IG.2407 - Signal Acquisition and Processing

Lab 2

IIR and FIR Filtering

G8

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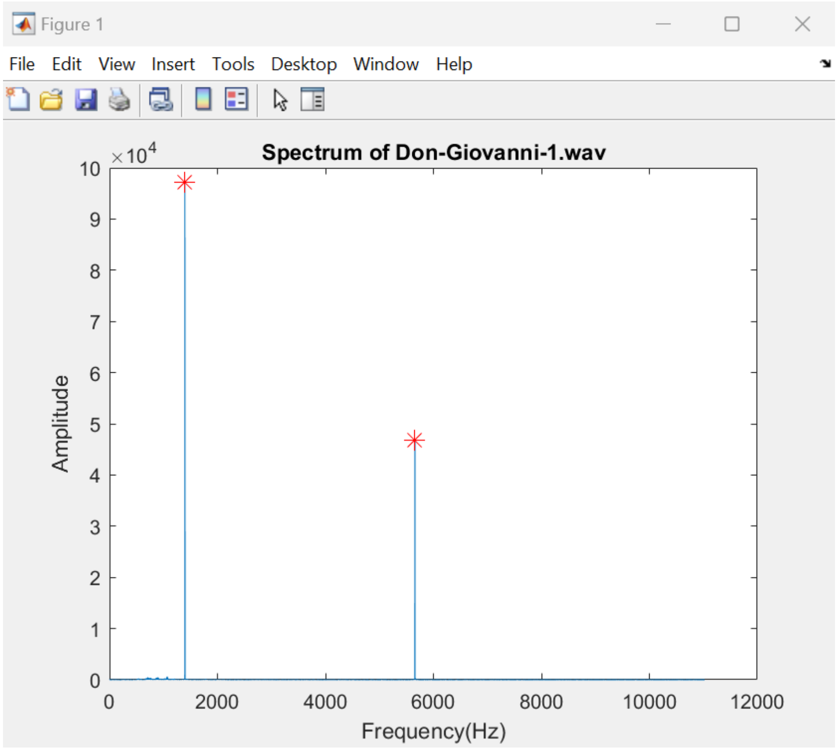
https://github.com/heyPetiteF/ISEP/tree/main/2402-2406/2-SIGNAL/LAB/LAB2

CONTENT

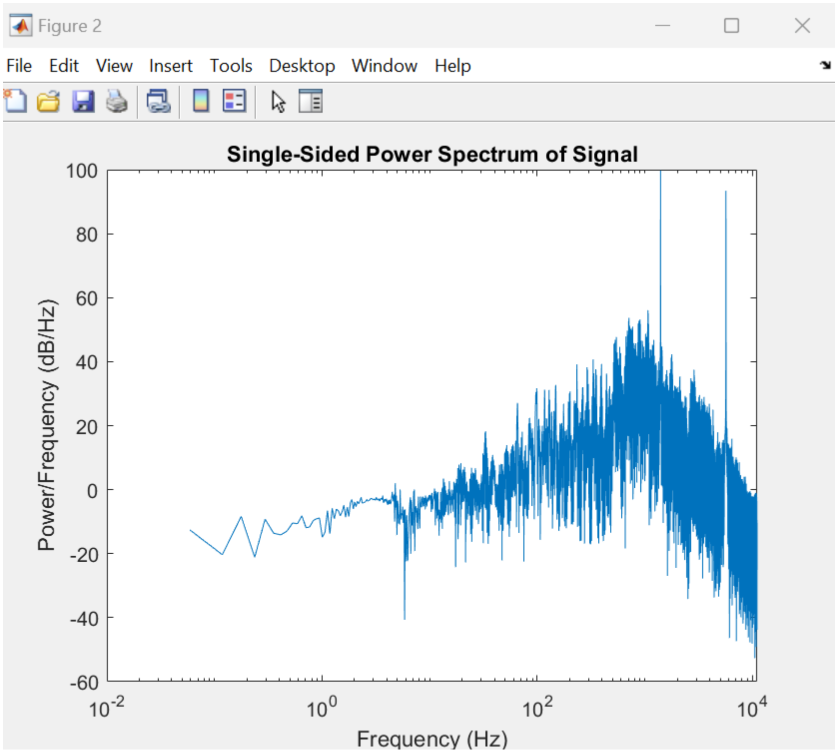
Part 1: IIR filter as a noise cancelling filter.

