

## Lab 3 Report and Grading Format Sampling Theorem

### 1 Report Format

1. Briefly explain what you did in the lab
2. Show the Results/plots, you can use excel or a snap shot or draw it by hand.
3. Matlab Code (print the code with the report and also send me the .m file, please save it as lab3\_yourname.m)
  - (a) Sample a 2KHz sinusoid signal, show the original signal and the sampled time domain signals. And also plot the magnitude spectrum.
  - (b) Pass the 2KHz sinusoid signal through a 5 KHz Low Pass Filter. Show the original time domain signal and the spectrum.
  - (c) Pass the sampled 2 KHz sinusoid signal through a 5 KHz Low Pass Filter. Show the spectrum. Justify the result
  - (d) Write a matlab code to find the Minimum Sampling Rate. You can work with any kind of the signal you like. For example: you can choose a 10KHz Sinc/square wave, find the min. sampling rate  $f_{smin}$ . Show time domain and magnitude spectrum for  $f_s < f_{smin}$ ,  $f_s = f_{smin}$ ,  $f_s > f_{smin}$ .
  - (e) MDSDR: Generate a 3KHz sinusoid signal (4V) and a 4KHz sinusoid (close to 0V). Add them as shown in Figure 5. Increase the amplitude of 4KHz wave, when do you notice the presence of this signal. Show the plots (time domain and spectrum).
  - (f) EXTRA CREDIT: In matlab generate a pulse train as shown in Figure 1 of the handout. Use this pulse train to sample a 2KHz sinusoid. Show the sampled time domain signal and the corresponding magnitude spectrum.
  - (g) Build an AM receiver (demodulator). Generate an AM signal with  $A_c = 10$  and  $f_c = 5KHz$ ,  $s(nT_s) = A_c[2.5 + m_1(nT_s) + m_2(nT_s)]\cos(2\pi f_c nT_s)$ . This is the same question as Lab 2b, use a different method to build the demodulator.
    - i. Peak detection: Consider the modulated (time domain) signal. You can get the maximum value of the time domain modulated signal at a point, store this value until the next peak and so forth. Then plot the results. The plot should give an approximate envelope. This is a crude envelope. Then it can be passed through a Low Pass filter to result in a smooth plot, LPF removes the jumps from one value to other.

- ii. hilbert transform: When you take the hilbert transformed signal, the result is  $s(nT_s) = A_c[2.5 + m_1(nT_s) + m_2(nT_s)]e^{(2\pi f_c nT_s)}$ . When the magnitude (absolute value) is taken the signal is obtained.
4. Answer the following questions
- (a) The questions at the end of the handout.
  - (b) Few other questions listed in the procedure.

## 2 Grading format

The lab will be graded for a total of 10 points. There is an option for extra credit, you will not loose any points.

1. 3.5 points for Steps 1.1 - 1.2
2. 3.5 points for Steps 1.3 (Matlab)
3. 3 points for Step 1.4 (for answering the questions)
4. Extra credit: what did you get from this lab (Example: low pass filter or anything, points will vary on the effort and the justification)

## 3 Feedback

I would appreciate if you take few minutes to answer the following questions. It will be helpful if you can write this part on a different paper, so that I can collect it. It will be helpful for future labs (definitely for next year). Thank you.

1. How do you rate the lab
2. What was explained properly and what was not
3. Did you find any concepts missing
4. Feel free to write any other comments (what did you like about the lab and what you didnt, so forth)

You can work in groups but everybody should submit an individual report. The lab report is due in a week.

If you have any other questions, you can send me an email at vijay@ele.uri.edu or stop by the lab Kelly 201.