4/8/2017

Module 1.

Data communication model.

Source → Transmitter → Transmission S/m → Receiver → Destination

Source:

generates the data to be transmitted.

Eg: telephone, PCs.

Transmitter:

transforms and encodes information in electromagnetic wave or pulses which can be transmitted across a transmission s/m.

Eg: modem.

Transmission system:

A single line or a complex network connecting source and destination.

Receiver:

Accepts the electromagnetic signal from the transmission system and converts into a form which can be handled by destination.

Destination:

takes the incoming data from the receiver.

> The communication system exchanges data b/w two parties.

> Transmission media could be guided media

(Eg. twisted pair, coaxial cable, optical fibre)

> Unguided media or wireless is by means of electromagnetic as air, water or sea water.

> The communication should be simplex, half duplex or full duplex.

Simplex

Signals are transmitted only in one direction.

ie, from transmitter to receiver.

Eg: radio stations

Half duplex:

Both stations can transmit & receive but only one at a time.

Eg: walkie - talkie.

Full duplex:

Both stations can transmit & receive simultaneously.

Eg: telephone, mobile phones.

7/8/17 Mon.

A periodic Analog Signal

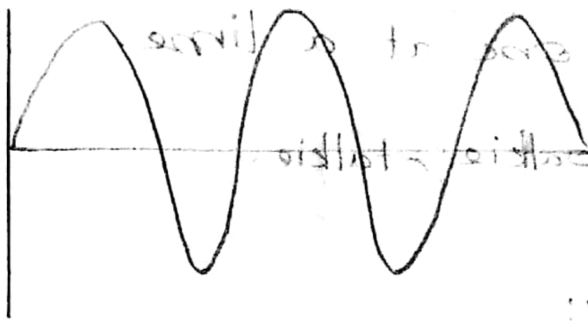
An analog signal is one in which signal intensity varies in a smooth wave over time or has no breaks/discontinuities in the signal.

A signal is $s(t)$ is continuous.

if $\lim_{t \to a} s(t) = a$ for all a.

Analog & D

> **Time Domain:**



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

Mathematically;

$s(t)$ is periodic if

$$s(t+T) = s(t)$$

$$-\infty < t < \infty .$$

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

Time domain concepts:

A sine wave is represented by 3 parameters.

* Peak amplitude (A)
* Frequency (f)
* Phase (ϕ)

Waveform is represented as.

Signal S at any instant t.

$$\boxed{S(t) = A\sin(2\pi f t + \phi)}$$

A - Peak Amplitude is the maximum value of the signal. (Strength of the signal over time).

f - frequency is the rate at which the signal repeats.

T - Period is the amount of time for one wave.

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

ϕ - Phase is the measure of a relative position in time within a single period of a signal.

λ - wavelength of a signal is the distance occupied by a single cycle.

(λ will be same if the wave is periodic) / Distance between any two points of corresponding phase of two consecutive cycles.

$$V = f\lambda \qquad V = \frac{\lambda}{T}$$

> ## Frequency Domain:

There will be second frequencies which are the integral multiples of a fundamental frequency and the period of the total signal is always equal to the period of fundamental frequency.

> ## Analog & Digital signal

Analog signals are continuous in nature and most of the data collected by sensors are analog in nature.

Eg: audio, video.

Digital signals are discrete in nature.

Signals are electric or electronic representation of data. Physical propagation of signals through a suitable medium/substance is called signalling. Communication of data by propagation & processing of signal is called transmission.

In any communication s/m point to point communication is done by electromagnet means. It could be through guided media or wired media.

Eg.: coaxial, twisted pair, fibre optics.

If $s(t+T) \neq s(t)$ Aperiodic

Digital signals are cheaper than analog signals. They are less susceptible to interferences changes but suffer more from attenuation

> Analog Transmission

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 and receiver data still will be intelligent.

Scanned by CamScanner

> Digital Transmission

is concerned with content of the signal but it can only be transmitted to a limited distance before attenuation makes it unintelligible.

Noise and other requirements endanger the integrity of data.

To achieve greater distance repeaters have to be used.

9/8/17 wed

> The repeaters recover the digital data from the analog signal and generates a new clean ~~analog~~ digital signal. The noise does not get cumulated.

> ADVANTAGES OF DIGITAL SIGNALLING

→ The VLSI technology have made the digital signalling much cheaper.

→ Data integrity is better

→ Better capacity utilisation

→ Better encryption technologies available makes it more secure.

→ Similar signals can be integrated and makes overall communication cost cheaper.

TRANSMISSION IMPAIRMENTS:

1. Attenuation: × Axenuation

Strength of signal falls off with distance for any transmission medium. For guided media exponential increase in attenuation is experienced. Hence a constant number of decibels per unit distance is introduced. For unguided media it is a more complex function of ~~attenuation~~ atmospheric conditions and parameters. The main concerns about attenuation are

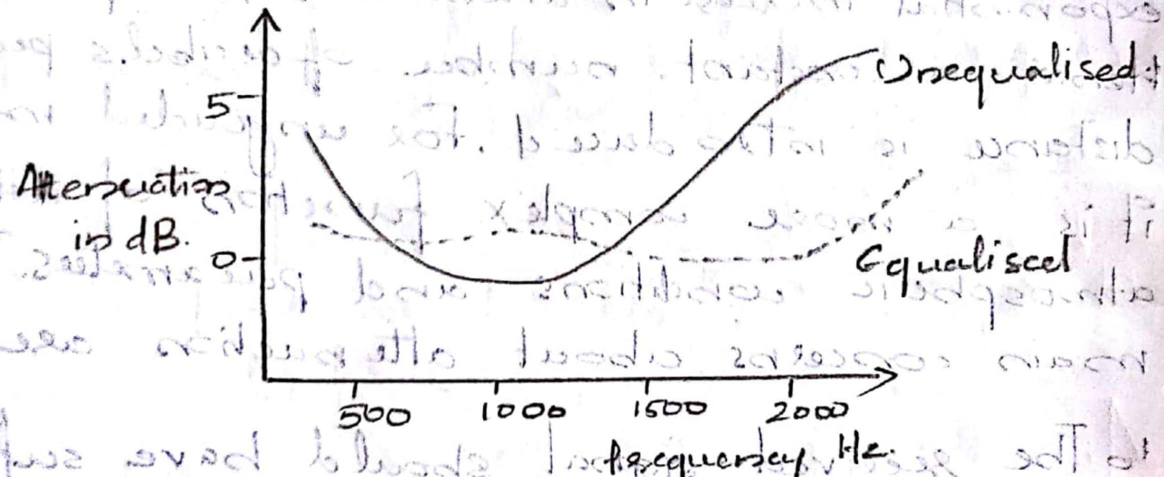
1. The received signal should have sufficient strength to be detected as a signal.

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 function of frequency. Hence for higher frequency attenuation is more.

Amplifiers and repeaters are used to alleviate the strength fading and distinction of signal from noise to some good extent.
If transmitter power is high, noise also gets high so raising the transmitter power to boost the signal has its limitation

→ Attenuation varies as a function of frequency especially for an analog signal. So
So the received signal gets distorted. So equalising techniques are used.



→ Especially for voice grade telephones which (20Hz-20kHz) smooths out attenuation effect

→ Another way is to amplify higher frequencies more as compared to lower frequencies.

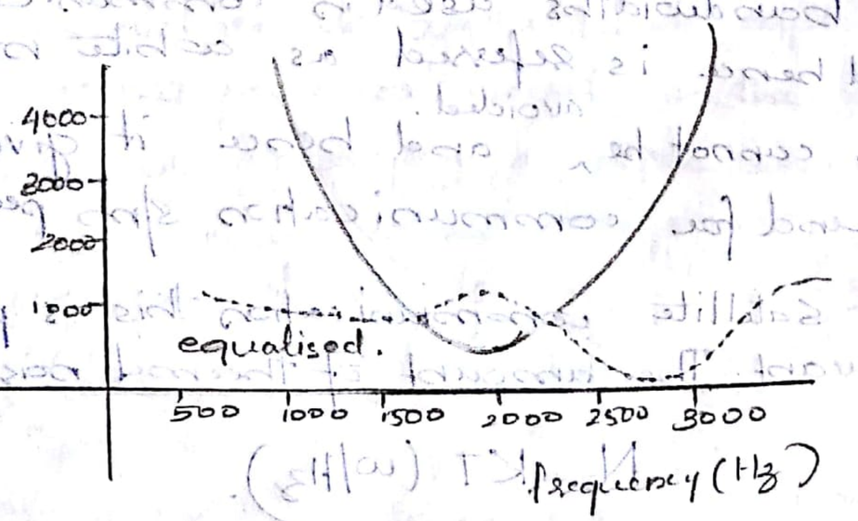
→ The upper end of voice band are attenuated more than the lower frequency

→ This certainly causes distortion of speech signal. The flattening of curve improves the quality of sound.

→ For digital signal attenuation causes less of a problem

2. Delay Distortion

Delay distortion occurs ~~with~~ because the velocity of signal through guided media ~~varying~~ varies with frequency. For a band limited signal the velocity tends to be higher near the centre frequency and falls off near the edges. Hence various frequency components arrive at the receiver at different times resulting in phase shifts b/w different frequencies. The effect is referred as delayed distortion. Since received signal is distorted due to various delays experienced for its constituent frequencies. This is more critical for digital data. Suppose the 1's component spill over other bits causing inter symbol interference it will limit the maximum bit rate of the transmission channel.



