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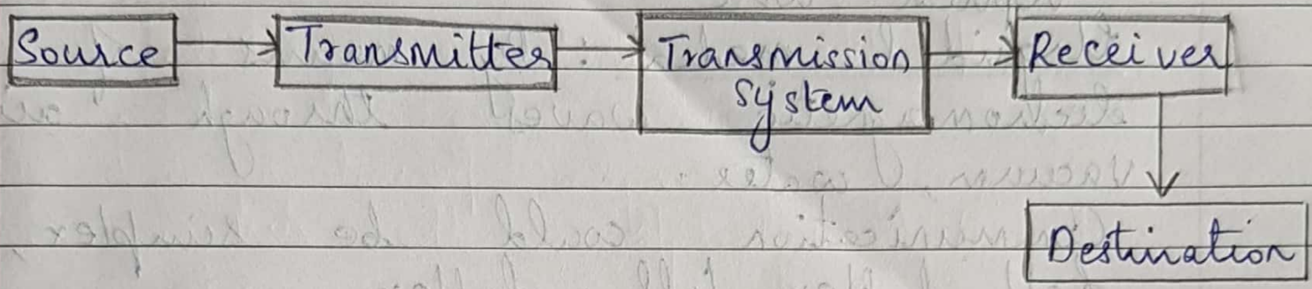
MODULE - I

DATA COMMUNICATION MODEL



* Source : generates the data to be transmitted.
  Eg: Telephone, a PC

* Transmitter : transforms & encodes information in electromagnetic waves or pulses which can be transmitted across a transmission system.

* Transmission System : A single line or a complex network connecting source & destination.

* Receiver : Accepts electromagnetic signal from the transmission system & converts into a form which can be handled by the destination.

* Destination : Takes incoming data from receiver.

Communication system exchanges data between two parties.

Transmission medium could be through guided media.
Eg: twisted pair, coaxial cable, optical wire.
Unguided media : By means of electromagnetic waves through air, vacuum, water.
Communication could be simplex, half duplex, full duplex.
Simplex : Signals are transmitted only in one direction ie; from transmitter to receiver. Eg: Radio stations
Half Duplex : Both stations can transmit & receiver but only one at a time. Eg: walkie talkie
Full duplex : Both stations can transmit & receive simultaneously.
Eg: Telephone

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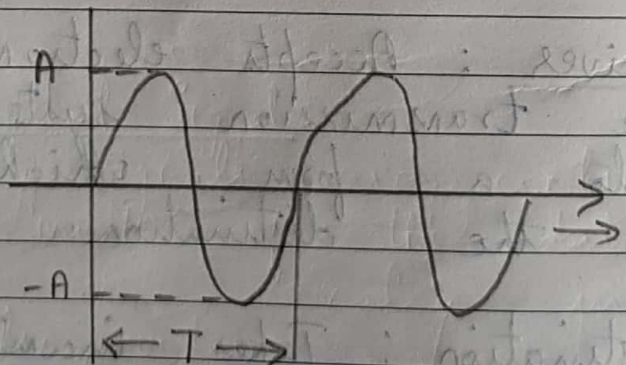
Signals

1) Periodic Signal



A periodic analog signal

An analog signal is a one ~~wave~~ in which the signal intensity varies in a smooth wave over time or has no breaks / discontinuities in the signal.

Mathematically,

$$\lim_{t \to a} s(t) = s(a)$$

* $V = \nu \lambda$

⇒ Analog & digital signals

Time domain : Consider the sine wave

A simplest form of a signal is a periodic signal ie, one in which the same signal pattern repeats over time.

Mathematically, it can be stated as

$s(t+T) = s(t)$, $-\infty < t < \infty$.

$$T = \text{period} = \frac{1}{f}$$

Time domain :

Sine wave : It is represented by 3 parameters peak amplitude A, frequency f & phase ϕ.

$$s(t) = A \sin(2\pi f t + \phi)$$

A: peak amplitude is the maximum value of the signal. (strength of the signal over time).

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f: frequency - rate at which the signal repeats.
T: Period - amt. of time for 1 wave

$$f = \frac{1}{T}$$

ϕ: Phase - measure of a relative position in time within a single period of a signal.
λ: Wavelength - the distance occupied by a single cycle.
- Distance b/w any two points of corresponding phase of 2 consecutive cycles.

$$\boxed{v = f\lambda}$$

Frequency domain:

There will be second frequencies which are integer multiples of fundamental frequency & the period of total signal is always = the period of fundamental frequency.

Analog & Digital Signal:

Analog signals are continuous in nature & most of the data collected by sensors are analog in nature. For eg: audio, video

Digital signals are discrete in nature. Signals are electric or electronic representations of data. Physical propogation of signals through a substance or medium which is suitable is called signalling.

Communication of data by propogation & process of signal is called transmission.

In any communication s/m, point-to-point comm. of data is done by electromagnetic signals. It could be through a wired or guided media.
Eg: coaxial, twisted pair, fibre optical

If $s(t+T) \neq s(t)$ then it is called an aperiodic signal.

Digital signals are cheaper than analog signals. They are less susceptible to interference. But they suffer more from attenuation.

Analog Transmission

It is a means of transmitting analog signals without regard to their content. (It may contain digital or analog data). It needs amplifiers to boost the signal for long run communication. Noise also gets amplified & cascading makes it more distorted. But analog communication can tolerate more distortion & in the receiver, data still will be intelligible.

Digital Transmission

It is concerned with content of the signal but it can be transmitted only to a limited distance before attenuation makes it un-intelligible. Noise & other impairements endager the integrity of data. To achieve greater distance, repeaters have to be used.

9/8/17 The repeaters recover the digital data from analog signal & generates a new clean analog signal. The noise (distortion) does not get cumulated.

Advantages :

1. The VLSI technology have made the digital signalling much cheaper.
2. Data integrity is better.
3. Better capacity utilization.
4. Better encryption technology is available. It makes it more secure.
5. Similar signals can be integrated & makes overall communication cost lesser.

~~TRANSITIONS~~

TRANSMISSION IMPAIRMENTS

1) Attenuation

* ~~As~~ Strength of signal falls off with distance for any transmission medium; for guided media, exponential increase in attenuation is experienced.

Hence, a constant no. of decibals per unit distance is introduced.

For unguided media, it is a more complex function of atmospheric conditions are parameters. (humidity)

The main concerns about attenuation are

1. The received signal should have sufficient strength to be detected as a signal.

2. It should have sufficient level of power as compared to the noise to be received without error.

3. Attenuation very often is an increasing fn. of frequency. Hence, for higher frequency, attenuation is more.

Amplifiers & repeaters are used to alleviate the strength failing & distinction of s/g from noise to some good extent.

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Noise :

In any data transmission, the s/g consists of the transmitted s/g, distortions by transmission s/ms & additional unwanted s/gs inserted b/w transmission & reception.

All undesired s/gs are referred as noise.

The types of noises are :

1) Thermal noise
2) Intermodulation noise
3) Cross talk
4) Impulsive Noise

Thermal Noise

It occurs due to the agitation of ēs. It is present in all transmission media & is a function of temperature. It is uniformly distributed across all bandwidths used in communication s/m & hence is referred as 'white noise'. This cannot be avoided & hence it gives an upper bound for comm. s/m performance.

For satellite comm., this is particularly relevant. The amt. of thermal noise is given by

$$N_0 = KT \ (W/Hz)$$

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Where N_0 : is the noise power density in W/Hz
K : Boltzmann Constant
$K = 1.38 \times 10^{-23}$ J/K
T : Temperature in K

Intermodulation Noise

When different fs share same transmission medium it results in intermodulation noise. It produce a freq. which is a sum or difference of two original s/gs or the multiples of those frequencies.

This is due to the non-linearity of transmitter, receiver and/or transmission medium intervention.
Excessive non linearity is caused by component malfunction or overloading of excessive s/g strength.

Cross Talk

It occurs by electrical coupling b/w nearby twisted pair or rare in case of coaxial cables carrying multiple s/gs.
It can also occur when microwave antenna pick up unwanted s/gs even if the antennas are highly directional.

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This is caused by spreading of s/gs during propogation.
Cross talk is of the same order of magnitude as the thermal noise.

Impulsive Noise

It is non-continuous. It consist of irregular pulses or noise spikes. It is highly unpredictable & of very short duration generally having relatively high amplitude.

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The source of an impulsive noise could be electromagnetic disturbance such as lightning or faults and flaws in communication s/ms.
It is generally a minor annoyance for ~~di~~ analog data such as short clicks & crackles. But no loss of intelligibility for the data. But its a primary source of error for digital data communication.

Channel Capacity

The channel capacity gives the maximum rate at which data can be communicated over a communication path under given conditions.

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Data rate : It can be expressed as bit per second at which the data can be communicated. (BPs)

Bandwidth : It is constrained by the transmitted s/g, transmitter & the transmission medium.
It is usually expressed as cycles per second or Hz.

The Nyquist Bandwidth:

In a noiseless channel, the only limitation of data rate is by the bandwidth of the s/g. Nyquist states that "If the rate of transmission is 2B, s/gs with frequencies no greater than B is sufficient for the s/g rate. The converse is also true. Given a bandwidth of B, the highest data rate carried by the channel is 2B."

$$C = 2B \log_2 L$$

(Channel capacity)

Shannon's Capacity Theorem:

– Claude Shannon 1944

In a noisy channel, one has

to consider the relationship b/w data rate, noise and error rate. At a given noise level, higher the data rate, higher will be the error rate.

Signal to noise ratio (SNR) is the ratio of the power of s/g to the power of noise.

$$SNR = 10 \log_{10} \frac{\text{s/g power}}{\text{noise power}}$$

This gives the possibility of making a code at ~~bit~~ this rate in a noisy channel without error.

$$\underset{(\text{bits/s})}{C} = B \log_2 (1+SNR)$$

B: bandwith
C: Channel capacity

Transmission Medium.

MODULE - II

TRANSMISSION MEDIUM

To transmit any sig from the transmitter to the receiver, the transmission medium is required.

Any transmission media will have transmission impairements such as noise, cross-talk, intermodulation frequencies etc and attenuation.

There are two types of transmission media.

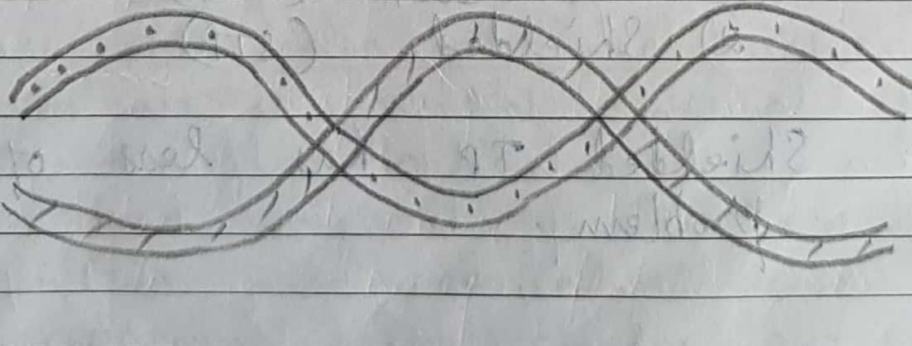
1) Guided
2) Unguided

Be it any medium, the factor we are interested in are the data rate, bandwidth, error rate. etc.

Usually, if bandwidth is greater you will have higher data rate.

Transmission impairements are highest for twisted pair & lowest for optical fibre. Coaxial cable stand in between.

Twisted Pair Cable:



They are separately insulated twisted together to lessen the interference. They are bundled into cables. They are usually installed in home wiring.

Transmission Characteristics:

Twisted pair can be used to transmit both analog & digital sigs.

Analog sigs need amplifiers at every 5-6 kms. Digital sigs need repeaters at every 2-3 kms.

This has limitations in distance, bandwidth & data rate.

Hence, usually used in home connections.

It is more susceptible to interference & noise due to electrical coupling.

Applications: Home connections from ~~PBX~~ PBX to head office, in subscriber loops etc.

It is usually less expensive than others.

2 types:
1) Unshielded (UTP)
2) Shielded (STP)

Shielded TP has less of interference problem.

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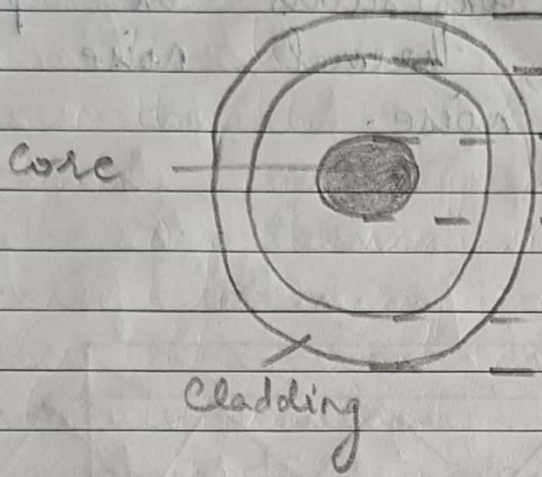
Coaxial Cables



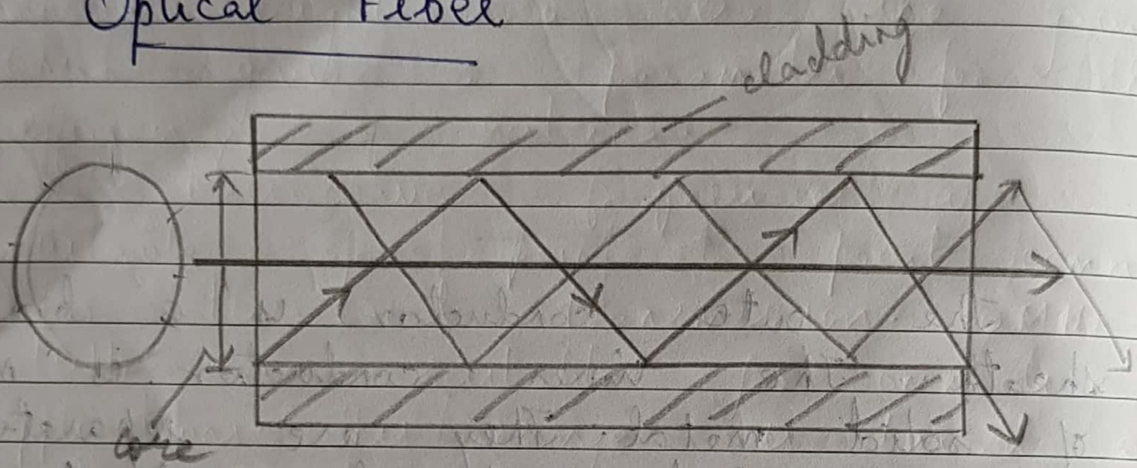
The outer conductor is a braided sheath. The inner conductor is made of solid metal. They are separated by an insulated material and is all covered by a padding. The material & thickness of padding depends on the application for which it is used for.

These cables can be used to operate over a wide range of frequencies. The typical applications are ~~telephone~~ television distribution, long distance telephone transmission, short run computer system's links, LAN. The transmission characteristics are generally in b/w twisted pair & fiber optic cables.

For long call analog transmissions amplifiers are needed every few kms. The spacing gets closer as fs get higher.

The highest bandwidth usually used for analog transmissions is 500 MHz. The principle constraints on performance are attenuation, thermal noise & intermodulation noise.

Optical Fiber



Physical description:

An optical fibre is a thin flexible medium capable of guiding an optical ray which has 2-125 μm diameter. Various types of glasses, plastic & silica.

An optical fibre cable has a cylindrical shape & consists of 3 concentric sections - a core, cladding and a jacket.

The core is the innermost section & consists of 1 or more very thin strands or fibres made of glass or plastic & has a diameter in the range of 8-100 μm.

Each fibre is surrounded by

its own cladding which again is made of glass/ plastic but has optical properties different from that of core so that the light would always be confined within it by the property of 'Total Internal Reflection.' The outermost layer is called a jacket which is composed of plastic or other materials to protect against moisture, abbration, crushing & other environmental dangers.

Application :

It can be put for long distance telecommⁿ, military applⁿ. & have the following properties which have ↑ its usage:

1) Greater s/g handling capacity.
2) Its smaller size & lighter weight.
3) Comparatively low attenuation.
4) Electromagnetic isolation.
5) Greater repeater spacing.

Typical Applications:

1) Long-haul trunks, metropolitian trunks.
2) Rural exchange trunks
3) Subscriber loops
4) Local Area Networks

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Wireless Transmission

1GHz to 740GHz range is used as microwave freq. for wireless transmission. Antennas are a must for wireless transmission. Antenna size, its shape tower height so also the gain which gives a parameter for the efficiency of antenna.

Microwave frequencies are suitable for point-to-point commn. for terrestrial transmission. 30MHz to 1GHz are suitable for omnidirectional applications. This range is known as radio range.

Effective area of the antenna is always related to its physical size & shape.

Terrestrial Microwave

Dish antennas, usually parabolic antennas are used for this purpose with a typical size of 3m in diameter, they are fixed rigidly at considerable height for LOS transmission to the receiving antenna.

Long distance transmission is achieved by a series of relay towers if it is out of distance for point-to-point transmission.

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Applications:

- Having microwave links for long haul telephone commⁿs.
- Closed circuit TV.
- For cellular phone.

Transmission Characteristics:

The main source of loss is attenuation. where

$$L = 10 \log \left(\frac{4\pi d}{\lambda}\right)^2$$

L: loss
d: distance of antenna
λ: wavelength

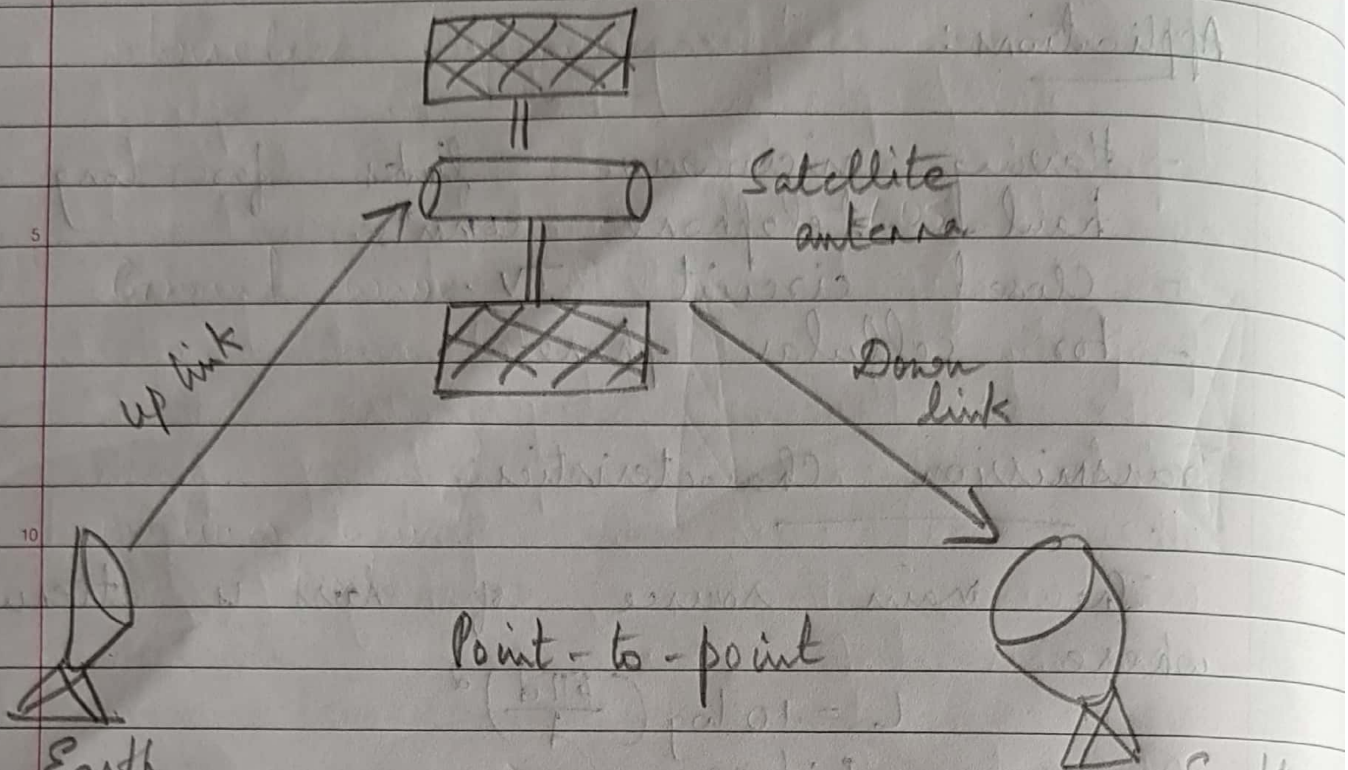
Usually, 1 to 4 GHz are used for this purpose. For long haul commⁿ 4 to 6 GHz are used. 12GHz is typical for cable TV antenna. For very short P to P commⁿ, we use 22 GHz.

It should be noted that for higher frequencies, antennas will be cheaper & smaller.

Repeaters are usually used in 10 - 100 kms.

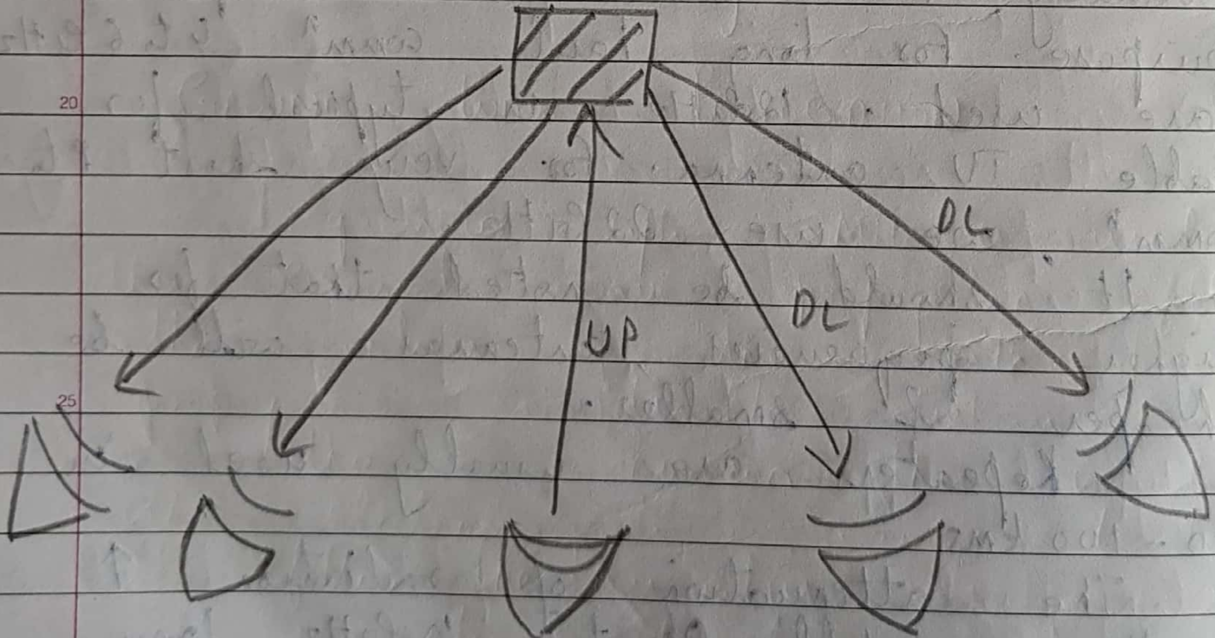
The attenuation of radiation ↑ with rainfall. About 10 GHz freq, this is more noticable.

Growing popularity of μwaves, interference also is a problem. Hence f bands are strictly regulated.



4/6 GHz – 3°
12/14 GHz – 6°

Broadcast



Satellite Microwave:

In this, the comm^n satellite act as a microwave relay station. It can link to transmitter/Receiver as in point-to-point transmission known as earth station.

A single satellite can operate on a number of freq. bands called transpond channels or transponders.

An up link (Earth station to satellite) is a different freq. of atleast 2GHz is used with a Down link (from satellite to ES) is used for this purpose. Either 4/6 GHz or 12/14 GHz are typical for this purpose. An earth angle or an angle spacing of 4° is required for 12/14 GHz & 3° for 4/6 GHz.

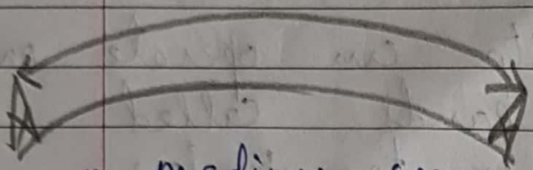
The satellite should be geo synchronous for comm^n purpose using microwave rays.

Applications:
- For TV distribution
- For long distance telephone comm^n.
- For private business networks.

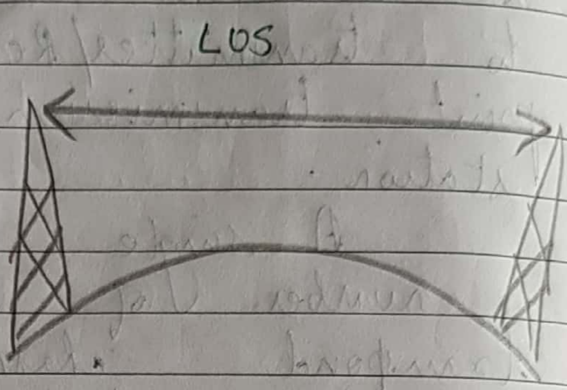
Some properties of satellite comm^n are for long haul comm^n, a delay is felt in vocal conversation also introduces problems in areas of error control & flow control.

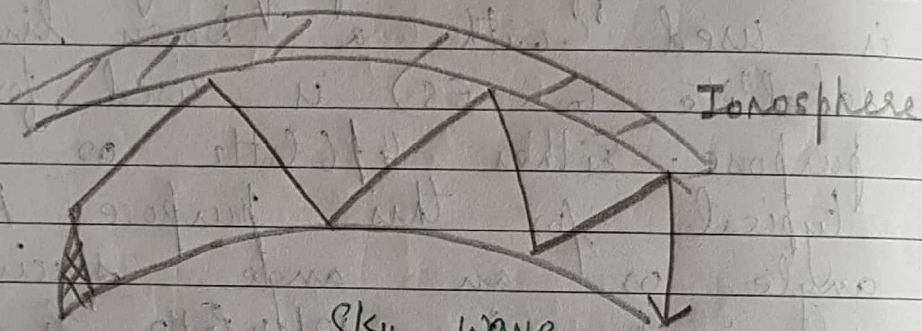
Wireless Propogation

Ground wave



medium wave bands.





Sky wave

GROUND WAVE PROPOGATION

Frequencies upto 2MHz is used for this. Ground wave propogation follows the contour of the earth & can propogate considerable distances well on the horizon. This is because the electromagnetic wave's tendencies to follow the curvature as a result of magnetic attraction and slowing the wavefront nearer.

This is typically used for ground wave commn for AM radio for medium band waves.

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SKY WAVE

This is used for amateur radio, CB radio & international broadcasts such as BBC.

In sky wave propogation, the s/g from earth based antenna is reflected from ionosphere back to earth. So after multiple reflection, SW s/gs can travel a number of waves bouncing back & forth b/w ionosphere

This can be picked up from 1000s of km.

LOS

Only fs above 30 MHz are used for this. Main problems faced are refraction and the optical LOS distance can be given by

$$d = 3.57\sqrt{h}$$

d: dist. travelled by electro magnetive wave
h: tower height
k: used to account for refraction
k can be taken as 4/3 for ordinary purposes.

If 2 antennas are used for

$$3.57\left(\sqrt{kh_1} + \sqrt{kh_2}\right)$$

will be the maximum distance achieved where h_1 & h_2 are heights of transmitting & receiving antennas.

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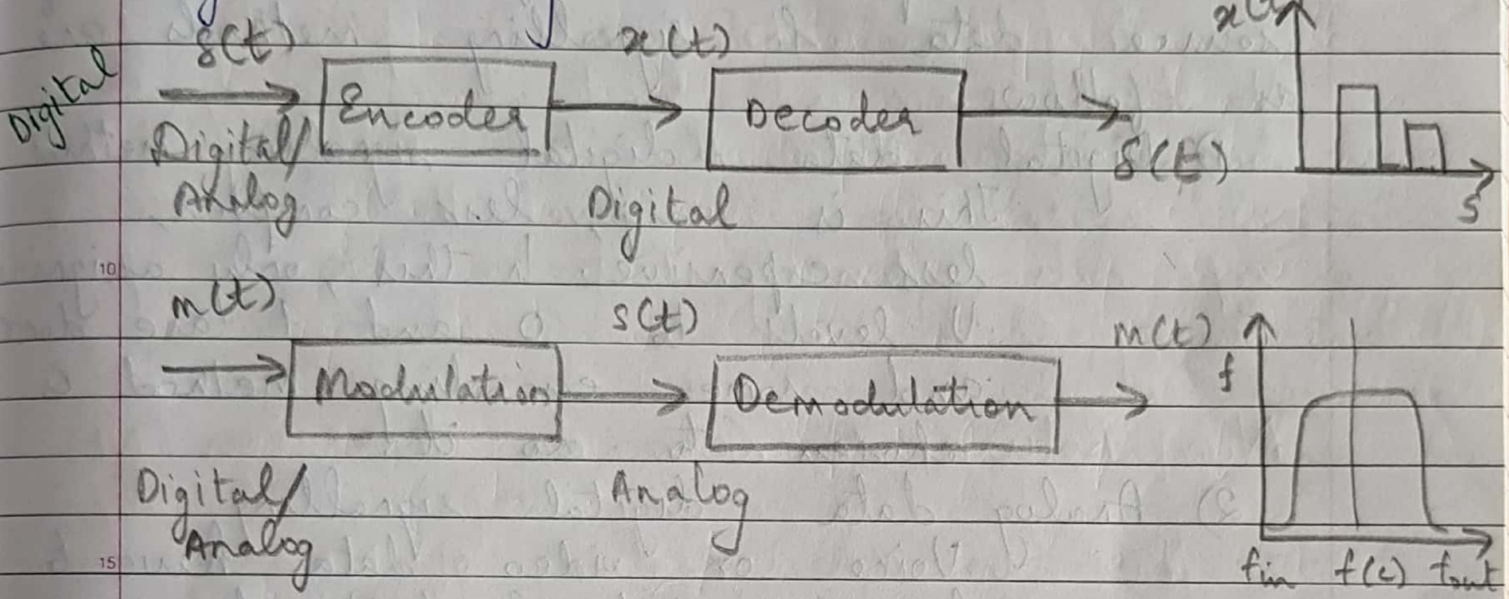
MODULE - III

Signal Encoding Techniques

Signal Encoding Methods:



The encoding depends on the source data & signalling methods employeed for transmitting the info. through the transmission medium. The data could be either digital or analog & signalling method could also be either digital or analog.

In digital signal modulation, the source could either be digital or analog. Use an encoder to assign values of voltages to make it into a digital form of EM impulses. The info. is gathered at the receiver in digital form. In analog signalling method, data could be either D or A. This is modulated with a carrier f & the base band is converted to a modulated analog wave which will be received as a band pass s/g around central f, in the receiver.

The possibilities of combinations of source data & signalling methods are as follows:

1) Digital data - digital signalling method

This is easy, less complex, less expensive. In this, only assigning V levels to 0 and 1 are done and some encoding method is chosen to do it.

2) Analog data - digital signalling

Voice or video data have to be sampled & digitized for Digital transmission & some modulation methods such as pulse code modulation to sample it & digitize it.

3) Digital data - Analog signalling

A modem converts D to A to be transmitted across an analog line such as telephone or optical fibre. Different types of shift keying such as amplitude shift keying, frequency shift keying or phase shift keying are used for this purpose.

4) Analog data - Analog signalling

This is very easy. Only carrier wave modulation of base band is necessary such as AM, FM or PM of the phase band of the modulated band.

Digital Data Digital Signalling

Unipolar – In this, the s/g elements have the same algebraic sign ie; either only +ve or only -ve.
For eg: 0V and +5V to represent 0 & 1.

Bipolar / Polar – One logic level is represented by a +ve sign & the other by a -ve sign.
For eg: +5V to represent 1 & -5V to represent 0.

Duration: The time to ~~#~~ encode 1 or 0 expressed in bits/s.

Modulation rate: The no. of elements modulated per second expressed in 'baud' & a clock pulse is used to synchronize the transmitter & receiver.

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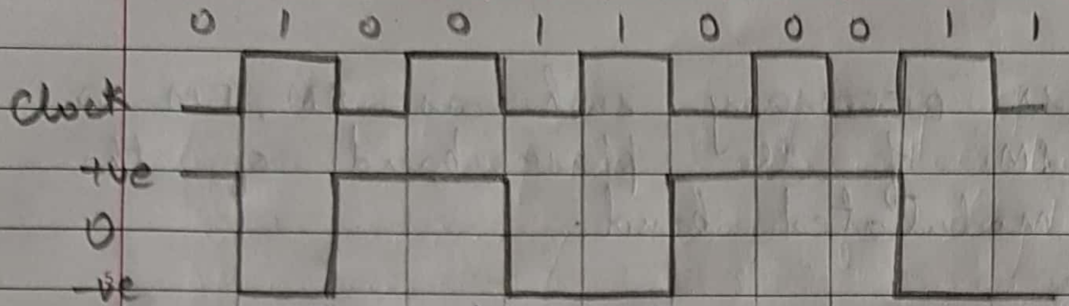
Non Return To Zero (NRZ):
(NRZ-L)

1 represented by a -ve voltage.
0 by a +ve voltage.

- Non return to zero is the most common, easiest way to transmit digital signal to use 2 different V levels for two binary digits.



They have the common property that V level remain constant during each bit interval & there is no transition & return to 0V level. Here a -ve V represents binary 1 & a +ve V represents binary 0.

Non Return To Zero on Inversion (NRZ-I)

This is a variation of NRZ known as NRZ-I.

Quite like NRZ-L, this also maintains a constant voltage pulse for the duration of a whole bit. The data itself are encoded as the presence or absence of a s/g transition at the beginning of a bit time.

