Analog and Digital communication

Modulation:

Process of change in characteristic of single with respect to change in the original message signal

Amplitute Modulation:

A process of change in amplitute of carrier signal with respect to original message signal (or) Modulation signal

Voltage Distribution:

Modulating $\rightarrow E_m \sin(2\pi F_m t)$

$$V_{am}(t) = [E_c + E_m \sin(2\pi f_m t)]$$

$$E_m = m F_c$$

$$E_c + m E_c \cdot \sin(2\pi F_m t)(\sin 2\pi f_c t)$$

$$[1 + m \sin(2\pi F_m t)] E_c (\sin 2\pi f_c t)$$

$$= F_c (\sin 2\pi f_c t) - \frac{E_c m}{2} \cos(2\pi F_c + F_m t)$$

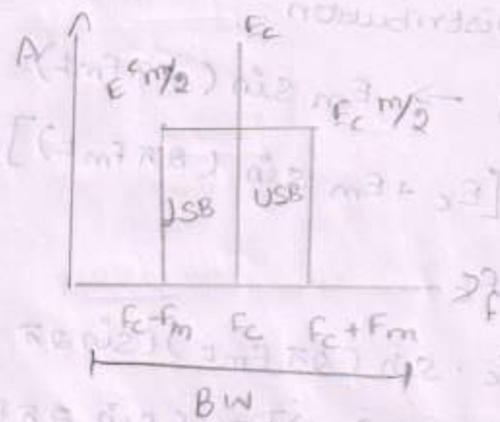
$$+ \frac{E_c m}{2} \cos(2\pi f_c - F_m(t))]$$

Frequency Spectrum and Band w

Modulation of known a coefficient of modulation The amount of change in the amplitude Present in the AM wave

Modulation Index defind as the ratio of E amplitute modulating signal carrier

$$m = \frac{E_m}{E_c}$$



BW (Band width) = $2F_m$ (max

Percentage Modulation

The percentage change in the amplitute of the output wave is called percentage modulation.

Modulation

$M = m \times 100$

critical | under | over modulation

$m = 1$ $m < 1$ $m > 1$

$E_c = E_m$ $E_c > E_m$ $E_c < E_m$

100% 50%

$V_{min} = 0$

$V_{max} = 2E_c$

Power Distribution:

$$P = \frac{V^2}{R}$$

where V refers to voltage
R resistance

$$P_c = \frac{(RMS)^2}{R}$$

$$RMS = \frac{E_c}{\sqrt{2}}$$

$$\frac{\left(\frac{E_c}{\sqrt{2}}\right)^2}{R} = \frac{E_c^2}{2R} = P_c$$

side Band:

$$P_{USB} = P_{LSB} = \frac{(AMP)^2}{2R}$$

$$= \frac{\left(\frac{mE_c}{2}\right)^2}{2R}$$

$$= \frac{(mE_c)^2}{8R}$$

$$= \frac{m^2 E_c^2}{8R}.$$

Sub $P_c = \frac{E_c^2}{2R}$

$$= \frac{m^2 P_c}{4}$$

$$P_{AM} = P_c + P_{USB} + P_{LSB}$$

$$= P_c + \frac{2m^2 P_c}{4}$$

$$P_{AM} = P_c \left[1 + \frac{m^2}{2}\right]$$

Analog and Digital communication

Unit-I : Fundamentals of Analog communication :

* Introduction :

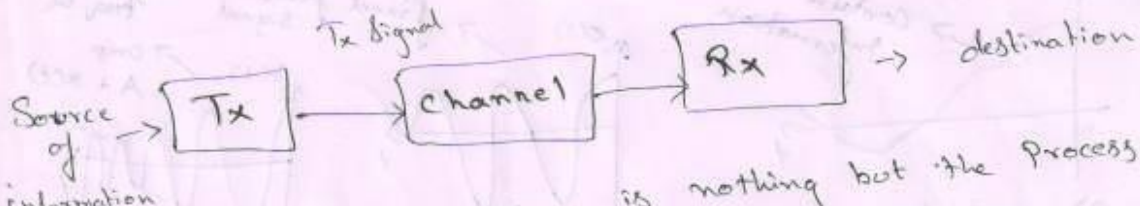
Need of communication System => to transmit information

* Bearing Signal refer as Baseband Signal.
  └> Base band is used the band of frequencies of the Signal from the Source.

* Transmitting technique
  - Analog
  - Digital

* Analog communication => Used the continous Signal to transmit the data [voice, image video etc]

* Simple Communication System :



Communication is nothing but the process of transmission of information from one pt to another.

* **Modulation:** The Process of Changing the characteristic of the carrier Signal w.r to the instantaneous change in the original msg Signal.

* Types of Modulation
  - Analog => Sinusoidal wave used as carrier
  - Pulse => Periodic Sequence of rectangular pulse
  - Digital

* **Need of Modulation:**

→ low freq or audio Signal are translated to higher freq. ↳ Small wavelength

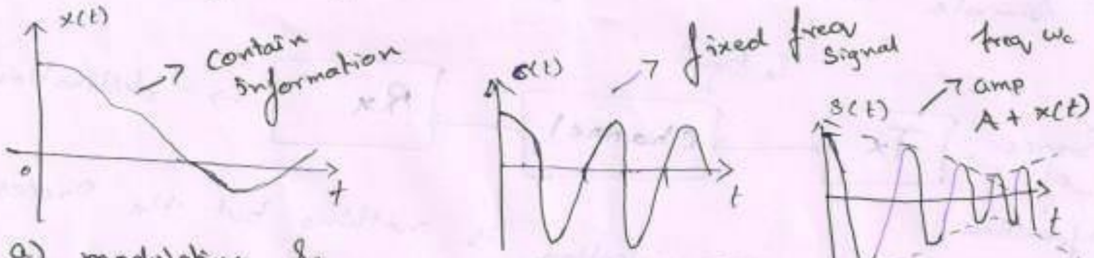
→ To remove interference.

→ Reduction of noise.

* **Principle of AM:**

↳ It is one of the Continuous-wave modulation System.

→ The Process of Changing the amplitude of carrier wave w.r to original Signal.



a) modulating Sg b) carrier Sg c) modulated Signal

$c(t) = A\cos\omega_c t$

$S(t) = x(t)\cdot\cos\omega_c t + A\cos\omega_c t$

$= [A + x(t)]\cos\omega_c t$

A → max amplitude of the Carrier wave.

ω_c → Carrier freq

Note: freq + Phase [constant]

amplitude is varied.

* AM Envelope :

=> Fig 3 AM wave time-varying amplitude
called Envelope of AM wave

-> Envelope of the modulated signal shape is same as the baseband signal.

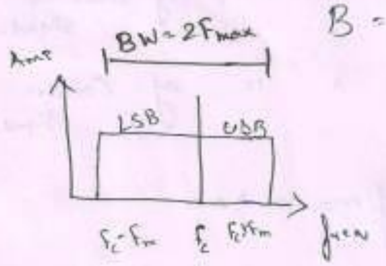
$S = [A + x(t)] \cos\omega_c t$ $E(t)$ => envelope of AM

$E(t) = A + x(t)$ -> baseband signal is recovered by envelope.

* Frequency spectrum and bandwidth :

=> Bandwidth defined as twice that of the highest modulating freq (f_{max})

$B = 2f_{(max)}$

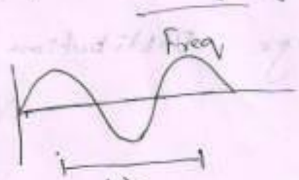


-> carrier freq

LSB -> Lower side band [below CF]

USB -> upper " " [above carrier freq]

Wave length (λ) = $\frac{\text{Speed of light}}{\text{Freq}}$



freq = 300K Hz to 3 MHz

$f_c - f_{m(max)}$ -> diff of freq

$f_c + f_{m(max)}$ -> sum of freq.

$$x(t)\cos\omega_c t = x(t)\left[\frac{1}{2}e^{j\omega_c t} + e^{-j\omega_c t}\right]$$

$$= \frac{1}{2}x(t)e^{j\omega_c t} + \frac{1}{2}x(t)e^{-j\omega_c t}$$

$$= \frac{1}{2}\left[x(\omega-\omega_c) + x(\omega+\omega_c)\right]$$

* Angle Modulation : freq modulation, Phase modulation

→ freq or phase of the carrier signal is varied acc w.r to msg signal, keeping amp constant.

→ (θ) Phase angle is varied

Carrier Signal ⇒ $m(t) = V_c \cos[\omega_c t + \theta(t)]$

* Application: Radio broadcasting, cellular radio, satellite comm, μ wave comm, etc.

* Phase modulation:-

→ Phase angle ϕ is varied w.r to modulating signal $x(t)$.

$$c(t) = A\cos(\omega_c t + \theta_0)$$

$\phi = \omega_c t + \theta_0 \Rightarrow \phi = \omega_c t$

$\phi_i = \omega_c t + k_p \cdot x(t)$ $\quad k_p \rightarrow$ Phase Sensitivity → radians/volt.

$S(t) = A\cos\phi_i$ → Phase modulated wave.

$$S(t) = A\cos[\omega_c t + k_p \cdot x(t)].$$

* Frequency modulation:

→ ω_i (instantaneous freq) varied with a msg signal $x(t)$. ω_c → unmodulated carrier freq.

Expression:

$$\omega_i = \omega_c + k_f \cdot x(t)$$

k_f → frequency sensitivity
↳ Hz/volt.

$C(t) = A\cos(\omega_c t + \theta_0)$
$C(t) = A\cos\phi$

Freq of modulated wave

$S(t)$ $A\cos\phi_i$ ⇒ A constant. ϕ changed.

$\phi = \omega_c t + \theta_0$

diff ⇒ $\frac{d\phi}{dt} = \omega_c$ or $\phi = \int \omega_c dt$

So $\phi_i = \int \omega_i dt$

$\phi_i = \int [\omega_c + k_f x(t)] dt$ (from 1st eqn)

$= \omega_c t + k_f \int x(t) dt$

$$\boxed{S(t) = A\cos\left[\omega_c t + k_f \int x(t)\, dt\right]}$$

* Frequency modulation:

→ ω_i (instantaneous freq) varied with a msg Signal $x(t)$. ω_c → unmodulated Carrier freq.

Expression:

$$\omega_i = \omega_c + k_f \cdot x(t)$$

k_f → frequency sensitivity
↳ Hz/volt.

$$c(t) = A\cos(\omega_c t + \theta_0)$$

$$c(t) = A\cos\phi$$

Freq of modulated wave

$$s(t) \quad A\cos\phi_i \quad \Rightarrow$$

A constant.
ϕ changed.

$$\phi = \omega_c t + \theta_0$$

diff ⇒ $\frac{d\phi}{dt} = \omega_c$ or $\phi = \int \omega_c dt$

So $\phi_i = \int \omega_i dt$

$$\phi_i = \int [\omega_c + k_f\, x(t)]dt \quad (\text{From } 1^{st} \text{ equ})$$

$$= \omega_c t + k_f \int x(t)\, dt$$

$$\boxed{s(t) = A\cos\left[\omega_c t + k_f \int x(t)\, dt\right]}$$

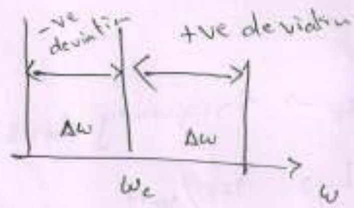
* Frequency deviation :

  → max change of instantaneous freq (ω_i) from Avg freq

$$\omega_i = \omega_c + k_f \cdot x(t)$$

$\Delta\omega$ → max freq deviation.

$$\Delta\omega = |k_f \cdot x(t)|_{max}$$



Note:

PM is often known indirect FM because PM → amt of Phase Shift is varied in ω_c (carrier freq) chang

PM → Phase angle varies linearly with x(t) (baseband sig

FM → " " " " integral of x(t)

Performance Table:

FM	PM
* $S(t) = V_c \sin[\omega_c t + m_f \sin\omega_m t]$	* $S(t) = V_c \sin[\omega_c t + m_p$
* Noise immunity better than AM + PM	* Better than AM
* SNR is better than PM	* worse than F
* FM widely used	* In mobile system

AM	Angle modulation
* Envelope of AM wave is dependent of $x(t)$	* Envelope is constant
* Zero crossing Perfect regularity	* it differs.

Bandwidth of FM $= 2[\Delta f + f_m]$

↳ depent of modulation index + signal freq

→ angle modulated wave sets of side bands, it has wider BW

* low index → $m < 1$
* medium " → $(1 < m < 10)$
* high " → $(m > 10)$

Δf → Peak freq deviation

f_m → modulating freq Hz

* According to Carson's Rule :

a) low modulating index $f_m >> \Delta F$

$B = 2 f_m$ Hz

b) for high modulating index

$B = 2 \Delta f$ Hz.

* Deviation Ratio :-

$$DR = \frac{\Delta f_{max}}{f_m (max)}$$

$\Delta f \to$ max Peak freq deviation (Hz)

$f_{max} \to$ max modulating Signal freq (Hz)

Types of FM :- Narrow band FM (k_f is small)

Wide band FM (k_f is large)

Modulation index of FM :-

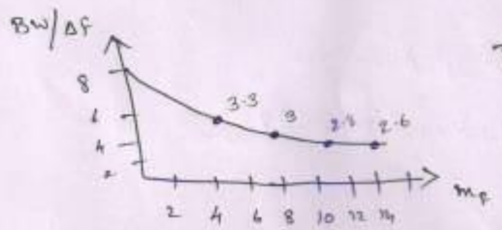
$$m_f = \frac{\text{freq deviation}}{\text{modulating freq}} = \frac{\Delta\omega}{\omega_m} = \frac{K_f V_m}{\omega_m}$$

Bandwidth:

↳ depent on modulation index (m_f)

$m_f \uparrow \Rightarrow$ BW also $\uparrow$

Universal Curve : $\to$ Schwartz determined graph for BW of FM



It shows variation of the BW B normalised with respect to Δf against m_f

$$B.W = 2(\Delta\omega + \omega_m) \quad ; \quad m_f = \frac{\Delta\omega}{\omega_m}$$

$$B.W = 2(m_f \cdot \omega_m + \omega_m) = 2(1 + m_f)\omega_m$$

Carson's rule:-

1) $\Delta\omega << \omega_m$ (ie) $m_f << 1$ => narrow band FM

$$BW = 2\omega_m^*$$

2) $\Delta\omega >> \omega_m$ (ie) $m_f >> 1$ => wideband FM

$$BW = 2(m_f)\omega_m$$

But $m_f \omega_m = \Delta\omega$

$$B.W = 2\Delta\omega^*$$

Note:

* AM is called linear type of modulation
  ↳ because there is no intermodulation (or) Cross - Product Sideband

Phase modulation : Analytic View!

-> PM deviation in carrier freq ω_c is linearly proportional to baseband freq ω_m

-> FM deviation is independent.

SSB - Single Side Band

Need of Modulation

Noise Calculation

Noise factor denoted by F

Noise factor F of an amplifier or any other networks is defind as ratio of available S/N Power ratios at the input to the available S/N Power ratio at the output.

$$\frac{P_{si}}{P_{Ni}} \times \frac{P_{No}}{P_{so}}$$

∴ S/N - singnal to the noise (unit less)

Raw Noise figure:

If the Noise factor expressed in decepial (unit) it is known as Noise figure for circuit

Noise figure = 10 log$_{10}$ F

Noise Temperature:

The available Noise Power is directly propostional to temperat and it is independent of value of resistance. This power specify in terms of temperature is called Noise Temperature.

Noise due to the several amplifier in cascade

Evaution of description of SSB Technique :-

[∵ SSB - Single Side Band]

In DSB the carrier frequency is occur there are In order to decrease the carrier we are going to SSB.

