UNIT–1 : Modulation: Analog, Pulse and Digital (PYQ Solution)

PART–A Answers (≈70 Words Each)

Q1. What is the fundamental need for modulation in electronic communication? (2025)

Modulation means changing a property (amplitude, frequency or phase) of a high-frequency carrier according to the message. We need it because:

- **Small antenna: antenna length ≈ λ/4; for a 3 kHz voice signal it would be 25 km, but for a 100 MHz carrier only 0.75 m.**
- **Multiplexing: many stations share the air using different carriers.**
- **Long range and less noise: high-frequency waves travel farther.**

Q2. Define 'Gain' and 'Attenuation' in the context of a communication system. (2025)

Gain is the increase in signal power by a circuit (amplifier): $A = P_{out}/P_{in}$, in decibels $G_{dB} = 10\log_{10}(P_{out}/P_{in})$.

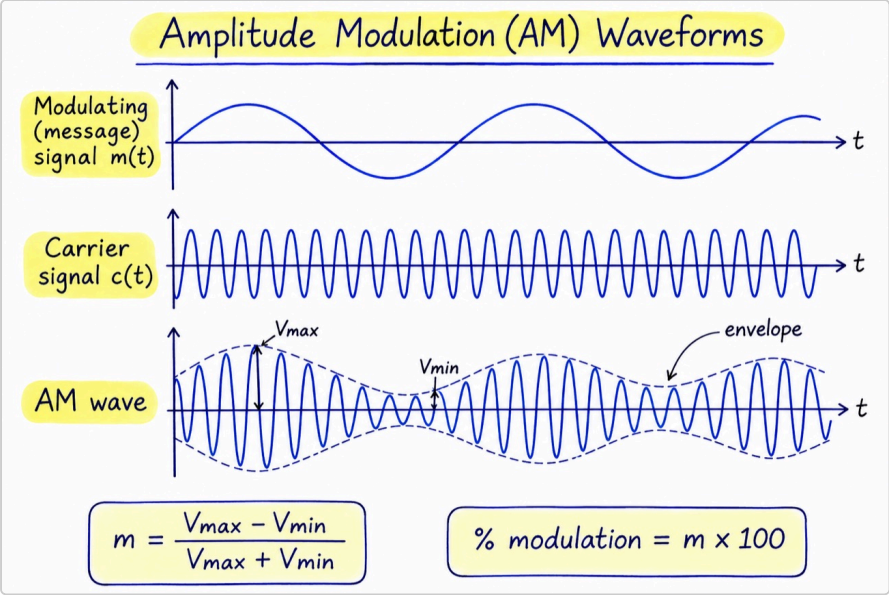
Attenuation is the loss of signal power while travelling through a cable, fiber or air: $P_{out} < P_{in}$, so the dB value is negative.

Example: +20 dB = 100 times gain; −3 dB = power becomes half.

Q3. Define modulation index and percentage modulation for an AM wave. (2022)

Modulation index (m) is the ratio of message amplitude to carrier amplitude: $m = V_m/V_c = \dfrac{V_{max} - V_{min}}{V_{max} + V_{min}}$.

Percentage modulation = m × 100%.



m must be between 0 and 1. m = 1 (100%) is ideal. m > 1 causes over-modulation and distortion.

Q4. Distinguish between low level and high level modulation. (2022)

Low level modulation	High level modulation
Modulation is done at low power, before the final amplifier	Modulation is done at the final (high power) amplifier stage
Needs **linear** amplifiers after it (class A/B)	Uses efficient **class C** amplifiers
Low efficiency, used in TV / SSB	High efficiency, used in AM broadcast

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Q5. Define the image frequency. (2022)

In a superheterodyne receiver, the image frequency is an unwanted signal that also produces the same intermediate frequency (IF) as the wanted signal.

$$f_{si} = f_s + 2f_{IF}$$

Example: f_s = 1000 kHz, IF = 455 kHz → image = 1910 kHz. It is rejected by the RF stage before the mixer.

