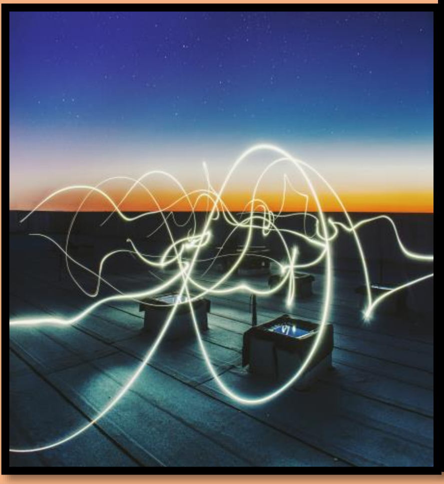  

COMMUNICATION SYSTEM FUNDAMENTALS

MODULATION TECHNIQUES

Nur Suhana binti Suhadi
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POLITEKNIK MALAYSIA
PORT DICKSON

DEP30013
COMMUNICATION SYSTEM FUNDAMENTALS

MODULATION TECHNIQUE

Perpustakaan Negara Malaysia Cataloguing-in-Publication Data

Nur Suhana Suhadi, 1973-
DEP30013 : COMMUNICATION SYSTEM FUNDAMENTALS : MODULATION TECHNIQUE / Nur Suhana binti Suhadi, Azilah binti Asri.
Mode of access: Internet
eISBN 978-967-2897-23-1
1. Modulation (Electronics).
2. Demodulation (Electronics).
3. Pulse modulation (Electronics).
4. Pulse techniques (Electronics).
5. Government publications--Malaysia.
6. Electronic books.
I. Azilah Asri, 1983-. II. Title.
621.381536

PUBLISHED BY:

Politeknik Port Dickson

KM14, Jalan Pantai, 71050 Si Rusa

Port Dickson, Negeri Sembilan

AUGUST 2021

ACKNOWLEDGEMENT

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We would like to convey our utmost gratitude to the Department of Polytechnic and Community College Education particularly the E-learning and Instructional Division (BIPD) for funding our e-book project.

We hereby declare that this module is our original work. To the best of our knowledge it contains no materials previously written or published by another person. However, if there is any, due acknowledgement and credit are mentioned accordingly in the e-book.

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MODULATION & DEMODULATION

MODULATION : The process of **changing** the characteristics of the **carrier signal** whether **amplitude, frequency or phase** according to the **change of the information signal**.

MODULATION PROCESS

Information Signal (Low Frequency) + **Carrier Signal (High Frequency)** → **MODULATED SIGNAL (HIGH FREQUENCY)**

The circuit that performs the modulation process is known as a **Modulator**. The information is either an analog or digital signal where it will combine with the carrier signal to produce a high frequency modulated signal.

DEMODULATION PROCESS

Modulated Signal (High Frequency) − **Carrier Signal (High Frequency)** → **DEMODULATED SIGNAL (LOW FREQUENCY)**

DEMODULATION : The inverse of modulation proses. It is a technique to extract the information signal from a modulated signal. The circuit that performs the demodulation process is known as a **Demodulator**. Demodulated signal is the Original Information Signal.

MODULATION & DEMODULATION

MODULATION & DEMODULATION NEEDS

1. To **increase** the frequency of an information signal.
2. **Converting** analogue signals to digital signals and vice versa.
3. Each station (source) can modulates its information to a different frequency band or channel to **avoid** interfering with each other.
4. To **maximise** the signal's bandwidth.
5. **Multiplexing** a larger number of input signals.
6. To **reduce** the size and height of antenna.
7. **Reduce** the complexity of the equipment.

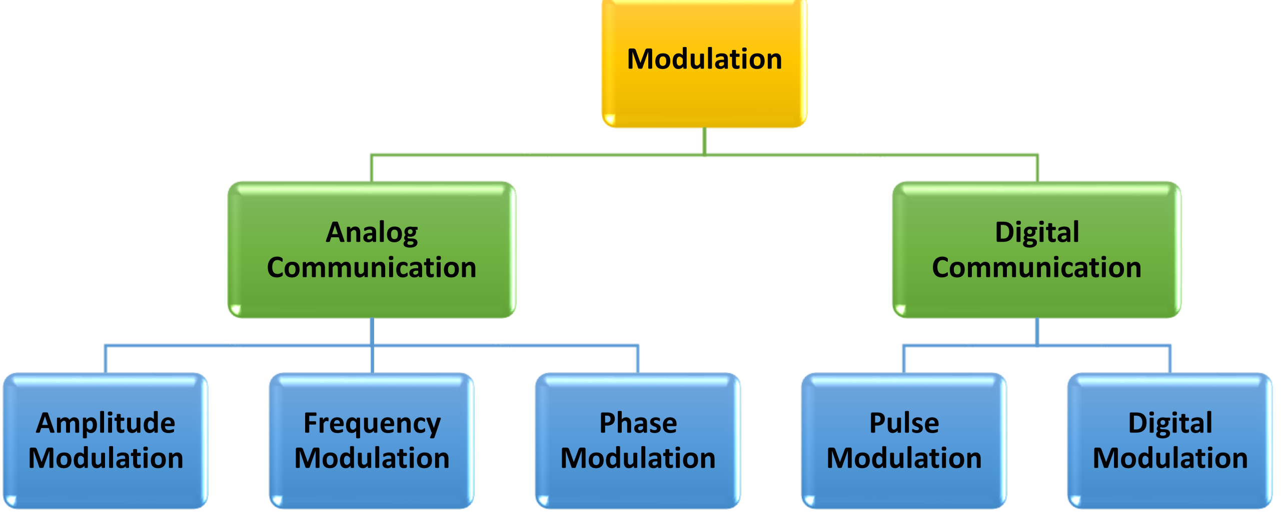
TYPES OF MODULATION

TYPES OF MODULATION

ANALOG MODULATION - The waveforms of the **Information Signal and the Carrier Signal are analog**.

PULSE MODULATION - The waveform of the **Information Signal is analog**, while the waveform of the **Sampling Signal is digital**.

DIGITAL MODULATION - The waveform of the **Information Signal is digital**, while the waveform of the **Carrier Signal is analog**.



TYPES OF MODULATION

ANALOG MODULATION

ANALOG MODULATION

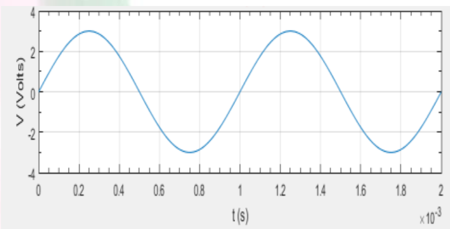
- Amplitude Modulation **(AM)**
  - **Amplitude (Vc)** of the **analog carrier signal** is varied proportional to the amplitude of analog information signal.
- Frequency Modulation **(FM)**
  - **Frequency (fc)** of the **analog carrier signal** is varied proportional to the amplitude of analog information signal.
- Phase Modulation **(PM)**
  - **Phase (Θc)** of the **analog carrier signal** is varied proportional to the amplitude of analog information signal.

AMPLITUDE MODULATION (AM)

AMPLITUDE MODULATION (AM) is the process of changing the amplitude (Vc) of analog carrier signal in proportion to the amplitude (Vm) of the analog information signal.

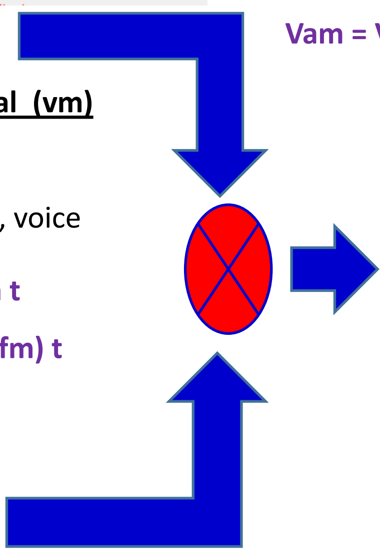
The amplitude (Vc) of the carrier signal is varied proportional to the amplitude (Vm) of information signal. The frequency (fc) and phase (ϴc) of carrier signal are remains unchanged.

When the amplitude (Vm) of information signals is increased, the amplitude (Vc) of carrier signal also increased and vice versa.



Vam = Vc sin ωc t + mVc/2 cos (ωc-ωm)t - mVc/2 cos (ωc+ωm)t

Vam = Vc (1 + m sin wm t) sin wc t

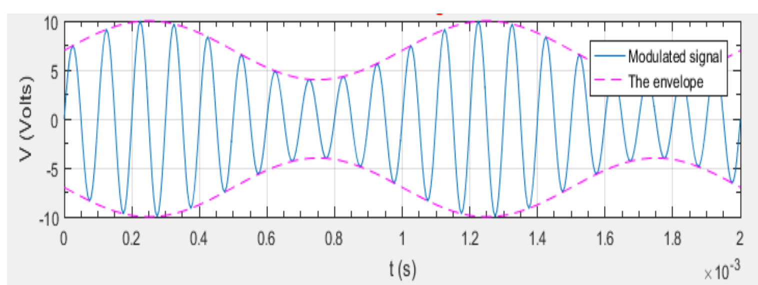


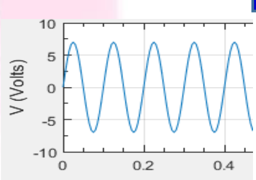
<u>Information Signal (vm)</u>

- Low frequency
- eg: audio signal, voice

Vm = Vm sin ωm t

Vm = Vm sin (2πfm) t





<u>Carrier Signal (vc)</u>

- High frequency.

Vc = Vc sin ωc t

Vc = Vc sin (2πfc) t

<u>Amplitude Modulated Signal (Vam)</u>

- High frequency
- The **amplitude (Vc)** of the carrier signal is varied proportional to the amplitude (Vm) of information signal
- **Frequency (fc) and phase (θc)** of carrier signal remain unchanged

AMPLITUDE MODULATION

AMPLITUDE MODULATION FREQUENCY SPECTRUM

AM has 2 sidebands

1) Lower Side Band (LSB)

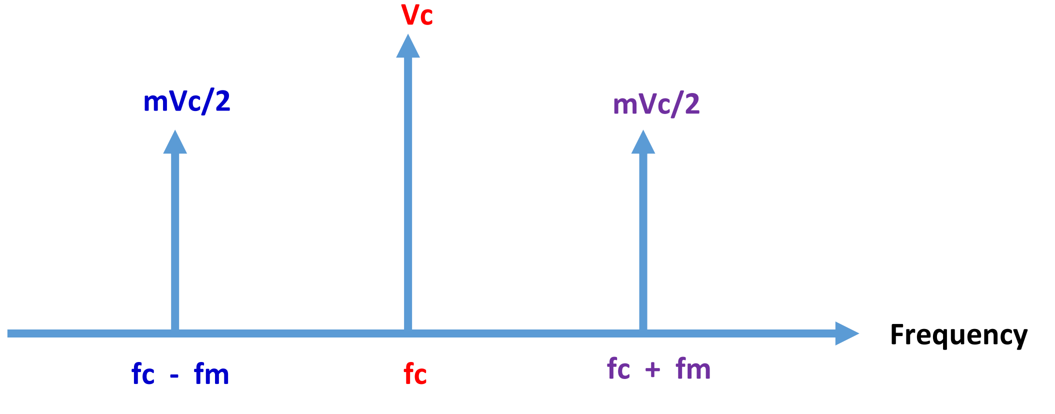
   fc – fm

2) Upper Side Band (USB)

   fc + fm

Vam = Vc sin ωc t + mVc/2 cos (ωc-ωm)t - mVc/2 cos (ωc+ωm)t

Carrier Frequency	LSB Lower Side Band Frequency	USB Upper Side Band Frequency
fc	fc - fm	fc + fm



AMPLITUDE MODULATION

AM BANDWIDTH, Bw

BANDWIDTH of Amplitude Modulation is different between the highest upper side frequency and the lower side frequency.

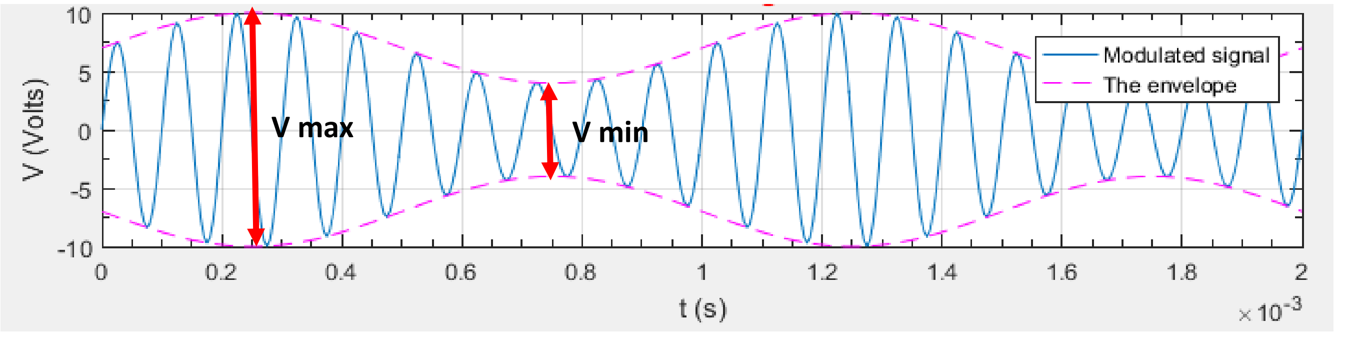
Bw = Upper Side Band – Lower Side Band

= (fc + fm) – (fc + fm)

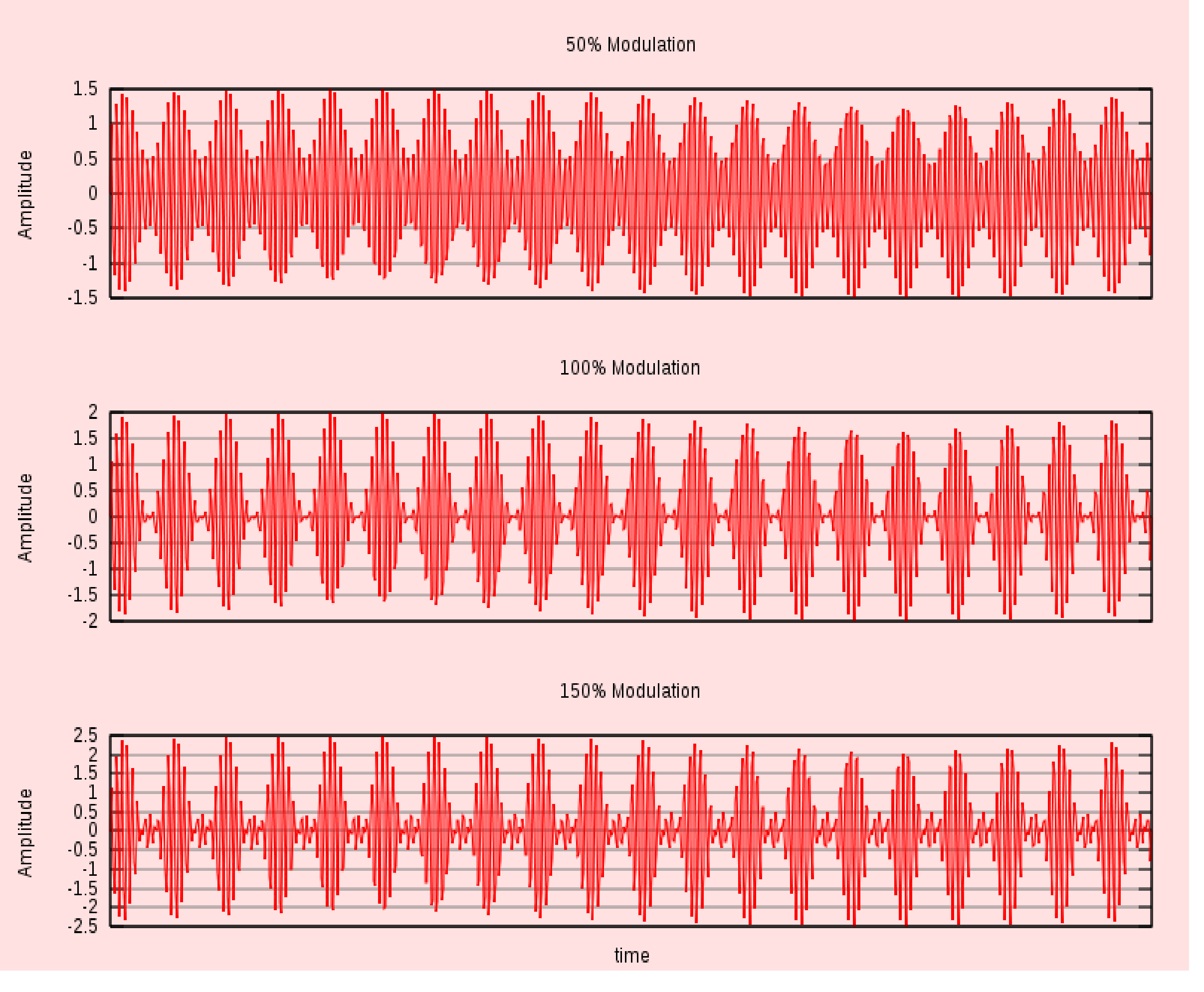
AM INDEX MODULATION, m

INDEX MODULATION is how much can the carrier amplitude change after being modulated by the message wave.