Advantage of SSB

* Conserves Band width
* Power requirement is reduced
* Noise at the reciever circuit is reduced
* Fading effect is absent

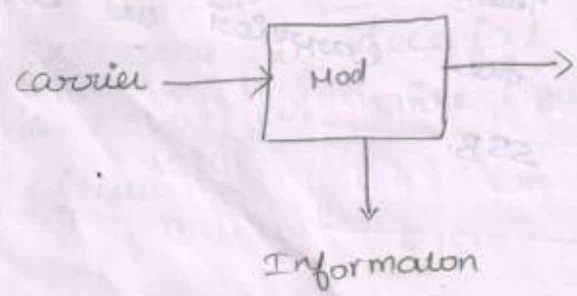
Fading effect:

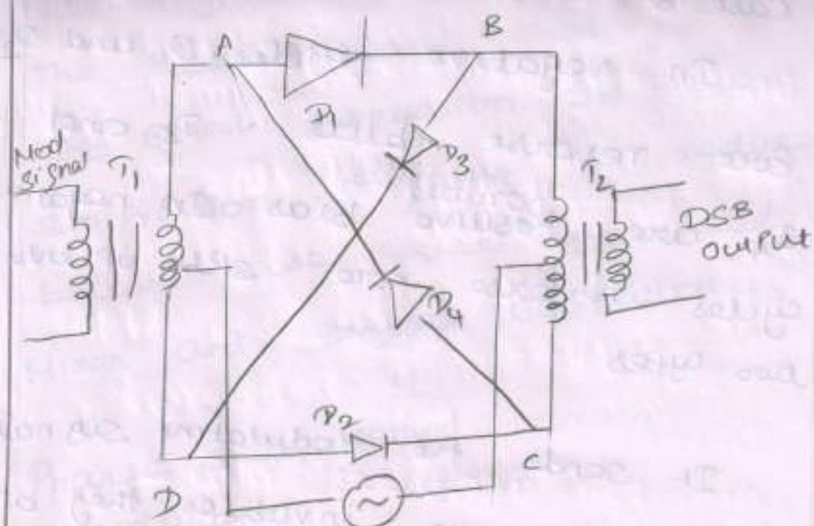
Three Side Bands are interference at that point is called fading effect

Disadvantage

complex & turning circuits

To suppression of carrier with the use of Base Modulat





T_1 - Input Transformer

T_2 - output Transformer

D_1, D_2, D_3, D_4 - Diaode.

In Postive cycles D_1 and D_2 are forward bias. they are split up into two equal and opposite so they are cancel each other. In forward Bias the power will be produced. In D_3 and D_4 are reverse bias. The carrier signal ~~it~~

Case 2:

In Negative cycles D_1 and are reverse bias. D_3 and D_4 are ~~Positive~~ forward bias In nega cycles reverse are split up in two types

It sends the Modulating sig n D_3 and D_4 are invisbles they direct connect with output.

To supress the side Band there are two types they are

* filiter method
* Pharse swit Shift Method

The Base Modular is suppresed the carrier and then go side Band suppression In that suppression Q value is reduced because we use Balanced Mixer and crystal oscillator.

Phase Shift Method :

Demodulation :

The process of extracting the information signal from the modulated signal

Naro Band FM	wide Band FM
In Naro Band FM modulation Index less than 1	Modulation Index is greater than 10
In Naro Band, Band width is $BW = 2F_{(max)}$	In wide Band $WB = 2(\delta + F_{max})$

Advantage of Angle Modulation :

Noise Immunity is high

There is is inforn

improvement in signal to Noise

Pow

power utilisation is

efficient.

Capture effect is high

Disadvange:

PM FM requires more

Band width.

circuit complexicity and

cost is high.

Need of Modulation

It is used to reduce the

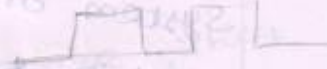
noise

Unit II

Digital Modulation (Discrete time data)

high frequency analog carrier wave is modulated with respect to the low frequency digital Information signal.

(Digital Radio)



* Amplitute Shift Keying
* Frequency Shift Keying
* Pharse Shift Keying.
* Quatratic Amplitute Modulation (QAM)

Application of Digital Modulation:

Relatively low speed voice Band communication Modeums

High speed data transmission System such as Digital subscriber line (DSL)

Digital microwave and Microwave Sattelite communication

cellular telephone are Personal communication syst

Definition for Bit Rate:

Information capacity of System or else simply the nu of bit transmitted during one second and expressed as Bit Per Second [BPS]

Bit:

The most Basic digital sym used to represent the information is the binary dig (or) bit

Baud rate

It is also change the rat of change of signal after encoding and modulation occurs.

$$\text{Baud rate} = \frac{1}{t_s}$$

t_s - Time of one signal in element

unit - seconds

Baud rate is otherwise is called as symbol rate, Transmission rate and modulation rate.

Binary system:

eg. BPSK - Binary Pharse Shift Key. In this system bit rate is equal to Modurate Baudrate

High level system:

BPS - Bit per second is also Greater than the Baud rate.

Information capacity

It is donoted as I

$I \propto B_W \times t$

Band width Increases at the same time Information capacity is also Increases. It is also known Hartley's Law

$$I = B \log_2 (1 + S/N)$$

This Thereom is known as Shanon ratio

S/N - Signal to Noise ratio (un

B - Bandwidth (Hz)

In $\log_{10}$ the formula is

$$I = 3.32 \, B \log_{10} (1 + S/N)$$

M Aray encoding :

$$M = 2^n.$$

M Aray encoding represents digit that corrosponds to nu of condition, level, or comina are possible for a given nu of binary.

$$\boxed{N = \log_2 M}$$

N - Number of bits

Nyquist Band width Equation

$$F_b = 2B_W$$

F_b - Information capacity (or) Channel capacity.

M - Number of condition.

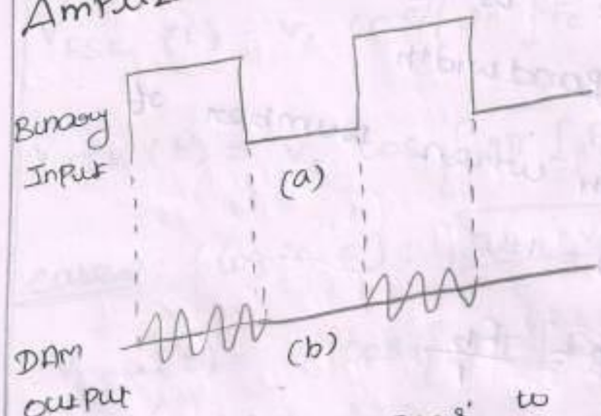
$$F_b = 2B \log_2 M$$

$$B = \frac{F_b}{\log_2 M}$$

B - Band width.

$$B = \frac{F_b}{N}$$

Amplitude Shift Key:



Binary Input signal goes to 0's & 1's. ASK Binary means is a special form of Amplitude Modulation. There are two logic 1 & 0

$$\boxed{V_{ask}(t) = \left[1 + V_m(t)\right]\left[\frac{A}{2}\cos(\omega_c t)\right]}$$

Case 1:

$$V_{ask}(t) = [1+1]\left[\frac{A}{2}\cos(\omega_c t)\right]$$

$$= 2\,\frac{A}{2}\cos(\omega_c t)$$

$$= A\cos\omega_c t$$

Case 2: (logic 0)

$$V_{ask}(t) = [1-1]\left[A/2\cos(\omega_c t)\right]$$

$$= 0\left[A/2\cos(\omega_c t)\right]$$

$$= 0$$

Bit Rate is equal to the Nyquist Bandwidth

Band width when number of bit $(N) = 1$

$$B = \frac{F_b}{1}$$

$$\boxed{B = F_b}$$

Frequency shift key:

$$V_{FSK}(t) = V_c \cos\left\{2\pi\left[F_c + V_m(t)\,\Delta f\right]t\right\}$$

Δf - Frequency deviation

$V_m(t)$ - Information signal

F_c - Frequency carrier

Frequency shift key is frequency carrier is change with respect to the information signal.

case 1: (logic 1) $\boxed{\text{Sub } V_m(t) = 1}$

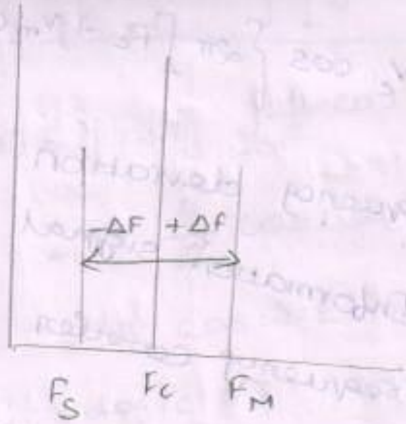
$$V_{FSK}(t) = V_c \cos\left\{2\pi\left[F_c + 1\,\Delta F\right]t\right\}$$

$$V_{FSK}(t) = V_c \cos\left\{2\pi\left[F_c + \Delta F\right]t\right\}$$

case 2: (logic 0): $\boxed{\text{Sub } V_m(t) = -1}$

$$V_{FSK}(t) = V_c \cos\left\{2\pi\left[F_c - 1\,\Delta F\right]t\right\}$$

$$V_{FSK}(t) = V_c \cos\left\{2\pi\left[F_c - \Delta F\right]t\right\}$$



F_M = Mark frequency

F_S = Space frequency

Frequency Deviation

$$\Delta f = \frac{|F_m - F_s|}{2} \quad (Hz)$$

Unit = Hz

$$B = \left| (F_s - F_b) - (F_m \mp F_b) \right|$$

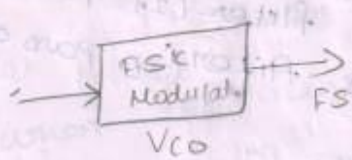
$$F_s - \cancel{F_b} - F_m \mp \cancel{F_b}$$

$$B = |F_s - F_m| + 2F_b$$

$$\boxed{B = 2\,|\Delta f + F_b|}$$

Transimiter

VCO - Voltage control oscillator



Truth Table

I/P	O/P
logic 1	F_m
logic 0	F_s

When logic 1 is then the input the output is F_m and is logic 0 output is F_s

FSK Receiver :

Three types of Reciever

* Non coherent
* Coherent
* PLL (Pharse lock loop)

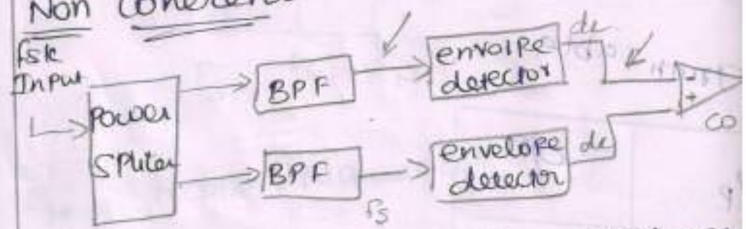
There are three types of filter

High Pass filter (It pass only

Low Pass filter (It pass only

Band Pass filter (It pass only

Non coherent



Envolpe detector is Estimat

Power estimation.

BPF - It allows specific f

range.

Power - compare two pow

and gives output.

Because no frequency is

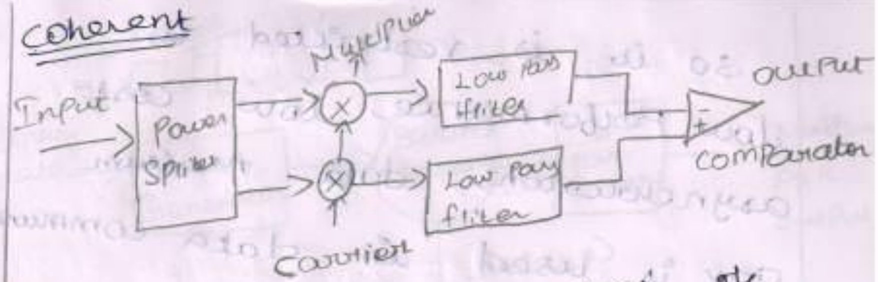
in demodulation process that is

synchronosed ethier in phase

frequency or both with resp

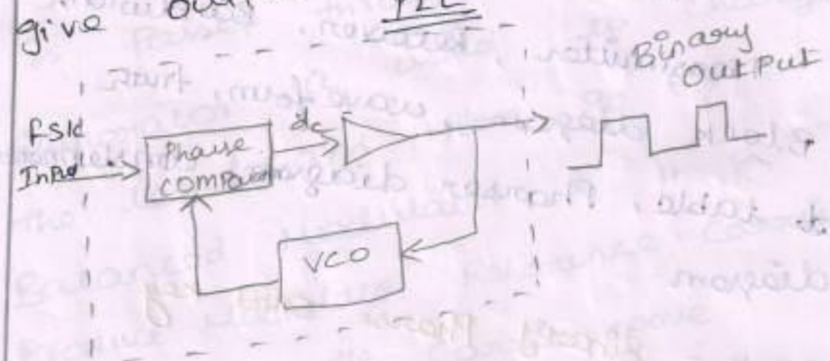
incoming FSK signal.

Coherent



when the input gives the Power Spliter split into two signal and the ~~sar~~ carrier is multiple and then go low pass filter the comparator then analysis low frequency give output

PLL



▷ → amplifier.

Demodulate using dc error voltage

Disadvantage

Other name of FSK is Binary FSK. poor error performance compare to the PSK, QAM

so it is restricted to low Performance, low cost asynchronons data Modeum FSK is used in data com over analog, voice band tel line.

Pharse Shift key:

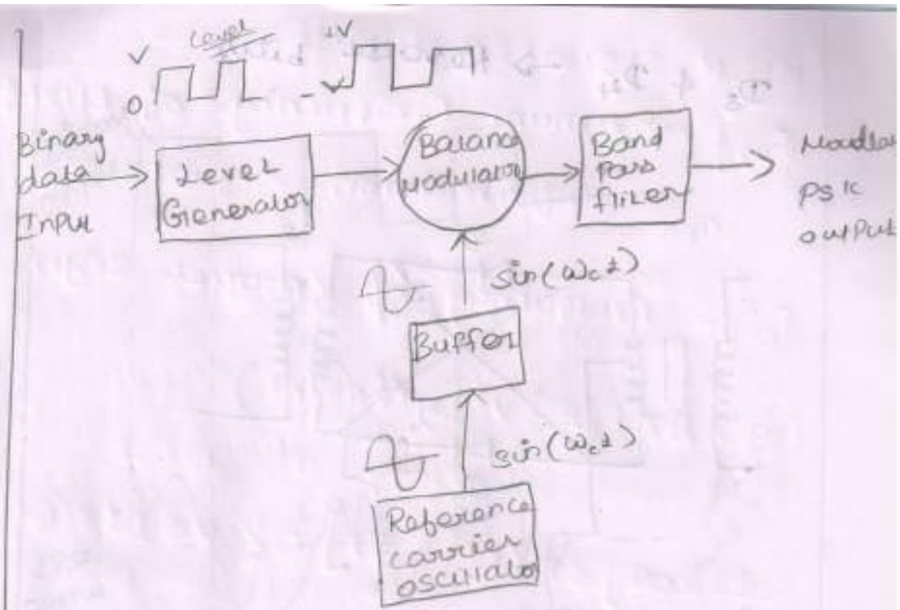
B PSK
Q PSK
8 PSK
16 PSK

Transimitter, Receiver, Bandwidth Block diagram, wave form, truth table, Pharser diagram, constell diagram.

BPSK - Binary Pharse Shift key

N = 1
M = 2

when M = 2 The input is 2 and the output is also 2.