11/8/17 Fri.

Noise:

In any data transmission the signal consists of the transmitted signal, distortion by the transmitted systems and additional unwanted signals inserted b/w transmitted noise.

All undesired signals are referred to as noise.

The types of noises:

1. Thermal noise
2. Intermodulation noise
3. Crosstalk
4. Impulsive Noise

1. THERMAL NOISE

Thermal noise occurs due to the agitation of electrons. It is present in all transmission media and is a function of temperature. Thermal noise is uniformly distributed across all bandwidths used in communication s/ms and hence is referred as white noise.

This cannot be avoided and hence it gives an upper bound for communication s/ms performance. For satellite communication this is particularly relevant. The amount of thermal noise is given by.

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

No : Noise power density in W/Hz.
K : Boltzmann constant.
$= 1.38 \times 10^{-23}$ J/K
T : Temperature in Kelvin (K)

2. INTERMODULATION NOISE

When different frequencies share the same transmission medium it results in intermodulation noise. It produce a frequency which is the sum or difference of two original signals or multiples of those frequencies. This is due to the non-linearities of the transmitter or receiver and/or transmission medium intervention. Excessive non-linearity is caused by component malfunction or overloading of excessive signal strength.

3. CROSSTALK:

Crosstalk occurs by electrical coupling between nearby twisted pair or rarely in the case of coaxial cables carrying multiple signals. It can also occur when microwave antennas pickup unwanted signals even if the antennas are highly directional. This is caused by spreading of signals during propagation. Crosstalk is of the same order of magnitude as thermal noise.

Scanned by CamScanner

4. Impulsive Noise:

Impulsive noise is non-continuous. It consists of irregular pulses or noise spikes. It is highly unpredictable and of very short duration generally having relatively high amplitudes.

14/8/17 Mon.

The source of an impulsive noise could be electromagnetic disturbance such as impulse lightning or faults & flaws in communication system. It is generally a minor annoyance (disturb-) for analog data such as short clicks and crackles but no loss of intelligibility for data. But is a primary source of error for digital data communication.

→ Channel Capacity:

Channel capacity is the maximum rate at which data can be communicated over a communication path under given conditions.

→ Data rate

Data rate can be expressed as bit/second at which data can be communicated.

→ Bandwidth.

Bandwidth is constrained by the transmitted signal, transmitter and the transmission medium. It is usually expressed as cycles/second or hertz.

→ **Nyquist Bandwidth**

In a noiseless channel the only limitation of data rate is by the bandwidth of the signal. Niquist suggest if the rate of bandwidth signal is 2B signals with frequencies no greater than B is sufficient for the signal rate. The converse is also true. Given a bandwidth B the highest data rate carried by the channel is 2B

Channel capacity

$$C = 2B\log_2 L.$$

→ **Shannon's Capacity theorem.**

In a noisy channel one has to consider the relationship between 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 power of signal to the power of noise measured as

$$SNR_{dB} = 10 \log_{10} \frac{\text{Signal power}}{\text{Noise power}}.$$

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

ie

$$C = B \times \log_2(1 + SNR)$$

(bits/s)

MODULE 2.

19/8/17 Mon. Transmission medium.

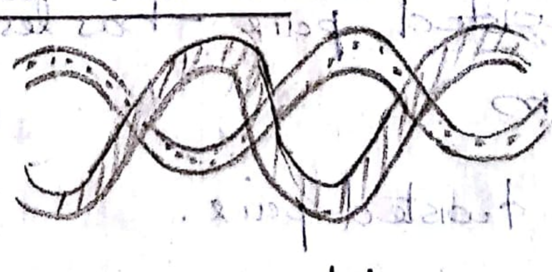
To transmit any signal from the transmitter to the receiver a transmission medium is required. Any transmission medium will have transmission impairments such as noise, crosstalk, intermodulation frequencies etc. and attenuation.

There are two types of transmission medium, guided medium and unguided medium.

Be it any medium, the factors that we interested are data rate, bandwidth, error rate etc...

Usually if bandwidth is greater, data rate is higher. Transmission impairments are highest for twisted pair cables and lowest for fibre optic cables. Coaxial cables stand in between.

→ Twisted Pair Cables



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

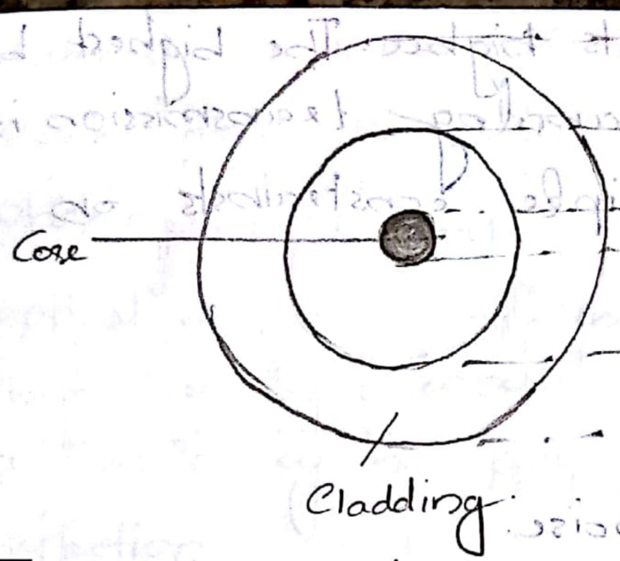
Twisted pair can be used to transmit both:
Analog signals need amplifiers for every 2-3km and digital signal needs repeaters for every 5-6km
Twisted pair cable has limitations in distance, bandwidth & data rate. Hence, usually used in home connections. It is more susceptible to interference and noise due to electrical coupling.
Typical applications are home connections from PBX to end office in subscriber loops etc.
Usually twisted pair cable is less expensive than other types.
It has two ~~pairs~~ types:

1> Shielded twisted pair. - has less of interference problems

2> Unshielded twisted pair.

16/8/17 Read

Coaxial Cable:



The outer conductor is a braided sheath. The inner conductor is made of solid metal. They are separated by an insulation material and its all covered by a ~~padd~~ padding. The material & thickness of the padding depends on which application the cable is being put for. Coaxial cable can be used to operate over a wide range of frequencies. The typical applications are:

1> Television distribution
2> Long distance telephone transmission
3> Short run computer systems links.
4> Local Area N/w (LAN)

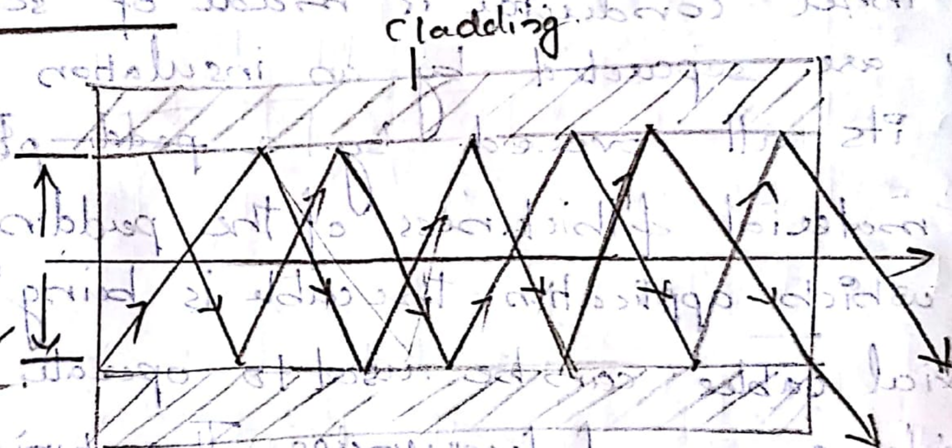
→ Transmission Characteristics

They are generally in b/w twisted pair cables & fibre optic cables. For long call analog transmission amplifiers are needed every few kilometers. The spacing gets closer.

as the frequency gets higher. The highest band usually used for analog transmission is 500 MHz. The principle constraints on performance are:

1> Attenuation
2> Thermal noise
3> Intermodulation noise

(C) Optical Fibre:



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

An optical fibre cable has a cylindrical shape and consists of three concentric sections a core cladding and a jacket

The core is the innermost section and it consists of ~~very~~ one or more very thin strands made of glass or plastic and has a

diameter in the range of ~~800μm~~ 8-10μm.
Each fibre is surrounded by its own cladding which again is made of glass or plastic but of optical properties different from that of core so that the light could always be confined within it. by the pply of total internal reflection.

The outermost layer is called a jacket. It is composed of plastic or other materials to protect against moisture, abbreation, crushing and other environmental dangers.

Application:

The fibre optics can be put for
1> Long distance communication
2> Military application
and has the following properties which has increased its usage:
1> Greater signal handling capacity
2> Its smaller size & lighter weight
3> Comparatively low attenuation
4> Electromagnetic isolation
5> Greater repeater spacing

Typical app are:

1> Long haul trunks.
2> Metropolitan trunks
3> Rural exchange trunks
4> Subscriber
5> Local Area N/W

23/8/17 Wed

Wireless Transmission

1 GHz - 14 GHz range is used as micro wave frequencies for wireless transmission. Antennas are a must for wireless transmission. The antenna size, its shape, the tower height and so also is the antenna gain which gives the parameter for the efficiency of the antenna.

