Lab 1 Preparation Using Simulink

Double Sideband Modulation With and Without Carrier

You should already know that functions suchas adders and multipliers are easily implemented in the Simulink software. However this is not the case in the real world. The Simulink is only a tool to predict what you expect will happen in the actual implementation of your design.

(**Please note:** that we are restricted to some license issues. So do not stress when you cannot open a block set. If you cannot open the block set at that time, that means all the licenses are used up. Make sure you do your prelab way before your prac so that you are sure that you will finish the prelab before the lab. **YOU WILL NOT BE ALLOWED TO ATTEND THE PRAC if your PRELAB IS NOT DONE before the lab and shown to one of the tutors running the lab**).

Objectives:

To investigate amplitude modulation techniques, frequency translation, synchronous and asynchronous detection.

1. **Modulation**

   1.1 Double Sideband Suppressed Carrier (DSB-SC)

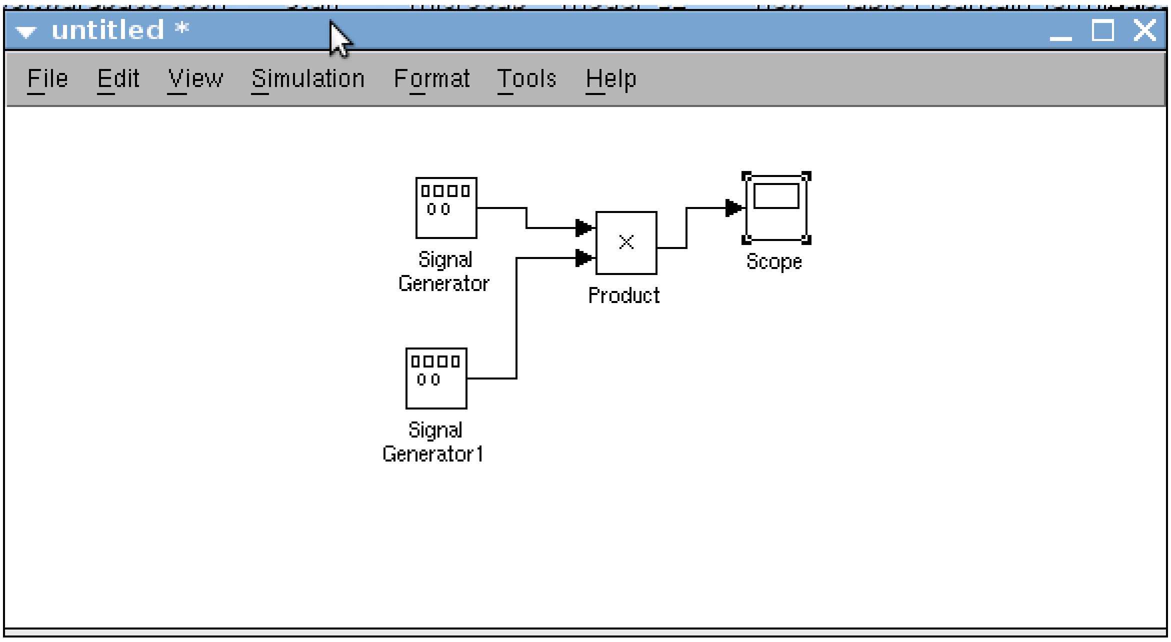

Figure.1

You will have to set up a DSB-SC modulator in Simulink. Your design will look similar to Fig.1. First set up the blocks as seen below and set the following parameters:

- First you need to set the system time. Set sample rate to 200kHz (this should be high enough to satisfy the Nyquist requirement). So sample time will be 5e-6 seconds.
- **Modulating signal: Frequency:** 1kHz, amp 1V (2V p-p) and leave the rest of the parameters as they first appeared. (the Signal generator (SG) on top)
- **Carrier signal:** Frequency 20kHz, amp 1V (2V p-p), (the SG at the bottom)