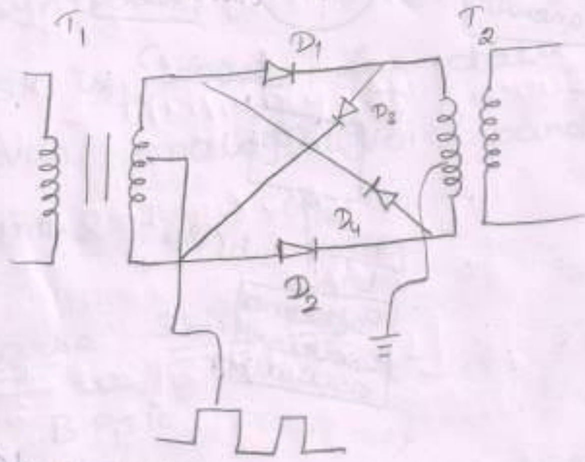
when the Input is Binary data is passed through the level Generator it used to change the level and then go to the Balanced Modulator it is a Product Modulator Reference Carrier to Generate the Carrier wave signal then ~~go to~~ give output

0° Phase shift → $\sin \omega_c t$

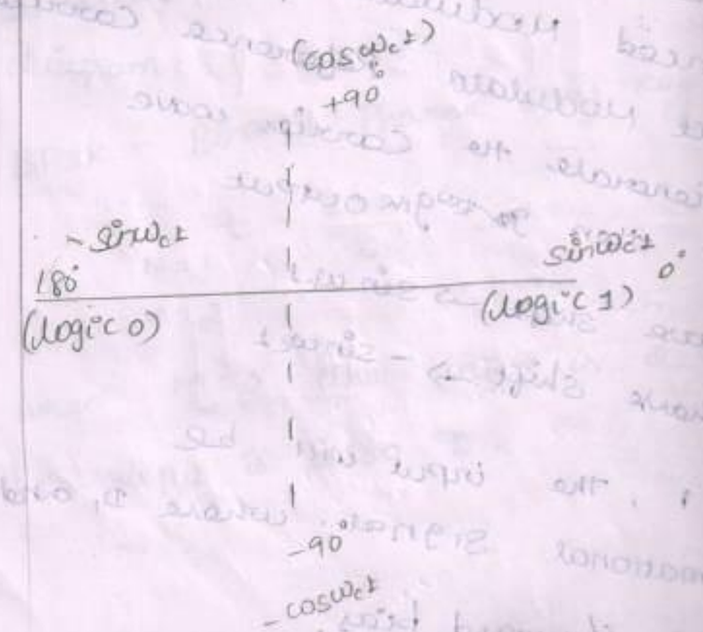
180° Phase shift → $-\sin \omega_c t$

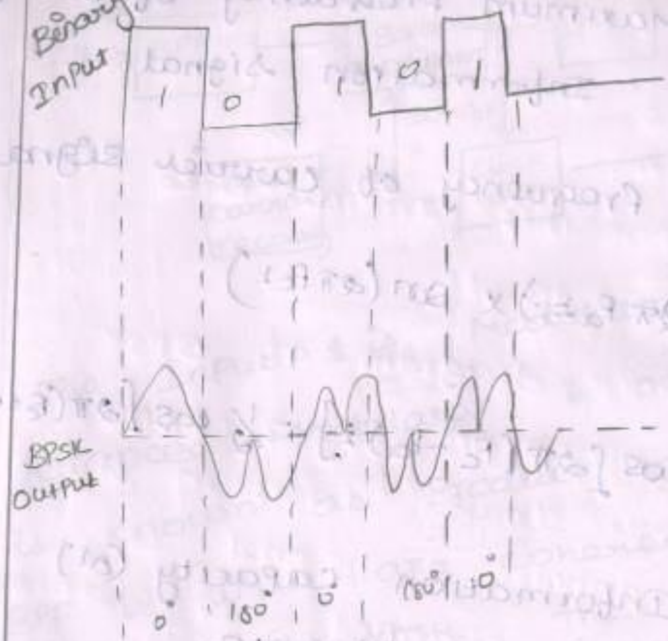
logic 1, the input will be Informational signal. where D_1 and D_2 → forward bias.

D_3 & D_4 → Reverse bias

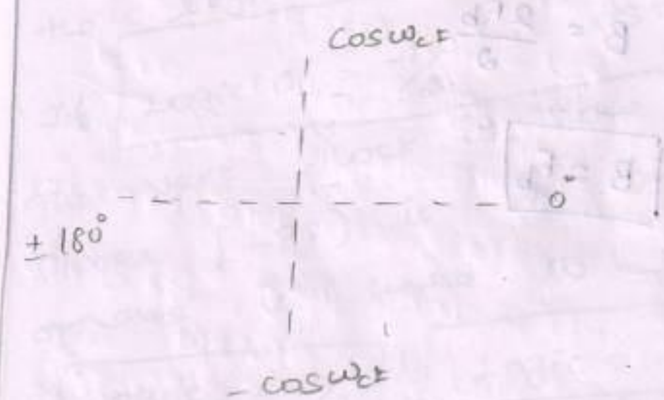


I/P	O/P
Logic 1	0°
Logic 0	180°





Binary output	Output Phase
Logic 0	180°
Logic 1	0°



BPSK Signal Bandwidth:

F_a - Maximum frequency of Information signal

F_c - frequency of carrier s

$(\sin 2\pi F_a t) \times \sin(2\pi F_c t)$

$\frac{1}{2}\cos[2\pi(F_c - F_a)t] - \frac{1}{2}\cos[$

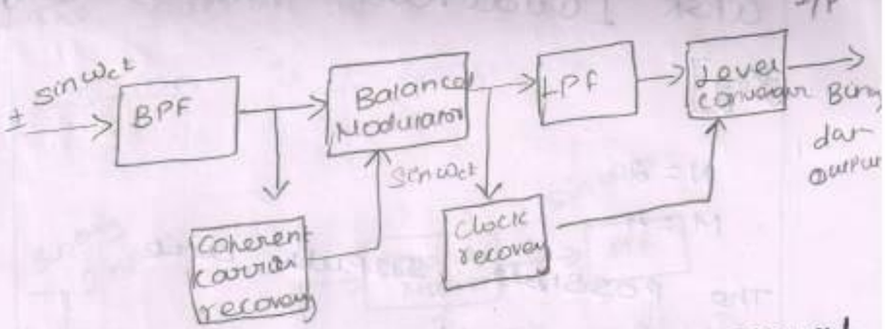
F_b - Information capacity (Input Bit rate.

$$F_a = \frac{F_b}{2}$$

$\frac{1}{2}\cos[2\pi(F_c - \frac{F_b}{2})t] - \frac{1}{2}\cos$

$$B = \frac{2F_b}{2}$$

$$\boxed{B = F_b}$$



The Input is Modulated signal Balanced Modulator is otherwise is known as Product Modulator BPF is input are coherent carrier recovery the working Banced Modulator works is add two $\sin\omega_c t$.

$$\boxed{\text{Logic } 1 = \frac{1}{2}(1 - \cos 2\omega_c t)}$$

If logic 0 is $-\sin\omega_c t$ and the logic 1 is $\sin\omega_c t$ is

$$\boxed{\text{Logic } 0 = -\frac{1}{2}(1 - \cos\omega_c t)}$$

If The Next Block is Low Pass filter (LPF) it is filter works on the is used to filter.

$$\boxed{\text{Logic } 1 = \frac{1}{2}} \qquad \boxed{\text{Logic } 0 = -\frac{1}{2}}$$

QPSK [Quatrature Pharse S
key]

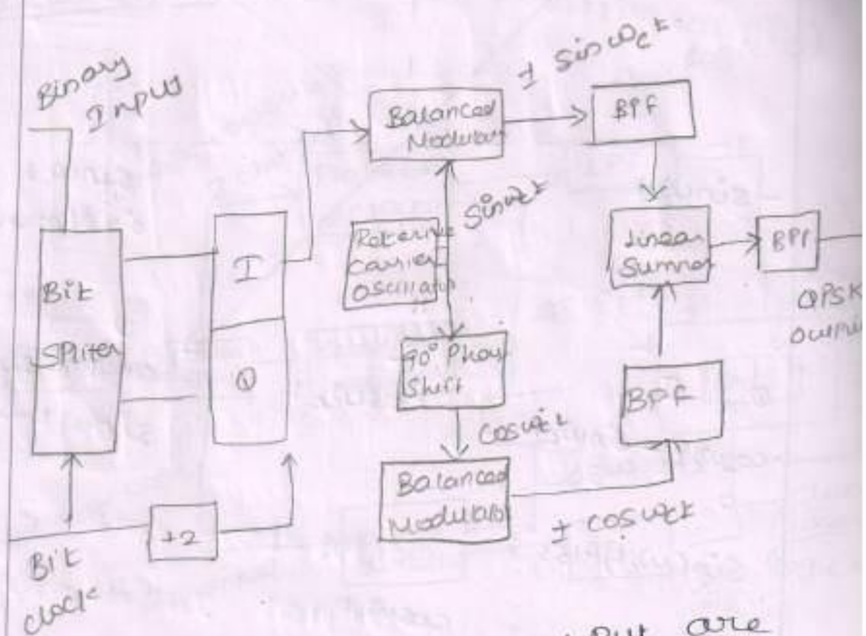
$N = 2$
$M = 4$

The possible Inputs are
00, 01, 10, 11

In Block diagram of QPSK
upper Position is I channel
and the lower position is
Q-channel. I-channel is
Inpharse channel. Q-char
is [90° outside of the carr
is called Q-channel.

In Block diagram of QP
the Balanced Modulator
two input one input is
user given (Binary input) and the other
input Retorive carrier osc
gives It gives $\sin \omega_c t$. and
the lower Position is ou
Pharse angle so 90° phar
shift add and then 90 gi
$\cos \omega_c t$. and the output
of the upper Posistion

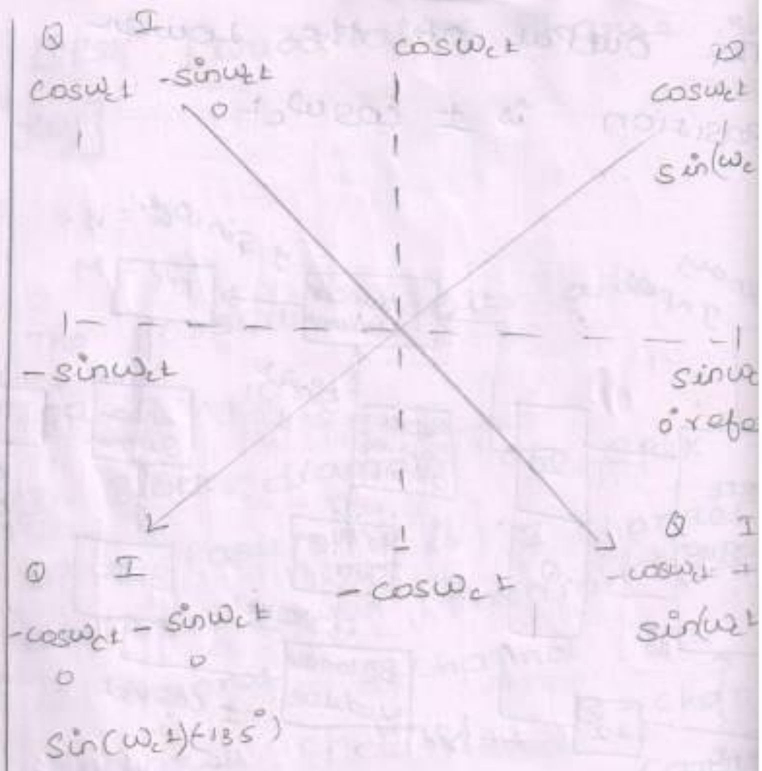
The output of the Lower position is $\pm \cos\omega_c t$.

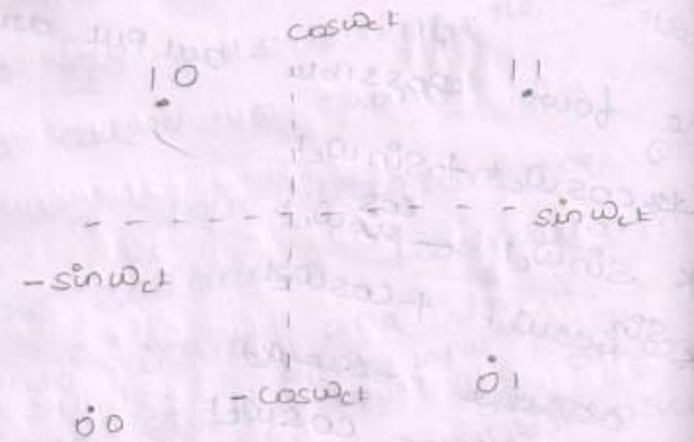


The four possible output are

* $\cos\omega_c t + \sin\omega_c t$
* $\sin\omega_c t - \cos\omega_c t$
* $\cos\omega_c t + \cos\omega_c t$ ~~cos ω_c t + sin ω_c t~~ $- \cos\omega_c t$
* $-\sin\omega_c t - \cos\omega_c t$

Binary Input		QPSK output Phase
Q	I	
0	0	$-135°$
0	1	$-45°$
1	0	$135°$
1	1	$45°$





Bandwidth of QPSK

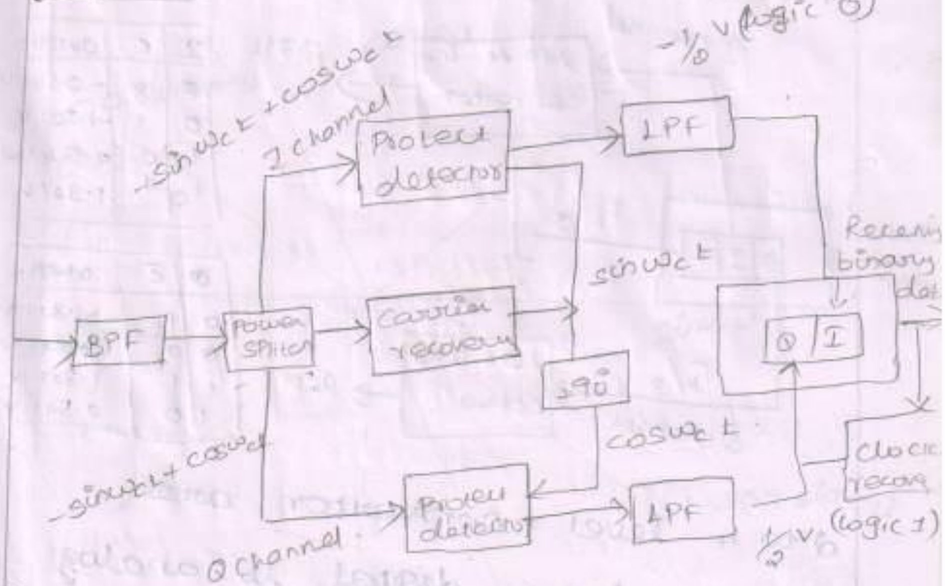
$\sin(2\pi f_a t) \times \sin(2\pi f_c t)$

$= \frac{1}{2}\cos(2\pi(f_c - f_a)t - \frac{1}{2}\cos[2\pi$

$f_a = \frac{f_b}{4} \Rightarrow 2f_a = \frac{2f_b}{4}$

$B = \frac{f_b}{2}$.

Receiver of QPSK:



$I = \{-\sin\omega_c t + \cos\omega_c t\}\{\sin\omega_c t\}$

$= -\sin^2\omega_c t + (\cos\omega_c t)(\sin\omega_c t)$

$= -\frac{1}{2}(1-\cos 2\omega_c t) + \frac{1}{2}\sin(\omega_c+\omega_c)t + \frac{1}{2}\sin(\omega_c-\omega_c)t$

$= -\frac{1}{2}V$ logic 0

$Q = \{-\sin\omega_c t + \cos\omega_c t\}(\cos\omega_c t)$

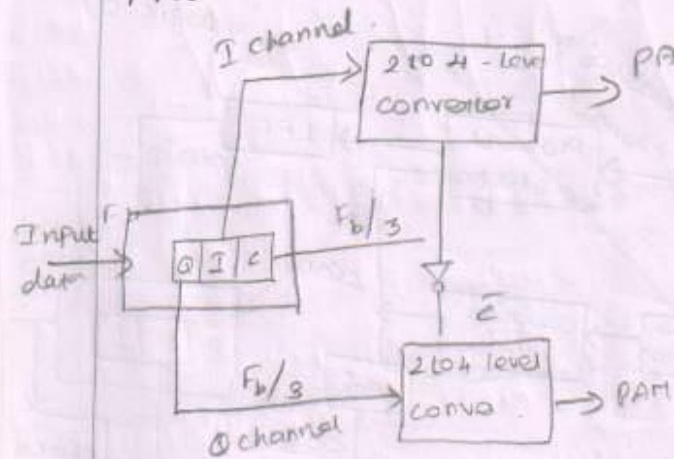
unwanted expression are filter

The expression is

$Q = \frac{1}{2}V$ logic 1

8 PSK :

Transmitter Block :



Out

I	C
0	0
0	1
1	0
1	1

Q	$\bar{C}$
0	1
0	0
1	1
1	0

2 to 4 level convertor are parrell input digital to anc compu convertor [DAC] . In transmiter end DAC . In recei end ADC [Analog to digital]

