

TASK 4 : DIGITAL FILTER DESIGN

BASIC INFORMATION:

A **Digital Finite Impulse Response (FIR) Filter** is one of the most fundamental building blocks in digital signal processing (DSP), widely used in applications such as audio and image processing, communications, and control systems. Unlike Infinite Impulse Response (IIR) filters, FIR filters are characterized by their simple and predictable structure, making them particularly attractive in many digital filtering scenarios.

The primary function of an FIR filter is to process a discrete-time signal by applying a weighted sum of its past and present input samples. The "**Finite**" in FIR refers to the fact that the filter's impulse response is of finite duration—meaning it has a finite number of non-zero coefficients. These coefficients define the filter's behavior, such as whether it acts as a low-pass, high-pass, band-pass, or band-stop filter.

The FIR filter operates on the principle of **convolution**, where the current output is a weighted sum of the input signal values, with each input value multiplied by a corresponding filter coefficient. This makes the filter inherently stable and easy to design, especially for applications that require linear phase responses, where the phase distortion of the signal is minimized.

MATLAB CODE :

```
Fs = 1000;
```

```
Fc = 100;
```

```
N = 50;
```

```
Wn = Fc / (Fs / 2)
```

```
b = fir1(N, Wn, 'low', hamming(N+1));
```

```
fvtool(b, 1, 'Fs', Fs);
```

OUTPUT:

