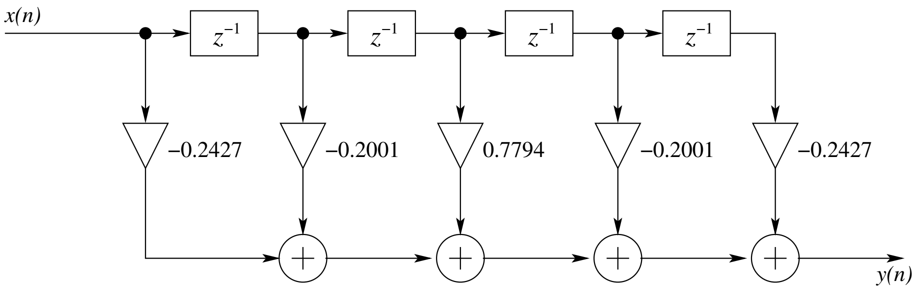
SGN-11007 Introduction to Signal Processing,
Exercise 4, 12.-14.9.2018

Pen & paper tasks should be done before the exercise session and Matlab tasks are done during the exercise session.

Task 1. (*Pen & paper*)

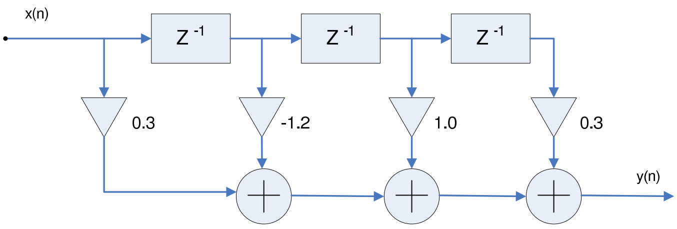
(a) What is the difference equation between the input $x(n)$ and the output $y(n)$ when the block diagram of the system is as shown below?

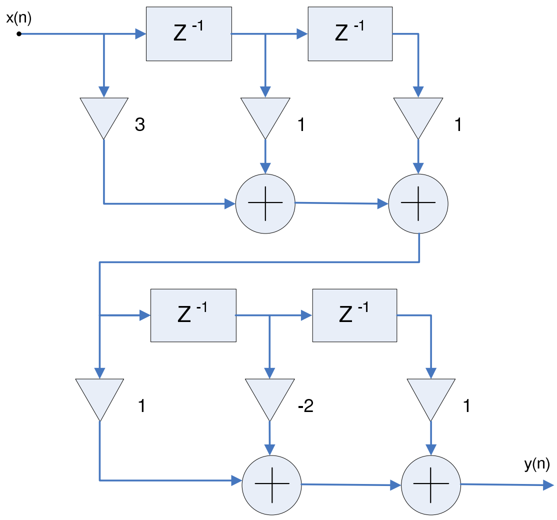


(b) Draw the block diagram for this system

$$y(n) = 0.11x(n) + 0.28x(n-1) - 0.01x(n-2) - 0.52x(n-3).$$

Task 2. (*Pen & paper*) What are the difference equations between the input $x(n)$ and the output $y(n)$ when the block diagrams of the systems are as shown below? The properties of convolution presented in Lecture 4 may assist in figuring out the right-hand system.





Task 3. *(Matlab)* Generate a one second long signal having frequency 1000 Hz with sampling rate 8192 Hz. The signal can be obtained using the formula

$$x(n) = \sin\left(\frac{2\pi n f}{F_s}\right),$$

where f is the desired frequency in Hertz and F_s is the sampling frequency in Hertz. In Matlab the variable n is a vector which contains the desired points in time, i.e., $(1, 2, 3, \ldots, 8192)$. Generate also signals having frequencies 2000 Hz and 3000 Hz and listen to all the results using the command `soundsc`. What happens when you exceed the Nyquist limit, that is, generate signals having frequencies 6000, 7000 and 8000 Hz?

Task 4. *(Matlab)* Let's simulate aliasing with Matlab.

Download the test signal `seiska.mat` from the course Moodle (`Ex_4.zip`) and load it to Matlab using the command `load`, which automatically loads it to the variable `x`. Listen to the original signal with the command `soundsc(x, F)`, where `F` is the sampling rate 16384 Hz. Decrease the sampling rate to half of the original using the command `y=x(1:2:length(x));` Thus, every second sample remains and the same result could have been obtained by taking samples initially at frequency 8192 Hz. As the signal actually includes frequencies up to 8192 Hertz, the high frequencies are aliased over the small ones. Listen to the result with the command `soundsc`. Compare that result to the one obtained by correctly removing too high frequencies (above 4096 Hz) before reducing the sampling rate (command `decimate`).

Task 5. *(Matlab)*

(a) Load the Matlab audio file `gong.mat` using command `load gong` to obtain signal `y` and sampling rate `Fs`. Familiarize yourself with the convolution command (`help conv`) and convolve the signal `y` and a vector formed from the coefficients of the Task 1 (a). Assign the result to the variable `z`. Listen to the original signal and compare it with the filtering result. The coefficients were selected in such a way that certain frequencies are removed in the convolution. Estimate which were removed frequencies.

(b) The impulse response may be human-designed (as in (a)) or measured e.g. in an echoing room. In the latter case, it models the acoustics of the room. Download the file `hall.wav` from the course Moodle (`Ex_4.zip`) and read it to Matlab using the command `audioread`. The file represents the impulse response of a large hall (and sounds like a single clap of hands in a church). Download also the file `seiska.mat`, and convolve it and the impulse response you downloaded. Listen to the signal before and after filtering. The result should sound like the speech was moved to a large hall.