I or Q bit determines the polarity of the output analog signal

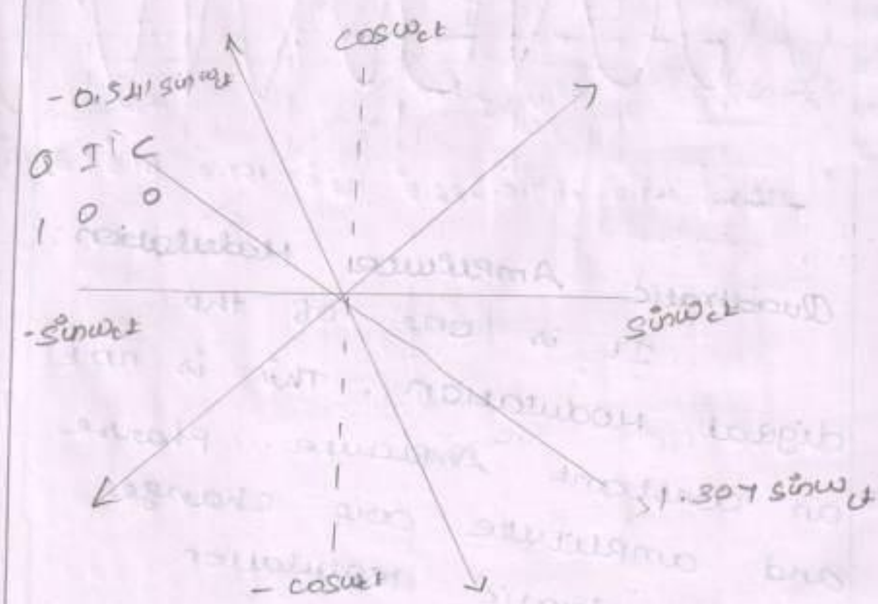
logic 1 → + Volt

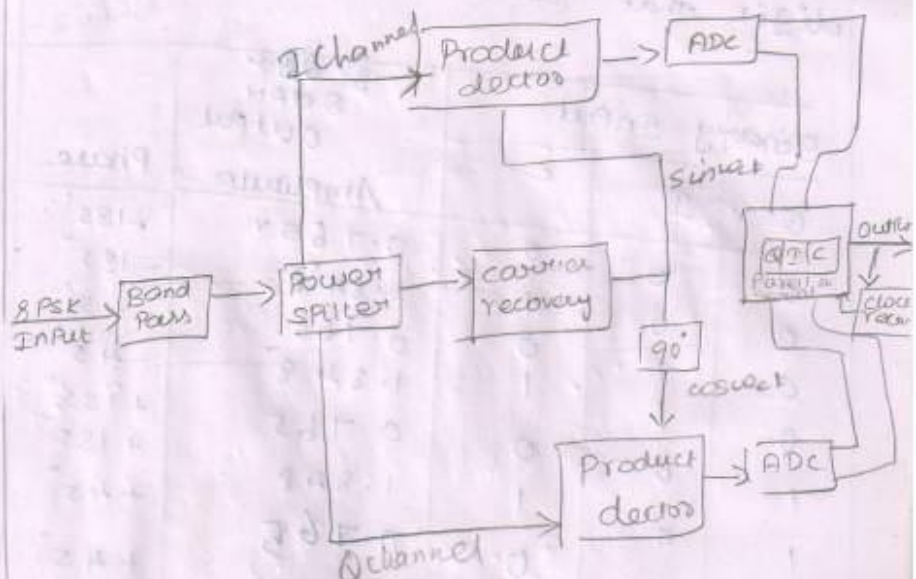
logic 0 → - Volt

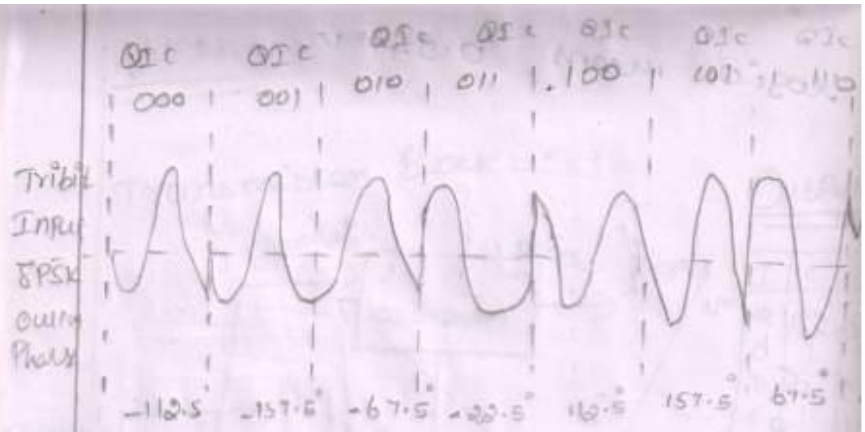
C or $\bar{C}$ determine magnitut

logic 1 means 1.307 V

logic 0 means 0.541 V





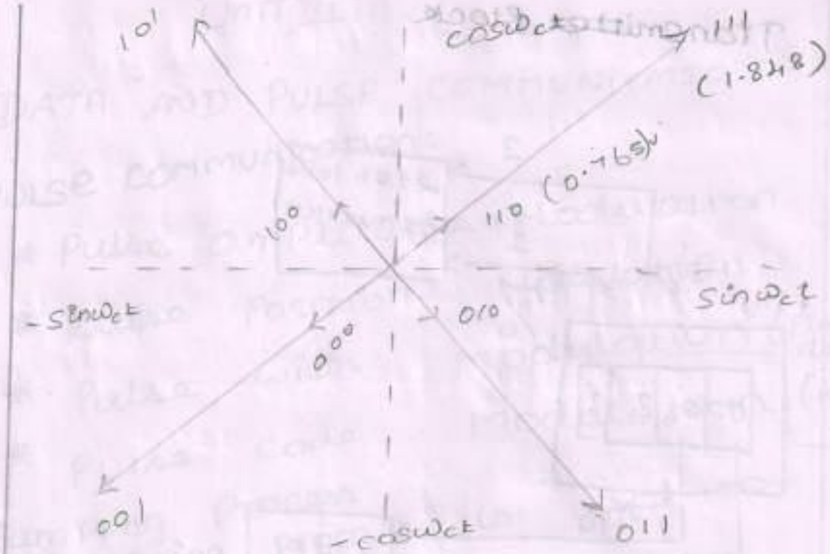


Quadratic Amplitude Modulation

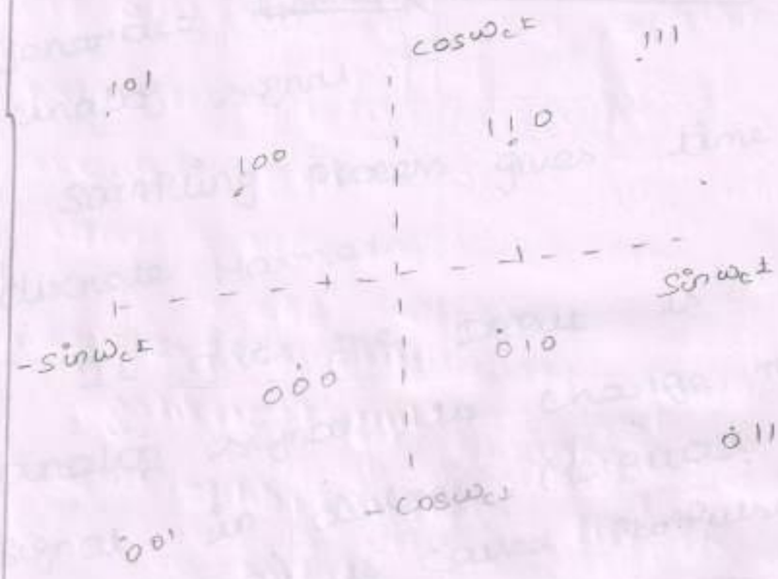
It is one of the digital modulation. This is not an constant Amplitude. phase and amplitude are change in Quadratic modulation

The M array represent the digit that correspond the number of

Binary Input			8 QAM output	
Q	I	C	Amplitude	Phase
0	0	0	0.765V	-135°
0	0	1	1.848	-135°
0	1	0	0.765	-45°
0	1	1	1.848	-45°
1	0	0	0.765	+135°
1	0	1	1.848	+135°
1	1	0	0.765	+45°
1	1	1	1.848	+45°



101
$\cos\omega_c t$
111
(1.848)
110 (0.765)
100
$-\sin\omega_c t$
$\sin\omega_c t$
010
000
001
$-\cos\omega_c t$
011



$\cos\omega_c t$
111
101
110
100
$\sin\omega_c t$
$-\sin\omega_c t$
010
000
011
$-\cos\omega_c t$
001

UNIT III

DATA AND PULSE COMMUNICATION

PULSE communication:

* Pulse amplitude Modulation.
* Pulse Position Modulation.
* Pulse width Modulation (or) (Pluse duration) (or) (Pluse Length)
* Pulse Code Modulation

Sampling Process

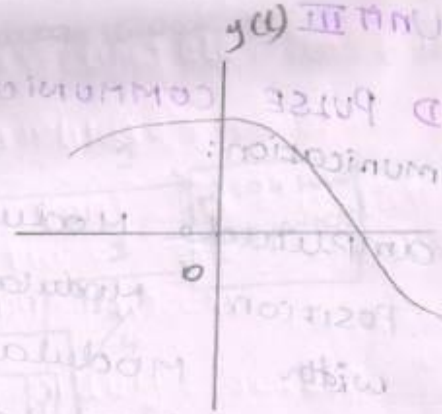
Sampling process is used to generate the pulse. form an analog signal.

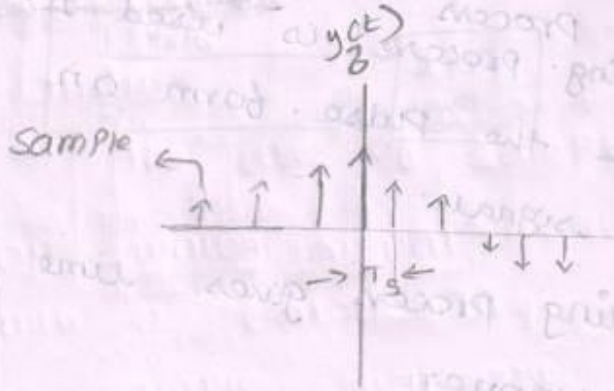
Sampling process gives time in discrete Manner.

It gives the Input is analog signal to change the signal in discrete (Digital Modulation) is called Pulse

Quantisation Process:

In Quantisation process both time and amplitute are discrete





Pulse Modulation representing the transion from Analog to digital communication pulse modulation is defind as the some Parameter of Pulse strain its vary in accordance with the message signal

There are two types of

* Pulse Modulation

* Analog Pluse Modulation
* Digital Pluse Modulation

Analog Pluse Modulation

In Analog Pluse Modulation is Periodic pluse strain is used as the carrier wave and some characteristic features of each pluse is varied in a continous manner in accordance with the corrosponding sample value of message signal

Analog Modulation → PAM

Analog Puse Modulation

In analog Puse Modulation basically information is transion in analog form but transmission takes in discreste time.

Digital Pluse Modulation

In digital Pluse Modulation the message signal is represented in a form

That is discrete in both time and amplitude thereby permitting its transion in digital form.

PCM - Pluse code Modulation

Sampling Process

Sampling process is used to describe in the time domain.

The use of sampling is convert an analog signal into a corresponding sequence of sample that are usally space uniform in time.

T_s - sampling Period

$\frac{1}{T_s}$ - sampling Rate (de

Sampling Theroem states the process of uniformly sampling a continous time signal of finite energy in a periodic spectrum with the Period equal to

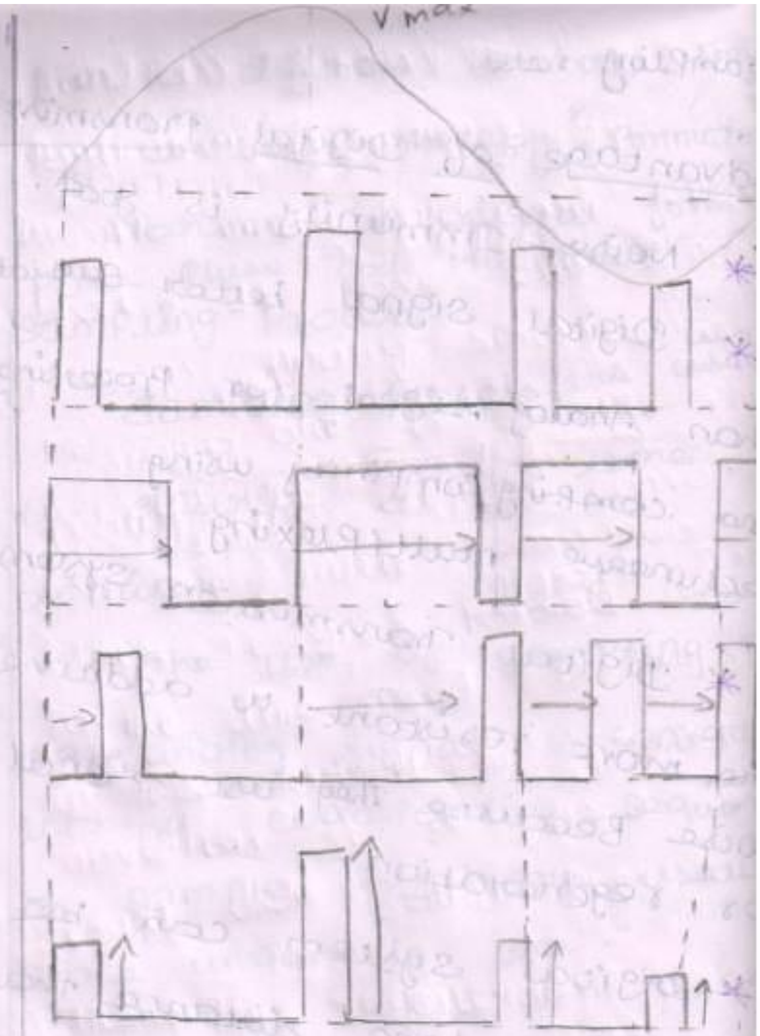
sampling rate

Advantage of Digital Transmission

* Noise Immunity is good.
* Digital signal better suited than Analog signal for Processing and combing combining using technique multiplexing
* Digital Transmission system are more resistant to additive noise Because they use signal for regeneration.
* Digital system can be Transported long distance than the Analog signal
* Digital system is simpler to measure and evaluate than analog signal.

Disadvantage of Digital Transmission

* Need More Band width
* Need of additional encoding and decoding circuits



Pluse width Modulation

Pluse width Modulation defind as the width of the constant amplitute pluse is varied propostional to the amplitute of the analog sig at the time the signal

sampled.

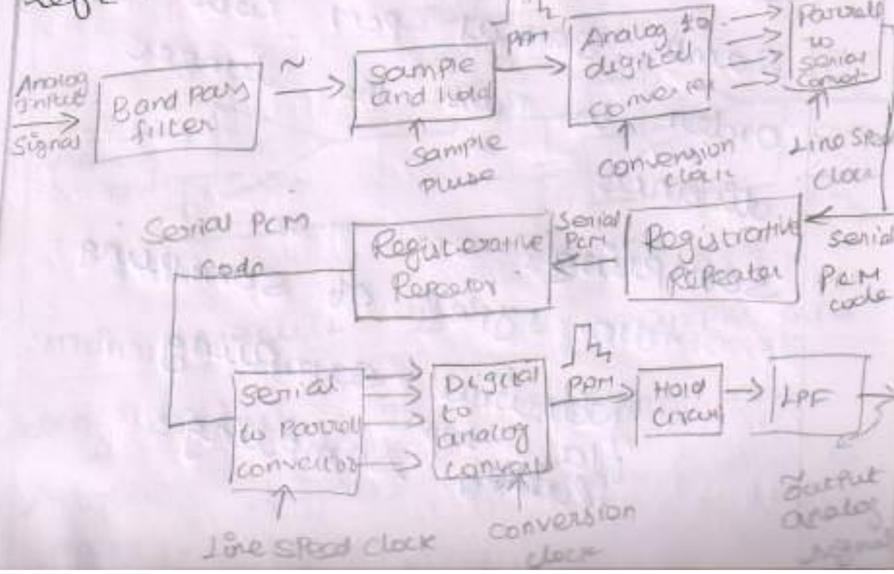
The maximum analog signal amplitude produces the width as ~~pluse~~ wider pluse

The minimum analog signal amplitute produce the width as narrow pluse

Pluse position

The highest amplitute sample produces as a pluse to the far right

lowest amplitute sample produces a pluse to the far left.



