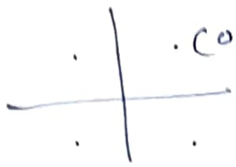
Rs Symbol rate = Baud rate. S/s → one symbol reflect one bit = Bit rate

Bit rate = Rb bPS

in some commun system. multiple Bits combined to form a single symbol. which result to higher. symbol rate. for given bit rate. Like in QAM. Like multiple bits can be encoded in a single symbol.



·(0,0,1) ← bits encoded to form one symbol.

So Symbol rate will be higher than bit rate Rb.

given as $Rb = Rs \times \log_2 M$

bit rate = Baud symbol rate $\times$

M is Possible no. of symbol

LTI - same signal shifted in time.

transmitter - objective is to transform the I/P signal. in a signal to adapted to transmitted over Physical or wireless medium.

Adaptation is Performed by modulation.

also include - channel encoding..

Receiver :- task to guess 'a' random variable - from y. The decision made by Receiver is denoted by â.

Receiver is also designed to. minimize the. Probability of errors.

SNR / BER - Signal to noise ratio. affects BER. Ratio of signal Power to noise Power.

high SNR means signal is strong than Power noise.

- BER - measure to accuracy of dc system. defined as no of bits are received in error. divided by no of bits transmitted.

Lower BER means highest. accuracy of system.

So Relation is as SNR increases the. BER decreases.

* increase the Bandwidth is more efficient way to increase the channel capacity. Or to reduce the transmitt Power.

* BW refers to the range of frequencies occupied by a signal. The BW of signal determines the. amount of information can be transmitted over Channel.

in digital BW is defined as the rate at which data is tx expressed as bits Per s

2nd chapter

* Energy signal - finite energy 0 power

Power signal - finite average power but infinite energy.

②

Jeddah

* For DC design, channel includes physical medium, antenna, RF front end (amplifiers, filters, up & down converters), D/A & A/D).

Guided → fiber, cable.

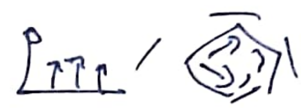
free propagation → mobile satellite channels.

* telephone channel - is bandlimited channel. attenuates the frequency above 3 KHz., It is dispersive channel - dispersed into another phase as i. colloid.

Channel is frequency selective - channel attenuates in different way & different frequency.

Equalization is needed. when there is dispersion.

multipath propagation make channel dispersion



* undesired component in signal - noise & distortion.

$$y(t) \Rightarrow s(t) + d(t) + n(t)$$

signal, distortion, noise.

Noise is independent from the signal.

distortion - is directly related to the signal is the result of a transformation T. of the signal performed by channel / system.

* distortion can be reduced (ideally to zero).

* the noise can be reduced only up to a certain limits.

If signal is attenuated & shifted in time it is distortion less form..

* To achieve distortionless transmission, the overall system response must have a constant magnitude response & its phase shift must be linear with frequency.

In other words. ① The channel must attenuate or amplify all frequency components equally

②. All the signal's components must arrive with identical time delay in order to add up coherently.

In Practical. in signal. will be distorted by passing through channel.

This means amplitude will be not just scaled & phase will not be linear

Equalizers. will be used to perform phase & amplitude counter. this distortion.

Classification of Noise.

Signal interference - crosstalk - multiple access interference,

External sources of noise - Atmospheric noise, cosmic noise, Artificial noise.

Internal source of noise - Thermal noise - shot noise - Flicker noise

Thermal noise:- from thermal agitation of electrons.

Shot noise: from grainy structure of the electric current.

Flicker noise:- from the structural imperfection of cathodes & circuits.

* thermal noise is modelled as Random Process :-

if a signal is similar to other shape then we use correlation to measure it. If cross correlation is high then signal is very similar to each other. & signal is easily detected.

Auto correlation: How much a signal is similar to itself. means we make integral & vary the τ of time shift when we get a peak so then the Rx is sync with tx.

$\boxed{X(F)^2}$ = energy spectral density of signal. If I take Fourier transform then we get Power spectral density of signal. how much the energy of signal is distributed in the frequency domain this is important because of BW of signal is related to the range of frequency in which most of energy is situated.

* one signal energy overlap with other called as interference & noise.

$$s(t) \rightarrow \otimes \rightarrow s(t) + i(t)/n(t)$$
$\uparrow$ $i(t)$ / $n(t)$

interference: independent from signal.
noise

I cannot avoid thermal noise. ⇒ when I measure the voltage at the end of resistor. The agitation of electronics. & random behavior of electron.

Random Process:- we need. Random process to deal with thermal noise.

$x(t) \rightarrow \boxed{G(f)} \rightarrow y(t)$ ~~at at~~ LTI give same output.

LTI. (LTI does not change frequency content).

If we sample a process at time x we get a random variable which is one. distributed PDF

We sample another variable we'll get another random variable.

(4)

Go back to noise. Model noise. Independent from any signal.

Thermal noise affects all frequency same.

The thermal noise is white.

PSD of random process  $N_0/2$.

in Radio comm we use electromagnetic waves. AM & FM.

Transmission - System.

Analog & digital baseband & Passband.

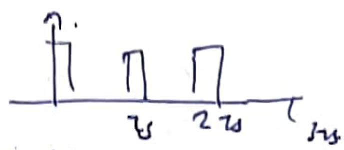
baseband → energy around 0.

Passband → energy at high frequency.

msg signal modulate in terms of pulses

→ modulate in some way. carrier is much high frequency.

Passband → QAM QPSK -

Baseband → modulated in terms of Pulse → 

changing amplitude - PAM.

PWM & PPM →

Passband → AM / coherent ; non coherent ← strictly related to Passband.

At Rx we don't have to do demodulation so is hard to know the ϕ.

if change in change freq of Tx & Rx which is called as doppler effect.

Transmitter :- The transmitter do mapping of from the message set A to the signal set S

adaptation: many : If signal is digital we need to transform it in "continuous time" waveform

Both in case of digital or analog :- The waveform must be compatible with the characteristic of the channel.

converting digital signal to continuous time signal through PAM PWM PPM. (Pulse shaping)

Filtering quantization & encoding.

Bandpass modulation :- is the process by which the information signal is converted to a sinusoidal waveform. (5)

combination of Amplitude, Phase, frequency is varied according to information to be transmitted

$$S(t) = A\cos(\omega(t)t + \phi)$$ message signal.

↑ radian frequency ↑ Phase.

modulated signal = $s(t) = $. $m(t), S(t)$.

(transmission should be like that it should be demodulated.).

Bandpass demodulation :- Coherent when the receiver exploits the knowledge of carrier phase to detect signal this is called coherent detection

There is availability at receiver a prototype of each possible arriving signal.

This prototype waveform try to duplicate the transmitted signal set in every respect.

even RF Phase.

Then receiver said to be Phase Locked to incoming signal.

Non coherent detection the receiver is simpler as it doesn't need to perform.

Phase estimation but the price paid is an increased probability of error

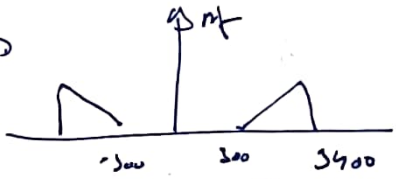
Demodulation means waveform recovery.

Detection the process of symbol decision.

The PLL is used in coherent to synchronized the phase of the local carrier signal with the phase of the incoming signal. because the phase may have affected by the channel distortions of noise so by using this we can compensate.