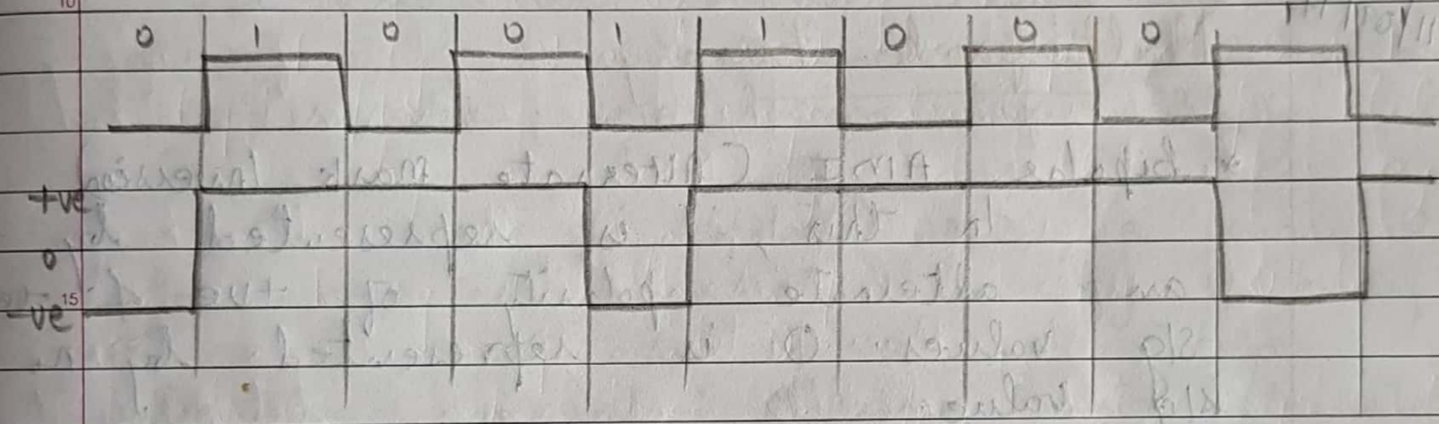
A transition at the beginning of a bit time denotes a binary 1 for that bit time. No transition indicates a binary 0. (Transition means a low to high or high to low).

NRZ-I is an example of differential coding. In this, the info. to be transmitted is represented in terms of changes b/w successive s/g elements rather than the s/g element itself.

The encoding of current bit is determined as follows:

If it is a 0 then, the current bit is encoded in the same s/g as preceded bit.

If it is a binary 1 then, the current bit is encoded with a different s/g than the preceding bit.



NRZ codes are the easiest to engineer and it makes an efficient use of the bandwidth.

The main limitation of these s/gs are the presence of a DC component & the lack of synchronization capabilities b/w the transmitter & receiver.

When there is a long strings of 1s or 0s in the NRZ-L or a long string of 0s in NRZ-I, in such circumstance a drift b/w timing of transmitter & receiver will result in loss of synchronization b/w the two.

Due to the simplicity & relatively low f response characteristics, NRZ codes are commonly used for digital magnetic recording.

Due to their limitations, they make the code unattractive for digital s/g transmission applications.

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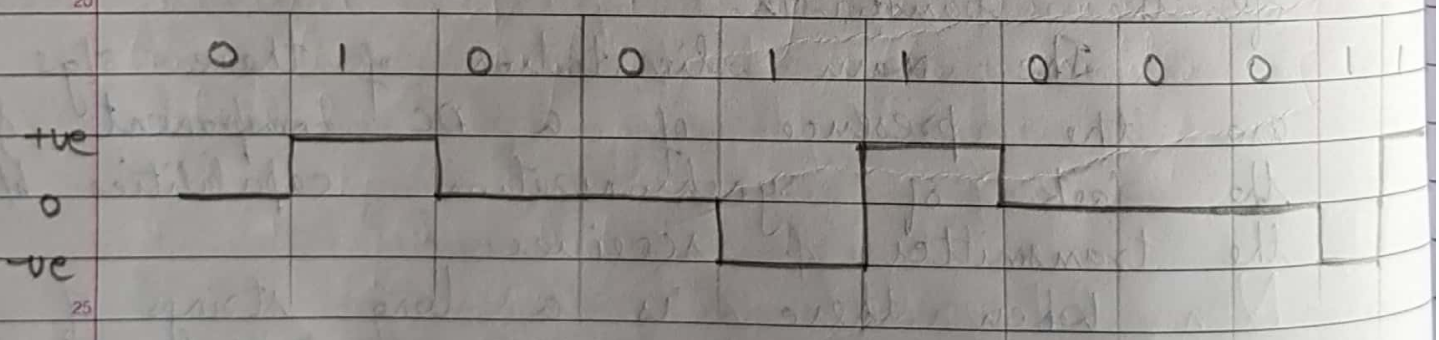
Multilevel Binary

* Bipolar AMI (Alternate Mark Inversion)

In this, 1 is represented by an alternate polarity of +ve & -ve s/g values. 0 is represented by no s/g value.

Assume, the preceding 1 had a -ve value.

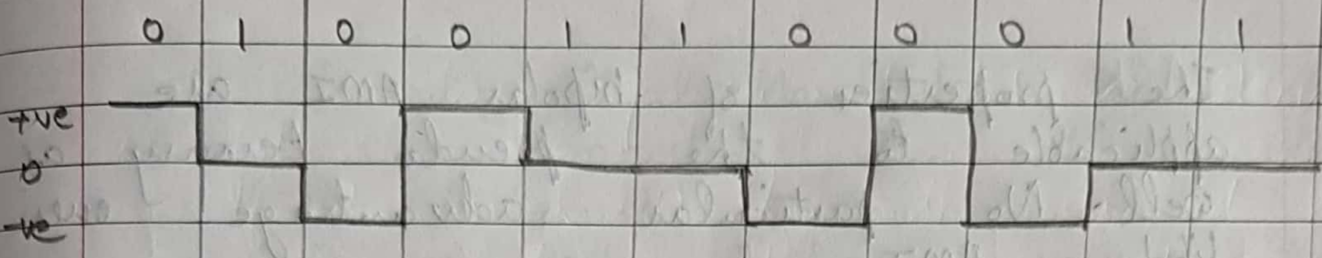


* Pseudo Ternary

1 - no s/g

0 - alternating +ve & -ve signals

Assume preceding 0 → -ve



It is reverse of bipolar AMI.

1 - absence of s/g

0 - alternate polarity (+ve or -ve)

The multilevel binary codes address some of the deficiencies of NRZ codes. These codes use more than 2 s/g levels.

In case of bipolar AMI, a binary 0 is represented by no line s/g & binary 1 is represented by a +ve integer pulse. The binary 1 pulses must alternate in polarity. This code has less bandwidth than NRZ codes but has the advantage that it has no DC component. It also has the advantage that any isolated error can be immediately detected, whether it deletes a pulse or add a pulse, which would be a violation of this property.

In pseudo ternary, the coding is represented by a reversal of bipolar AMI, the 1 being represented by absence of s/g & 0 by alternate polarity in +ve or -ve s/gs.

The properties of bipolar AMI are applicable to the pseudo ternary as well. No particular advantage over bipolar AMI.

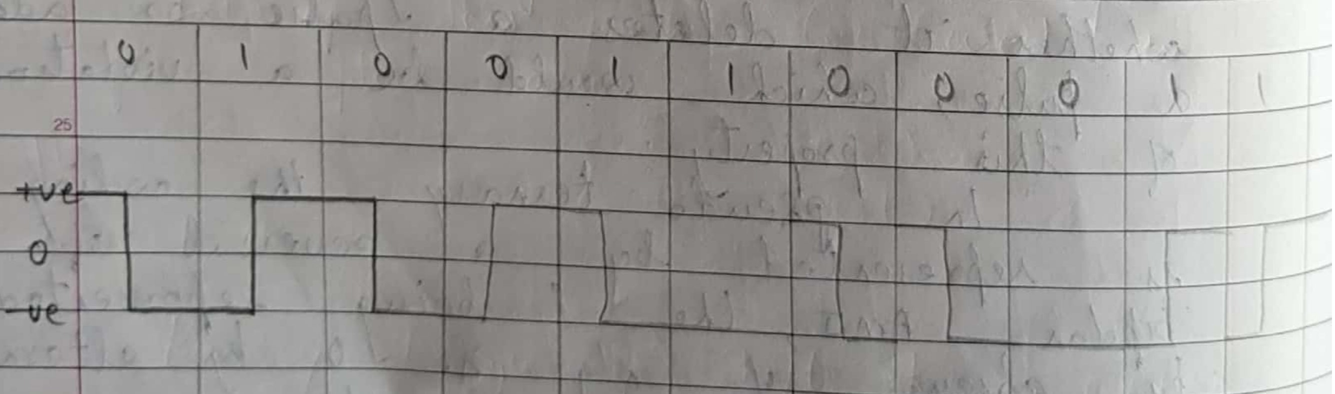
Biphase

→ Manchester → Differential Manchester.

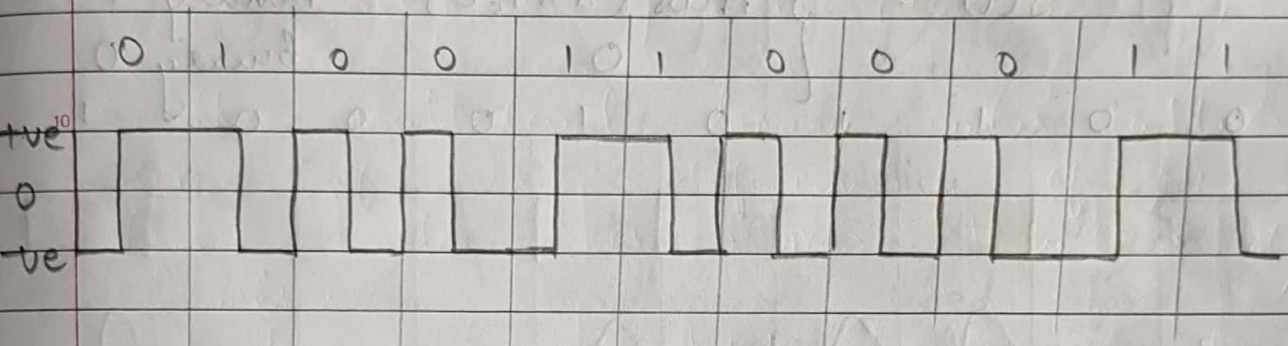
In biphase, another set of coding technique which overcomes the limitations of NRZ.

There are 2 of these Manchester & differential manchester.

In Manchester, there is a transition at the middle of each bit period. 0 is represented by a transition from high to low in the middle of interval & 1 by a transition from low to high in the middle of interval.



In differential Manchester, there is always a transition in the middle of the interval.
0 is represented by a transition at the beg. of the interval & 1 by no transition at the beg. of interval.



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Digital Data, Analog Signals

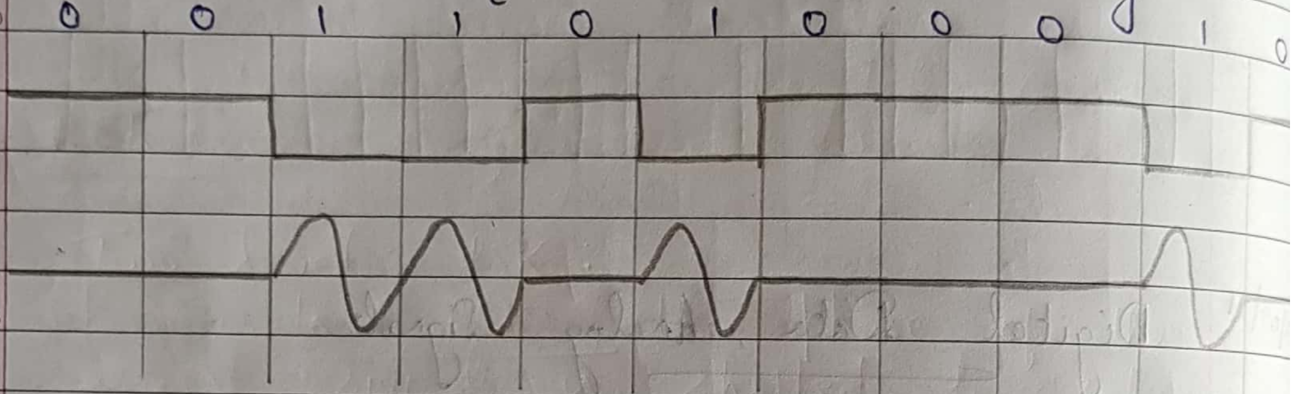
When you have to transmit digital data through a s/m of public telephone network, you have to convert the digital data to an analog slg pattern. A MODEM is used to for digital to analog conversion & back ie A to D conversion. In the above case, a MODEM produce slg in the voice frequency range.

The same technique is used for connecting to microwave frequencies also. The methods used are traditionally amplitude shift keying (ASK), frequency shift keying and phase shift keying.

1. ASK

Binary values 0s & 1s are represented by two different amplitudes of carrier frequency.

$$s(t) = \begin{cases} A\cos(2\pi f_c t) & \text{Binary } 1 \\ 0 & \text{Binary } 0 \end{cases}$$



where carrier signal is $A\cos(2\pi f_c t)$

ASK is susceptible to sudden gain changes and is rather an inefficient modulation technique. The limitation used for voice grade lines is upto 1200 bps. This can be used in fiber optics also.

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2. FSK

Binary Frequency Shift Keying (BFSK)

$$s(t) = \begin{cases} A\cos(2\pi f_1 t) \rightarrow \text{Binary } 0 \\ A\cos(2\pi f_2 t) \rightarrow \text{Binary } 1 \end{cases}$$

The most common use of FSK is binary frequency shift keying in which if two binary values 0 & 1 are represented by two different frequencies near the carrier frequency as shown above.

f_1 & f_2 are typically offset from carrier frequency by equal but opposite amounts. FSK is less susceptible to error than ASK. In voice grade lines, it is used typically upto 1200bps. It is also commonly used for high frequency from 3 to 30MHz. In Multiple FSK (MFSK) in which more than 2 frequencies are used can also be employed which is more bandwidth efficient but is also more prone to error.

In MFSK,

$$s(t) = A\cos(2\pi f_i t) \qquad 1 \leq i \leq M$$

$$f_i = f_c + (2i - 1 - M) f_d$$

f_c : carrier freq.
f_d : different freq.
M : no. of different signal elements

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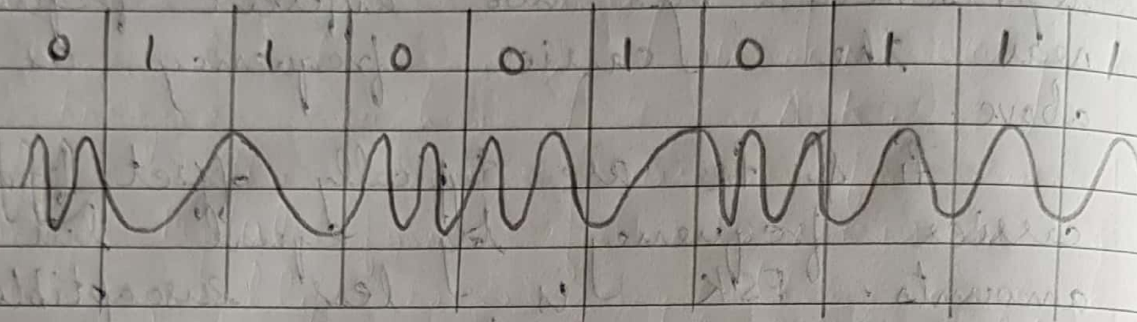
2. ~~BFSK~~

BFSK :



3. PSK

Two - level PSK :

In PSK, the phase of the carrier signal is shifted to represent the data. The simplest scheme ie, two - level PSK uses 2 phases to represent the 2 binary digits & is known Binary PSK (BPSK).

$$s(t) = \begin{cases} A\cos(2\pi ft) \\ A\cos(2\pi ft + \pi) \end{cases}$$

$$= \begin{cases} A\cos(2\pi ft) & \text{- Binary 1} \\ -A\cos(2\pi ft) & \text{- Binary 0} \end{cases}$$

An alternate form of two - level PSK is differential PSK (DPSK) in which a binary 0 is represented by sending a signal burst of the same phase as the previous burst & a binary 1 is represented by sending a s/g burst of opposite phase to the preceding one.

Hence the name DPSK.

The phase shift is with reference to the previous bit transmitted rather than some constant reference signal.

Four-level PSK:

$$s(t) = \begin{cases} A\cos(2\pi f_c + \pi/4) & - 11 \\ A\cos(2\pi f_c + 3\pi/4) & - 01 \\ A\cos(2\pi f_c - 3\pi/4) & - 00 \\ A\cos(2\pi f_c - \pi/4) & - 10 \end{cases}$$

A more efficient way of using the BW can be achieved if each s/g element represents more than 1 bit.

In Quardrature PSK use shifts of odd multiples of $\pi/4$ as is previously shown.