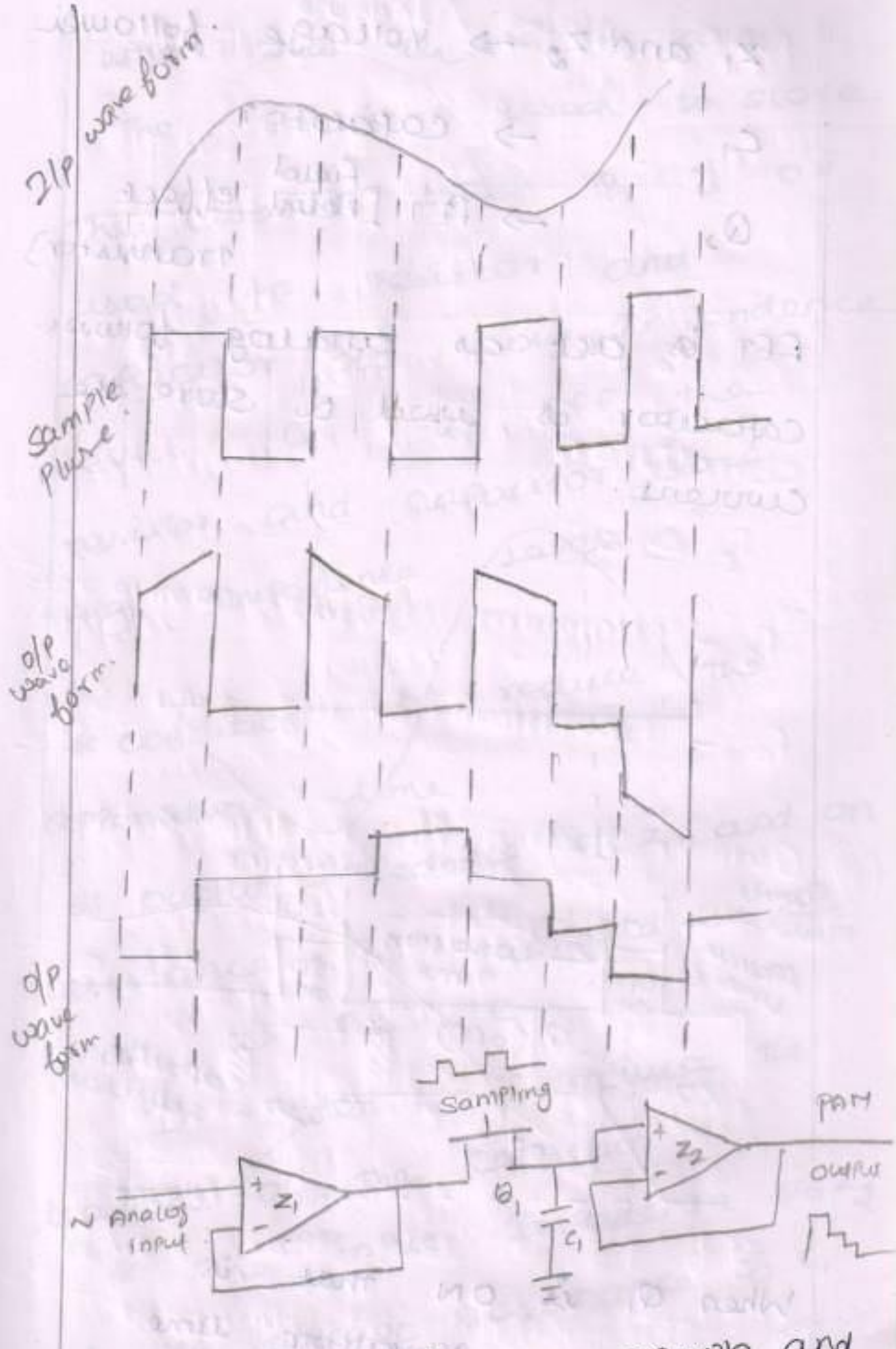
Band Pass filters are allow
Particular frequency and th
sample and hold are imp
Block. It gave two inp
one is Analog sinwave
the other one is sample
Analog and digital conver
give one input Parrell conversion
clock. It gives Parrall are
so the line speed is inpu
in the Parrell to serial
convertor it is used to
serial code. Serial PCM co

Then The regenerative rep
cir is used to repeat th
Same serial pcm code in
order to pass the longer
distance

sampling:

Two types of samplin

Natural sampling
flat top sampling



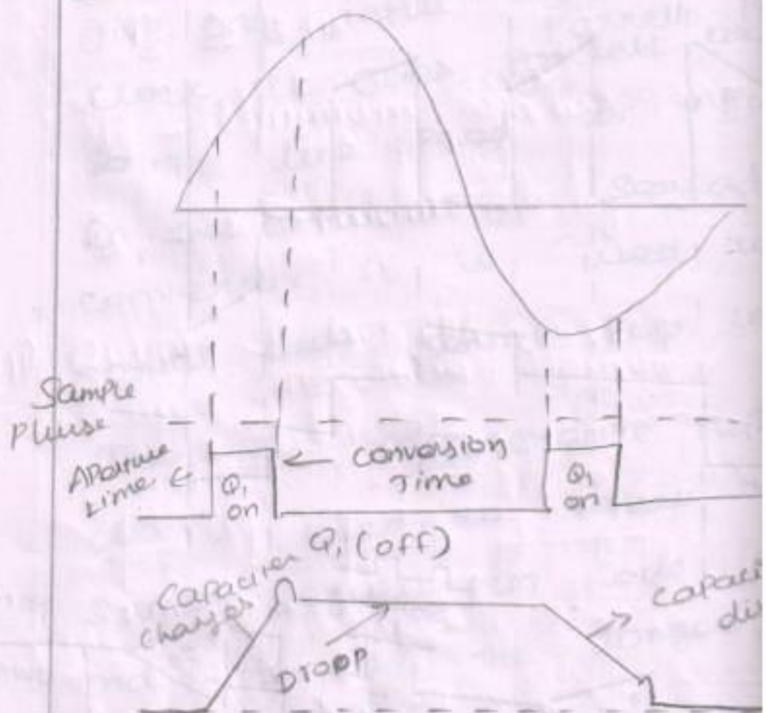
This circuit is a sample and hold circuit. The Input was Analog signal Z_4 or Output is

Z_1 and Z_2 → voltage fo

C_1 → capacitor

Q_1 → FET [Field effe
Tran

FET is act as analog
capacitor is used to store
current.



When Q_1 is ON that is
denoted as Aperature time
and when Q_1 is off that
denoted as conversion time

When the Q_1 and C_1 are . The C_1 is used to store the current when Q_1 is ON used to resistor and capacitor are low Impendence level. Q_1 is off the resistor and capacitor are High Impedence level.

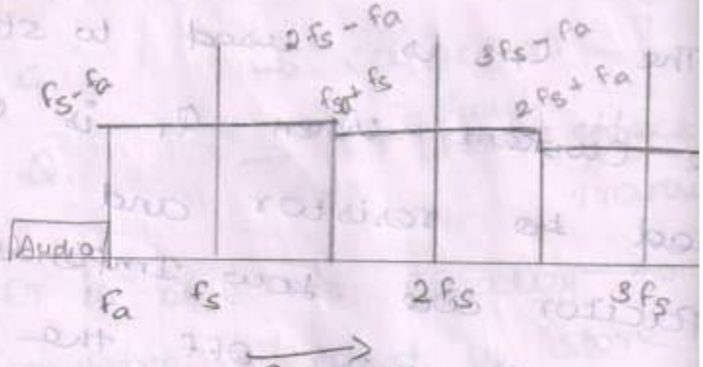
* condition to reduce the apperature time

* output Impedance of z_1 and on resistance of Q_1 should be as small as possible.

In order to reduce the leakage of the capacitor :-

* The input Impedence of z_2 and ~~the~~ leakage resistance of C_1 as high as possible.

(i) Non - Alising Alaysing



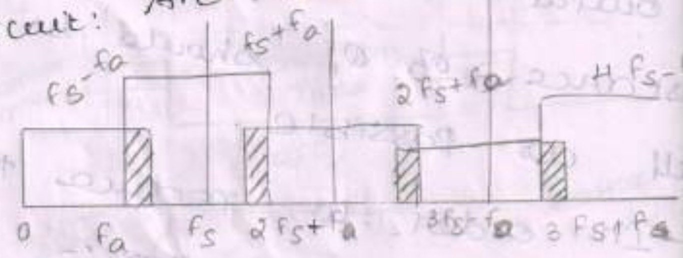
condition :

$f_s \geq 2 f_a$

f_s – Minimum Nyquist sample

f_a – Maximum analog Input

(ii) Output Spectrum of sample and circuit : Ano Alising : Alaysing :



If f's is less than 2 ti
fa . This signal get disstorted
which is called as fold over
distraction or Alising Alaysing

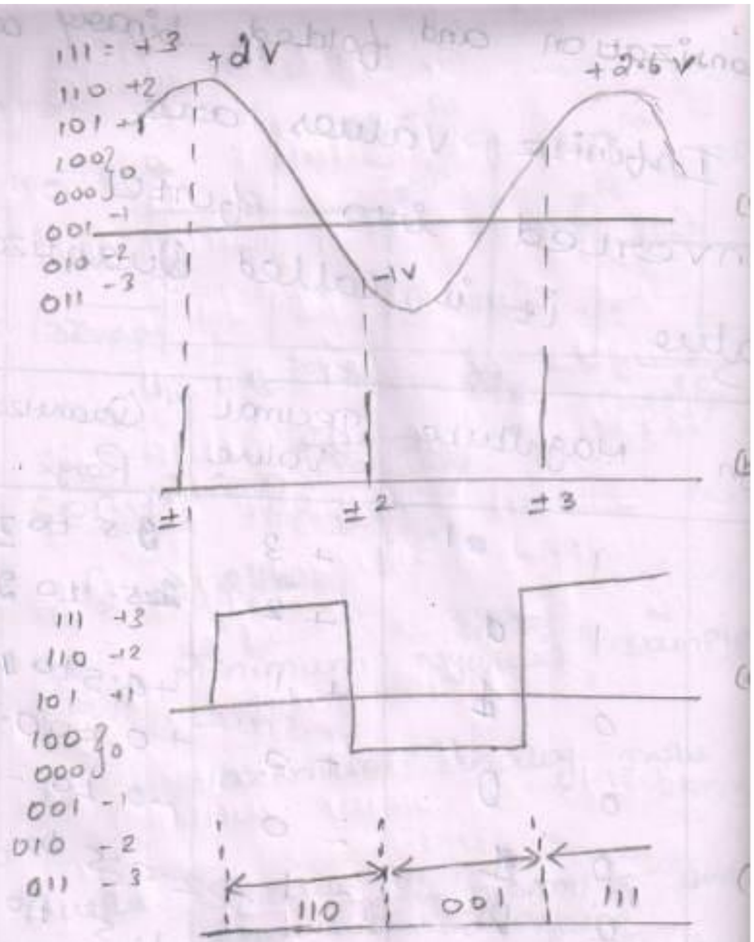
Quantization and folded binary codes

Infinite values are converted into finite value it is called Quantization.

Sign	Magnitude		Decimal Value	Quantization Range
1	1	1	+ 3	2.5 to 3.5
1	1	0	+ 2	2.5 to 2.5
1	0	1	+ 1	+0.5 to 1.5
1	0	0	+ 0	+0 to 0.5
0	0	1	− 0	−0 to −0.5
0	0	0	− 1	−0.5 to −1.5
0	1	1	− 2	−1.5 to −2.5
0	1	0	− 3	−2.5 to −3.5

This is another name is also called 3 bit magnitude or folded binary codes.

The decimal value



This graph is represented a sample and hold circuit ~~are~~ need The last diagram represent the table views

Quantization is the process of converting one infinite numb of possibilities in finite num of condition.

A magnitute difference between disadjoint is called Quantum Queue.

5 Is also ~~called~~ called resolution.

Any round off error in the tramistor in the signal are reproduced when the code is convertor back to the reciver. This error is called Quantisation error.

The folded PCM code's is sample voltage / resolution

The Quality of the PAM girls can be improved using a PCM codes with more bits, reduce the Magnitute of Quantum and improving the resolution.

Dynamic Range:

The number of PCM bit transmitter per sample can be Analyse by dynamic Range so dynamic range is defind the ratio of the largest possible Magnitute to the smallest possible Magnitute can be decoded by a DAC in the receiv

Dynamic range is denot by DR.

$$DR = \frac{V_{max}}{V_{min}} \quad (\text{unit less})$$

V_{min} → Resolution or Qu

V_{max} → Maximum V Ma

DR → Dynamic Range.

Dynamic range will be expressed in DB. (deci

Error Analysis:-

Exceptation of the rate at which error will occurs is ca Probability of error P[E]

The system error perfo is called bit Error Rate [

Error Probality Error P[E]

It is means the proba assumption of the error in the probality. It dete the error ony assumptio

Error detection:

Redundancy Checking

* Vertical redundancy c
* check some longitudinal redundancy
* cycling redundancy c
* longitudinal redunda chec

Vertical Retundancy checking:

when the input is passed in odd parity and the acutal output is also called odd parity. when any odd 1's are change, error occur. it is called verticalt Retundancy checking

odd parity - No of ~~bits~~ 1's in odd.

even parity - No of 1's in even.

Advantage:

~~No~~ ~~ex~~ mostly No error accur.

It is also known as
* Marking parity
* No parity
* Ignored parity.

Always same - Marking Pa

~~Ao~~ Always zero - Ignored
Pari

No information - No pari
send.

Check some :-

If the tramitter and
receiver has a same
bit error not occur.
If transmitter and receiv
has a different bit error
occur.

1) International standard organi
(ISO)

2) ITU-T - International Telecomm
Union - Telecommunication S

3) IEEE - Insitute of electrica
and electronics
engineering

4) ANSI - American National Standard Insitute

5) EIA - Electronic Industries Association

6) TIA - Telecommunication Industry Association

7) IAB. - Internet architecture board.

8) IETF - Internet engineering Task force

9) IRTF - Internet Research Task force

Longitudinal Redundancy checking: (LRC)

* It generates a parity to determine the error it is called message parity.

* Longitudinal LRC, a XORing of character codes which gives the message.

* VRC - Vertical redundancy ce
extraordinary or Xoring the bit
with in a single character.

* The group of character that
contain a message is called.
frame or Block of data

* The bit sequence for the
LRC is called Block check
sequence (BCS) or frame
check sequence.

\-> BCS (or) FCS

Advantage:

Multiple bit are erro
can identify.

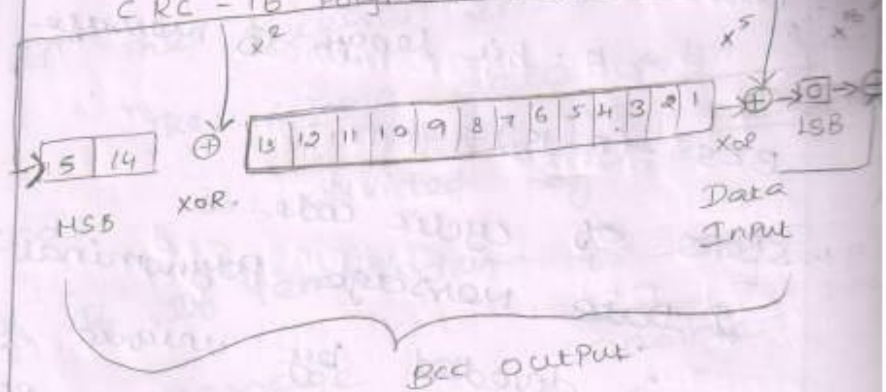
It is also passed throug
long distance

Disadvantage:

* It will not detect trans
when an even number of

character as an error in same bit position.

CRC - 16 Polynomial, $G(x) = x^{16} + x^{15} + x^{2} + \dots$



$$\frac{G(x)}{P(x)} = Q(x) + R(x)$$

Cycling Redundancy Checking:-

It can be 0 otherwise XOR condition called as convolution coding scheme. It can able to detect the 99.99%. transmission error CRC code is commonly called as CRC 16. 16 bits are used for the Block check sequence CRC is consider as systematic

Truth table to be output

0	0	0
1	1	0
1	0	1
1	1	1

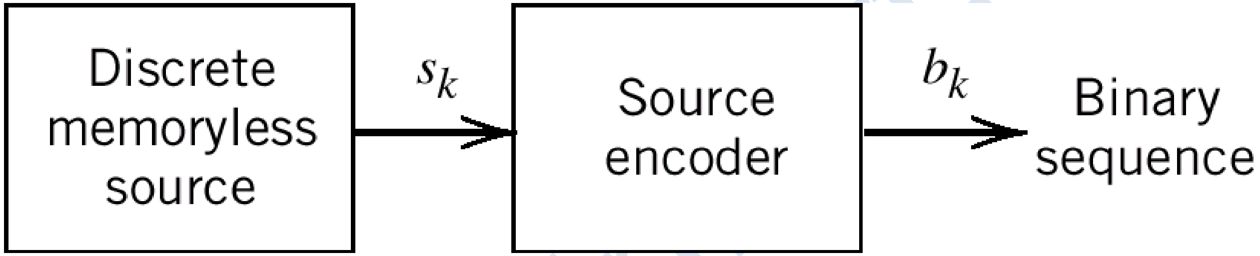
Unit-4

DIGITAL COMMUNICATION

Coding

Source Coding

Source encoding is the efficient representation of data generated by a source.