AM modulation

modulating signal →

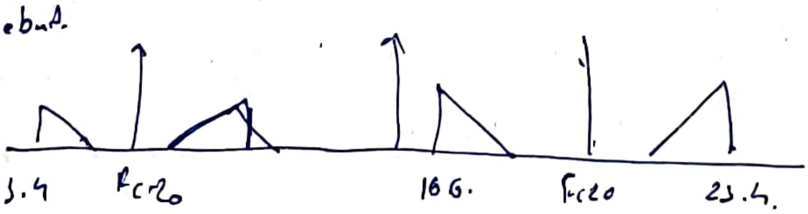


frequency Hz

baseband

BW = W.

Double sideband
DSB



BW = 2W.

AM is wasteful of BW

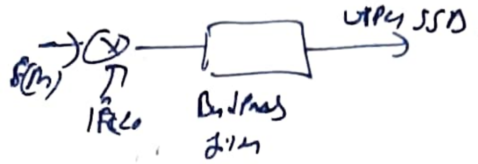
Simplest form of modulation. $v(t) = s(t)\cos(\omega_c t + \phi_0)$.

AM → It does not contain the carrier. also the amplitude modulation, it is significantly linear.

DSB-SC – linear modulation. – 2 BW wastage of BW

SSB → –||–

USB → –||–

SSB → 

6

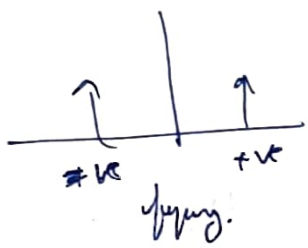
Very efficient.

Complex representation.

At lower frequency the filtering is easy than at higher frequency.

$x(t) = 2\cos(\omega_c t)$

$= e^{j\omega_c t} + e^{-j\omega_c t}$ Euler's formula.



+ve freq later   −ve frequency component   s(t).

Real signal – ve frequency component is redundant. & we ignore it

Analytic signal is complex signal containing both magnitude & phase information of real signal.

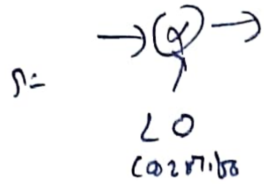
Complex envelope ~~~~ of a signal is a representation of the signal. that contains only the amplitude information. or magnitude of signal.

quadrature component of v is called hilbert transform of v.

It is convenient to translate the positive frequency mode around the origin so to achieve a base band signal which is the representation of the original bandpass signal & it is called as complex envelope.

In phase & quadrature are 2 ~~~~ orthogonal components that are used to represent a real signal. In phase – corresponds to the cosine component of the real signal. & carries information about the magnitude & phase of positive frequency. while the quadrature component corresponds to the sine component of the real signal & carries the information about the magnitude & phase of negative frequencies

* Super heterodyne Rx :-



sk + cos wt

To remove the multiplied frequency if i want to remove we use Low Pass filter.

Superheterodyne Rx makes this decoding process slow as we need it. at Lower frequency we can do filtering easy than that at higher frequency.

over sampling reduces the effect of distortion on the received signal.

* Pulse band transmission :- The sequence of digit is called baseband signal. For transmission of binary bits we need to convert them in waveform compatible to channel.

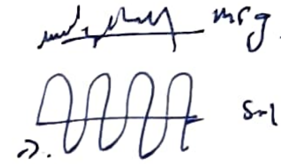
For baseband compatible waveform are pulses.

↓

Pulse modulation

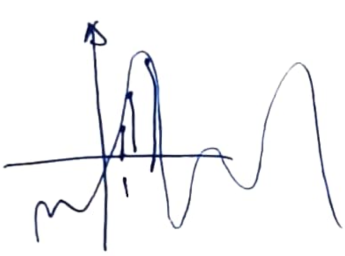
↓

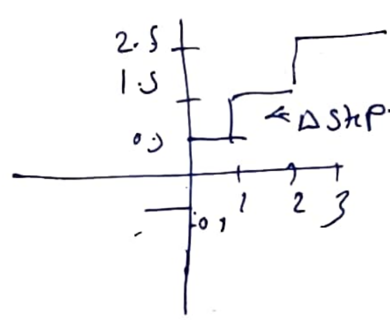
In continuous wave modulation the sine wave is varied accordingly to msg signal



In pulse modulation the parameters of pulses are varied according to the msg signal.

PAM PWM PPM → when the sample are not quantised we have analog pulse modulation. when they are quantised we have digital pulse modulation.





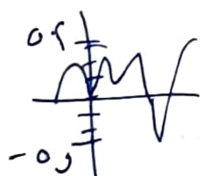
Δ step size are uniform. - 1

For quantizer suppose I have the Amplitude value of sampled signal

↓

Quantizer

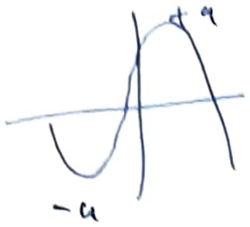
↓ at output of quantizer I have finite no of level as 0.5 - 1.5, 2.5.

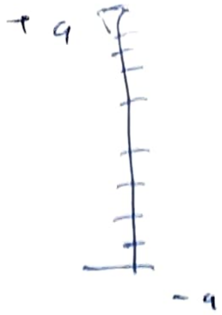


I will increase the step size.

Automatic gain controller to increase the amplitude if I have uniform quantizer. & receive the Low energy signal from -0.5 to +0.5

8

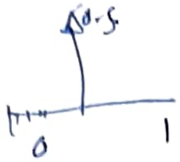




$\Delta = \frac{2a}{N}$

Quantization error will be between.

$\left(-\frac{\Delta}{2}, \frac{\Delta}{2}\right)$



This error can be modelled as uniform random variable $e \in U\left(-\frac{\Delta}{2}, \frac{\Delta}{2}\right)$

Higher Δ higher the error.

delta Δ defines ds.

N no of levels in my quantization.

Δ =

If delta is low my quantization error will be low.

increase N to decrease Δ SQNR will increase.

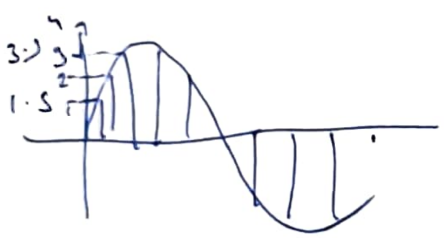
if choose N big. So I will have to ~~tx~~ more bits. So more BW

This $N = 2^{\nu}$ → is different the $\log_2 M$.

10000000.

if I use QPSK.

00 we use 4 different ...
01
10 ← this is different from quantization.
11



Natural	1.3	3.6	2.3
Quantized	1.5	3.5	2.5

No modulation only. assign ~~bits~~ code to each value.

PCM assign bits of data to each value. 001 111 0.1

QPSK each ~~...~~ symbol carry 2 bits

*modulation comes after quantization.

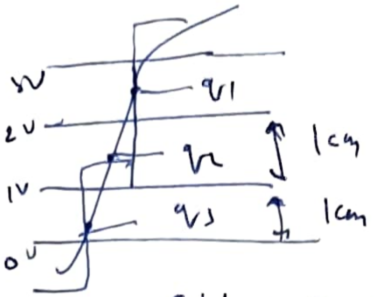
Non uniform Quantization :-

(5)

For a high varying sig.

