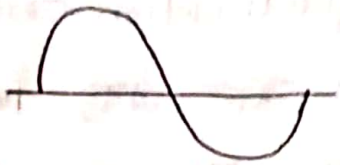
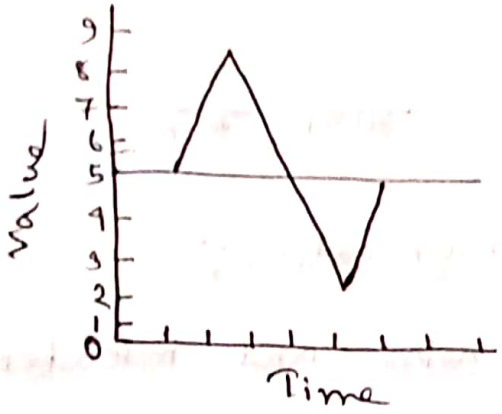
Chapter - 11

Digital Communication

→ analog signal - sine wave

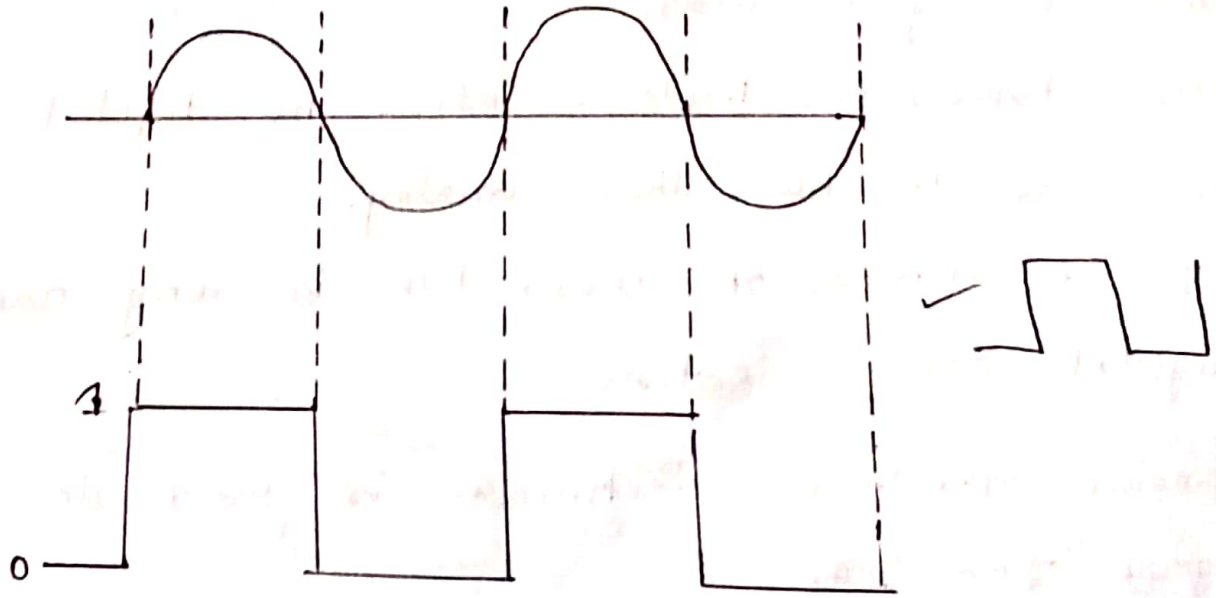


→ analog to digital



digitalization

⊛ Binary representation of a sine-wave signal:



* Positive সমস্ত value → 1

Negative " " → 0

* sine wave এ অনেক গুলো continuous point আছে, সেই Point গুলোকে কমানোর জন্য digitalization করা হয়। সেগুলো একটা fixed range এ নিয়ে আসা হয় এবং কিছু linear lines দিয়ে wave কে express করা হয়, যার ফলে noise এর effect অনেক কমানো possible. এরপর হটি value 0, 1 এ convert করা হয়

* digital data এ কেন নিয়ে আসা হয়?

* Advantages of digital communication:

1) The effect of distortion, noise and interference is much less in digital signals as they are less affected.
2) Digital circuits are more reliable.
3) Digital circuits are easy to design and cheaper than analog circuits.
4) The hardware implementation in digital circuits, is more flexible than analog.
5) The occurence of cross-talk is very rare in digital communication.
6) Spread spectrum technique is used to avoid signal jamming.

