} = H(s)\, X(s) \qquad 5.1$$

where s_i is the ith pole of the transfer function H(s) and X(s) is the Laplace transform of x(t)

$$h(t) = \sum_{i=1}^{M} A \, e^{-s_i t}$$

The essence of impulse invariant method is to obtain a DT filter whose impulse response h(n) is exactly equal to sampled values of h(t) that are T seconds apart, where T is the sampling interval.

Z-transform of h(n) = H(z) can be written as

$$H(z) = \sum_{i=1}^{M} \frac{A_i z}{z-e^{s_i T}} \qquad 5.2$$

This is a transfer function of DT filter corresponding to CT filter. However, by this method we cannot design high pass, band pass and band stop filters directly. We have to use frequency transformations to convert low pass to any filter. Bilinear transform (BLT) is mostly commonly used technique for designing of IIR filters. This transform provides a nonlinear one-to-one mapping of the frequency points on $j\omega$ axis in S-plane to those on the unit circle in the Z-plane. It also allows us to implement digital high pass filters from their counterparts.

BLT is a transformation which relates points on S and Z-plane as given below.

$$s = \frac{z-1}{z+1} \text{ or } z = \frac{1+s}{1-s}$$

The phenomenon of frequency warping is associated with bilinear transform.

The advantage of the bilinear transform method is that it enables the existing analog filter design theory to be applied directly to the design of discrete filters. For a given filter it is possible to calculate the exact pass band ripple, transition band-width and minimum stopband attention.

Three steps involved in BLT design:

(1) Prewarp the critical frequency ω_{D1} to obtain analog frequency ω_{A1}

$$\omega_{A1} = \tan\left[\frac{\omega_{D1} T}{2}\right]$$

(2) Obtain the scaled transfer function $\hat{H}(s)$ from H(s)

$$\hat{H}(s) = H(s)\Big|_{s = s/\omega_{A1}} = H\left(\frac{s}{\omega_{A1}}\right)$$

(3) Apply the BLT transformation,

$$H(z) = \hat{H}(s)\Big|_{s = (z-1)/(z+1)}$$

*Design of Low Pass Filter Using BLT Method:

The transfer function of simple RC low pass filter with band width 1 rad/sec. is

$$H(s) = \frac{1}{s+1} \qquad 5.3$$

Bilinear transform yields

$$H(z) = \frac{1+z^{-1}}{A+Bz^{-1}} \qquad 5.4$$

where A = 1+1/a, B = 1-1/a

and $a = \tan\left[\frac{\omega_{D1} T}{2}\right]$, $\omega_{D1} = 2\pi Fc$

The transfer functions of HPF, BPF and BSF have been obtained by similar process. The digital filter is implemented from the system function. The equation 5.4 is system function of the LPF.

In this work discrete low pass, high pass, band pass and band stop filters have been designed. For simulation a program in BASIC is developed, which gives magnitude and phase response of a filter for any cutoff frequency.

Program Description and Features:

The given program simulates low pass, high pass, band pass and band stop filters. It gives the magnitude and phase response in graphical representation, that helps to observe the characteristic curve, pass band, stop band, transition region, ripple etc.. The response is displayed as a graph on video screen, the hard copy of which can also be taken. The number of points in the graph are user selectable, maximum points being 200. The aliasing problem can be observed by keeping F_S smaller than F_B. There is no restriction on cutoff frequency of the filter. The software is completely menu-driven. Initially main menu will appear on the screen which helps to select only one of the four filters. On each parameter entry the program displays what is to be entered next. A program for tolerance analysis of the filter is also supplied with this program, which calculates the change in cutoff frequency and change in component values due to component tolerance for the corresponding analog low pass and high pass filter.

The facilities in the software and some notable features are highlighted below:

1 The maximum as well as minimum amplitude corresponding to magnitude and phase response are displayed on the screen.

2 Computed values of magnitude, phase and input frequency upto the frequency band of interest with increment of F_B/Np are displayed. An option is provided to have the hard copy of the same.

3 Option is provided to display the magnitude as well as phase response either in graphics or in text (character) mode.

4 The hard copy is available with the help of "Shift Prt Scr" keys.

5 Provision is made to observe magnitude as well as phase response keeping the same input parameters by pressing the key "Q" when monitor displays the response.

6 The response with change in parameters can also be observed, while the new parameters are being entered, the old parameters are also displayed.

7 The software is adaptive in nature in the sense that while displaying the response it automatically chooses the scale on X and Y axes and selects the X axis position on the screen at optimum position so as to display positive and or negative points with maximum resolution.

