ACS 503 Signal Analysis Quiz 3 – Fall 2020

You may use any of the material from the class on this quiz. That includes notes, homework, and anything on the Canvas site for ACS597. You may not discuss the problems or their solutions with any other person; the work you submit must be your own.

Unless the operations are obvious, please explain your methods and your answers. Just giving a value with no explanation is not sufficient. Furthermore, giving an explanation can help me to give partial credit for errors. The due date is posted on the Canvas calendar.

When I ask you to "check" or "test" things, I'm not asking you to *prove* something, I'm looking for a test of at least some aspect of your solution, a check that will increase confidence in your approach. If you are in doubt about the intent of a question, explain the point of uncertainty and explain your approach.

1. (30 pts) Two-Dimensional Array

Suppose that you're working with a two-dimensional array of four receiving transducers. By two-dimensional, I mean that the element positions can be described completely by (x,y) coordinates—all elements have the same z coordinate, which you can take to be zero. The elements are located at the corners of a rectangle and have the following (x,y) coordinates in meters:

(0.0, 0.0), (2.0, 0.0), (0.0, 1.0), (2.0, 1.0) for elements 1, 2, 3, and 4 respectively.

For one measurement, you are given the x- and y-components of the slowness vector associated with the incident plane wave,

$S_x = 0.475\text{e-}3$ s/m; $S_y = 0.475\text{e-}3$ s/m,

and that the z-component is zero.

a. Make a sketch of the elements and include the direction of the slowness vector. Don't worry about the length of the slowness vector—its units are not meters—just show the correct direction.

b. Is the array more likely to be in air or in the ocean? Why?

c. For the plane wave with the above slowness vector, what are the times of arrival at elements 2, 3, and 4, if using element 1 as the reference (so the arrival time at 1 is 0.0 s)?

For another measurement with a different signal (but still arriving as a plane wave at the array), you measure the following arrival times:

$t_1 = 0.0$; $t_2 = 0.598$ ms; $t_3 = -0.635$ ms; $t_4 = -0.003$ ms.

d. Find the x and y components of the slowness vector for this plane wave.

e. What is the sound speed?

2. (40 pts) Quadratic Frequency Sweep

A pulse with a frequency that varies quadratically with time (i.e., traces part of a parabola in time-frequency space) can be useful in imitating some natural sounds. In this problem, you'll construct an approximation to the dominant component of a crow's "caw". It won't sound much like a crow because there are other significant components but it's a place to start.

a. Write a function to take as input three time/frequency pairs (i.e., a 3x1 vector of times and a 3x1 vector of frequencies) and fit those three points with a frequency function that is quadratic in time. You can either generate the coefficients of the second-order polynomial or write the function to produce a vector of frequency values given an input vector of time values. You may use the polynomial-fit function in MatLab/Octave/Python or you can write the relevant matrix equation (three equations in three unknowns) and solve directly.

b. How can you check to be sure your function produces reasonable results? Execute that check and show the results.

c. Modify that function to also generate the corresponding instantaneous phase. As before, you can either find the coefficients of whatever polynomial properly expresses the phase or produce a vector of phase values given an input vector of time values.

d. Use the following three time/frequency pairs,

(0.145 s, 1020 Hz), (0.190 s, 1410 Hz), (0.280 s, 990 Hz),

to generate a pulse with an instantaneous frequency that varies quadratically. Create a vector that is 0.4 seconds long and insert the pulse so that it starts at t = 0.145 s and ends at t = 0.280. Before and after this interval the values in the vector should be zeros. When the pulse is not zero, the amplitude of the oscillation peaks should be 0.9 WU. (In practice, you'd smooth the leading and trailing edges of the pulse—don't worry about that in this problem.)

e. How would you test the pulse to be sure the frequency variation is what you intended? Execute that check and show the results.

f. The wav file, Bird_call_iso1.wav, contains a pair of actual crow "caw"s. Using some tool from this class, compare your synthetic "caw" to the actual recording. Describe what you would do to make the synthetic "caw" sound more realistic. You don't have to execute these improvements, just describe what you'd do.

3. (30 pts) Partial-Correlation Filter

The following equation,

$$y_n \quad = \quad r\, y_{n-1} \quad + \quad \sqrt{1-r^2}\; x_{n-1} \qquad ; \qquad 0.0 < r < 1.0$$

suggests a simple time-domain filter with input, x, output, y, and a single parameter, r. In fact, this filter is sometimes used to filter random, white noise in order to produce a noise-like signal that is more representative of natural noise processes. The constant, r, specifies how much similarity there is between adjacent output samples.

a. What does this filter do if $r = 0$? If $r = 1$?

b. Find the a and b filter coefficients so that you can use the MatLab *filter* function to implement the filter.

c. How would you test the filter? Try that test (or tests) and discuss the results.

d. Find and plot the magnitude and phase of the frequency response of this filter from 0 to $fs/2$ Hz using a sampling frequency, fs, of 5000 and $r = 0.85$.

e. According to literature on this filter, the value $f_x = -f_s * \ln(r)/(2\pi)$ gives a frequency that has a special meaning with regard to the frequency response. (The "ln" here is the MatLab *log* or natural log.). What relationship do you see?