7) The configuring process of digital signals is easier than analog signals.

8) Digital signal can be saved and retrieved more conveniently than analog signals.

9) The capacity of the channel is effectively utilized by digital signals.

(As the signals are digitized, there are many advantages of digital communication over analog communication, such as –) ↑

⊛ Elements of digital Communication:

The elements of digital communication system is represented by the following block diagram for the ease of understanding.

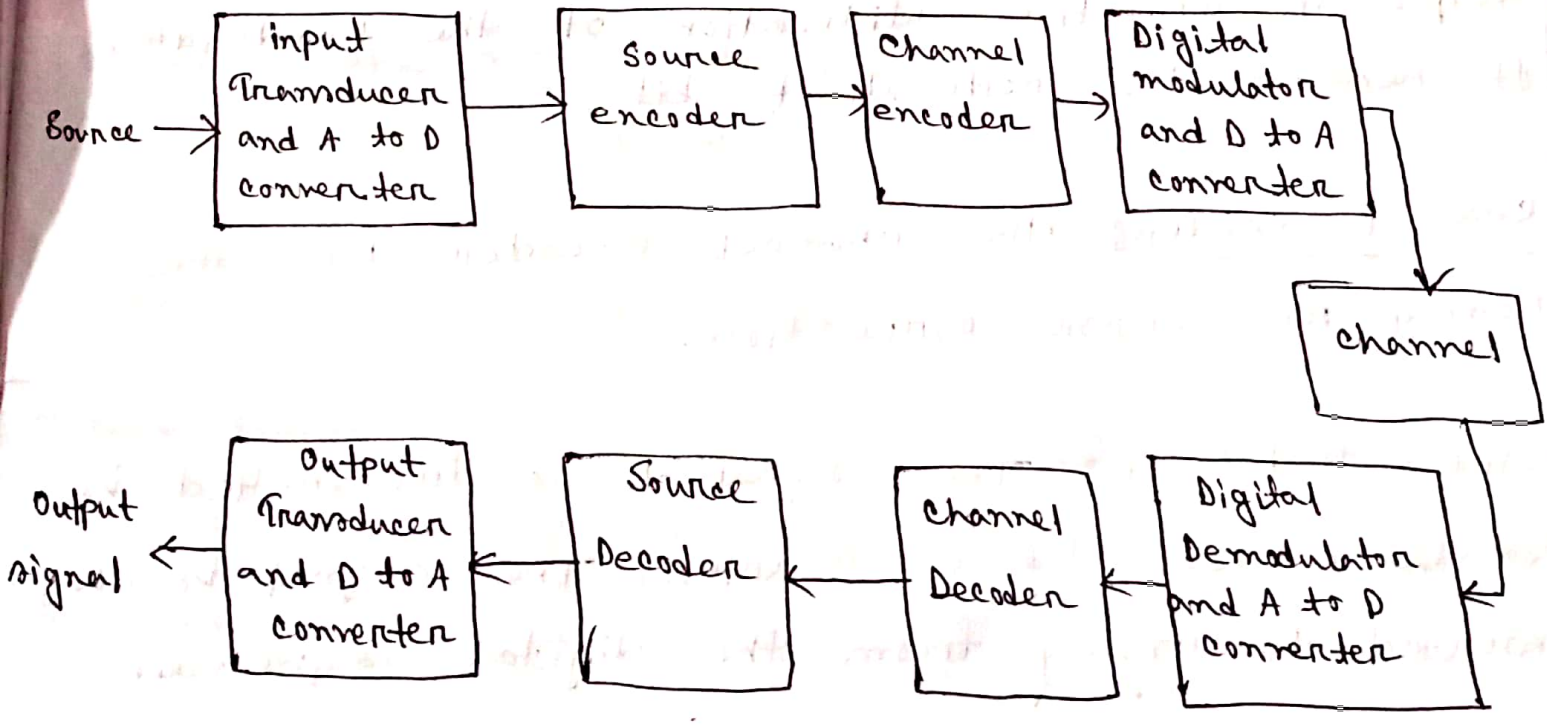


fig: Basic elements of a Digital Communication System

following are the sections of the digital communication system.

Source: The source can be an analog signal.

Example: A sound signal.

Input Transducer: This is a transducer which takes a physical input and converts it to an electrical signal. Example: Microphone. This block also consists of an analog to digital converter where a digital signal is needed for further processes.