1. Signal Analysis

- Fourier transform an audio file.
- Plot the spectrum of an audio file.
- Use “findpeaks” function to mark two peaks.



- These two peaks are which need to be removed.



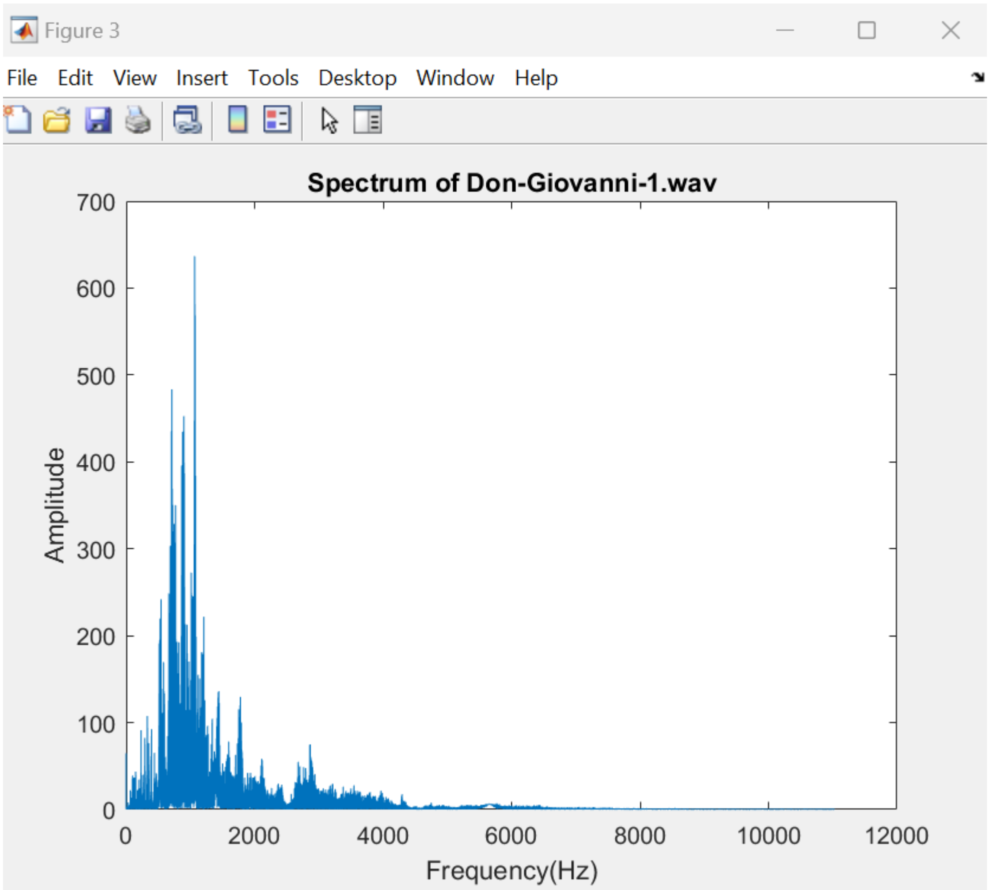
- In the single-side power spectrum image can also clearly see that there are two frequency points, which are noise points.