Q6. Distinguish between PAM and PWM. (2025)

PAM	PWM
Amplitude of pulses changes with the message	**Width** of pulses changes with the message
Width and position are fixed	Amplitude and position are fixed
More affected by noise	Better noise immunity
Lower bandwidth	Higher bandwidth

Q7. List two advantages of Digital Modulation over Analog Modulation. (2025)

1. **Better noise immunity: the receiver only decides 0 or 1, so noise can be removed by regeneration.**
2. **Security and error control: data can be encrypted and errors can be detected and corrected.**

Also: easy multiplexing (TDM), works with computers and storage, efficient use of bandwidth (QPSK, QAM).

Q8. What is frequency translation? (Expected)

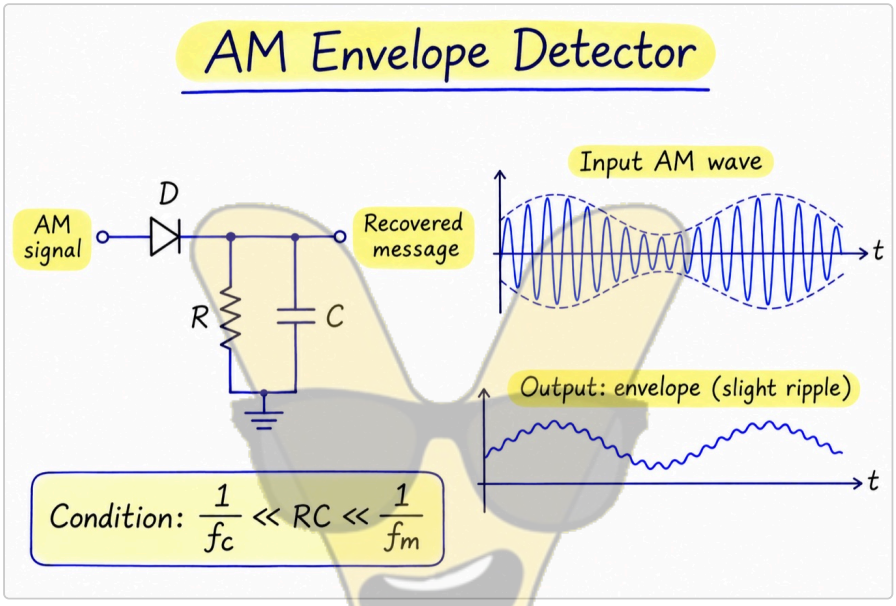
Frequency translation means shifting a signal from one frequency band to another without changing its information. It is done by a mixer (multiplying with a local oscillator), which produces sum and difference frequencies $f_1 \pm f_2$.

Examples: modulation shifts voice up to the carrier band; a superheterodyne receiver shifts all stations to one fixed IF (455 kHz).

PART–B Answers (≈250 Words Each)

Q1. Describe the process of demodulation for an Amplitude Modulated (AM) signal using a simple envelope detector. (2025)

Demodulation is getting the original message back from the modulated wave. In AM the message is in the **envelope** (outline of the peaks), so a simple **envelope detector** is enough.



Circuit: a diode D in series, followed by a resistor R and capacitor C in parallel.

Working:

1. **Rectification:** the diode conducts only on positive half-cycles, so the negative half is removed.
2. **Charging:** on each positive peak the capacitor charges quickly through the diode up to the peak value.
3. **Discharging:** between peaks the diode is off and C discharges slowly through R.
4. So the capacitor voltage follows the **peaks**, i.e. the envelope = the message (with small ripple).
5. A DC blocking capacitor and low-pass filter remove the DC and ripple.

Choice of time constant:

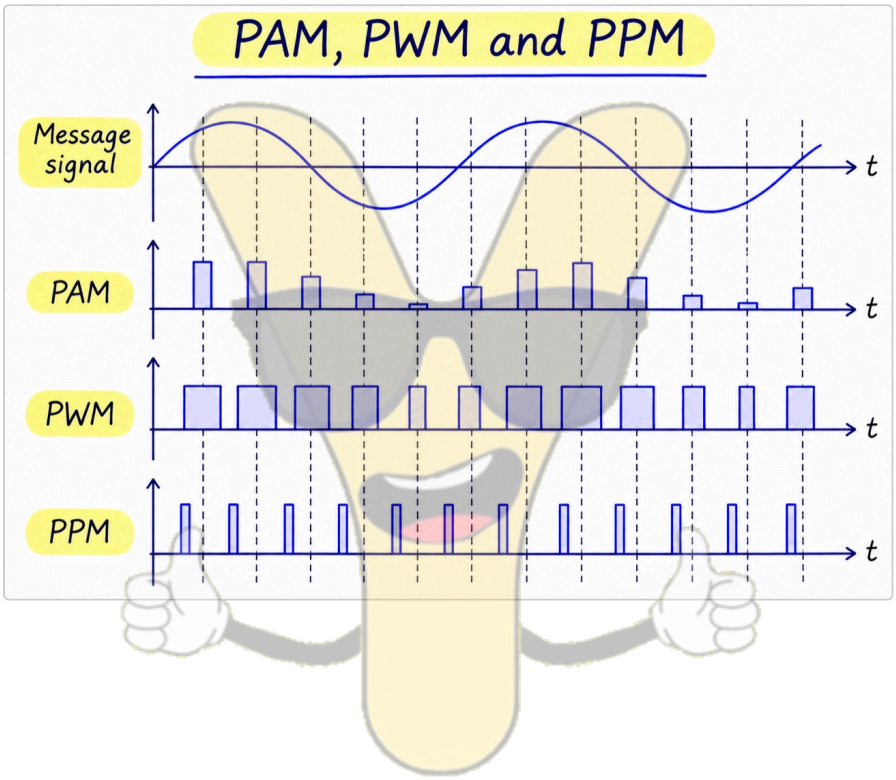
$$\frac{1}{f_c} \ll RC \ll \frac{1}{f_m}$$

- If RC is too small → output follows the carrier (ripple).
- If RC is too large → output cannot follow fast falls of the envelope (**diagonal clipping**).

Advantages: simple, cheap, no local oscillator needed. **Limitation:** works only for normal AM (DSB-FC) with m ≤ 1, not for DSB-SC or SSB.

Q2. Compare PAM, PWM and PPM. (2022)

In **pulse modulation**, the message is sampled and a train of pulses carries it. One property of the pulse changes with the message.



Point	PAM	PWM (PDM)	PPM
What changes	Amplitude	Width (duration)	Position (time)
Fixed	Width, position	Amplitude, position	Amplitude, width
Noise immunity	Poor (noise adds to amplitude)	Better	Best
Transmitted power	Varies	Varies with width	Constant
Bandwidth	Low	High	High
Synchronisation	Not needed	Not needed	Needed
Generation	Sample and hold	Comparator with sawtooth	From PWM using monostable
Use	Inside PCM, TDM	Motor control, class-D audio	Optical / remote control

Simple idea: PAM is "taller or shorter pulses", PWM is "wider or narrower pulses", PPM is "earlier or later pulses".

Q3. Derive an expression of quantization error (in terms of power) for linear quantization. (2022)

Quantization rounds each sample to the nearest fixed level. The difference between the actual sample and the rounded value is the **quantization error**.

Let the signal swing from −V to +V and let there be $L = 2^n$ levels (n bits). Step size:

$$\Delta = \frac{2V}{L}$$

The error e lies between $-\Delta/2$ and $+\Delta/2$, and every value is equally likely, so its probability density is uniform: $f(e) = 1/\Delta$.

Quantization noise power (mean square error):

$$N_q = \int_{-\Delta/2}^{\Delta/2} e^2 \frac{1}{\Delta}\, de = \frac{1}{\Delta}\left[\frac{e^3}{3}\right]_{-\Delta/2}^{\Delta/2} = \frac{\Delta^2}{12}$$

Signal to quantization noise ratio for a full-scale sine wave (power $V^2/2$):

$$\frac{S}{N_q} = \frac{V^2/2}{\Delta^2/12} = \frac{3}{2}L^2 \quad \Rightarrow \quad (S/N_q)_{dB} = 1.76 + 6.02\,n$$

Result: each extra bit improves SQNR by about **6 dB**. Example: 8-bit PCM → 1.76 + 48.2 ≈ 50 dB.

Q4. Give the comparison of Pulse code modulation, Delta modulation, Adaptive delta modulation and Differential pulse code modulation. (2022)

All four convert analog signals to digital, but they code different things.