Source encoder: This source encoder compresses the data into minimum number of bits. This process helps in effective utilization of the bandwidth. It removes the redundant bits.

Channel encoder: The channel encoder does the coding for error correction.

Digital Modulator: The signal to be transmitted is modulated here by a carrier. The signal is also converted to analog from the digital sequence.

Channel: The channel or a medium, allows the analog signal to transmit from the transmitter end to the receiver end.

Digital Demodulator: This is the first step at the receiver end. The received signal is demodulated as well as converted again from analog to digital.

Channel Decoder: The channel decoder, after detecting the sequence, does some error corrections.

Source Decoder: The source decoder recreates the source output.

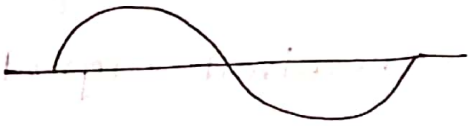
Output transducer: This is the last block which converts the signal into the original physical form, which was at the input of the transmitter. It converts the electrical signal into physical output. Example: loud speaker.

Output Signal: This is the output which is produced after the whole process. Example: the sound signal received.

Pulse Code Modulation (Digital Modulation)

→ ⊓⊔ → pulse

→ যখন frequency কম অনুযায়ী pulse আসতে থাকে



sine wave input

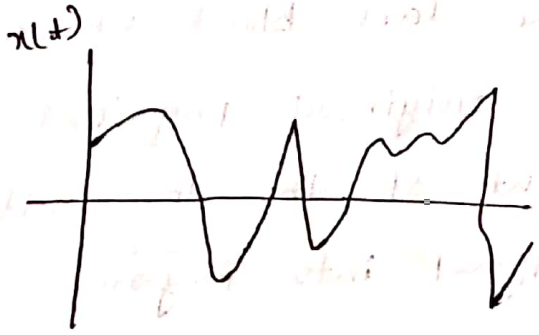
PCM output

top — 1

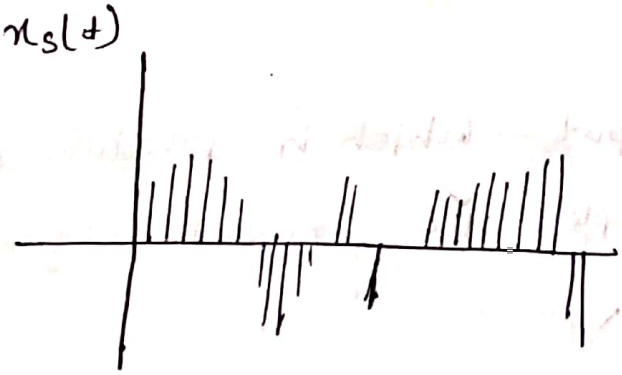
low — 0

* mainly 2টি step → analog data কে digital data → convert করা হয়ে থাকে

1) Sampling
2) Quantization



→ continuous-time signal



→ sampled signal

Sampling rate

$$\text{Sampling frequency} = \frac{1}{T_s}$$

$$= f_s$$

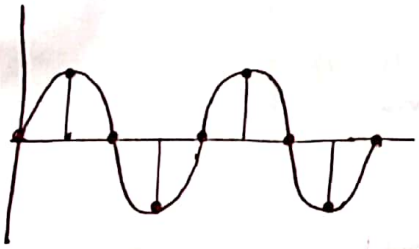
where, $T_s \rightarrow$ sampling time

$f_s \rightarrow$ sampling frequency / rate

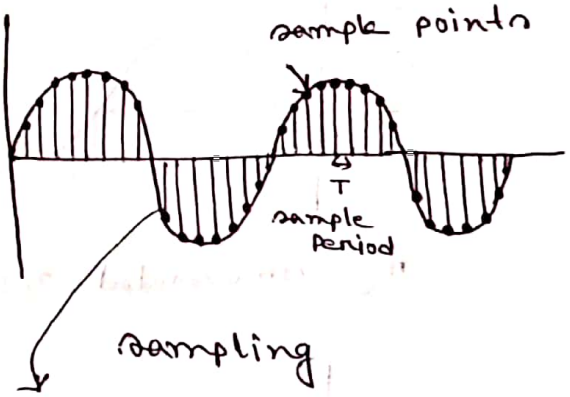
(analog frequency তে কয়টা sample তুলি তার পরিমাণ)