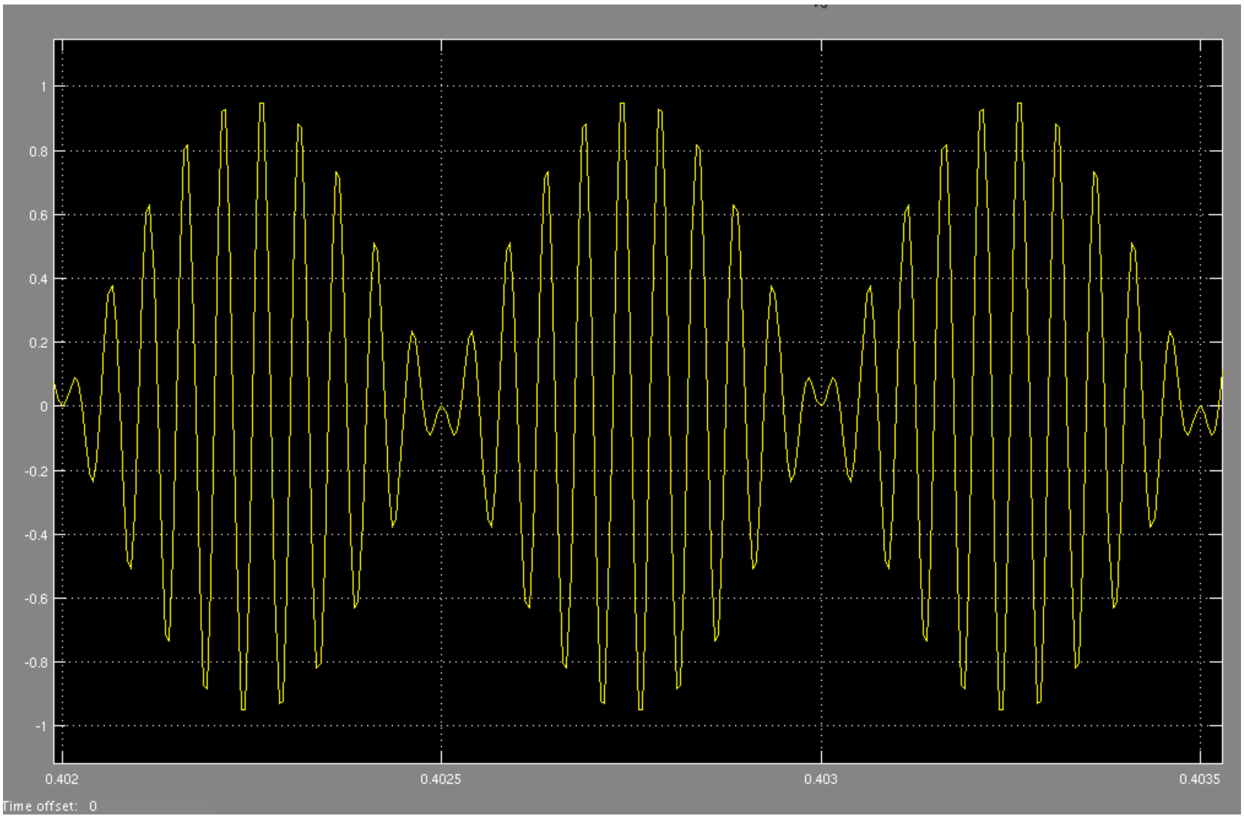

Figure.2

The output of the multiplier should look something like the graph in figure.2. When using zoom in the analysis window.
Save your file with a suitable filename.

1.2 Double Sideband Large Carrier (DSB-LC)

DSB-LC can be obtained by adding some DC voltage to the modulating signal.
First set up the tokens as seen in the figure, then set up the following parameters of the indicated block:

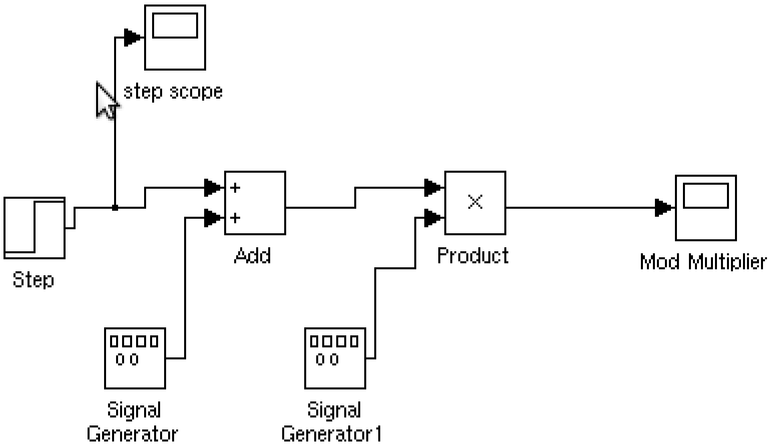

Figure.3

- Use the same system time as 1.1
- **Step Function generator:** start at 0 sec, Initial value - 2, Final value – 2.
- **Modulating signal:** frequency 1kHz, Amp 1V (2V p-p)
- **Carrier Signal:** Frequency 20kHz, Amp 1V (2V p-p)

By adjusting the amplitude of the step function you are changing the DC voltage and hence the modulation index. Try setting the amplitude of the step function (DC) to 0V. What do you observe in the modulated signal, what is the modulation index? What DC value would give you a modulation index of exactly 1?

When you actually build this circuit, you will be given an already built AD633 multiplier module which allows you to change the modulation index by adjusting a potentiometer.

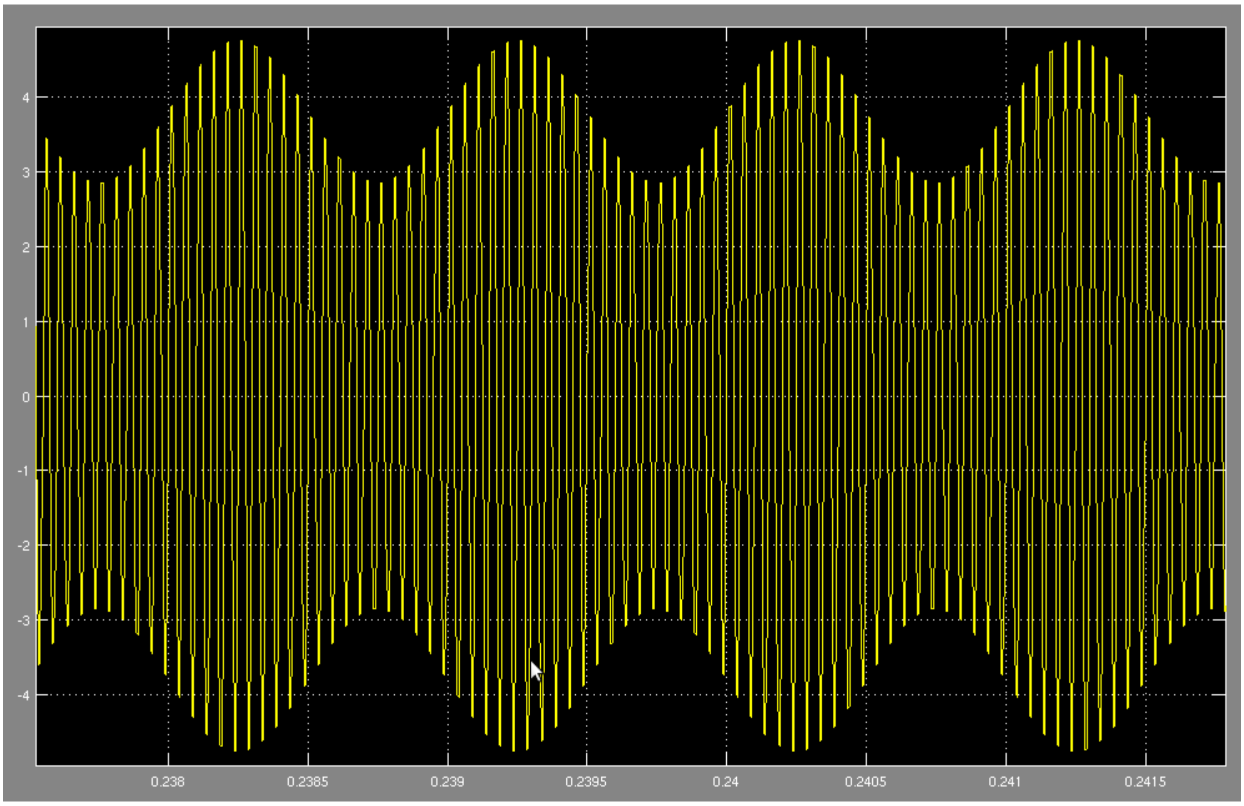

Figure.4

When the step is set to 4V, the output of the multiplier should look like fig.4 in the analysis window (zoom in to see the graph). Save your file with a suitable filename.

