FILTERING

M2 AI — SIGNAL PROCESSING

SIGNAL AND SYSTEMS

A signal is recorded, and distorted, by a sensor

A signal is (almost) always linked to the notion of "system"

System: Functional block that reacts to an input excitation signal and produces a response signal after applying a function to the input signal

S is a "functional" that applies to a signal and returns another signal

FILTERS: LINEAR SYSTEMS INVARIANT IN TIME

A filter is a linear system that is invariant over time

Let be a filter with an impulse response . Then

For finite sequences (digital signals) of size , 2 possibilities

- Zeros-padding: and
- Circular convolution: and are supposed to be periodic (different results if the period is chosen to be or !)

FILTERING IN THE FOURIER DOMAIN

Let a filter with an impulse response :

then (if the Fourier transform exists):

For finite sequences (digital signals), the underlying convolution is circular: Fourier transform must be done after zeros-padding !

Filtering a signal acts directly on the spectrum

IDEAL FILTERS AND REALIZABLE FILTERS

Ideal filter cut some frequencies while other are preserved. 4 types of filters: Low pass, High pass, Band pass and Band cut filter.

A filter is realizable iff its impulse response is stable and causal

Ideal filters are not realizable (not causal)

Two kind of realizable numerical filters:

- Finite Impulse Response (FIR) filters:
- Infinite Impulse Response (IIR) filters:

DIGITAL FILTERS

Ideal filters can be implemented in the frequency domain (but not in real time)

FIR filters are stable, but need a lot of coefficients

IIR filters can be unstable, but accurate.

Classical IIR filters: Butterworth, Tchebychev I & II, Elliptical

TO DO: FIR DELAY EFFECT

Data:

Any (short) sound file

Goal:

The FIR filter for delay effect can be implement thanks to the following input-output equation (￼ is the input and ￼ is the output):

￼

Where ￼ is the attenuation factor and ￼ is the time delay

- Implement the delay effect in the time domain
- Determine the impulse response of the filter (numerically)
- Provide the Frequency response of the filter (numerically)

TO DO: IIR DELAY EFFECT

Data:

Any (short) sound file

Goal:

The IIR filter for delay effect can be implement thanks to the following input-output equation (is the input and is the output):

Where is the scaling factor, is the attenuation factor and is the time delay

- Implement the delay effect in the time domain
- Determine the impulse response of the filter (numerically)
- Provide the Frequency response of the filter (numerically)
- Is this implementation always stable ?
- Discuss the parameters
- Compare with the FIR implementation

TO DO: IMAGE FILTERING

Data:

Any image

Goal:

For a given image, implement the following filters and discuss their effects

- Gradient filter
- Sobel filter
- Averaging filter
- Gaussian filter