Coursework Assignment for Signal Processing

- General terms: due on Monday 24/1/2005, 2pm, to the course office; solutions will be poster on 24/1/2005 after 2pm at www.ecs.soton.ac.uk/ sw1/elec3029/elec3029.html.
- Digital Filtering:

  Given $H_1(z) = 1 + 2z^{-1} + z^{-2}$, $H_2(z) = 1 - z^{-2}$, $H_3(z) = 1/(1 - .5z^{-1})$.

  1. What can you expect of the magnitude responses: lowpass, highpass or bandpass characteristics?
  2. What can you expect about the phase responses: linear/non-linear and minimum/maximum/non-minimum phase characteristics?
  3. Calculate the frequency responses $H_i(e^{j\Omega})$ to $H_i(z)$.
  4. If the $H_i(e^{j\Omega})$ are excited by uncorrelated Gaussian noise, state the power spectral densities (PSD) of the filter outputs.
  5. How does the PSD at the filter output alter if the filter input has a mean (DC offset) μ?
  6. Argue how you could approximate $H_3(z)$ by an FIR filter. Is this sensible in terms of implementation cost and stability?

- Adaptive Digital Filtering:

  1. Using a labelled generic adaptive filter diagram, derive the LMS algorithm.
  2. If your input signal is $x[n] = \cos(\pi n)$, is the Wiener Hopf solution for a 2-tap filter viable?
  3. What happens to the Wiener-Hopf solution, if the input signal is additionally corrupted by white Gaussian noise (zero mean, variance σ^2)?
  4. Calculate the analytic inverse for a channel $C(z) = 1 + \frac{1}{4}z^{-1}$. What would you expect an LMS adaptive filter with N coefficients to converge to for a white noise input?
  5. Discuss how you could adapt a channel equaliser for a maximum phase system, and exemplify that it works for $C(z) = \frac{1}{4} + z^{-1}$. [Perhaps verify in Matlab with an LMS adaptive equaliser.]

- Data Conversion

  1. 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.
  2. Compared to an ADC sampling at 16kHz with same world length: (i) how can you relax the analogue anti-alias filter and (ii) calculate the SQNR at the output of both systems. Comment on how you could adjust the word length in both structures to achieve identical SQNR.
  3. Assume the z-transform model $Y(z) = X(z) + (1 - z^{-1})N(z)$ for an ADC with input $X(z)$, uncorrelated observation noise $N(z)$ and output $Y(z)$. Argue how this ADC can benefit from your oversampled ADC architecture in further enhancing the SQNR.