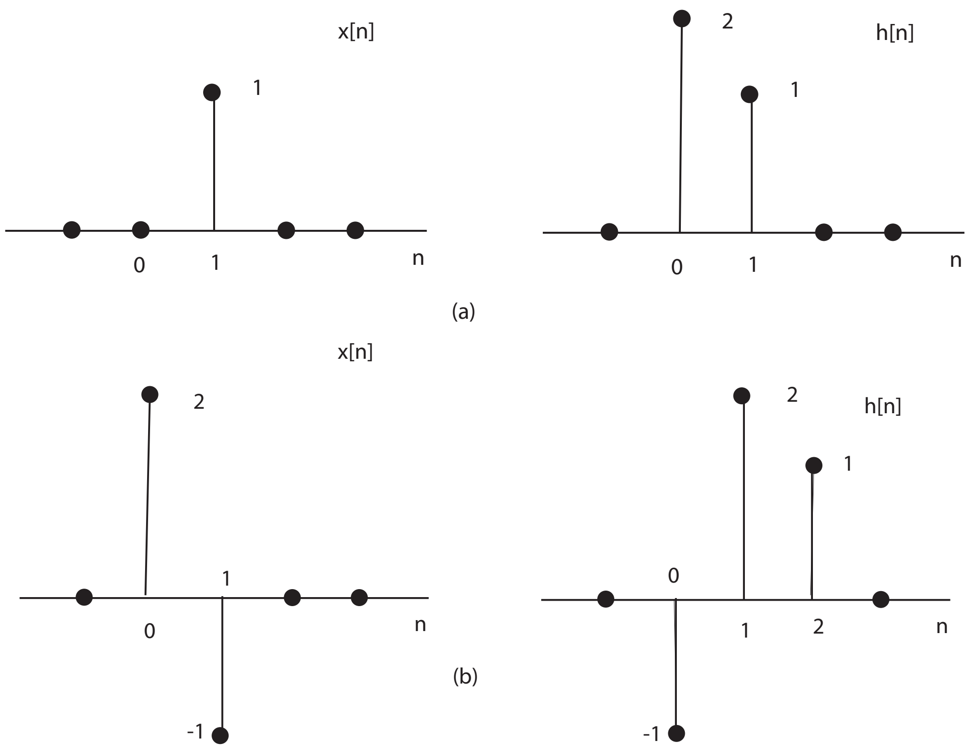
ECE 431
Digital Signal Processing
Homework 1: Discrete-Time Signals and Systems Review

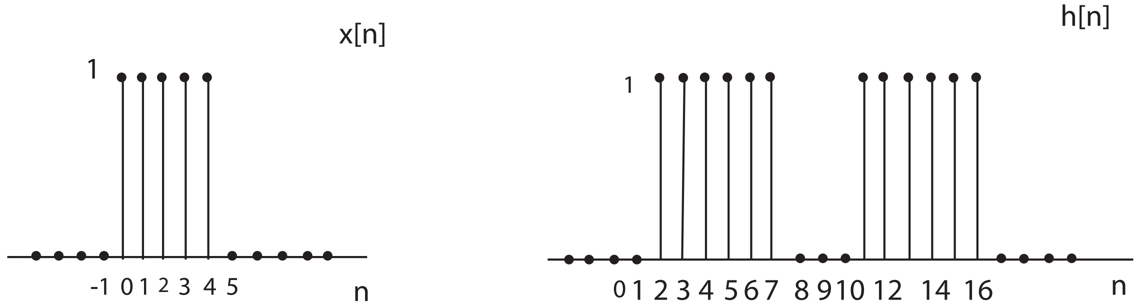
Due at beginning of class Friday, September 15, 2006

1. Oppenheim & Schafer with Buck (OSB) problem 2.22.

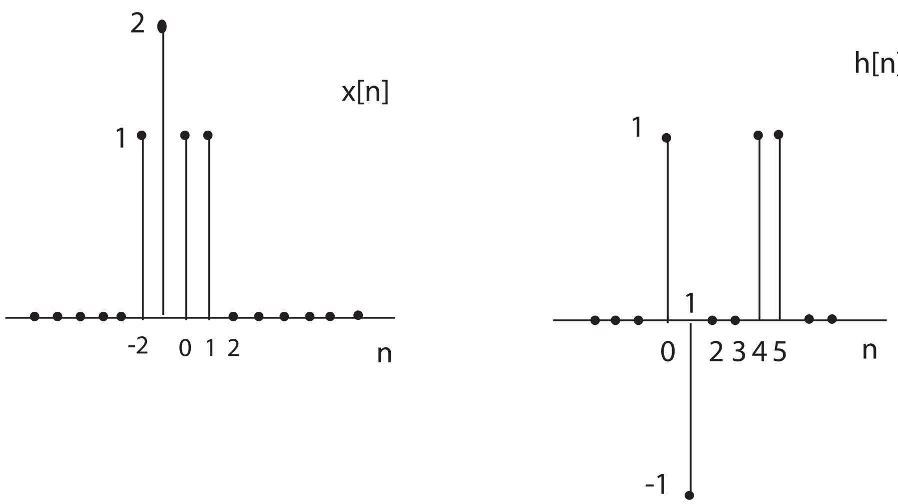
For each of the pairs of sequences in Figure $P2.22-1$, use discrete convolution to find the response to the input $x[n]$ of the linear time-invariant system with impulse response $h[n]$.

