Geol 335.3

Lab #7: Fourier Transforms and Filtering

Many signals represent functions of a single variable, like records of ground motion or sound are functions of time, sound, or video can be represented by time series.

In this exercise, you will familiarize yourself with forward and inverse Fourier transforms. Copy to yourself and unpack the zipped archive. Open Matlab in that directory and execute the `fftlab.m` script.

The script should give you a large window with six plot panels (produced by `subplot` command in Matlab). If the window is too large or too small for your screen, you can adjust the dimensions by changing values of field '`Position`' in the call of function `figure()` in script `fftlab.m`. The four values shown like '[100,100,1600,1800]' are the coordinates of the lower-left corner of the window on your screen, and the window width and height, in screen pixels.

The two panels on the left are the real and imaginary parts of the time domain. The time sampling interval is 1 ms, and the total record is 32 ms long.

The two panels in the middle are the real and imaginary parts of the signal in frequency domain. The panels on the right (with red symbols) show the amplitude and phase spectra. Zero time is shown in the middle of time panels, and zero frequency is shown as the open circle in the bottom.

For clear displays, all plots are normalized to unit peak complex magnitudes. The scaling values are printed in the titles of upper plots.

Operate the script by pressing keyboard keys 'c' (to clear the plots) and 'q' (to exit) and the left and right mouse buttons to construct waveforms and spectra. A brief summary of the commands is printed along the bottom of the window.

Assignments

Perform the following tests and answer questions. Make plots of your work by using figure menu operations in Matlab.

1) (8%) Note the total frequency range. How does it relate to the time sampling interval (1 ms)? What is the sampling interval of the frequency?

Note that the frequency axis in the middle panels extends from $-f_N$ to f_N, where f_N is the Nyquist frequency.

2) (8%) Using the left mouse button, **put in a 2-ms wide "*boxcar*" function** (square wave) (a box of 3 samples of equal values) centered on the origin. What is the Fourier transform of this signal (shown in the middle plots and spectra on the right)? How wide is the main lobe? Print out the plot out and mark it up.

3) (8%) Press 'c' to clear the displays and **create another boxcar function** with double width of the square-wave function. How wide is the transform now? Repeat with a yet wider boxcar. What property of the Fourier transform does this test illustrate?

4) (8%) Clear the display and **create another boxcar function** of the same width but with its rightmost sample at the time origin. How does the transform compare to the one before? What property of the Fourier transform does this illustrate?

5) (8%) Clear again and use the right mouse button to **put a single impulse** at the origin. What does its transform look like?

6) (8%) **Move the impulse** 2-5 ms away from the zero grid point. What happens to the transform?

7) (8%) **Put in two pulses equally spaced around the origin** two units away from the origin. This is called an even impulse pair. What is characteristic about its transform?

8) (8%) **Make one of the pulses negative**. This is called the odd impulse pair. What does the transform look like?

9) (8%) Put a **single pulse** at 0 kHz on the **real part of the frequency domain**. Repeat this test for a pulse at –0.2 kHz or +0.2 kHz. Describe the characteristic features of the time-domain signal. Is it real- or imaginary-valued? Is it an even or odd function of time? Is it periodic, with what period? What phases do you see in the bottom-right plot?

10) (8%) Repeat step 9) with **imaginary-valued** pulses in **the frequency domain** and answer the same questions.

11) (8%) Repeat step 9) with **an even impulse pair at** ±0.2 kHz. Answer the same questions.

12) (8%) Finally, **experiment with low-pass, high-pass, or band-pass filtering**. Put one pulse in the time domain, as in steps 5) or 6). By clicking the left mouse button in the upper-right plot (amplitude spectrum), you can construct a shape of the filter. The filter will be shown by a green line. Press key 'f' to clear the filter, and then you can pick it again. When the filter is shaped, press 'a' to apply it to the records. Describe how the time-domain and frequency-domain signals change.

    Note that in the frequency domain (middle plots), filtering is applied to both the positive and negative frequencies.

    (4%) Try putting two pulses of different amplitudes and press 'a' again. You should see how the filtered pulses overlap in time.

Make more experiments with the transform tool if you like.

Hand in:

Zipped directory or Word or PDF document containing answers to the above questions and images.