Signal Acquisition and Processing **IG 2407**

Final Exam

Lecture notes allowed

Duration: 2h00

Exercise n°1:

An audio signal whose spectrum is limited to 20kHz was digitized at a sampling frequency $F_e = 96\mathrm{kHz}$.. The ADC used has a resolution of 20 bits and operates over an amplitude range [-2V, 2V].

1. Compute the memory space required to store 1 second of audio signal.
2. Does the choice of the sampling frequency conform to Shannon's theorem? Justify your answer.
3. Compute the quantization step.
4. Compute the quantization noise power.
5. We assume that the crest factor, F_c, of the audio signal is equal to 3dB. Compute the signal to noise ratio at the ADC output.

A digital signal processor (DSP) is used to calculate the Discrete Fourier Transform (DFT) of this signal over 1024 points.

6. What minimum computational load (in terms of number of multiplications and additions per second) must this DSP have to be able to achieve this DFT?
7. What happens to this calculation load if we use a fast calculation algorithm (FFT)?
8. What is the cutoff frequency of the reconstruction filter applied at the output of the DAC?

Exercise n°2:

We consider a digital filter with two poles $P_0 = \frac{\sqrt{2}}{2} e^{j\pi/4}$ et $P_0^* = \frac{\sqrt{2}}{2} e^{-j\pi/4}$.

1. Compute the transfer function of this filter.
2. Does this filter have a FIR or an IIR structure? Justify your answer.
3. Write the difference equation corresponding to this filter.
4. Is this filter causal? Justify your answer.
5. Compute the frequency response of this filter.
6. Compute the modulus of the frequency response of this filter.
7. What's the type of this filter (band-pass, low-pass, high-pass, other)?
8. Plot the implementation scheme corresponding to this filter.

Exercise n°3:

We consider a real white noise $b(n)$ with variance σ^2 . This noise passes through a digital filter defined by its transfer function $A(z) = 1 - \sqrt{2}z^{-1} + z^{-2}$. We note $y(n)$ the signal at the output of the filter.

1. Does this filter have a FIR or an IIR structure? Justify your answer.
2. Compute the modulus of the frequency response of this filter.
3. What's the type of this filter (band-pass, low-pass, high-pass, other)?
4. Compute the autocorrelation function $r_{yy}(p)$ of the signal $y(n)$ at $p = 0$, $p = 1$, $p = 2$.
5. Give the general form of $r_{yy}(p)$.
6. Is the signal $y(n)$ white? justify your answer.
7. Give the expression for the power spectral density $P_y(f)$ of the signal $y(n)$. (We will give the most factorized expression).
8. Compute the power spectral density value for frequencies 0 and $3\frac{F_e}{8}$.

Good luck…