**BITS, PILANI - DUBAI CAMPUS**

**Knowledge Village**

| | |
|---|---|
| **IV-Year – 2nd SEMESTER 2003-2004** | **Date: 3/6/2004** |
| **Digital Signal Processing (EEE UC415)** | **Time: 3 Hours.** |
| **Max Mark: 80** | **Weightage: 40%** |

**Comprehensive Exam (Closed Book)**

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**Q1}**

A) Design a bandstop digital IIR filter with a Butterworth magnitude to satisfy the following specifications:

| | |
|---|---|
| Stopband | 400 Hz- 600 Hz |
| Lower passband | 0-300 Hz |
| Upper passband | 700 Hz-1000 Hz |
| Passband ripple | 3 dB |
| Stopband attenuation | 20 dB |
| Sampling frequency | 2000 Hz |

Determine the coefficients of the digital filter using the BZT method.

B) Write a MATLAB program to design an IIR digital filter that meets the specifications mentioned in part (A). Use the signal processing toolbox to determine its coefficients and to plot its spectrum. [8+4]

**Q2}**

A) Design a highpass digital FIR filter to meet the following specifications using the window method. Incorporate, in your answer, the type of the window and the reason for your choice

| | |
|---|---|
| Stopband attenuation | 55dB |
| Passband edge frequency | 4 kHz |
| Transition width | 1 kHz |
| Sampling frequency | 10 kHz |

B) Using MATLAB and signal processing toolbox, write an m-file program that will determine the filter coefficients and plots its spectrum. [7+4]

**Q3}**

A) An IIR digital filter is described by the following transfer function:

$$H(z) = \frac{0.0606 + 0.0483z^{-1} + 0.1047z^{-2} + 0.11047z^{-3} + 0.0483z^{-4} + 0.0606z^{-5}}{1 - 2.0998z^{-1} + 3.1022z^{-2} - 2.6061z^{-3} + 1.4443z^{-4} - 0.04134z^{-5}}$$

Determine the filter realizations using:

1- Cascade structure.
2- Parallel structure.

B) Write a MATLAB program that is used to factorize the transfer function into second order sections and to find all coefficients. [8+3]

**Q4}**

An audio signal is sampled at 10 kHz, in which it is required to separate the frequency components below 250Hz. The filter has a passband $0 \le f \le 220$ Hz and a transition band of $220 \le f \le 250$ Hz. The filter specifications are:

Passband ripple $\delta_p = 10^{-2}$

Stopband attenuation $\delta_s = 10^{-4}$

$$\text{Filter length } N = \frac{-10 \operatorname{Log}_{10}(\delta_p \, \delta_s) - 13}{14.6 \, \Delta f} + 1$$

where $\Delta f$ is the transition width.

Design an efficient two stage decimator, specifying the sampling frequencies, the passband ripple and stopband attenuation, edge frequencies and the filter length of each stage.

[10]

**Q5}**

**A)** Show that the adaptive filter turns itself off when there is no correlation between the interference signal, $x_x$, and the contaminated signal $y_k$.

**B)** The output signal from an adaptive noise chancellor is given by:

$e_k = y_k - \mathbf{X_k}^T \mathbf{W_k}$

Where $\mathbf{W}_k$ is the adaptive filter weight vector

$\mathbf{X}_k$ is the input vector.

$Y_k$ is the contaminated signal with noise.

Starting with this equation, derive:

1) the discrete Wiener-Hopf equations
2) the basic LMS algorithm.

[4+4+4]

**Q6}**

**A)** State why the RLS adaptive algorithm may become unstable after certain number of recursions? What do you suggest to solve this problem?

**B)** Compare the following:

1- Stability, rate of convergence and computational complexity of LMS and RLS adaptive algorithms.
2- Finite word length effects in FIR and IIR digital filters.
3- Hamming window and Kaiser window.
4- Adaptive filters and fixed coefficients filters.

[4+8]

**Q7}**

**A)** Draw the block diagram of the internal architecture of the TMS320C5X DSP digital signal processor. Mark all units, busses and elements.

**B)** Write a TMS320C5X assembly language program that will be used to generate and find the sum of 1, 2, 3, …, n integers.

[6+6]

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**IV-Year – 2nd SEMESTER 2003-2004**
**Digital Signal Processing (EEE UC415)**
**Test-II (Open Book)**

**Max Mark: 40**
**Weightage: 20%**
**Time: 50 Min**
**Date: 25/04/04**

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**Note: All questions carry equal marks.**

**Q1}**

An IIR digital filter is described by the following transfer function:

$$H(z)=\frac{0.1852\,z^5 + 0.8354\,z^4 + 1.5893\,z^3 + 1.5893\,z^2 + 0.8354\,z + 0.1852}{Z^5 + 1.5435\,z^4 + 1.5565\,z^3 + 0.8273\,z^2 + 0.8584\,z + 0.03414}$$

Draw the filter realization using:

a. Cascade structure.
b. Parallel structure.