27/09/17

Analog Data, Digital Signalling

– Digitization of analog data

Analog data have to be converted to digital signals & usually it is termed as digitization. This is nothing but an analog to digital conversion.

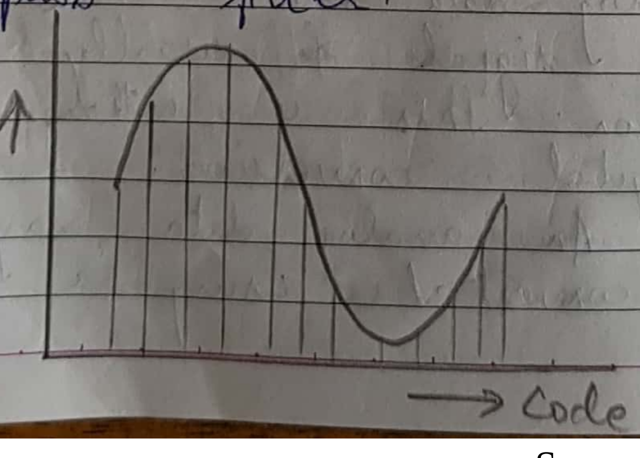
An analog data say voice s/g, can be converted using a digitizer & the net result will be a digital data say for eg: NRZ-L reach at the receiver. If needed to be converted back to an analog sig can be demodulated using MODEM using some technique such as ASK.

The device used for converting Analog to Digital form for transmission & subsequently recovering the original analog data from digital is known as CODEC (Coding Decoding). The technique used here is pulse code modulation.

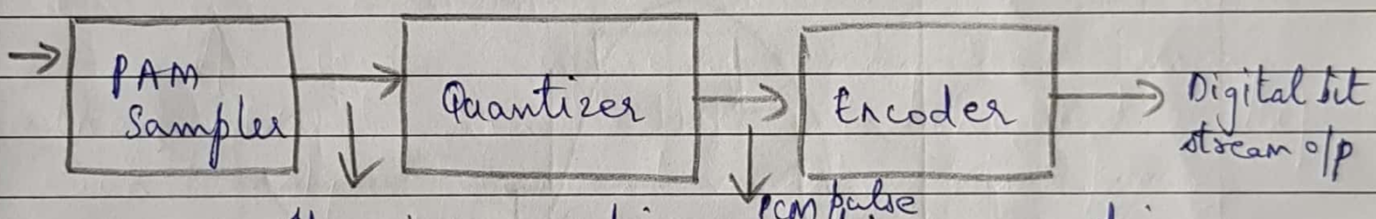
Pulse Code Modulation

The pulse code modulation goes according to the sample theorem. The sample theorem says that if a signal f(t) is sampled at regular intervals of time & at a rate twice the highest signal frequency, then the samples contain all information of the original signal.

The original signal may be reconstructed from the samples using a low pass filter.



The pulse code modulation starts with continuous time, continuous amplitude analog i/p s/gs which a pulse amplitude mode sampler will convert to discrete time, continuous amplitude signals or PAM pulses. which can be quantized using a quantizer & the result will be a discrete time, discrete amplitude signal that is, pulse code modulation pulses. which can be i/p to an encoder & the o/p will be digital bit stream o/p in the required format.



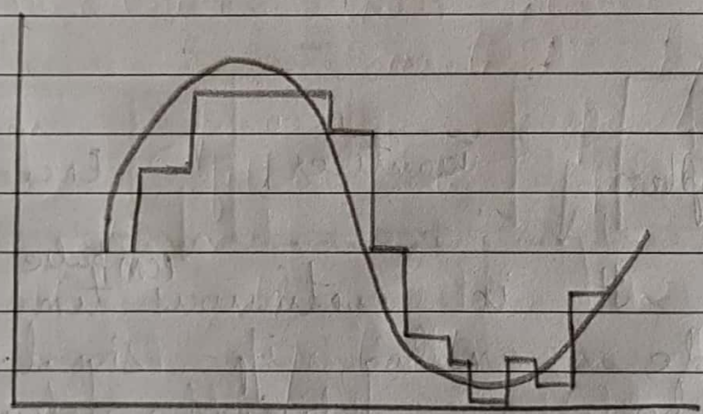
It is continuous time, continuous amplitude analog i/p signal. At this it will be a discrete time, continuous amplitude s/g & quantizer will have o/p discrete time, discrete amplitude signal. Quantizer will have o/p as PCM pulses.

A PCM scheme can be refined using a technique known as non-linear encoding that is effectively the quantization levels are not equally spaced. With equal spacing, the problem occurs that the mean absolute error for each sample is the same regardless of the signal level. As a consequence, the lower amplitude values are relatively more distorted.

By using greater number of quantizing steps for signals of large amplitude, a marked red^n in the overall s/q distortion can be achieved.

The non-linear encoding can significantly improve the PCM SNR ratio.

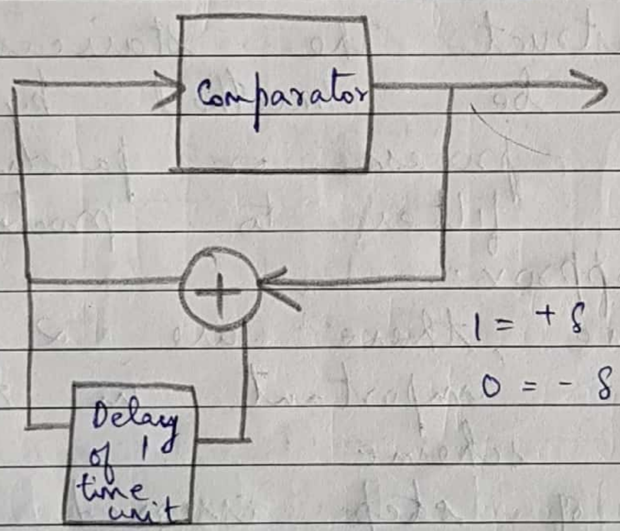
DELTA MODULATION



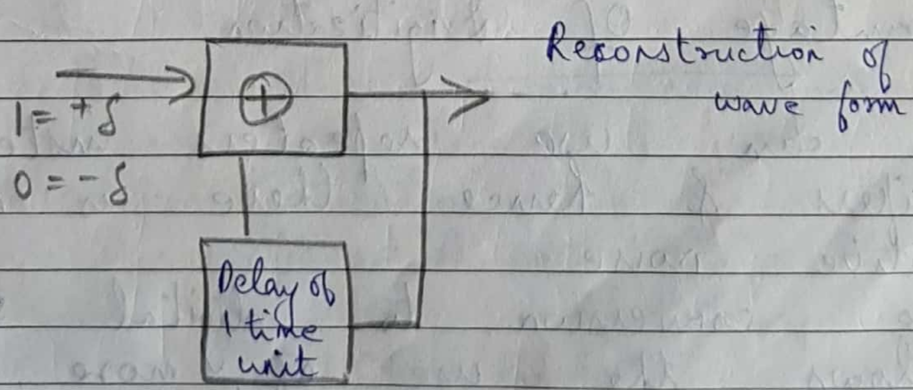
An analog input is approximated by a staircase function that goes up or down by a quantization level delta. At each sampling interval, since the stair case function has binary behaviour, the o/p can be represented by a single binary digit for each sample. That is, a bit stream is produced by approximating the derivatives of the analog s/g rather than its amplitude. A binary 1 is generated if the staircase function is to go up during the next interval & binary 0 if it is otherwise.

The transition that occurs at each sampling interval is chosen so that the stair case function tracks the original analog waveform as closely as possible.

Transmitter



Receiver



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At transmitter, at each sampling time the analog sig i/p is compared to the most recent value of the approximate staircase function.
If the value of sampled waveform exceeds that of staircase function, a binary 1 is generated. Otherwise, a 0 is generated.

The o/p of the delta modulation process is a binary sequence that can be used at the receiver to reconstruct the staircase function. It can be smoothed by some integration process or passing it to low pass filter to produce an analog approx.
Here, there are 2 parameters that are important in the delta modulation scheme
1) Size of step assigned to each binary digit δ.
2) Sampling rate

Advantage Of Digitization

1) We can use repeaters instead of amplifiers & hence there is no additive noise.
2) The conversion to digital signalling allows the use of more efficient digital switching techniques.

4/10/17

Analog Data, Analog Signalling

Reasons for Analog Modulation:

1) In many occasion you need higher frequencies for analog transmission. So you will have to modulate it with a carrier wave.

In wireless comm., it is almost impossible to transmit the base band signals with the practical antennas.

2) The modulation makes it possible for frequency division multiplexing. This is a very effective way of transmission.

Amplitude Modulation

The signal $s(t) = [1 + n_a x(t)] \cos 2\pi f t$ where the $\cos 2\pi f t$ is the carrier, $x(t)$ is the input signal and both are normalized to unit amplitude, n_a is the modulation index which is the ratio of the amplitude of i/p signal to the carrier. This scheme is known as 'double side band transmitted carrier' (DSBTC)

The other techniques are frequency and phase modulation. Both are variations of angle modulation where

$$s(t) = A_c \cos [2\pi f t + \phi(t)]$$

and for phase modulation

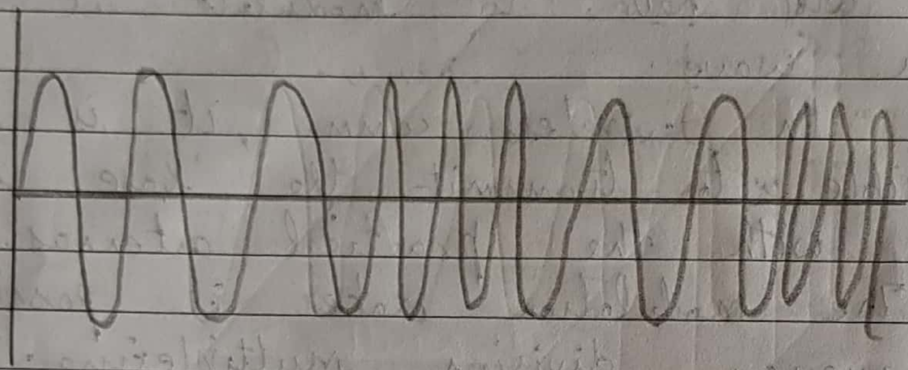
$$\phi(t) = n_p m(t)$$

where n_p is the phase modulation index. For frequency modulation

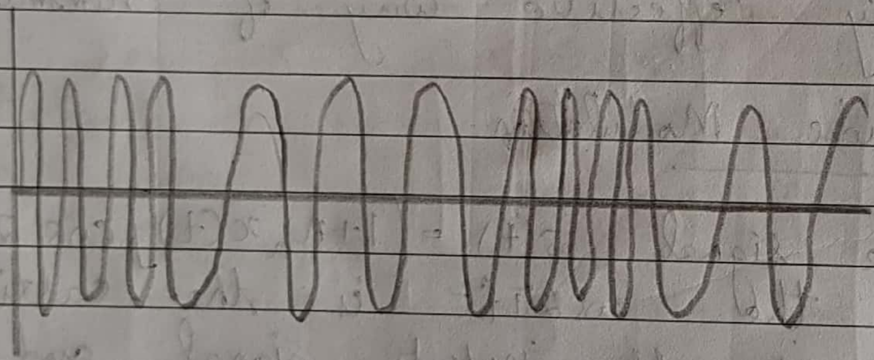
$$\phi'(t) = n_f\, m(t)$$

where n_f is the frequency modulation

Phase modulation



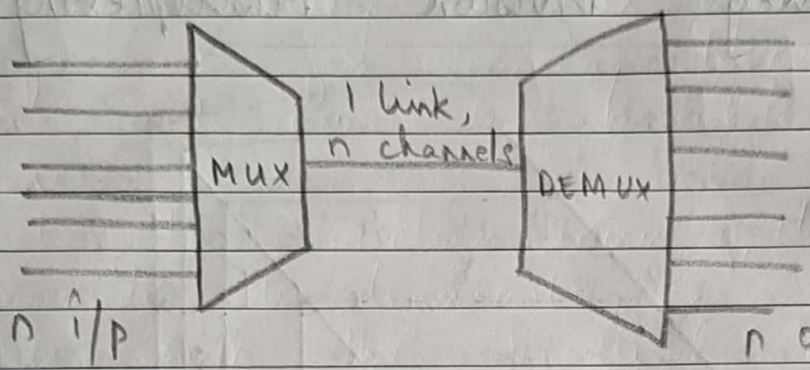
Frequency modulation



It is difficult to distinguish PM and FM without the knowledge of freq. function. Just as in AM, both FM and PM result in a signal whose bandwidth is centered at the carrier frequency.

Module - IV

Multiplexing



With two devices, connected by a point-to-point link, care should be taken so that the data link will not become a ~~burden~~ bottle neck between the two stations.

For efficiency, it should be always possible to share the capacity of the data link.

A generic term for such sharing is called as multiplexing. It is commonly applied between trunks or long-haul networks in high capacity & also in fiber, coaxial or microwave links.

It is used because higher the data rate, the more cost effective the transmission facility will become.

ie, ~~the~~ for a given distance the cost per kbps declines with an increase in data rate of transmission facilities.

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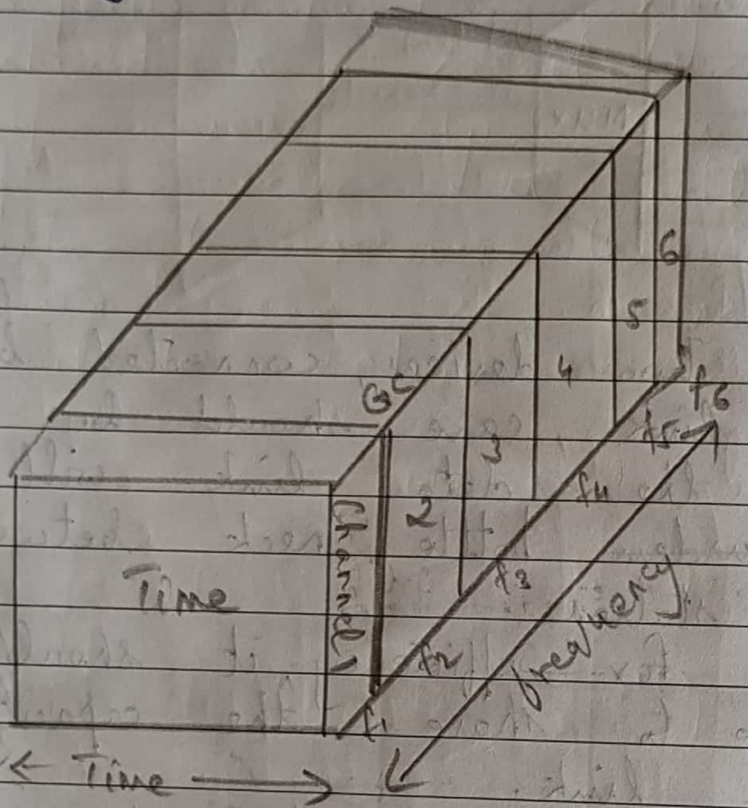
In most cases the individual data commⁿ devices require relatively modest data rate supports.

Frequency Division Multiplexing (FDM)

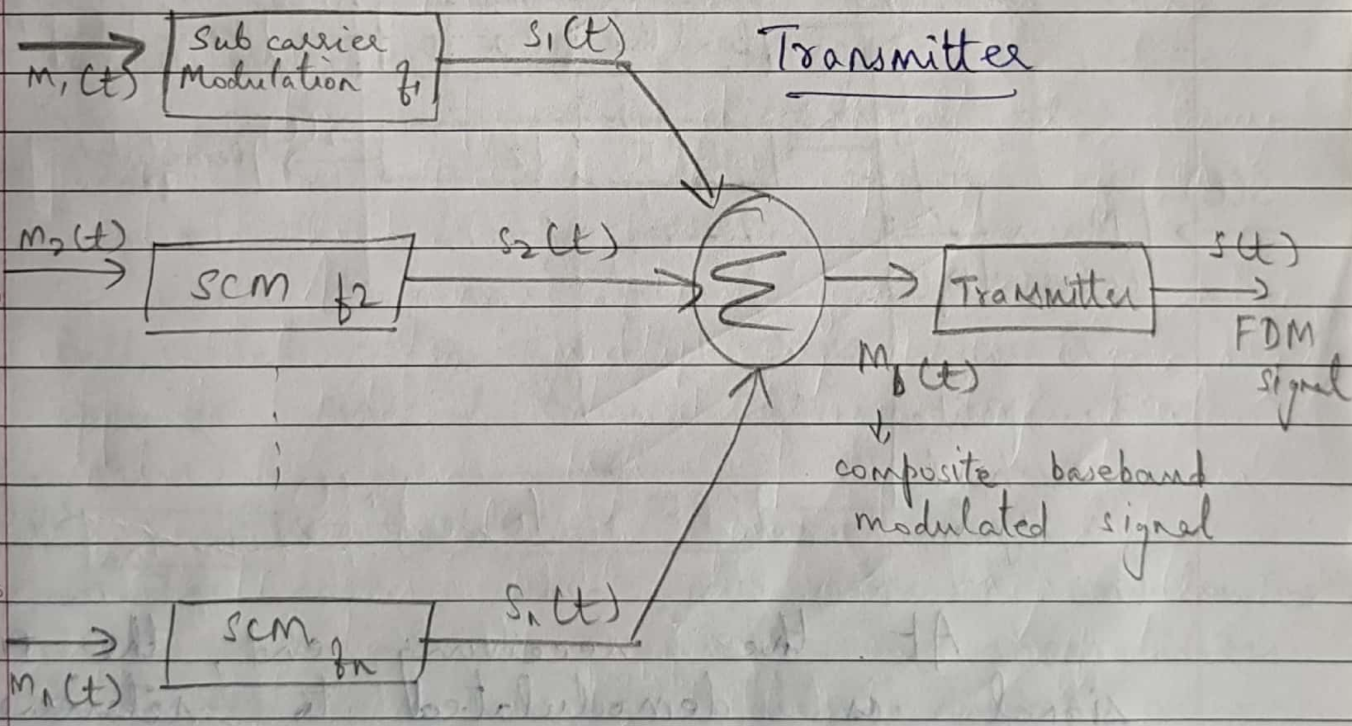


GC : Guard channel

FDM is possible when the useful bandwidth of the transmission medium exceeds the required bandwidth of the signal to be transmitted.

~~The~~ A number of signals can be carried simultaneously if each signal is modulated into a different carrier f. and if the carrier freq. are sufficiently separated so that the BWs of the signals do not significantly overlap. As shown in the figure, 6 signal sources can be fed to

a MUX which modulates each signal to a different frequency from ($f_1, f_2, \ldots f_6$).
Each signal will require a certain B/W centered on its carrier frequency referred to as a channel.
To prevent interference, channels are separated by Guard bands which are unused portions of the spectrum.
The composite sig. is transmitted across the medium. It could be either digital or analog. If it is D, the conversions to A should be done using modems.

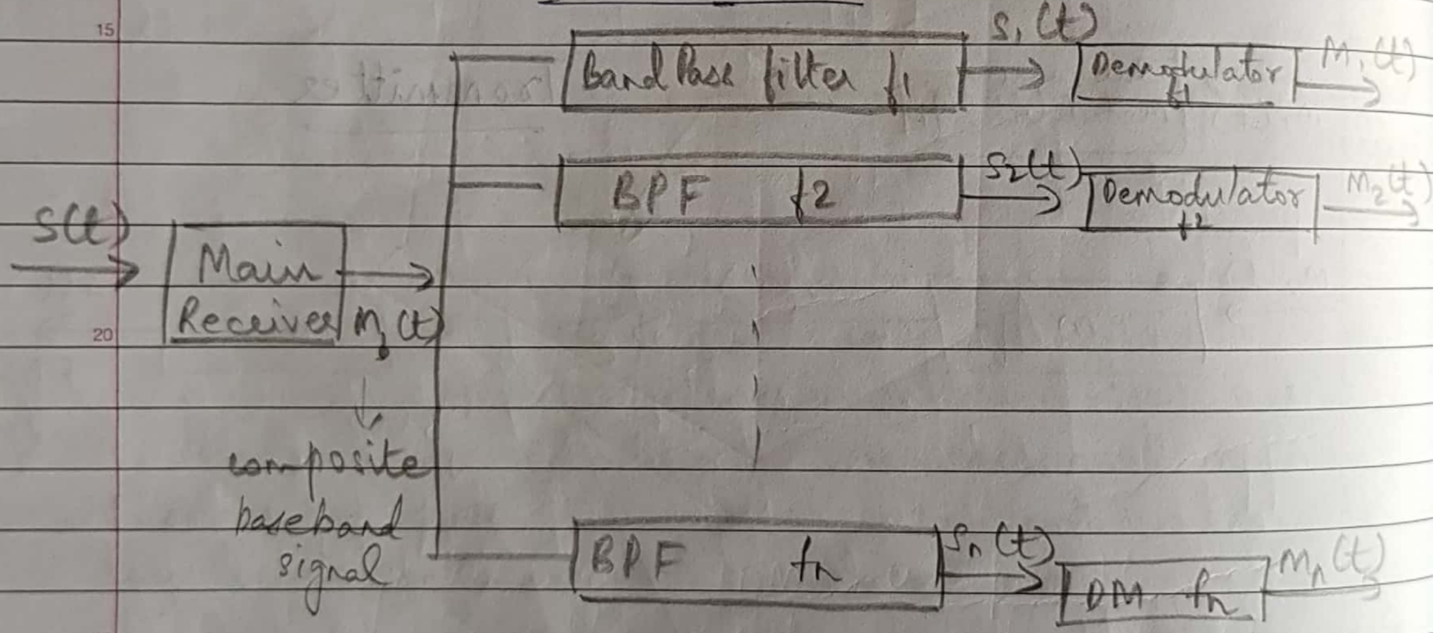


A number of analog signals are to be multiplexed from $m_1(t), m_2(t) \ldots m_n(t)$ to the same transmission medium.
Each signal $m_i(t)$ is modulated

into a carrier frequency f_i & each is referred to as a sub carrier.

The resulting analog modulated signals are then summed to produce a composite baseband signal $M_b(t)$. f_i should be selected so that BWs of various signals do not overlap or else it will be impossible to recover the original signal. This signal can be transmitted over a suitable medium.

Receiver Side



At the receiving end, the FDM signal is demodulated to retrieve the $M_b(t)$ which is passed through n band pass filters, each filter centered around f_i to produce the individual signals $m_1(t)$, $m_2(t)$, ... $m_n(t)$. As it is evident, this multiplexing technique has 2 problems to face :

1) Cross talk between the adjacent frequency channels.
Sufficiently part guard bands should prevent it to a good extend

2) The other problem is intermodulation noise

6/10/17

Wavelength Division Multiplexing

It is actually another form of frequency division MUXing. The true potential of optical fiber was fully exploited only when multiple beams of light at different freq. could be transmitted on the same fiber. This is commonly known as wavelength division MUXing.

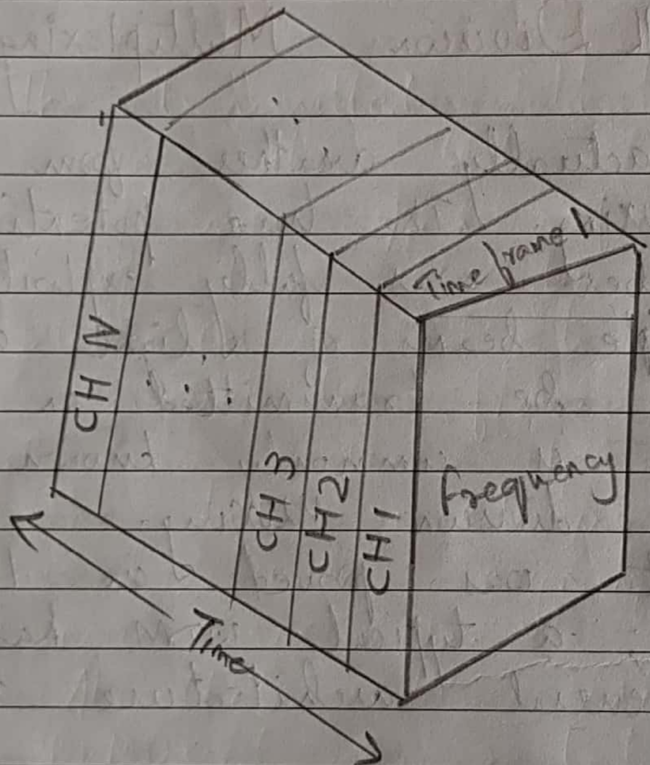
It was proved in 1997 at Bell Lab ; a typical WDM has the same general architecture as FDM systems.

A number of sources which generate laser beams at different wavelengths are sent to a MUX, which consolidates the sources for transmission over a single fiber line.

Optical amplifiers which are spaced typically 10km apart amplify the wavelengths simultaneously.

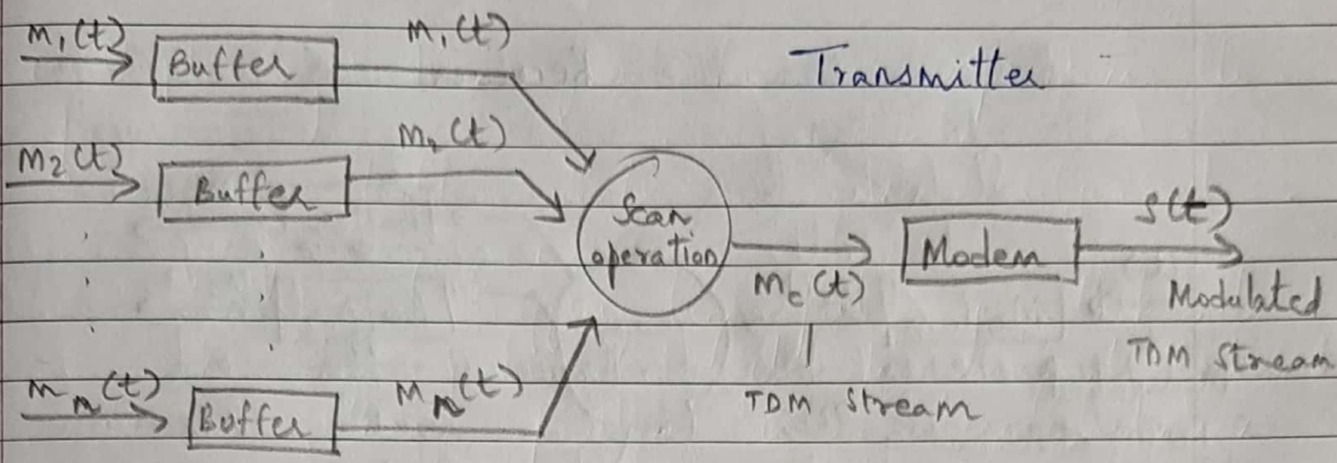
At the receiving end, the composite s/gs will be DEMUXed where the component channels are separated and send to receivers at destination point. Most WDM operate in the 1550nm range & use a 50GHz spacing.

Synchronous Time Division Multiplexing (TDM)



Synchronous TDM is possible when the achievable data rate of the medium exceeds the data rate of the digital signals to be transmitted. Multiple signals can be carried on a single transmission path by interleaving portions of each signal in time.

The interleaving can be done either either at bit level or block level (as bytes of characters).

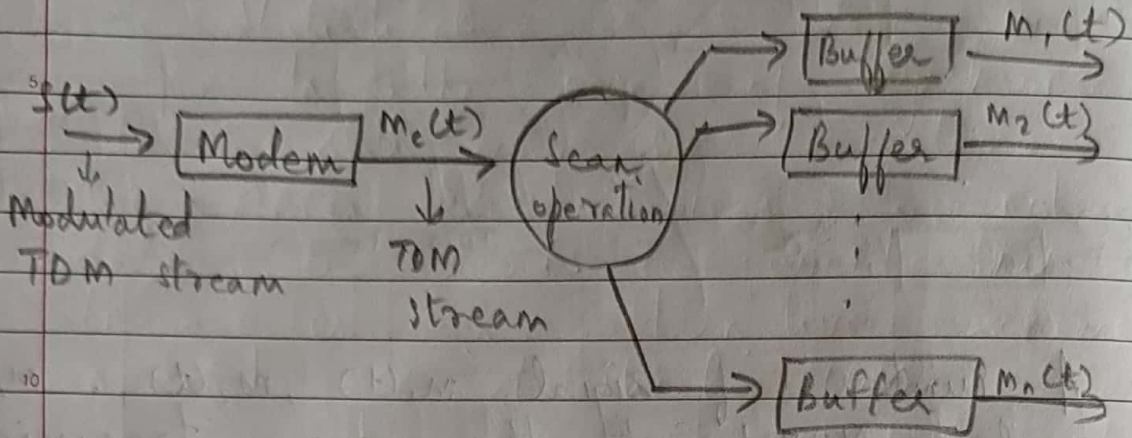


A number of signal $m_1(t), m_2(t), \ldots m_n(t)$ can be MUXed into the same transmission medium as shown in the figure. The incoming data from each source are briefly buffered. Each buffer is typically one bit or one character in length. The buffers are scanned sequentially to form a composite data stream $m_c(t)$. The scan operation is sufficiently rapid so that each buffer is emptied before more data can arrive.

Thus, the data rate of $m_c(t)$ must be atleast equal to the sum of the data rate of $m_i(t)$, $i = 1, 2, \ldots n$.

The digital s/g $m_c(t)$ may be transmitted directly or passed through a modem if you require an analog s/g to be transmitted.

Receiver



The transmitted data may have a format ie; the data are organized into frames. Each frame contains a cycle of time slots. The sequence of slots dedicated to each source from frame to frame is called a channel.

The byte interleaving technique can be used with asynchronous & synchronous sources.

At the receiver, the interleved data are DEMUXed & routed to the appropriate destination BUFFER.

For each i/p source $M_i(t)$, there is an identical output destination which will receive o/p data at the same rate at which it was generated.

The word synchronous is used because the time slots are pre-assigned to source & are fixed.

The time slot for each source is transmitted whether or not the source has data to send.

09/10/17

* Pulse Stuffing

~~In pulse stuffing~~, In synchronous time division MUXing if each source has a separate clock, any variation among the clocks could cost loss of synchronization. Also, in certain cases, the data rates of i/p data strings are not related by a simple rational number.

For both the problems, the remedial measure is a technique known as pulse stuffing. With pulse stuffing, the outgoing data rate of MUX excluding the frame bits, will be higher than the sum of maximum instantaneous incoming rates. The extra capacity is used by stuffing dummy waves or pulses into each incoming signal until it's rate is raised to that of a locally generated clock signal.

The stuffed pulses are inserted at fixed locations in the MUX frame format so that they can be identified & removed at the DEMUX.

Statistical TDM

~~Statistical~~

In the case of synchronous TDM, the slots may be sometimes wasted, when certain devices do not have data to send. To avoid this wastage, there is an alternative scheme named as 'statistical' TDM.

This uses the property of data transmission by dynamically allocating time slots on demand. Even if there are 'n' i/p lines, only 'k' time slots are available on the TDM frame where $k < n$.

For i/p, the function of the MUX is to scan the i/p buffers, collect the data until a frame is filled and then send the frame.

On the o/p, the MUX receives a frame & distributes the slots of data to the o/p buffers.

Since, in this scheme, all the attached devices are not transmitting all the time, the data rate of the multiplexed line is less than the sum of the data rates of the attached devices.

Thus, in a statistical TDM can use a lower data rate to support as many devices as a synchronous MUX. Or in other words, statistical TDM can support more devices with the same data rate as compared to synchronous TDM.

Since data arrive from and are distributed to I/o lines in an unpredictable manner, address info. is ~~refer~~ required to assure proper delivery.

Hence there is more overhead per slot for statistical TDM since the slot has to carry address as well as data.

The frame structure used by statistical MUX has impact on performance. Usually statistical TDM use synchronous protocols such as HDLC frames.

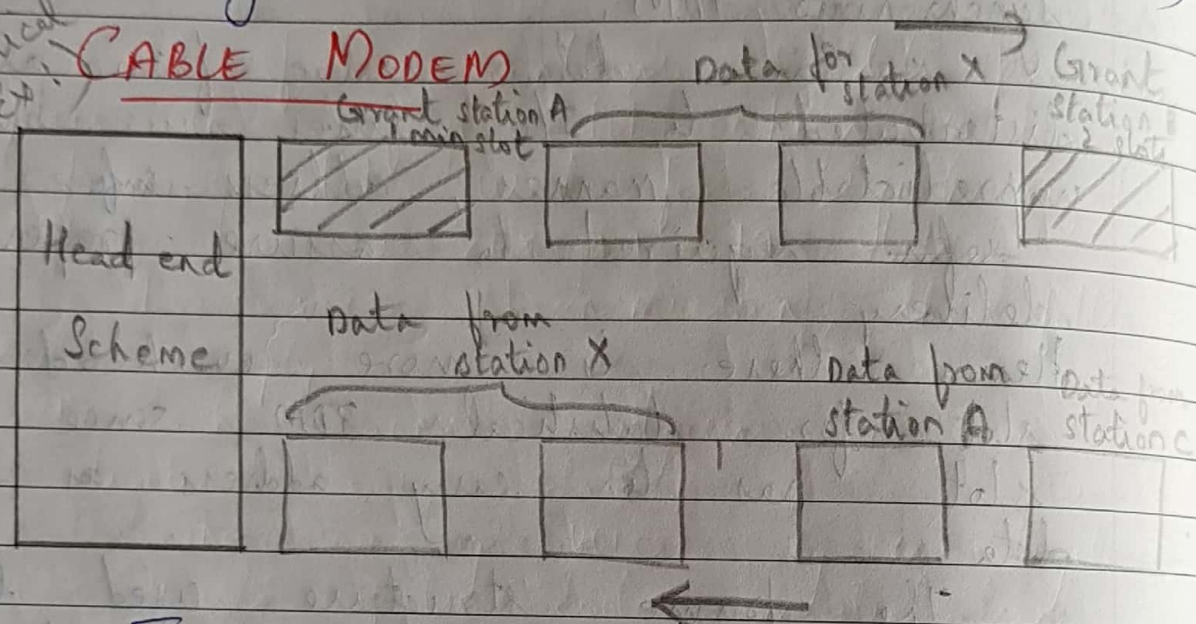
In this the source is identified by the address the length of data frame is available & its end is marked by the end of overall frame.

