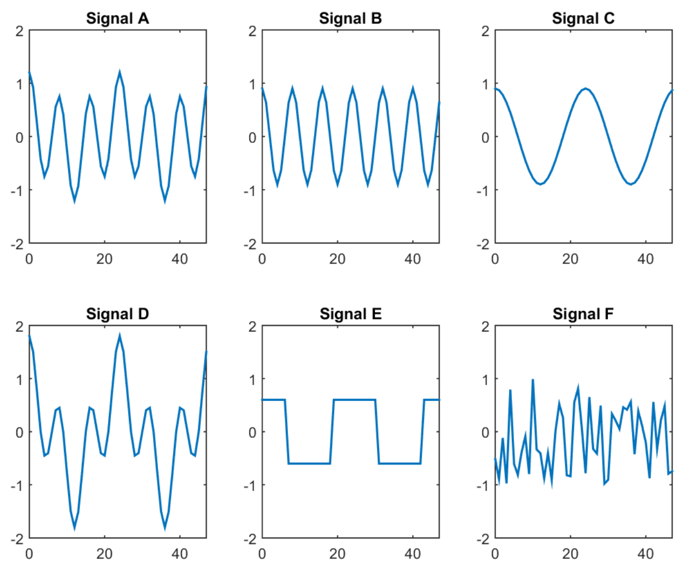
FREQUENCY DOMAIN AND FILTERING

SECTION 2 QUESTION 1 (1/2 points)

Each signal (labelled A to F) shown in the figures below, consists of 48 samples. The horizontal axis is the sample index (n).



The figures below (labelled 1 to 6) show the amplitude spectra of the Fourier Series expansions of these signals, but in random order. The horizontal axis is frequency index (k), which runs from 0 to 24.

Modulation

Topic 10: Summary and Review

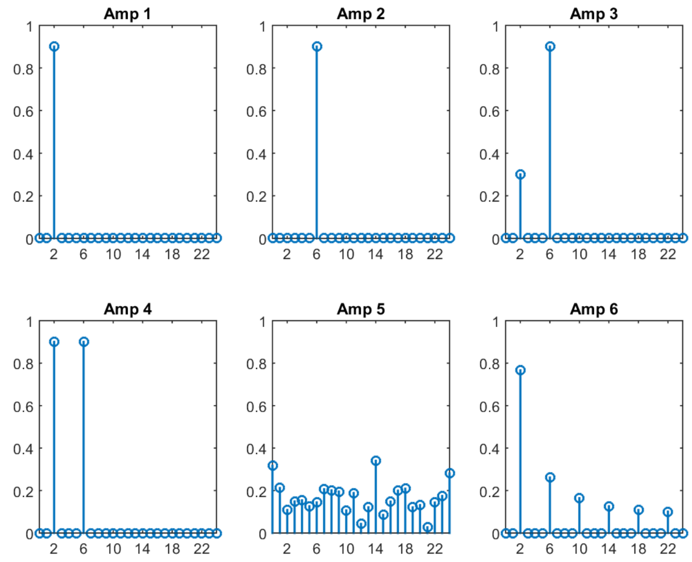
Final Exam

Final Exam
Final Exam due Dec 07, 2015 at 16:00 UTC

MATLAB download and tutorials

MATLAB Sandbox

Post Course Survey



For each signal, find the corresponding amplitude spectrum (Amp 1 - 6) and enter its numerical label in the space provided.

Signal A:

3

3

Signal B:

4

4

Signal C:

1

1

Signal D:

5

5

Signal E:

6

6

Signal F:

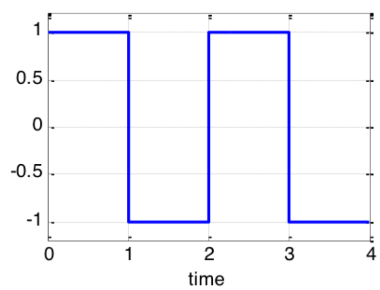
2

2

You have used 2 of 2 submissions

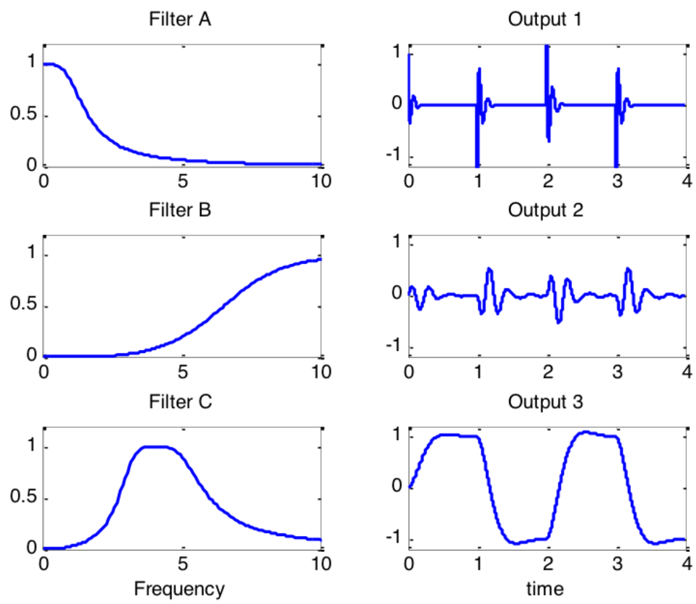
SECTION 2 QUESTION 2 (0.67/2 points)

Suppose that the waveform shown below is input to three different filters. The horizontal axis shows time in seconds. The vertical axis shows signal amplitude.