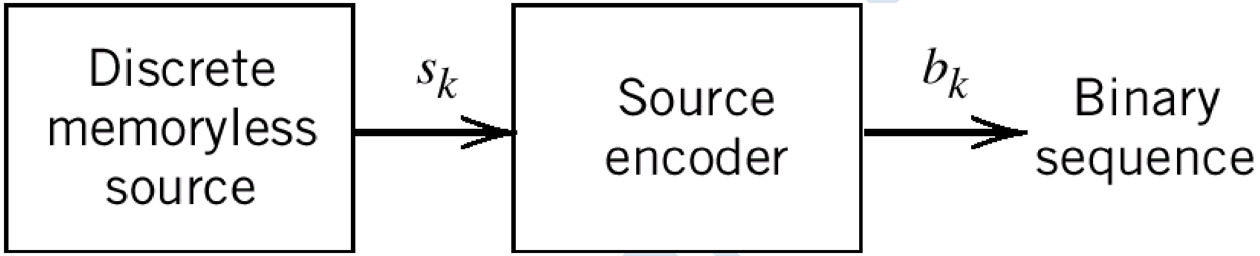


For an efficient source encoding, knowledge of the statistics of the source is required.

If some source symbols are more probable than others, we can assign short code words to frequent symbols and long code words to rare source symbols.

Source Coding

Consider a discrete source whose output of k different symbols s_k is converted by the source encoder into a block of 0s and 1s denoted by b_k



Assume that the kth symbol, s_k occurs with probability p_k , k=0,1…..K-1.
Let the binary code word assigned to symbol s_k have length l_k (in bits)

Therefore the average code-word length of the source encoder is given by

$$\overline{L} = \sum_{k=0}^{K-1} p_k l_k$$

Source Coding

Let L_{min} denotes the minimum possible value of code-word length

The Coding efficiency of the source encoder is given by

$$\eta = \frac{L_{\min}}{\overline{L}}$$

Data Compaction

Data compaction is important because signals generated contain a significant amount of redundant info and waste communication resources during transmission.

For efficient transmission, the redundant info should be removed prior to transmission.

Data compaction is achieved by assigning short description to the most frequent outcomes of the source output and longer description to the less frequent ones.

Some source-coding schemes for data compaction:-

- Prefix coding
  - The Huffman Coding
  - The Lempel-Ziv Coding

Prefix Coding

A prefix code is a code in which no code word is the prefix of any other code word

Example: Consider the three source codes described below

Source Symbol	Probability of Occurrence	Code I	Code II	Code III
s_0	0.5	0	0	0
s_1	0.25	1	10	01
s_2	0.125	00	110	011
s_3	0.125	11	111	0111

Prefix Coding

Source Symbol	Probability of Occurrence	Code I	Code II	Code III
s_0	0.5	0	0	0
s_1	0.25	1	10	01
s_2	0.125	00	110	011
s_3	0.125	11	111	0111

Is Code I a prefix code?

It is NOT a prefix code since the bit 0, the code word for s_0, is a prefix of 00, the code word for s_2 and the bit 1, the code word for s_1, is a prefix of 11, the code word for s_3.

Is Code II a prefix code?

Yes

Is Code III a prefix code?

No

A prefix code has the important property that it is always uniquely decodable

Prefix Coding - Example

Source Symbol	Code I ✓	Code II ✗	Code III ✗	Code IV ✓
s_0	0	0	0	00
s_1	10	01	01	01
s_2	110	001	011	10
s_3	1110	0010	110	110
s_4	1111	0011	111	111

Prefix code?

Huffman Coding – a type of prefix code

Basic idea : Assign to each symbol a sequence of bits roughly equal in length to the amount of information conveyed by the symbol.

Huffman encoding algorithm:

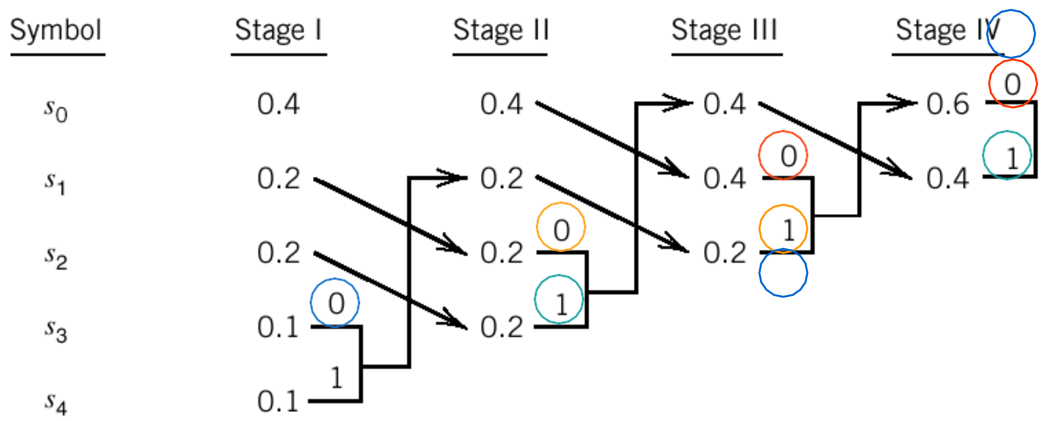
Step 1: The source symbols are listed in order of decreasing probability. The two source symbols of lowest probability are assigned a 0 and 1.

Step 2: These two source symbols are regarded as being combined into a new source symbol with probability equal to the sum of the two original probabilities. The probability of the new symbol is placed in the list in accordance with its value.

The procedure is repeated until we are left with a final list of symbols of only two for which a 0 and 1 are assigned.

The code for each source symbol is found by working backward and tracing the sequence of 0s and 1s assigned to that symbol as well as its successors.

Huffman Coding – Example



(*a*)

Symbol	Probability	Code word
s_0	0.4	00
s_1	0.2	10
s_2	0.2	11
s_3	0.1	010
s_4	0.1	011

(*b*)

Step 1: The source symbols are listed in order of decreasing probability. The two source symbols of lowest probability are assigned a 0 and 1.

Step 2: These two source symbols are regarded as being combined into a new source symbol with probability equal to the sum of the two original probabilities.

The probability of the new symbol is placed in the list in accordance with its value.

The procedure is repeated until we are left with a final list of symbols of only two for which a 0 and 1 are assigned.

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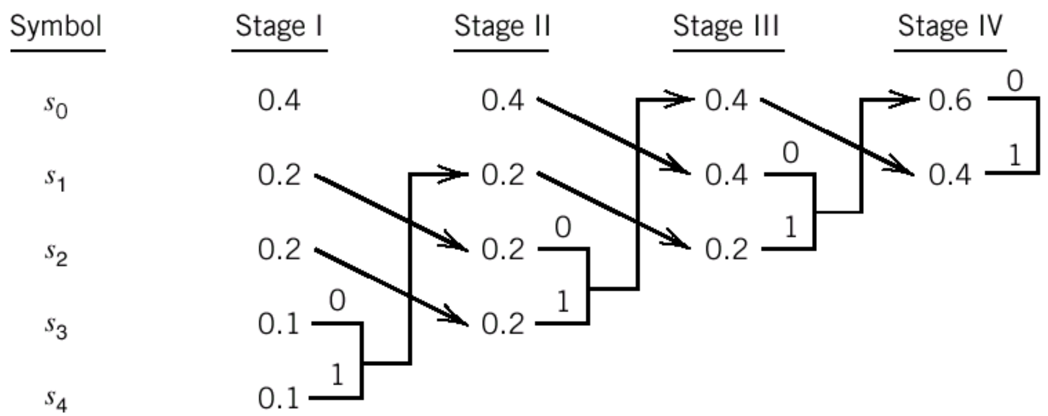
Huffman Coding – Average Code Length



(a)

Symbol	Probability	Code word
s_0	0.4	00
s_1	0.2	10
s_2	0.2	11
s_3	0.1	010
s_4	0.1	011

(b)

$$\bar{L} = \sum_{k=0}^{K-1} p_k l_k$$

= 0.4(2) + 0.2(2) + 0.2(2) + 0.1(3) + 0.1(3)
= 2.2

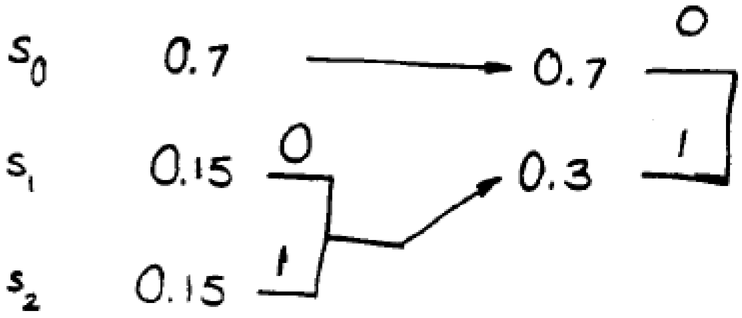
Huffman Coding – Exercise

Symbol	S_0	S_1	S_2
Probability	0.7	0.15	0.15

Compute the Huffman code.

What is the average code-word length?

Huffman Coding – Exercise



The Huffman code is therefore

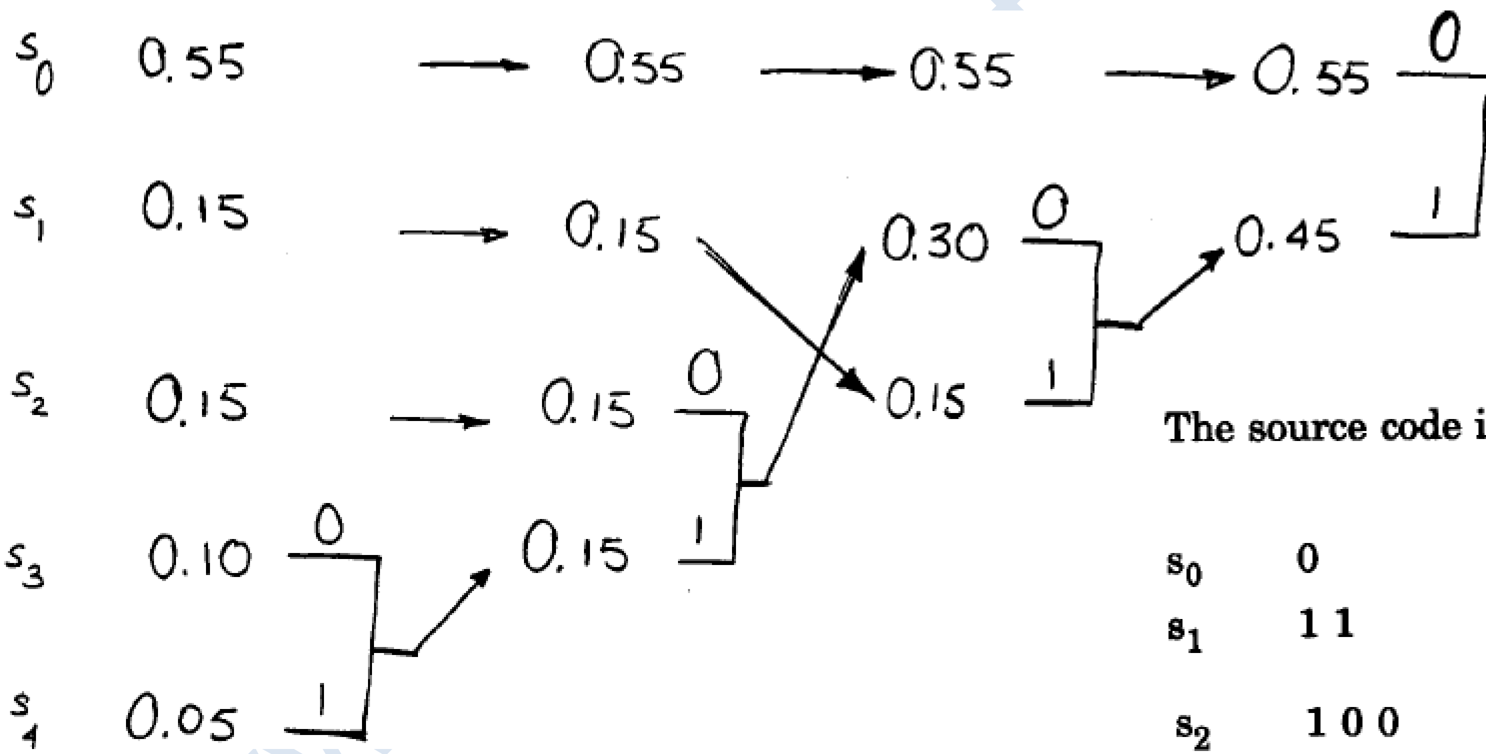
s_0	0
s_1	1 0
s_2	1 1

The average code-word length is

$$\bar{L} = 0.7(1) + 0.15(2) + 0.12(2)$$
$$= 1.3$$

Huffman Coding – Two variations

When the probability of the combined symbol is found to equal another probability in the list, we may proceed by placing the probability of the new symbol as high as possible or as low as possible.



The source code is therefore

s_0	0
s_1	1 1
s_2	1 0 0
s_3	1 0 1 0
s_4	1 0 1 1

Huffman Coding – Two variations

The average code-word length is therefore

$$\bar{L} = \sum_{k=0}^{4} p_k l_k$$

$$= 0.55(1) + 0.15(2) + 0.15(3) + 0.1(4) + 0.05(4)$$

$$= 1.9$$

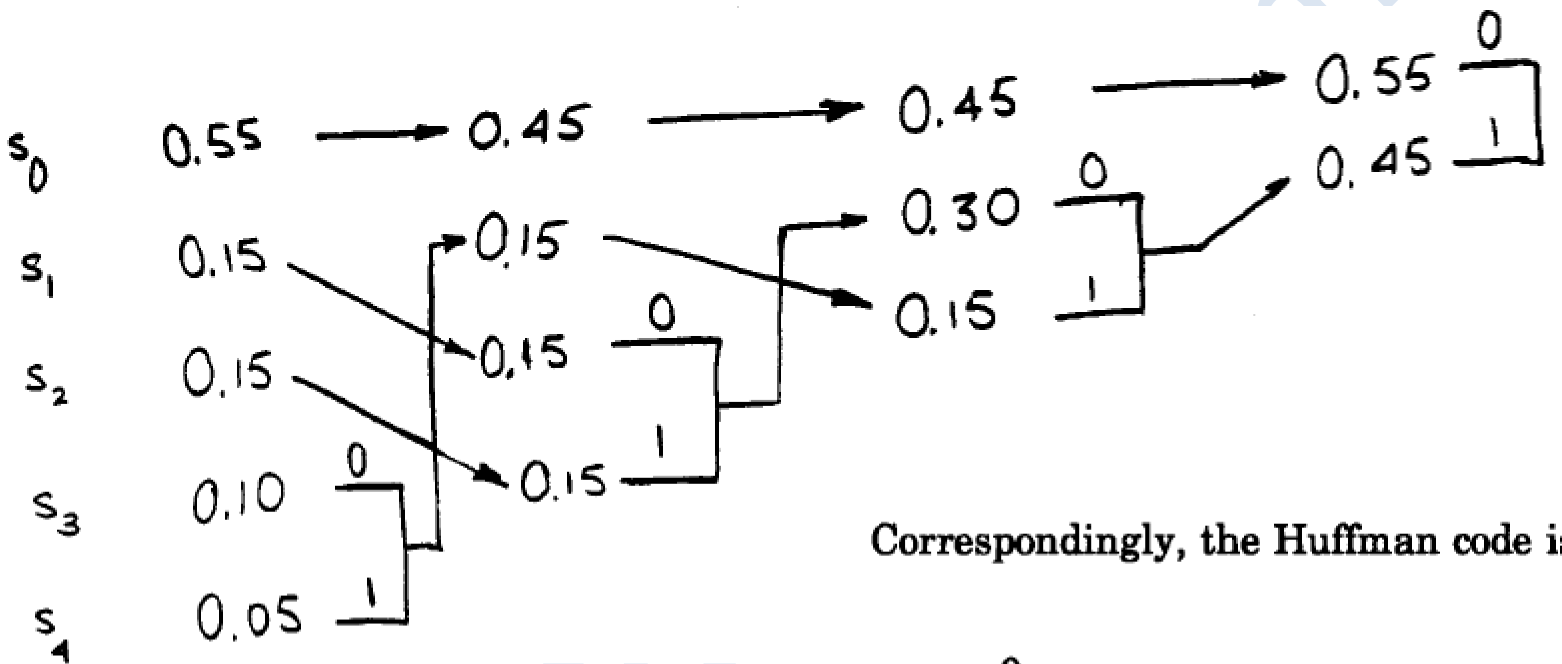
The variance of $\bar{L}$ is

$$\sigma^2 = \sum_{k=0}^{4} p_k (l_k - \bar{L})^2$$

$$= 0.55(-0.9)^2 + 0.15(0.1)^2 + 0.15(1.1)^2 + 0.1(2.1)^2 + 0.05(2.1)^2$$

$$= 1.29$$

Huffman Coding – Two variations



Correspondingly, the Huffman code is

s_0	0
s_1	1 0 0
s_2	1 0 1
s_3	1 1 0
s_4	1 1 1

Huffman Coding – Two variations

The average code-word length is

$$\bar{L} = 0.55(1) + (0.15 + 0.15 + 0.1 + 0.05)(3)$$
$$= 1.9$$

The variance of $\bar{L}$ is

$$\sigma^2 = 0.55(-0.9)^2 + (0.15 + 0.15 + 0.1 + 0.05)(1.1)^2$$
$$= 0.99$$

Which one to choose?