Nyquist Rate

$f_s = 2W$ [W = analog signal এর highest frequency



মূল signal



sampling

ডিজিটাল কয়টা sample তুলি 2র তার rate হবে highest frequency এর 2গুন।

This rate of sampling is called as Nyquist rate

Quantization:

→ sample গুলোকে তিন্ন মানের গুলোতে Packatization করা।

ⓐ Pulse Modulation

* Pulse amplitude analog amplitude এর মত।

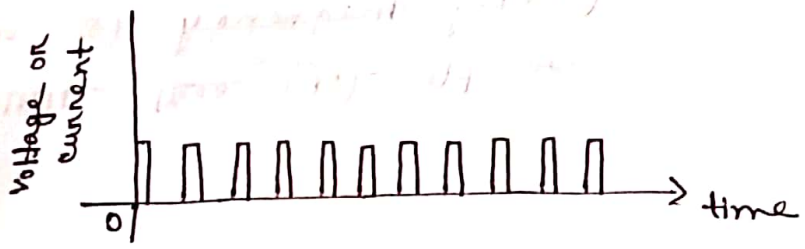


fig: unmodulated pulse carrier waveform

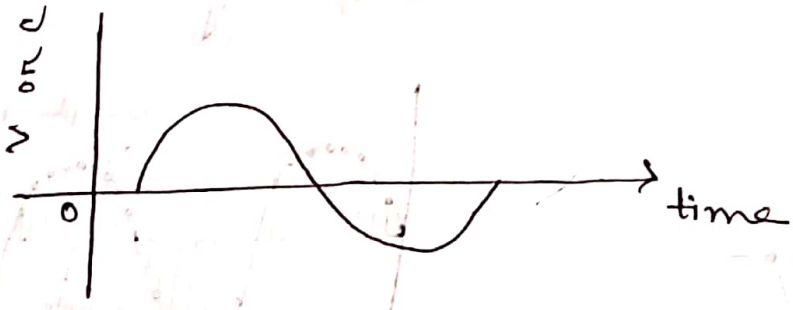


fig: sinusoidal modulating signal

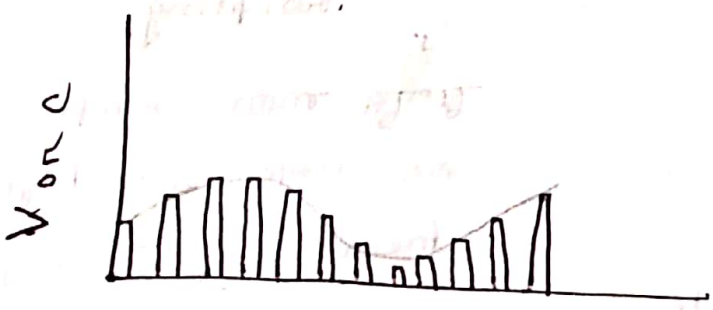


fig: pulse amplitude modulated carrier wave (PAM)

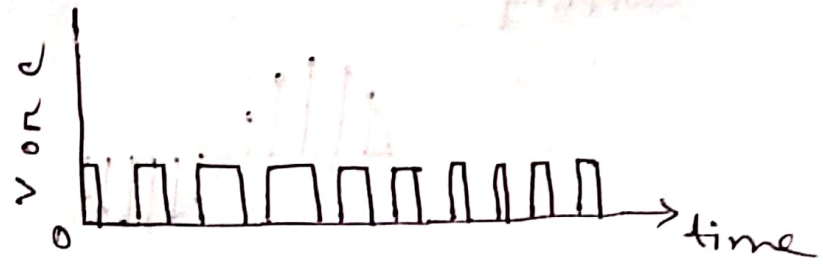


fig: pulse width modulated carrier wave (PWM)