Microwave freq are suitable for point to point communication for terrestrial transmission. and 30 MHz to 1 GHz are suitable for omnidirectional applications. This range is known as radio range.

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

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 heights for line of sight transmissions to the receiving antennas.

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

Typical applications:

1> For having microwave links.
For long hold telephone communications.
2> For closed circuit TV.
3> For cellular phones is widely being used.

Transmission Characteristics

The main source of loss is attenuation

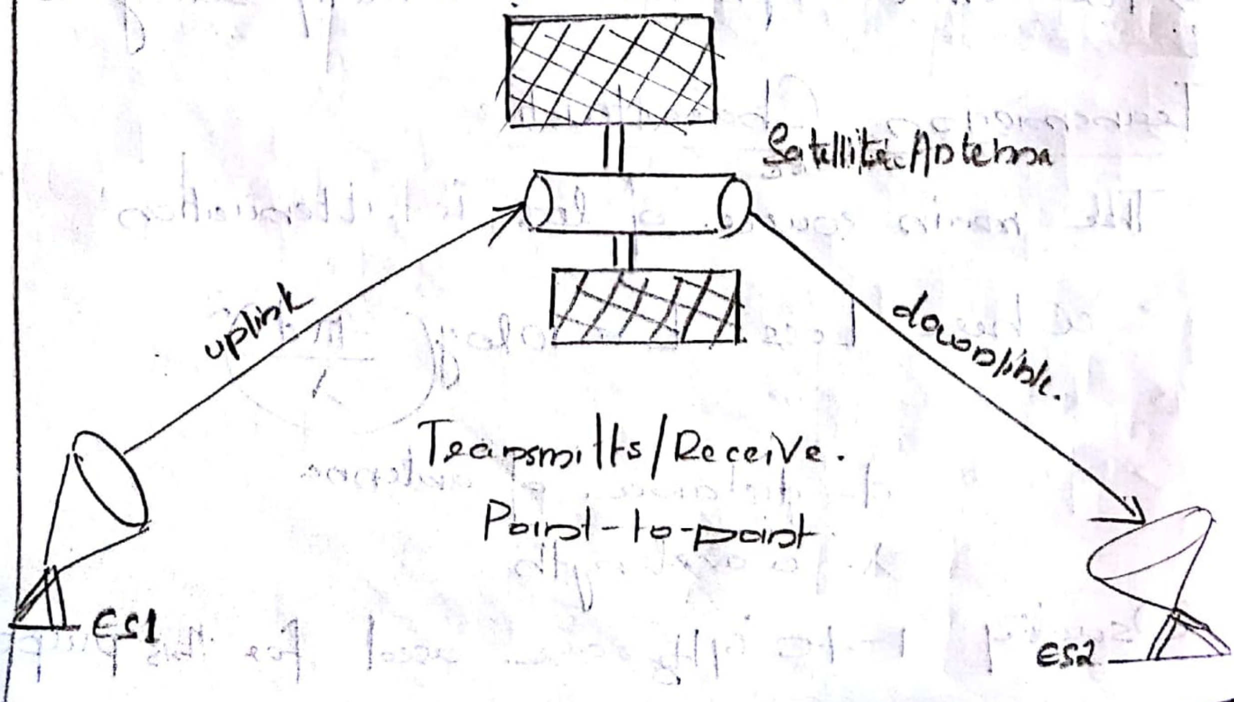
where Loss $L = 10\log\left(\frac{4\pi d}{\lambda}\right)^2$

d- distance of antenna
λ- wavelength

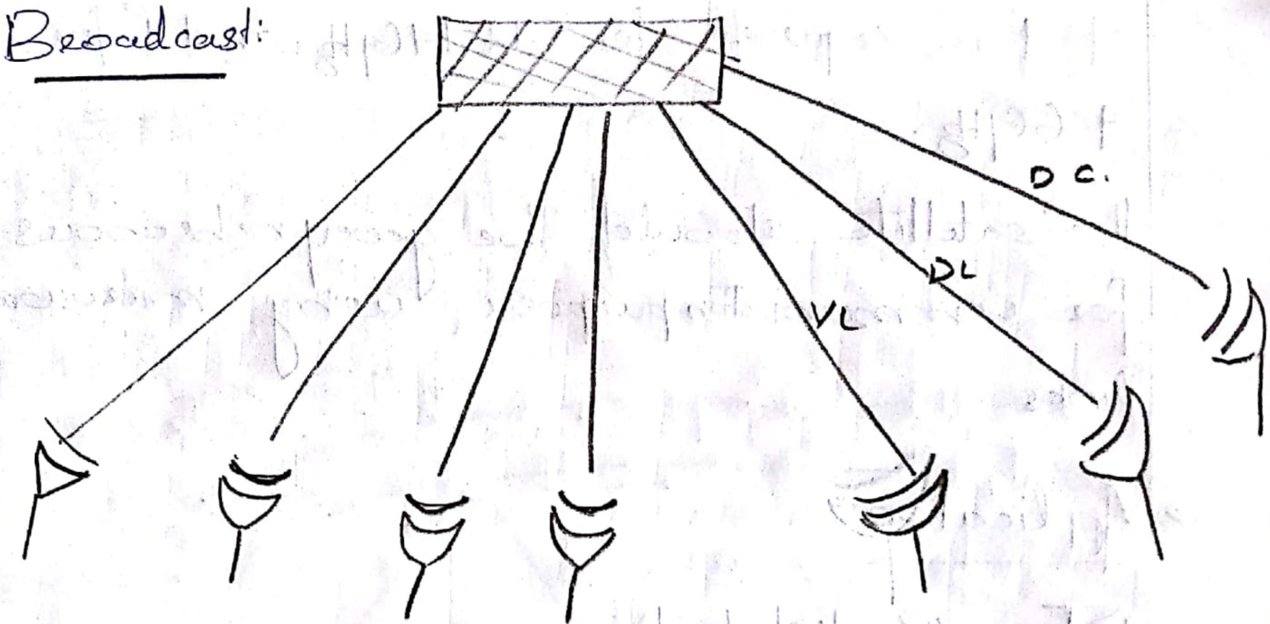
Usually 1-40 GHz are used for this purpose

For long hold communication 4-6 GHz is used.
and 12 GHz is typical for cable TV antenna.
For very short point-to point communication 22 GHz are used.
It should be noted that for higher frequency antennas could be cheaper and smaller.
Repeaters are used within 10-100 kms.
Attenuation of radiation increases with rainfall
Above 10 GHz frequencies this is more noticeable
With growing popularity of microwaves interference ~~is~~ also is a problem. Hence frequency bands are strictly regulated.

Satellite Communication



Broadcast:



In this the communication satellite act as a microwave relay station. It can link two transmitter ~~and~~ per receiver. known as earth station
A single satellite can operate on a number of frequency bands called transponder channels or transponders. An uplink (earth station to satellite) with a difference of frequency of atleast 2GHz is used ~~for~~ what downlink (from satellite to earth station) 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 12-14GHz and 3° for 4-6GHz.

The satellite should be geosynchronous for communication purposes using microwave links.

→ Applications.

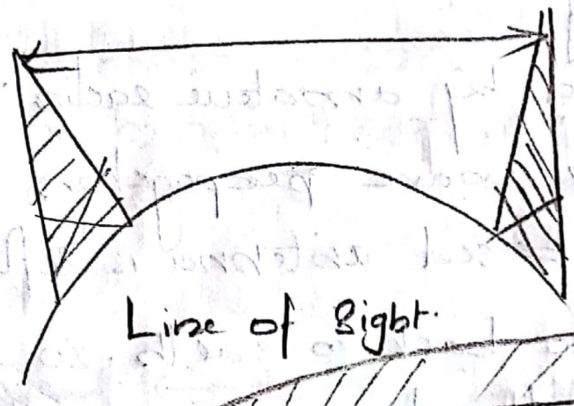
1> For TV distributions
2> For long distance telephone communications
3> For private business networks

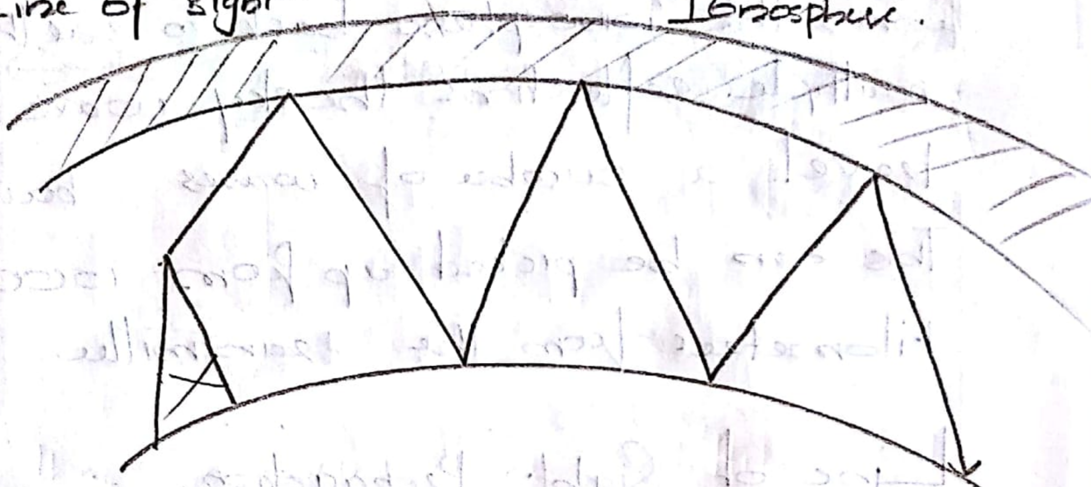
→ Some properties of the Satellite Communication

For long hold communication a delay is felt in vocal conversation. It also introduces problems in areas of error control and flow control

→ Wireless Propagation

Ground wave





* **GROUNDWAVE PROPAGATION:**

Frequencies upto 2MHz is used for this purpose Ground wave propagation follows the contour of the earth and can propagate considerable distances well over the visual horizon. This is because the electromagnetic waves tendency to follow earth curvature as a result of the magnetic attraction & slowing the wavefront nearer. This is typically used for AM radio application for medium band waves

Sky wave Propagation:

This is used by amateur radio.