Bigger amplitude will have bigger Δ. ~~further width~~

low signal will have smaller Δ



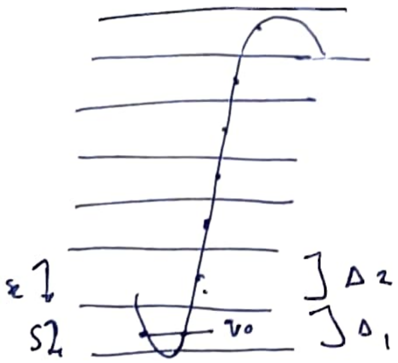
Step size (s) Δ.

Quantization level q1, q2, q3

$$\text{Step size} = \frac{V_H - V_L}{Q} = \frac{3-0}{3} = \underline{1 \text{ step size}}$$

(q1 q2 q3)

decide a step size



all values in Δ_1 region are matched to q_0.

5.19 → 5.2 = Error quantization error.

decrease the step size. that means more interval.

But q will increase

if $Q = 8$. so q_0 to q_7 $Q = 2^v$.   So Q will need 3 bits.

if Q increase then my bit rate & BW will increase.

SQNR → if needed high. increase NQ. also BW width will increase.

Quantization

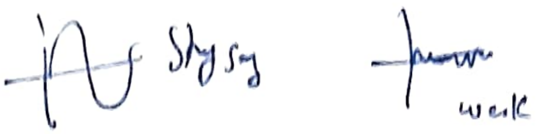
- uniform: step size constant
- non uniform: step size is not constant,
  ↳ depends on amplitude.

Original signal & Quantized signal error.

Noise Power in uniform $= \frac{\delta^2}{12}$

In weak signal, SNR is weak.

For strong the SNR is strong & for weak it is weak so we move to non uniform.



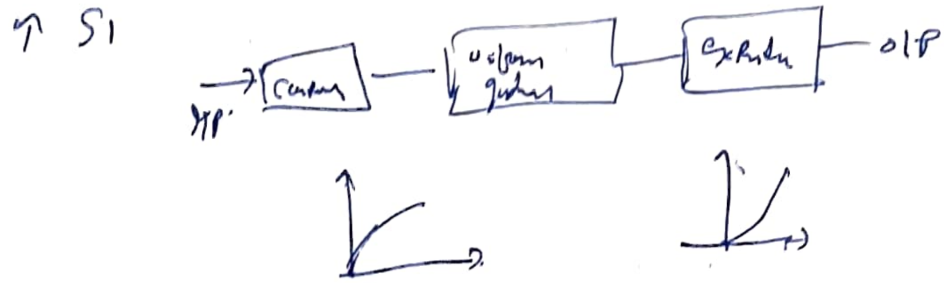
$\frac{S_1}{N_q} \uparrow$ $\qquad$ $\frac{S_1}{N_q} \downarrow$

As we saw the step size is variable in non uniform.

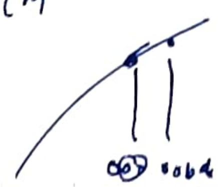
Companding.

At Tx.
weak signal are amplified
strong signal are attenuated

At Rx
weak signal are attenuated
strong signal are amplified.



DPCM



more redundant bits are their & same information is transmitted again.

So highly correlated samples are encoded in PCM.

So when a sampled signal $\quad 0 \cdot 1 \; 3 \quad 6 \; 10 \quad$ amplitude PCM

difference $1-0$, $3-1 = 2$, $6-3$, $10-6$.

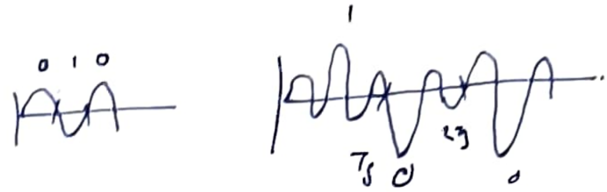
Case 1
$VH - VL = 0 - 10 = 10$

Case 2 $VH - VL$. $1 - 4 = 5$ $\qquad$ $0-10$ to $1-4$. difference of the signal.

So $VH - VL$ decrease. So step size & quantizer are 1.

Delta modulation we are not sending the whole information of signal.
we are just sending whether the ~~signal is high or lower~~
Present sample is higher than or lower than the approximated value.

ISI.

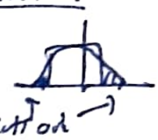


000000000

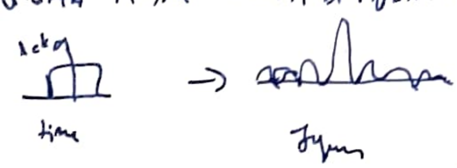
000 → one signal → one waveform
001 → one symbol. → one waveform.

The energy of symbol is interface with each other so I have ISI.
&
Even I don't have overlap. but they will overlap in Rx. →.

Because my channel is Band Limited some frequency are passed & others are attenuated.



Limitation in time → will be infinite. in frequency



energy of my pulse is spread.

So if I pass to band limited  just pick up a range of frequency. in

So if I have something limited in frequency then I have will have something infinite in time. Vice versa. So energy is spread over long period in time.

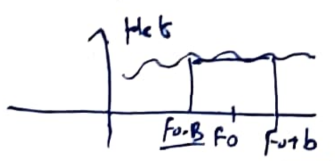
If I sample this still I will get the replica of frequency transmitted signal.

So the symbol previously tx will have effect.

Bandwidth of channel control this

Rs symbol rate baud rate.

So Band channel is frequency selective..

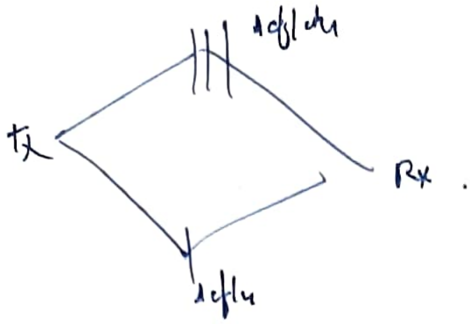


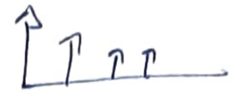
delay & attenuation of frame. it is distortionless channel.

Some frequency are passed & other are attenuated.

Selectivity in frequency domain is ISI in time domain.
If I have the ISI i will say the channel is frequency selective.

(12)





I will receive the reflection of the signal. ~~we~~ & energy of signal is spread on longer Period

& this is called delay spread.

The Fourier transform of $\uparrow$ $\perp$ flat.

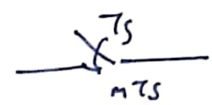
This is not flat because we have no of delays

we will get frequency selective.

<u>Condition to avoid ISI</u> :-

To remove ISI then to do equalization.

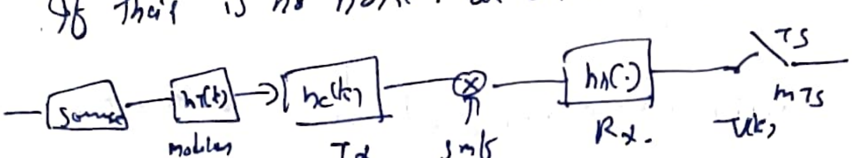
design Tx & Rx to avoid ISI



Each symbol is tx by sample Period Ts so sample at Ts Period

noise is not dual at Rx. after filter.

If their is no noise. we will do the foll. to Avoid ISI.