2. Filter Design – IIR

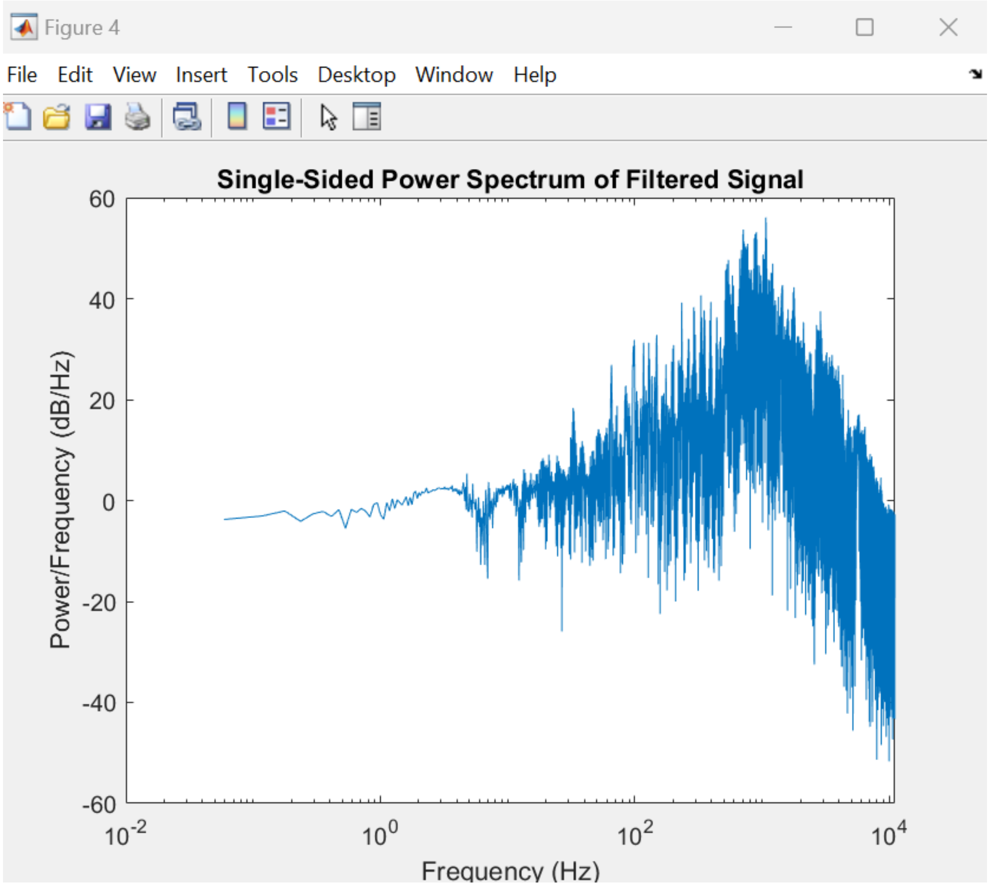
- Use the “fdesign.notch” function to create a notch filter design object & implement it as a digital filter through the “design” function.
- **N**: the order of the filter, here is 2.
- **F0**: the filter center frequency, already obtained when marking peak before.
- **Q**: the quality factor, which is inversely proportional to the bandwidth of the filter.
- **butter**: instructs the design function to use the Butterworth method to implement filter design.
- **bn**: numerator coefficient of the filter.
- **an**: denominator coefficient of the filter.

3. Results

- After filtering, can see that there are no obvious noise points through Spectrum figure:



- There also no noise points in single-Sided Power Spectrum figure:



- Play the filtered audio and you can hear clear singing without harsh noise.
- Filtering successful.

Part 2: FIR filter as moving average filter.

1. Signal Analysis

- The Convolution:

$$y_n = \sum_{k=0}^{N-1} x_{n-k} h_k = \frac{1}{N}\sum_{k=0}^{N-1} x_{n-k}$$

- Impulse Response Function:

$$h[n] = \frac{1}{N} \ for\ 0 \leq n < N$$

- **Transfer Function** (Z transform of IRF)**:**

$$H(z) = Z\{h[n]\} = \frac{1}{N}\sum_{n=0}^{N-1} z^{-n} = \frac{1}{N}\frac{1-z^{-N}}{1-z^{-1}}$$

- Frequency Response (Transfer Function on the unit circle, z = $e^{j\omega}$):

$$H\left(e^{j\omega}\right) = \frac{1}{N}\frac{1-e^{-j\omega N}}{1-e^{-j\omega}} = \frac{1}{N}\frac{\sin\left(N\omega/2\right)}{\sin\left(\omega/2\right)} e^{-j(N-1)\omega/2}$$