The amplitude responses of the three filters, labelled by A, B and C are shown on the left hand side of the figure below. The horizontal axes of these plots show frequency in hertz. The vertical axis is the filter amplitude response. The outputs of these filters, labelled by 1, 2 and 3, are shown on the right hand side of the figure below *in random order* (i.e.

Output 1 is not necessarily the output of Filter A). The horizontal axes of these plots show time in seconds, and the vertical axes show signal amplitude.



For each filter, find the integer label of its corresponding output, and enter the number in the space provided below.

Filter A:

2

2

Filter B:

1

1

Filter C:

3

3

You have used 1 of 1 submissions

Consider the continuous-time signal

$$x(t) = 4\cos(2\pi(100)t) + 6\cos(2\pi(500)t) + 2\cos(2\pi(2000)t)$$

where t is measured in seconds.

Suppose that $x(t)$ is sampled starting at t = 0 at a sampling frequency of F_s = 5kHz for $N = 100$ samples. Let $x(n)$ be the sampled data waveform, which can be expressed as

$$x(n) = A_o + \sum_{k=1}^{50} A_k \cos(2\pi f_k n) \qquad \text{for } n = 0, \ldots, 99$$

where

$$f_k = \frac{k}{N}$$

is the normalized frequency in units of cycles per sample.

SECTION 2 QUESTION 3 PART A (2/2 points)

Calculate the length of time (in seconds) that the waveform was sampled.

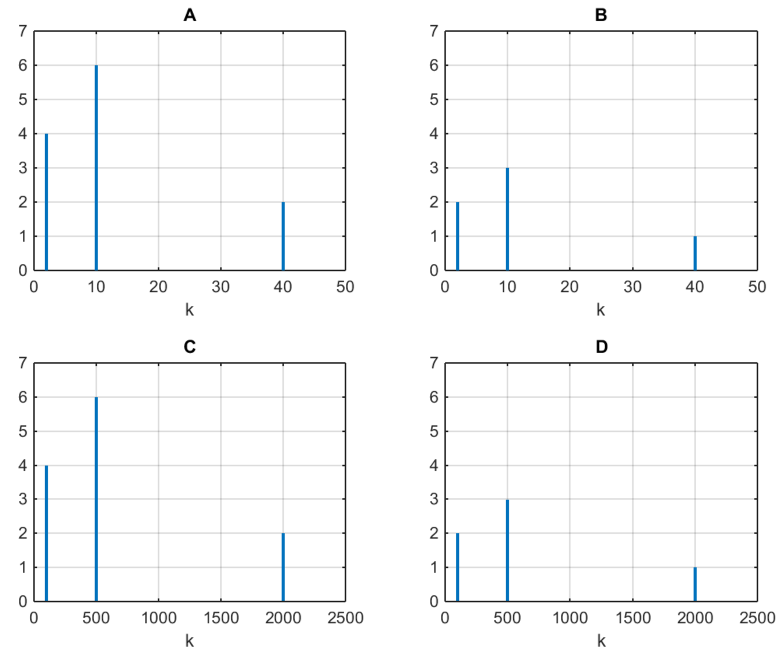
0.02

0.02

You have used 1 of 1 submissions

SECTION 2 QUESTION 3 PART B (2/2 points)

Which of following figure shows the correct amplitude spectrum of the Fourier series expansion of x(n)? The horizontal axis shows frequency index k. The vertical axis shows the amplitude of the corresponding Fourier component, A_k.

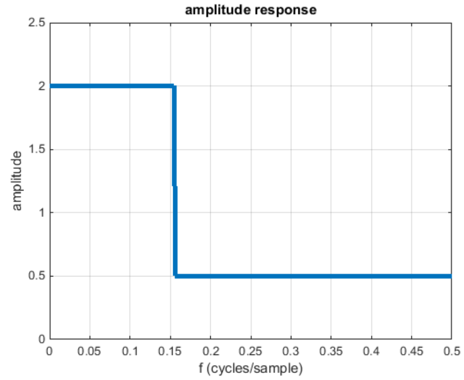


- A ✔
- B
- C
- D

You have used 1 of 1 submissions

SECTION 2 QUESTION 3 PART C (1.33/2 points)

Suppose $x(n)$ is input to a discrete time channel with amplitude spectrum shown below.



where f is the normalized frequency in cycles/sample.

Denoting the output by $y(n)$, which can be written as the sum of three frequency components,

$$y(n) = \sum_{i=1}^{3} B_i \cos(2\pi f_i n)$$

where f_i are the normalized frequencies, and these components are numbered in order of increasing frequency, where f_1 is the lowest and f_3 the highest. Give the magnitude and frequency of these components. Give the amplitudes to three decimal places.