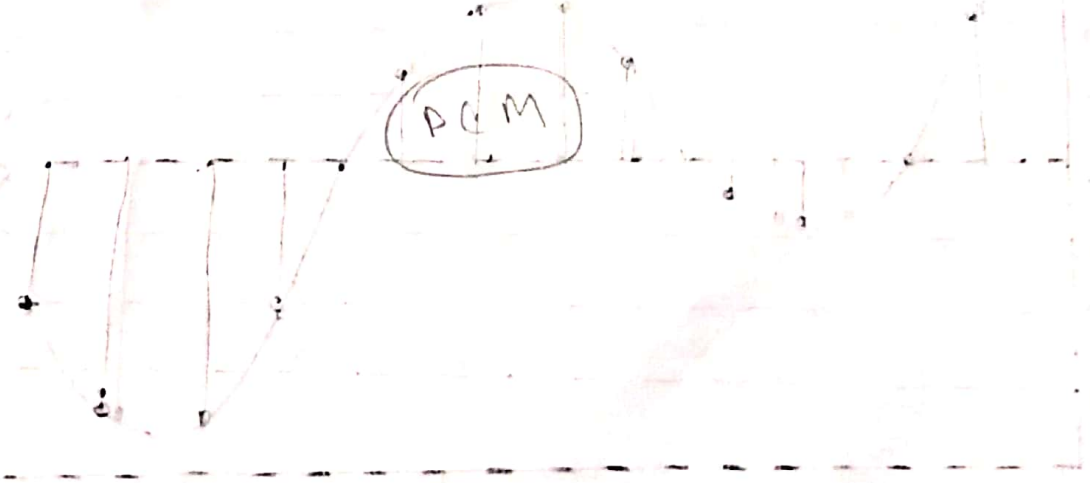
amplitude high হলে শব্দ আস্তে low হলে চুরে যাবে

fig: pulse position modulated carrier wave (PPM)

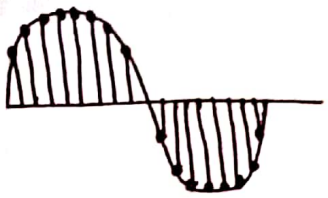
high amplitude এর frequency collect করব, low frequency বাদ দিবো.

fig: pulse frequency modulated carrier wave (PFM)

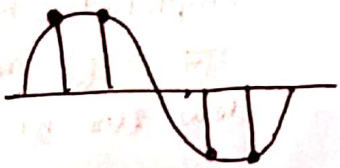
PCM



Sampling



high sample rate



→ Low sample rate

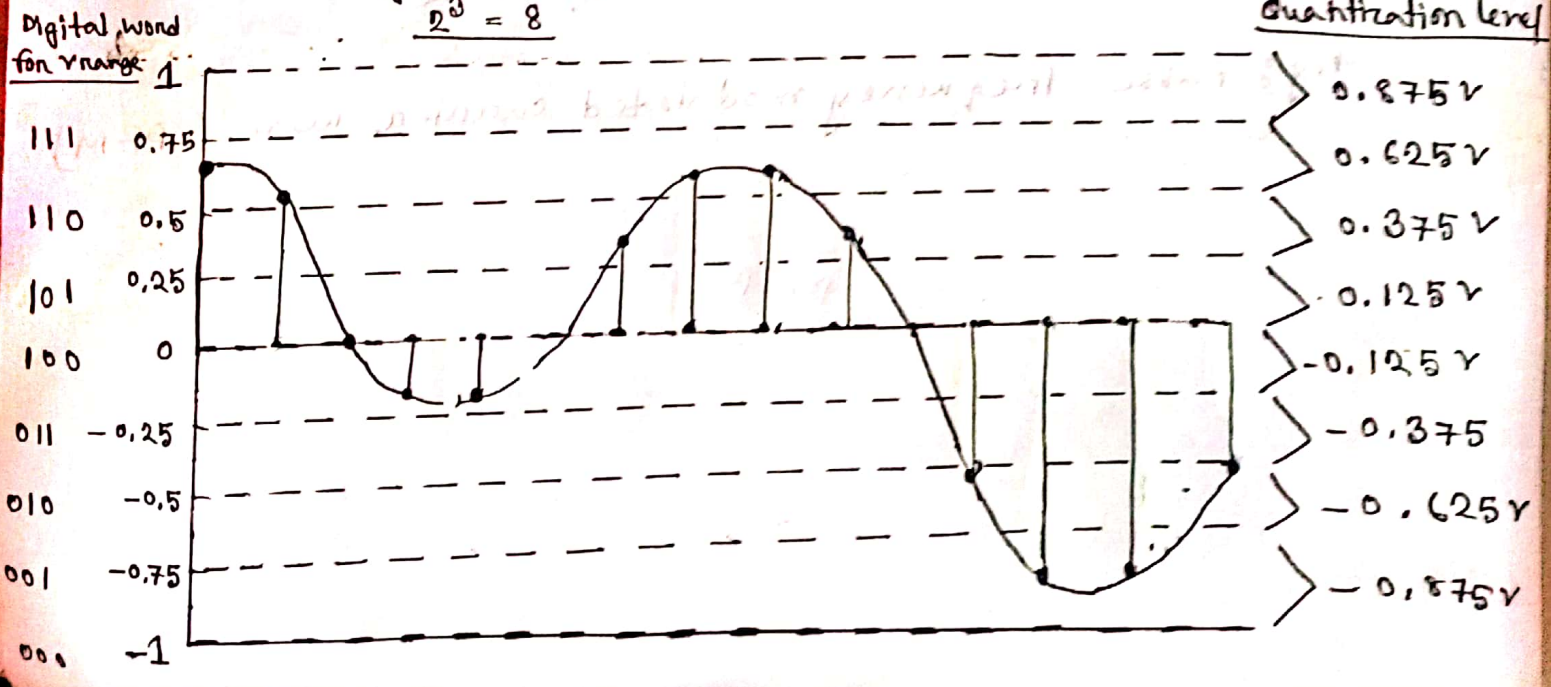
* যত বেশি sample data হবে তত বেশি down করে original continuous data এর কাছাকাছি data পাওয়া যাবে।

