



o Conveyor Belt

The ramp input test the motor's ability to gradually accelerate, ensuring the belt operates smoothly under increasing load without instability.

4) We were able to generate various types of signals in Matlab

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1. The Sampling Theorem (Nyquist-Shannon) states that a continuous signal can be completely represented by discrete samples if the sampling rate f_s is greater than twice the maximum frequency f_m of the signal: $f_s \geq 2f_m$

Aliasing Effect: Aliasing occurs when a signal is sampled below the Nyquist rate $f_s \geq 2f_m$. Higher frequency components fold back into the lower frequencies, causing distortion and misrepresentation of the signal.

Eliminate Aliasing:

- 1) Increase the sampling rate to be greater than $2f_m$.
- 2) Use a low-pass filter (anti-aliasing filter) to remove high-frequency components before sampling.

2. Nyquist Criterion

The Nyquist Criterion states that to avoid aliasing and perfectly reconstruct a signal, the sampling rate must be at least twice the maximum frequency in the signal: $f_s \geq 2f_m$. This ensures the signal is sampled sufficiently to preserve all its frequency components.

4. Aliasing in Control Systems (e.g., Robotic Arms, CNC Machines)

- Impact of Aliasing: In control systems like robotic arms and CNC machines, aliasing can lead to inaccurate feedback, causing jerky movements, position errors, or misalignment in machine operations.
- Proper Sampling Rates: Ensuring a sampling rate higher than twice the maximum signal frequency avoids aliasing, allowing the control system to maintain precise position tracking, smooth motion, and accurate control.

5. Result and Conclusion

Result: When the sampling rate is greater than twice the signal's frequency ($f_s > 2f_m$) there is no aliasing, and the signal can be accurately reconstructed. If the sampling rate is insufficient ($f_s < 2f_m$), aliasing occurs, causing errors.

Conclusion: Proper sampling rates are crucial for accurate feedback control in systems like robotic arms and CNC machines. Aliasing can severely affect system performance, and by following the Nyquist criterion and using anti-aliasing filters, control systems can function accurately and reliably.

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1. Properties of DFT

The Discrete Fourier Transform (DFT) has several important properties that are useful in signal processing. Here are the properties for **Linearity**, **Time Shifting**, and **Parseval's Theorem**:

1 Linearity:

The DFT is a linear operation, meaning that the DFT of a linear combination of two signals is the same as the linear combination of their individual DFTs.

- Mathematically:

$$\text{DFT}\{ax[n] + by[n]\} = a \cdot X[k] + b \cdot Y[k]$$

Where:

- $x[n]$ and $y[n]$ are the input sequences.
- a and b are constants.
- $X[k]$ and $Y[k]$ are the DFTs of $x[n]$ and $y[n]$, respectively.

.2 Time Shifting:

Time shifting in the time domain results in a phase shift in the frequency domain. If the signal is shifted by m samples:

- Mathematically:

$$\text{DFT}\{x[n - m]\} = X[k] \cdot e^{-j2\pi \frac{km}{N}}$$

Where:

- $x[n - m]$ is the time-shifted sequence.
- $X[k]$ is the DFT of $x[n]$.
- N is the total number of samples.

.3 Parseval's Theorem:

Parseval's theorem relates the total energy in the time domain to the total energy in the frequency domain:

- Mathematically:

$$\sum_{n=0}^{N-1} |x[n]|^2 = \frac{1}{N} \sum_{k=0}^{N-1} |X[k]|^2$$

Where:

- $x[n]$ is the input sequence.
- $X[k]$ is the DFT of $x[n]$.

This theorem essentially states that the energy of a discrete-time signal is conserved between the time and frequency domains.

3. DFT in Audio Signal Compression (e.g., MP3)

In audio signal compression techniques like **MP3**:

- DFT** (or its efficient implementation, the **FFT**) is used to convert audio signals into the frequency domain.
- The signal is transformed into a **spectral representation**, which shows the **magnitude** of each frequency component.
- Compression Process:**
 - In the frequency domain, perceptually irrelevant components (like high-frequency sounds that are inaudible to the human ear) are discarded.
 - This reduces the amount of data required to store or transmit the audio signal.
- The compressed signal is then stored or transmitted, and when decoded, an **Inverse FFT (IFFT)** is applied to reconstruct the audio signal.

4. DFT in Vibration Signal Analysis

DFT is commonly used in vibration analysis to identify **dominant frequencies** in mechanical systems. For example:

- Mechanical failures**, like bearing wear, misalignment, or imbalance, often manifest as **specific frequency patterns** in vibration signals.
- By performing a **DFT** on the vibration signal, we can isolate these dominant frequencies:
 - Dominant frequencies** in the DFT output correspond to the mechanical resonant frequencies or fault frequencies.
 - Monitoring these frequencies over time allows for **predictive maintenance** to detect failures before they cause significant damage.

Results:

DFT/IDFT enable signal transformation between time and frequency domains. Magnitude and phase reveal frequency components. In audio compression, DFT helps reduce file size by

discarding irrelevant frequencies. In vibration analysis, DFT identifies dominant frequencies for predicting mechanical failures.

Conclusion: DFT is essential for frequency analysis, aiding in efficient audio compression and predictive maintenance in mechanical systems.

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3) Circular Convolution for Fault Detection

Circular convolution compares a measured signal from machinery with a reference signal, amplifying similarities and highlighting differences. Fault signatures appear as deviations in the output. Using the frequency domain, it identifies fault-related harmonics efficiently, aiding in detecting issues like bearing defects, imbalance, or misalignment.

4) understood circular convolution and implemented algorithm in math lab of discrete time sequences

S-5

2 Z-Transform for Discrete PID Design

The Z-transform converts a continuous PID controller into a discrete form for digital implementation. By substituting s with its Z-domain equivalent, the PID transfer function is discretized into a difference equation:

$$u[n] = K_p e[n] + K_i \sum e[n]T + K_d \frac{e[n] - e[n-1]}{T}$$

This equation controls the DC motor in a robotic arm, minimizing position error. The motor's dynamics are modeled in the Z-domain, and K_p , K_i , and K_d are tuned for optimal performance, ensuring precise motion control.

3. We were able to implement z transform and inverse z transform for the above equation

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3. FT for Detecting Mechanical Faults

The Fast Fourier Transform (FFT) converts vibration signals from rotating equipment into the frequency domain. Faults like imbalance or misalignment create specific frequency patterns:

- **Imbalance:** Detected as a peak at the rotational frequency.
- **Misalignment:** Identified by peaks at multiples (harmonics) of the rotational frequency.

By analysing the frequency spectrum, FFT highlights these fault-related signatures, enabling quick and precise diagnostics.

4. computation of fft using inbuilt function and without using inbuilt function was performed using MATLAB

S-7

2. FIR and IIR Filters in Drone Accelerometer Data

- **FIR Filters:**
Provide stable filtering with no feedback. Used to remove noise (e.g., high-frequency vibrations) while preserving signal phase, crucial for accurate orientation and motion tracking.
- **IIR Filters:**
Use feedback for efficient filtering, requiring fewer computations. Ideal for real-time applications like stabilizing the drone by smoothing out accelerometer data for control systems.

Both filters ensure clean and reliable sensor data for precise flight control.

3. we studied general dsp function, transform functions, filter functions, compression methods to design using dsp tools