$S_0(t) = x_1(t) * \left(h_T(t) * h_c(t) * h_{R}(t)\right)$ $\leftarrow$ PAM sym

$P(t)$ $\leftarrow$ filter

So overall system is given by

Signal in input $x_s(t)$ to filter $\boxed{P(t)}$ $\leftarrow$ shape is $P(t)$.

$$v(t) = \sum_{m=-\infty}^{+\infty} A_m \, P(t - mT_s)$$

Want to sample at $(t - mT_s)$ to get A_m.

but $v(mT_s) = A_m P(0) + \sum_{m \neq n} A_m P(\cancel{t - T_s})\,((m-n)T_s)$

this is what I want + this $\leftarrow$ This is ISI. which over lap with our signal.

Condition to satisfy.

$$P_m(T_S) = \begin{cases} 1 & m = 0 \\ 0 & m \neq 0 \end{cases}$$

$i = 0$
$i \neq 0$ in time domain.

This is my $p(t)$

I will sample at T_s interval. when it is 0

so other ~~energy~~ is 0

I will not manage to sample at T_s. Every time. ← Problem.

So there will be contribution of other symbol.

$\boxed{p(t)}$ FT→ $\boxed{P(f)}$

↑ pulse before sample

Condition on Fourier transform of pulse before sampler.

⟶ signal ⟶ repetition in frequency if I take F.T.

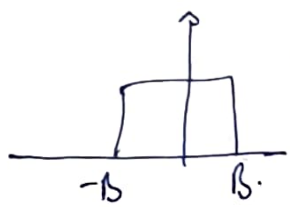
$$P(mf_s) = \sum_{m=-\infty}^{+\infty} P\left(f + \frac{m}{T_s}\right) \leftarrow \text{train of pulses.}$$

this $\sum_{m=-\infty}^{+\infty} P\left(f + \frac{m}{T_s}\right) = \underline{\text{const}}$.

This spectrum is equal to constant.

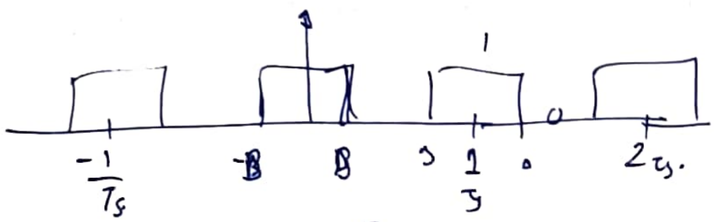
Spectrum of P(t) pulse.

P(f) assume this is rect.



BW is limited by the BW of channel. & channel is Band Limited.

So I sample so I get.



← Not a constant when 0 1

Condition do not verify. $\sum_{-\infty}^{\infty} P\left(f + \frac{m}{T_s}\right) = \text{const} \neq$ not verify.

This condition is verify

Assume that pulse before sampler is a rect ~~so~~ P in frequency is rect

& in time $\underline{P(t)}$ = Sinc. Pulse before sample is Sinc

then this condition verify when.

So $\frac{1}{T_s} - B_c = B_c$.

$\frac{1}{T_s}$ = Bandwidth. $\underline{R_s}$.

if $R_s = 2B_c$ then. it is done

or BW is $\frac{R_s}{2}$ then it is satisfied.

we can say,

if R_S is higher than $2B_W$

its [illegible]

now if

if R_S is lower.

some get overlapping. $R_S < 2B_C$.

overlapping will not harm ik [illegible].

Assuming [illegible] PF)

$R_S < 2B_C$.

this overlapping will give me. Flat

[illegible] infinite no of [illegible] will give me Flat.

Summary:

B_C Bandwidth $\quad R_S$ [illegible]

B_C higher. then

if $B_C > \frac{R_S}{2}$ infinite [illegible].

if $B_C = \frac{R_S}{2} \Rightarrow$ only [illegible] can [illegible] [illegible] same.

if $\boxed{B_C < \frac{R_S}{2}} \rightarrow$ will [illegible] have. ISI.

If use Raised cosine then B_W is not B_{min} but it will $B_C B_{min}(1+\alpha)$ $\underline{\text{excess } B_W}$

* Spectral efficiency

$\eta = \frac{R_b}{b}$ ← bits/s. ← B_W [illegible] B_W needed for one

$\frac{R_S}{2} = \frac{R_b}{\log_2 m} = B_{min}$.

$B_W = \cancel{R_S}\ \frac{R_b}{2\log_2 m}$

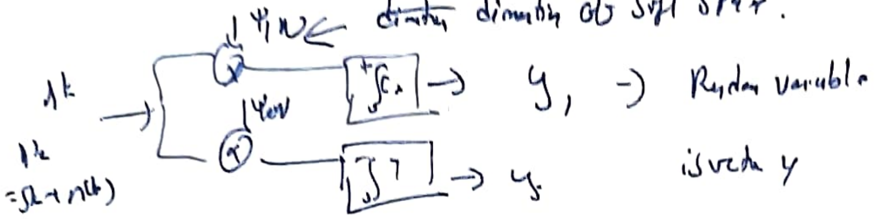
$\eta = \frac{R_b}{\frac{R_b}{2\log_2 m}} \times \Rightarrow$

$\eta = 2\log_2 m$

Optimum detector method: cross correlator (using the signal to noise ratio.

o/p of demodulator is vector which contains all the information to reconstruct data.

detection: How to minimize the probability of error when I do the demodulation.



Signal + noise are modelled as random variable random process.

y — [detector] → $\hat{m}_i$: we guess the estimated msg of the transmitted. if I don't make error otherwise I will ...

vector

↑ guess what I have tx.

So that I guess what I have tx

we use MAP - Maximum A - Posteriori Probability

Probability of transmitted one of the possible msg m_1, m_2, m_3 given I have receive y. I watch $\underline{y}$ and try to understand what is the probability that if I receive y I have transmitted m_1.

$$P(m_i / y) \leftarrow \text{A-Posterior Probability.}$$

So if I tx s_2 then I will watch the y vector & the probability after y_2 will be max as I have transmitted S_2 so we choose m_2

so we have to maximize $\underline{\underline{MAP}}$.

So tx signal is $s_m(t) = \sum_{i=1}^{N} s_i \, \psi_i(t)$

Rx sig $r(t) = s_m(t) + n(t)$

o/p of correlator: $y_k = \int_0^T r(t)\, \psi_k(t)\, dt = \underbrace{\int_0^T \sum_i s_i \psi_i(t) \cdot \psi_k(t)\, dt}_{\text{Ortho normal}} + \underbrace{\int_0^T n(t) \psi_k}_{\text{r.v. random variable}}$.

o/p of correlator will be S_K this is projection of transmitted signal over the dimension + noise.

↑ this is no.

$y_k = s_k + n_k$ ← o/p will be random variable.

$E(y_k) = s_k + E(n_k)$ ← ~~mean of~~

Noise mean is zero & variance is $\frac{N_0}{2}$.

when to calculate the variance of Random variable

autocorrelation of Noise is delta.

PSD is Flat

Go to theory

y_1 → [detector] → $\hat{m}$ → msg optimal.

our detector optimal detector chooses the msg such that my probability will be higher than other. (AP).

$$P(m_i / y) > P(m_j / y)$$

we choose the msg m_i such that this our probability is higher than any other A Posteriori Probability

So I need to calculate all the Probability but this is not practical.

but we get help from bayes theorem.

bayes tell me that this Probability can be written as function of A-Priori Probability

$$\underset{\text{A Posteriori}}{P(m_i/y)} = \frac{P(m_i)\, f(y/m_i)}{f(y)}$$

if I tx m_i I will get y i.e. I have transmitted

This PDF is also called as likelihood function.