$$m = \frac{(V\ max) - (V\ min)}{(V\ max) + (V\ min)}$$



AMPLITUDE MODULATION



AMPLITUDE MODULATION

POWER IN AMPLITUDE MODULATION

Power Total, P_T

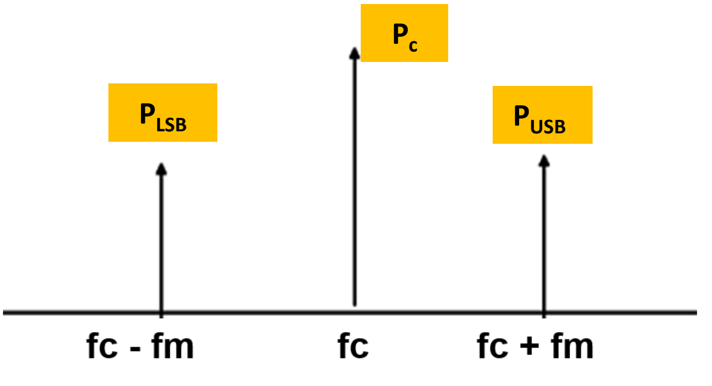
$$P_T = \frac{V_C^2}{2R} + \frac{m^2 V_C^2}{8R} + \frac{m^2 V_C^2}{8R}$$

$$P_T = P_c \left(1 + m^2 / 2\right)$$

Or

$P_T = P_c + P_{USB} + P_{LSB}$; where $[P_{USB} = P_{LSB}]$

$P_T = P_c + 2\,P_{USB}$



AMPLITUDE DEMODULATION

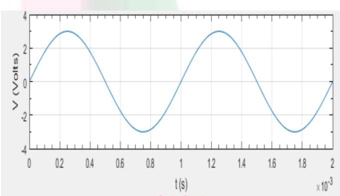
AMPLITUDE DEMODULATION is the process of **recovering the information wave** that has been transmitted. Most DSB system uses diode detector, transistor detector, oscillated radio frequency (TRF) and super heterodyne detector.

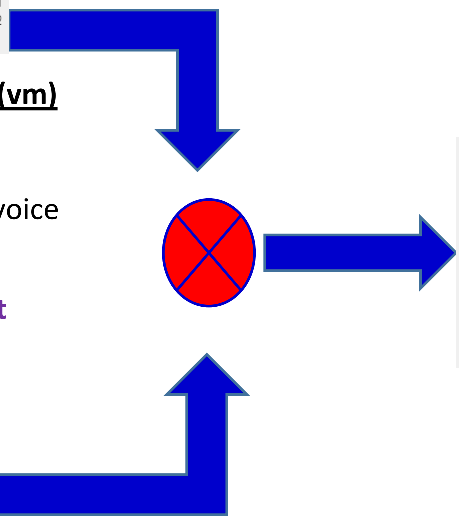
FREQUENCY MODULATION (FM)

FREQUENCY MODULATION (FM) is the process of changing the frequency (fc) of analog carrier signal in proportion to the amplitude (Vm) of the analog information signal.

The frequency (fc) of the carrier signal is varied proportional to the amplitude (Vm) of information signal. The Amplitude (Vc) and phase (Θc) of carrier signal remain unchanged.

When the amplitude (Vm) of information signals is increased, the frequency(fc) of carrier signal also change and vice versa.



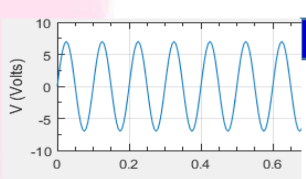


Information Signal (vm)

- Low frequency
- eg: audio signal, voice

vm = Vm sin ωmt

vm = Vm sin (2πfm)t



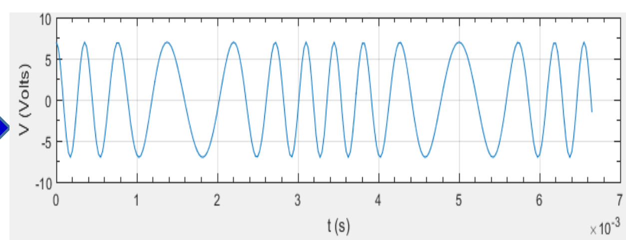
Carrier Signal (vc)

- High frequency

vc = Vc sin ωct

vc = Vc sin (2πfc)t

Vfm = Vc Cos (ωct + mf sin ωmt)



Frequency Modulated Signal (Vfm)

- The frequency (fc) of the carrier signal is varied proportional to the amplitude (Vm) of information signal
- Amplitude (Vc) and phase (θc) of carrier signal remain unchanged.

FREQUENCY MODULATION

FM MODULATION INDEX, mf

mf = fd/fm

mf = modulation index
fd = frequency deviation/different
fm = message signal frequency

ADVANTAGES OF FM TO AM

1) Less distortion.

Any undesirable amplitude in the power of the AM signal at the transmission can produce an amplitude change, distorting the radio sound.

2) Less interferences.

However, because the amplitude of an AM carrier wave contains information, factors like hum from adjacent equipment and sparks from passing cars can alter the amplitude and interfere with the wave.

3) Can carry more data .

Higher frequencies can carry more data than lower frequency waves.

DISADVANTAGES OF FM TO AM

1) FM suffers more attenuation than AM signal

Attenuation is directly proportional to frequency.

2) FM signals propagate as line-of-sight

If there is a large hill between transmitter and receiver, the receiver will not receive any signal AM carrier waves have much longer wavelengths than FM carrier waves. At low frequencies , AM signal can bend around obstacles like mountains and buildings better than FM waves and can travel greater distances.

3) FM is more complicated

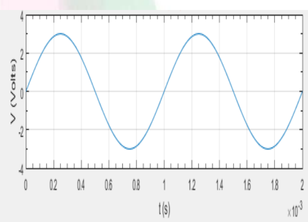
AM consists of a very simple detector circuit. This allows inexpensive production of medium wave broadcast receivers. AM circuit is cheaper and not complex compared to FM.

PHASE MODULATION (PM)

PHASE MODULATION (PM) is the process of changing the Phase (θc) of analog carrier signal in proportion to the amplitude (Vm) of the analog information signal.

The Phase (θc) of the carrier signal is varied proportional to the amplitude (Vm) of information signal. The Amplitude (Vc) and frequency (fc) of carrier signal are remains unchanged.

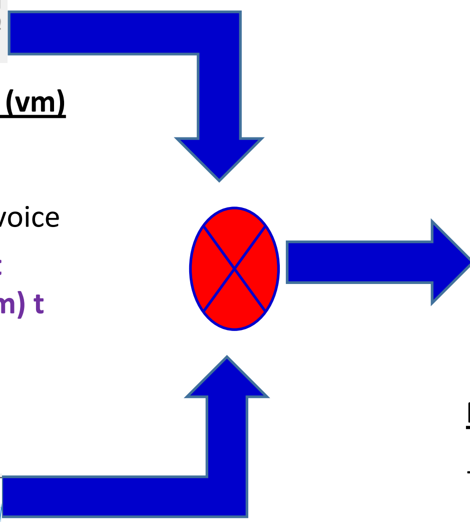
When the amplitude (Vm) of information signals is increased, the Phase (θc) of carrier signal also change and vice versa.

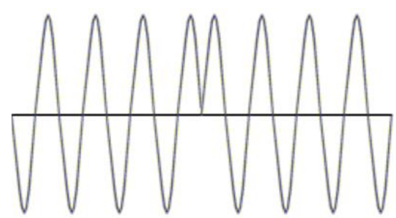


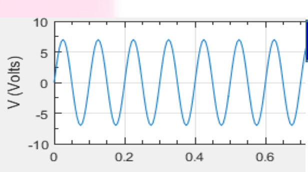
<u>Information Signal (vm)</u>

- Low frequency
- eg: audio signal, voice

$vm = Vm \sin \omega m\ t$
$vm = Vm \sin (2\pi fm)\ t$







<u>Carrier Signal (vc)</u>

- High frequency

$vc = Vc \sin \omega c\ t$
$vc = Vc \sin (2\pi fc)\ t$

<u>Phase Modulated Signal</u>

- The Phase (θc) of the carrier signal is varied proportional to the amplitude (Vm) of information signal
- Amplitude (Vc) and frequency (fc) of carrier signal remain unchanged.

COMPARISON BETWEEN AM, FM & PM

AM	FM	PM
The **carrier's amplitude changes** in accordance to the message signal's amplitude	The **carrier's frequency changes** in accordance to the message signal's amplitude	The **carrier's phase changes** in accordance to the message signal's amplitude
Frequency (fc) and Phase (θc) of carrier signal remain unchanged	**Amplitude (Vc) and Phase (θc)** of carrier signal remain unchanged	**Amplitude (Vc) and Frequency (fc)** of carrier signal remain unchanged
	The receivers on FM radios are less complicated than PM	The receivers on PM are more complicated than FM because it need proper synchronization process

DIGITAL COMMUNICATION SYSTEM

DIGITAL COMMUNICATION SYSTEM is the method of transferring information from one location to another using digital signals. There are **2 techniques** of digital communication , namely **digital transmission** and **digital radio.**

DIGITAL TRANSMISSION

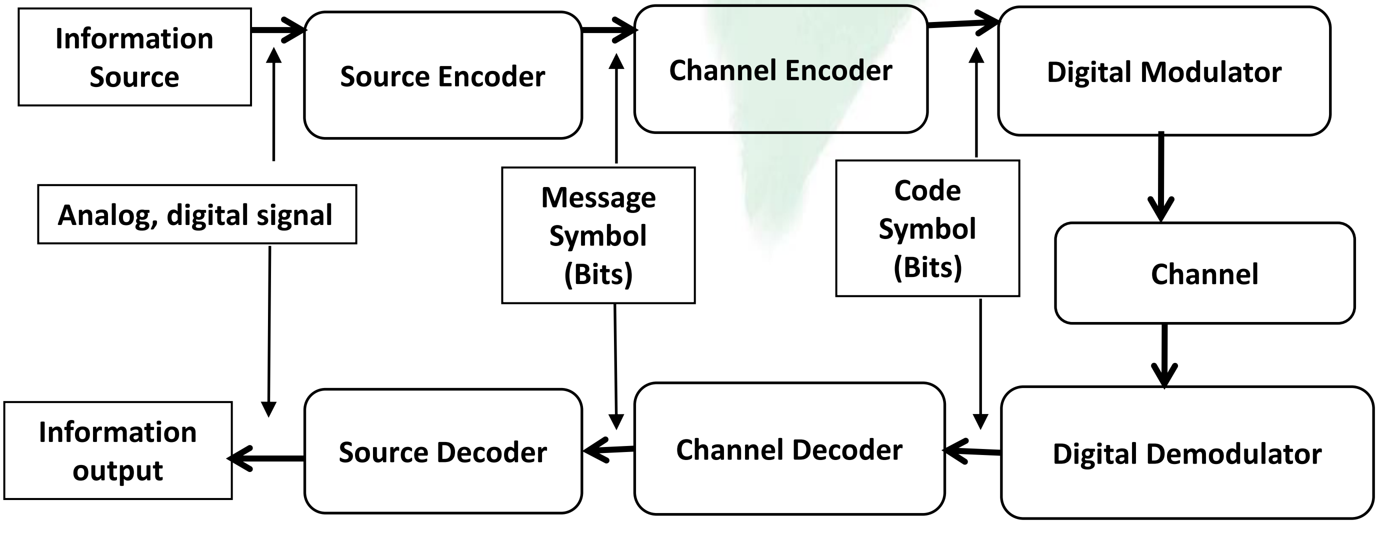
DIGITAL TRANSMISSION is a digital signal which will be transmitted between two or more points in the communication system . The original source information in digital transmission may be digital or analog signal but the carrier signal must be in digital signal. If the information signal is in analog form, it must be converted to digital pulses using Pulse Code Modulation (PCM) technique and converted back to analog form at the receiver. Since digital pulses cannot be propagated over a wireless communication medium, a transmission medium such as a metallic cable or a fibre optic cable was needed for digital transmission.

DIGITAL RADIO

The transmission of analogue signals between two or more sites in a communication system is known as **DIGITAL RADIO**. The original source information must be in digital form where the carrier signal and modulated signal are in analog form. Since the modulated signal is in analog form, the transmission medium could be either wireless or metallic cable or fibre optic cable.

DIGITAL COMMUNICATION SYSTEM

DIGITAL COMMUNICATION SYSTEM BLOCK DIAGRAM



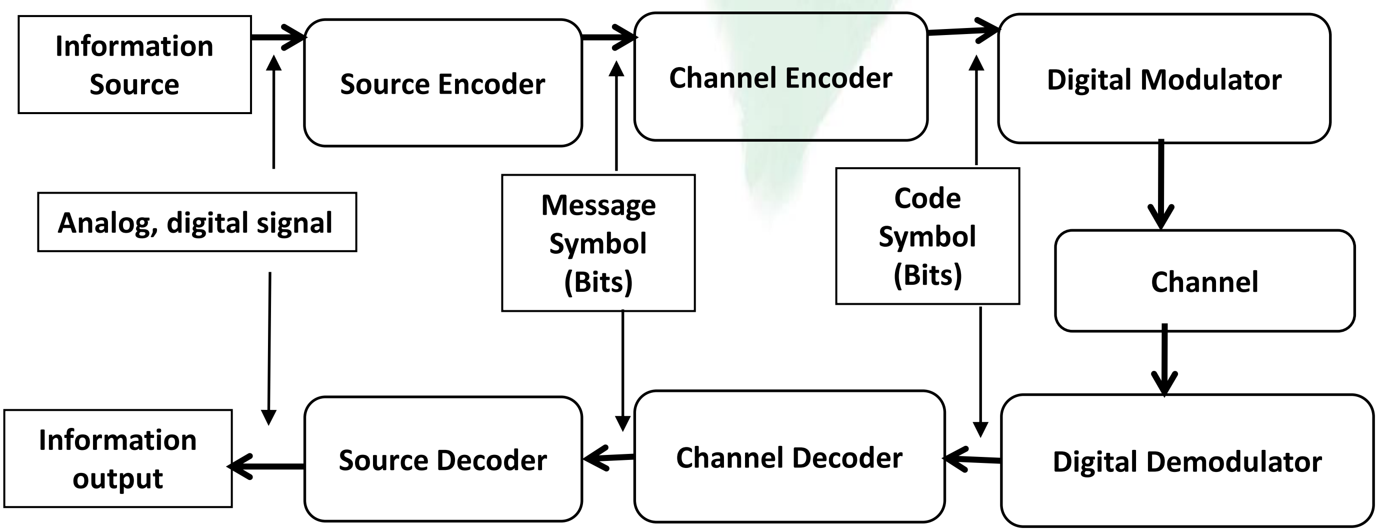
Block Diagram of Digital Communication System

TRANSMITTER

1. **Information Source** – The information source can be analog or digital.
2. **Source Encoder** – Convert the information signal from into digital signals by formatting and compressing the signals. The bits are then grouped together to form message symbols. Consider the PCM and Character Encoding (ASCII code) processes.
3. **Channel Encoder** – Error correction coding. It may reduce the probability of error by adding redundancy to the message symbols and converting them to code symbols/words.
4. **Digital Modulator** - Convert the serial bits (digital waveform) into electric signals (analog waveform).

