GX568

OLLSCOIL NA hÉIREANN, GAILLIMH
THE NATIONAL UNIVERSITY OF IRELAND, GALWAY

SUMMER EXAMINATIONS 2001

B.E. DEGREE IN ELECTRONIC ENGINEERING
B.E. DEGREE IN ELECTRONIC AND COMPUTER ENGINEERING

DIGITAL SIGNAL PROCESSING

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Duration of Examination : ***Three*** hours

Instructions: Answer **five** questions.
All questions carry equal marks.

1. (a) A discrete-time system is shown in Figure 1(a).

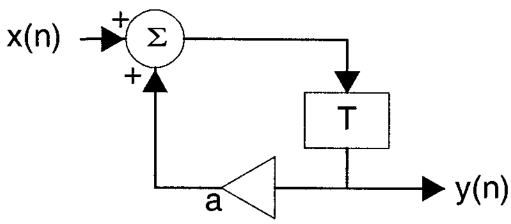

Figure 1(a)

Calculate the impulse response, $h(n)$, for n=0..4, and obtain an expression for the frequency response, $H(\theta)$, of the system. Hence, obtain an expression for the magnitude response and calculate the cutoff frequency of the system (in radians) if a = 0.88.

(b) With the aid of a suitable graphical example, describe the process of convolution of two signals, in terms of the operations performed on the signals.

The first non-zero value in a finite length sequence $x(n)$ occurs at n=-6 and has a value $x(-6)$ = 3. The last non-zero value occurs at index n = 24, and has a value $x(24)$ = -4. What is the index of the first nonzero value in the convolution of $x(n)$ with itself, and what is its value? What is the index and value of the last non-zero value of the convolution?

2. (a) Explain what is meant by windowing, and why it is necessary in spectral analysis of discrete-time signals, or in FIR filter design. Citing suitable examples of windows, discuss the issues involved in choosing a suitable window for a particular application.

(b) A discrete-time system has impulse response

$$h(n) = \alpha^n u(n)$$

[cont'd]

Obtain an expression for the frequency response of the system, and hence calculate the phase spectrum. From the phase spectrum, obtain an expression for the group delay of the system.

Use the fact that

$$\frac{d}{dx}\tan^{-1}\left[\frac{-a\sin x}{1-a\cos x}\right]=-\frac{a^2-a\cos x}{1+a^2-2a\cos x}$$

(c) The Fourier transform of the sequence $a^n u(n)$ is given by:

$$\frac{1}{1-ae^{-j\theta}}$$

Using this fact, find the output of a system with impulse response $h(n) = a^n u(n)$, when the sequence $b^n u(n)$ is input to the system (hint: recall the convolution property of the Fourier transform).

3. (a) Using the Window method, and starting from the frequency response, design an FIR lowpass filter that has a cutoff frequency of 0.3π radians. If the filter is going to be used at a sampling rate of 8 kHz, choose the filter length such that the latency of the filter is less than 10 msec.

(b) Using the impulse-invariant transformation, transform the following analogue filter into a corresponding digital filter:

$$H(s)=\frac{2}{(s+2)(s+5)}$$

Hence, write the difference equation for the filter.

(c) Explain why the gain of a discrete-time filter designed using the impulse-invariant transformation is different from that of the continuous-time filter on which it is based.

4. (a) Using the pole-zero placement method, design a digital band-pass filter with the following characteristics:

(i) Complete rejection of the input signal at DC and half the sampling frequency
(ii) A narrow passband centred at 500 Hz
(iii) A 3 dB bandwidth of 100 Hz

The sampling frequency is 4 kHz. Also, sketch the pole-zero map of the filter.

(b) In biomedical signal processing applications (e.g. electrocardiogram measurements), interference from the 60 Hz mains voltage (in North America) is common. Design a narrowband notch filter to reject this interference, where the sampling rate is 500 Hz. You may assume that a 3 dB notch bandwidth of 20 Hz is sufficiently narrow. Sketch the pole-zero map of the filter.

(c) Using the bilinear transformation, determine the transfer function of the digital equivalent of a first-order RC analogue filter, whose normalised (i.e. $\omega_c = 1$ rad/s) transfer function is given by:

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

Assume a sampling frequency of 2 kHz, and a cutoff frequency of 300 Hz.

5. (a) A decimation filter is required that reduces the sampling rate of a signal by a factor of 16. Draw a block diagram of a suitable multi-stage system that implements the required operation, and comment on the complexity of the filters required for implementing each stage of the system. Also, specify whether each stage would be implemented using FIR or IIR filters, and explain your choice.

(b) A signal processing application requires that the sampling frequency of a signal be changed from 2 kHz to 3 kHz. Design a rate-changing system to achieve this, and sketch approximately the spectra of the signals at various points in the system, as well as the frequency responses of any filters required (it is not necessary to actually determine the coefficients of the filters).

(c) A digital modulator is shown in Figure 1. The input signal has a bandwidth that extends from 500 Hz to 1 kHz, and the sampling frequency is 8 kHz. The carrier frequency of the modulator is 2 kHz.

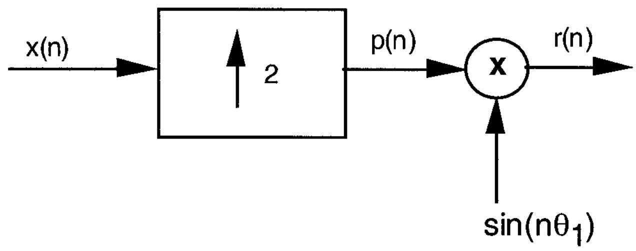

Figure 1.

(i) Determine the sampling frequency of each of the signals in Figure 1.
(ii) Sketch $|X(\theta)|$ against θ and ωT, where T is the sampling period of $x(n)$.
(ii) Sketch $|P(\theta_p)|$ and $|R(\theta_p)|$ against θ_p and ωT, where θ_p is the digital frequency variable for the upsampled signal.

6. (a) With the aid of a block diagram, describe the model for speech production, and derive an expression for the transfer function between the speech signal and the excitation.

(b) The autocorrelation function and the Average Magnitude Difference Function are to be used to extract the pitch of speech from a male speaker with a fundamental frequency of 130 Hz. Draw a rough sketch of the expected value for each of the two functions, indicating clearly the time differences between peaks and nulls. The sampling frequency is 8 kHz.

(c) Briefly describe the basis of the Linear Predictive Coding method of speech analysis (there is no need to do a detailed derivation of the equations). Discuss the relationship between the order of the prediction filter and the prediction error.

7. (a) With the aid of equations, describe the operation of the Least Mean Squares (LMS) algorithm, and discuss the stability and convergence speed of the algorithm.

(b) An adaptive filter is to be used in an acoustic echo cancellation problem, where the input signal to the adaptive filter has a peak amplitude of 3.7 and a peak to average ratio of 4.9. The length of the filter is 256 coefficients. Calculate a suitable upper limit for the step size of the filter, based on this information.

(c) Describe the normalised LMS algorithm for adaptive filtering, and outline the advantages that such a filter would have, and in what applications.