Signal Processing - Spectral Analysis

All questions are weighted the same!

A digital filter H is implemented with the sampling frequency f_s=16 kHz. It has the impulse response $h[n]$ with the length N=4. Using a DFT the figure below gives the frequency response of the filter H (see figure 1).

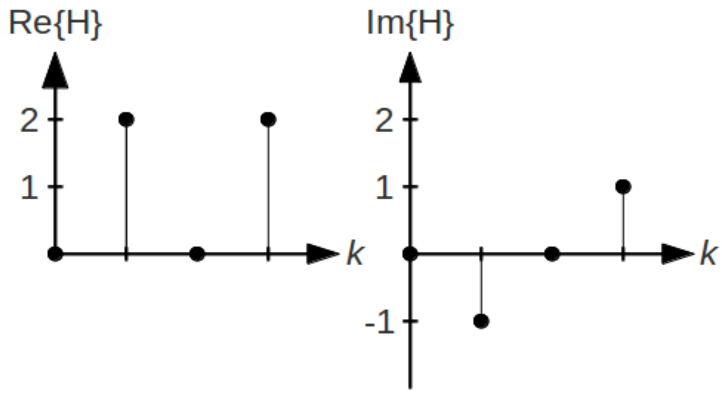

Figure 1. The frequency response of the digital filter H.

1) Find the time sequence $h[n]$.

2) Put frequency numbers/values (for each k) on figure 1.

We now consider a continuous sinusoidal signal with the frequency f=2 kHz, i.e. $s(t)=\cos(\Omega t)$, where $\Omega=2\pi f$ and t is the time. This signal is sampled at f_s and we call the sampled signal $x[n]$.

3) Find $x[n]$ when we use a rectangular window with the length L=4.

The sampled sinusoidal signal $x[n]$ is now sent through the filter H and the output we call y[n].

4) Find $y[n]$.

At the first glance, figure 1 suggests that no frequency around 2 kHz is let through the filter H, but $y[n]$ is <u>**not**</u> as we might expect a zero-sequence.

5) Explain why $y[n]$ is not just a sequence of zeros.

6) Compute the gain of the filter H at 2 kHz.

7) Sketch the filter H's continuous amplitude response $|H(\omega)|$ and include frequencies on the x-axis and amplification on the y-axis.

The continuous signal $s(t)$ is now sampled again at f_s but this time we use a Blackman window with the length L=128, and secondly we analyze the outcome using a 128-points DFT.

8) What is the frequency resolution (distance between the ks) using such an analysis and what frequency does k=117 correspond to? If we wish to find the amplitude at 2 kHz, what value must k then take?

9) What is the “effective frequency resolution” of the analysis, i.e. its ability to separate nearby frequencies?