DIGITAL COMMUNICATION SYSTEM

DIGITAL COMMUNICATION SYSTEM BLOCK DIAGRAM



Block Diagram of Digital Communication System

RECEIVER

5. **Channel** - Transmitting signals from transmitter to receiver .
6. **Digital Demodulator** – Demodulation process and converts the electric signals back to the code symbols.
7. **Channel Decoder** - Reassemble the original message symbols using the channel encoder's code symbols and the redundancy in the received data.
8. **Signal Decoder** - Convert the message symbols back to the original source information signal.

DIGITAL COMMUNICATION SYSTEM

ADVANTAGES OF DIGITAL COMMUNICATION

The advantages of digital transmission compared to analog transmission are :

1. **Noise Immunity**

   Digital signals are less affected by distortion, noise, and interference.

2. **Reduction of errors**

   By using error detecting and correcting codes, the probability of an error occurring can be minimised.

3. **Ease of processing and multiplexing**

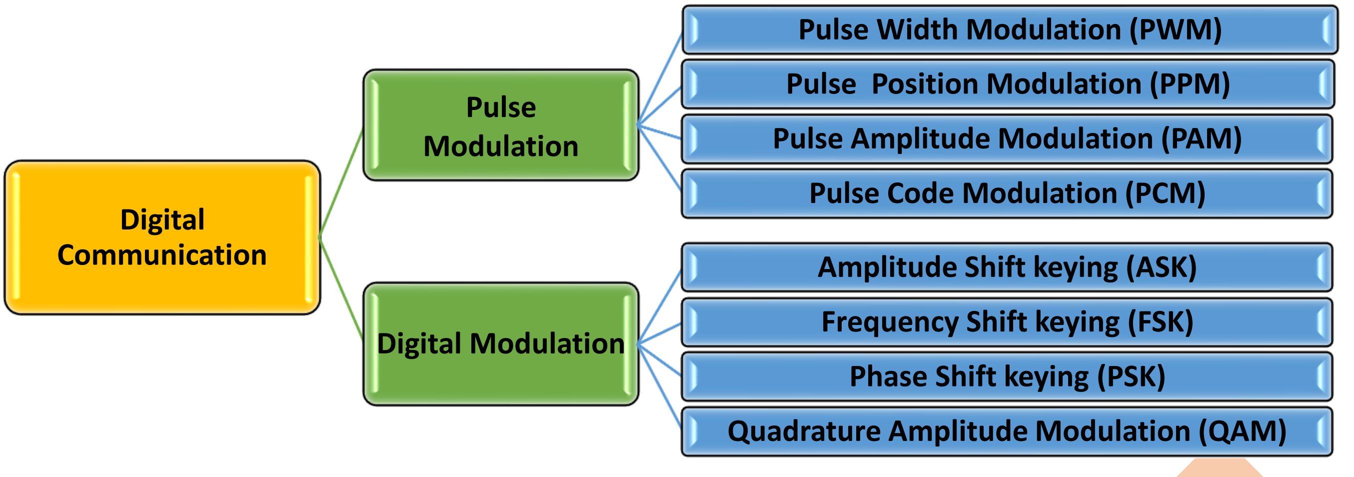
   Combining digital signals using Time Division Multiplexing TDM is more easier.

4. **Inexpensive digital circuitry**

   Digital circuits are more reliable, easy to design and cheaper than analog circuits.

5. The capacity of the channel is **effectively utilized** by digital signals.

TYPES OF DIGITAL COMMUNICATION



M-ARY ENCODING

An M-ary transmission is a type of digital modulation in which two or more bits are sent at the same time instead of one at a time and it can reduced channel bandwidth.

The mathematical expression for number of condition or level or combinations is:-

$M = 2^n$, where, M = number of conditions or levels or combinations or stage.

n = number of bits.

The mathematical expression for the number of bits is:-

$2^n = M$ where $n = \log_2 M$.

Example 1

Calculate the number of levels if the number of bit is 3.

Solution

n = 3 bit , $M = 2^3 = 8$ levels

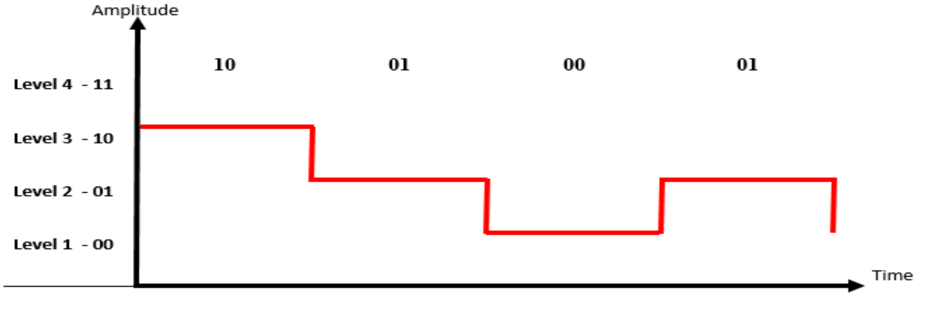
Example 2

There are 4 levels to a digital signal. If the data is 10010001, calculate how many bits are required per level and draw the digital signals.

Solution

$$n = \log_2(4)$$

$$n = \frac{\log 4}{\log 2} = \underline{\underline{2\text{ bits}}}$$



PULSE MODULATION TECHNIQUE

PULSE MODULATION TECHNIQUE

PULSE MODULATION TECHNIQUE is the process of changing the characteristics of the sampling pulse whether width, position and amplitude according to the change of the information signal to produce sampled signal. The characteristics of sampling pulses signal will be varied in proportion with amplitude of information signal. Pulse Modulation cannot use free space as transmission line.

In Pulse Modulation the waveform of the Information Signal is analog and the Sampling Signal is digital pulse . The process of Pulse modulation will convert an analog signal to a digital signal (ADC) and vice versa for digital transmission.

PULSE MODULATION

- **Pulse Width/Duration Modulation (PWM @ PDM)**
- **Pulse Position Modulation (PPM)**
- **Pulse Amplitude Modulation (PAM)**
- **Pulse Code Modulation (PCM)**

PULSE MODULATION TECHNIQUE

4 TECHNIQUES FOR PULSE MODULATION

1) **Pulse Width/Duration Modulation (PWM @ PDM)**

   PWM/PDM is the process of changing the width of the sampling pulses in proportion to the amplitude of the analog information signal. The amplitude and position of the sampling pulses remains unchanged. When the amplitude of information signals is increased, the width of sampling pulse become wider and vice versa.

2) **Pulse Position Modulation (PPM)**

   PPM is the process of changing the position of the sampling pulses in proportion to the amplitude of the analog information signal. The amplitude and width of the sampling pulses remains unchanged. When the amplitude of information signals is increased, the position of sampling pulse the farther to the right and vice versa.

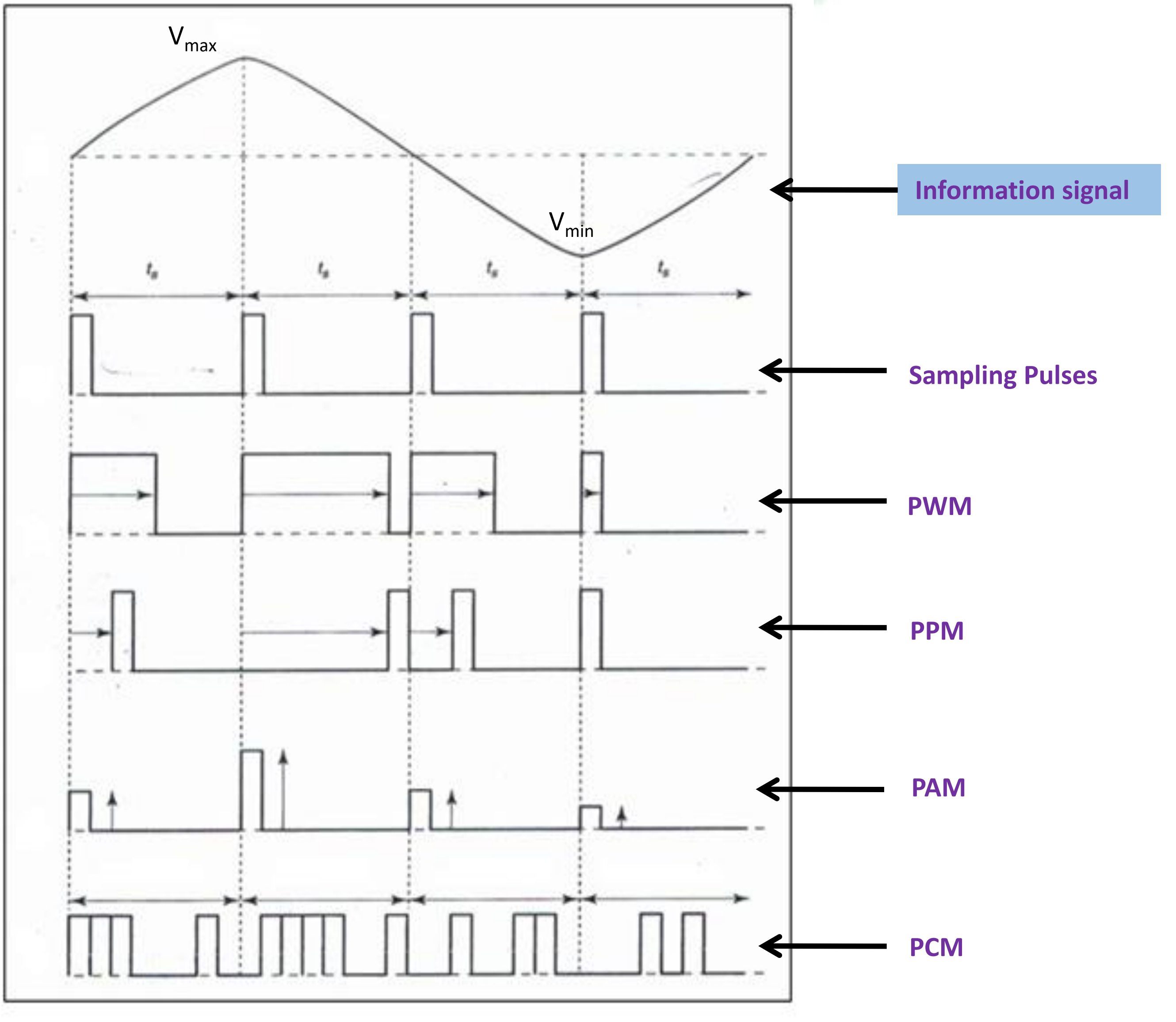
3) **Pulse Amplitude Modulation (PAM)**

   PAM is the process of changing the amplitude of the sampling pulses in proportion to the amplitude of the analog information signal. The width and position of the sampling pulses remains unchanged. When the amplitude of information signals is increased, the amplitude of sampling pulse also increase and vice versa.

4) **Pulse Code Modulation (PCM)**

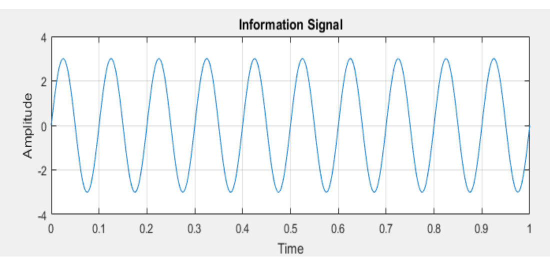
   PCM is a technique for converting an analogue signal into a digital signal that can then be transmitted via a digital communication network as a modified analogue signal. Since PCM is binary, there are only two possible states: high and low (0 and 1).

PULSE MODULATION TECHNIQUE

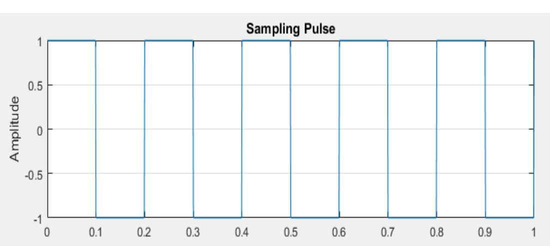


PULSE WIDTH/DURATION MODULATION (PWM/PDM)

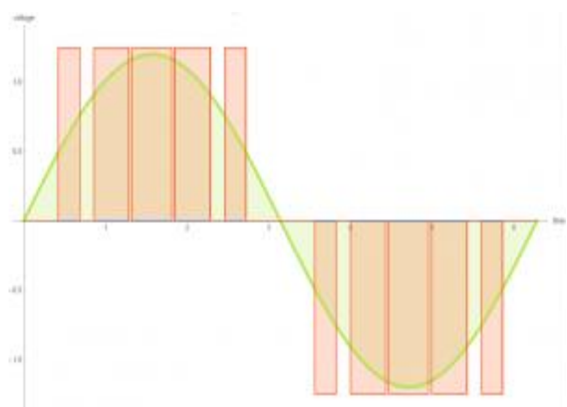
PWM / PDM is the process of changing the width of the sampling pulses in proportion to the amplitude of the analog information signal. The amplitude and position of the sampling pulses remains unchanged. When the amplitude of information signals is increased, the width of sampling pulse become wider and vice versa. PWM/PDM is a analog pulse modulation technique.



Analog Information Signal



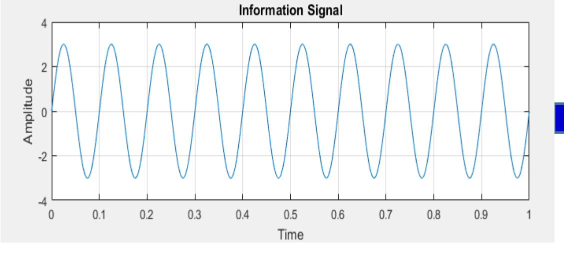
Sampling Pulse



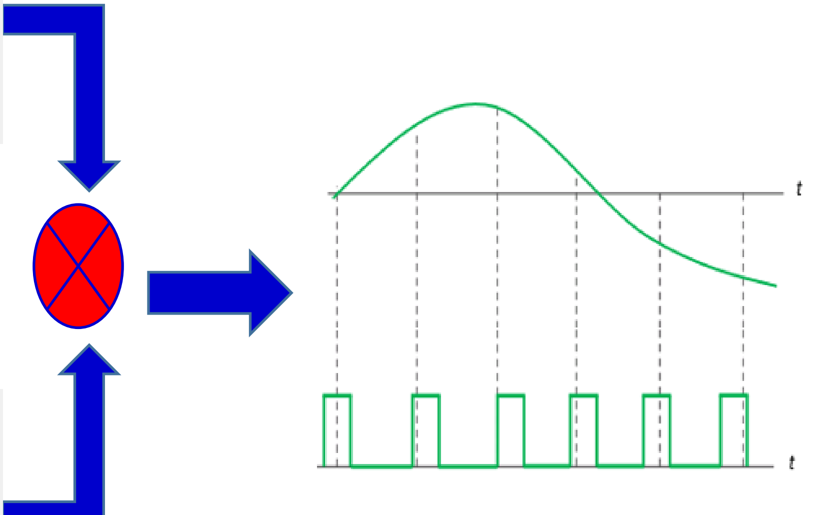
PWM / PDM Signal

PULSE POSITION MODULATION (PPM)

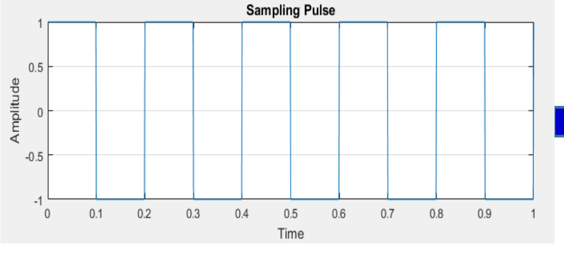
PPM is the process of changing the position of the sampling pulses in proportion to the amplitude of the analog information signal. The amplitude and width of the sampling pulses remains unchanged. When the amplitude of information signals is increased, the position of sampling pulse the farther to the right and vice versa. PPM is a analog pulse modulation technique.



