EE123 Spring 2018 Miki Lustig
Digital Signal Processing

Assignment 7

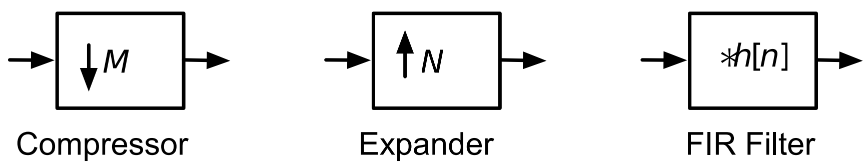
Due March 21^{st} 2018

1. Self-grade Homework 6.

2. Oppenheim and Schafer Exercise 4.61

3. *From Midterm II, sp'11:*
   Design a system that takes a band-limited discrete time input $x[n]$ sampled at a rate $f_{s,1} = 1$ sample/second, and outputs the same signal sampled at $f_{s,2} = 1.5$ samples/second.

   The processing blocks you have are shown below, you can choose to connect them in any order, but you can use each block only once.

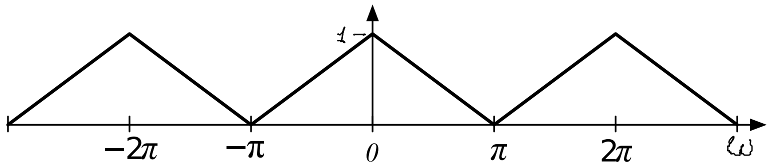


   These are

   - A compressor that reduces the sampling rate by a factor of M. It keeps one out of every M samples, and throws the others away.
   - An expander that increases the sampling rate by a factor of N. It inserts $N-1$ zeros after every input sample.
   - An ideal filter, for which you must specify the frequency response.

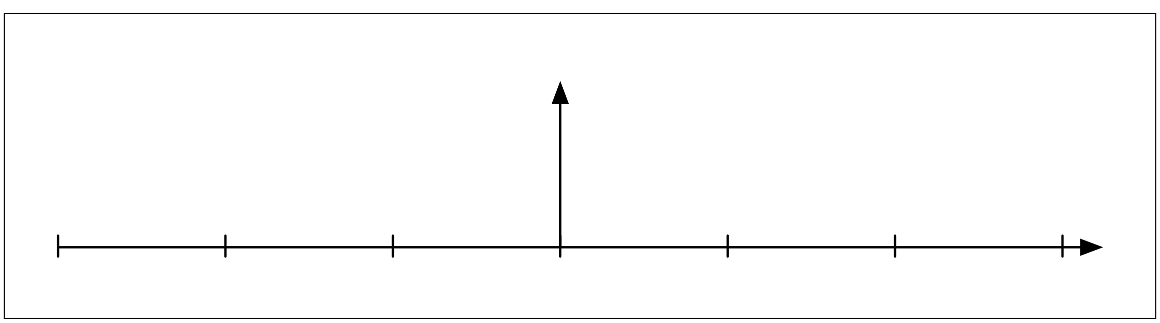
   a) Draw a block diagram of your system, and specify M, N and $H(e^{j\omega})$:

b) The input spectrum looks like this

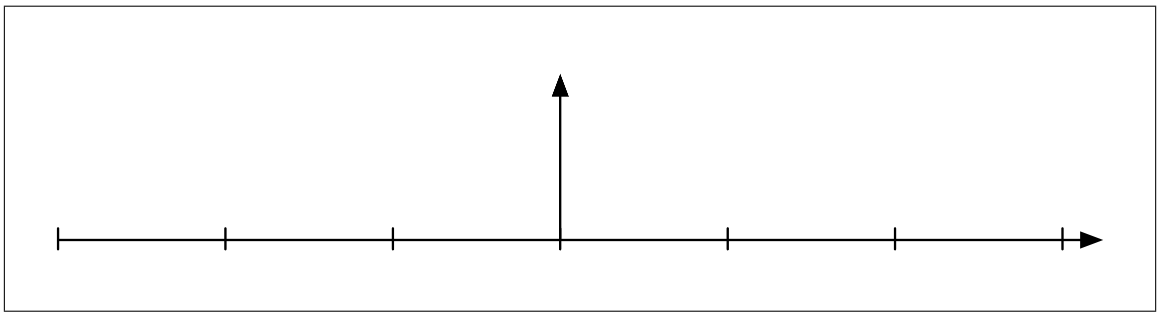


Sketch the output spectrum after each block of your system. Label your axes. Use radian frequencies ω. If aliasing occurs, overlay it with dashed lines.