The ~~MAA~~ MAP criteria becomes.

$$\frac{P(m_i)\, f(y/m_i)}{f(y)} > \frac{P(m_j)\, f(y/m_j)}{f(y)}$$

This MAP criteria can be written as the function of A Priori Probability

if msg are equiprobable. So MAP criteria becomes

$$f(y/m_i) > f(y/m_j)$$

Maximum likelihood criteria.

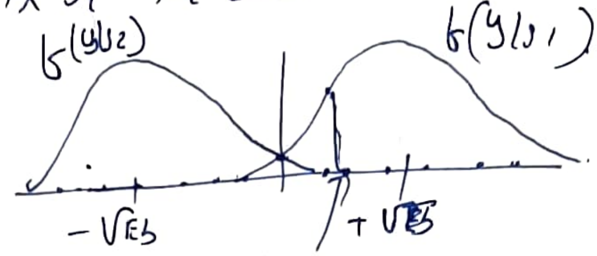
Antipodal ← constellation.

$y = I\sqrt{E_b} + n$.

$f(y/s_1)$ if I tx s_1 then denote PDF of y.

$y = \sqrt{E_b} + n$.

→ This is random variable.

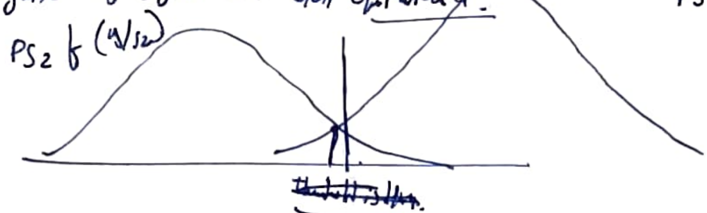


Suppose I receive a value y = this.

My maximum likelihood will choose the probability of tx s_1 is high.

~~when~~. $\boxed{y \geq 0}$ ~~whether the~~ whether the received is higher than zero or not

~~hence symbol~~ if symbol are ~~non~~ not equiprobable.



~~threshold~~

MAP is a threshold detection

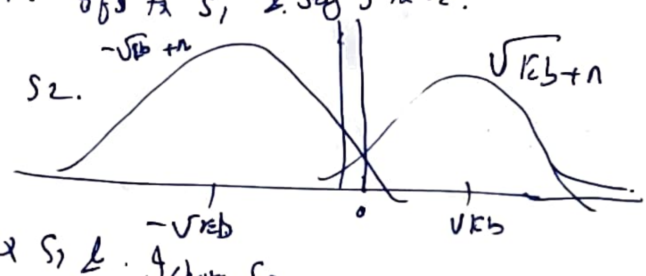
then the threshold is different.

Suppose P_{s_1} is higher than P_{s_2}.

α differs from 0.

α depends from variance of noise.

let's calculate the probability of error. If I tx s_1 & say I tx s_2.



So Probability of I tx s_1 & I chose s_2.

I Receive the $+\sqrt{E_b} + N_0$ & my value fall in -ve ~~any~~ area.

$$P(y > 0 / s_2) \times P(s_2) + P(s_1)\, P(y < 0 / s_1).$$

y higher than zero. but I tx s_2. so error.

y less than zero. I tx s_1 so error

So we need to calculate when this happen as $\sqrt{E_b} + n$ ← noise random variable. which can go from $-\infty$ to ∞.

Same calculate: $P(y<0/S_1) = P(\sqrt{Eb}+n<0) = P(n<-\sqrt{Eb})$ (18)

less than 1 — Gaussian — Gaussian PDF.

Mean is zero & variance is $\frac{N_0}{2}$.

$$= \frac{1}{\sqrt{2\pi \frac{N_0}{2}}} \int_{-\infty}^{-\sqrt{Eb}} e^{-\frac{x^2}{N_0}} dx \qquad Q_x = \frac{1}{2\pi}\int_{-\infty} e^{-\frac{t^2}{2}}$$ (18)

Use the Q function.

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{x} e^{-\frac{t^2}{2}}$$

Q function is the PDF of standard Gaussian random variable.

Calculate Q function calculate the area of the tail of the Gaussian distribution. then the

So do change of variable.

$$\frac{x^2}{N_0} = \frac{t^2}{2} = \quad t = \sqrt{\frac{2}{N_0}}\, x.$$

So this integral becomes.

$$= \frac{1}{2\pi} \int_{-\infty}^{-\sqrt{\frac{2}{N_0}Eb}} e^{-\frac{t^2}{2}} \times \sqrt{\frac{N_0}{2}}\, dt. \qquad \text{Similar to Q function.}$$

$$\text{So } \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\sqrt{\frac{-2Eb}{N_0}}} e^{-\frac{t^2}{2}} dt \qquad \text{Q function calculated in } \sqrt{\frac{-2Eb}{N_0}}$$

$$P(y<0/S_1) = Q\left(\sqrt{\frac{2Eb}{N_0}}\right) \Rightarrow \text{monotonically decreases as argument increases.}$$

Gaussian N_x

Higher the SNR Lower the errors.

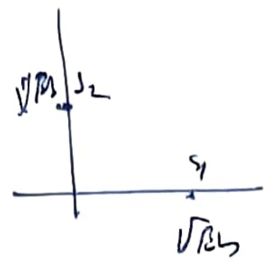
$$P(y>0/S_2) = Q\left(\sqrt{\frac{2Eb}{N_0}}\right)$$

For equiprobable binary antipodal,

$$\boxed{P_e = Q\left(\sqrt{\frac{2Eb}{N_0}}\right)}$$

Orthogonal Binary

Such as PPM



Hamming matrix for the Carrier

2.

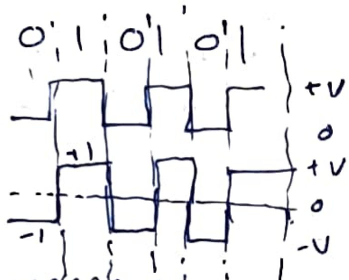
$P_{eun} = Q\left(\sqrt{\frac{Eb}{No}}\right)$ ← Peun is higher than anti Pol $\frac{2Eb}{No}$ with same SNR.

To get same Peun ~~of~~ like anti Pol. $\sqrt{\frac{2Eb}{No}}$ we have need to increase SNR.

in

--- modulation. ---

BPSK :- Phase of carrier is varied according to the msg signal.

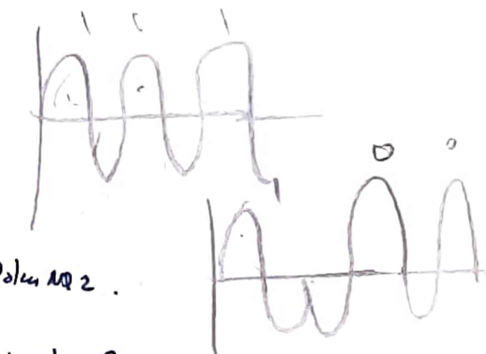


unipolar NRZ

Converted unipolar to

Polar NRZ.

multiply $F_c \times$ Polar NRZ.