Analog Information Signal



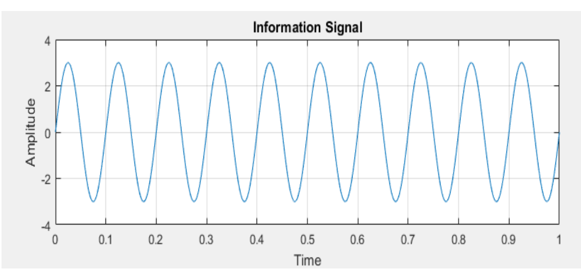
PPM Signal



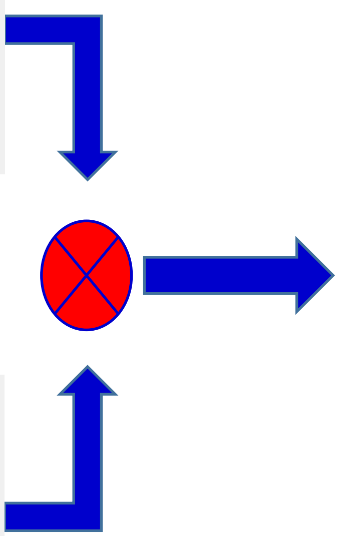
Sampling Pulse

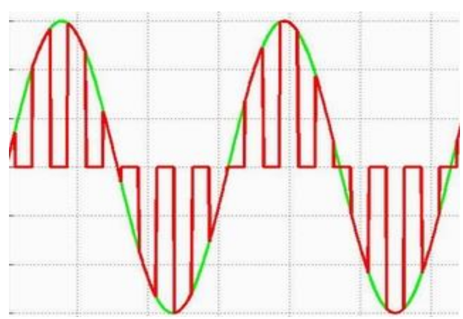
PULSE AMPLITUDE MODULATION (PAM)

PAM is the process of changing the amplitude of the sampling pulses in proportion to the amplitude of the analog information signal. The width and position of the sampling pulses remains unchanged. When the amplitude of information signals is increased, the amplitude of sampling pulse also increase and vice versa. PAM is a analog pulse modulation technique.

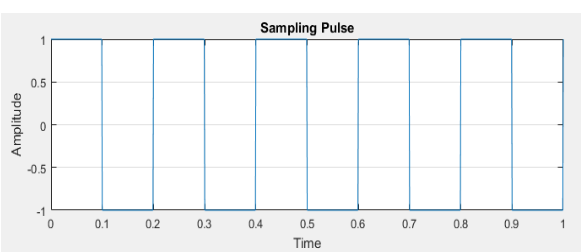


Analog Information Signal





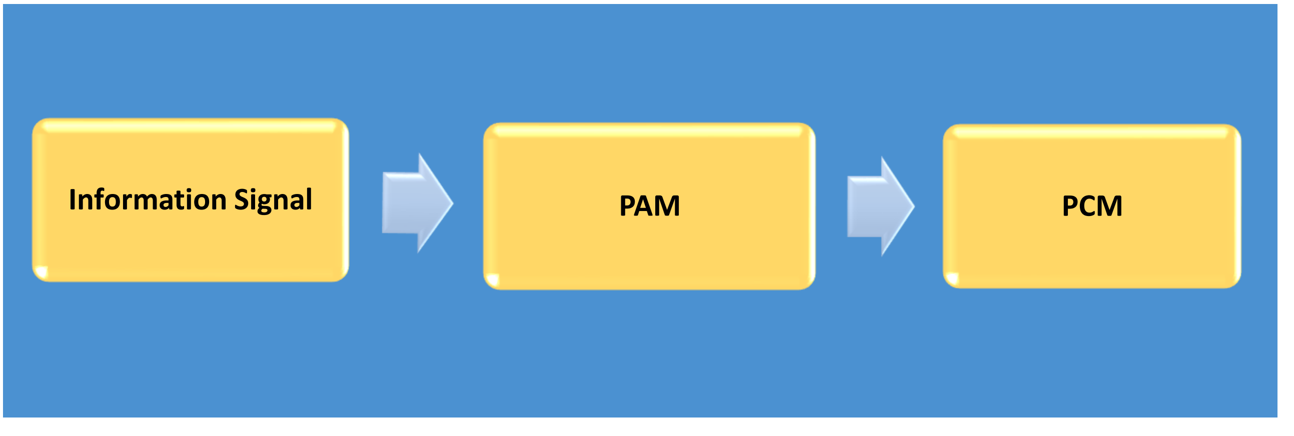
PAM Signal

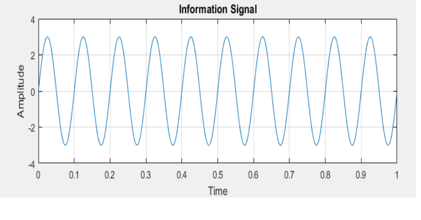


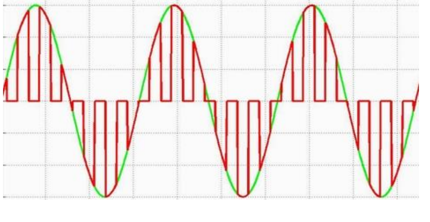
Sampling Pulse

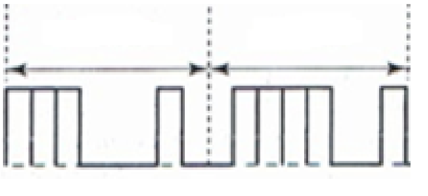
PULSE CODE MODULATION (PCM)

PCM is a technique for converting an analogue signal into a digital signal that can later then be transmitted via a digital communication network as a modified analogue signal. Since PCM is binary, there are only two possible states which is high (1) and low (0). PCM is a digital pulse modulation technique to convert analog signal to digital signal. The only modulation technique for digital transmission that is widely used is digitally encoded modulation. Any analog information data should be converted into a digital signal for digital transmission in digital transmission.









PULSE CODE MODULATION (PCM)

PCM TECHNIQUES

- Delta Modulation PCM
  - is an analog-to-digital and digital-to-analog signal conversion technique
  - used for voice transmission where accuracy is not a top priority
- Delta Sigma Modulation PCM
  - is a method for converting analog signal into digital signal (ADC)
  - used to convert high bit count into lower bit count which is convert digital signals into analog signal (DAC)
  - is the most powerful mode of conversion

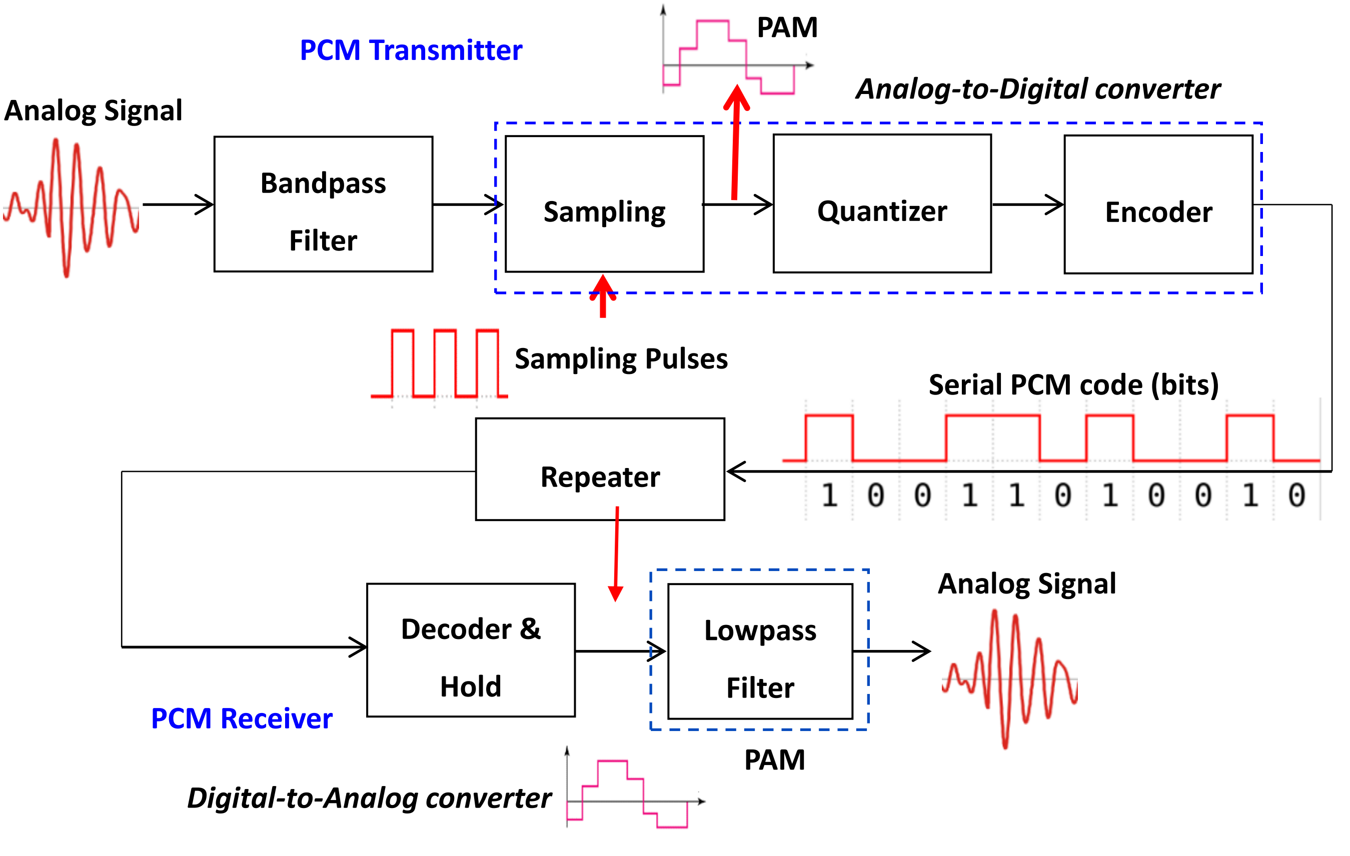
PULSE CODE MODULATION (PCM)

APPLICATION OF PULSE CODE MODULATION TECHNIQUE

- Analog to Digital Converter (ADC)
- Digital to Analog Converter (DAC)
- Multiplexing in Digital Telephony
- Private Automatic Branch Exchange
- Digital Audio Recording

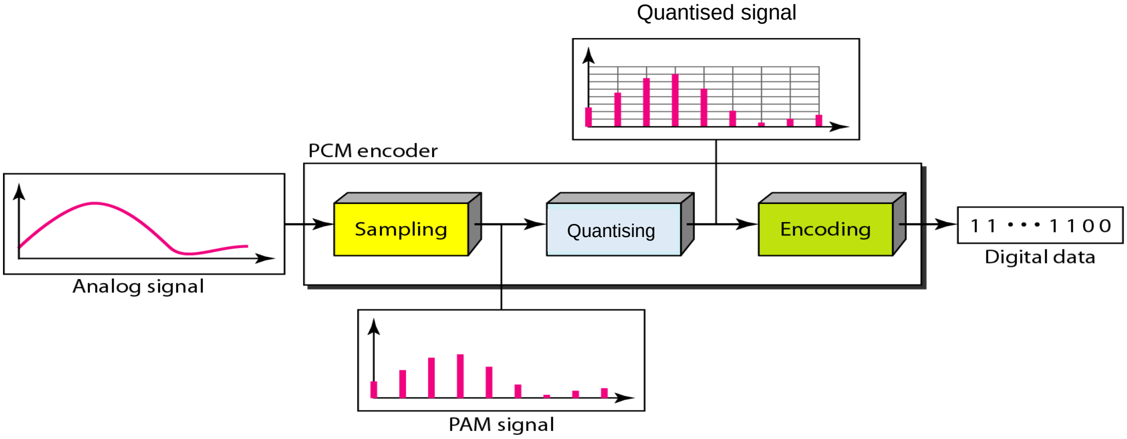
PULSE CODE MODULATION (PCM)

BLOCK DIAGRAM OF PCM



PULSE CODE MODULATION (PCM)

BLOCK DIAGRAM OF PCM (TRANSMITTER)

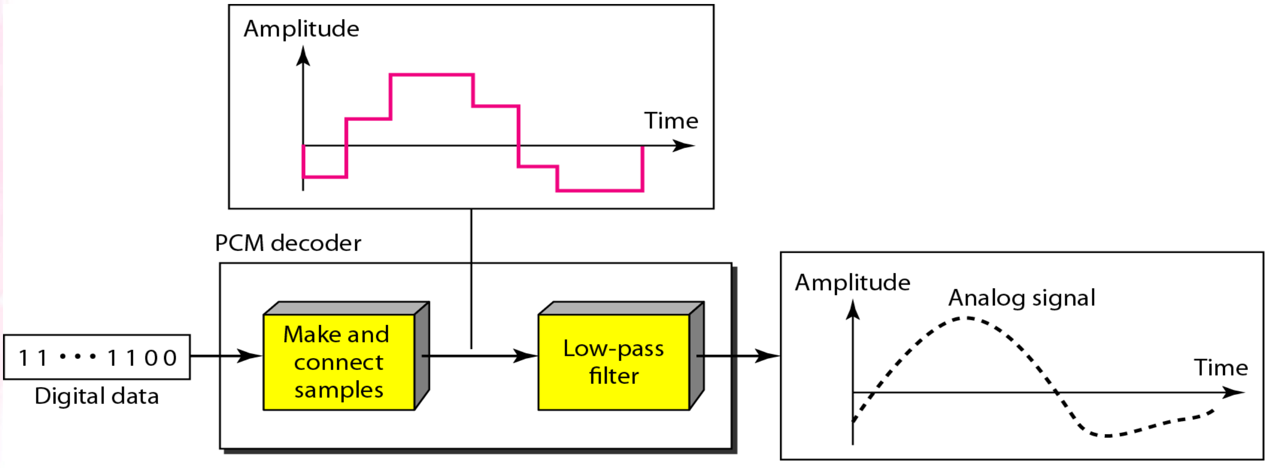


1) Sampling - The conversion of an analog signal into a digital pulse train(PAM signal). When the analog is sampled, a series of short pulses is produced, each representing the waveform's amplitude at the time of sampling.

2) Quantising - The process of changing the sampled PAM signal into quantised PAM signal by rounding off the amplitude of sampled signal to quantisation levels, L.

3) Encoding - Transform the quantized PAM signal into a series of binary pulses.

BLOCK DIAGRAM OF PCM (RECEIVER)



4) PCM decoder – Converts the binary pulses signal back to PAM signals and low pass filter used to recover the original signal.