This scheme works well at light load but is inefficient when the load is heavy.

To improve efficiency, we can allow multiple data sources to be packaged in single frame but then, some means have to be deviced to specify length of data for each source.

Thus, statistical TDM sub frame consists of a sequence of data ~~frames~~ fields each labelled with address & length (Frame formats)

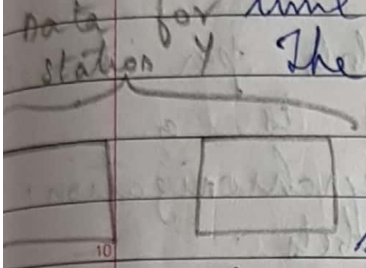
CABLE MODEM (Statistical TDM Ex)

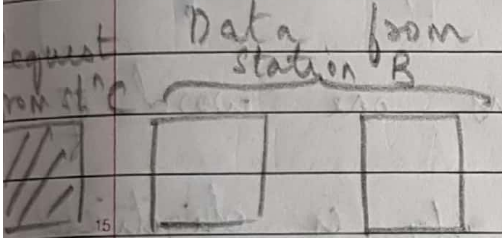


To support data transfer to and from a cable modem, a cable TV provider dedicates two channels one for transmission in each direction. Each channel is shared by a number of subscribers & there is an allocating scheme for transmission.

A form of statistical TDM is used from the head end to the subscriber the schedular delivers data in the form of small packets. If more than 1 subscriber is active, each subscriber only a fraction of the down stream capacity. The down stream is also used to grant the time slots to the subscriber.

When subscriber has data to transmit, it must first request the time slot on the shared upstream channel. Each subscriber is given dedicated time slots for the request purpose. The head end schedular responds to a request packet by sending back an assignment of future time slots to be used by the subscriber. Thus, a number of subscribers can share the same up-stream channel without conflict.





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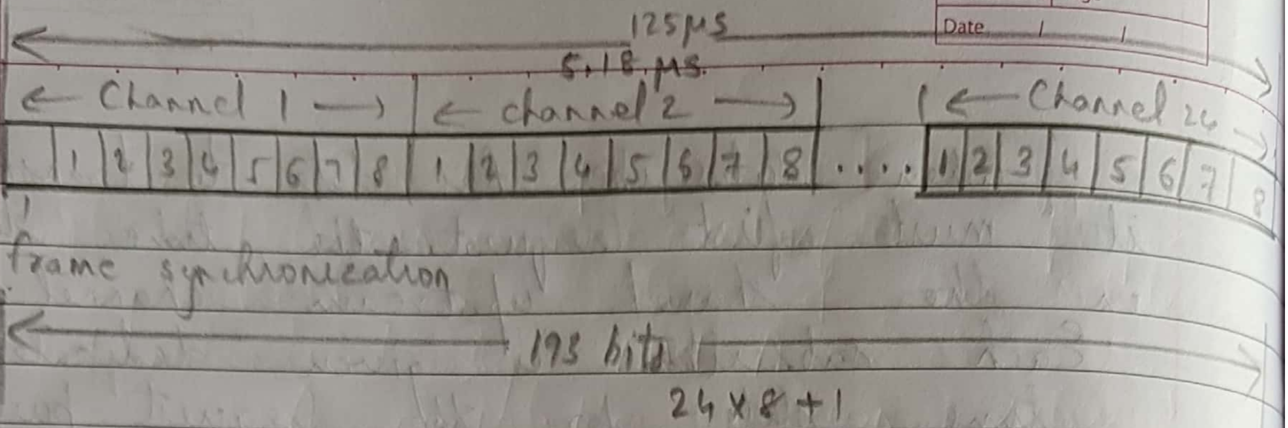
Digital Carriager System

DCS was first used in US for long distance communication for voice grade signals over high capacity transmission links such as optical fiber, coaxial cable & mwave.

This happened in the time when telecommⁿ N/w where passing on to digital technology by adopting synchronous TDM transmission structure.

Later, identical hierarchy had been adopted under ITUT.

The basis of TDM hierarchy is DS-1 transmission format which MUXes 24 channels.



Each frame contains 8 bits per channel, plus the first bit which is a framing bit used for synchronization. So $24 \times 8 + 1 = 193$ bits.

For voice transmission, the following rules apply:

1) Each channel contains one word of digitized voice data.

The analog voice sig is digitized using pulse code modulation at the rate of 8000 samples/sec with a frame length of 193 bits

$8000 \times 193 = 1.544$ M bps.

For 5 of every 6 frames, 8 bit PCM samples are used. For every 6^{th} frame, each channel contains a 7 bit PCM word + a signalling bit. The signalling bits form a stream for each voice channel, which contains network control & routing information.

The same DS-1 format is used to provide digital service also. The same data rate is used. In this case, 23 channels of data are provided. The 24^{th}

channel position is reserved for special sync. byte which allows faster & more allowable reframing in case of a framing error.

Within each channel 7 bits per frame are used for data. The 8th bit is used to indicate whether a channel contains user or s/m control data. A data rate of 56kbps can be provided per channel.

Lower data rates are provided by sub-rate MUXing. For this technique, an additional bit is roped from each channel to indicate which subrate MUXing rate is being provided.

DS-1 format can be used to carry a mixture of voice & data channels. In this case, all 24 channels are used. No sync byte is provided.

For higher level MUXing, it is achieved by interleving bits from DS-1 inputs.

SONET / SDH

Synchronous Optical Network : SONET
Synchronous Digital Hierarchy : SDH

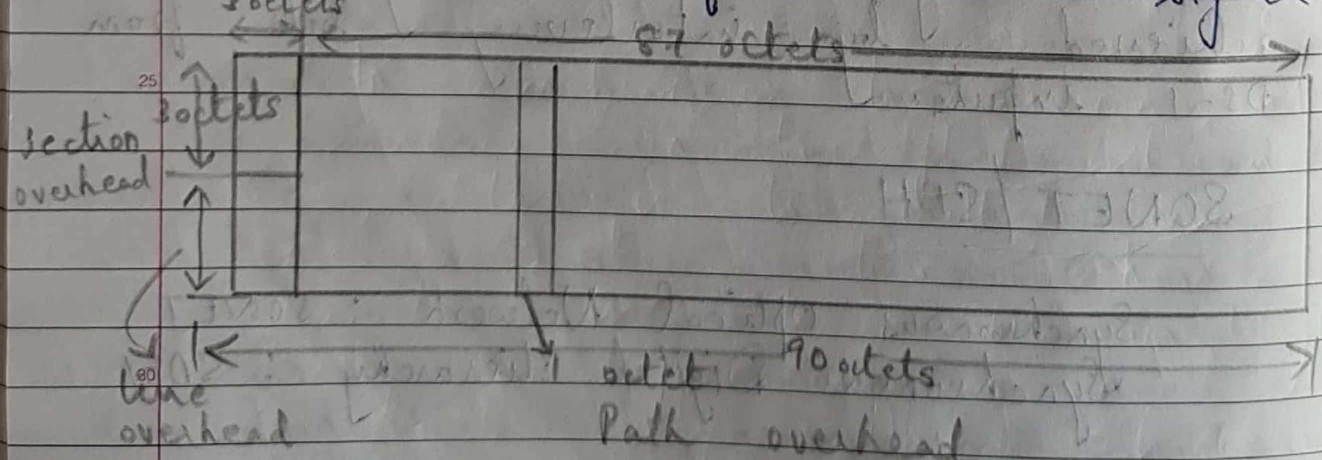
The SONET specification is for an optical transmission interface proposed by Bell Core Co. & was later standardised by ANSI · A.

A compatible version refered to as synchronous digital hierarchy has been published by ITU-T.

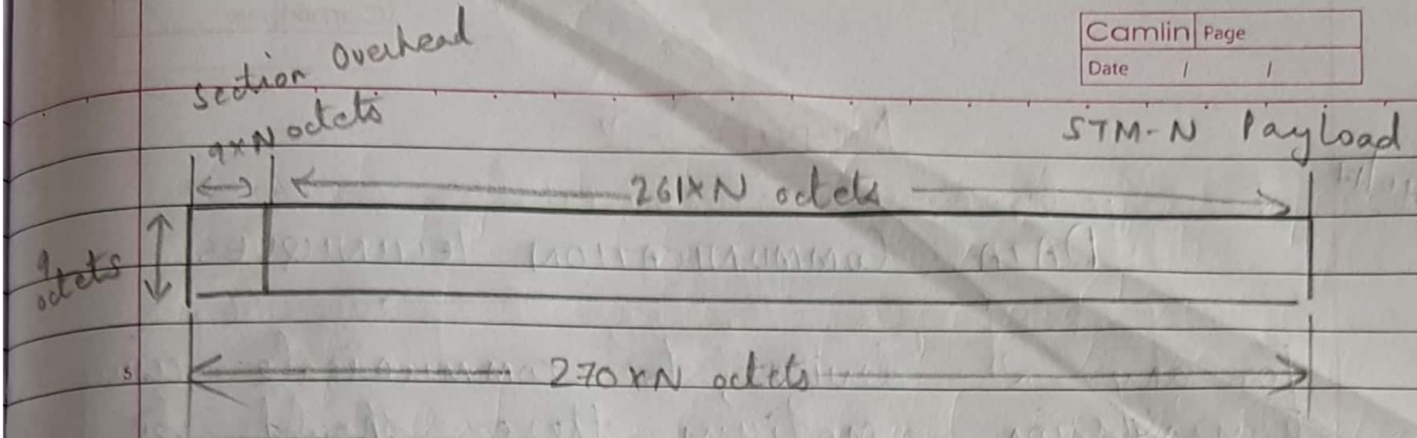
SONET is intended to provide specification for high speed digital transmission through optical fiber.

SONET hierarchy

The SONET defines a hierarchy of standardised digital data rates. At the lowest level, refered as STS-1 (synchronous transport signal) or OC-1 (optical carrier level) is 51.84 Mbps. This is used for the lower rate signals such as DS-1C, DS-2 etc. The multiple STS-1 signals can be combined to form STS-N signal



STS-1 Frame format.



STM - N format

The frame format

The basic SONET building block is STS-1 format. It consists of 810 octets & is transmitted once in every 125 µs. with an overall overhead of 51.84Mbps The first three columns of the frames are devoted to overhead octets. Overhead octets are section, line & path overhead. The remainder of the frame is payload. The payload includes a column of path overhead, line overhead which contains a pointer where the path overhead starts.

MODULE - V

23/10/17

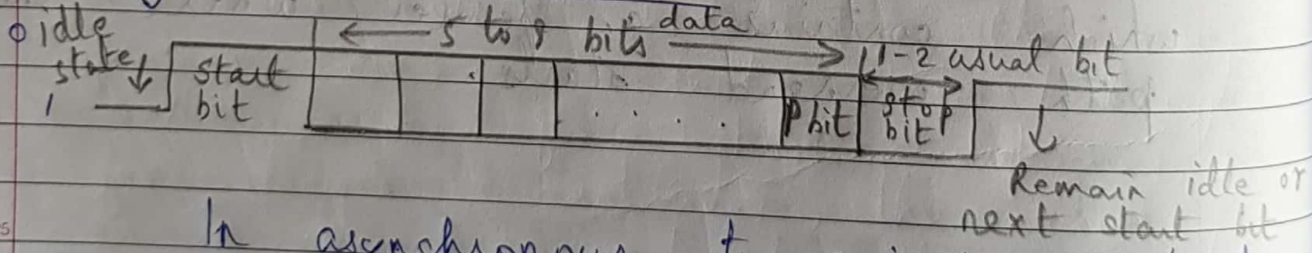
DATA COMMUNICATION TECHNIQUES

For any 2 devices, to communicate using a transmission medium, there should be some agreement to exchange data. The timing ie; the rate, duration & spacing of bits must be same for the transmitter & receiver.

The two common methods usually employed are synchronous & asynchronous data communication.

The number of errors that seeps in is also another problem. The error rate has to be limited so that there is effective communication between transmitter & receiver.

Asynchronous Communication



In asynchronous transmission, instead of sending un-interrupted streams of data, data is transmitted one character at a time where each character is 5 to 8 bits in length. Timing or synchronisation need be maintained only within each character & the receiver has an opportunity to resynchronise at the beginning of each new character.

When no character is being send or transmitted, the transmission line remains in an idle state.

Suppose, we are using NRZ-L signalling method which is commod for asynchronous transmission. The idle state would be the presence of a -ve voltage on the line.

The beginning of the character is signalled by a start bit with a value of binary 0. This is followed by 5 to 8 bits that actually make the character. The bits of character are transmitted beginning with the least significant bit. The parity bit is send as the last bit. Either an even parity or odd parity is used.

There could be framing errors due to noise conditions or cumulation of error bits.

Asynchronous transmission is cheap & simple but has an overhead of 20% due to the presence of start & end bits which doesn't contain any information.

Synchronous Communication

8 bit flag	Control fields	Data // Fields //	Control fields	8 bit flag
↑ Preamble				↑ Postamble

Synchronous Frame

With synchronous transmission, bits are transmitted in a steady stream without start or stop.

The blocks may be may bits in length. To prevent timing drift b/w transmitter & receiver, their clocks must be synchronized.

One method is to provide a separate clock line b/w transmitter & receiver. One of the sides would pulse the line regularly with one short pulse per bit time.

The other side uses these regular pulses as a clock. This technique will work for short distances, but for longer distances, the clock pulses are also subject to the same impairments as data & timing error can occur.

The other alternative is to embed the clocking info. in the data s/g itself.

With synchronous transmission, there is another level of synchronization required. To allow the receiver to determine the beginning & end of the block of data.

Each block of data begins with a preamble bit pattern & ends with a postamble bit pattern.

In addition, other bits are added to block which convey control info. used in the data link layer.

The data along with preamble, postamble & control info. together is called a frame.

The exact format of frame depends on the data link control procedure used.

Typically, the frame starts with a preamble called flag which is 8 bits in length. The same flag is used as postamble.

The receiver looks for the occurance of flag pattern to sig the start of the frame which is followed by control fields, data fields, more control fields, finally the postamble flag.

For sizable blocks of data, synchronous transmission is more effective than asynchronous transmission.

Asynchronous transmission requires 20% or more overhead.

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Parity Check

There is even and odd parity check. Uses the simplest error detection scheme. It can detect a single error. Even parity is usually used for synchronous transmission & odd for asynchronous.

The use of parity bit is not full proof since noise impulses are often long enough to destroy more than 1 bit especially at higher data rates.

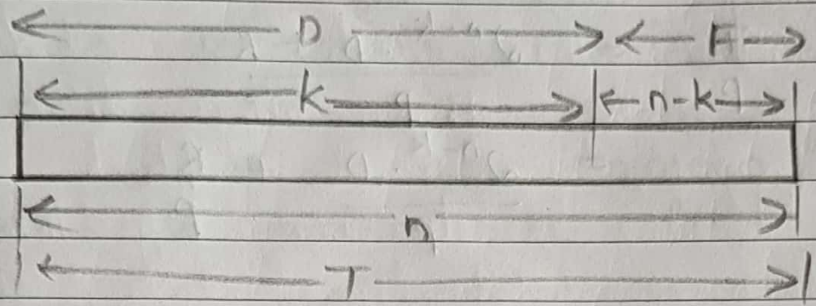
Cyclic Redundancy Check (CRC)

Given a 'k' bit block of bits or message. The transmitter generates an (n-k) bit sequence as frame check sequence (FCS) so that the resulting frame which consist of 'n' bits is exactly divisible by some pre-determined number.

The receiver then divides the incoming frame by that number, and if there is no remainder, assumes that there is no error.

We define T to be an n-bit frame that is to be transmitted. D is a k-bit block of data or message, that is the first k bits of T.

F is the (n-k) bit FCS, that is, the last (n-k) bits of T.



P is the pattern of n-k+1 bits which is the pre-determined divisor.
We would like to have that T/P has no remainder (provided message is error free). It is clear that T is

$$T = 2^{n-k} . D + F$$

Effectively, padding D with zeroes & adding F effectively concatenating D & F.
We want T to be exactly divisible by P. Suppose, we divide 2^{n-k} D by P.

$$\frac{2^{n-k} \cdot D}{P} = Q + \frac{R}{P} \quad \text{—} \quad (1)$$

There is a quotient & a remainder. Since the division is modulo 2, the remainder is always atleast 1 bit shorter than divisor. Now, we will use this remainder as our FCS.

$$T = 2^{n-k} . D + R$$

Let us see whether the T/P has no remainder.

$$\frac{T}{P} = \frac{2^{n-k} \cdot D + R}{P}$$

$$= \frac{2^{n-k} \cdot D}{P} + \frac{R}{P}$$

$$= Q + \frac{R}{P} + \frac{R}{P}$$

$$= Q + 0$$

$$= Q$$

From eqⁿ ①, $\frac{2^{n-k} \cdot D}{P} = Q + \frac{R}{P}$.

