Multimedia Signal Processing 1st Module

Pre-examination

23 November 2009

Ex.1 (Pt.5)

Consider the filter H(z) whose Transfer Function is: $H(z)=\dfrac{1+0.81z^{-2}}{1+0.49z^{-2}}$.

1. Define its block scheme in terms of inputs (x(n)) and outputs (y(n)).
2. Provide its zeros-poles diagram and draw an approximate behavior of its amplitude and phase response at different normalized frequencies.
3. If the filter is a "minimum phase" transform it into a "maximum phase" or vice-versa preserving the same amplitude response. Provide a new phase diagram accordingly to the modifications.

Ex.2 (Pt.5)

Consider a filter whose impulse response is

$$h(n)=\delta(n)-\delta(n-1)+2\delta(n-2)-2\delta(n-3)+3\delta(n-4)-3\delta(n-5).$$

1. Provide a polyphase implementation of its decimation of order 3 (M=3) and the associated structure (block diagram).
2. Using a decimation of order 2 (M=2) define a perfect reconstruction filter bank ($H_0(z)$, $H_1(z)$,$G_0(z)$, $G_1(z)$).

Ex.3 (Pt.5)

A Random Variable X has a Uniform Probability Density Function (pdf) between θ-½ and θ+½ (i.e. rect(x- θ)). Three realizations of X are x_1=1.5, x_2=1.7, x_3=2.5.

1. Represent and quote the likelihood function L(θ| x_1,x_2,x_3).
2. How would change L if the 3 samples would be x_1=1.6, x_2=1.7, x_3=2.4? Provide a quoted representation

Ex.4 (Pt.5)

An autoregressive model for parametric spectral estimation of a random process provide us with a one pole filter (a_1=-0.5)for a white noise of power σ^2=4.

1. Find the autocovariance values r(0), r(1) and r(2) of the final model.

Ex.5 (Pt.5)

Load the waveform 'mtlb' and determine the sampling frequency Fs.

Consider a LTI digital filter characterized by the following difference equation:

h(n) = -0.9 y(n-1) + x(n) + x(n-1)

Filter the input sequence x(n) with h(n) computing the convolution by means the OLA approach:

segment the signal x(n) into overlapping frames of length 20ms using a Hamming window at 50% overlap.

Ex.6 (Pt.5)

Load the waveform 'mtlb' and determine the sampling frequency Fs.

Decimate the signal to the sampling frequency: F = (p/q)Fs [p=4 and q=5].

Design the appropriate low-pass filters using the Matlab's function 'fir1'.