- **Magnitude Response:**

$$\left|H\left(e^{j\omega}\right)\right| = \frac{1}{N}\frac{\sin\left(N\omega/2\right)}{\sin\left(\omega/2\right)}$$

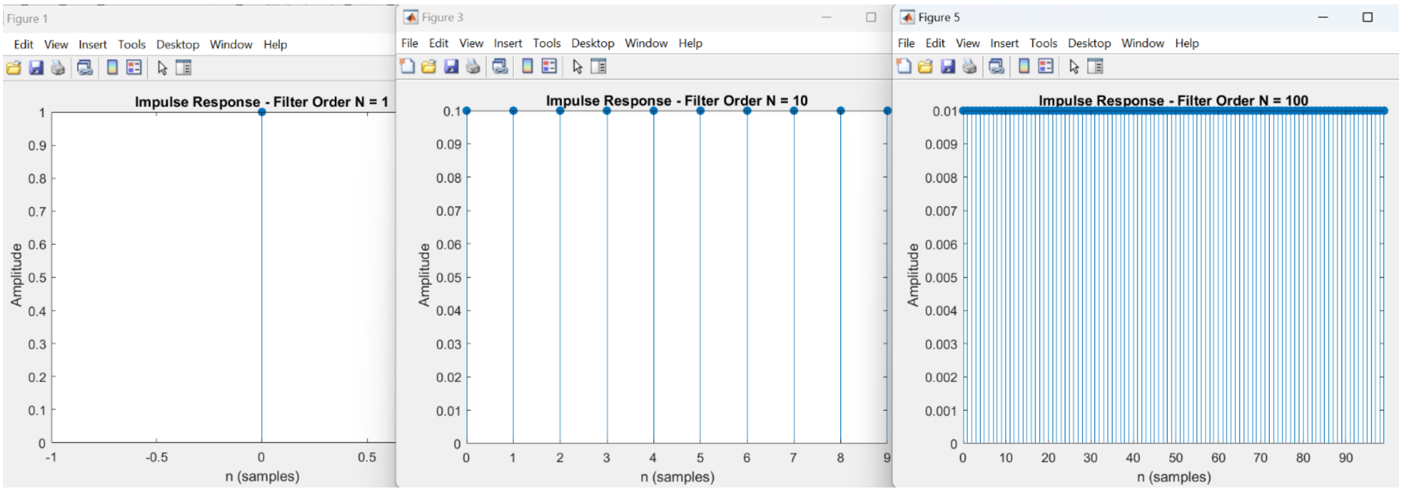
- **Phase response:**

$$\angle H\left(e^{j\omega}\right) = -\frac{(N-1)\omega}{2}$$

2. Filter Design – FIR

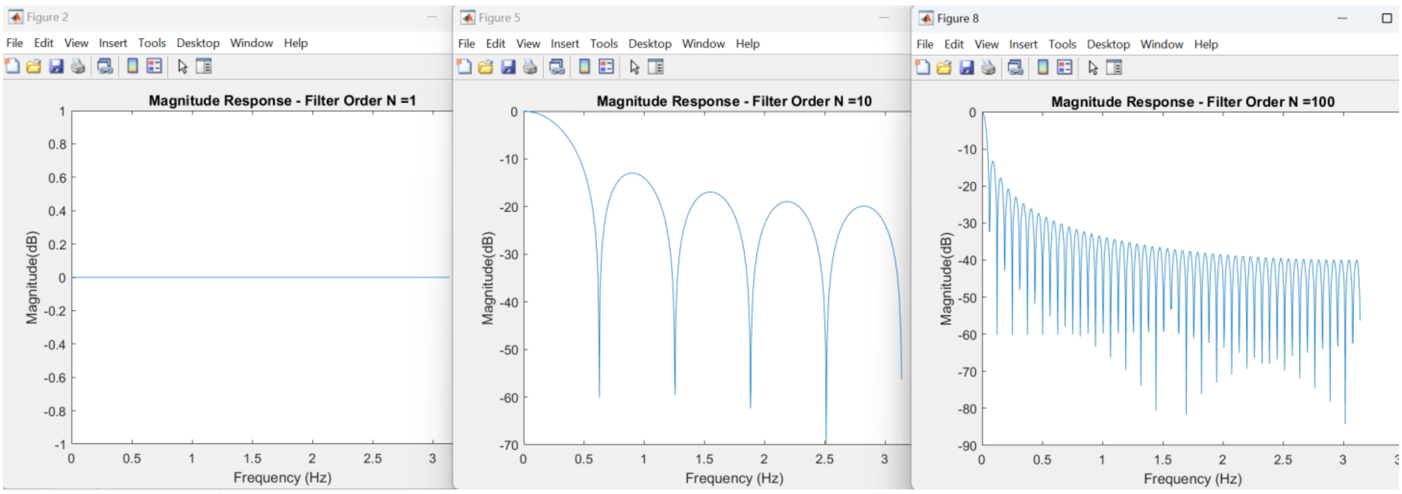
- Create a quantity of length N and value 1/N as the coefficient of the FIR filter.
- Calculate and plot the impulse response using the “impz” function.
- Calculate and plot the frequency response using the “freqz” function.
- Observe changes in frequency response and impulse response by changing the value of N.