Here, there is no remainder & T is exactly divisible by P.
Thus, this FCS is easily generated by $\frac{2^{n-k} D}{P}$ and use the $(n-k)$ bit remainder as FCS. On reception, the receiver will divide T by P and if it gets no remainder, it concludes that there is no error, else it shows there is error.

The pattern P is chosen to be 1 bit longer than desired FCS & the exact bit pattern chosen depends on the type of error detected.

Eg: D = 1010001101 (10 bits)
P = 110101 (6 bits)
n = 15, k = 10, n-k = 5
The message is multiplied by 2^5.
Padding with 5 zeroes
101000110100000
This is divided by P

```
          1100010 1
110101 | 101000110100000
         110101
         011101 01
          110101
          00011000 10 0
             110101
             0011101010
               110101
               0001110
```

Q = 1101010110
R = 01110
~~Q+R= [illegible]~~
If there is no error, the received frame should exactly be divided by P.

For a given frame of bits, additional bits that can constitute an error detecting code are added by the transmitter. This code is calculated as a function of the other transmitted bits. Typically, for a data block of k bits, the error detection algorithm leaves an error detection code of $(n-k)$ bits where $(n-k) < k$.

The error detection code, which is referred as check bits is appended to the data block to produce a frame of n bits which is then transmitted. The receiver separates the incoming frame into k bits of data & $(n-k)$ bits of error detection code. The receiver performs the same error detecting calculation ~~&~~ on the database & compares this value with the value of incoming error detection code. There is an error detected only if there is a mismatch.

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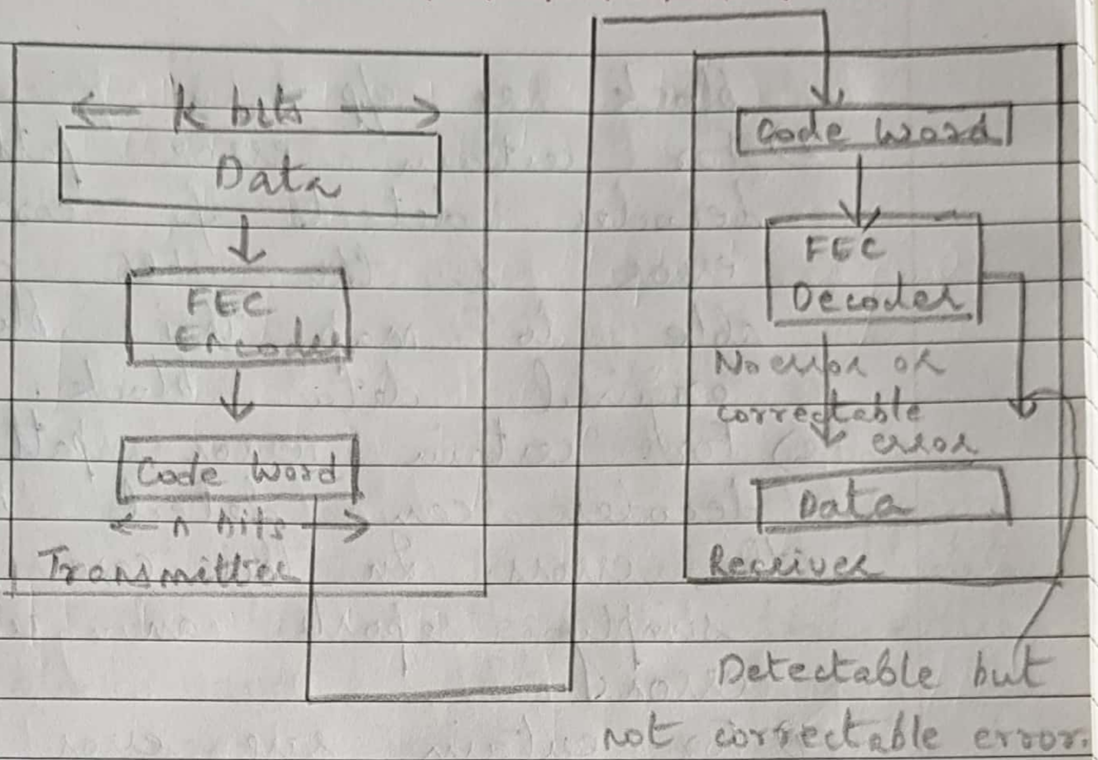
ERROR CORRECTION

The figure shows a general method for correction of codes with forward error correction (FEC) method.



On the transmission end, each k bit block of data is mapped into an n-bit block (where $n > k$.) called a code word using FEC Encoder. The code word is then transmitted.

During transmission, the signal is subject to impairments which may produce bit errors in the signal.

At the receiver, the incoming signal is demodulated to produce a bit string which is similar to the original code word but may contain errors. This block is passed through an FEC decoder with the following possible results

1) If there are no bit errors, the i/p to the FEC decoder is identical to the original code word & the decoder produce this original data

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block as o/p.

2) For certain error patterns, the decoder detects & corrects the error & the FEC decoder is able to map this block to the original data block.

3) For certain error pattern, the decoder can detect but not correct the errors. In this case, decoder simply reports an uncorrectable error.

4) For certain rare error patterns, the decoder do not detect any error & maps the incoming 'n' bit data block to a 'k' bit block which differs from the original k bit block.

In general, the FEC algorithm takes a k bit block as input & adds (n-k) check bits to that block & produce an n-bit block. These extra bits helps to correct any bit errors which might have come across during transmission.

HAMMING DISTANCE

The hamming distance $d(v_1, v_2)$ b/w two n bit binary sequences v_1 and v_2 is the number of bits in which v_1 & v_2 disagree.

For eg: $v_1 = 011011$
$v_2 = 110001$
$d(v_1, v_2) = 3$

Let us consider a block code technique for error correction. Suppose we wish to transmit blocks of data of length 'k' bits. Instead of transmitting each block as k bits, we map the k bit sequence into a unique n bit code word.

$k = 2$ $n = 5$

Data block	Code word
00	00000
01	00111
10	11001
11	11110

Received : 00100

$d(00100, 00000) = 1$
$d(00100, 00111) = 2$
$d(00100, 11001) = 4$
$d(00100, 11110) = 3$

From the example, the transmitted message 00100 has a minimum distance with first code word & it is corrected as the first code word.

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But for this to take place, there should be only one minimum distance valid code word, else the correction is not possible.

The ratio of redundant bits $(n-k)/k$ is called the redundancy of the code & the ratio of data bits to the total bits k/n is called the code rate.

For a code consisting of the code words $w_1, w_2, \ldots w_s$ where $s = 2^k$, the minimum distance d_{min} of the code is defined as

$$d_{min} = \min_{i \neq j} [w_i, w_j]$$

It can be shown that for any given +ve integer t, if a code satisfies

$$d_{min} \geq 2t + 1$$

then the code will be able to correct all errors upto and including t bits of errors. If $d_{min} \geq 2t$ then all errors less than or equal to $(t-1)$ bits can be corrected, errors of t bits can be detected but need not always be corrected.

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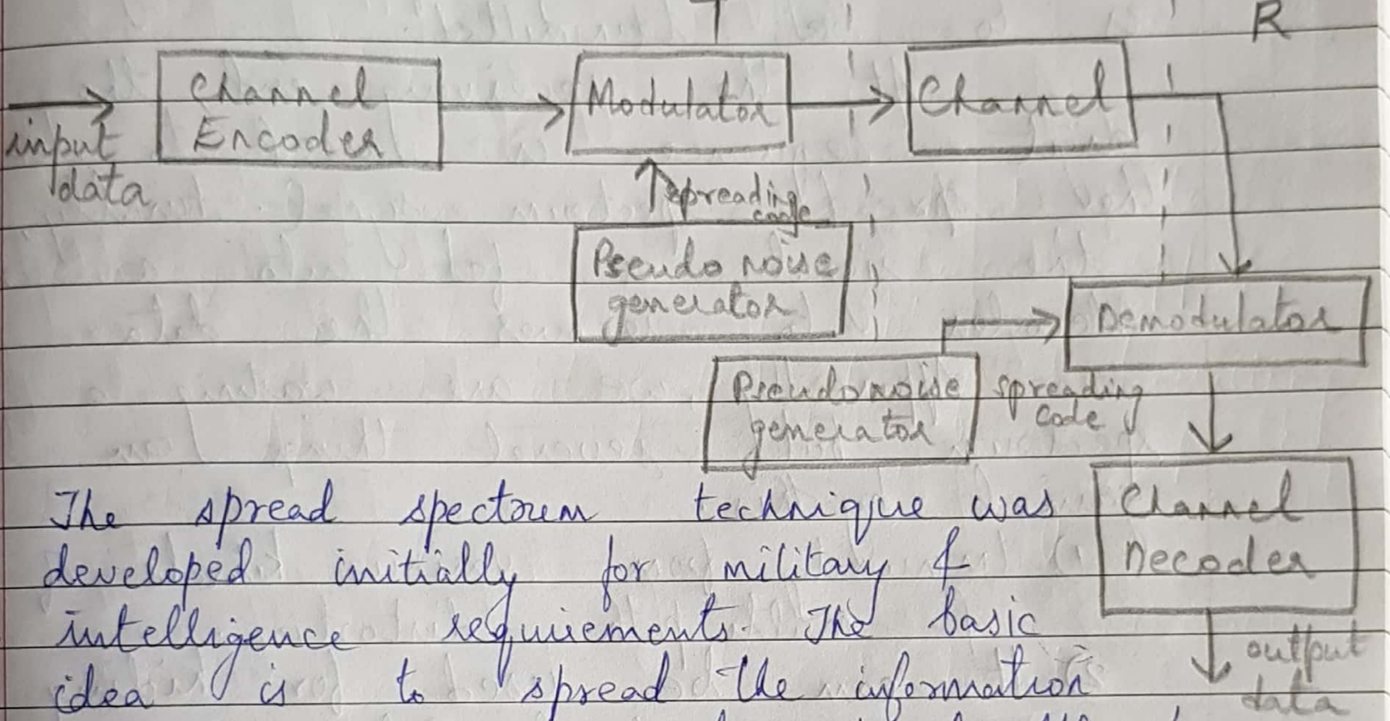
MODULE - VI

Spread Spectrum



The spread spectrum technique was developed initially for military & intelligence requirements. The basic idea is to spread the information signal over a wider bandwidth to make jamming & interception more difficult.

Two types of spread spectrum are used: Frequency Hopping spread spectrum and Direct Sequence Spread spectrum.

The figure shows the key characteristics of the spread spectrum. The i/p is fed into the channel encoder which produces an analog signal with a relatively narrow bandwidth around some central frequency. This signal is further modulated using a sequence of digits known as spreading code or spreading sequence.

Spreading code is usually generated by a pseudo noise generator or pseudo random number generator.

The effect of this modulation is to spread the spectrum or to increase the BW significantly of the signal which is to be transmitted.

At the receiving end, the same digit sequence is used to demodulate the spread spectrum signal. Finally, the signal is fed into a channel decoder to recover the data.

Apparently, we are wasting a wide spectrum but several things can be gained by this.

1) Immunity from various kinds of noise & multi path distortion.
2) It can be used for hiding & encrypting signals. Only a recipient who knows the spreading code can recover the encoded information.
3) Several users can independently use the same higher BW with very little interference. This property is used in cell-phones with a technique known as CDMA.

The Pseudo Random numbers are generated by an algorithm using some initial value known as 'seed' value. The algorithm is a deterministic algorithm & produces a sequences of numbers which though not are statistically random, it is good enough as a random code and hence known as pseudo random number.

So, without knowing the algorithm & seed, it will be impractical to predict the sequence.

Hence, only a receiver which shares this information with a transmitter will be able decode the sig successfully.

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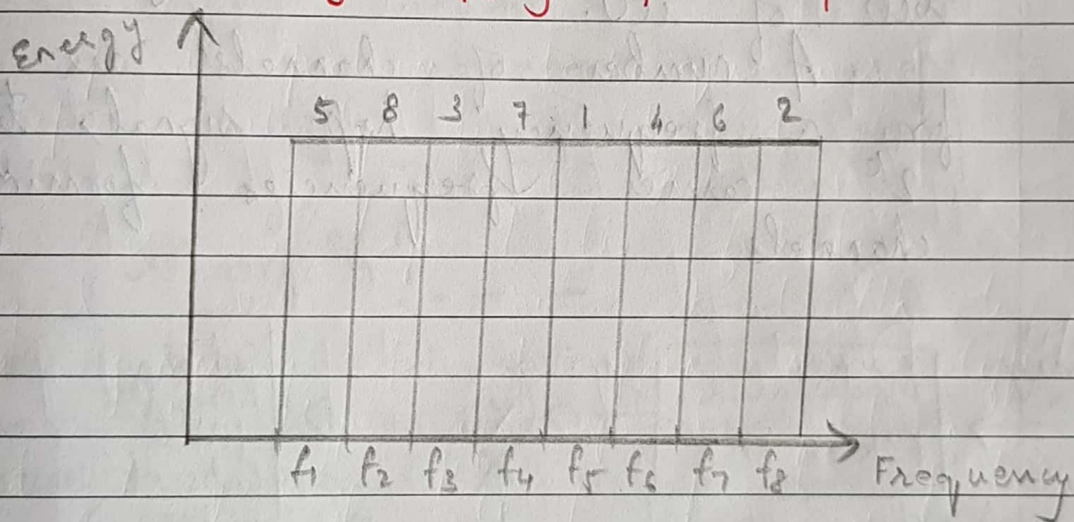
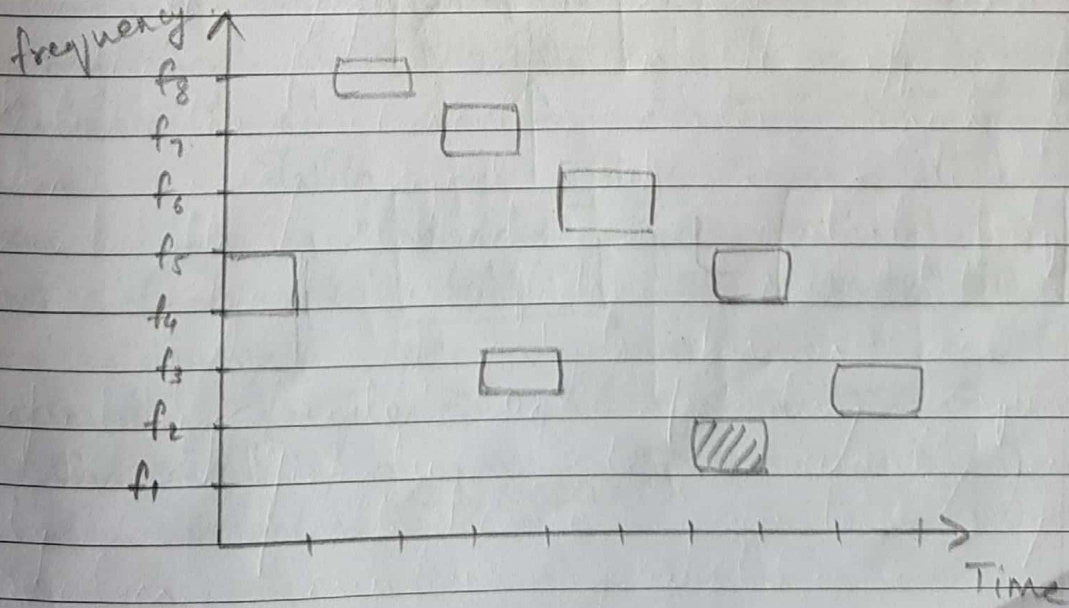
Frequency Hopping Spread Spectrum (FHSS)

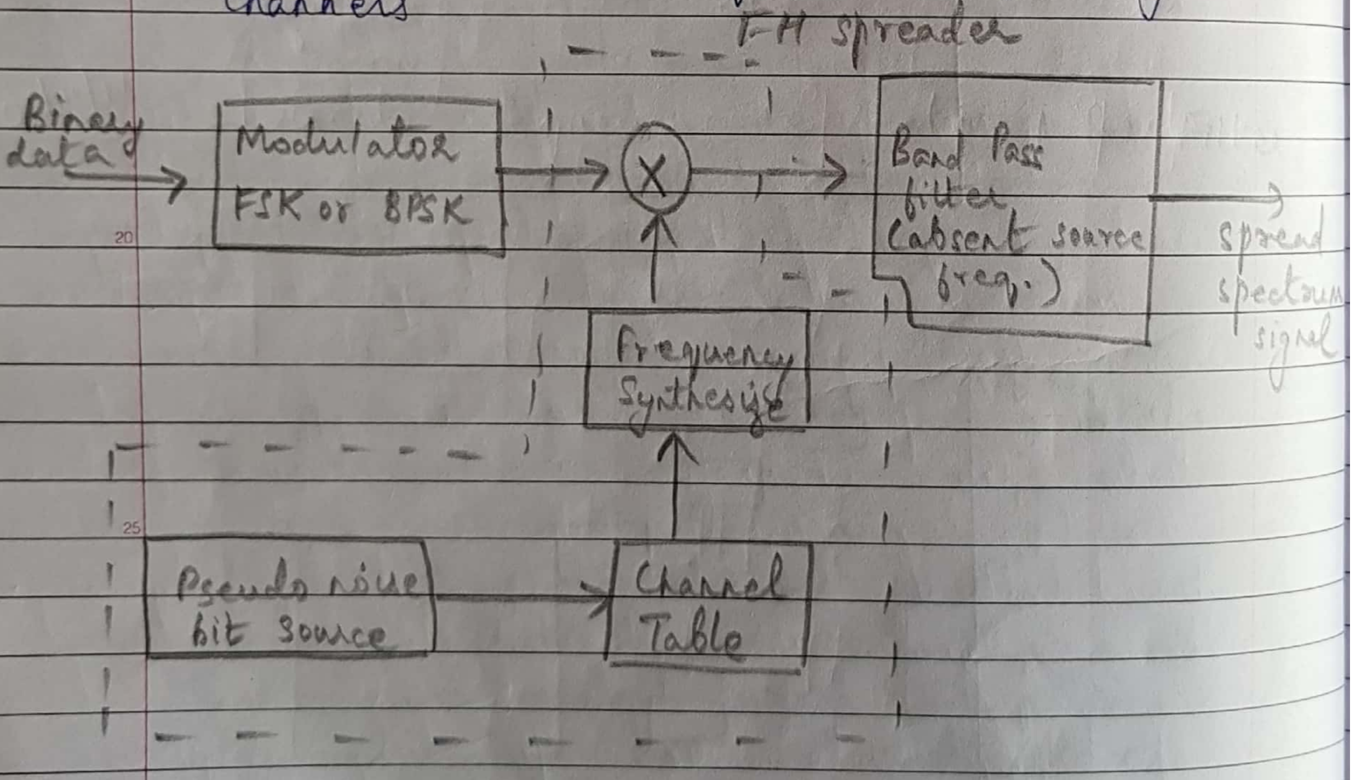




With FHSS, the signal is broadcast for a seemingly random series of radio frequencies hopping frequency to frequency at fixed intervals.