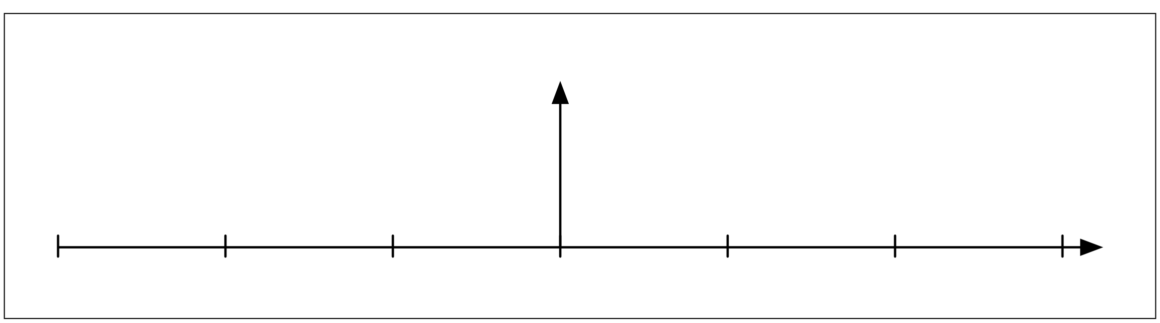
i) Spectrum after Block 1:



ii) Spectrum after Block 2:



iii) Spectrum after Block 3:



c) Find the impulse response $h[n]$ of the filter. You can assume it is ideal instead of an FIR.

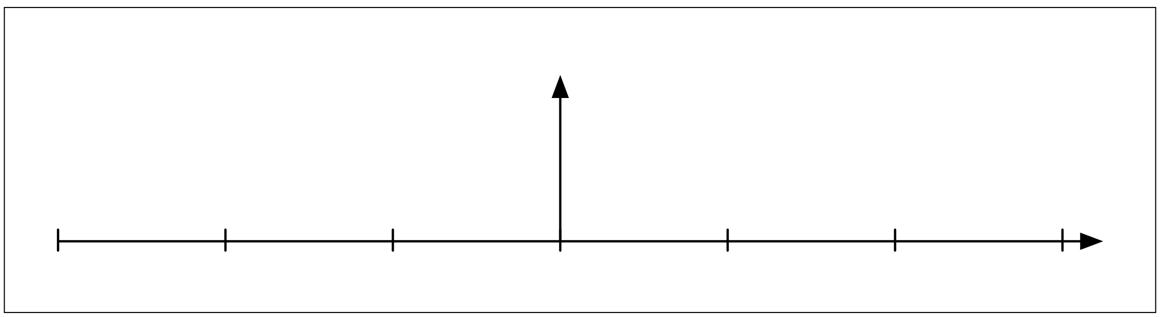
$h[n] =$

d) Instead of using the ideal $h[n]$ FIR filter, you decide to use a triangular window (linear interpolation) with a gain a.

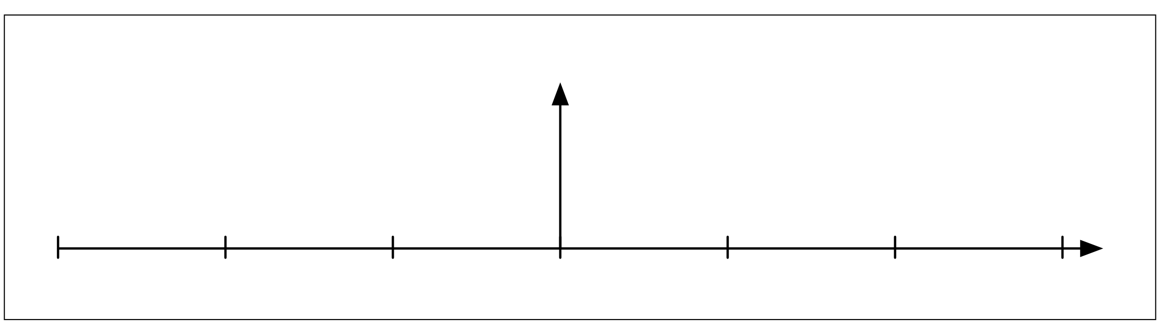
$$h[n] = \begin{cases} a(1 - |n/N|) & |n| < N \\ 0 & \text{otherwise} \end{cases}, \qquad H(e^{j\omega}) = a\frac{\sin^2(\frac{\omega}{2}(N+1))}{(N+1)\sin^2(\frac{\omega}{2})}$$