2. Demodulation

2.1 Synchronous Demodulation

NOTE: change the stop time to 10ms from configuration parameters

It can be shown mathematically that when a modulated signal is multiplied with the same carrier used in modulation, the original signal can be recovered by eliminating the high frequency components using an ideal LPF.

We could use Butterworth filters available to us in Simulink.

Open the file you saved in section 1.1 (DSB-SC modulator) then setup the tokens as seen in Fig.5 and connect the output of the modulator to the multiplier as shown in Fig.5.

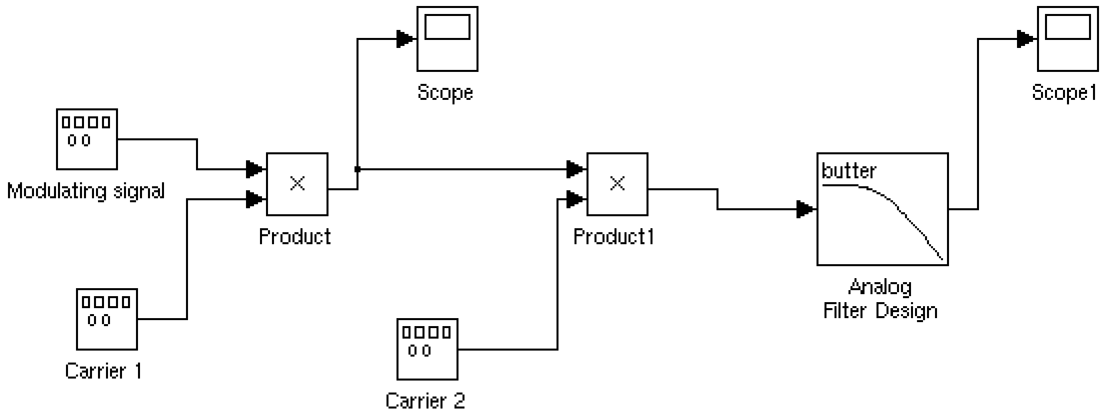

Figure.5

Then set the following parameters of the indicated token:

- **Carrier Signal**: Frequency 20kHz, Amp 1V (2V p-p).
- **Low Pass filter:** should have a cut off just above 1 kHz, but we can put it at 2kHz (still far away from the carrier) and an order of 1.
  NOTE: In the lab, one will have to build a RC LPF. The cutoff is at 1/(2*pi*RC).