The sky wave propagation the signal from earth based antenna is reflected from the ionosphere back to earth. So after multiple reflections the sky wave signal can travel a number of waves bouncing back to receiver. This can be picked up from 1000s of kilometers from the transmitter.

Line of Sight Propagation.

Only frequencies above 30MHz are used for this. Main problems faced are refraction ~~and~~. The optical line of sight distance can be given by.

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

d - distance travelled by em wave
h - tower height

There is an adjustable factor k to account for refraction.

$k = 4/3$ for ordinary purpose

If two antennas are used for transmitter

and receiver. $3.57(\sqrt{kh_1} + \sqrt{kh_2})$
will be the max dist achieve where
h_1, h_2 - heights of transmitting & receiving antenna.

25/8/17 Fri.

~~Signal~~ ~~Encoding~~ Techniques.

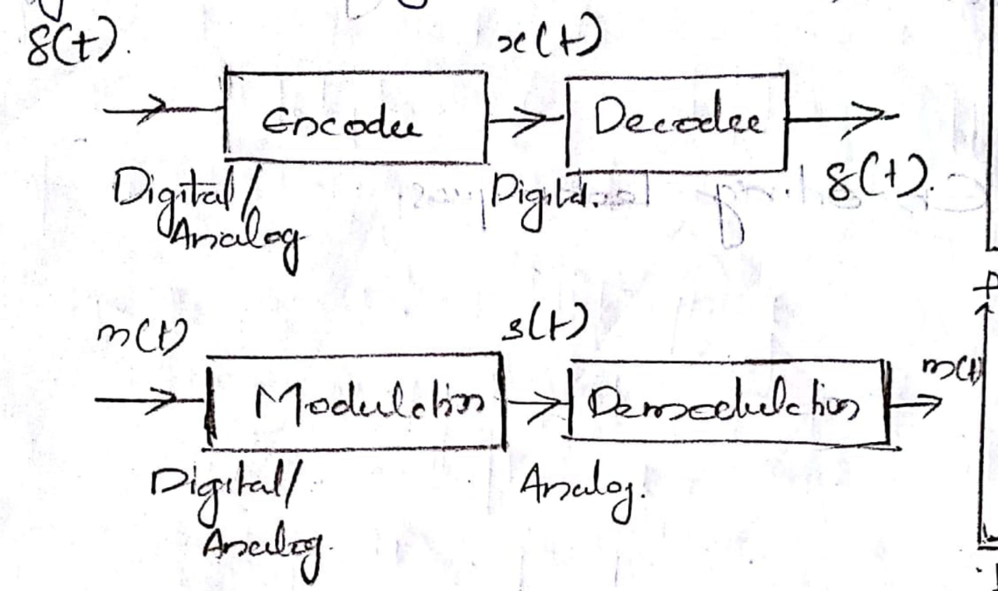
MODULE 3.

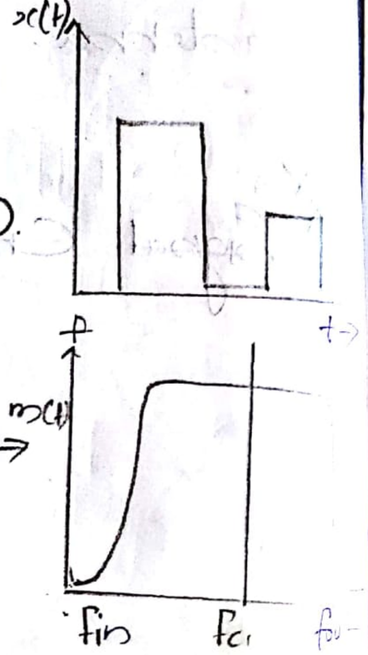
25/8/17
Fri

Signal Encoding Techniques

Digital Signal.

Analog Signal





The encoding depends on the source data and signalling method employed for transmitting the information through the transmission media. The data could be either digital or analog. Signalling method also could be digital or analog.

In the digital signal modulation the source could be either digital/analog. You use an encoder to assign value of voltages to make it into a digital form of em pulses. The information is gathered at the receiver in digital form.

In the analog signalling method the data could be either digital/analog. This is modulated with a carrier frequency & the base band is converted to a modulated analog wave which is received as a bandpass signal ~~centre~~ at centre frequency in the receiver.

The possibilities of combinations of the source data and signalling method are as follows

1> Digital data digital signalling method.

Easy, less complex, less expensive. In this assigning voltage 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 and digitized for digital transmission and some modulation methods such as pulse code modulation method to sample it & digitalize it is used.

3> Digital Data & Analog Signal

A MODEM converts digital data to analog to be transmitted across an analog line such as telephone / optical fibre. Different types of shift keying method such as

1> Amplitude Shift keying
2> Frequency shift keying or
3> Phase Shift keying

are used for the purpose.

4> Analog data & Analog Signal.

This is very easy as only carrier wave modulation of base band is necessary such as Amplitude modulation, frequency modulation phase modulation of the base band to the modulated band.

→ (c) Unipolar

In this the signal elements have the same algebraic sign ie, either +ve / -ve.

Eg: 0V and +5V to represent -0 & +1

→ Bipolar/Polar -

One logic level is represented by a +ve sign and other by a -ve sign.

Eg: +5V to represent 1 &
-5V to represent 0

→ Duration

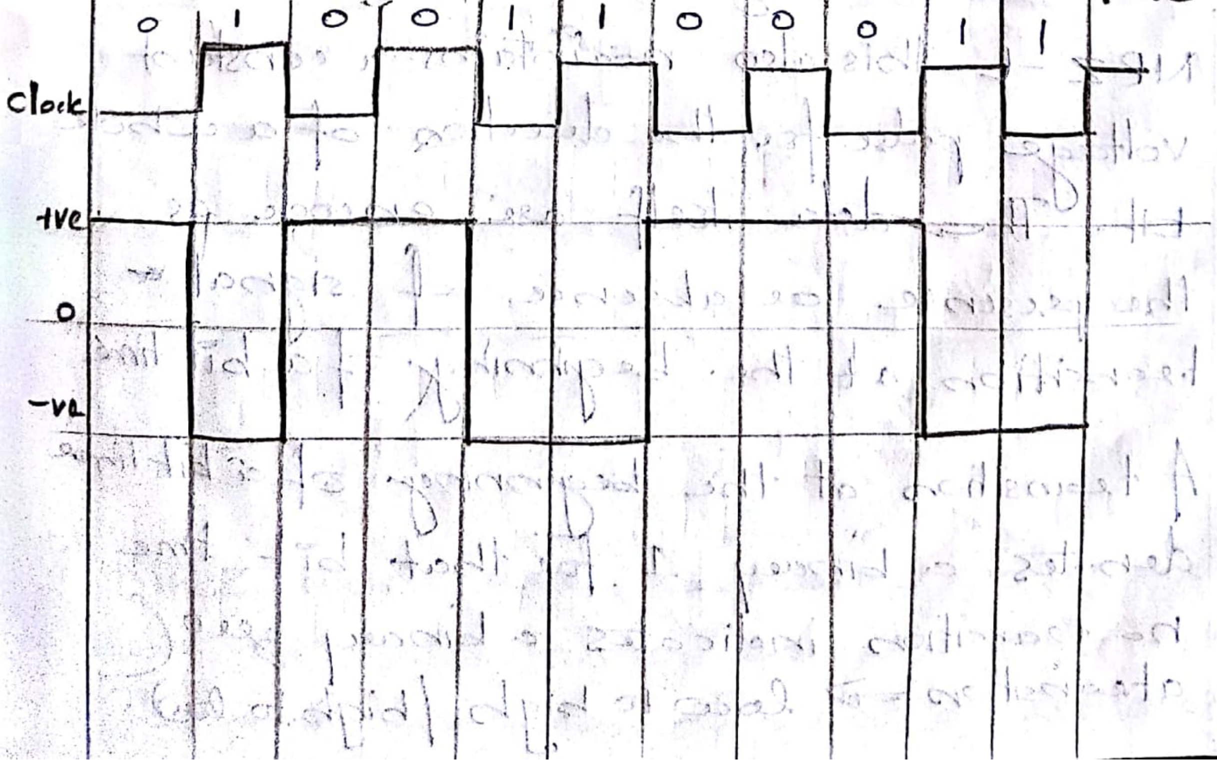
It is the time to encode a 1 or 0 expressed in bits/second. Both modulation rate is the number of modulation signal expressed in baud and a clock pulse is used to synchronize transmitter & receiver.