In addition to computing convolutions by hand, write your *own* Matlab function to compute the convolution of two finite-length sequences and plot the results and print out your Matlab code.





(c)



(d)

Figure P2.22-1

2. OSB 2.34

The input-output pair shown in Figure $P2.34-1$ is given for a LTI system.

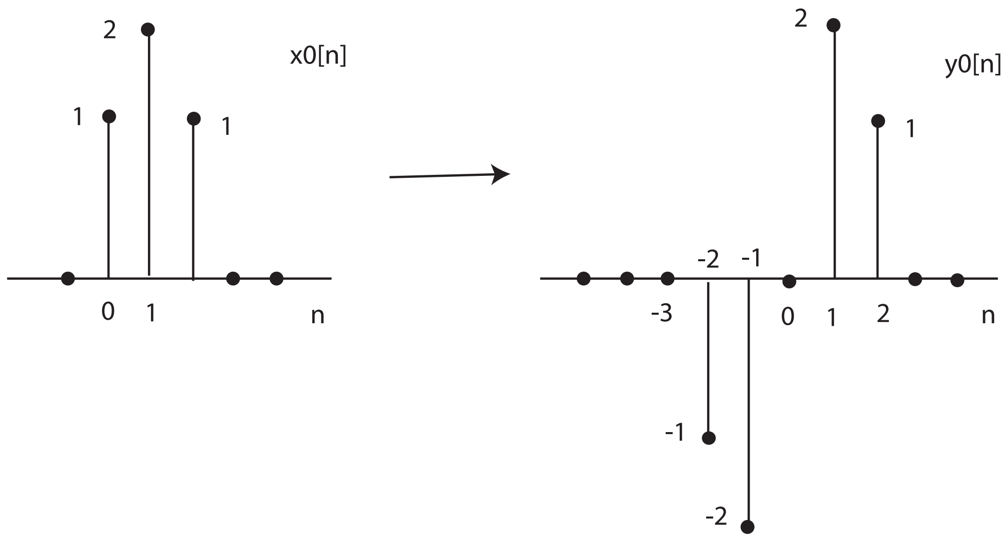


Figure P2.34-1

Determine the response to the input $x_1[n]$ in Figure $P2.34-2$.

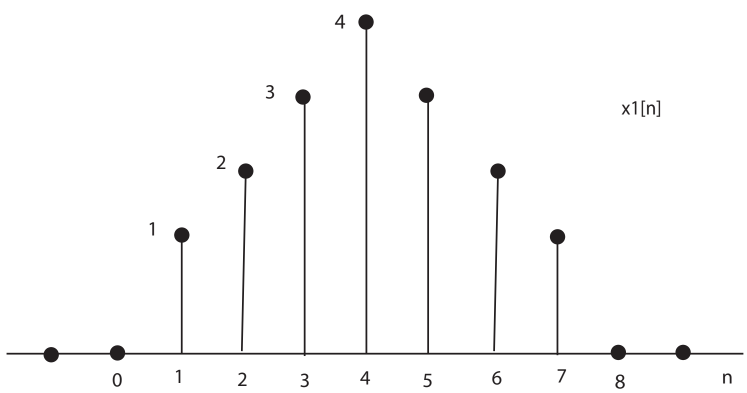


Figure P2.34-2

3. Consider an LTI system whose impulse response is $h[n] = (0.4)^n\, u[n]$, where $u[n]$ is the unit step function (i.e., $u[n] = 0$, for $n < 0$, and $u[n] = 1$, for $n \geq 1$).

a. Compute and sketch the magnitude of the frequency response of the system. Is the system highpass, bandpass, or lowpass?

b. Let $x[n] = \frac{1}{2}\cos(\pi n) + 2\cos(\frac{\pi}{2}n)$ be the input to the system. Compute the output $y[n] = h[n] * x[n]$.

c. Suppose instead that $x[n] = \delta[n] + 1/2\delta[n-1] - 1/4\delta[n-3]$ is the input to the system. Compute the output $y[n] = h[n] * x[n]$.

4. Digital Music Processing. Download the Matlab data file
`http://www.ece.wisc.edu/~nowak/ece431/mclips.mat`
which includes two music clips, and the Matlab program
`http://www.ece.wisc.edu/~nowak/ece431/music.m`
The program will playback these clips for you The clip x is the original, and the clip y has been a digitally processed (in Matlab) to produce an "echo" effect.

a. Design and implement a *linear* digital filter in Matlab that processes the original clip x to produce an echo with a 0.1 second delay (this should sound similar to y). Turn in the derivation for your echo filter and your Matlab implementation code.

b. Design and implement a *nonlinear* digital filter in Matlab that produces a saturation effect (i.e., distortion). Specifically, mimic the effect that pushing an amplifier into saturation would have by "clipping" x whenever the magnitude of its amplitude is greater than a certain level (e.g., $|x[n]| > 0.7$).

5. The DTFT of a discrete-time signal/system $h[n]$ is given by

$$H(\omega) = \sum_{k=-\infty}^{\infty} h[k]e^{-j\omega k}$$

and the inverse DTFT is

$$h[n] = \frac{1}{2\pi}\int_{-\pi}^{\pi} H(\omega)e^{j\omega n}d\omega$$

Prove that the inverse DTFT formula is valid. Hint: Replace $H(\omega)$ by the DTFT formula.