

ECE 320 SIGNALS AND SYSTEMS II
Matlab Project II
Fall 2002

Posted: Saturday, October 12, 2002

Due: Monday, November 4, 2002

This project has two main objectives. The first objective is to reinforce your understanding of discrete-time filters and convolution. In the first exercise you will study noncausal finite impulse response (FIR) filters and see how they can be used for *interpolation*. The second objective is to develop your intuition about the continuous-time Fourier transform (CTFT). In the second exercise you will use audio signals to explore various properties of the CTFT. As a part of this exercise, you will need to use a discrete-time filter to interpolate an audio signal.

You **MUST** work in groups on this project. Groups assignments will be emailed on October 12, 2002. Each group will turn in a single writeup. The writeups must include all of the analytical (*i.e.*, pencil/paper) work, Matlab plots and code, and relevant explanations. A list of guidelines for preparing the writeup of this project are given below. Failure to comply with these guidelines will result in a grade of **ZERO** for the project.

- The report must be neatly handwritten or typed, and all pages must be numbered.
- All plots must be neatly annotated with x-axis and y-axis labels and a title.
- When referring to plots in the text, you should do at least one of the following:
 - use figure numbers, e.g., “Figure 1 is a plot of the signal $x[n]$.”
 - cite the page number they are on, e.g., “The figure at the top of page 4 is a plot of $x[n]$.”
- All Matlab code must be well-documented and should be included in the report.

1 Preliminaries

To prepare for this project, you should review some of your previous work from this semester. First, review your solution to problem 2.44b from Problem Set 3. You should find this extremely helpful in answering one question assigned in Section 2.

Second, review your solution to part f of Problem ECE320-5 from Problem Set 4 (where you used the `filter` command). If you did not do so for ECE320-5, you may want to do the tutorial exercises in Section 2.2 of *Computer Explorations in Signals and Systems*. You do not need to turn in these tutorial exercises.

2 FIR Filters and Interpolation

Do all of the exercises in Section 2.6 of *Computer Explorations in Signals and Systems* by Buck, Daniel, and Singer. Your writeup should include answers to all of the questions in the book and any other observations you make as you complete the exercises.

Hint:

In the Intermediate Problems, you are asked to insert zeros between the samples of a signal. While this could be accomplished by typing the samples of the expanded signal (including the zeros) into Matlab, there

are easier ways to do it. Matlab's `reshape` command is very useful for expanding a signal. To see how `reshape` might be used, try the following commands:

```
x=[1 2 3 4 3 2 1];  
tmp=[x; zeros(1,7)];  
xnew=reshape(tmp,14,1);
```

How is the vector `xnew` related to the vector `x`? Try `help reshape` for more information about the `reshape` command.

3 Properties of the Continuous-Time Fourier Transform

Do all of the exercises in Section 4.3 of *Computer Explorations in Signals and Systems* by Buck, Daniel, and Singer. Your writeup should include answers to all of the questions in the book and any other observations you make as you complete the exercises.

Notes:

- You may want to use the `soundsc` command instead of `sound`. The `soundsc` command automatically scales the signal and plays it as loud as possible without clipping. This may enable you to hear the signals better.
- In addition to listening to the audio signals, you should also plot them. Include the plots in your report.