26/8/17 Sat.

NRZ CODES.

→ Not return to zero (NRZ)-L

0 → +ve voltage
1 → -ve voltage



It is the most common easiest way to transmit digital signals. To use two different voltage levels for two binary digits. They have the common property, that voltage level will remain constant during each bit interval and there is no transition/return to the zero voltage level.

Here a negative voltage represents binary 1 and a positive voltage represents binary 0.

→ Non - return to zero on inversion (NRZ-I)

This is the variation of NRZ. known as Non - return to zero on I. Quiet like NRZ-L this also maintains a constant voltage pulse for the duration of a whole bit. The data itself are encode as the presence or absence of signal 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 zero (a transition ⇒ low to high / high to low).

NRZ-I is an example of differential encoding. In differential coding, the information to be transmitted is represented in terms of changes between successive signal elements rather than the signal element itself.

· Encoding of the current bit is determined as follows:-

* If the current bit is binary zero then the current bit is encoded with the same signal as preceding bit.

* If the current bit is a binary 1 then the current bit is encoded with a different signal 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 signals are:

* Presence of dc component
* Lack of synchronisation capability b/w the transmitter & receiver.

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

Due to their simplicity and relatively low frequency response characteristics, NRZ-codes are commonly used for digital magnetic recording.

~~Due~~ Their limitations make the code unattractive for signal transmitting applica.

11/9/17
Mon.

→ Multilevel Binary

Suppose: Preceeding 1 → represented by -ve. value

Bipolar AMI (Alternate Mark Inversion)

In this 1 is represented by an alternate polarity of positive and negative signal values and 0 is represented by no signal value.

→ Pseudo Ternary

Assume: Preceding 0 → -ve.

In this 1 → no signal

0 → an alternate polarity of +ve & -ve signal.

- The multilevel binary codes address some of the deficiencies of NRZ-codes. These codes use more than two signal levels

In the case of Bipolar AMI a binary zero is represented by no line signal and binary '1' represented by a positive integer pulse. The binary '1' pulses must alternate in polarity. These codes have 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 the Pseudo ternary, the coding is represented by a reversal of bipolar AMI Binary '1' being represented by absence of the signal and '0' by the alternate polarity of positive & negative signals.
- The properties of Bipolar AMI are applicable to pseudo ternary as well. There is no particular advantage over

Bipolar AMI .

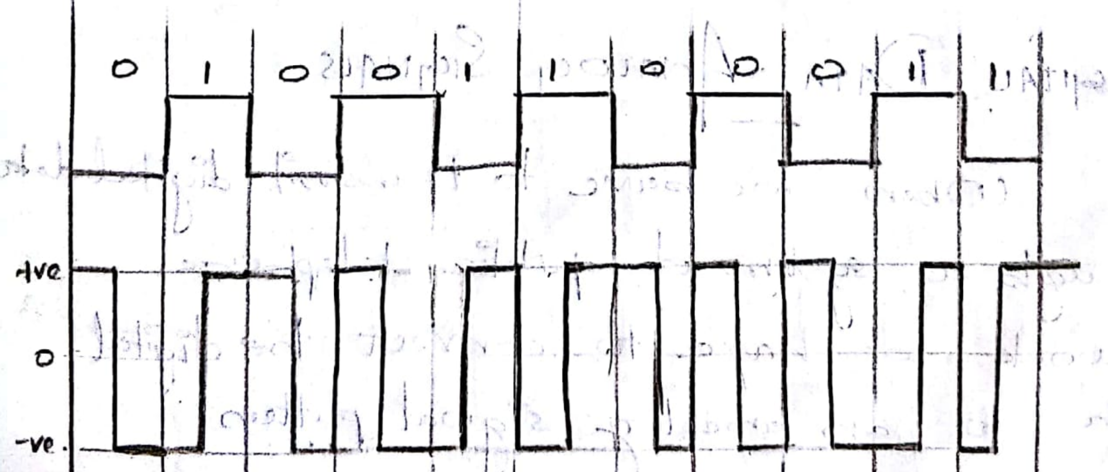
→ Biphase

In the Biphase, uses another set of coding techniques which overcomes the limitation of NRZ-codes.

These are two codes.

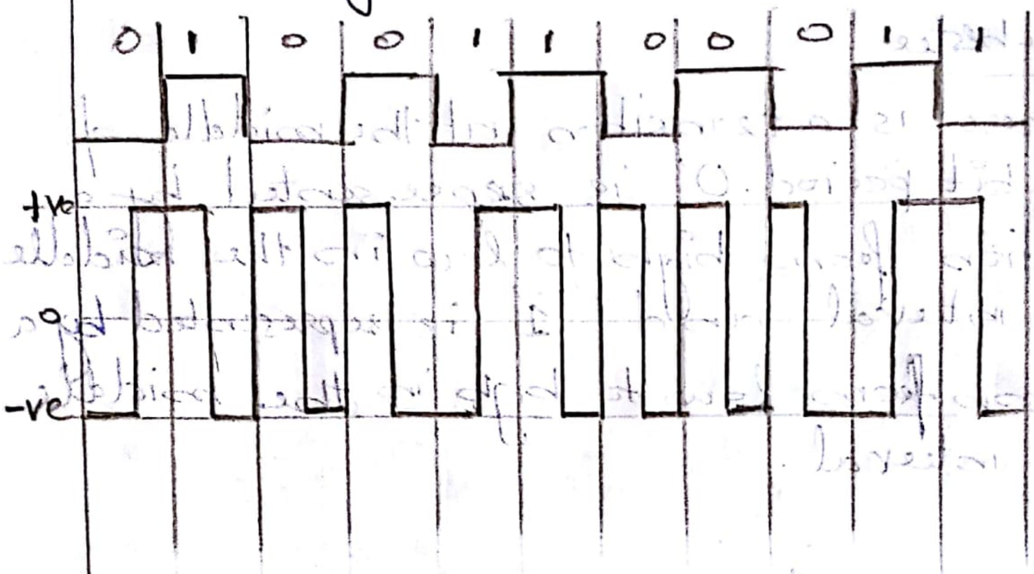
1> Manchester.

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 the interval and '1' is represented by a transition from low to high in the middle of the interval.



2> Differential Manchester

In the differential manchester there is always a transition in the middle of the interval. '0' is represented by a transition at the beginning of the interval & 1 by no transition at the beginning of the interval



13/9/17 Wed

DIGITAL DATA : ANALOG SIGNALS

When we have to transmit digital data through a system of public telephone network we have to convert the digital data to an analog signal pattern

A modem is used for this digital to analog conversion and back. In the above case a modem produce signals in the voice frequency range. The same technique is used for connecting two microwave frequencies.

The methods used traditionally:

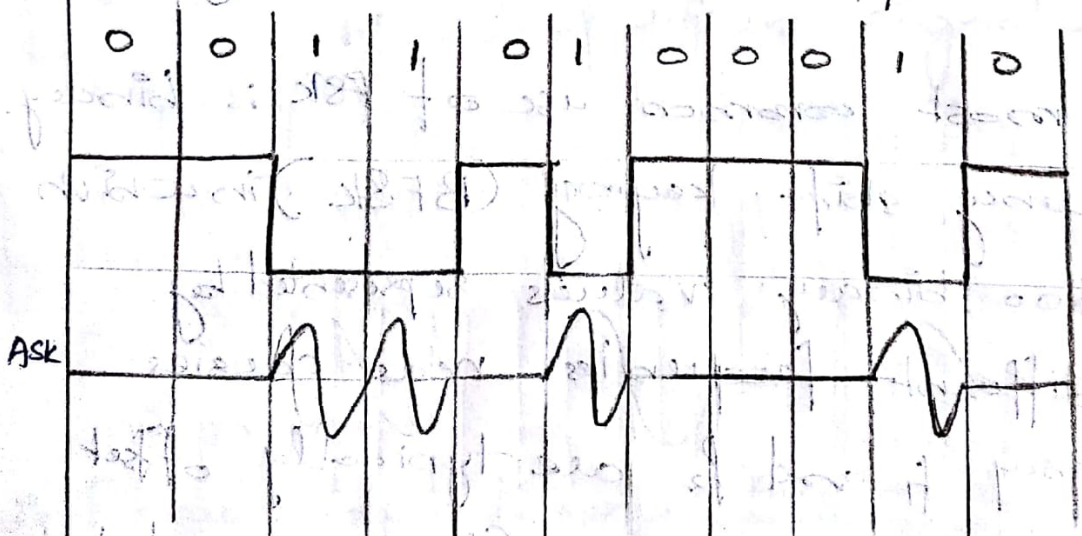
- Amplitude shift keying (ASK)
- Frequency shift keying &
- Phase shift keying.