PULSE CODE MODULATION (PCM)

3 STEP TO DIGITIZE AN ANALOG SIGNAL

Analog to Digital Converter (ADC)

1) Sampling

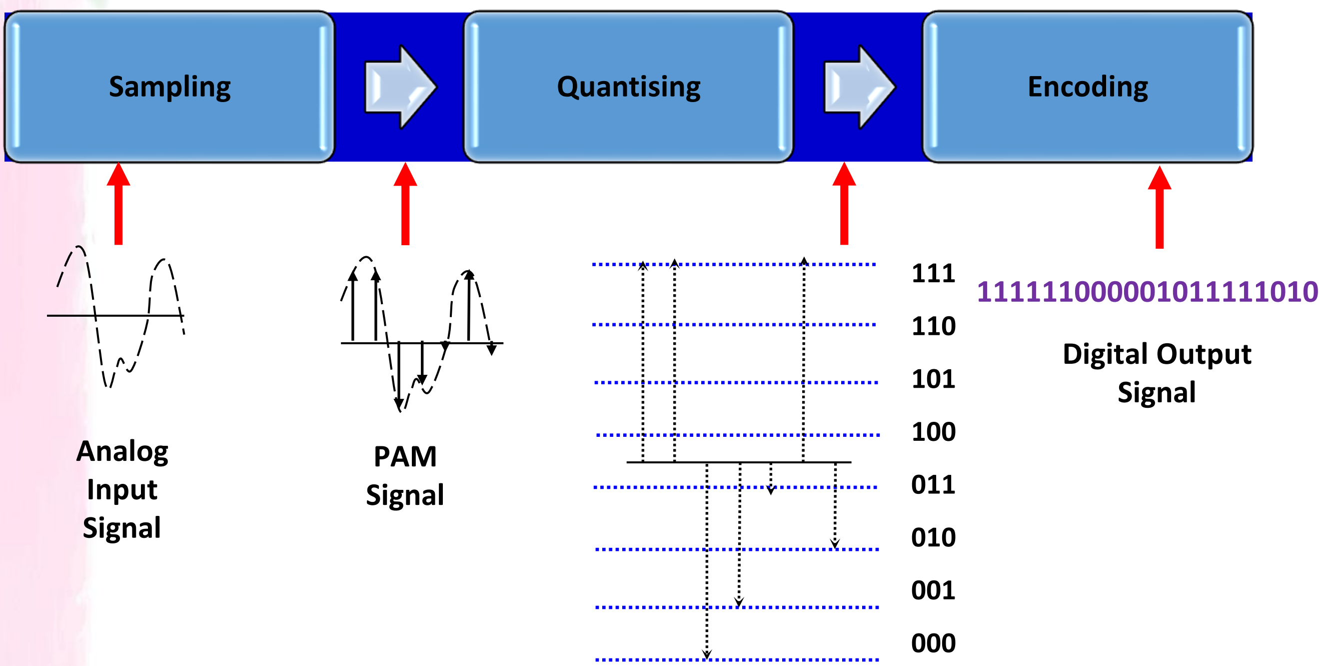
- The process of creating pulses with an amplitude equal to the amplitude of the analog signal.

2) Quantising

- The process of changing the sampled PAM signal into quantised PAM signal by rounding off the amplitude of sampled signal to quantisation levels, L.

3) Encoding

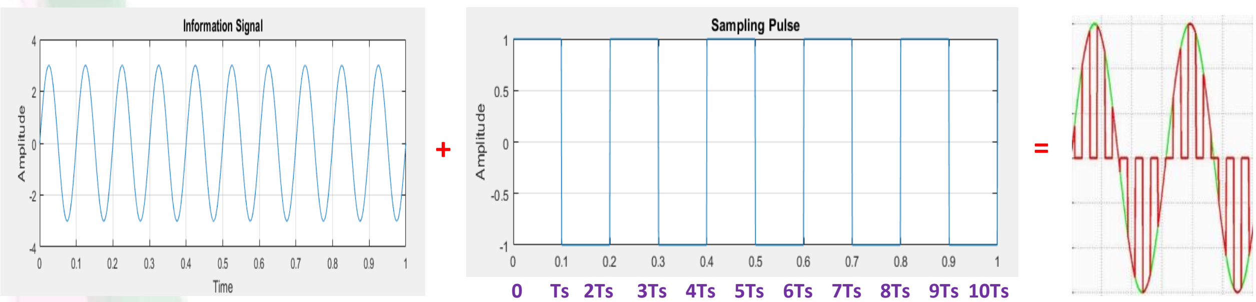
- Convert the quantized values to digital output signal.



PULSE CODE MODULATION (PCM)

SAMPLING

SAMPLING is the process of sampling an information signal by sampling a pulse signal . It transforms an analog signal to a discrete-time digital pulse signal. The signal that was sampled is a PAM signal. The higher sampling rate(Fs),the shorter sampling interval (Ts), the more the recovered signal.

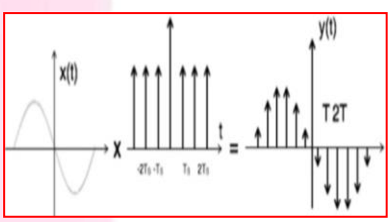


Information Signal **Sampling Pulse** **PAM Signal**

Ts = Sampling Interval or Sampling Period

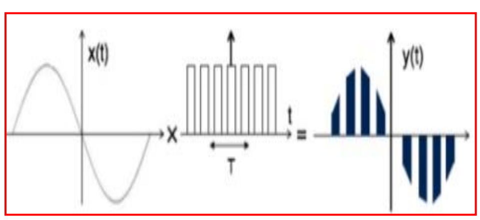
Fs = 1/Ts , Sampling Rate or Sampling Frequency

TYPES OF SAMPLING METHOD



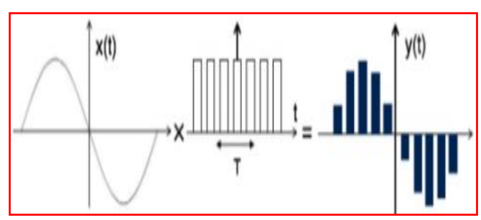
IDEAL SAMPLING

The amplitude of each pulse equals to modulating signal at the point of time.



NATURAL SAMPLING

The amplitude of each pulse is directly proportional to the modulating signal, and then follows the amplitude of the modulating signal.



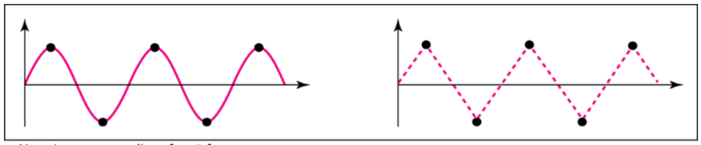
FLAT-TOP SAMPLING

The amplitude of each pulse is directly proportional to the instantaneous modulating signal at the time of pulse.

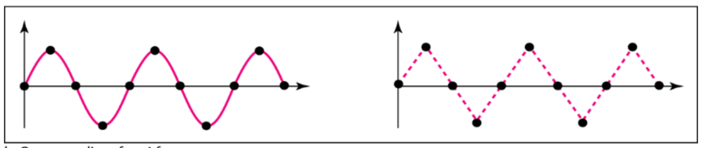
PULSE CODE MODULATION (PCM)

SAMPLING

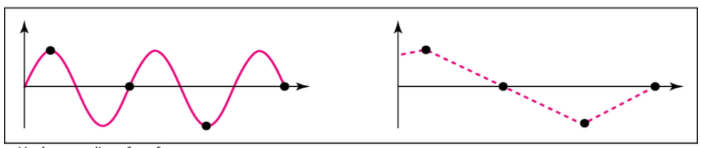
Sampled sine wave for different sampling



a. Nyquist rate sampling: $f_s = 2f$



b. Oversampling: $f_s = 4f$



c. Undersampling: $f_s = f$

SAMPLING THEOREM

According to **NYQUIST'S THEOREM**, a sampling frequency(Fs) must be sampled at least twice its highest frequency of information signal (fm).

$$F_s \geq 2f_{m(max)}$$

EXAMPLE :

The frequency spectrum of a signal has a bandwidth of 500 Hz with the maximum frequency of 600Hz. .Express the Sampling frequency and Sampling Period.

ANSWER :

$F_s = 2f_{m(max)} = 2\ (600\text{Hz}) =$ **<u>1200 Hz samples/s</u>**

Ts = 1/ fs = 1/1200Hz = **<u>0.83ms</u>**

PULSE CODE MODULATION (PCM)

QUANTISATION

QUANTISATION is the process of changing the sampled PAM signal into quantised PAM signal by rounding off the amplitude of sampled signal to quantisation levels, L. The number of quantisation stages, L, is calculated using the M-ary formula.

$\mathbf{L = 2^n}$ n = number of bits per level

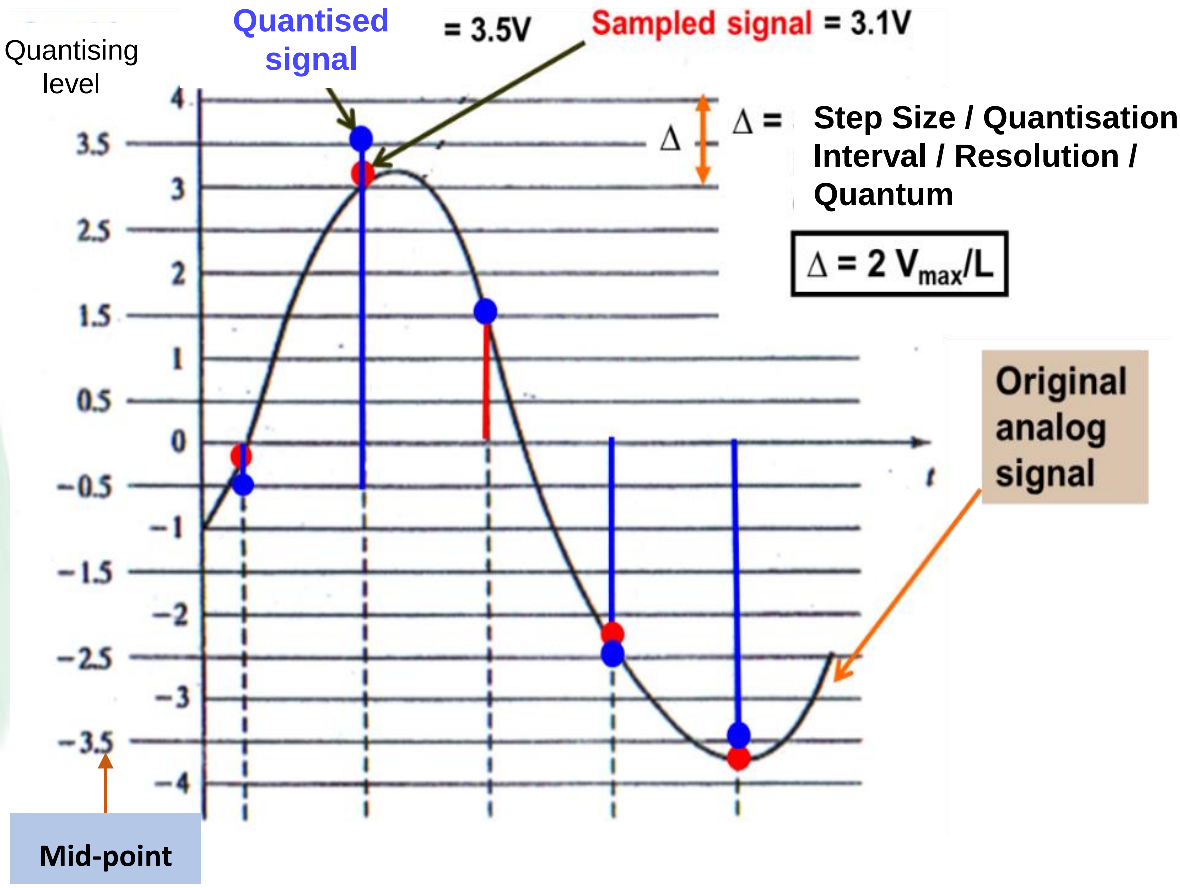
L = number of quantisation level

The more L levels used, the more accurately an analog signal can be identified during signal recovery at the receiver. This is because the more bits (n) and quantisation levels (L) used, the lower the quantisation error (Qe) and the more reliable the original signal recovery. The maximum length of a PCM code is 8 bits and 256 levels.

$\mathbf{L = 2^8}$ = 256 level

PULSE CODE MODULATION (PCM)

QUANTISATION



1) Amplitude of sampled PAM signal is between Vmax = 4V and Vmin = -4V.

2) 3 bit PCM code which is n=3 , the Quantisation Level is;

$L = 2^n$

$L = 2^3 = 8$ level

3) Step size, Δ = 2 Vmax /L

= 2(4)/8 = 1V

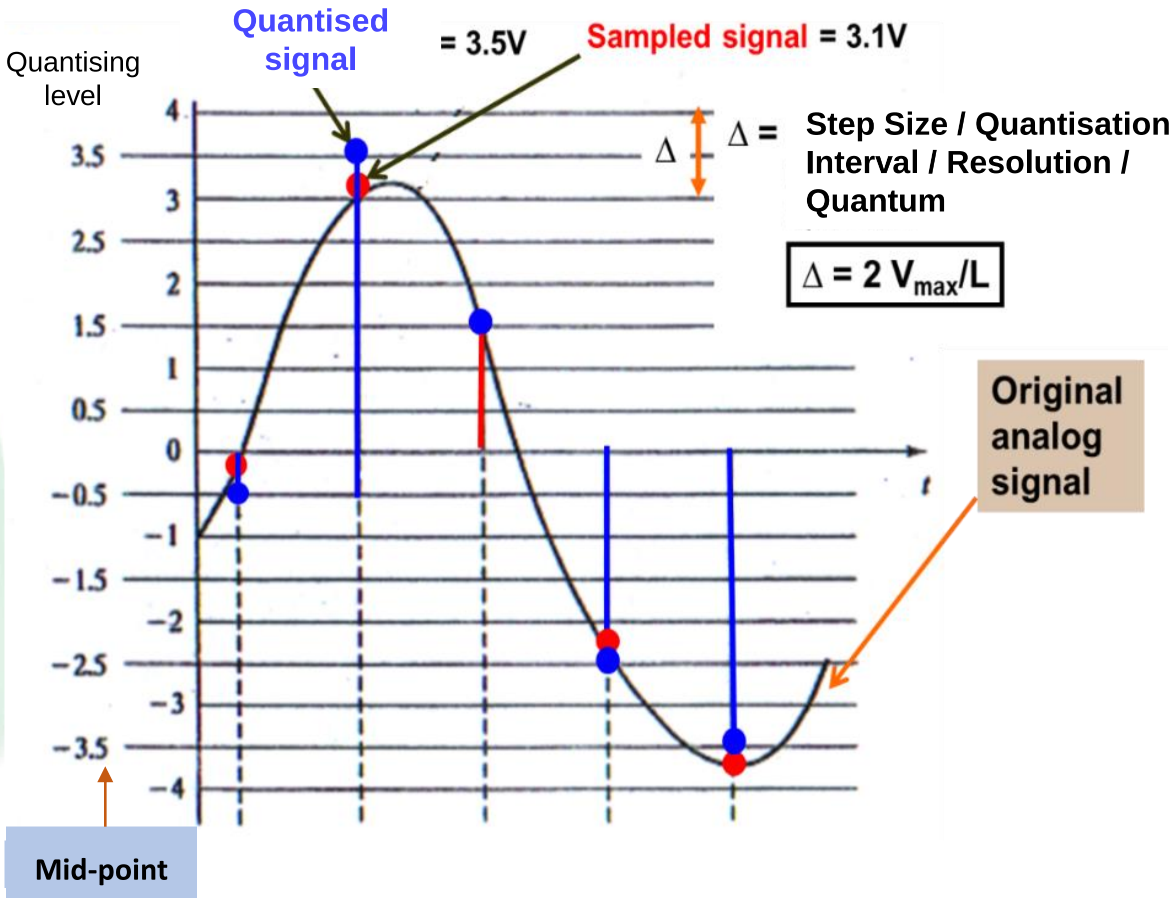
4) The 8 zones are: (-4V to -3V), (-3V to -2V), (-2V to -1V), (-1V to 0V), (0V to 1V), (1V to 2V), (2V to 3V) and (3V to 4V).