1) When N=1&10&100, the impulse response figures.



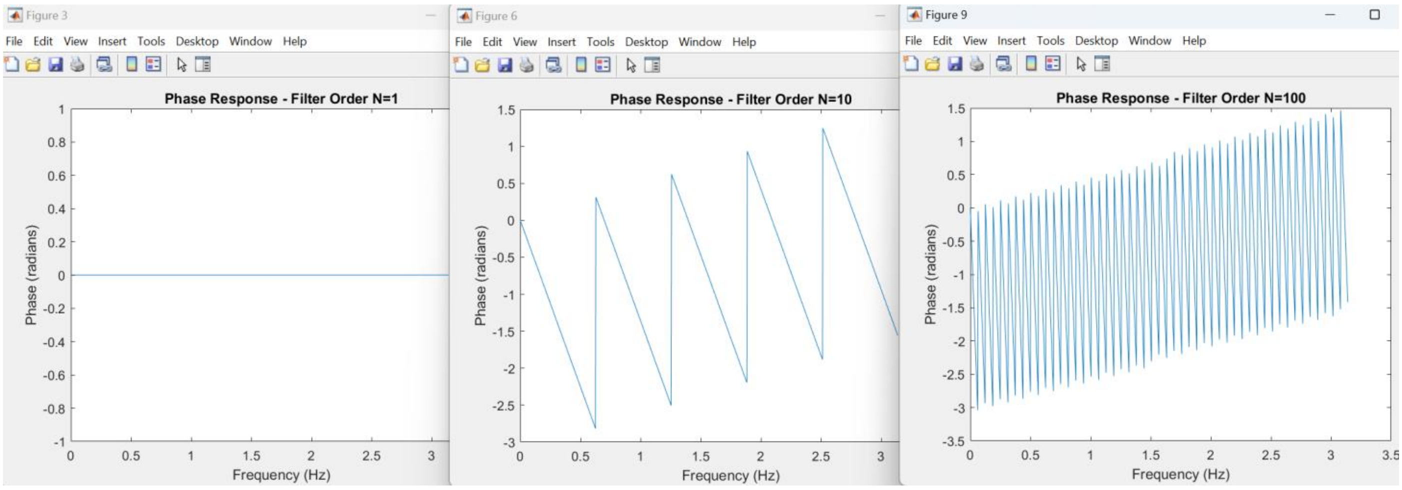
- N=1: There is actually no filtering effect. The impulse response only has one sample of 1.
- N=10: The impulse response shows ten values (all 0.1), indicating that each input sample is one of the ten numbers that is averaged.
- N=100: The impulse response here is very dense, showing 100 values (all 0,01).
- As N increases, the impulse response broadens in time, the filter has a longer temporal "memory" of the input signal.

2) When N=1&10&100, the magnitude response figures.



