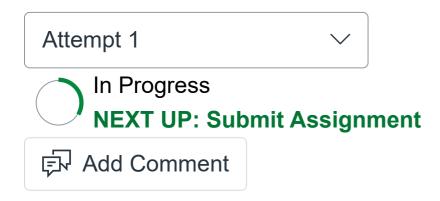
Coding #05: Frequency Magnitude/Phase Responses & Filter Properties

10 Points Possible



Unlimited Attempts Allowed

2025/9/24 to 2025/10/8

∨ Details



Narrative:

Our surveillance team has stumbled upon a curious cache of digital filters in enemy transmissions. At first, the numbers look meaningless—but filters never lie. If we classify their design, we may deduce how the enemy is shaping their signals.

Later, one of our field agents intercepted a garbled audio tape—clearly urquan.wav, but run through a mysterious filter. Rumor from our inside source says the adversary's filter is a **bandpass system**, tuned to let only a narrow range of frequencies slip through. To

respond in kind, we'll need to identify their design and reconstruct a copy from scratch. Only then can we decode what lies hidden in the passband.

This case is divided into two parts.

Provided Tools

```
get_filter_info function: get filter info.p
(https://ufl.instructure.com/courses/540008/files/100385461?wrap=1)
(\downarrow)
(https://ufl.instructure.com/courses/540008/files/100385461/download?
download frd=1)
get_filtered_audio function: get filtered audio.p
(https://ufl.instructure.com/courses/540008/files/100385465?wrap=1)
(https://ufl.instructure.com/courses/540008/files/100385465/download?
download frd=1)
ba2pz helper function: ba2pz.m
(https://ufl.instructure.com/courses/540008/files/99382863?wrap=1) \(\psi\)
(https://ufl.instructure.com/courses/540008/files/99382863/download?
download frd=1)
pz2ba helper function: pz2ba.m
(https://ufl.instructure.com/courses/540008/files/100251216?wrap=1)
(https://ufl.instructure.com/courses/540008/files/100251216/download?
```

pzlot helper function: pzplot.m

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download frd=1)

(https://ufl.instructure.com/courses/540008/files/99382867/download? download_frd=1)

urquan.wav audio file: urquan.wav

(https://ufl.instructure.com/courses/540008/files?preview=100385468)

Problem 1: Filter Identification

You are provided a .p file:

This returns numerator and denominator filter coefficients **b** and **a**.

Your task: Analyze this filter and determine its properties.

Specifically, you must submit **six classification variables** in a .mat file:

- filter_type 1 for low-pass, 2 for high-pass, 3 for all-pass,
 for band-pass, 5 for stop-band
- stability_type 1 if the filter is stable, 0 if unstable.
- [fir_type] [1] if FIR, [0] if IIR.
- linear_phase_type 1 if linear phase, 0 if not.
- $[invertible_type]$ [1] if invertible, [0] if not.
- minimum_phase_type 1 if minimum phase, 0 if not.

Hint: You may use the helper functions provided:

- [p, z] = ba2pz(b, a)
- [b, a] = pz2ba(p, z)
- pzplot(b, a)

You can use help ba2pz, and help pz2ba to learn about their use.

Submission for Problem 1:

A .mat file containing the five scalar variables above.

Problem 2: Reconstructing the Hidden Bandpass Filter

You are also given another .p file:

```
[y, x, Y, X, w] = get_filtered_audio(ufid);
```

The <code>.p</code> file produces a filtered version <code>y</code> and original version <code>x</code>. The signal <code>y</code> is filtered using an **unknown bandpass filter**. Also provided is the discretized DTFT of the filtered signal <code>y</code> and discretized DTFT of the original signal <code>x</code>. The vector <code>w</code> is the angular frequencies (horizontal axis) for <code>y</code> and <code>x</code>. From the DTFT, it should be clear the signal has been **restricted to a narrow slice of the spectrum**—a telltale signature of a bandpass design.

Your task:

Design your own filter with numerator and denominator coefficients b and a such that when applied to x, the result matches y. You should use as **few filter coefficients as possible** while achieving the effect. Note that while overall gain is an important filter variable, we will not be grading it in this assignment. What is important is that you get the center frequency and shape.

In addition, determine the **center angular frequency** (in normalized frequency between $-\pi$ to $+\pi$) we of the bandpass filter and provide

it as a variable.

Hint: You may want to use the following MATLAB functions:

- filter → (https://www.mathworks.com/help/matlab/ref/filter.html)
- soundsc ⇒ (https://www.mathworks.com/help/matlab/ref/soundsc.html)

Submission for Problem 2:

A .mat | file containing:

- b, a your chosen filter coefficients
- wc the center frequency of the bandpass filter (in rad/s between $-\pi$ to $+\pi$)



Final Deliverable

Please submit a single .mat file (e.g., case5 results.mat) that includes:

- filter type, stability type, fir type, linear phase type, invertible type, minimum phase type (Problem 1)
- b, a, wc (Problem 2)

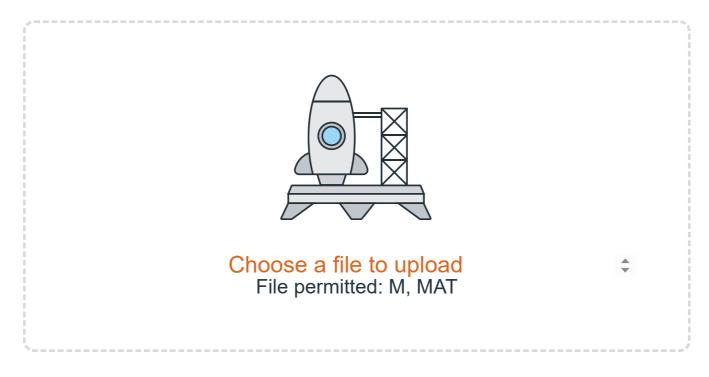
All variables must be defined exactly as above.

Choose a submission type









or



Submit Assignment