1) Amplitude Shift keying:

Binary values 0 and 1 are represented by two different ~~amplitudes~~ values of carrier frequency.

$$A\cos(2\pi f_c t) \rightarrow \text{Binary } 1$$

$$0 \rightarrow \text{Binary } 0.$$



where the carrier signal is

$$A\cos(2\pi f_c t)$$

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

25/9/17
Mon

2) Frequency Shift Keying (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 (BFSK) in which the two binary values represented by two different frequencies near carrier frequency f_1 and f_2 are typically offset from carrier frequency (f_c) by equal but opposite amounts. FSK is less susceptible to errors than ASK. In voice grade lines, it is used typically upto 1200 bps. It is also commonly used for high frequency from 3MHz to 30MHz.

In multiple FSK, in which more than 2 frequencies are used can also be employed. It is more bandwidth efficient but is also more prone to error.

In MFSK.

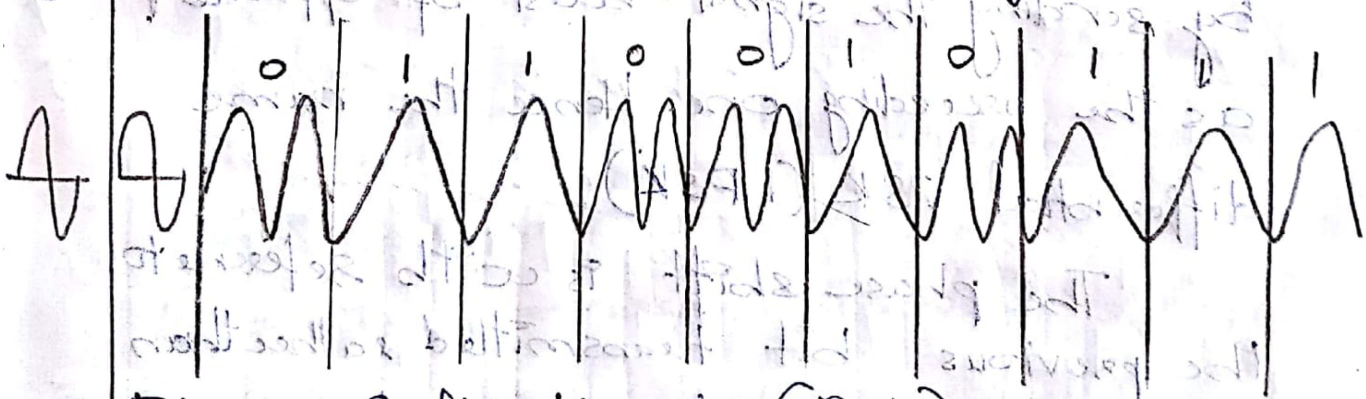
$$s(t) = A\cos(2\pi f_i t) \quad 1 \le i \le M.$$

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

f_c = carrier freq.

f_d = different freq.

M = no of different signal elements



3) Phase Shift Keying (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 used 2 phases to represent the two binary digits & is known Binary PSK (BPSK)

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

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

An alternate form of two level PSK is differential PSK in which a binary 0 is represented by sending a signal burst and a binary 1 is represented by sending the signal burst of opposite phase as the preceding one. Hence the name differential PSK (DPSK). 27/9/17 Wed

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

Four level PSK:

QPSK

~~$S(t) = \{ A\cos$~~

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

A more efficient way of using bandwidth can be achieved, if each segment element represents more than one bit. In quadrature PSK use shifts of odd multiples of $\pi/4$ as is previously shown.

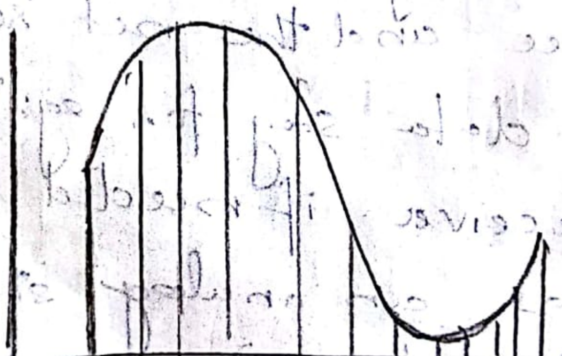
Analog data digital signals.

27/9/17 Wed.

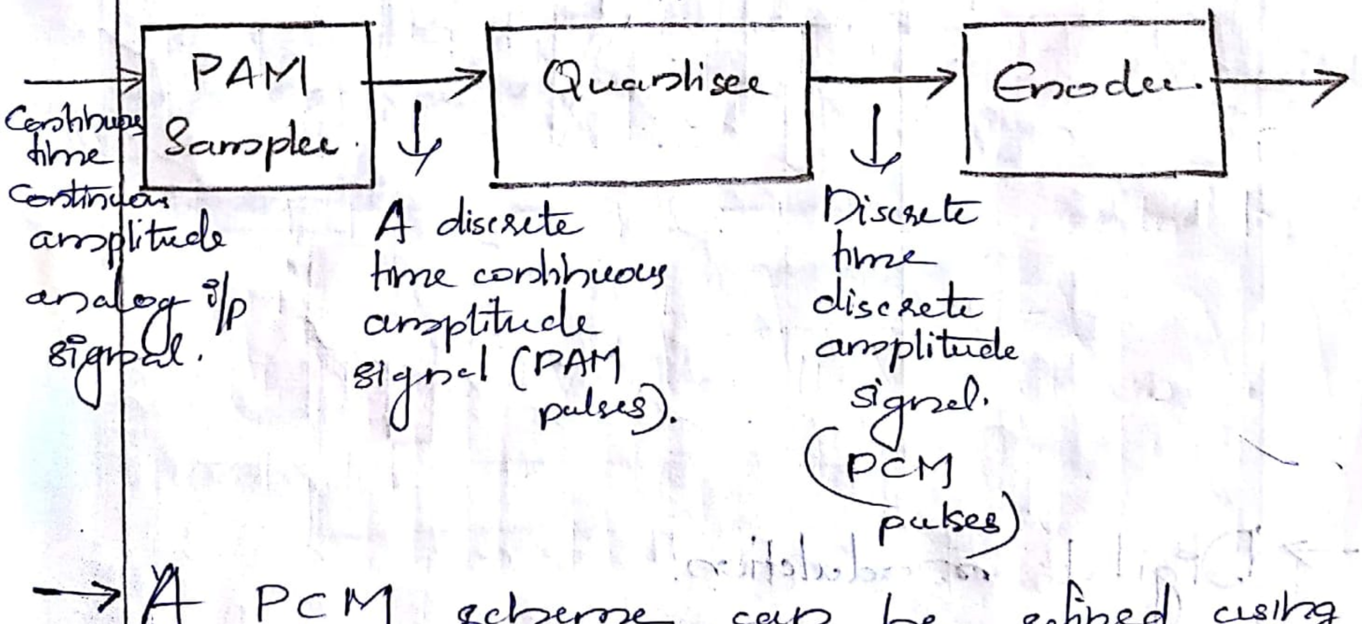
Analog data have to be converted to digital signals and usually it is termed as digitalisation. This is nothing but an analog to digital conversion. An analog data say a voice signal can be converted using a digitalisee and the net result will be a digital data say for eg: NRZ-L which at the receiver if needed to be converted back as an analog signal can be demodulated using a modem ~~so~~ with some technique such as ASK (Amplitude Shift Keying). The device used for converting analog data into digital form for transmission and subsequently recovering the original analog data from the digital is known as CODEC (Coding Decoding).

Pulse Code Modulation:

The pulse code modulation goes according to the sample theorem. The sample theorem states that if a signal f(t) is sampled at regular intervals of time and at a rate twice the highest signal frequency then the samples contain all information of the original signal. The f(t) (original signal) may be reconstructed from the samples using a low pass filter.

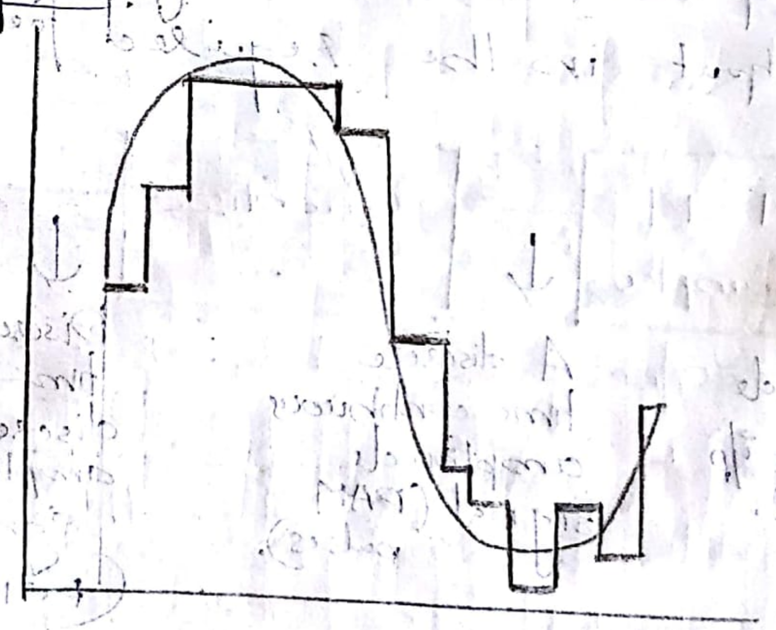


Pulse code modulation starts with conti. time continuous amplitude analog input signals which a pulse amplitude mode sampler will convert to discrete time ~~cont~~ continuous amplitude signals or PAM pulses which can be quantised using a quantiser and the result will be a discrete time, discrete amplitude signal ie, pulse code modulation pulses which can be input to an encoder and the output will be digital with string output in the required format.