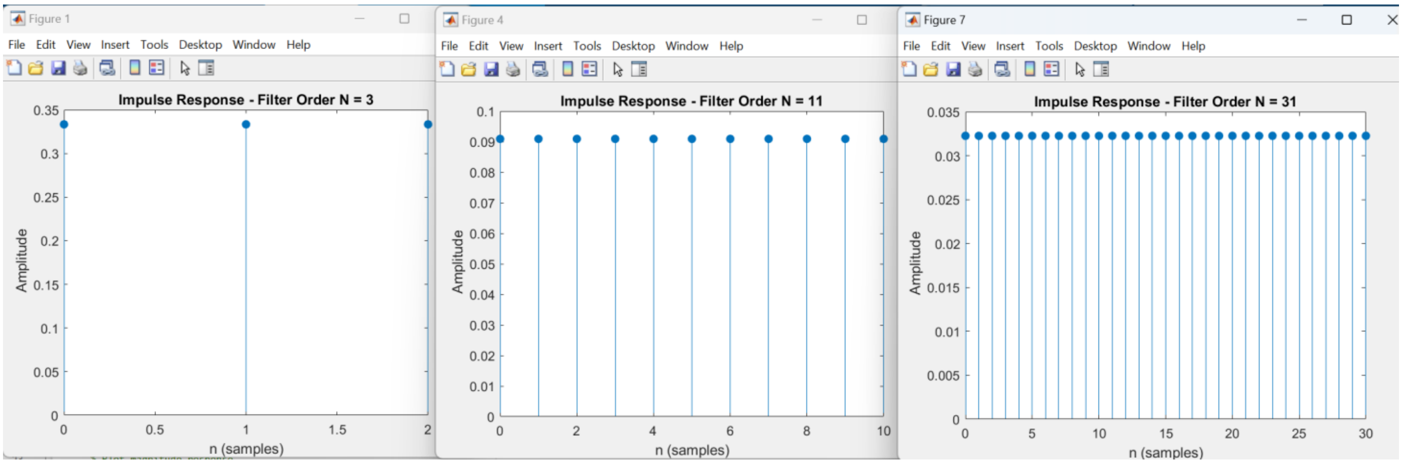
- N=1: The magnitude response is a flat line, indicating that all frequency components are passed.
- N=10: The magnitude response shows an obvious sine wave shape with large fluctuations, signal components at certain frequencies are amplified or attenuated.
- N=100: The magnitude response becomes more oscillatory, the main lobe (the main peak area) becomes narrower, and the side lobes (fluctuations) become more numerous, which means the filter becomes more selective.

3) When N=1&10&100, the phase response figures.