Point	PCM	DM	ADM	DPCM
What is coded	Full sample value	Only up or down (1 bit)	Up/down with variable step	Difference between sample and prediction
Bits per sample	n (e.g. 8)	1	1	Fewer than PCM (3–4)
Bandwidth	Highest	Lowest	Low	Medium
Main problem	Quantization noise	Slope overload and granular noise	Reduces both problems	Prediction errors
Step size	Fixed	Fixed	Changes with signal slope	Fixed
Complexity	Medium	Simplest	More than DM	More (needs predictor)
Use	Telephony (64 kbps), CD audio	Simple voice links	Voice coding	Speech, video compression

Key idea: neighbouring samples are similar, so DM, ADM and DPCM send only the **change**, which saves bits.

Q5. Derive the expression of error probability of BPSK signal employing coherent reception. (2022)

In **BPSK**, bit 1 → $s_1(t) = +\sqrt{2E_b/T}\cos\omega_c t$ and bit 0 → $s_0(t) = -\sqrt{2E_b/T}\cos\omega_c t$. On a one-dimensional line they are two points at $+\sqrt{E_b}$ and $-\sqrt{E_b}$.

Coherent receiver: multiply the received signal by the same carrier, integrate over one bit (correlator), and decide: output > 0 → 1, output < 0 → 0.

The correlator output is $x = \pm\sqrt{E_b} + n$, where n is Gaussian noise with mean 0 and variance $N_0/2$.

Error when 1 is sent: x becomes negative.

$$P_e = P(n < -\sqrt{E_b}) = \int_{-\infty}^{-\sqrt{E_b}} \frac{1}{\sqrt{\pi N_0}} e^{-n^2/N_0}\, dn$$

Put $z = n/\sqrt{N_0}$:

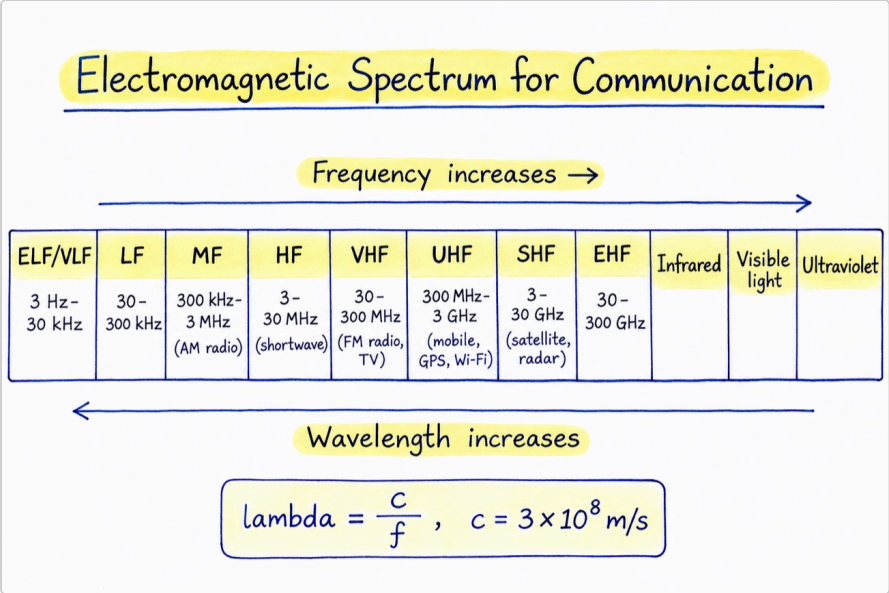
$$P_e = \frac{1}{\sqrt{\pi}} \int_{\sqrt{E_b/N_0}}^{\infty} e^{-z^2}\, dz = \frac{1}{2}\,\text{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right) = Q\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

By symmetry, error for 0 is the same, so this is the **average bit error probability**.

Meaning: BPSK has the lowest error rate of the basic schemes (3 dB better than coherent ASK/FSK). Example: E_b/N_0 = 9.6 dB gives $P_e \approx 10^{-5}$.

