

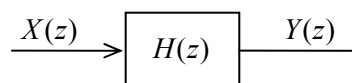
Digital Signal Processing
Tutorial Questions Booklet

Digital Signal Processing

Tutorial Sheet 1

Sampling and z-transforms

- 1*. Write down an expression in the z-domain for $Y(z)$ in terms of $X(z)$ and $H(z)$.



- 2*. Write down an expression for $X(z)$ in terms of $x(n)$.
- 3*. Write down an expression for $y(n)$ in terms of $x(n)$ and $h(n)$.
- 4**. Show that your expression for $y(n)$ in question 3 is related to your expression for $Y(z)$ in question 1 by the z-transform.
- 5***. Find the inverse z-transform of the system $X(z) = \frac{7z^2 - 5z}{z^2 - 2z - 3}$ for the following regions of convergence.
- (a) $|z| < 1$
 - (b) $|z| > 3$
 - (c) $1 < |z| < 3$.
- State in all cases whether the system is causal, anticausal or non-causal.

Digital Signal Processing

Tutorial Sheet 2

Discrete Fourier Transforms

- 1*. Find the Discrete Fourier Series coefficients for the sequence

$$x_p(n) = 10 \sin(2\pi n / 3) .$$

- 2**. Find the magnitude spectrum of the sequence [1, 1, 2, 3, 3]. Don't use software tools on a computer unless you write you own.

- 3**. A speech signal is sampled at 8 kHz. A DFT is computed of a block of 1024 samples of the data.

- (a) What is the time duration of the block?
- (b) What is the frequency resolution of the DFT?
- (c) If only 512 data samples were taken but the block was zero padded to restore the size to 1024, what would be the new answers to (a) and (b)?

- 4***. A sequence $y(n)$ is constructed from a finite duration sequence $x(n)$ of length 8 samples in the following manner

$$y(n) = \begin{cases} x(n/2) & \text{for } n \text{ even} \\ 0 & \text{for } n \text{ odd} \end{cases}$$

Determine $Y(k)$ in terms of $X(k)$ where $Y(k)$ and $X(k)$ are the DFTs of $y(n)$ and $x(n)$ respectively.

Digital Signal Processing

Tutorial Sheet 3

Digital Filters

- 1**. Design a causal linear phase FIR filter with 8 taps and a cut-off frequency of $\pi/4$ using the frequency sampling method with a rectangular window.
- 2*. Find an expression for the d.c. gain of the first order lowpass filter
- $$H(z) = k \frac{1 + z^{-1}}{1 - cz^{-1}}, \quad |z| > |c|$$
- 3*. For the filter of Question 2, find the value of k for unity gain at d.c.
- 4**. For the filter of Question 2 with unity d.c. gain, find the value of c to give -3 dB gain at $\omega = \pi/2$.
- 5**. Sketch the magnitude response for the filter of Question 2 with $k = 1$ and
- (i) $c = -0.9$
 - (ii) $c = -0.5$
 - (iii) $c = 0$
 - (iv) $c = 0.5$
 - (v) $c = 0.9$.

Digital Signal Processing

Tutorial Sheet 4

Convolution

1*. Given that $w(n) = x(n) * y(n)$, show that $W(z) = X(z).Y(z)$.

2**. Show that $x_{3p}(n) = \sum_{l=0}^{N-1} x_{1p}(l).x_{2p}(n-l)$ given that $X_p(k)$ is the DFS of $x_p(n)$ and that $X_{3p}(k) = X_{1p}(k).X_{2p}(k)$.

3***. Perform the linear convolution of the sequences $x(n)$ and $h(n)$ using (i) the overlap add procedure and (ii) the overlap save procedure.

$$x(n) = [3, 2, 1, 4, 1, 2, 3, 2, 1]$$
$$h(n) = [1, 1]$$

The suggested block size is 3.

You may use MATLAB to compute the FFTs but otherwise work this problem on paper.

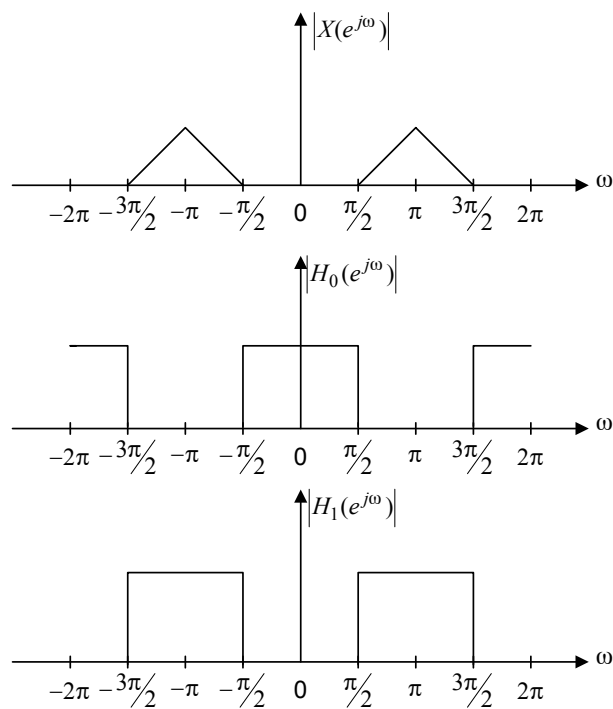
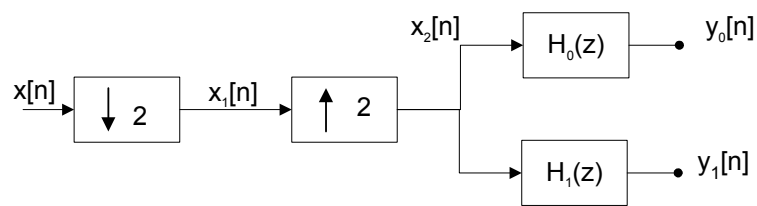
Digital Signal Processing

Tutorial Sheet 5

Multirate Signal Processing

1**.

For the system below, sketch $X_1(z)$, $X_2(z)$, $Y_0(z)$ and $Y_1(z)$.



2*.

Design a scheme for reducing the sampling rate of a signal to 0.75 of its original sampling frequency. Sketch the magnitude of the frequency response of any filters employed.

3**.

Given a filter $H(z) = 1 + 2z^{-1} + 3z^{-2} + 4z^{-3}$, decompose this filter using type 1 polyphase decomposition into two filters $E_0(z)$ and $E_1(z)$. Sketch the signal flow graph for both the original filter and the polyphase filter.