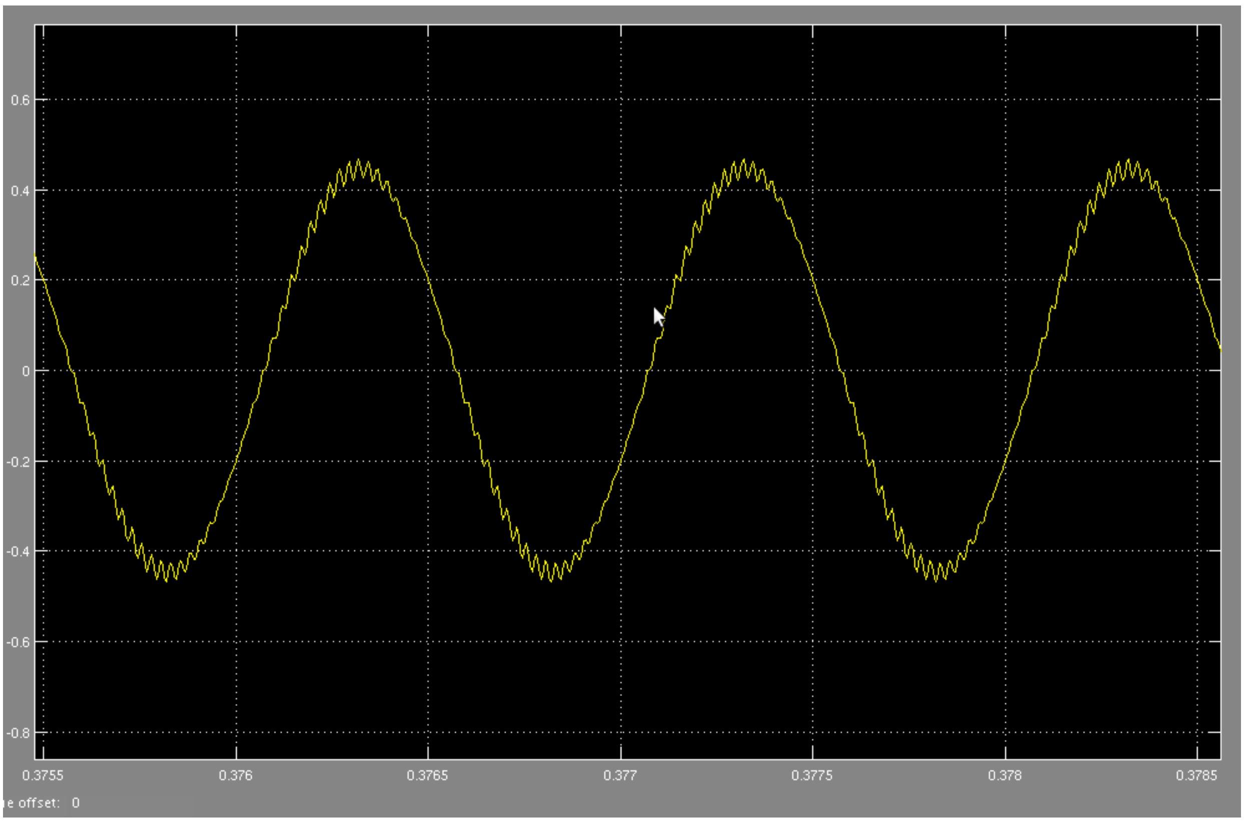

Figure.6

Because the LPF is so crude you cannot recover a perfect replica of the modulating signal (as seen in Fig.6).

2.2 Asynchronous Demodulation

Open the file you saved in section 1.2 and add blocks for Asynchronous Demodulation shown in figure 7.

Note: You could initially set the "spectrum scope" buffer size to 1024 samples. The buffer will then store a duration T=1024*5E-6 = 0.00512 sec, which implies a frequency resolution of 1/T = 195Hz. This will probably provide sufficient resolution to see the sidebands nicely. If not, then increase the total simulation time and the buffer size (which must be a power of two e.g. 2048, 4096 etc.)

Set the following parameters of the indicated token:

- Gain: Can be found under "common used blocks". Set the gain to 3.
- Envelope Detector: in our case, we will take the absolute value of the signal.
- LPF: cut off at 2kHz and a 4th order filter.

