APPLIED SIGNAL PROCESSING 2015

EXERCISE 1

A. Design of IIR filter.

The file `http://www.abo.fi/~htoivone/courses/sigbe/signal3.dat` contains a discrete-time sequence $\{x(nT_s)\}$ representing an audio signal which has been sampled with the sampling frequency $f_s = 11025$ Hz. The signal is corrupted with high-frequency noise concentrated in the frequency region above approximately 2750 Hz. For comparison, the original noise-free signal $\{s(nT_s)\}$ can be found in
`http://www.abo.fi/~htoivone/courses/sigbe/signal.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: 2750 Hz
- stopband attenuation: > 50 dB
- width of transition band: 250 Hz
- maximum passband deviation: 0.05 dB

Select 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 noise-corrupted signal $\{x(nT_s)\}$, and the noise-free signal $\{s(nT_s)\}$ as in (ii). Comment on the result.

B. Synthesis of notes with the plucked-string filter.

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 plucked-string 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.