Sketch the output spectrum after the second and third blocks of your system. Label your axes. If aliasing occurs, overlay it with dashed lines.

i) Spectrum after Block 2:

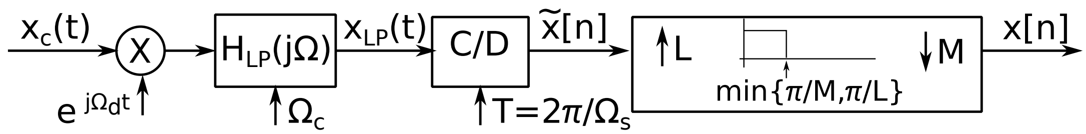


ii) Spectrum after Block 3:

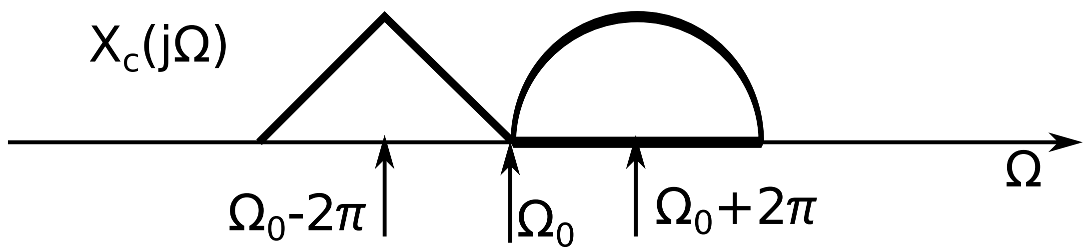


4. *From Midterm II, sp'14:* Continuous to Digital

Consider a continuous to digital system that consists of a demodulation by Ω_d, ideal analog low-pass filter with a cutoff Ω_c, ideal C/D (without LP filter) with a sampling rate $T = \frac{2\pi}{\Omega_s}$, and an ideal digital $\frac{L}{M}$ resampling (with an appropriate $\min\{\frac{\pi}{M}, \frac{\pi}{L}\}$ LP filter).



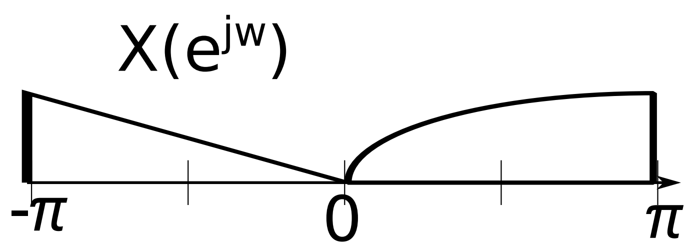
Also consider a signal with an unknown gain, $x_c(t)$, with the following spectrum, $X_c(j\Omega)$:



For each of the following possible outputs of the system, find the parameters Ω_d, Ω_c, Ω_s and L/M that will produce it. Also, plot qualitatively the intermediate spectrums $X_{\text{LP}}(j\Omega)$ and $\tilde{X}(e^{j\omega})$. It is possible that a parameter combination that produces the output does not exist. In that case, mark the appropriate box and explain why.

The analog H_{LP} filter has limits: $2\pi \leq \Omega_c \leq 8\pi$. When given a choice, choose the **lowest** Ω_c

a) The spectrum $X(e^{j\omega})$ is:



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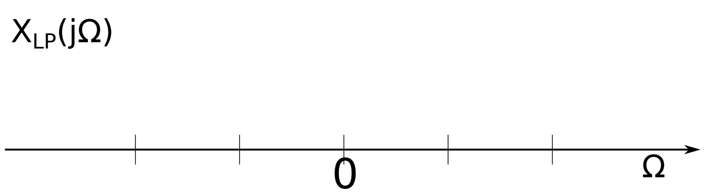
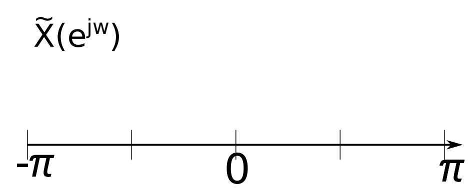
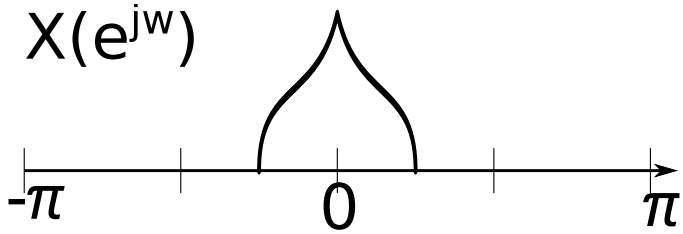


