EXERCISES SESSION

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Ex 1: Filtering in time and frequency

- Given
  - the filter h=[-1,0,1]
  - the signal x=[3 1 0 2 1 5]
  - time indices starting both from zero:

1. compute y as the linear convolution of h and x in the time domain
2. compute y2 as the linear convolution of h and x in the frequency domain
3. Plot the squared error between y and y2
4. Plot the two signals y and y2 together

Ex 2: Allpass filters

- Given a fir filter $B(z) = -2z^{-3} + 2z^{-1} + 3$,

1. implement an all-pass filter equal to H(z)=B(z)/A(z)
2. plot the poles and the zeros of the all pass filter in the z-plane
3. compute its frequency response from 0 to π and plot its modulus and phase (two subplots), with normalized frequency
4. compute and plot its impulse response: is it accurate?

Ex 3: Additive synthesis

- Open the wav file «flute.wav» as x and try to synthetically (and empirically) replicate it as a sum of $\mathrm{K}=3$ sinusoids y

1. Compute and plot the spectrogram of x
2. From a random frame of the spectrogram, find the K main sinusoids using the provided function `findPeaks`
3. Find the frequencies f_k and, from the STFT, the (time-variant) magnitudes $A_k(t)$
   - Normalize A_k with respect to the maximum over all k=1,…,3
4. Compute $y = \sum_k^K A_k \cdot \cos(2\pi f_k t)$ using a loop
   - Interpolate A_k using the function `interp` and L=<u>hopsize</u> of the spectrogram window
5. Normalize it: $y = y/\max(|y|)$ and listen to the result

Ex 4: Changing domains

- Given a filter with
  - $z = \left[0.8, -0.8, 0.8e^{-j\frac{\pi}{3}}, 0.8e^{+j\frac{\pi}{3}}\right]$
  - $p = [0 + 0.5j, 0 - 0.5j, -0.6 + 0.5j, -0.6 - 0.5j\ ]$

1. Plot the z-plane
2. Compute the difference equation of the filter
3. Plot the transfer function and impulse response of the filter in $H(e^{j\omega_k})$ in N=1024 points

Ex 5 Pitch tracking

Given the audiofile «flute.wav»

1. Load the signal in x
2. Find the pitch frequency of the note **p** (in Hertz) by means of the autocorrelation
3. Compute the spectrogram of the signal
4. Plot the magnitude of the STFT for a generic frame with the frequency axis in Hertz
5. Stem over the previous figure, the sample corresponding (or closest to) **p**

Ex 6. Overlap and Save Lowpass filter

Given the 16kHz-sampled audiofile Toms_diner_16.wav

1. Load the file in the signal x
2. Design a FIR lowpass filter b with cut-off frequency=6,000 Hz
   1. use the function fir1 and M=32 weights
3. Process the signal x with the filter h=b using the Overlap and Save technique
   1. Use block size N=1024
4. Plot two subplots with the original signal x and the filtered version

Ex 7. Minimum phase filter

Given a filter H(z)=B(z)/A(z) write a function isMinimumPhase (bz,ap) such that:

1. If bz and ap are row vectors, it considers them as the coefficients of the difference equations b and a
2. If bz and ap are column vectors, it considers them as the zeros and the poles, respectively
3. It returns -1 if the inputs are not correct (input must be either two row vectors, or two column vectors)
4. It returns 1 if the filter is minimum-phase and zero if it is not
5. Test the function in the aforementioned scenarios

Ex8: Overlap and Add Lowpass filtering

Given the 16kHz-sampled audiofile Toms_diner_16.wav

1. Load the file in the signal x
2. Design a FIR lowpass filter b with cut-off frequency=6,000 Hz
   1. use the function fir1 and M=32 weights
3. Process the signal x with the filter h=b using the Overlap and Add technique
   1. use block size N=400 choose the kind parameters of the window you prefer
4. Plot two subplots with the original signal x and the filtered version

Ex 9 multirate processing with rational factor

Given the 16kHz-sampled audiofile Toms_diner_16.wav

1. Load the file in the signal x
2. Create y from x with frequency rate =12 kHz
3. Plot two subplots with the original signal x and the modified version