5) This midpoint is called quantisation level, L. The midpoints are: -3.5V, -2.5V, -1.5V, -0.5V, 0.5V, 1.5V, 2.5V, and 3.5V.

6) Each sampled signal's amplitude is quantised (rounding-off) to the midpoint (quantisation level). For example in the second sample has sampled amplitude value of 3.1V. After quantisation, the sampled amplitude value is quantised to 3.5V level.

PULSE CODE MODULATION (PCM)

QUANTISATION



1) There are **2 types of Quantising** method which are;

i. Uniform Quantisation – uniform step size, Δ

ii.Non-uniform Quantisation – non-uniform step size, Δ

2) The figure is Uniform quantisation where the step size is uniform for each zone.

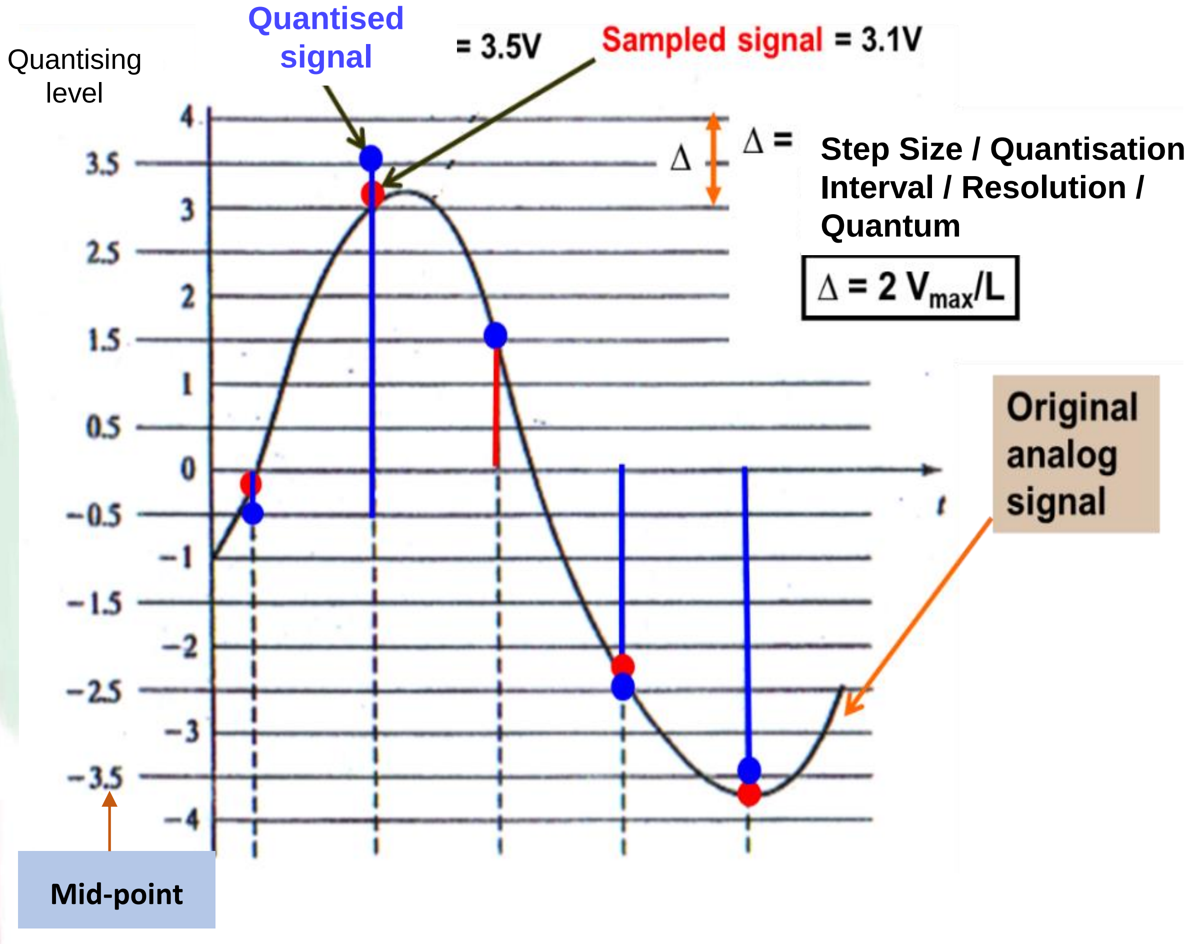
3) Uniform quantisation scheme is wasteful for speech signals because many of quantisation levels, L are rarely used.

4) Non-uniform quantising method is more suitable for speech signals because the step size could be adjusted depending on the amplitude of signal (smaller step size for lower amplitude and larger for higher amplitude).

5) Non-uniform quantisation is commonly used because the uniform quantisation is not efficient for a signal that has smaller amplitude.

PULSE CODE MODULATION (PCM)

QUANTISATION



1) Since quantisation is a process of approximation, thus error (Qe) is introduced during signal quantising.

2) Quantisation error (Qe) is the difference between sampled and quantised value.

3) Example, the sampled value is 3.1V, while the quantised value is 3.5V. So;

Q_e = quantised value – sampled value

= 3.5V – 3.1 = **0.4V**

4) Quantisation error (Q_e) is also called **Quantisation noise (Q_n)** where the **maximum** error is **Q_e** = ± Δ/2 for Uniform Quantisation.

PULSE CODE MODULATION (PCM)

SIGNAL TO QUANTISATION NOISE POWER RATIO (SQR)

The SQR (Signal-to-Quantisation Noise Power Ratio) in dB is defined as follows :

SQR(dB) = 6.02n + 1.76 dB where n = number of bits

Values of SQR depends on the number of bits, n. This is due to the fact that as the number of bits(n) increases, so does the number of quantisation levels(L), and the step size(Δ) decreases. As the step size(Δ) gets smaller, resulting in smaller errors and the more accurate the recovery signal.

Example :

What is the SQR in dB if the number of quantisation, L= 8 and L=16

$$n = \log_2 L = \log_2 8 = \frac{\log 8}{\log 2} = \underline{\underline{3bits}}$$

$$\text{SQR(dB)} = 6.02(3) + 1.76 = \underline{\underline{19.82\text{dB}}}$$

$$n = \log_2 L = \log_2 16 = \frac{\log 16}{\log 2} = \underline{\underline{4bits}}$$

$$\text{SQR(dB)} = 6.02(4) + 1.76 = \underline{\underline{25.84\text{dB}}}$$

PULSE CODE MODULATION (PCM)

BIT RATE

BIT RATE is defined as the number of bits transmitted in one second, given in bits per second (bps).

Bit Rate, R = F_S x n where , F_S = sampling rate

n = number of bits per sample

EXAMPLE :

Calculate the bit rate of human voice having 16 bits per sample.

SOLUTION :

The bandwidth of human frequency is 300 – 3400 Hz

Sampling Rate, $F_S = 2f m_{max}$ = 2 x 3.4kHz = **6800 samples/s**

Bit Rate, R = F_S x n = 6800 x 16 bits = **108.3k bps**

PULSE CODE MODULATION (PCM)

ENCODING

ENCODING is a process of converting the **quantised signal** into a **decimal code number**.

The decimal code number is then transformed to the binary sequence that corresponds to it.

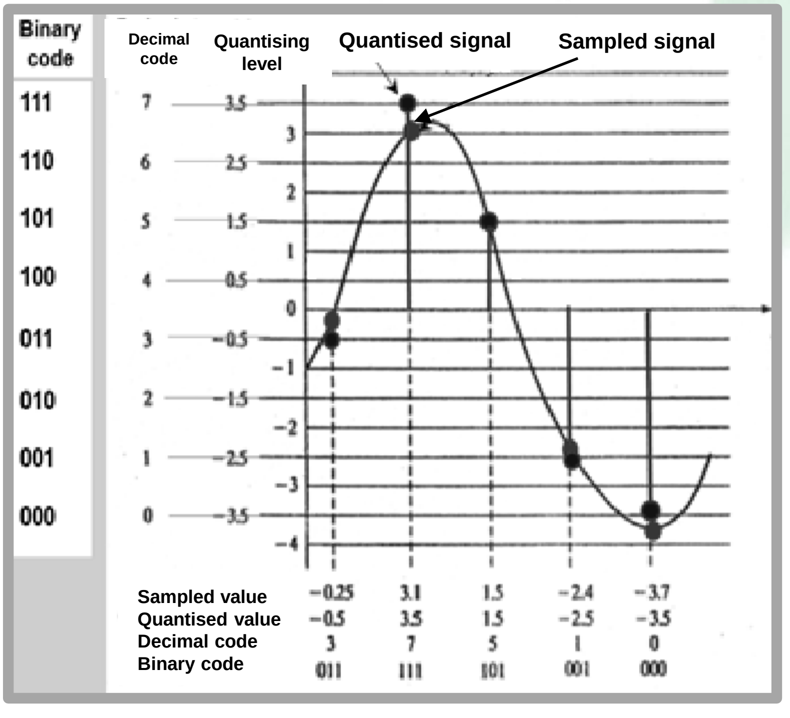
The number of bits(n) for each level of code number is determined by the number of quantisation levels(L) used to quantise the samples using the M-ary formula.

$$\boxed{n = \log_2 L}$$

The quantising and encoding operations are usually performed in the same circuit which is called analog-to-digital converter (ADC).

PULSE CODE MODULATION (PCM)

ENCODING



- Assign the **decimal** code number **0** to the level **-3.5V**, the code number 1 to level -2.5V, and so on until level +3.5V.
- Each **decimal code number (0 - 7)** has its own **3 bits binary code** representation, ranging from **000** for code number **0** to **111** for code number **7**.
- The binary sequences (digital signal) that produce from PCM are **011 111 101 001 000.**

The **Decimal Code Number** for each quantisation level is converted to its representative **binary sequence** using **Binary Code.**

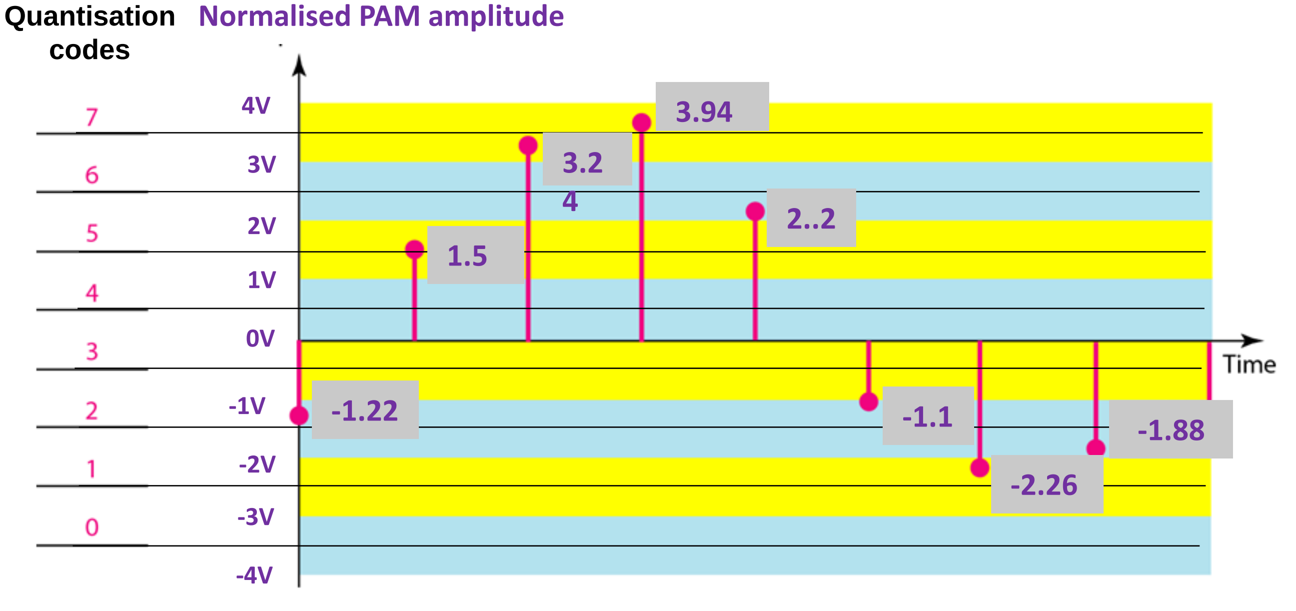
Decimal Code	Binary Code
7	111
6	110
5	101
4	100
3	011
2	010
1	001
0	000

PULSE CODE MODULATION (PCM)

ENCODING

EXAMPLE :

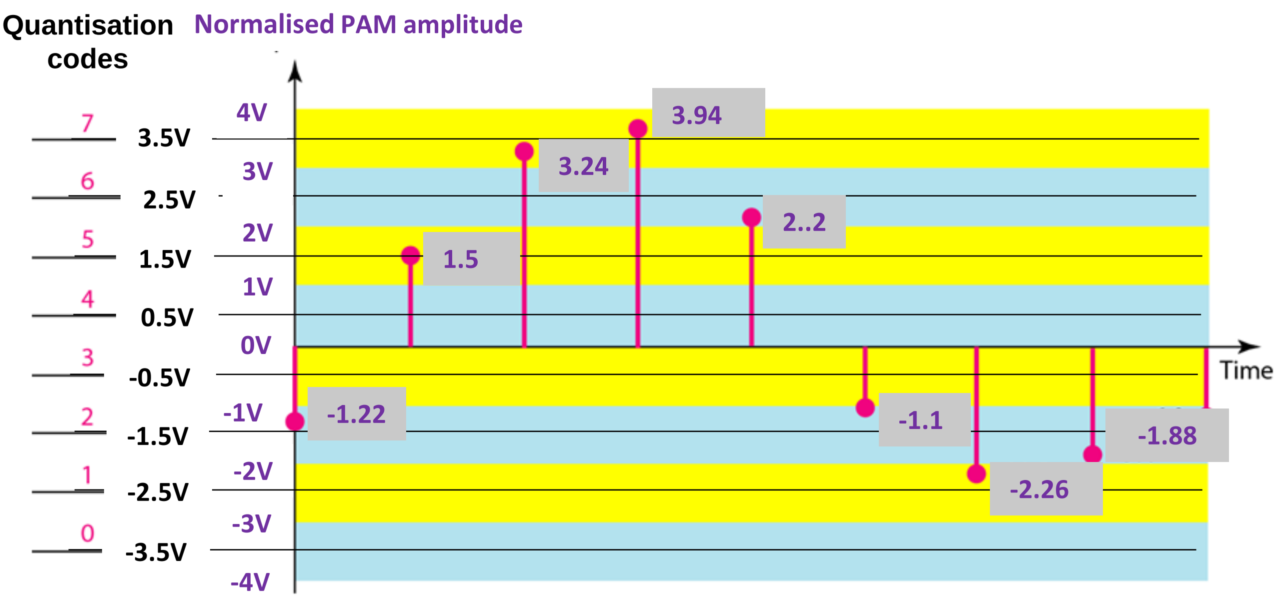
Encode the following quantised signal and find the quantisation error for each samples.