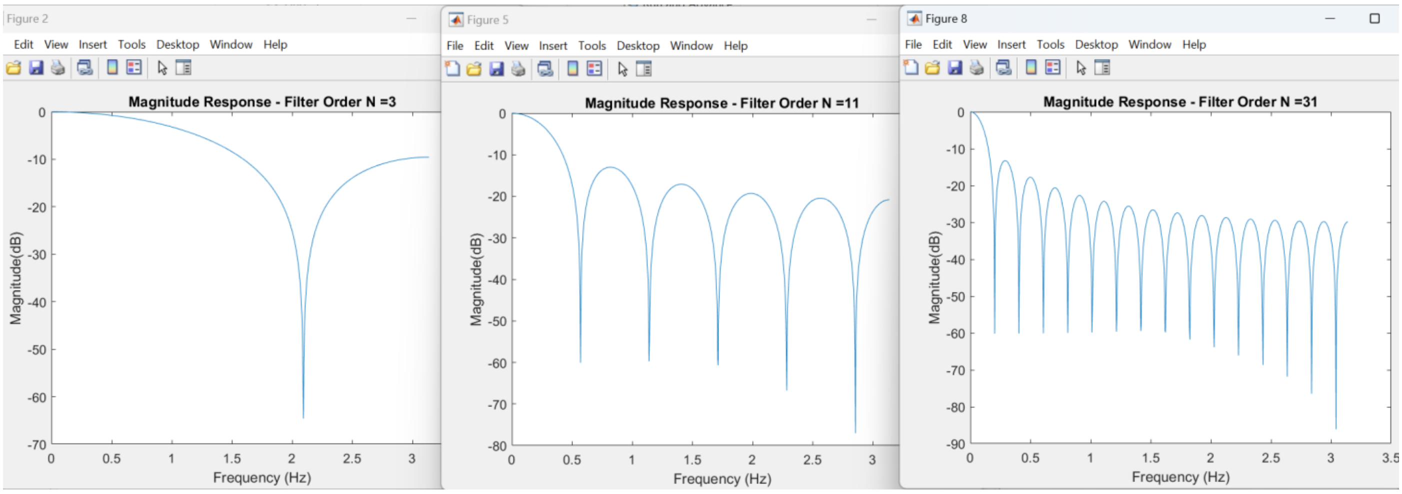


- N=1: it is almost a constant.
- N=10: a trend in phase as a function of frequency.
- N=100: the phase response becomes very volatile.
- As N increases, the phase response becomes more complex.

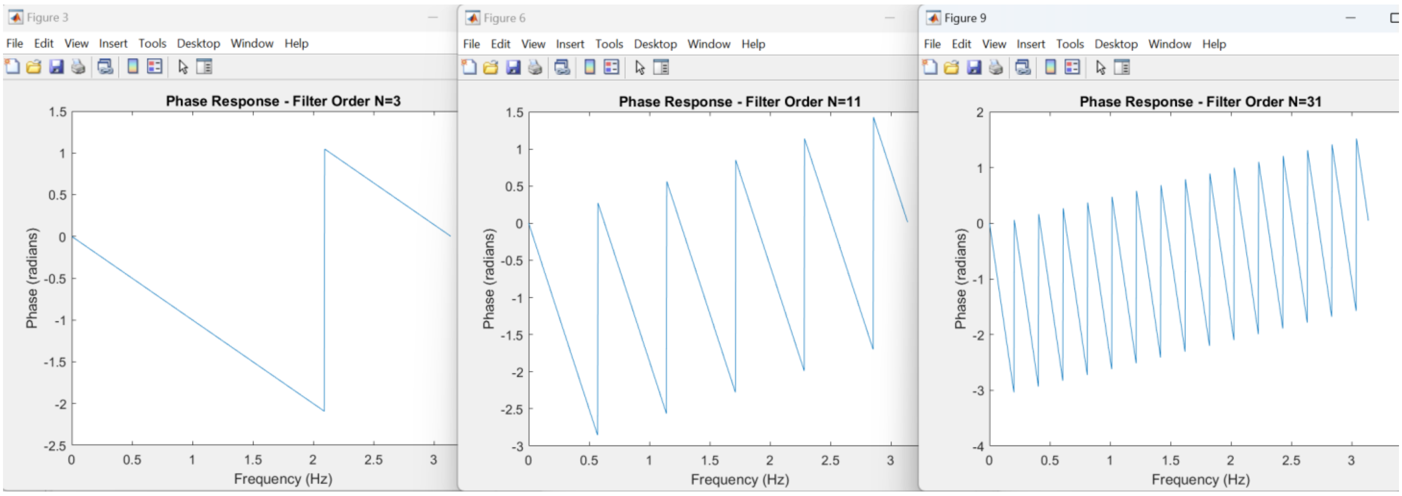
4) When N=3&11&31, the impulse response figures.



5) When N=3&11&31, the magnitude response figures.



6) When N=3&11&31, the phase response figures.



3. Results

When increasing the order N of an FIR filter:

- **Narrower Bandwidth**

  The main lobe of the filter's frequency response becomes narrower, enhancing frequency resolution. For broadband signals, higher-order filters can suppress adjacent unwanted frequencies more effectively.

- **Increased Phase Complexity**

  The phase response becomes more erratic with frequency as N increases. This increased phase complexity can lead to signal phase distortion, especially in parts of the frequency spectrum where the phase changes rapidly.

- **Greater Latency**

  A higher filter order N introduces more latency due to the need to process a greater number of historical input samples.

Appendix (Code)

Part1

https://github.com/heyPetiteF/ISEP/blob/main/2402-2406/2-SIGNAL/LAB/LAB2/LAB2PART1.m

Part2

https://github.com/heyPetiteF/ISEP/blob/main/2402-2406/2-SIGNAL/LAB/LAB2/LAB2PART2.m