# 2nd Coursework Assignment for Signal Processing

- General terms: due on Tuesday 24/1/2006, 2pm, to the course office; solutions will be poster on 24/1/2005 after 4pm at www.ecs.soton.ac.uk/ sw1/elec3029/elec3029.html. This coursework contributes 5% to the total mark.

- Digital Filtering:

  Given is the transfer function $H(z) = 1 + \frac{5}{2}z^{-1} + z^{-2}$.

  1. Without explicitly calculating the magnitude response, state the filter gain for $\Omega = 0$ and $\Omega = \pi$.
  2. What are the phase properties: is $H(z)$ linear / non-linear phase, and if the latter, would it be minimum, non-minimum, or maximum phase?
  3. Calculate the frequency response $H(e^{j\Omega})$ to $H(z)$.
  4. State a minimum phase system $H_{\min}(z)$ and a maximum phase system $H_{\max}(z)$ that have the same magnitude response as $H(z)$.
  5. Given is the channel transfer function $C(z) = 1 + 2z^{-1}$. Calculate a (not necessarily causal but stable) equaliser $W(z) = C^{-1}(z)$.
  6. Relaxing the equaliser to $C(z) = z^{-10}W^{-1}(z)$, find an FIR implementation of the causal part of $C(z)$.

- Adaptive Digital Filtering:

  1. Using a labelled generic adaptive filter diagram, derive the LMS algorithm.
  2. Write down a short Matlab programme to simulate the LMS algorithm to identify an "unknown" system $C(z) = 1 - \frac{1}{2}z^{-2} + \frac{1}{3}z^{-3}$. Your adaptive filter should have four coefficients, and you should select an appropriate step size. Your input signal should be about 1000 samples long and consist of (i) Gaussian white noise and (ii) a sinusoid with normalised angular frequency $\Omega = \pi/2$.

     For cases (i) and (ii), provide plots of the instantaneous squared error. For case (ii), provide a justification for the adaptive filter coefficient after convergence (hint: study the frequency response).
  3. Why would practitioners prefer the LMS over the implementation of the Wiener-Hopf solution?

- Data Conversion

  1. The preconditioning stage of an ADC is adjusted such that the mean input power is 10% of the power of a sinusoid extending over the full input range of the quantiser characteristic, in order to avoid potential clipping in the quantisation stage. How must the word length $R$ be selected to guarantee an SQNR of at least 63 dB?
  2. An audio signal should be sampled at 16kHz. The ADC operates at 256kHz and is followed by a digital postprocessing stage comprising a digital anti-alias filter and a decimator. Sketch the requirement for the analogue anti-alias filter at the ADC input.
  3. Assuming an ideal digital anti-alias filter, calculate and compare the SQNR for the system in (2) with an ADC sampling directly at 16kHz. How would the wordlengths be selected to achieve the same SQNR?
  4. Derive the equivalent digital model of a first order $\Sigma - \Delta$ device, such that $Y(z) = X(z) + (1-z^{-1})N(z)$ for an ADC with input $X(z)$, uncorrelated observation noise $N(z)$ and output $Y(z)$. Calculate the PSD of the quantisation noise at the output of the device, and explain how technique can extract resolution in addition to oversampling.