**Aim ↬ To study the concept and fundamentals of Sampling Theorem.**

**Software Required ↬** MATLAB

**Theory ↬**

Filters are crucial components in signal processing, employed to manipulate and enhance signals by removing unwanted frequencies. The primary types of filters include low-pass, high-pass, and band-pass filters, each serving distinct purposes based on their frequency response characteristics.

A low-pass filter [LPF] allows signals with frequencies below a specified cutoff frequency [fc] to pass through while attenuating higher frequencies. This type of filter is commonly used in applications where it is necessary to remove high-frequency noise from a signal. The mathematical representation of a low-pass filter's transfer function is given by:

$$H(s) = \frac{1}{1 + [s/\omega_c]^{2n}}$$

where $\omega_c = 2\pi f_c$ is the cutoff angular frequency and n is the filter order. A higher order results in a steeper roll-off, leading to more effective attenuation of unwanted high-frequency components.

In contrast, a high-pass filter [HPF] permits signals with frequencies above a certain cutoff frequency to pass through while blocking lower frequencies. High-pass filters are valuable in applications where it is essential to isolate high-frequency signals, such as in audio processing to remove low-frequency hum or rumble. The transfer function for a high-pass filter can be expressed as:

$$H(s) = \frac{[s/\omega_c]^{2n}}{1 + [s/\omega_c]^{2n}}$$

The band-pass filter [BPF] combines the characteristics of low-pass and high-pass filters, allowing a specific range of frequencies to pass while attenuating those outside this range. Band-pass filters are particularly useful in applications like communication systems, where it is essential to isolate specific frequency bands. The transfer function for a band-pass filter can be defined as:

$$H(s) = \frac{[s/\omega_{c1}]^{n_1}[s/\omega_{c2}]^{n_2}}{[s/\omega_{c1}]^{n_1} + [s/\omega_{c2}]^{n_2}}$$

where $\omega_{c1}$ and $\omega_{c2}$ are the lower and upper cutoff frequencies, respectively.

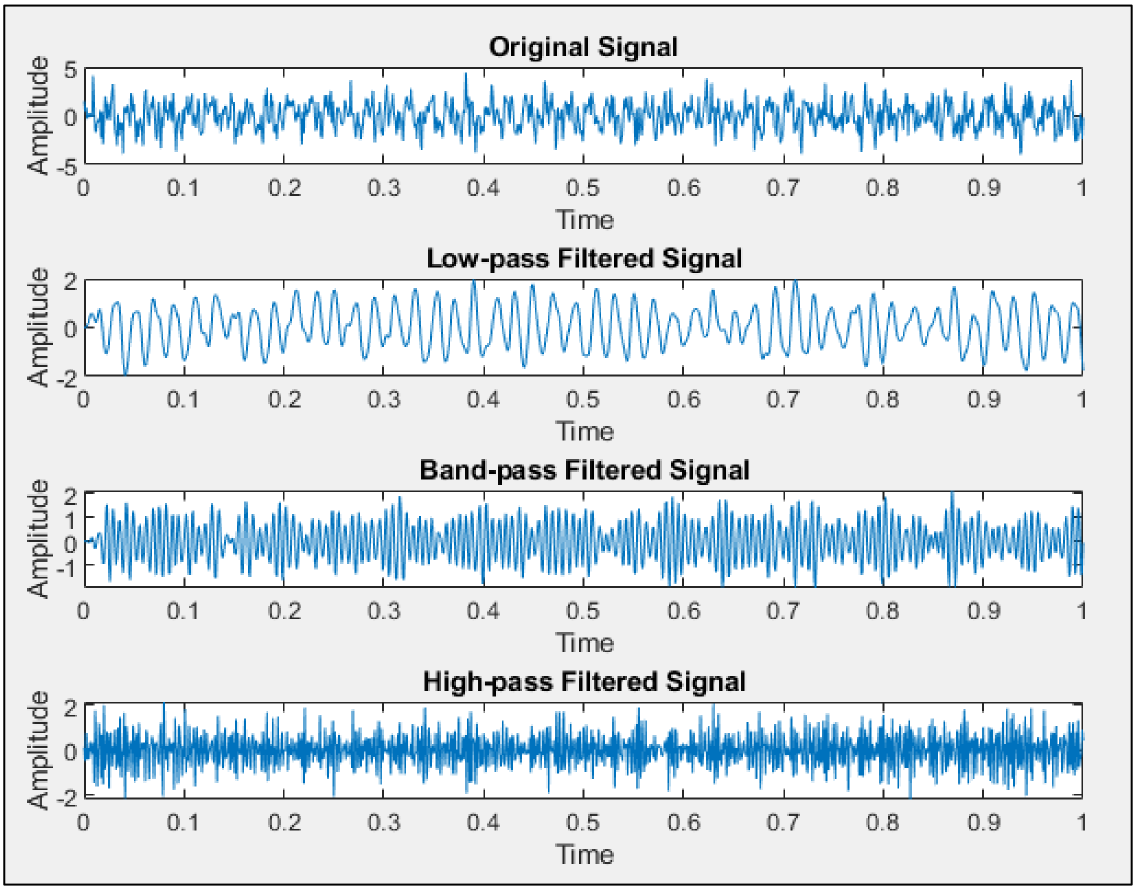
The design of these filters often involves selecting the appropriate cutoff frequencies and the filter order to achieve the desired frequency response. The Butterworth filter is widely used for its maximally flat frequency response, making it ideal for various applications

## Code ↬

```
% Filters
clc;
clear;
close all;

fs = 1000;
t = 0:1/fs:1;
x = sin(2*pi*50*t) + sin(2*pi*150*t) + randn(size(t));

fc_low = 100;
fc_high = 200;
order = 6;

[b_low, a_low] = butter(order, fc_low/(fs/2), 'low');
filtered_low = filter(b_low, a_low, x);

[b_band, a_band] = butter(order, [fc_low, fc_high]/(fs/2), 'bandpass');
filtered_band = filter(b_band, a_band, x);

[b_high, a_high] = butter(order, fc_high/(fs/2), 'high');
filtered_high = filter(b_high, a_high, x);

figure;
subplot(4, 1, 1);
plot(t, x);
title('Original Signal');
xlabel('Time');
ylabel('Amplitude');

subplot(4, 1, 2);
plot(t, filtered_low);
title('Low-pass Filtered Signal');
xlabel('Time');
ylabel('Amplitude');

subplot(4, 1, 3);
```

```
plot(t, filtered_band);
title('Band-pass Filtered Signal');
xlabel('Time');
ylabel('Amplitude');

subplot(4, 1, 4);
plot(t, filtered_high);
title('High-pass Filtered Signal');
xlabel('Time');
ylabel('Amplitude');
```

## Output ↬




## Result ↬

The results successfully verified the need for sampling at or above the Nyquist rate to prevent aliasing and ensure accurate signal reconstruction, highlighting differences in frequency spectra.

## Conclusion ↬

In conclusion, the experiment validated the Sampling Theorem, emphasizing the importance of sampling at rates equal to or greater than twice the maximum frequency to maintain the original signal's integrity. This principle is crucial for applications in telecommunications and audio processing.

## Precautions ↬

- Ensure the sampling rate is set correctly according to the Nyquist criterion to prevent aliasing.
- Use an anti-aliasing filter prior to sampling to limit the bandwidth of the input signal.
- Verify the accuracy of the signal generation formulas to avoid computational errors.
- Label and title plots clearly to differentiate between original and reconstructed signals effectively.