

## ANALYTICS FOR INDUSTRIAL INTERNET 2018

### EXERCISE 1

#### Part A:

##### Design of IIR filter.

The file [https://abacus.abo.fi/ro.nsf/W/temp/\\$File/solo.dat](https://abacus.abo.fi/ro.nsf/W/temp/$File/solo.dat) contains a discrete-time sequence  $x(nT_s)$  representing an audio signal which has been sampled with the sampling frequency  $f_s = 48000$  Hz. The signal is corrupted with high frequency noise concentrated in the frequency region above approximately 4000 Hz. For comparison, the original noise-free signal  $\{s(nT_s)\}$  can be found in [https://abacus.abo.fi/ro.nsf/W/temp/\\$File/solo\\_c.dat](https://abacus.abo.fi/ro.nsf/W/temp/$File/solo_c.dat)

- (i) The high-frequency noise in  $\{x(nT_s)\}$  can be eliminated by filtering the signal using a low-pass filter. Determine the filter orders of the standard digital IIR filter prototypes required to achieve the filter specifications
  - stopband corner frequency: 4000 Hz
  - stopband attenuation:  $> 50$  dB
  - width of transition band: 500 Hz
  - maximum passband deviation: 0.05 dBSelect the filter having the lowest order, and plot its frequency response magnitude and phase.
- (ii) Compute the filtered signal  $\{y_f(nT_s)\}$  obtained by filtering the signal  $\{x(nT_s)\}$  using the filter in (i). Plot a sequence (for example for  $n = 1000 \cdots 1100$ ) of the filtered signal  $\{y_f(nT_s)\}$ , the noise-corrupted signal  $\{x(nT_s)\}$  and the original noise-free signal  $\{s(nT_s)\}$ . Determine approximately how much the filtered signal appears delayed due to phase shift of the filter. Is there any phase distortion due to nonlinear phase response of the IIR filter? Listen to the signals and comment on their differences.
- (iii) Compute the filtered signal  $\{y_{ff}(nT_s)\}$  obtained by filtering the signal  $\{x(nT_s)\}$  by zero-phase filtering using the filter in (i). Plot the filtered signal  $\{y_{ff}(nT_s)\}$ , the noisecorrupted signal  $\{x(nT_s)\}$ , and the noise-free signal  $\{s(nT_s)\}$  as in (ii). Comment on the result.

#### Part B:

##### Removal of periodic disturbance.

The file [https://abacus.abo.fi/ro.nsf/W/temp/\\$File/xp.dat](https://abacus.abo.fi/ro.nsf/W/temp/$File/xp.dat) contains an audio signal which has been sampled with the sampling frequency  $f_s = 8192$  Hz. The signal is contaminated with a periodic disturbance. Listen!

- (i) Determine the period and frequency components of the periodic disturbance.
- (ii) Design a filter which removes the periodic disturbance. Apply the filter to the audio signal, and listen to the filtered signal. Has the disturbance been satisfactorily removed?

#### Part C:

**Synthesis of notes with the plucked-string filter. (OPTIONAL)** Determine the parameters  $L$  and  $a$  of a tunable plucked-string filter which generates an audio signal with a specified fundamental frequency  $f_0$  (given in Hz) when the sampling frequency is  $f_s$  (Hz). Determine the transfer function from input signal  $x$  to output signal  $y$  of the tunable pluckedstring filter in Fig. 1.5 in the lecture notes. Use the above results to write a program which plays a given sequence of notes, where the damping  $R$  and the duration times are tunable. Test the program on the note sequence 165, 330, 392, 233, 220, 330 Hz. Use the note durations 0.7 sec with intervals of 0.5, 0.25, 0.25, 0.25, and 0.5 seconds between the notes.