B_1 =

8.0

8.0

f_1 =

0.01

0.01

B_2 =

12.0

12.0

f_2 =

0.1

0.1

B_3 =

20

20

f_3 =

0.4

0.4

You have used 1 of 1 submissions

SECTION 2 MATLAB QUESTION (1/1 point)

In this exercise, your task is to characterize the frequency response of an unknown channel, which is simulated by the function **channel_f2_2015**. To do that, you have to transmit a set of sinusoidal functions through the channel, and measure the amplitude of the sinusoid at the output.

In the initial code, we initialize the variable **flist**, which contains a list of 100 **normalized** frequencies at which we wish to measure the amplitude response. The code then initializes the variable **h_rx**, which also has length 100. This is intended to hold the peak-to-peak amplitudes of the responses of the channel to input cosine waves with the frequencies stored in **flist**. To be precise, **h_rx** should be a vector whose **i**-th element contains the peak-to-peak response of the channel to a cosine function with amplitude 1, length **nsamp** samples, and frequency **flist[i]**. However, in this initial code, this variable is not modified after initialization. We leave it to you to fill in the rest.

The next part of the code creates an input signal of length **nsamp** containing all zeros, and sends it through the channel. This is just to illustrate the use of the function **channel_f2_2015**. Feel free to modify this part of the code as necessary. The final part of the code is used to plot the frequency response of the channel in Figure 1. To do that, we assume that the amplitudes of the testing functions are 1, hence we divide the peak-to-peak response by 2 (see Lab 3).

Your task is to create cosinusoidal functions used to characterize the channel, send them through the channel, analyze the peak-to-peak amplitude of the received signals and store them inside the variable **h_rx**. When you compute the peak-to-peak amplitude, be reminded to skip past the transient response of the channel, which you can assume to last for 200 samp You are free to use any of the functions used in the previous labs.les.

Modify the code between the lines

% % % % Revise the following code % % % %

and

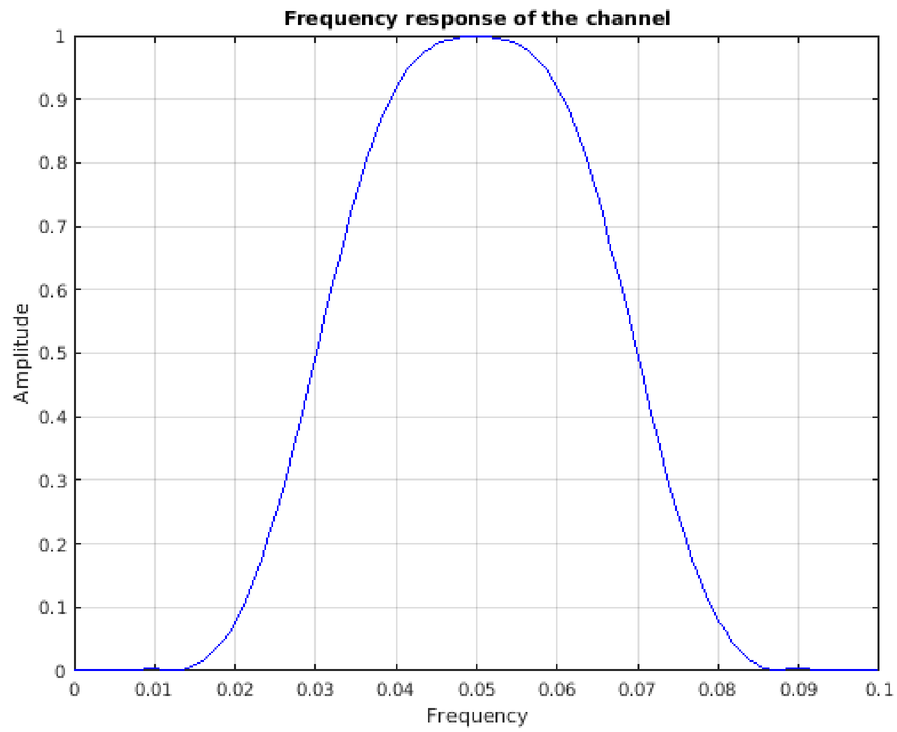
% % % % Do not change the code below % % % %

Please, do not change other parts of the code. Once you have successfully checked your work, inspect Figure 1 and use the insight gained to answer the questions below. To get credit for the MATLAB part, you must compute the vector **h_rx** correctly.

```
1 %% code
```

Correct

Figure 1



You have used 1 of 5 submissions

SECTION 2 MATLAB QUESTION PART A (1/1 point)

What type of filter is the channel?

- Low Pass
- High Pass
- Band Pass ✔
- Band Stop

You have used 1 of 1 submissions

SECTION 2 MATLAB QUESTION PART B (1/1 point)

We define the frequency that characterizes a channel according to the type of filter the channel implements. For a low/high pass filter, this frequency is the 3dB cut-off frequency. For a band pass/stop filter, this frequency is the center frequency, i.e. the frequency at which the amplitude response achieves its maximum value for a band pass filter, or its minimum value for a band stop filter.

By visual inspection, what is the frequency that characterizes the above channel (in units of normalized frequency)?

0.05

0.05