8 For hard copy of responses load MS-DOS graphics file before execution of the filter program. At the time of printing use "Shift Prt Scr" keys.

9 The message "Please wait" indicates that CPU is busy with computations.

10 The main menu contains two more options, Help and Exit. Help option gives operating information and features of the program, while Exit option transfers control to MS-DOS.

11 The program is made available in the form of executable file.

Digital filter consists of microprocessor system, ADC and DAC. The microprocessor we have used is 8085 with the ADC 0809 and DAC 0808. PPI-8255 is used for I/O ports at addressees 08H, 09H, 0AH, and 0BH. ADC 0809 is kept in I/O map at address 09H. A separate clock frequency is generated for clock input of ADC using IC 4049. Channel '0' is used in our system for the input. As the input range of DAC is 0 to 5V, the negative part of the input waveform has to be clamped above zero level which is done by a clamper circuit. Port C of 8255 is used in BSR (bit set reset) mode to provide handshake signals to ADC. PC1 is used to read the end of conversion signal. PC4 is used to give ALE signal to ADC 0809. PC5 gives output enable (OE) signal to ADC and PC6 is used to give start of conversion signal to ADC. The circuit diagram of ADC 0809 is shown in Fig. 4.2 The DAC 0808 is connected to port A of 8255 at address 08H.

Algorithm for LPF:

The system function of low pass filter is given by

$$H(z) = \frac{V_o(z)}{V_i(z)} = (1+z^{-1})/(A+Bz^{-1}) \qquad 5.5$$

Upon simplification this equation 5.5 gives

$$V_o = 1/A\ (V_i+V_i z^{-1})-B/A(V_o z^{-1}) \qquad 5.6$$

The algorithm to compute V_o using a microprocessor-based system is as follows. In the equation 5.6 z^{-1} indicates unit delay or delay by one sample.

1) First compute the constants 1/A and B/A from the filter parameters.

$A = 1+1/a \qquad B = 1-1/a$

where $a = \tan\left[\frac{\omega_{D1} T}{2}\right]$

$\omega_{D1} = 2\pi Fc$, Fc - is cutoff frequency

2) Store the initial I/O conditions corresponding to the zero input signal and store the corresponding digital equivalent code (V_o). This data form z^{-1} for the next input sample.

3) Read the value of input data after each time 'T' through port 09H.

4) Compute V_o using the equation

$$V_o = 1/A\ (V_i + V_i Z^{-1}) - B/A(V_o Z^{-1})$$

5) Refresh delayed values of V_i and V_o. For the current sampled V_i, the previous V_i acts as $V_i z^{-1}$. Similarly, for the present V_o to be computed previous V_o acts as $V_o z^{-1}$.

The flowchart of the algorithm and ALP to implement the filter have been given in Chapter IV.

Alternatively numerical solution method has also been worked out for the design of LPF. Using this method the system transfer function H(z) is obtained by following transformation:

$$s \rightarrow \frac{1-z^{-1}}{T}$$

where T - sampling period.

For LPF the system function yields

$$V_o = \frac{aT}{aT+1} V_i + \frac{1}{aT+1} V_o z^{-1}$$

The algorithm for the implementation is as follows:

1) Compute the constant 'a' using the relation

$a = 2\pi Fc$, Fc - filter cutoff frequency

and $T = \frac{1}{F_s}$, F_s - is the sampling frequency

2) Evaluate the constants

$$A = \frac{aT}{1+aT} \quad \text{amd} \quad B = \frac{1}{1+aT}$$

3) For zero input store the value zero for V_o which will act as $V_o z^{-1}$ for the next input sample.

4) Read the I/P and multiply it by 'A'. To this add the delayed O/P after multiplying by B.

5) O/P the computed V_o to DAC.

6) Refresh the value of $V_o z^{-1}$ and repeat from step 4.

The main problem involved in implementing the digital filters is the speed improvement. Although the use of fast converters is the obvious choice the high cost precludes their use in general purpose applications. The selection of microprocessor chip is equally important to increase the throughput. The following table compares various versions of 8055 chips.

Processor	Crystal MHz	State time nS(T)
8085 AH-1	11.976	167
8085 AH-2	10.000	200
8085 AH	6.250	320
8085 AH	6.144	325.5

The throughput can however, be largely improved using the DSP processors such as TMS 320 series, DSP 2100, DSP 56001, VSP-325, DSP 16A, DSPi, etc.

Sesmic wave processing can provide valuable information on the behaviour of earthquakes. A proper digital filter to record these signals will be a basic tool. These signals can be processed by DSP methods to evolve a model that can furnish meaningful information.

cCc