Quatizing / Encoding:

→ sample গুলোকে round করবে।

→ quantization level $2^N - 1$ or $1 - 2^N$

→ 3 bit analog to digital converter এর জন্য total $2^3 = 8$



* quantizer's resolution, $q = \frac{v_{max} - v_{min}}{2^N}$ (v_{max} → highest amplitude, v_{min} → lowest amplitude)

(interval- প্রতিটি level এর মধ্যে)/ $= \frac{1-(-1)}{8}$

distance $= 0.25\ v$

* 101, 100, ... → প্রত্যেকটা sample এর জন্য একটা একটা code.

* The binary representation of the above signal is:

110 101 100 011 011 100 110 110 100 010 000 000 001

* এই signal এর জন্য 1 sec এ কয়টা করে sample আসবে / কয়টা bit আসবে:

bit rate, $R_b = \frac{3 \text{ bits}}{\text{sample}} \times \frac{2000 \text{ samples}}{\text{sec}}$

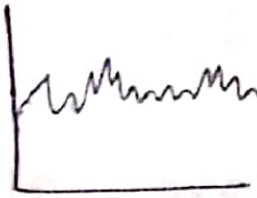
$= 6000$ bit/sec (bps)

no. of bit per sample × 1 টা frequency এ কয়টা sample আছে.

⊛ Quatization error / Quantization Noise:

→ The difference between an input value and it quantized value is called Quantization error.

* noise কমানোর উপায় 2টি — যদি no. of bit increase করা যায় অর্থাৎ level যত বাড়বে noise কমবে



→ 3 bit

→ 4bit (amplitude difference কম 2টি)

(*) Basic Step of Pulse Code Modulation:

1) Sampling:

→ sample voltages at fixed signals along the waveform.

→ how often we measure voltage
→ the more often the better frequency
→ makes it twice its frequency.

2) Quantization:

→ samples are converted into discrete values called quantizing values.
→ greater number of levels, greater the accuracy of representation of the signal.

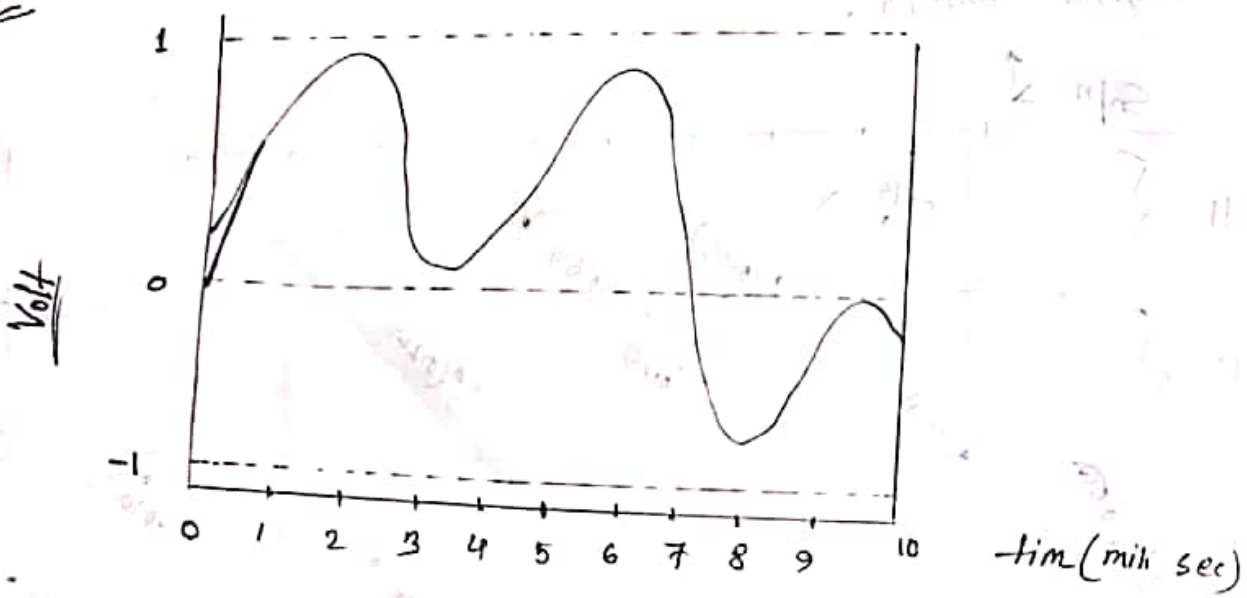
(*) PCM: PCM is a method used to digitally represent sampled analog signals.