Huffman Coding – Exercise

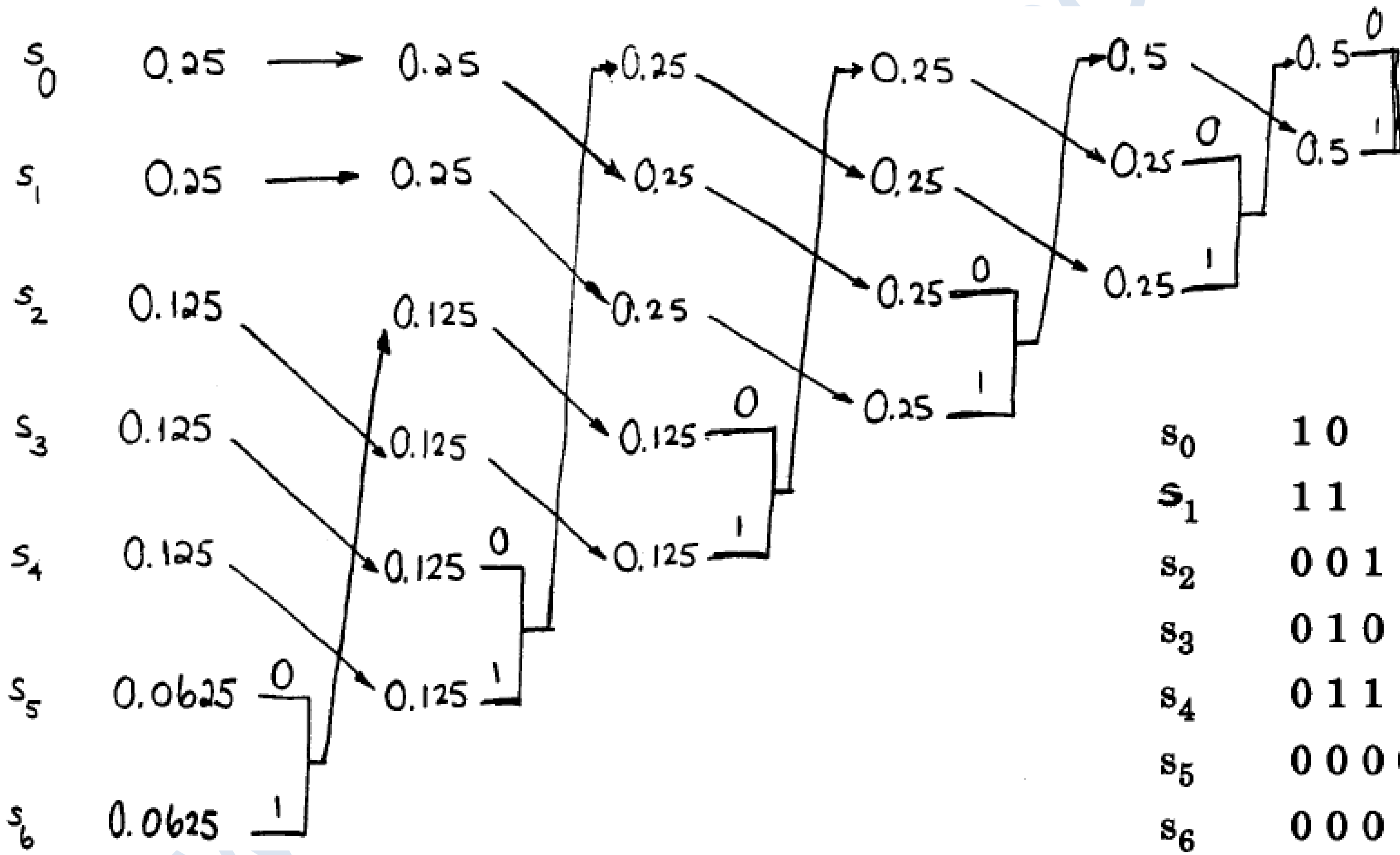
Symbol	S_0	S_1	S_2	S_3	S_4	S_5	S_6
Probability	0.25	0.25	0.125	0.125	0.125	0.0625	0.0625

Compute the Huffman code by placing the probability of the combined symbol as high as possible.

What is the average code-word length?

Huffman Coding – Exercise Answer

Symbol	S_0	S_1	S_2	S_3	S_4	S_5	S_6
Probability	0.25	0.25	0.125	0.125	0.125	0.0625	0.0625



s_0	1 0
s_1	1 1
s_2	0 0 1
s_3	0 1 0
s_4	0 1 1
s_5	0 0 0 0
s_6	0 0 0 1

Huffman Coding – Exercise

s_0	1 0
s_1	1 1
s_2	0 0 1
s_3	0 1 0
s_4	0 1 1
s_5	0 0 0 0
s_6	0 0 0 1

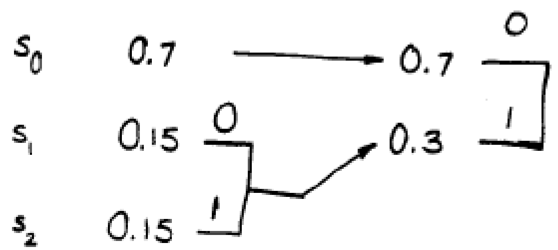
The average code-word length is

$$\bar{L} = \sum_{k=0}^{6} p_k l_k$$

$$= 0.25(2)(2) + 0.125(3)(3) + 0.0625(4)(2)$$

$$= 2.625$$

Huffman Coding – extended form



The Huffman code is therefore

s_0	0
s_1	1 0
s_2	1 1

(b) For the extended source we have

Huffman Coding – extended form

Symbol	s_0s_0	s_0s_1	s_0s_2	s_1s_0	s_2s_0	s_1s_1	s_1s_2	s_2s_1	s_2s_2
Probability	0.49	0.105	0.105	0.105	0.105	0.0225	0.0225	0.0225	0.0225

Applying the Huffman algorithm to the extended source, we obtain the following source code

s_0s_0	1
s_0s_1	0 0 1
s_0s_2	0 1 0
s_1s_0	0 1 1
s_2s_0	0 0 0 0
s_1s_1	0 0 0 1 0 0
s_1s_2	0 0 0 1 0 1
s_2s_1	0 0 0 1 1 0
s_2s_2	0 0 0 1 1 1

Lempel-Ziv Coding – a type of prefix code

Basic idea : Parse the source data stream into segments that are the shortest subsequences not encountered previously

Consider an input binary sequence 000101110010100101…

Assume that the binary symbols 0 and 1 are already stored

Subsequences stored 0,1

Data to be parsed 000101110010100101…

With symbols 0 and 1 already stored, the shortest subsequence encountered for the first time is 00, so

Th

Lempel-Ziv Coding – a type of prefix code

Lempel-Ziv Coding – Exercise

Encode the following sequence using Lempel-Ziv algorithm assuming that 0 and 1 are already stored

11101001100010110100….

Lempel-Ziv Coding – Exercise Answer

Encode the following sequence using Lempel-Ziv algorithm assuming that 0 and 1 are already stored

1110100110001011010O….

<u>Initial step</u>

Subsequences stored: 0

Data to be parsed: 1 1 1 0 1 0 0 1 1 0 0 0 1 0 1 1 0 1 0 0 ...

<u>Step 1</u>

Subsequences stored: 0, 1, 11

Data to be parsed: 1 0 1 0 0 1 1 0 0 0 1 0 1 1 0 1 0 0 ..

<u>Step 2</u>

Subsequences stored: 0, 1, 11, 10

Data to be parsed: 1 0 0 1 1 0 0 0 1 0 1 1 0 1 0 0 ....

Lempel-Ziv Coding – Exercise Answer

Encode the following sequence using Lempel-Ziv algorithm assuming that 0 and 1 are already stored

11101001100010110100….

Step 3

Subsequences stored: 0, 1, 11, 10, 100

Data to be parsed: 1 1 0 0 0 1 0 1 1 0 1 0 0 ...

Step 4

Subsequences stored: 0, 1, 11, 10, 100, 110

Data to be parsed: 0 0 1 0 1 1 0 1 0 0 ...

Step 5

Subhsequences stored: 0, 1, 11, 10, 100, 110, 00

Data to be parsed: 1 0 1 1 0 1 0 0 ....

Lempel-Ziv Coding – Exercise Answer

Encode the following sequence using Lempel-Ziv algorithm assuming that 0 and 1 are already stored

11101001100010110100….

Step 6

Subsequences stored: 0, 1, 11, 10, 100, 110, 00, 101

Data to be parsed: 1 0 1 0 0 ...

Step 7

Subsequences stored: 0, 1, 11, 10, 100, 110, 00, 101, 1010

Data to be parsed: 0 ....

Lempel-Ziv Coding – Exercise Answer

Encode the following sequence using Lempel-Ziv algorithm assuming that 0 and 1 are already stored

11101001100010110100….

Numerical positions	1	2	3	4	5	6	7	8	9
Subsequences	0,	1,	11,	10,	100,	110,	00,	101,	1010
Numerical representations			22,	21,	41,	31,	11,	42,	81

Wireless Communication Systems
unit-5

Radio Communication

- *Radio* or *radio communication* means any transmission, emission, or reception of signs, signals, writing, images, sounds or intelligence of any nature by means of electromagnetic waves of frequencies lower than three thousand gigacycles per second (3000 GHz) propagated in space without artificial guide.
- Examples of radio communication systems:
  - Radio broadcasting.
  - TV broadcasting.
  - Satellite communication.
  - Mobile Cellular Telephony.
  - Wireless LAN.
  - Multimedia communication & Mobile Internet

History

- 1864: Maxwell describes radio wave mathematically
- 1888: Hertz generates radio waves
- 1890: Detection of radio waves
- 1896: Marconi makes the first radio transmission
- 1915: Radio tubes are invented
- 1948: Shannon’s law
- 1948: Transistor
- 1960: Communication Satellites
- 1981: Cellular technology

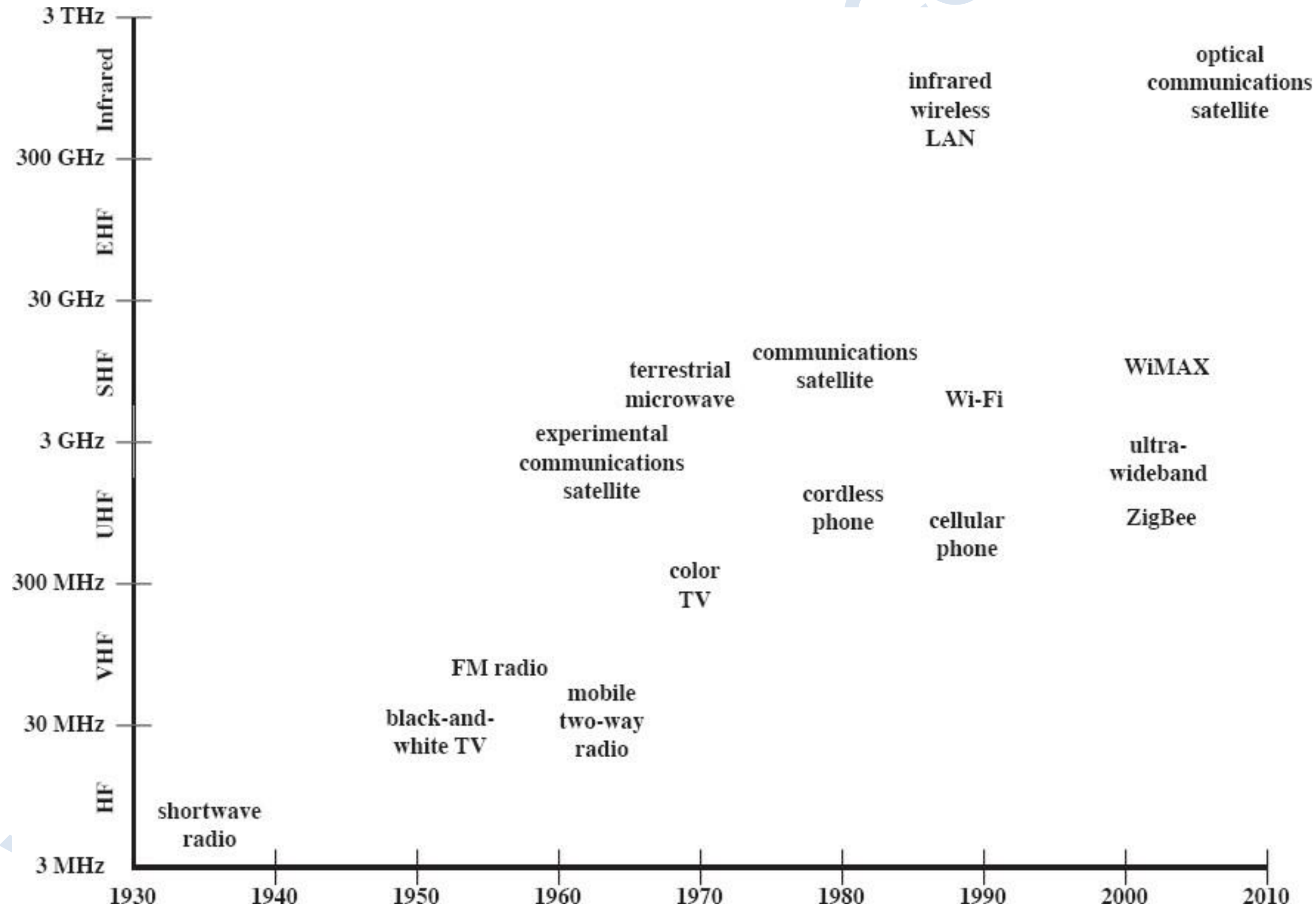
Classification of radio spectrum

Term	Wavelength	Frequency	Application
ELF	1000 -100 km	300-3000 Hz	Time and Frequency Normals, Navigation, Underwater Communication, Remote sensing under ground, Maritme telegraphy
VLF	100 -10 km	3-30 kHz	Time and Frequency Normals, Navigation, Underwater Communication, Remote sensing under ground, Maritme telegraphy
LF	10 -1 km	30-300 kHz	Long distance communication (fixed and marite), Broadcasting, Naviagation, Radio beacons
MF	1000 -100 m	300-3000 KHz	AM broadcasting, naviation, radio beacons, distress frequencies.
HF	100 -10 m	3-30 MHz	Fixed point to point communication, Mobile maritime aeronautical, land services, military communication, amateur radio and broadcasting
VHF	10 -1 m	30-300 MHz	Broadcasting, TV, FM, Mobile services for maritime, aeronautical and land, Wireless microphones, Meteor burst communicaiton
UHF	100 -10 cm	300-3000 MHz	Broadcasting TV, satelites, Personal telephone systems, radar systems, fixed and mobile satelite services
SHF	10 -1 cm	3-30 GHz	Fixed services, Fixed statelite services, Mobile serivces, Remote sensing
EHF	10 -1 mm	30-300 GHz	Frequency assaignments up 60 GHz

