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COLÁISTE NA hOLLSCOILE
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GALWAY

SEMESTER 1 EXAMINATIONS 2000-01

B.Sc. (Honours) Applied Physics and Electronics
B.Sc. (Honours) Physics and Computing

Paper III: Digital Signal Processing (EP412)

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Time Allowed: TWO hours	Answer THREE questions

Q.1 Briefly define or explain the following quantities as used in Discrete Time (DT) systems: (a) the impulse response h[n]; (b) the system difference equation; (c) the transfer, or system, function H(z); (d) the DT frequency response H(ω), and, (e) the Direct Form II (DF II) block diagram. How, if at all, are these quantities related to each other?

A DT system, with input x and output y, has the following difference equation:

$$y[n] + 0.8y[n-2] = x[n] - x[n-2]$$

Deduce the system transfer function H(z). Draw up the DF II block diagram and the pole zero plot for the system. What can you say about the general nature of this system as a digital filter? Calculate the numerical values of the first 5 terms in the system impulse response h[n].

Q.2 Answer *ALL THREE* parts.

(a) State the Sampling Theorem, and define or explain the terms: Nyquist Rate, Nyquist Frequency, Folding Frequency, and Aliasing.

(b) Give a short proof of the Sampling Theorem, by defining an ideally sampled signal and considering its Fourier Transform (or frequency spectrum).

(c) A signal, consisting of a mixture of 3.5, 5.4 and 6.2 kHz sinusoids, is sampled with a sampling interval T = 100 μsec. Calculate the sampling rate and the alias frequencies of the original sinusoids, and also the Nyquist Rate and Folding Frequency for this signal at this sampling rate.

Q.3 Say what is meant by a *digital filter*. State the relationship between the DT (Discrete Time) system function H(z) of a digital filter and its frequency response function H(ω). Explain how a pole-zero diagram of H(z) in the z-plane, together with the z-plane unit circle, can be used to estimate the DT filter frequency response.

Say what are meant by *IIR, FIR* and *linear phase* digital filters. Describe, in full, the window method of indirect FIR digital filter design. Explain the suitability of this method for the design of linear phase filters, and state why linear phase is desirable in most practical filters.

Q.4 Give a brief comparison of the BLT (Bilinear Transform) and II (Impulse Invariance) methods of Discrete Time (DT) system design. Explain how the BLT avoids the problem of aliasing, and say what the resulting disadvantage of the BLT approach is. How is this disadvantage overcome in practical filter design?

A DT Low Pass Filter (LPF), with a sampling interval T = 10 sec, is to be designed with its cut-off (-3 dB) frequency = 12.5 kHz and its stop band edge (-20 dB) frequency = 15 kHz. Give a general outline of the steps in the procedure necessary to design the DT filter, working from a Continuous Time (CT) prototype Butterworth LPF and using the BLT (indirect design method) to transform this to DT.

Calculate the required CT cut-off and stop band edge frequencies numerically, but simply describe in general terms (without numerical calculations) all the other steps in the design. You may quote, if you know them, any relevant formulae which you think are important

Q.5 Answer any *TWO* of the following.

(a) Discuss and explain, in detail, the main relative drawbacks and advantages of Digital Signal Processing (DSP) systems compared with Continuous Time Signal Processing systems.

(b) Describe and compare the three different methods that can be used for the practical implementation of a Discrete Time (DT) difference equation by a DSP system. In order to be more specific in your answer, you may assume that the difference equation is of order 6.

(c) A DT system has the following difference equation:

$$y[n] - 0.5y[n-1] = x[n] + 2x[n-1] + x[n-2]$$

Find the system function H(z), and use it to find the first three terms in the system impulse response h[n] by the two methods of
(i) a direct series expansion of H(z), done by Long Division, and
(ii) a partial fraction expansion of H(z) followed by use of z-transforms.

Note that the z-transform of the DT series $x[n] = a^n$ is $\dfrac{1}{1 - az^{-1}}$.