Q6. Explain the electromagnetic spectrum used for communication. (Expected)

The **electromagnetic spectrum** is the full range of EM waves arranged by frequency. All wireless communication uses some part of it. Frequency and wavelength are related by $\lambda = c/f$, with $c = 3 \times 10^8$ m/s.



Band	Frequency	Uses
VLF / LF	3 kHz – 300 kHz	Submarine, navigation
MF	300 kHz – 3 MHz	AM radio
HF	3 – 30 MHz	Shortwave, amateur radio
VHF	30 – 300 MHz	FM radio, TV
UHF	300 MHz – 3 GHz	Mobile phones, GPS, Wi-Fi, Bluetooth
SHF (microwave)	3 – 30 GHz	Satellite, radar
EHF (mm-wave)	30 – 300 GHz	5G mm-wave, radar
Infrared, light	~THz	Optical fiber, TV remote

Simple idea: higher frequency → more bandwidth (more data) and smaller antennas, but shorter range and more blocking by walls and rain. The spectrum is limited, so ITU and governments allocate bands by licence.

PART–C Answers (≈500 Words Each)

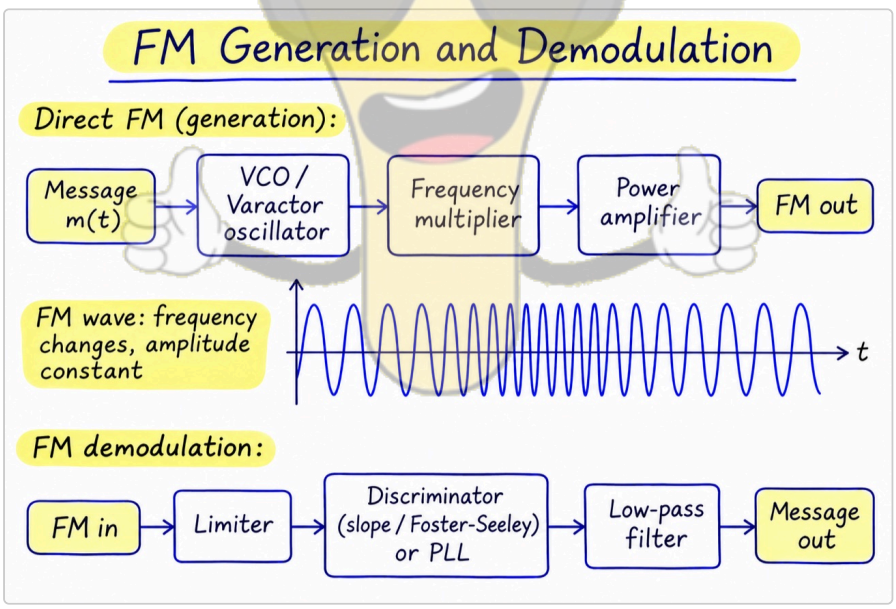
Q1. (a) Explain the generation and demodulation process of Frequency Modulation (FM) using suitable block diagrams. (b) A signal has a power of 50 mW at the input of an amplifier. If the amplifier has a gain of 20 dB, what is the output power in Watts? (2025)

(a) Frequency Modulation

In **FM**, the **frequency** of the carrier changes with the message amplitude, while the amplitude stays constant:

$$s(t) = A_c \cos\left(2\pi f_c t + \beta \sin 2\pi f_m t\right), \quad \beta = \frac{\Delta f}{f_m}$$

Δf = frequency deviation, β = modulation index. Bandwidth by **Carson's rule**: $BW = 2(\Delta f + f_m)$. Example: FM radio Δf = 75 kHz, fm = 15 kHz → BW = 180 kHz.



Generation:

1. **Direct method:** the message drives a **varactor diode** in the tank circuit of an oscillator (or a **VCO**). The capacitance changes with voltage, so the frequency changes with the message. Simple but frequency drifts, so AFC is used.
2. **Indirect (Armstrong) method:** first make narrow-band FM using a stable **crystal oscillator** and a phase modulator (integrate the message first), then use

frequency multipliers and mixers to reach the required deviation and carrier. Very stable.

