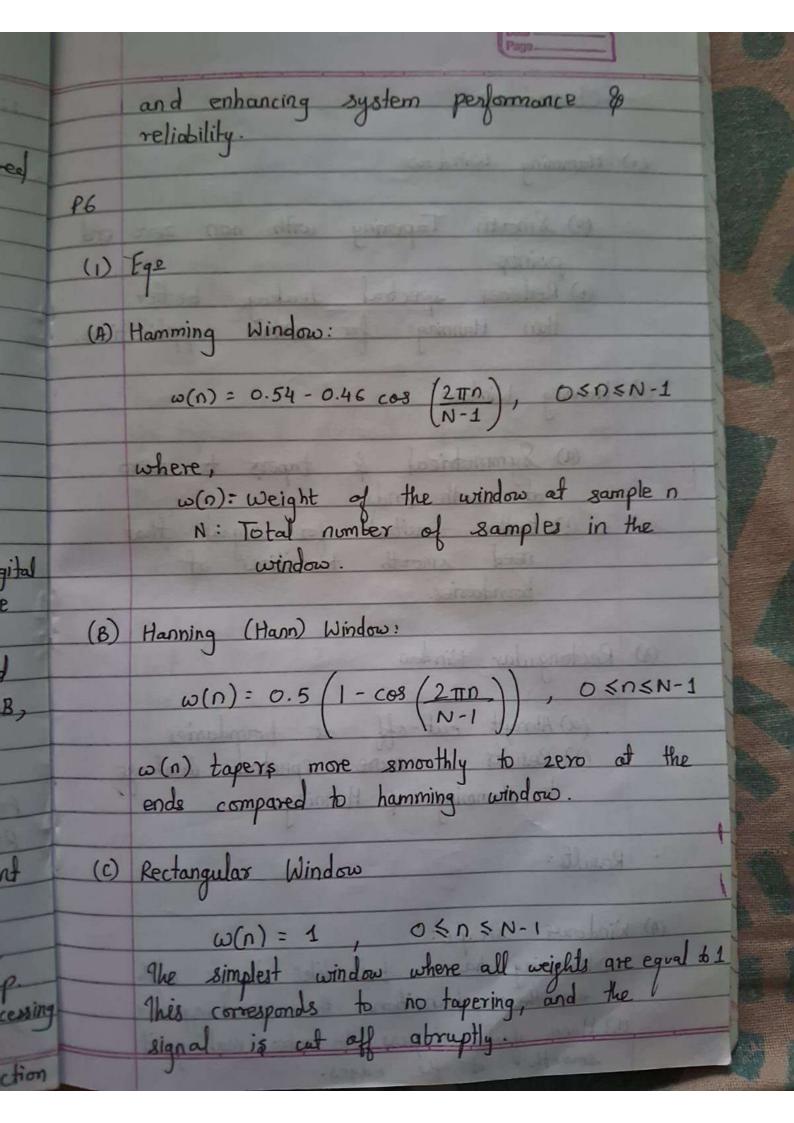
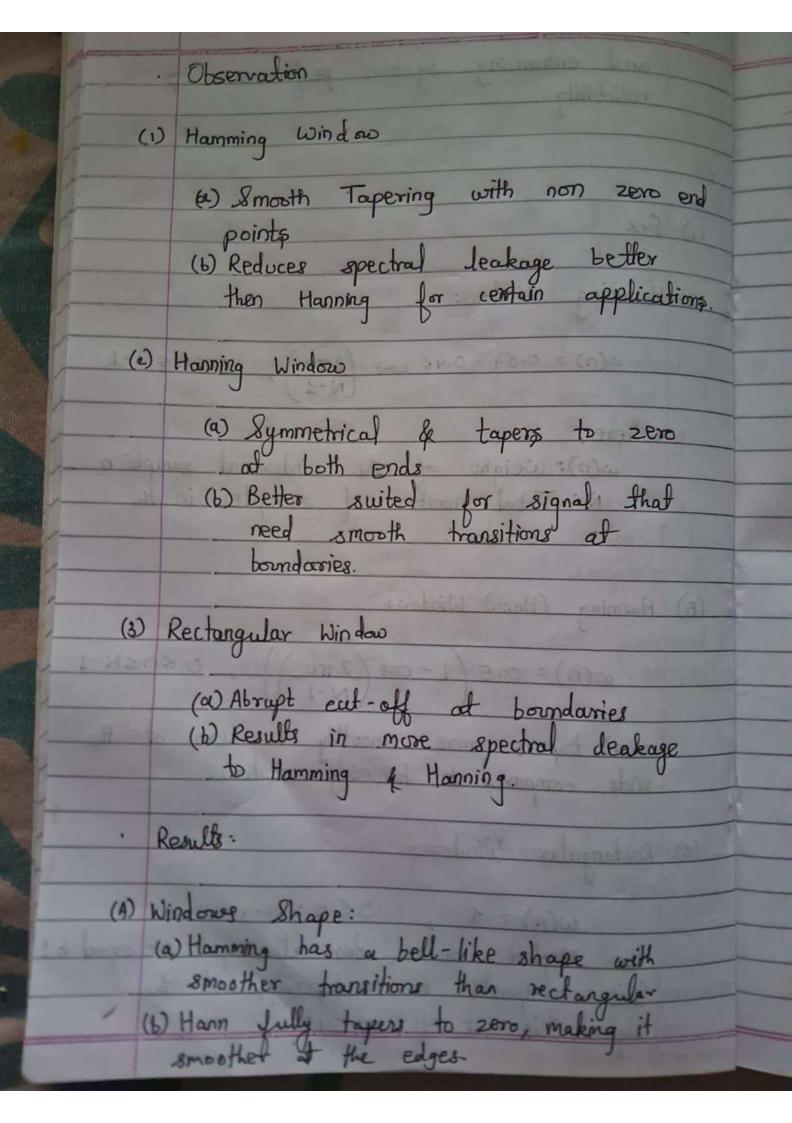
Describe how vibration signals are acquired in a mechanical system. 91 P6 Vibration signal in a mechanical system are acquired using sensors like accelerometer, velocity, displacement placed at critical points to capture motion. These sensors converts mechanical vibrations into electrical signal which are then conditioned using amplification & filtering The signals are transmitted to a Data Acquisition System (DAQ) for analog-to-digital conversion, ensuring a sufficient sampling rate to meet the Nyquist criterion.