← carrier which is higher frequency. much higher than B

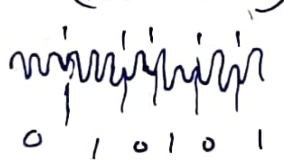
-cos wt traveL

$S_o f_c >> \frac{1}{T_s}$, $T_s >> \frac{1}{F_s}$.

bound let T_s, $R_s = \frac{1}{T_s}$

0: $-p \times \cos wt$ and $1 \times = 1 \times \cos wt$

So a^2 $(\cos wt + 0)$ $(\cos wt + \pi)$ Phase change from 0 to π

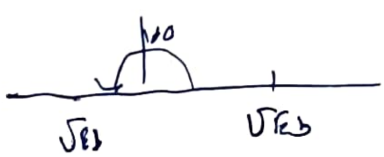


← ϕ Shift

$\sqrt{2Ps}\cos wct$ $+1$

$\sqrt{2Ps}\cos(wct+\pi) \to 0$

One dimensional.



$d_{12} = \sqrt{Eb} + \sqrt{Eb}$

$2\sqrt{Eb}$ euclidean distance

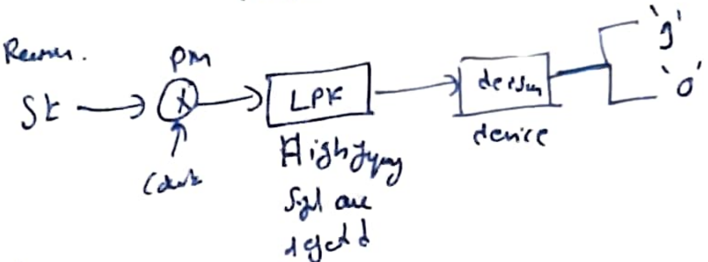
which is more than FSK & ASK.

Bw. is $2F_b$ which is equal to ASK.

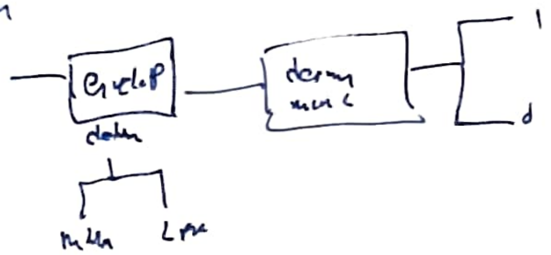
BPSK Generation.

(20)

Coherent Receiver.



Non Coherent



Envelope detector doesnt work in with square law device because it will convert -ve value into +ve counterpart.

So my non coherent receiver is not possible in BPSK.

So non coherent receiver dont require oscillator. So I dont have non coherent receiver it will be problematic.

in BPSK, 180° ϕ
0 represent with $\cos \omega_c t + 0$
1 represent with $\cos \omega_c t + \pi$

QPSK. bits change corresponds to 45° phase change.
⇒ 2 bits are combined ⇒ 90° phase ~~shift~~ change.

1	0	$\to \pi/4$
0	0	$\to 3\pi/4$
0	1	$\to 5\pi/4$
1	1	$\to 7\pi/4$.

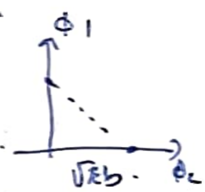
waveform can be written as function of 2
PSK dont have higher order.
PSK not power efficient. QAM is more power efficient.

$S_k = b_0(t)\cos \omega_c t + b_e(t) \sin \omega_c t$.

So $\sqrt{P_s T_b}\sqrt{\frac{2}{T_b}}\cos \omega_c t + \sqrt{P_s T_b}\sqrt{\frac{2}{T_b}}\sin \omega_c t$

ϕ_1 $\qquad$ ϕ_2

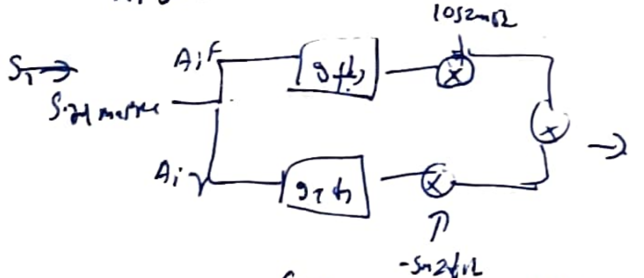
QPSK uses less BW in this case. as two bits are combined.



with the Pythagoras then I can find the euclidean distance. $d_{12} = \sqrt{2E_b}$.

Tx.

~~Receiver~~ of QPSK :-
MPSK.



→ digital phase modulation.

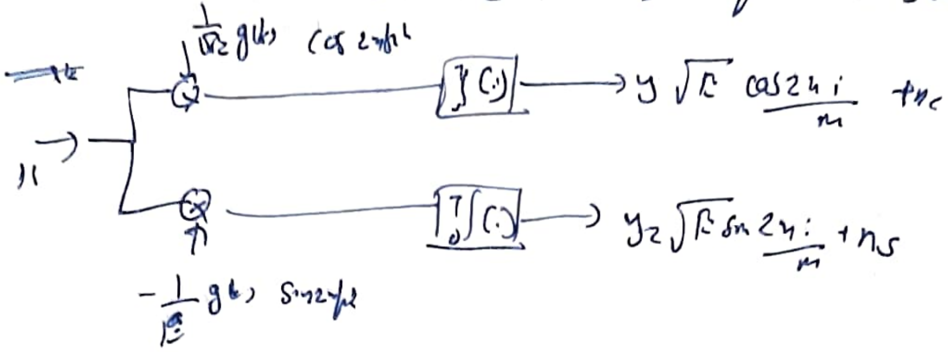
$s_i = \left(\sqrt{E}\cos \frac{2\pi i}{m}, \sqrt{E}\sin 2\pi i/m\right)$

MPSK demodulation & detection: carrier phase estimation the received signal can be represented as (21)

$r(t) = s_i(t) + n(t)$:

$\left(g_T(t) A_{ic} + n_c(t)\right)\cos(2\pi f_c t) - \left(g_T(t) A_{is} + n_s(t)\right)\sin 2\pi f_c t$.

noise is in n_c in phase & in quadrature n_s component.



As in phase & in quadrature components are zero mean & uncorrelated.

$E(n_c n_s) = 0$ $\quad = \frac{N_0}{2}$.

in coherent we derive phase & frequency.

we also get doppler effect. So phase is not constant. ~~So PLL~~

The optimum detector projects the received signal vector on each of M possible transmitted signal vectors & selects the vector corresponding to the largest projection.

$\Downarrow$

$y \cdot s_i$

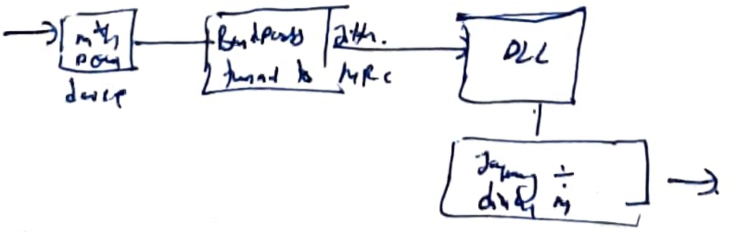
However all signal have same energy.

we calculate the angle rather than the distance.

$$\boxed{\theta = \tan^{-1} \frac{y_2}{y_1}}$$

we assume coherent demodulation that the transmitter is perfectly synchronized with the receiver in both ~~domain~~ time & frequency.

* So we need to generate phase coherent carrier at the receiver by using the received signal.



Bit error probability is the same as BPSK

$Q\sqrt{\frac{2E_b}{N_0}}$

$P_e = Q\left(\sqrt{\frac{d_{ij}^2}{2N_0}}\right)$.