PULSE CODE MODULATION (PCM)

ENCODING

SOLUTION :

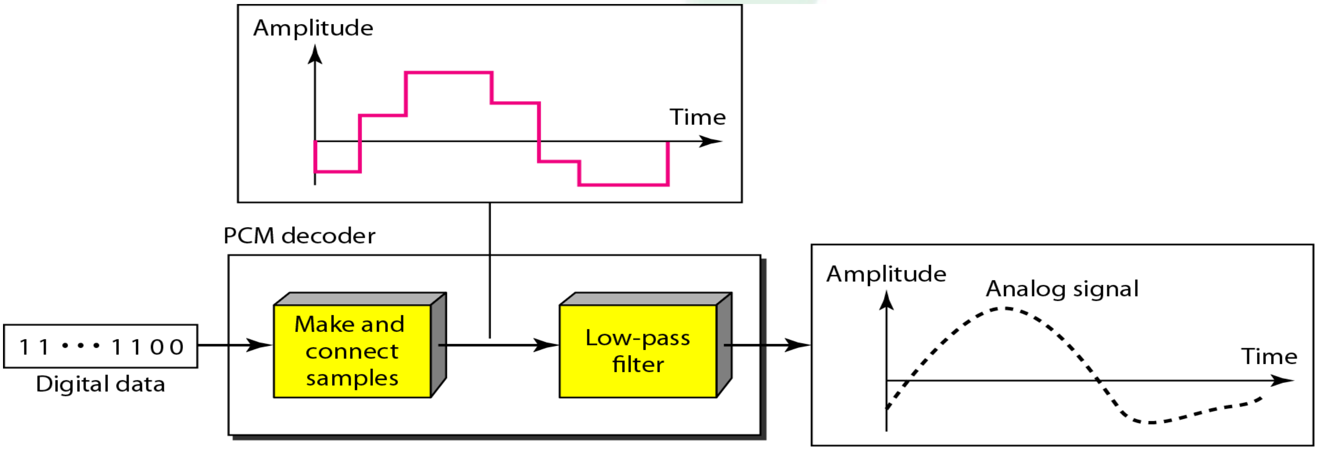


Normalised PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88
Normalised quantised values	1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50
Normalised error	-0.28	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38
Quantisation code	2	5	7	7	6	2	1	2
Encoded words	010	101	111	111	110	010	001	010

PULSE CODE MODULATION (PCM)

DECODING

BLOCK DIAGRAM OF PCM DECODING (RECEIVER)



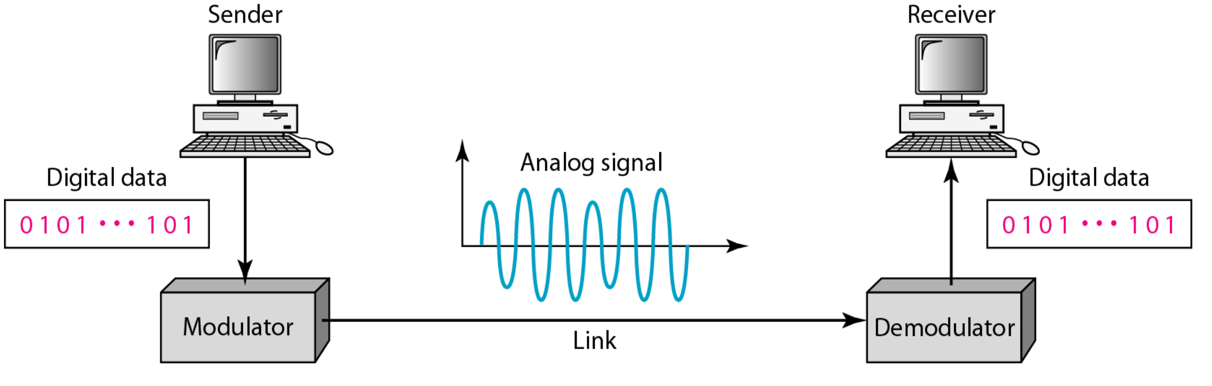
- To recover an analog signal from a digitised signal :

1. Use a **decoder and hold** circuit that holds the amplitude value of a pulse till the next pulse arrives. This will produce **a staircase PAM signal.**
2. This PAM signal will pass through a **low pass filter** which has the same cutoff frequency as the original information signal at sender. The filter will **smooth** the staircase amplitude of PAM signals into an analog signal.

- If the original info signal is **sampled at** (or **greater than**) Nyquist Sampling Rate AND if there are **enough Quantisation levels,** the original signal would be recovered back with less distortion.
- The higher the value of quantisation level L, the less distorted a signal is recovered.

DIGITAL MODULATION

DIGITAL MODULATION is the process of changing a characteristic of an analog carrier signal according to the change in amplitude of the digital information signal. In digital modulation techniques, the data input signal needs to be converted to an analog signal because the digital signal cannot be transmitted through a free space medium and at the receiver, the analog signal will be converted back to a digital signal form.



DIGITAL MODULATION

DIGITAL MODULATION

Amplitude Shift Keying (ASK)

Amplitude (Vc) of the analog carrier signal is varied proportional to the amplitude of digital information signal.

Frequency Shift Keying (FSK)

Frequency (fc) of the analog carrier signal is varied proportional to the amplitude of digital information signal.

Phase Shift Keying (PSK)

Phase (ϴc) of the analog carrier signal is varied proportional to the amplitude of digital information signal.

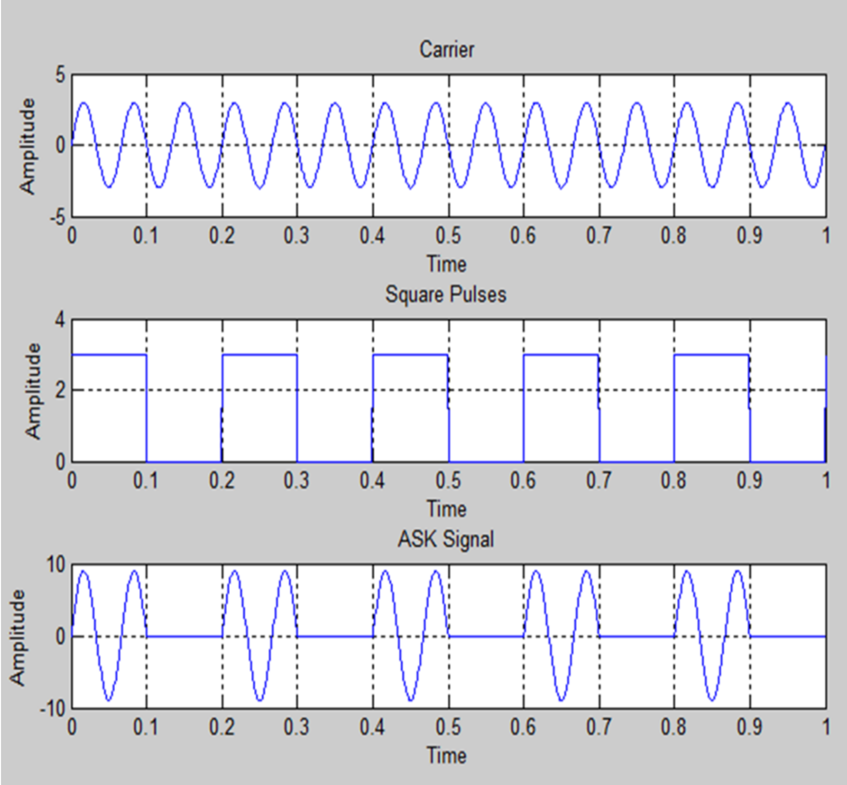
Quadrature Amplitude Modulation (QAM)

is a combination of Amplitude Shift Keying and Phase Shift Keying.

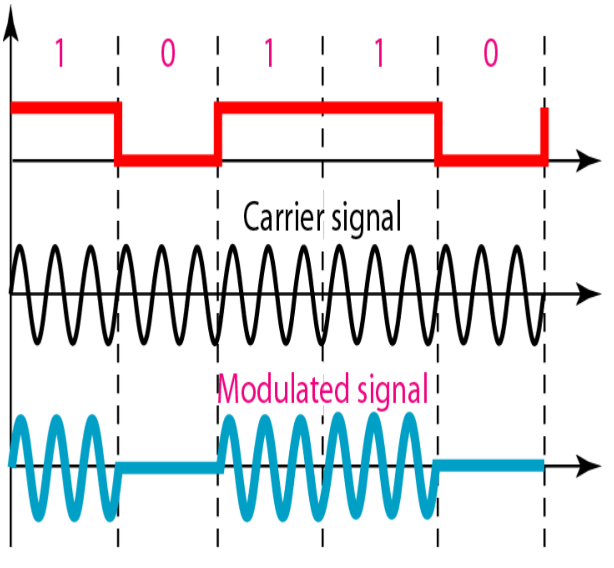
DIGITAL MODULATION

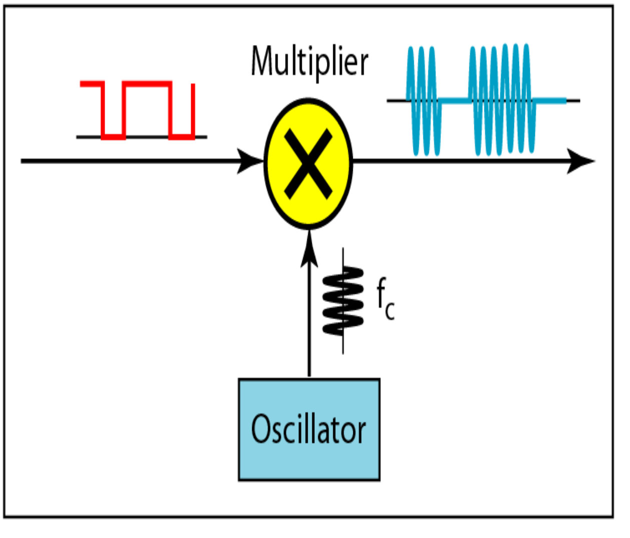
Amplitude Shift Keying (ASK)

- Amplitude (Vc) of the analog carrier signal is varied proportional to the amplitude of digital information signal.
- Changing the Carrier Signal's Amplitude.
- Frequency and phase carrier remain constant.
- Bit ‘1’, the output carrier signal has the constant amplitude.
- Bit ‘0’, the carrier signal has no amplitude.
- Advantage: Simple and require less bandwidth.
- Disadvantage: Easily affected by noise.



IMPLEMENTATION OF BINARY ASK

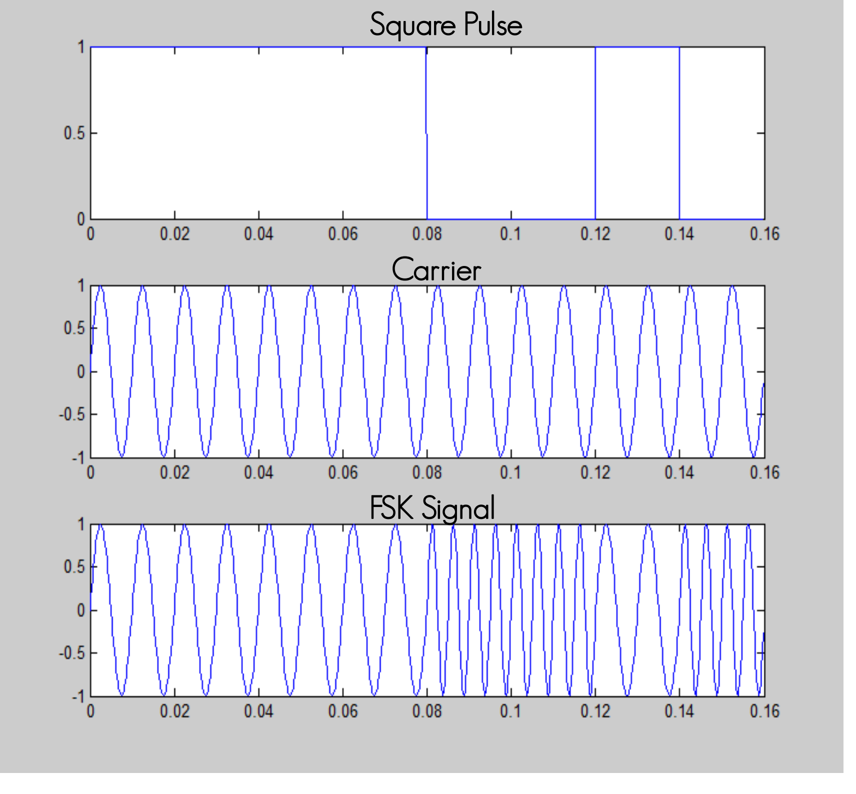




DIGITAL MODULATION

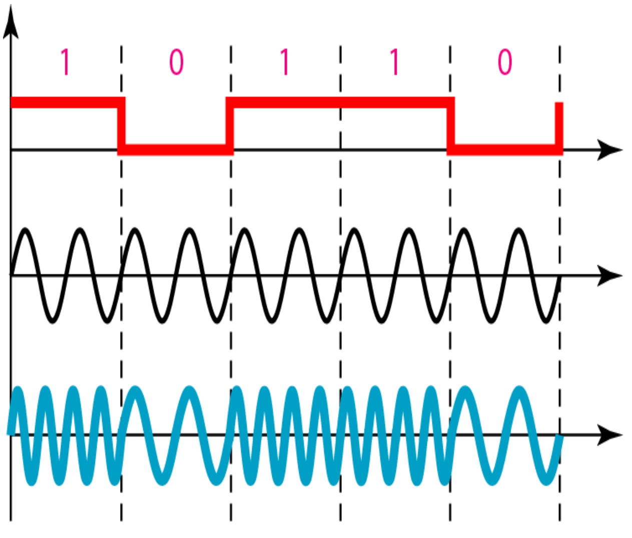
Frequency Shift Keying (FSK)

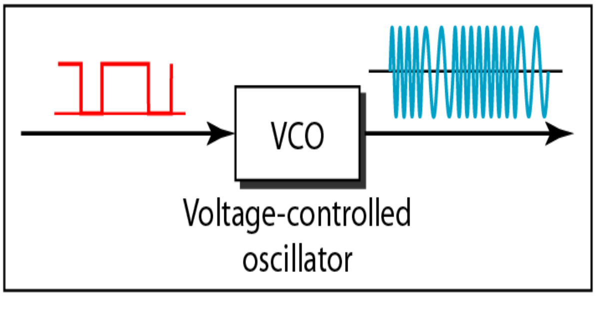
- Frequency (fc) of the analog carrier signal is varied proportional to the amplitude of digital information signal.
- Changing the Carrier Signal's Frequency.
- Amplitude and phase carrier remain constant.
- Bit '1'. Frequency carrier signal equal to f1.
- Bit '0'. Frequency carrier signal equal to f2.
- Advantage: Less affected by noise.
- Disadvantage : Require large bandwidth.



