TTT4120 Digital Signal Processing
Problem Set 8

The main topic for this problem set are stochastic processes and autoregressive modelling. Relevant chapters from the textbook are 5.3.2, 12.2-12.3. The maximum score for each problem is given in parentheses.

Problem 1 (2 points)

A random signal $x(n)$ is generated by filtering white Gaussian noise $w(n)$ with variance $\sigma_w^2 = \frac{3}{4}$ by a causal filter with transfer function

$$H(z) = \frac{1}{1 + \frac{1}{2}z^{-1}}.$$

Note that this is the same signal as in Problem 2 of Problem Set 7.

(a) Which type of process is $x(n)$? Justify your answer.

(b) Find the coefficients of the optimal first- and second order predictor.

Problem 2 (3 points)

A random process $x(n)$ is given by

$$x(n) = w(n) - 0.4w(n-1),$$

where $w(n)$ is white Gaussian noise with variance $\sigma_w^2 = 1$.

(a) Which type of process is $x(n)$? Justify your answer.

(b) Find the autocorrelation function $\gamma_{xx}(l)$ and the power density spectrum $\Gamma_{xx}(f)$ for this process.

(c) Calculate the coefficients of the optimal first, second and third order predictor for the process $x(n)$. (You can use Matlab to solve the Yule-Walker equations). Compute also the corresponding prediction error variances. Comment on the results.

(d) Write an expression for the power density spectrum estimate based on an AR[p] model.

Plot the calculated power density spectrum together with its estimates based on AR[1], AR[2] and AR[3] models. (Use Matlab functions `freqz`.)

Which of the AR models gives the best approximation of the MA process? Justify your answer.

Problem 3 (5 points)

In this problem you should make a vowel transformer that lets you record a vowel, and then transforms it to any desired Norwegian vowel pronounced in your own voice. This problem requires the use of headphones and microphone.

One sample of each Norwegian vowel can be downloaded from It's learning. Start by converting one of those vowels to other vowels.

When recording your own vowels you should use the sampling frequency $F_s = 8$ kHz, and make sure to extract only the file part containing the vowel before proceeding.

Hint: Vowels can be regarded as stationary signals, and are well modeled as AR[10] processes.

Useful Matlab functions are `audioread`, `audiorecorder`, `lpc`, `filter` and `sound`.