3. A **power amplifier** (class C allowed, because amplitude is constant) feeds the antenna.

Demodulation:

1. **Limiter:** clips amplitude variations (noise), because information is only in frequency.
2. **Discriminator:** converts frequency changes into voltage changes. Types: slope detector, **Foster-Seeley discriminator**, ratio detector, or a **PLL** (phase locked loop) whose control voltage follows the input frequency.
3. **Low-pass filter / de-emphasis:** gives the original message.

Advantages of FM: much better noise immunity, constant power, good audio quality.
Disadvantage: needs larger bandwidth.

(b) Numerical

Given: P_{in} = 50 mW, Gain = 20 dB.

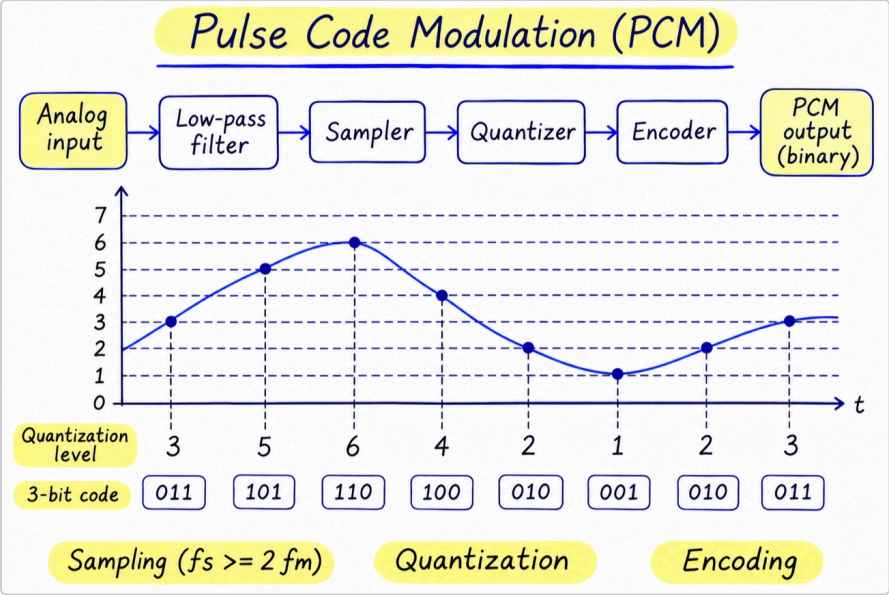
$$G = 10 \log_{10} \frac{P_{out}}{P_{in}} \Rightarrow \frac{P_{out}}{P_{in}} = 10^{20/10} = 100$$

$$P_{out} = 100 \times 50 \text{ mW} = 5000 \text{ mW} = \mathbf{5 \text{ W}}$$

Q2. (a) Describe the process of Pulse Code Modulation (PCM). Clearly explain the three main steps involved: sampling, quantization and encoding. (b) Compare ASK, FSK and PSK digital modulation techniques based on their bandwidth efficiency, noise immunity and complexity. (2025)

(a) Pulse Code Modulation

PCM converts an analog signal (like voice) into a stream of binary numbers. It is the base of digital telephony (64 kbps per call) and CD audio.



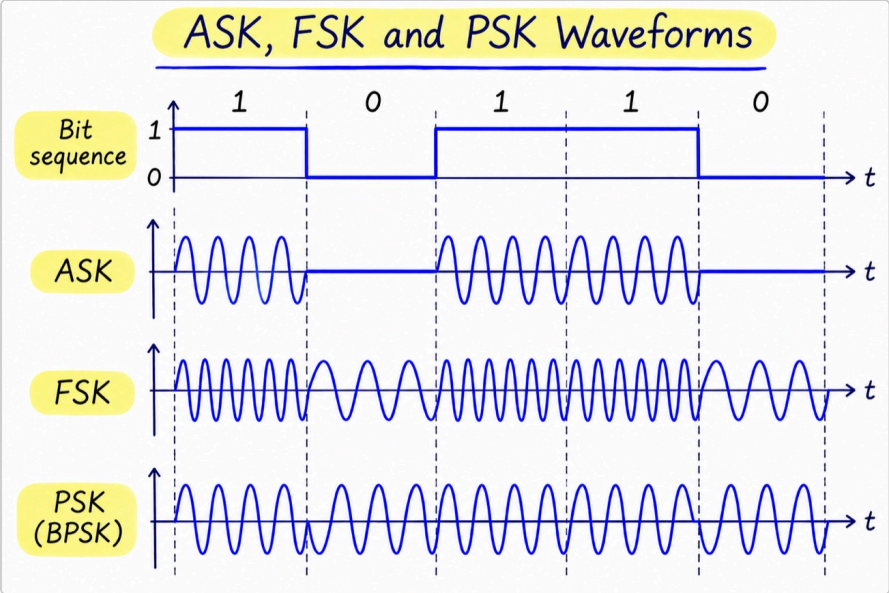
Block diagram: Low-pass filter → Sampler → Quantizer → Encoder → (channel with regenerative repeaters) → Decoder → Reconstruction filter.