DIGITAL MODULATION

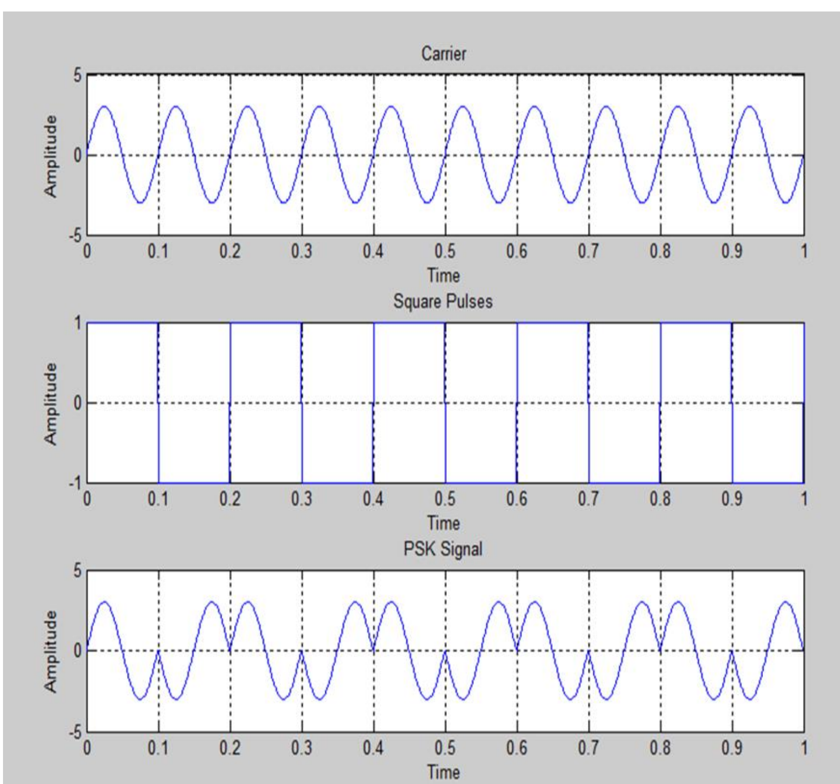
IMPLEMENTATION OF BINARY FSK





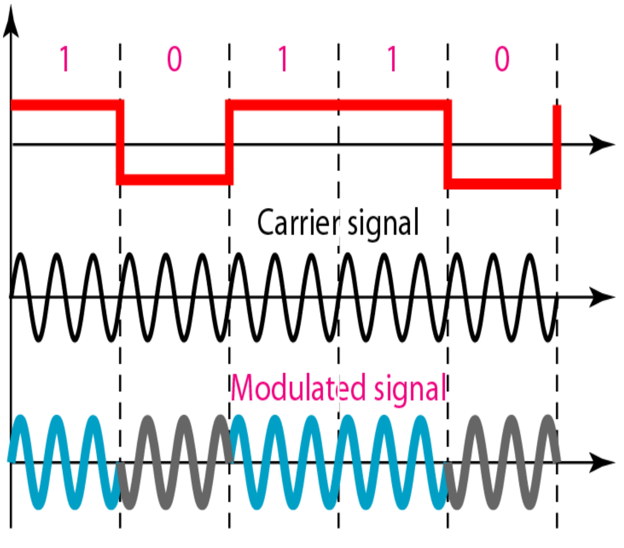
Phase Shift Keying (PSK)

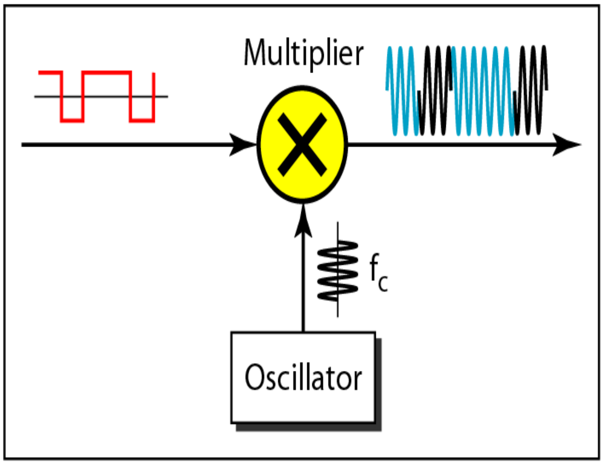
- Phase (Θc) of the analog carrier signal is varied proportional to the amplitude of digital information signal.
- Changing the Carrier Signal's Phase.
- Amplitude and Frequency carrier remain constant.
- The number of output phases is defined by $M = 2^n$.
- 2 phases are possible for the carrier which are logic '1' and logic '0'. One phase represents a logic '1' and other phase represents logic '0'.
- Advantages : 1) Less affected by noise compared to ASK
  2) Require bandwidth less than FSK.
- Disadvantage: Difficult to detect phase shift in case of phase difference is too small.

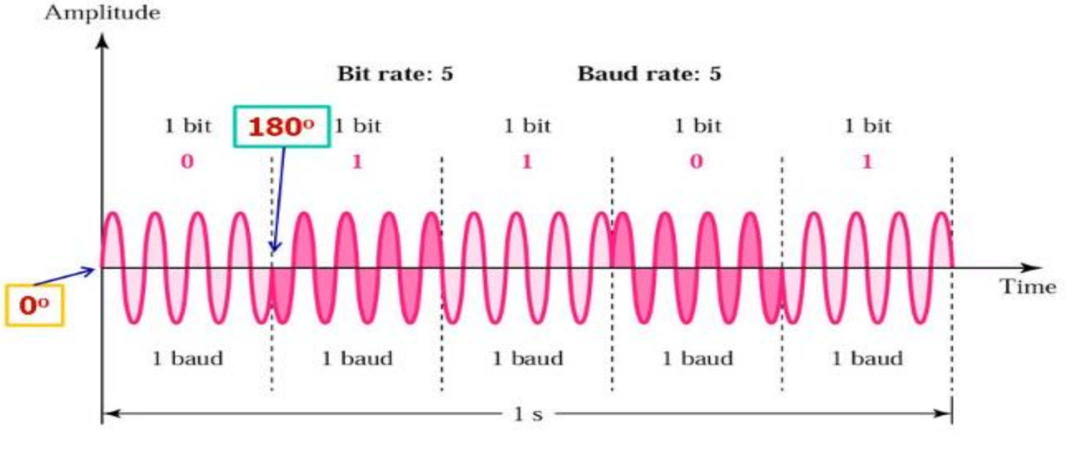


DIGITAL MODULATION

IMPLEMENTATION OF BINARY PSK



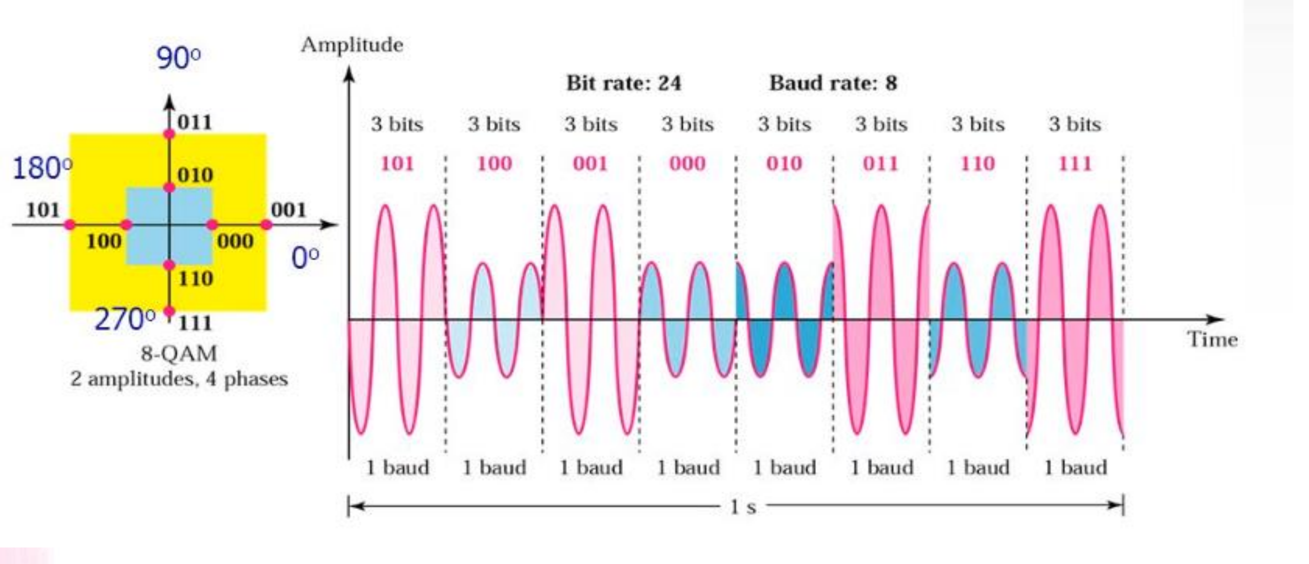




DIGITAL MODULATION

Quadrature Amplitude Modulation (QAM)

- Is a combination of Amplitude Shift Keying and Phase Shift Keying.
- Changing amplitude and phase of carrier signal
- Amplitude and Frequency carrier remain constant.
- Advantages : 1) Less affected by noise compared to ASK
  2) Require less bandwidth
- Disadvantage : Complex demodulation technique



SECTION REVIEW

1) Define modulation in communication system.

2) State the importance of a modulation process in communication system.

3) State the characteristics of a demodulation process.

4) With the aid of a diagram, explain the Amplitude Modulation (AM).

5) With the aid of a diagram, explain the Frequency Modulation (FM).

6) Explain the following pulse modulation techniques:
   i) Pulse Width Modulation (PWM).
   ii) Pulse Amplitude Modulation (PAM).

7) Given binary data 0111, sketch the data pulse, carrier signal and the output waveform of Amplitude Shift Keying.

8) A digital signal uses 4 levels to represent the data in the system. Calculate the number of bits needed per level and draw the digital signal for data 001110.

9) M-ary is a term derived from the binary word. If a digital signal has 2 levels, calculate how many bits are needed per level and sketch the digital signals if the input data is 10110001.

10) A signal in the frequency range 300 to 3400 Hz is limited to a peak swing of 8V. The signal is sampled using a minimum sampling rate for digital transmission and samples are quantized to 8 evenly spaced level. Calculate the frequency sampling, number of bits and transmission bit rate.

SECTION REVIEW

ANSWER

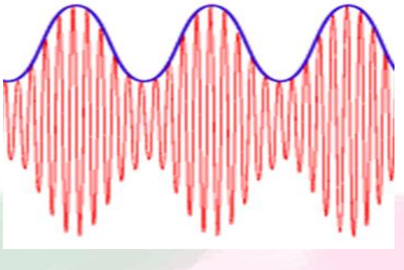
1) Modulation in communication system

    Modulation is a process of changing one or more properties of the high Frequency analog carrier signal in proportion with the values of amplitude information signal.

2) Importance of a modulation process in communication system
    a) To increase the frequency of information signal.
    b) To convert the analog signal to digital signal and vice versa.
    c) To avoid interfering with each other.
    d) To increase the bandwidth of the signal.
    e) To multiplex more number of signal.
    f) To reduce the antenna height and size.
    g) To reduce equipment complexity.

3) State the characteristics of a demodulation process
    a) Reverse process of modulation.
    b) Process extracting the information signal from the modulated-carrier signal.
    c) Done at receiver.
    d) Restore the original information signal.

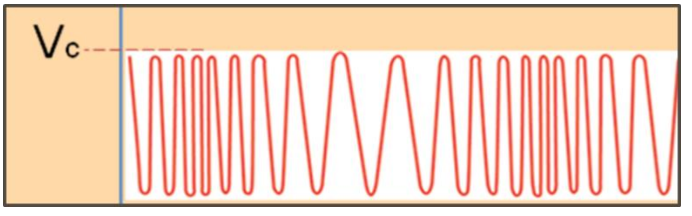
4) The diagram of Amplitude Modulation (AM) and explanation.



SECTION REVIEW

Amplitude (A) of carrier signal is varied proportional to the amplitude of analog Information signal. Frequency (W) and Phase (P) of carrier signal are constant. The higher amplitude of Information signal, the higher amplitude of carrier signal

5) The diagram of Frequency Modulation (FM) and explanation..



Frequency (F) of carrier signal is varied proportional to the amplitude of analog Information signal. Amplitude (A) and Phase (P) of carrier signal are constant. The higher amplitude of Information signal, the higher frequency of carrier signal.

6) Explanation of the following pulse modulation techniques:

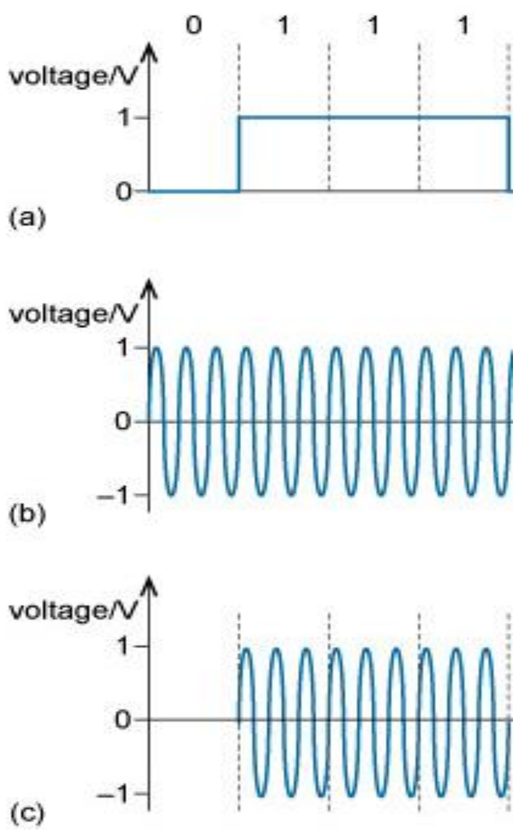
<u>Pulse Width Modulation (PWM)</u>
Width (W) of pulses is varied proportional to the amplitude of analog Information signal. Amplitude (A) and Position (P) of pulses are constant. The higher amplitude of Information signal, the wider of Pulses.

<u>Pulse Amplitude Modulation (PAM)</u>
Amplitude (A) of pulses is varied proportional to the amplitude of analog Information signal. Width (W) and Position (P) of pulses are constant. The higher amplitude of Information signal, the higher amplitude of Pulses.

SECTION REVIEW

ANSWER

7) Given binary data 0111, sketch the data pulse, carrier signal and the output waveform of Amplitude Shift Keying.



Data signal (information signal)

Carrier signal

ASK modulated signal

8) A digital signal uses 4 levels to represent the data in the system. Calculate the number of bits needed per level and draw the digital signal for data 001110.

Number of level = 4
Number of bits, n

$$n = \log_2(4)$$

$$n = \frac{\log 4}{\log 2} = \underline{\underline{2\text{ bits}}}$$

Level 4 (11)		11	
Level 3 (10)			10
Level 2 (01)			
Level 1 (00)	00		

SECTION REVIEW

ANSWER

9) M-ary is a term derived from the binary word. If a digital signal has 2 levels, calculate how many bits are needed per level and sketch the digital signals if the input data is 10110001.

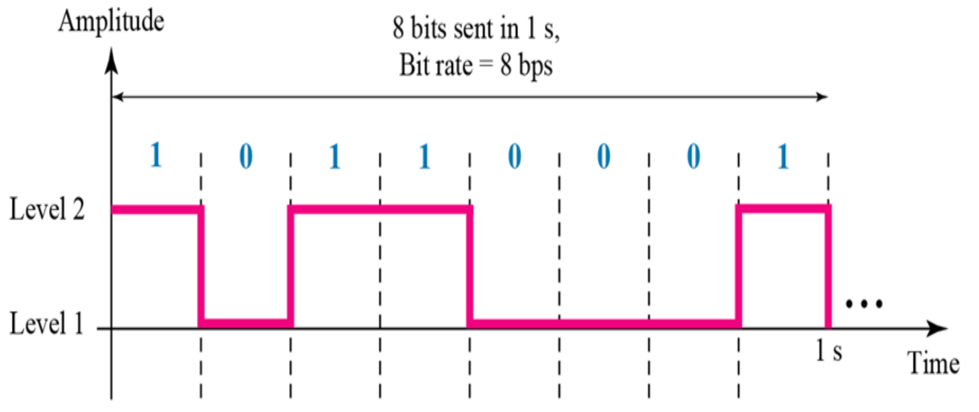
M = 2

$$n = \log_2 M$$

$$n = \log_2(2)$$

$$n = \frac{\log 2}{\log 2}$$

$$n = \underline{\underline{1\,\text{bit}}}$$



10) A signal in the frequency range 300 to 3400 Hz is limited to a peak swing of 8V. The signal is sampled using a minimum sampling rate for digital transmission and samples are quantized to 8 evenly spaced level. Calculate the frequency sampling, number of bits and transmission bit rate.

<u>Maximum sampling rate,</u>

Fs = 2fm(max)
= 2(3400)
= 6800 Hz @ 6.8kHz

<u>Number of bits,</u>

n = log2 L
= log2 8
= log 8/log 2
= 3 bits

<u>Bit Rate,</u>

R = Fs x n
= 6800 x 3
= 20.4 kbps

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e ISBN 978-967-2897-23-1