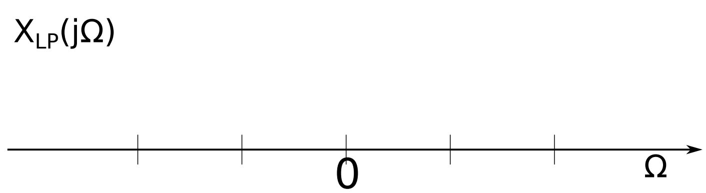


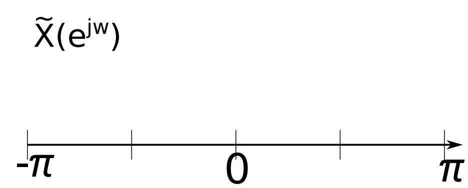
Parameters

$\Omega_d =$	$\Omega_c =$	$\Omega_s =$	$L/M =$	N/A ☐

b) The spectrum $X(e^{j\omega})$ is:



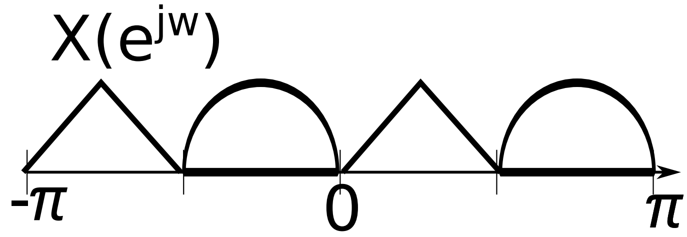


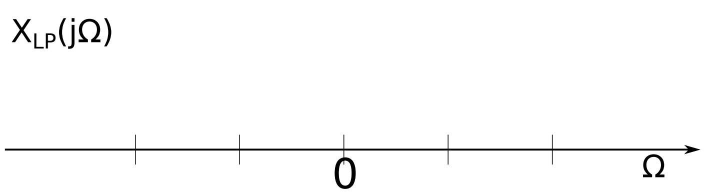


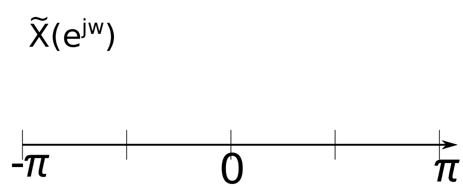
Parameters

$\Omega_d =$	$\Omega_c =$	$\Omega_s =$	$L/M =$	N/A ☐

c) The spectrum $X(e^{j\omega})$ is:



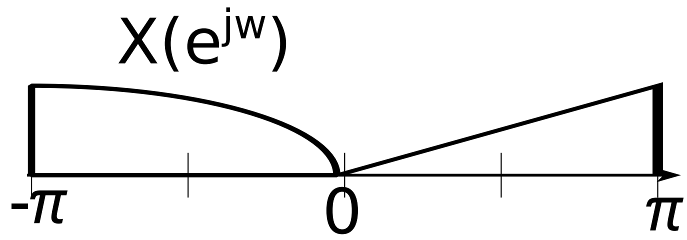


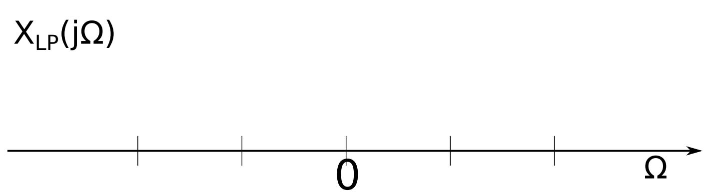


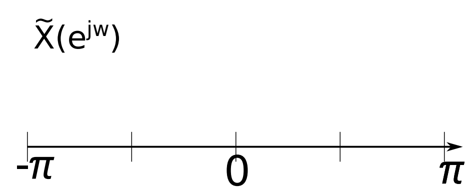
Parameters

$\Omega_d =$	$\Omega_c =$	$\Omega_s =$	$L/M =$	N/A ☐

d) The spectrum $X(e^{j\omega})$ is:







Parameters

$\Omega_d =$	$\Omega_c =$	$\Omega_s =$	$L/M =$	N/A ☐