**Q2}**

Design a sample rate converter which will change a digital audio signal (DAT) x(n) sampled at 48 kHz to a CD audio compatible signal y(m) with 44.1 kHz sample rate.

a. Specify the up and down sample rates.
b. Can this be implemented so that no information in x(n) is lost in output y(m)? Why?
c. Assuming that the filter used in the sample rate converter is ideal, what should be its corner frequency?
d. What will be the minimum FIR filter length, N, to be used?

**Q3}**

An audio signal has a passband signal of $0 \le f \le 75$Hz and a transition band of $75 \le f \le 80$ Hz. It is required to interpolate by a factor of 50. Design a two stage decimator. Specify the sampling frequencies, . Filter specifications are:

Passband ripple $\delta_p = 10^{-2}$

Stopband attenuation $\delta_s = 10^{-4}$

$$\text{Filter length } N = \frac{-10 \operatorname{Log}_{10} \delta_p \delta_s - 13}{14.6\,\Delta f} + 1$$

**Q4}**

a) Why the RLS algorithm converges faster than LMS algorithm?
b) Show that the adaptive filter turns off when there is no correlation between the interference signal, $x_k$, and the contaminated signal $y_k$.
c) What are the limitations of the RLS algorithm? Mention alternative adaptive algorithms which will avoid these limitations, explain briefly.

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**IV-Year – 2nd SEMESTER 2003-2004** **Max Mark: 8**

**Digital Signal Processing (EEE UC415)** **Time: 30 Min.**

**Quiz-1**

Q1} A Butterworth IIR bandpass filter is required to satisfy the following specifications:

| | |
|---|---|
| Passband | 200-300 Hz |
| Lower stopband edge | 50 Hz |
| Upper stopband edge | 450 Hz |
| Stopband attenuation | 0.001 |
| Passband ripple | 0.01 |
| Sampling frequency | 1000 Hz |

Using BZT and **MATLAB** obtain the magnitude frequency response of the filter. Also plot the-zero diagram of the filter.

Q2} Design a Butterworth lowpass digital IIR filter using BZT and MATLAB

| | |
|---|---|
| Cutoff frequency | 500 Hz |
| Sampling frequency | 4 kHz |
| Filter order | 3 |

Determine the filter coefficients. Plot the magnitude response and the pole-zero diagrams.

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**IV-Year – 2nd SEMESTER 2003-2004** **Max Mark: 8**

**Digital Signal Processing (EEE UC415)** **Time: 30 Min.**

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**IV-Year – 2nd SEMESTER 2003-2004**

**Digital Signal Processing (EEE UC415)** **Date: 21.3.2004**

**Max Mark:40** **Test-I (Closed Book)** **Time: 50 Min.**

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**Q1}** Design a bandpass IIR filter, using bilinear z-transformation (BZT), with the following specifications:

| | |
|---|---|
| lower stopband edge frequency | 100Hz |
| upper stopband edge frequency | 900Hz |
| lower passband edge frequency | 300Hz |
| upper passband edge frequency | 700Hz |
| passband ripple | 3dB |
| stopband attenuation | 20dB |
| sampling frequency | 2000Hz |

Assuming Butterworth magnitude-frequency response, determine the filter order and its coefficients. [14]

**Q2}**

**A)** Calculate the coefficients of a lowpass FIR digital filter given the following specifications and using the window method:

| | |
|---|---|
| Passband edge frequency | 2kHz |
| Transition width | 0.5kHz |
| Stopband attenuation | >50dB |
| Sampling frequency | 5kHz |

[12]

**Q3}**

**A)** Explain the concept of linear-phase in digital filters and mention the condition for linear-phase.

**B)** In the window method, for designing FIR digital filter design, compare the performance various window types.

**C)** Explain briefly the advantages and disadvantages of the window method for designing FIR digital filters.

**D)** What are the disadvantages and disadvantages of IIR filters, compared to FIR filters?

[14]

**BITS, PILANI - DUBAI CAMPUS**
**Knowledge Village**
**IV-Year – 2nd SEMESTER 2003-2004**

**Digital Signal Processing (EEE UC415)** **Date: 21.3.2004**
**Max Mark:40** **Test-I (Closed Book)** **Time: 50 Min.**

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