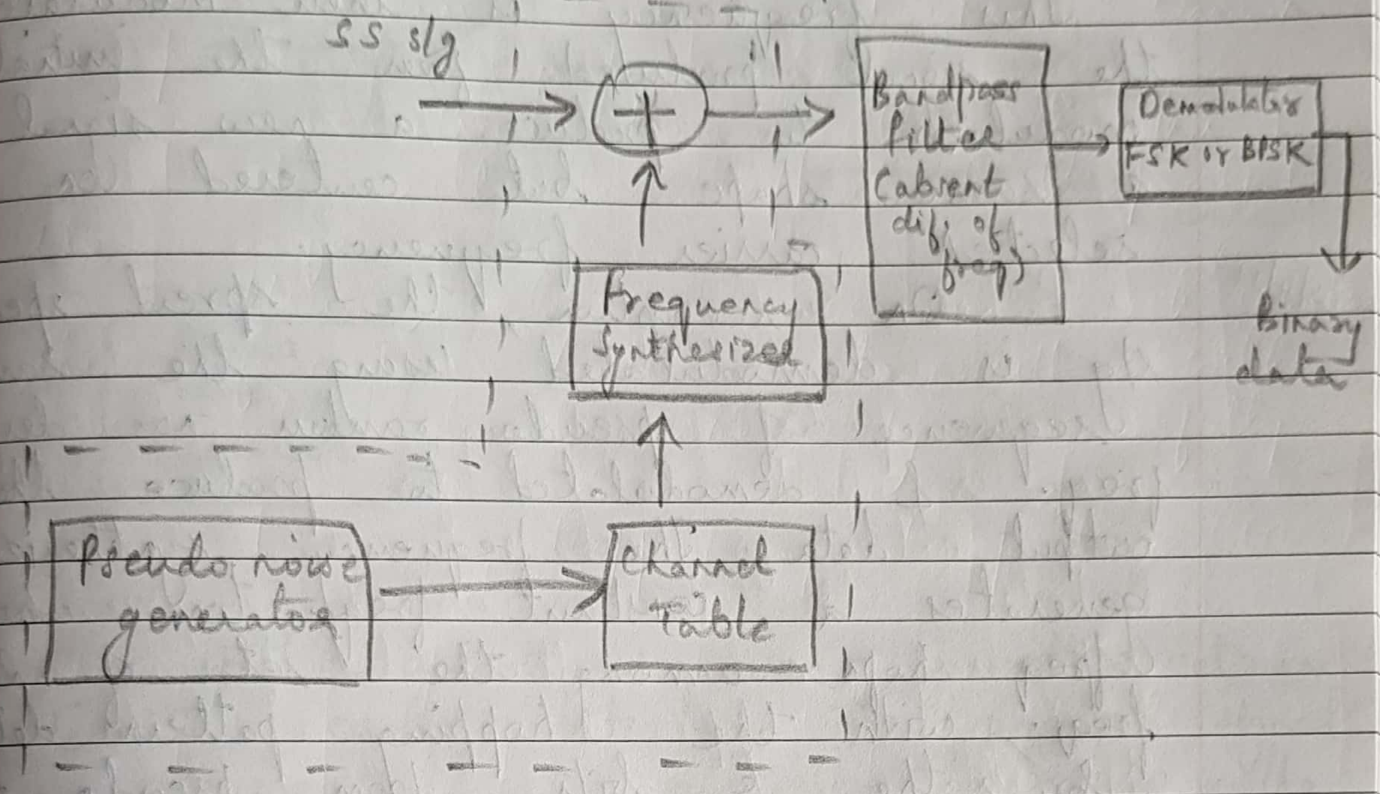
A receiver hopping between frequencies in synchronization with transmitter picks up the message. An eaves dropper will hear only unintelligible blips. Attempts to jam a signal on one frequency succeed only at knocking out few bits of it.

A number of channels are allocated for a frequency hop signal, typically 2^k carrier frequencies forming 2^k channels



Transmitter

FH Despreader



Receiver

A typical FH system is shown in the figure. For transmission, binary data are fed into a modulator using some digital to analog encoding scheme (FSK | BPSK). The resulting signal is centered on some base frequency. A pseudo noise or a pseudo random no. source serves as an index into a table of freq. which is referred as the spreading code.

Each k bits of the pseudo random no. source specifies one of the 2^K carrier frequencies. At each successive interval, a new carrier frequency is selected.

This frequency is then modulated by the signal produced from the initial modulator to produce a new signal with a same shape but centered on the selected carrier frequency.

On reception, the spread spectrum sig is demodulated using the same sequence of pseudo random no. derived freq. & demodulated to produce the output data. The frequency synthesizer generates a constant freq. tone whose freq. hops among the set of 2^k freq., with the hopping pattern determined by the k bits from pseudo no. sequence.

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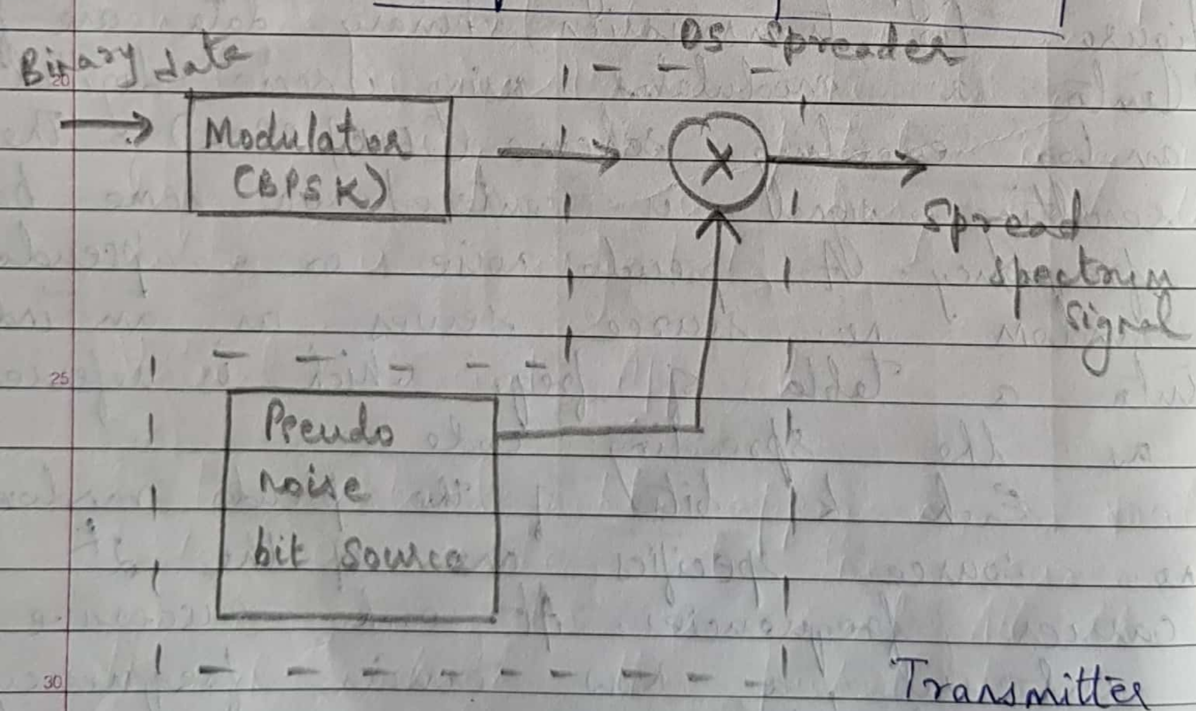
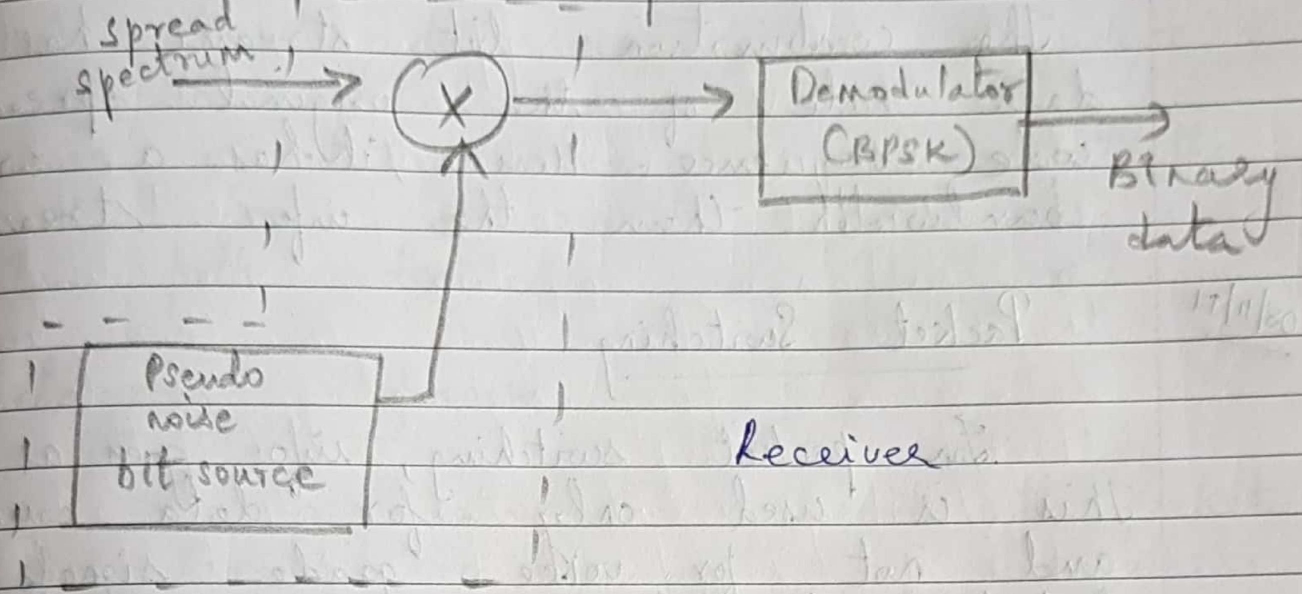
Direct Sequence Spread Spectrum



DS Despreader



Receiver

With direct sequence spread spectrum (DSSS), each bit in the original signal is represented by multiple bits in the transmitted signal using a spreading code. The spreading code spreads the signal across the wider frequency band in direct proportion to the number of bits used.

Suppose we are using a 10 bit spreading code, it will spread the s/g across a frequency band 10 times greater than a 1 bit spreading code.

One technique with the DSSS is to combine the digital info. stream with a spreading code bit-stream using XOR. So an info. bit of 1 inverts the spreading code bits in the combination while an info. bit of 0 causes the spreading code bits to be transmitted without inversion.

The combination bit stream has a data rate of the original spreading code sequence. Hence, it has a wider bandwidth than the info. stream.

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Packet Switching

In packet switching, info. goes as packets. This is used only for data commn. and not for voice grade signals. Packets will have routing information depending on whether it follows a data gram approach or a virtual circuit approach.

The packet size also varies.

As compared to circuit switching, even when a node is busy, packets can be send to a node in a usual circumstance. The delay or latency of delivering the packets may increase if the node is busy or the route is congested.

Virtual Circuit Approach

In virtual ckt. approach, a routing table is made at the source from which a data is send. All the nodes through which the packet has to traverse will be there in the routing table. In this approach, packets are send & received in sequence. The total delay will be the sum of the delay experienced at the nodes from source to destination. The path decided is not altered.

Datagram Approach

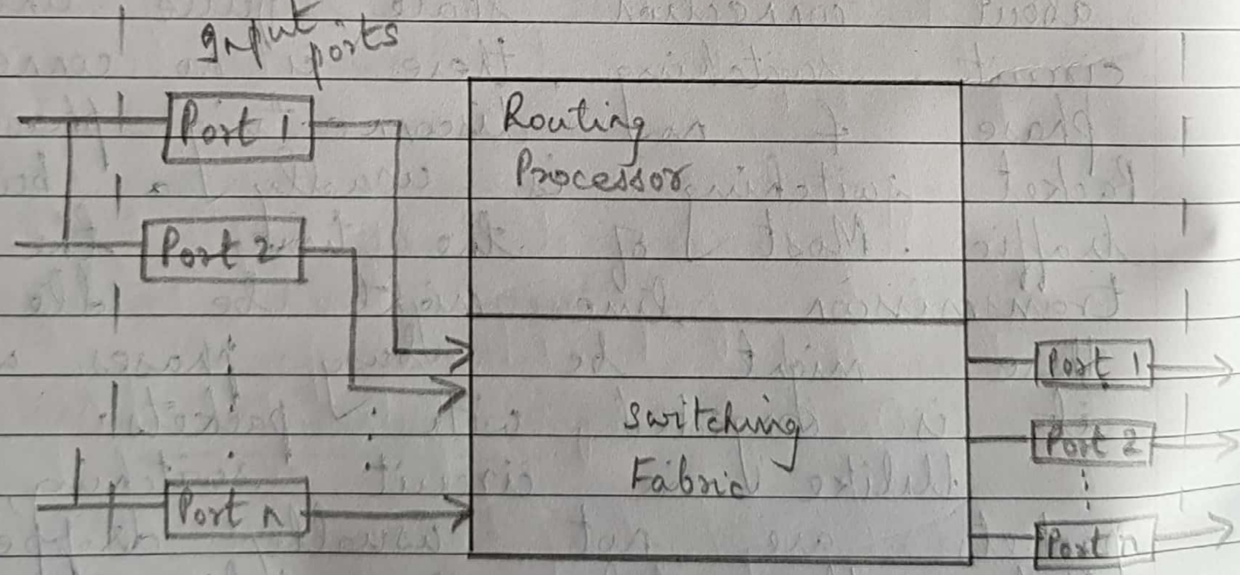
In a datagram network, the packet is treated independently of all others even when it is a part of multi-packet transmission. Packets are known as datagram in this approach. Datagram network is known as a connectionless n/w. It means that the packets which does not keep info. about connection state, quite unlike a circuit switching, there is no connecting phase & no disconnecting phase. Packet switching is usually a burstive traffic. Most of the times, the transmission line might be idle & there might be busy phases when it is congested with packets.

Unlike in circuit switching, packets are not usually dropped. But the delay in delivering the packets might vary depending on the traffic, processing delay at the node or other factors.

In datagram approach, the destination information remains the same whole through its travel. At each node, the next node is decided depending on distance, congestion, availability of the node etc. Due to this in the datagram approach, datagrams travel through multiple or varied paths.

They might reach the destination out of sequence. The packets may have to be reorganized according to their sequence no.

Structure of a Packet Switch:



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Input Port

An i/p port performs the physical and datalink fns. or the packet switch. The bits are constructed from the received s/gs. The packet is decapsulated. Errors are detected & corrected. The port will be having buffers or queues to hold the packet before it is directed to the switching fabric.

Output Port

The o/p port performs the same fns. as the i/p port but in the reverse order.

First, the outgoing packets are queued, then packets are encapsulated in a frame & finally a physical layer functions are applied to the frame to create a s/g to be send on line.

Routing Processor

The routing processor performs the fns. of the N/w layer. The destination address is used to find the address of next port & o/p port no. from which packet is send out

Switching Fabric

The most difficult task in PS is to move the packet from i/p queue to o/p queue. The speed with which it can be done affects the overall delay in packet delivery.

Packet switches are specialized mechanisms which uses a variety of switching fabrics. A simplest type is a cross-bar switch.