→ A PCM scheme can be refined using a technique known as non linear encoding ie, effectively the quantisation 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 quantising steps for signals of large amplitude a marked reduction in the overall signal distortion can be achieved. The non linear encoding can significantly improve the pulse code modulation, SNR ratios.

→ Delta Modulation:



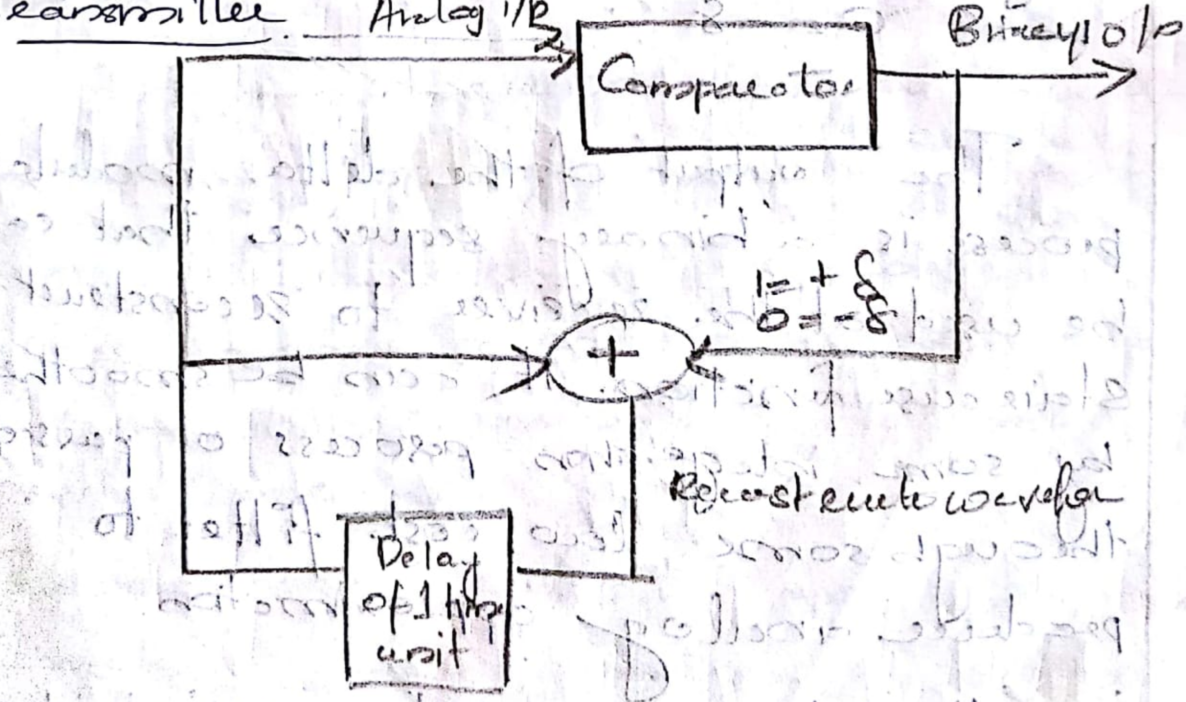
→ Digital modulation:

An analog i/p is approximated by a staircase function that goes up or down by a quantisation level delta at each sampling interval since the staircase function has binary behaviour the output can be represented by a single binary digit for each sample i.e., a pitch is produced by approximating the derivatives of the analog signal rather than its amplitude.

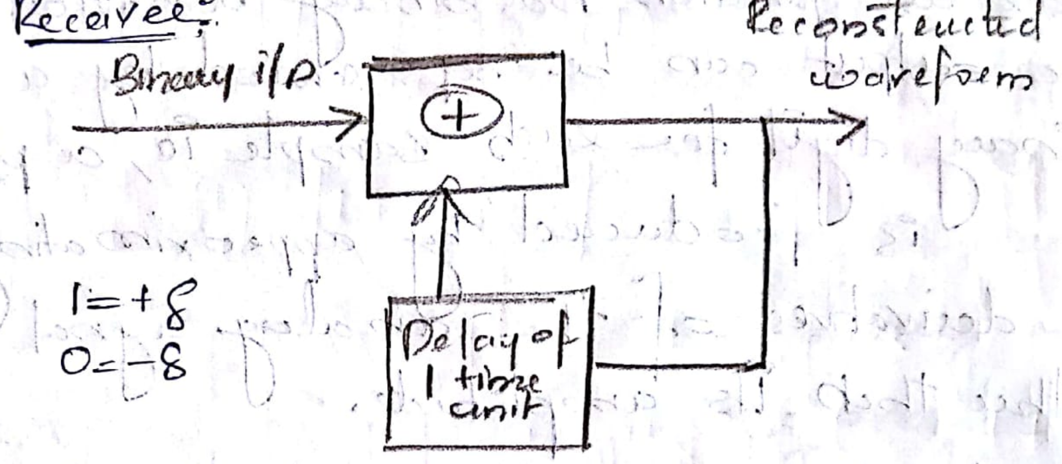
A binary 1 is generated if the staircase function is to go up during the next interval and a binary zero if it is otherwise.

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

Transmitter



Receiver:



At the transmitter at each sampling time the analog signal input is compared to the most recent value of the approximate staircase function. If the value of the sampled waveform exceeds that of the staircase function a binary 1 is generated. Otherwise a zero is generated

1=+8 0=-8

The output 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 passed through some low cost filter to produce analog approximation.

Here two parameters are important

in the delta modulation scheme:

1> Size of step assigned to each binary digit δ.

2> Sampling rate.

Advantages of Digitising:

1> We can use repeaters instead of amplifiers and hence there is no additive noise.

2> Conversion to digital signalling allows the use of more efficient digital switching techniques.

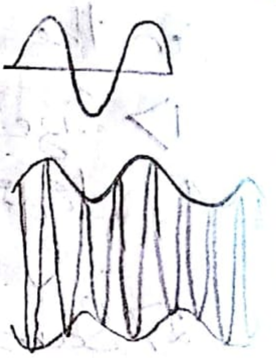
4/10/2017 Wed

Reasons for analog modulation.

In many occasions you need higher frequencies for analog transmission so you have to modulate it with a carrier wave. In wireless communication it is almost impossible to transmit the base band signals with the practical antennas

* The modulation makes it possible for frequency division multiplexing which is a very effective way of transmission.

Amplitude Modulation (AM)



The signal

$$s(t) = [1 + n_a x(t)] \cos 2\pi f t$$

where $\cos 2\pi f t$ is the carrier.

$x(t)$ = input signal, and both are normalised to unit amplitude

n_a is the modulation index which is the ratio of the amplitude of the input signal to the carrier. This scheme is known as double sideband transmitted carrier (DSBTC). The other techniques are frequency modulation & phase modulation. Both are variations of angle modulation.

For

$$\boxed{s(t) = A_c \cos(2\pi f t + \phi(t))}$$

and for phase modulation

$$\boxed{\phi(t) = n_p m(t)}$$

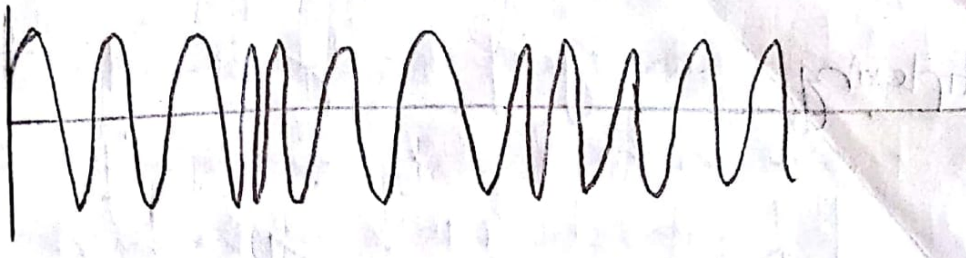
where ~~$\phi(t)$~~ n_p is phase modulation index

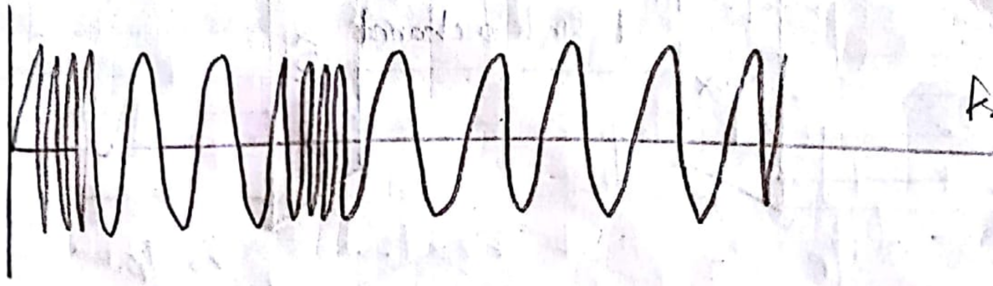
For frequency modulation

$$\boxed{\phi'(t) = n_f m(t)}$$

Scanned by CamScanner

$n_f \rightarrow$ freq modulation index.