1. **Sampling:** the signal is measured at regular instants. By the **Nyquist theorem**, $f_s \geq 2f_m$. Voice (up to 4 kHz) is sampled at 8 kHz. The low-pass (anti-aliasing) filter removes frequencies above $f_s/2$ so that **aliasing** does not occur. Output = PAM pulses.
2. **Quantization:** each sample is rounded to the nearest of L fixed levels ($L = 2^n$). This adds a small **quantization error** (noise power $\Delta^2/12$). Telephony uses non-uniform (μ-law / A-law) quantization so that weak sounds get finer steps.
3. **Encoding:** each level number is written as an n-bit binary code. Example: level 5 with 3 bits → 101. Telephony: 8 bits × 8000 samples/s = **64 kbps**.

At the receiver: regenerative repeaters clean up noisy pulses; the decoder turns codes back into levels, and a low-pass filter smooths them into the analog signal.

Advantages: high noise immunity, regeneration, easy TDM, encryption.
Disadvantage: needs more bandwidth, quantization noise.

(b) ASK vs FSK vs PSK



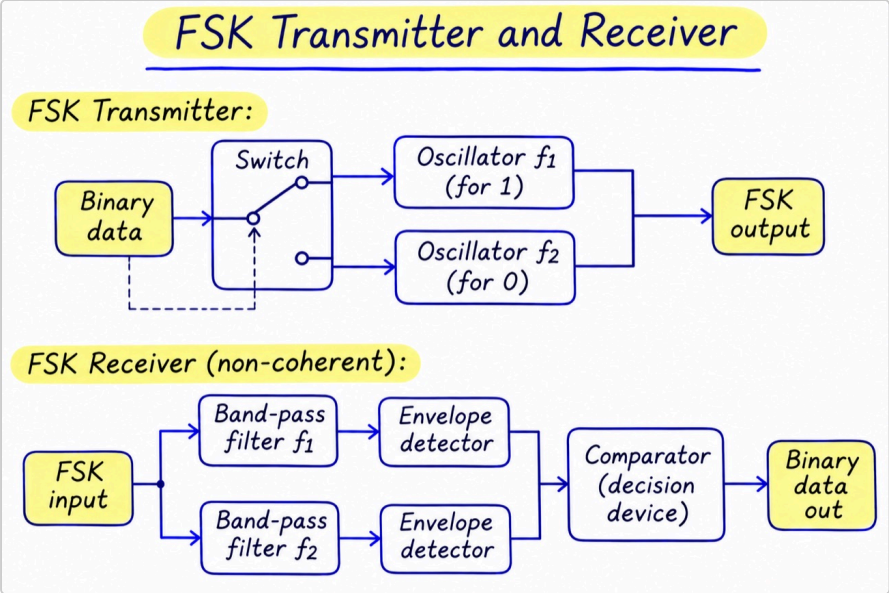
Point	ASK	FSK	PSK (BPSK)
What changes	Amplitude (on/off)	Frequency (f1 for 1, f2 for 0)	Phase (0° / 180°)
Bandwidth	$2R_b$ (efficient)	$2R_b + \|f_1 - f_2\|$ (widest)	$2R_b$
Bandwidth efficiency	Medium	Lowest	Good (QPSK doubles it)
Noise immunity	Poor (noise changes amplitude)	Better	Best (lowest error rate)
Complexity	Simplest	Medium	Complex (needs coherent carrier)
Use	Optical fiber (on-off keying), RFID	Low-speed modems, Bluetooth (GFSK)	Wi-Fi, satellite, cellular

Conclusion: PSK gives the best performance, ASK is the simplest, FSK is in between with the most bandwidth.