Doubling M data deteriorates the performance by a factor as to 6dB

Error will increase as I increase M. with same Bandwidth using efficiently.

8PSK → 6dB.

to get high spectral efficiency. I need to increase SNR by 6dB.

while Mqam is more power efficient.

need to increase by 3dB.

For qAM PSK we need only 2 correlator and 2 match filter.

Bit error rate also depends on mapping of k bits into the the corresponding signal phases.

Non coherent PSK: DPSK :- Here Phase synchronization is eliminated using differential encoding.

Here. Encoded the information in phase difference between successive signal transmission.

to generate DPSK there are 2 steps.

① differential encoding of binary bits.

② Phase shift keying.

1 0 0 1 0 0 1 1.

1 1 0 1 1 0 1 1 1

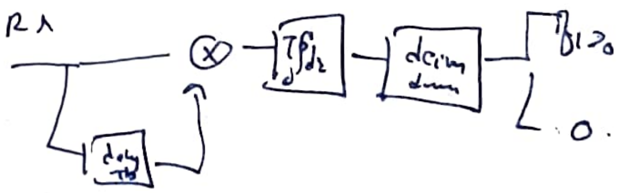
one have transition. transition 0

Previous bit 1 → No transition. — 1

1 0 1 1 0 0 1 0

1 1 0 0 0 1 0 0 1

$0\ \pi\ \pi\ \pi\ 0\ \pi\ \pi\ 0$



BW is reduced.

but probability of error is high.

Energy Needs

$P_{DPSK} = P_e = \frac{1}{2}\exp\left(-\frac{E_b}{N_0}\right)$

is power efficient.

DPSK integrate.

$$y = \int_0^{T_b} r(t)\, r(t - t_b)\, dt = \int_0^{T_b} \cos(\omega_c t + \psi_k + \phi)\cos(\omega_c t + \psi_{k-1} + \phi)\, dt$$

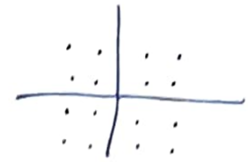
So constant ϕ becomes irrelevant.

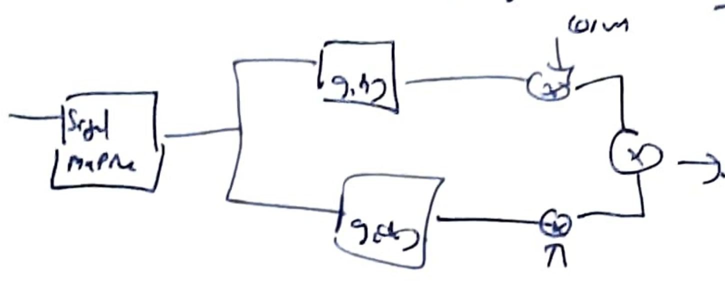
if $\psi_k - \psi_{k-1} = 0$ then $y > 0$. if $\psi_k - \psi_{k-1} = \pi$ then $y < 0$.

QAM → . If I could also change the amplitude.

Then we are also increasing the SNR.

we can also increase the spectral efficiency.





It is specific combination of ASK & PSK.

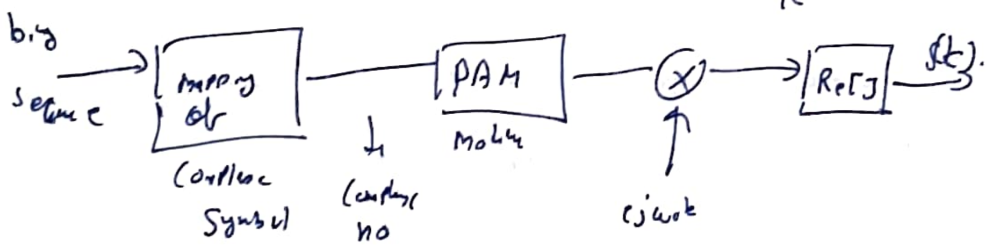
Both amplitude & phase change in different symbol.

QAM is implemented by modulating 2 PAM baseband signal being phased in quadrature.

2 PAM s.

$$\sum_k a_k g_T(t-kT) \quad e \quad \sum_k b_k g_T(t-kT).$$

$$s(t) = \sum a_k g_T(t-kT)\cos\omega_c t + \sum_k b_k g_T(t-kT)\sin\omega_c t$$



Here QAM becomes QPSK like.

BW of baseband is same as passband · both we use both in phase & quadrature channel. otherwise the bandwidth would be double.

So low energy is good for us.

BW in passband gets double 2B.

The min bandwidth is double

Bmin in PSK & QAM is twice.

~~Space in QAM it is~~ 3dB. SNR needed will not increase by 6 dB.

it will increase by 3 dB

QPSK — robust to noise

QAM → gives more bits ← noise.

Constellation → satisfy in ϕ or frequency constellation.

32APSK → less distortions non linearity in amplitude.

21 24

MFSK → f_1-f_2- f are orthogonal over different frequencies.

2FSK

in FSK if we increase the spectral efficiency do not increase.

FSK consume more BW. it is suitable for satellite communication.

to decode FSK modulation we need matched filter & MPLL.

Suppose we use 2 FSK → we use 4 matched filter. which allow us to avoid the ϕ & carrier sync

ϕ is not important. in FSK

* MFSK is more energy efficient than MPSK.

But it can achieve higher data rate per unit of bandwidth so it is more energy efficient than other.

to improve bandwidth efficiency & decrease signal to noise ratio we can

* using channel coding. Such as LDPC → to achieve high data rate with less bandwidth.

* As M increases, bandwidth efficiency of MPSK/MQAM increases, but bandwidth efficiency of MFSK decreases.

Tradeoff bandwidth efficiency – Power efficiency.–

maximum rate at which info is transmitted without error.

* detection compute the ~~cross~~ in the 2 frequency & pick the larger or use LP detection.

3 possible actions to reduce the error probability –

1. Increase Bandwidth.
2. Increase the SNR by increasing the tx power.
3. Increase system complexity.

channel encoding is very important in power limited system such as satellite.

* BPSK & QPSK have same E_b/N_0, probability of error. Hence QPSK is better bandwidth efficiency.

* M-QAM → is superior than MPSK.

* MPSK → M-QAM increases bandwidth efficiency and achieves lower energy efficiency.

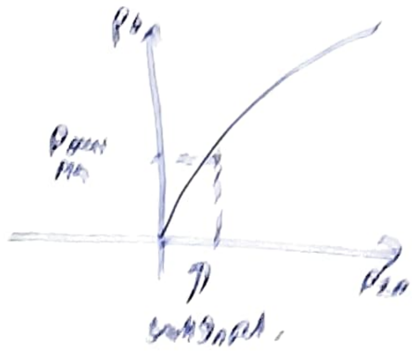
* MFSK → higher energy efficiency and reduced bandwidth efficiency.

* In satellite system power efficiency is more valuable than in terrestrial system. Scheme under modulation is difficult. (OQPSK)

Information resides in phase and in amplitude is called as constant envelope.

FSK & PSK are constant envelope modulation which are desirable for some situations where amplitude is ... in the satellite system with power amplifiers working in the non linear region.

affected badly BPSK QPSK.