The acquired digital signal are processed and analyzed using software like MATLAB, Result & Conclusion The MATLAB analysis revealed dominant (c) vitration frequencies at some Hertz, indicating potential resonance near the system's operational range. The exp. demonstrated MATIAB effectiveness in processing le analyzing vibration signals. Vibration analysis is crucial for early lault detection

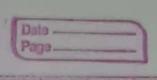




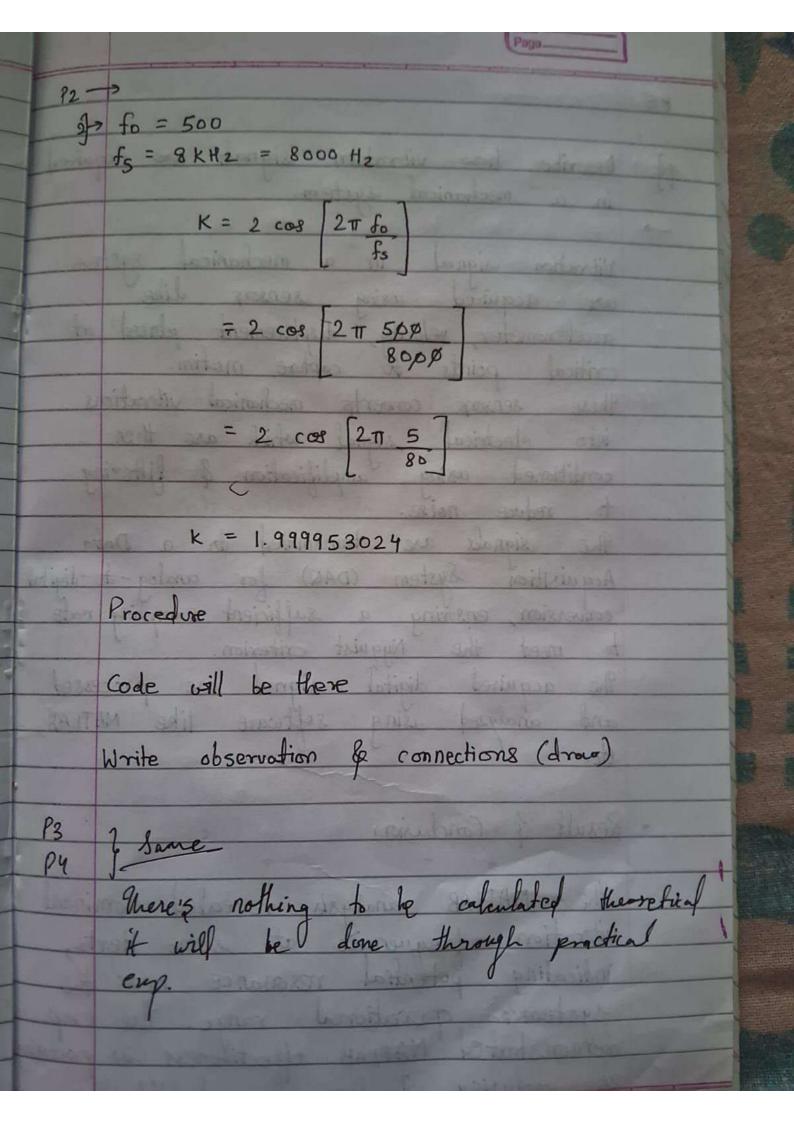
(c) Rectangular is flat across the entire Con chisions: MATLAB provides built-in functions for generating these windows simplifying analysis. Hanning ideal for app. requires smooth po topering & minimal spectral leakage Hamming useful for general purpose windowing, especially when non-zero end points are acceptable Rectangular provides the simplest form but with higher spectral deakage, switches for signals with no tapering requirements

P-7	The self of advertisation of
1	Eqo:
	· Aller Alle
(A)	General equ
	H(s) = 1
	$H(s) = 1$ $\sqrt{1 + \left(\frac{\omega}{\omega_c}\right)^{2}}$
	10 · C-
	ω: freq ωc: cutoff freq n: Filter order
	n: Filter order
(0)	Table 1 1.
(0)	Transfer function.
	H(s) = B(s)
	A(s)
	B(s): Numerodor Polynomial designed to
	ersure Butterworth characteristics.
	for an n-order filter
	$A(s) = \sum_{k=1}^{n} (5 + \omega_c e^{j(2k-1)\pi t/2n})$
	K=1

1) The sampling theorem states that a continuous - time signal can be completely reconstructed from its samples if the sampling freq is at least twice the highest freq component in the signal. sample a signal without aliasing, the sampling rate must be at least twice the signal frequency. Is > 2 fm, This ensure that the simpled signal retains all the freq. info. of the original signal to allow for precise reconstruction Of Aliasing can lead to incorrect feedback causing instability, vibrations, oscillations for ex.: A robotic arm may move erratically if its controller misinterprets aliased position or velocity signals. Proper sampling rate ensures accurate signal reconstruction & precise feedback enabling smooth & reliable operation of such system. 05 Result: (a) fs > 2 fm: The signal is accurately sampled & reconstructed.



- (b) fs 2fm: The signal is sampled with minimal distortion but is at the limit of accurate reconstruction.
- (c) fs < 2fm: Aliasing occurs, leading to significant distortion but is at the limits of accurate reconstruction.



DSP Practicals as per slip P1 -> of what's sampling rate? The sampling rate also known as sampling frequency, refers to the number of samples taken per second when converting a continuous analog signal into discrete digital signal. It's measured in Hertz (H2) (92) Draw waveform for 1/p & 0/p by changing the sampling rate \$ \$\$ Result & Conclusion: Signal without aliasing, the sampling rate must be at least twice the highest freg in the signal.