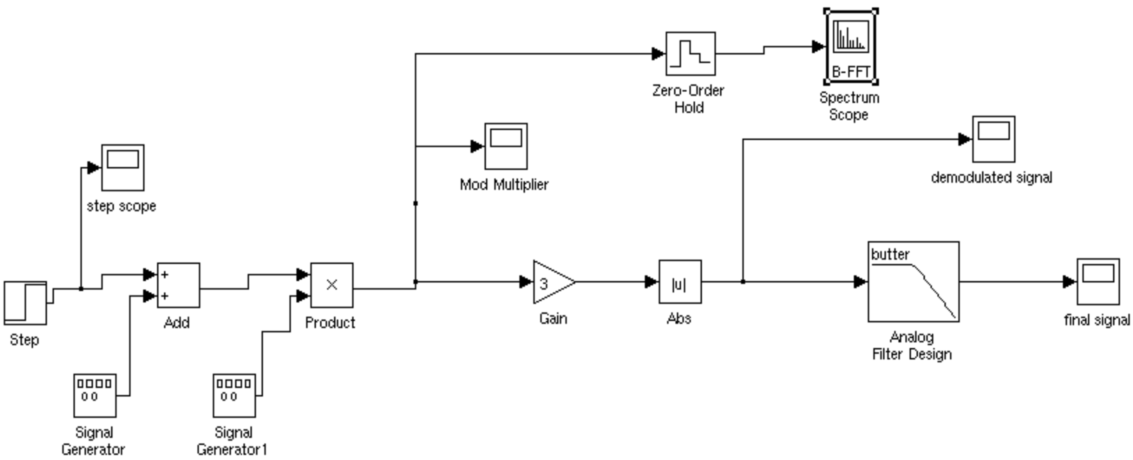

Figure.7

The output of the filter should look like the one displayed in figure.8 when the step function is set to 2V.

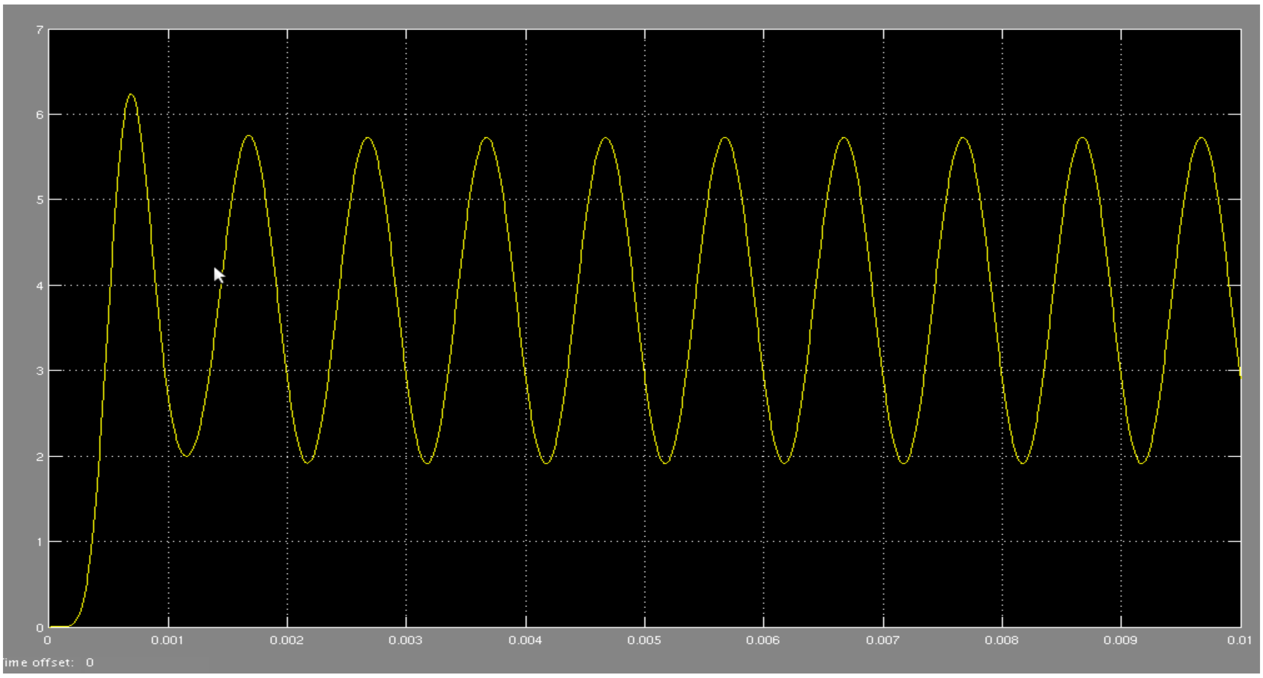

Figure.8

Questions:

Can you think of a way to centre the final output at 0V??

This type of demodulation will work for DSB-LC but not for DSB-SC, can you see why?

To see how (in)effective this type of demodulation works on DSB-SC set the step function to 0V and observe the output of the LPF.