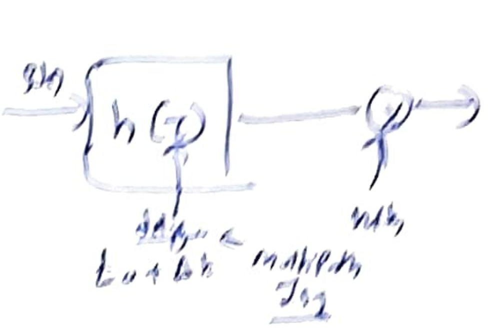


Answer.

DVB-S2 → point-to-point communication.

Zigbee → for more distance ... less ...

Equation → s(t)



if constellation is rotated between 0 & T

$\int_0^T$ →

coherent phase shift

← if we see the waveforms & if they are not coherent they will distort ...

Estimate the delay & add this will solve our Problem.

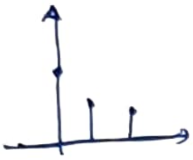
← This thing is called delay spread

opposite of delay spread is coherence Bandwidth $B_c = \frac{1}{T_{spread}}$

If we do Fourier transform of this h(τ)

if we receive one pulse no spread again we get constant

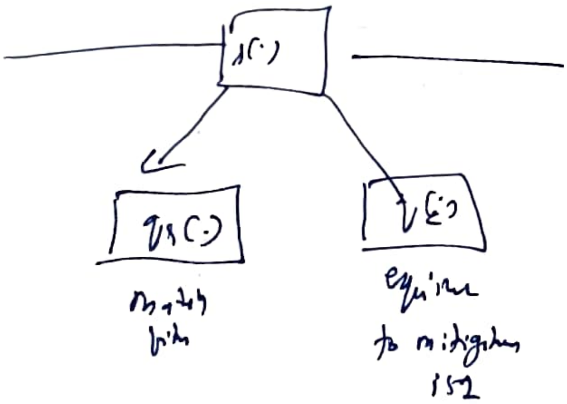
means each frequency is attenuated is same amount..



Behaviour of channel will be different for different frequency. You will get frequency selectivity.

Objective of the equalizer. to remove the effect of channel & have equivalent channel. to have flat response

At the end after equalizer the channel response of channel should be flat.



noise ~~add~~ white

Assume we know the channel. using Pilot Pulses & estimate the channel.
& Perform the demodulation & equalization.

Design the Receiver to have the minimum ~~square~~ mean square error.

So at the end the optimal Receiver should have this frequency response

i.e Product of 2 Parts. ① Fourier transform of Pulses before match filter

So it is the combination of transmit filter & channel

Pilot Sym ~~Blue~~ rate.

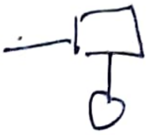
design to reduce mean square error.

minimize the mean square error.

Product of 2 Part the optimum Rx here.

F.T of Pulse of

cascade of Tx filter & channel.



how the. whitening the noise → matched filter

whitening the noise

make the noise white.

The frequency response of matched filter is the F.T of Pulse here (t) g(t).

in case of white Noise it is just matched filter.

matched filter is implemented in analog domain & second Part in digital domain.

frequency response of digital filter.

Sample by $1/T$.

So this Part is working to remove the ISI

or mitigating the ISI.

* Ix symbol & they are uncorrelated zero mean
① ~~x~~ n consider uncorrelated zero mean symbol

$\boxed{MF} \to \boxed{EQ}$ filter.

then using the equalizer is called as. zero force

* if noise is negligible.

zero forcing I do the inverse of frequency domain.
~~Note~~. I apply the filter whose ~~fr~~ response is inverse in frequency domain.
I will remove the effect of the ISI by channel.

* MSE → ~~the~~. the error depends from noise & ISI
the optimum Rx Should reduce both ~~noise~~ ISI noise in terms of MMSE.

optimum Rx will do the best. ~~[illegible]~~

with noise if I apply ZF
then Low noise at denominator $\frac{1}{G(\omega)}$
So it Pay number the amplify noise

in cable not known the channel.

in wireless we don't know channel.

* if we do need to adapt digital filter · is done in digital

~~Soon~~ Soon implement after ~~digital~~ sampling.

I remove the noise & then sample & the implement ~~digital~~ optimal Rx in digital.

Matched filter & Equalizer & used in digital.

Equi..   h(t)

no delay on

RT

Captilian

So freq response of channel is Flat

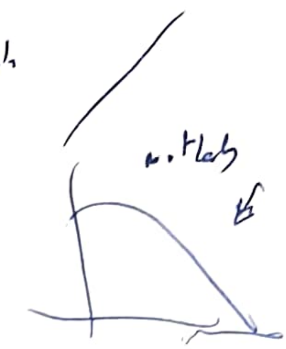
So all freq is attenuated in same way.

if channel introduce an ISI

So behaviour of channel will be different

So channel will be freq selective.

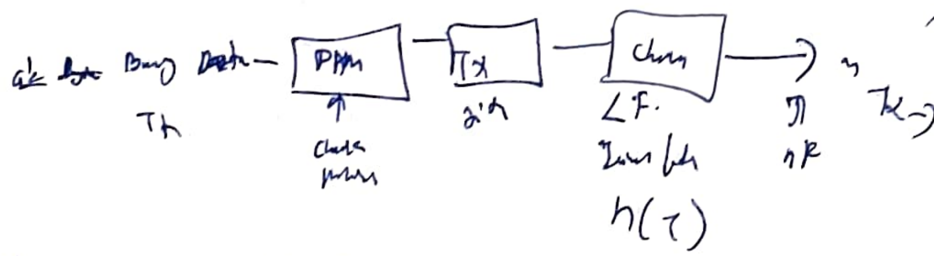
So we need the channel response to be Flat

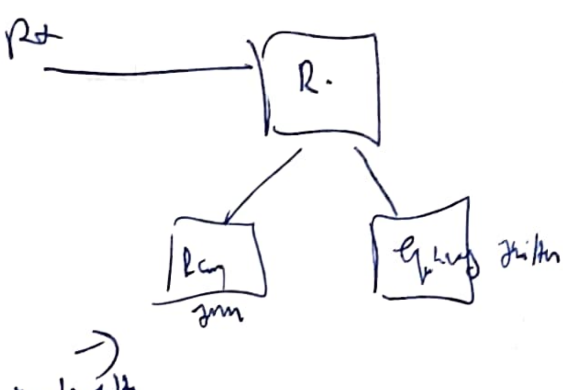


Means freq selective so we not able to make difference between the points.

So we need to do some thing

So we apply a filter at receiver called as equaliser.





Assume noise is not white & change by freq.

not flat

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If the ~~[illegible]~~ filter satisfy the nyquist criteria then optimum Receiver 2nd Part of receiver will be constant as I will not require equalizer.

we get what is expected.

~~[illegible]~~ this Nyquist is found when there is no noise.

when there is noise we have to be careful

~~[illegible]~~ even when symbol are correlated even if nyquist criteria is satisfied then also I need equalizer.

even I don't have ISI will need equalizer because the channel which is shown is filter with memory. ~~it introduce the.~~

The symbol have already correlation. ~~that mean~~ already have memory that correlation means what I tx now depends on what I tx before.

for this correlation I need equalizer & equalizer will remove this correlation.

This correlation makes the demodulation hard.

In cable medium I know the channel. In wireless I don't know channel I estimate it to adapt the channel.

If I have to adapt something update continuously is always done in digital after sampling.

after I remove the noise I sample & then I implement my optimum receiver Part is digital. This is the combination of matched filter & equalizer. is digital.

* Then we use the algorithm to adapt & estimate the coefficient of by adaptive filter. digital

* MSE