(*) Nyquist Rate: The Nyquist rate must be twice the highest bandwidth of the message signal $m(t)$.

$$f_S = 2W$$

W = highest frequency of analog signal.

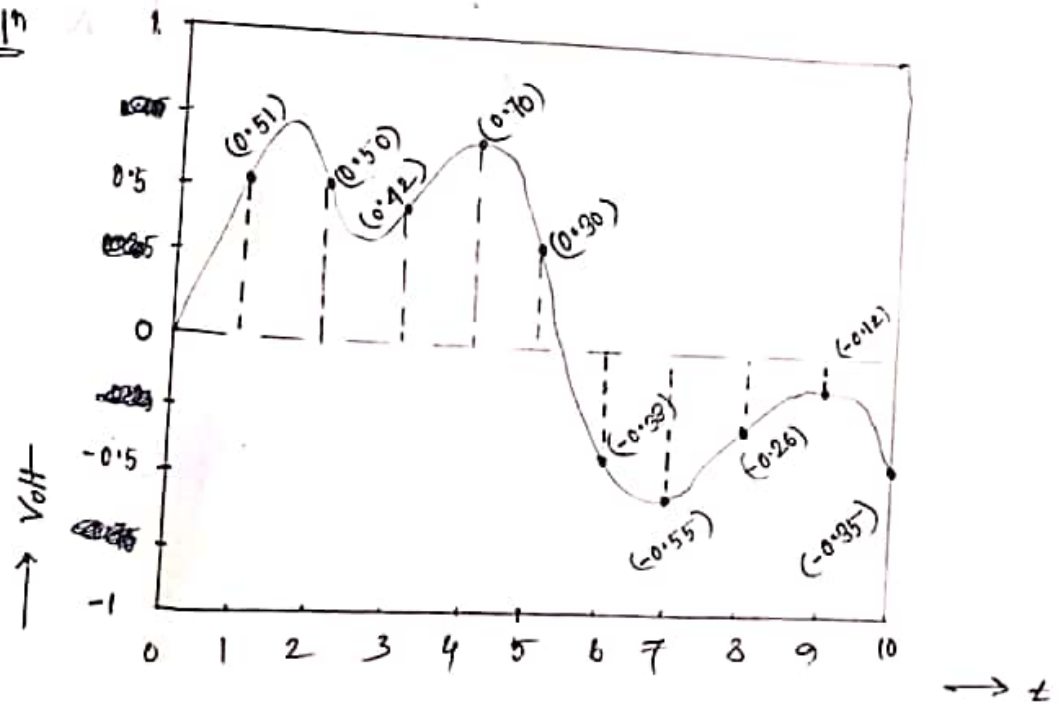
23.2



Consider the analoge waveform. this waveform is sampled at a 500 Hz rate & quantized with a 2 bit quantizer (A/D Converter) the input rang is -1 to +1 V.