2.3 View Signal in the Frequency Domain.

View the Spectra of the various signals in the Frequency Domain(using 'FFT')
NOTE: One may need to simulate for a longer length of time to obtain nice looking FFT (adjust buffer values) – HINT! Use frequency resolution to work it out

Figure 9 shows the FFT on the output of the multiplier. (do it for more points at the output)

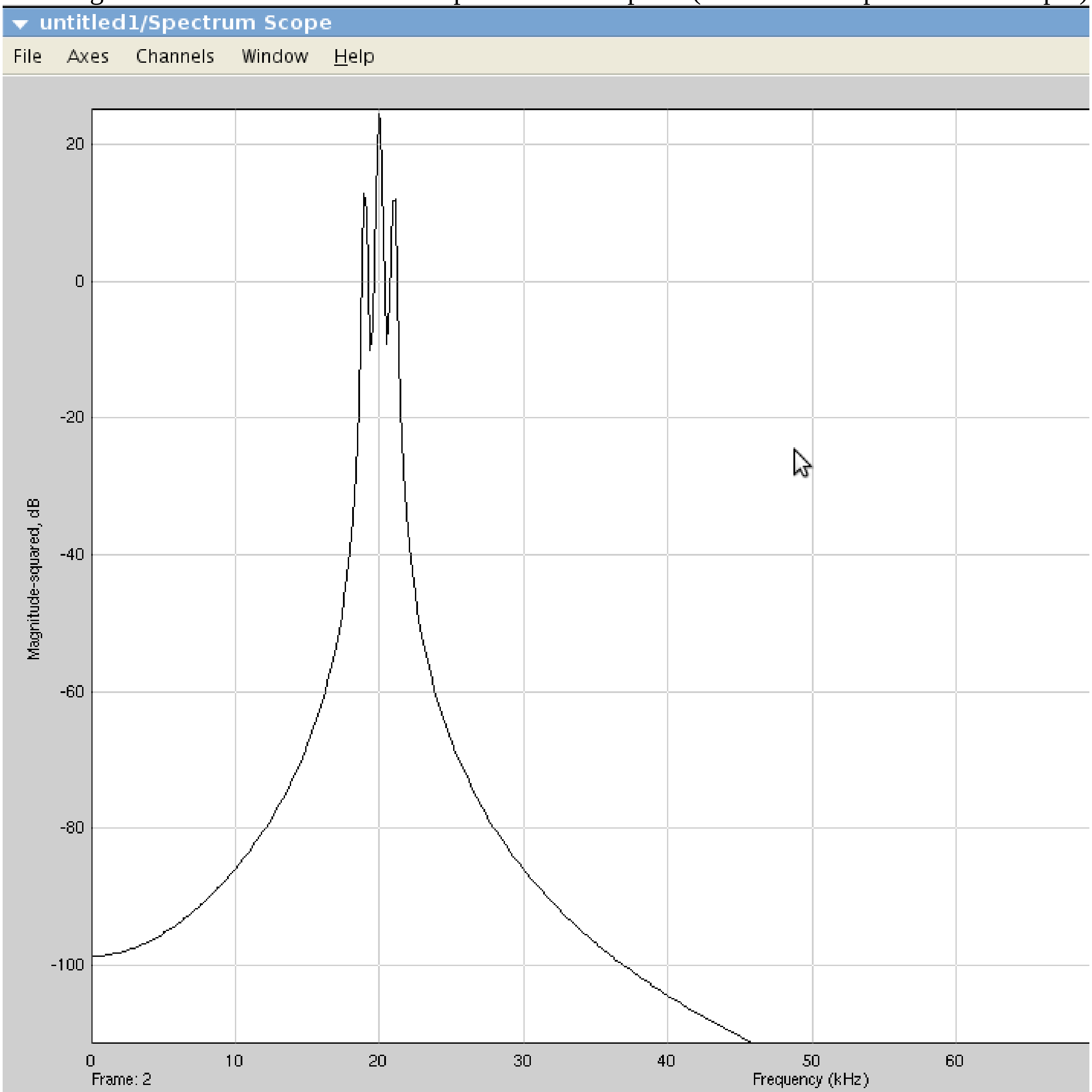

Figure.9