Q3. Explain the transmitter and receiver for the following: (a) FSK (b) QPSK. (2022)

(a) FSK (Frequency Shift Keying)

In **FSK**, bit 1 is sent as frequency f_1 and bit 0 as frequency f_2, with constant amplitude.



Transmitter: two oscillators produce f_1 and f_2. The binary data controls an electronic switch: when the bit is 1, f_1 is connected to the output; when 0, f_2 is connected. (Another method: data drives a VCO directly, giving continuous-phase FSK.)

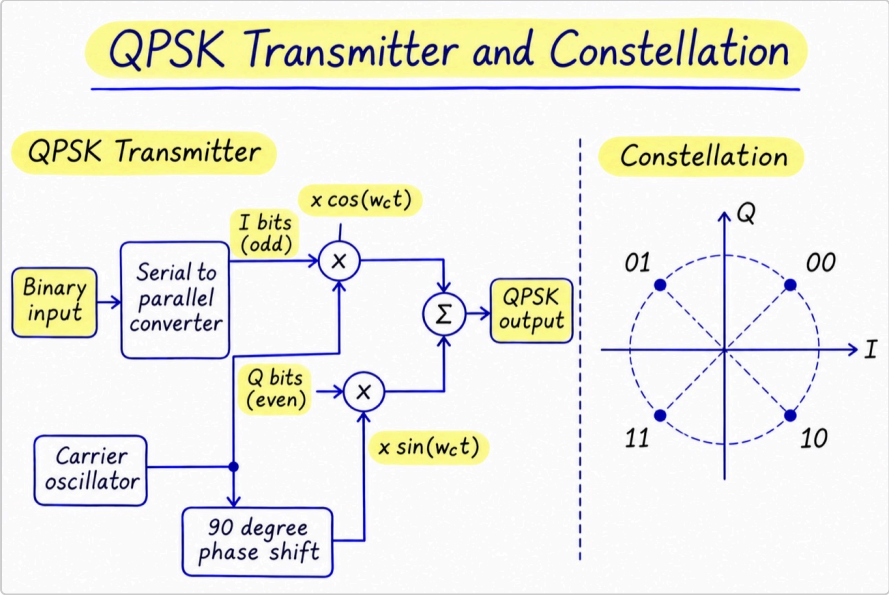
Receiver (non-coherent):

1. The received signal goes to two **band-pass filters**, one tuned to f_1 and one to f_2.
2. Each filter output goes to an **envelope detector**.
3. A **comparator** compares the two outputs: if the f_1 branch is larger → 1, else → 0.

A coherent receiver uses two correlators (multiply by $\cos 2\pi f_1 t$ and $\cos 2\pi f_2 t$) and is more accurate.

(b) QPSK (Quadrature Phase Shift Keying)

QPSK sends **2 bits per symbol** using 4 phases (45°, 135°, 225°, 315°). So it needs half the bandwidth of BPSK for the same bit rate.



Transmitter:

1. A **serial-to-parallel converter** splits the bit stream into two streams: odd bits → **I (in-phase)**, even bits → **Q (quadrature)**, each at half the bit rate.
2. A polar NRZ encoder converts bits to ±1.
3. The I stream multiplies $\cos \omega_c t$; the Q stream multiplies $\sin \omega_c t$ (from the same oscillator through a 90° phase shifter).
4. An adder combines both: $s(t) = I \cos \omega_c t + Q \sin \omega_c t$. Each dibit (00, 01, 11, 10) gives one of 4 phases.

Receiver (coherent):

1. A carrier recovery circuit regenerates $\cos \omega_c t$ and $\sin \omega_c t$.
2. The received signal is multiplied by both and integrated over one symbol (two correlators).
3. Each branch decides its bit by sign (+ → 1, − → 0).
4. A **parallel-to-serial converter** combines the I and Q bits into the original stream.

Advantage: same error rate as BPSK but double data rate in the same bandwidth. Used in satellite, CDMA, Wi-Fi and DTH.