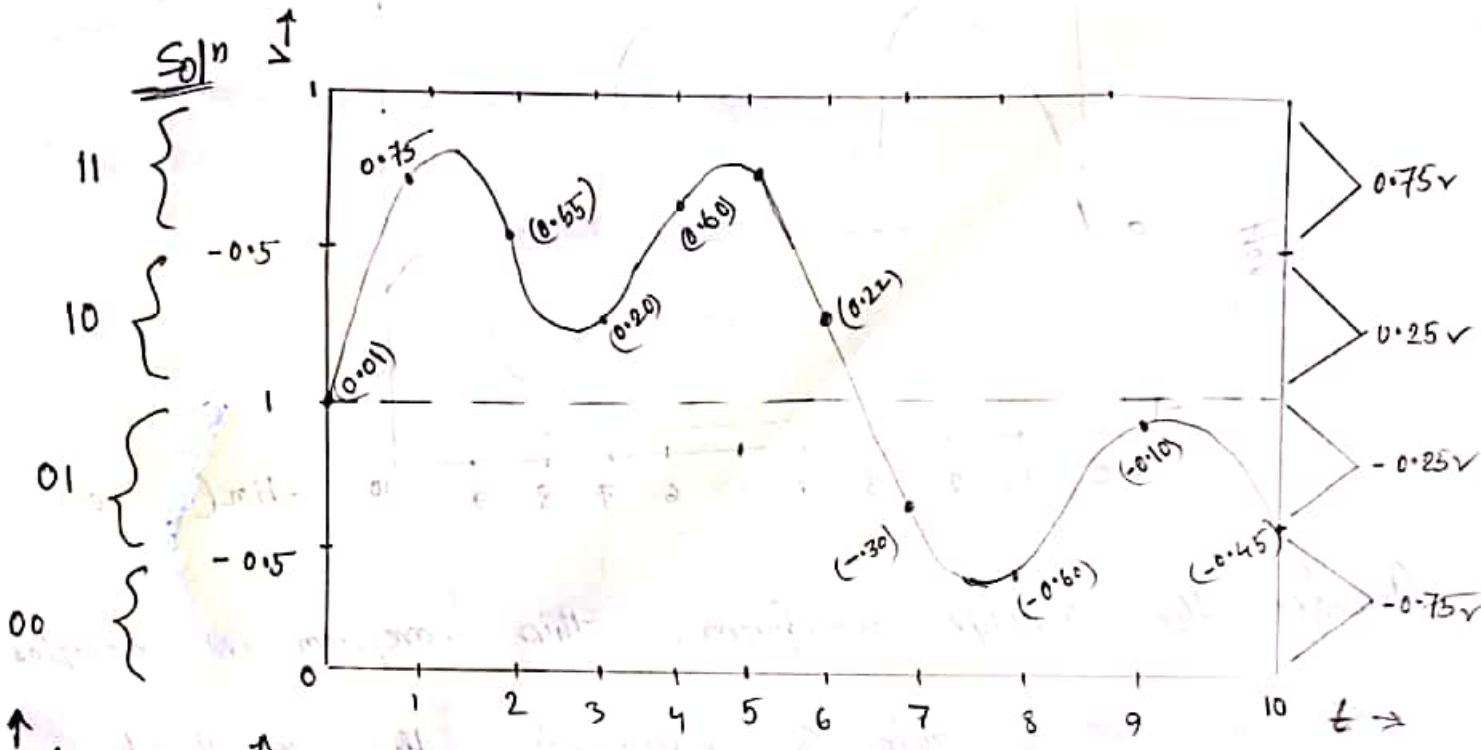
@ Circle the sample points (first sample is at time t=0 sec)

Soln



(b) Indicate the quantization intervals & corresponding digital words.

Soln



Digital words

Intervals

Interval, $q = \frac{V_{max} - V_{min}}{2^N}$

$= \frac{1 - (-1)}{2^2}$

$= 0.5$ V.

given,

$V_{max} = 1$

$V_{min} = -1$

$N = 2$

(c) Indicate the digital word assign to each sample point.

Soln:

Sample point	Digital word
0.01	10
0.75	11
0.55	11
0.20	10
0.60	11
0.22	10
-0.30	01
-0.60	00
-0.10	01
-0.45	01

(d) what is the stream of binary bits generated after the A/D Conversion is complete?

Soln:

Binary representation:

10 11 11 10 11 10 01 00 01 01

(e) what is resulting bit rate from A/D?

Soln

we know,

bit rate,

$$R_b = Nf = 2 \times 500$$

$$= 1000 \text{ bits/sec}$$

$\amalg$

given,

$f = 500$ Hz or $\frac{\text{Sample}}{\text{sec}}$

$N = 2$ bit