Cours de traitement des signaux biomédicaux
1ère séance Matlab

Useful commands
plot : graph of a signal (points linked).
subplot : several graphs on the same figure.
stem : graph of a signal ("numerical" style).
freqz : filter frequency response.
filter : output of a filter.
filtfilt : output of a filter with forward and backward filtering.
fir1 : FIR filter design.
buttord, cheb1ord, cheb2ord, ellipord: determination of filter order
butter, cheby1, cheby2, ellip: IIR filter synthesis.
resample : signal resampling.
hilbert : analytic signal computation.
unwrap : phase unwrapping.

Additional commands
filtre_hilbert : FIR Hilbert filter design (inverse FT and Hamming window).
diffe : FIR differentiating filter design (inverse FT and Hamming window).

Remarks
- The convention used Matlab for filter design is not 0 to 0.5 for normalized frequencies, but from 0 to 1 as a portion of π radians for the pulsation (frequency times 2π). For instance to have a normalized cutoff frequency of 0.2 for a lowpass filter, one must insert a parameter value w = 2*0.2 = 0.4
- The command freqz requires the number of frequency values at which the response is computed. Use at least 1000, it is so cheap.
- The command freqz uses the numerator and denominator coefficient vectors b and a of the filter transfer function. With an FIR filter, use for b the filter impulse response, and a = 1. Same thing for filter.
- With freqz if no output variable is asked for, a graph of the amplitude response (in dB) and of the phase response is automatically generated, of course with the convention mentioned above.
- To visualize the amplitude response with normalized frequencies, the simplest is to issue :
  ```
  >> [h,w]=freqz(b,a,1000) ;
  >> plot(linspace(0,0.5,1000,abs(h))
  ```
- To generate a sinusoid with frequency f and length n :
  ```
  >> x=sin(2*pi*f*(1:n)) ;
  ```
- In fir1, if an order n is specified, the filter length is n+1.

Experiment 1: transient of FIR filters, and the importance of linear phase

The signal in **ecg.dat** consists of several seconds of ECG sampled at 500 Hz. The small oscillations visible mainly between the PQRST complexes are due to 50 Hz interference. It can be suppressed with lowpass filtering, but care must be exercised not to deform the complexes.

The following commands generate a linear phase FIR lowpass filter with length 101 and cutoff frequency 40 Hz (thus normalized frequency 0.08) and visualize its response (2000 frequency values). Check why.

```
>> g = fir1(100,0.16) ;
>> freqz(g,1,2000) ;
```

The following commands generate a linear phase IIR (Butterworth) lowpass filter with length 101 and cutoff frequency 40 Hz (thus normalized frequency 0.08) with a transition band between 35 Hz and 45 Hz, a bandpass ripple of 0.5 dB, and 20 dB attenuation, and visualize its response (2000 frequency values). Check why.

```
>> [N,Wn] = buttord(0.14,0.18,0.5,20) ;
>> [b,a] = butter(N,Wn) ;
>> freqz(b,a,2000) ;
```

You can check that the two filters have similar amplitude responses by plotting them together. But apply both of them to a sinusoid of length 150 and normalized frequency 0.05, and plot the outputs together. Which filter has the shortest transient?

Then filter the ECG using both filters:

```
>> ecg1 = filter(g,1,ecg) ;
>> ecg2 = filter(b,a,ecg) ;
```

And forward-backward filtering:

```
>> ecg3 = filtfilt(b,a,ecg) ;
```

Use subplot to have the initial signal above all filter outputs.

1.1 Is the 50 Hz interference well cancelled?

1.2 Which filter distorts the QRS complexes the less?

1.3 Are the complexes distorted by the forward-backward filtering?

Experiment 2: Design of a differentiating filter with inverse FT – application to the estimation of an instantaneous frequency.

The file **mersurepoutre.dat** contains a signal, sampled at 1000 Hz, corresponding to the oscillation generated on a micro-beam by a culture of synchronized cardiac cells, that is, with self-organized, periodic excitation. Perform the following operations :

1. sub-sampling by a factor 100 with resample (the frequencies of interest are much lower than 1000 Hz, so no use in having many samples).

2. lowpass filtering using filtfilt with an IIR filter synthetized with :

```
>> [N,wn]=cheb1ord(0.06,0.1,0.5,20);
>> [b,a] = cheby1(N,0.5,wn);
```

You can check that filtfilt does not introduce a delay in the output, while filter does.

3. Compute the analytic signal of the filter output using Matlab command hilbert. You can check by plotting together the initial signal and the modulus of the analytic signal that the latter yields the signal envelope.

4. Compute the instantaneous unwrapped phase of the signal. With xa the analytic signal, this unwrapped phase phi is obtained with:

```
>> phi  = unwrap(angle(xa));
```
Finally, all that remains to do is to extract the instantaneous frequency by differentiating the instantaneous phase with the filter generated by diffe, and dividing it by 2π. Use a filter of length 11, and compensate for the delay. You can check that the impulse response is odd, due to the fact the frequency response is purely imaginary.

Experiment 3: event detection

File **accel.dat** is a recording (sampling frequency 40 Hz) from an accelerometer on the wrist of a health worker. The goal is to detect the disinfection episodes (clinical soap on the hands and rubbing). It is known that in the recording disinfection takes place between time indices 900 and 1100.

1. To detect this disinfection episode let us design a bandpass filter using:

```
>> [N,Wn]=buttord([0.14,0.16],[0.13 0.17],0.5,20);
>> [b1,a1]=butter(N,Wn);
```

How inspection of the signal plot can justify the bandpass type for the filter, and the particular choice of the filter central frequency? Why should filtfilt be used to filter the signal?

2. The instantaneous power of the filter output can then be estimated using a lowpass filter designed using:

```
>> [N,Wn]=buttord(0.04,0.02,0.5,20);
>> [b2,a2]=butter(N,Wn);
```

and by filtering the squared output signal (why?), of course using filtfilt again.

3. Perform the same operations using a bandpass filter with a passband three times as large, and a transition region twice as large. Comment on the relative merits of the two approaches.