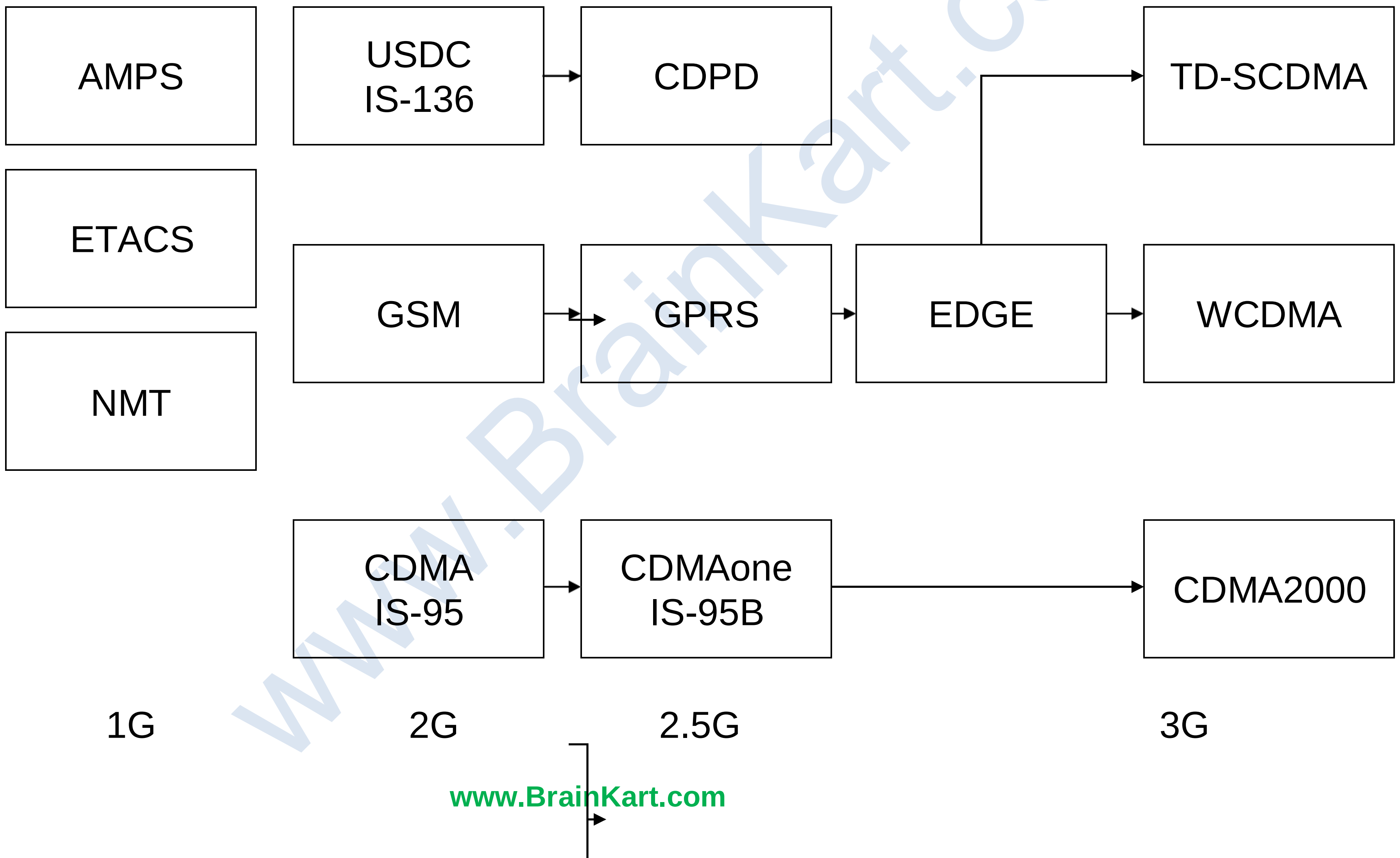
The Radio Spectrum

- The frequency spectrum is a shared resource.
- Radio propagation does not recognize geopolitical boundaries.
- International cooperation and regulations are required for an efficient use of the radio spectrum.
- The International Telecommunication Union (ITU) is an agency, within the UN, that takes care of this resource.
  - Frequency assignment.
  - Standardization.
  - Coordination and planning of the international telecommunication services.

Evolution of Wireless Systems



Evolution of Cellular Systems



LTE-Long Term Evolution

- High spectral efficiency
- Very low latency
- Support of variable bandwidth
- Simple protocol architecture
- Simple Architecture
- Compatibility and inter-working with earlier 3GPP Releases
- Inter-working with other systems, e.g. cdma2000
- FDD and TDD within a single radio access technology

Other Technologies

- WLAN
- Bluetooth
- Sensor networks (Zigbee and IEEE 802.15.4)

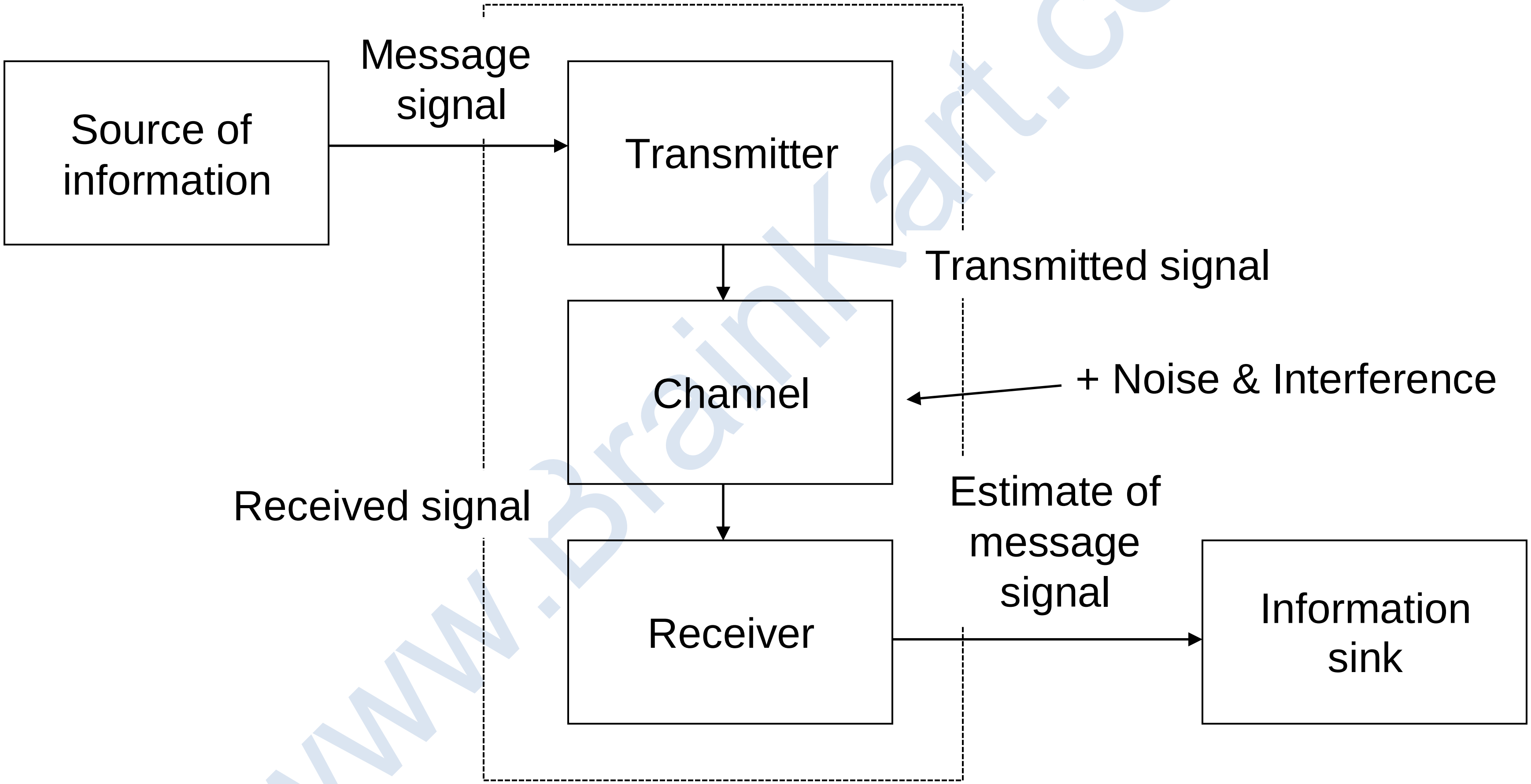
Radio Communication

- Three main problems:
  - The path loss
  - Noise
  - Sharing the radio spectrum

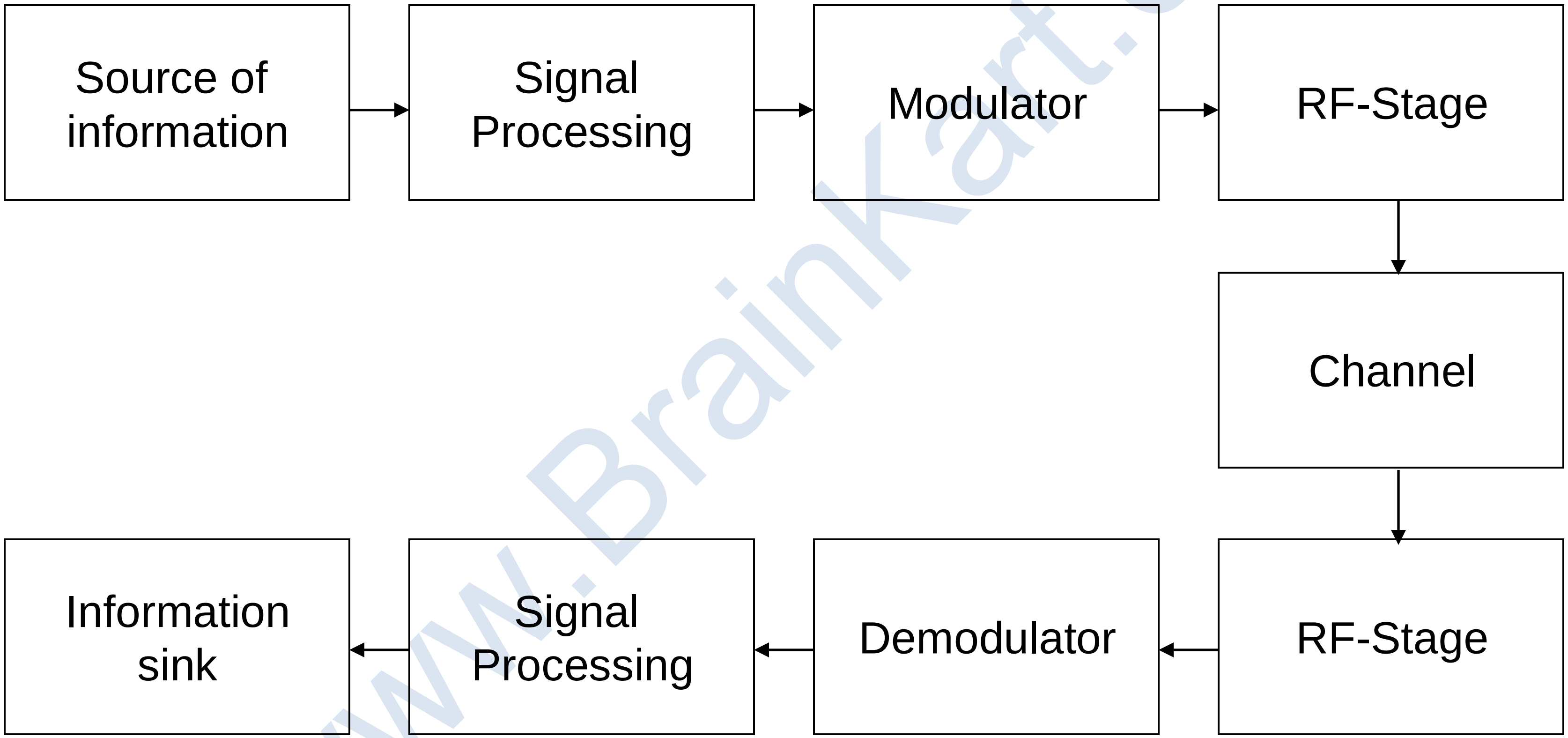
Challenges Today

- The Revenue Gap
  - Flat rate models for wireless broadband increase the demand for bandwidth but do not increase revenue.
  - Cost is roughly proportional to bandwidth
- Energy Consumption
  - Energy consumption of the ICT industry is roughly 2%
  - Communication is increasing rapidly
  - Energy cost is also increasing

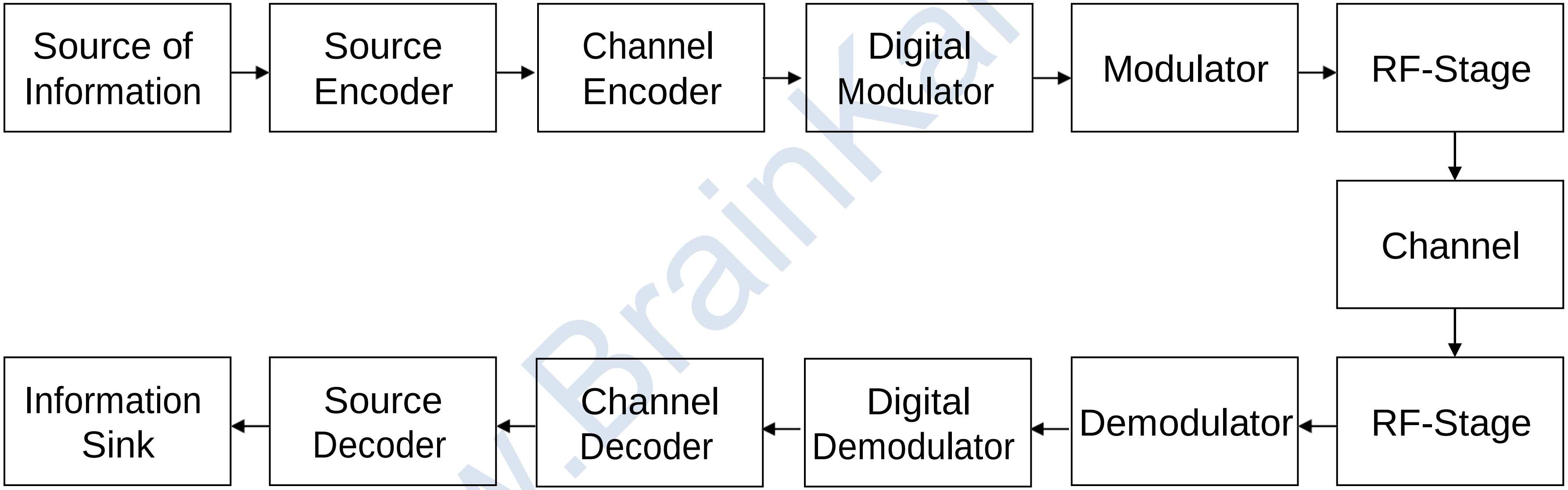
Communication Systems



Analog Communication System



Digital Communication System



decibels

- The *bel* is a logarithmic unit of power ratios. One *bel* corresponds to an increase of power by a factor of 10 relative to some reference power, P_{ref}.

$$P_{[bel]} = \log_{10}\left(\frac{P}{P_{ref}}\right)$$

- The bel is a large unit, so that decibel (dB) is almost always used:

$$P_{[dB]} = 10\log_{10}\left(\frac{P}{P_{ref}}\right)$$

- The above equation may also be used to express a ratio of voltages (or field strengths) provided that they appear across the same impedance (or in a medium with the same wave impedance):

$$V_{[dB]} = 20\log_{10}\left(\frac{V}{V_{ref}}\right)$$

[Saunders, 1999]

decibels

Unit	Reference Power	Application
dBW	1 W	Absolute power
dBm	1 mW	Absolute power *P* [dbW] = *P* [dBm] - 30
dBμV	1 μV	Absolute voltage, typically at the input terminals of a receiver
dB	any	Gain or loss of a network
dBμV/m	1 μV/m	Electric field strength
dBi	Power radiated by and isotropic reference antenna	Gain of an antenna
dBd	Power radiated by a half-wave dipole	Gain of an antenna 0 dBd = 2.15 dBi

CDMA

Frequency reuse
Large coverage
High spectrum capacity
High Privacy
Soft Handoff
Good Voice quality (using Voice Coding)
Perfect Power Control
Smooth migration to 3G