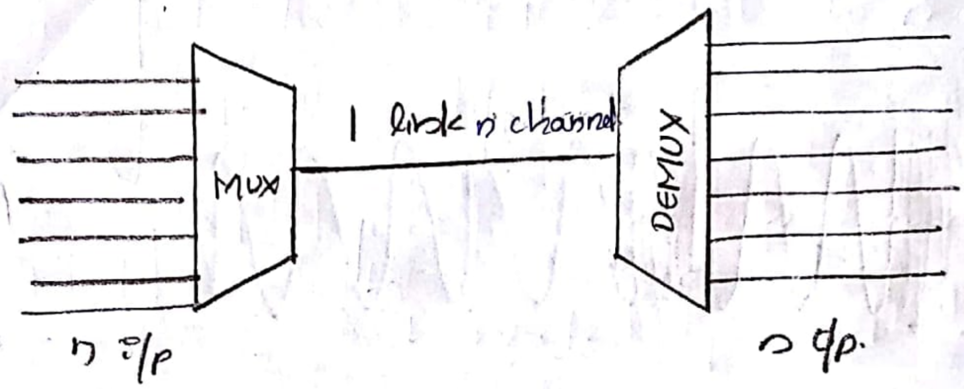
It is difficult to distinguish phase modulation & freq uency modulation without the knowledge of modulation function. Just as the amplitude modulation both frequency modulation and phase modulation result in a signal whose bandwidth centred at the carrier frequency.

~~Multiplexing~~

7/10/2017
wed.

MODULE 4

Multiplexing



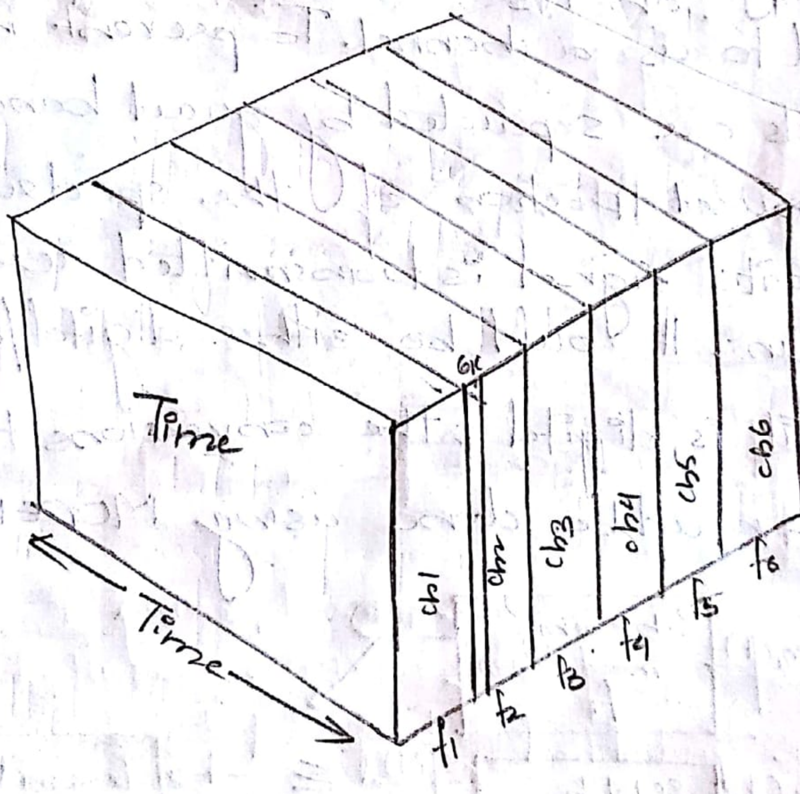
With two devices connected by a point to point link care should be taken so that the data link do not become a bottleneck between the two stations.

For efficiency it should be always possible to share the capacity of the data link. A generic term for such share is called as multiplexing. It is commonly applied between trunks on long hold networks with high capacity and also in fibre coaxial or microwave links.

It is used because higher the data rate the more cost effective the transmission facility become. ie, for a given distance the cost per kbps declines with an increase in the data rate of transmission facility.

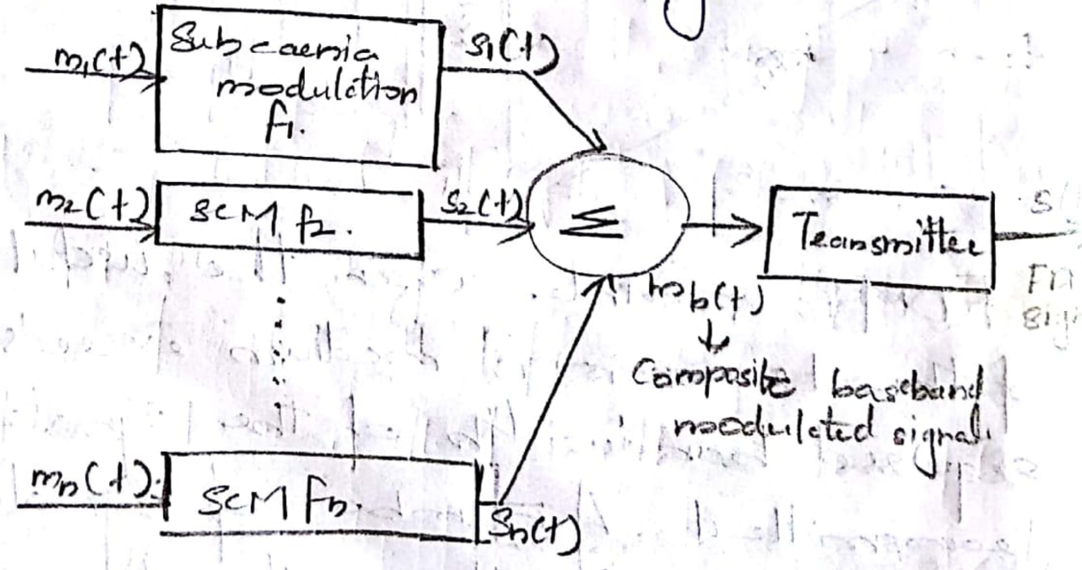
In most cases the individual data communication devices require relatively modest data rate support.

→ Frequency Division Multiplexing (FDM).

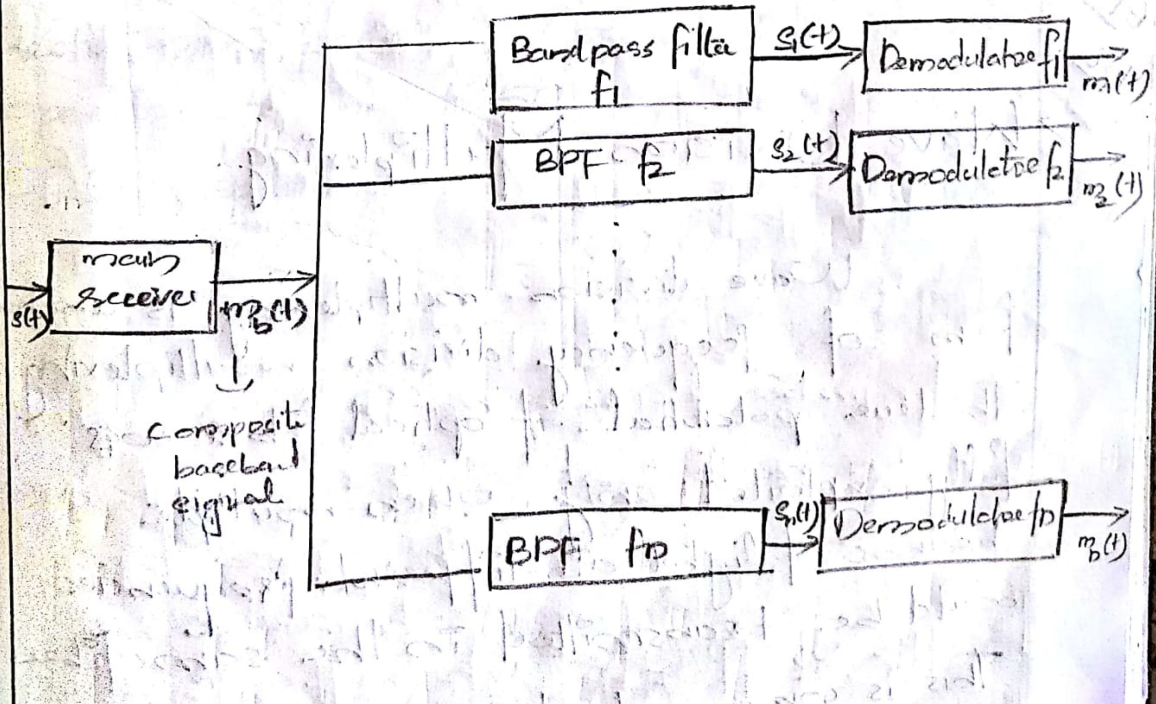


FDM is possible when the useful bandwidth of the transmission medium exceeds the required bandwidth of the signals to be transmitted. A number of signals can be carried simultaneously if each signal is modulated into a different carrier frequency and if the carrier frequencies are sufficiently separated so that the bandwidths of the signals do not significantly overlap, ~~as~~

As is shown in the figure ~~six~~ 6 signals source can be fed into a MUX which modulates each signal to a different frequencies ($f_1, f_2, \ldots$). Each signal require a certain bandwidth centered on its carrier frequency referred to as a channel. To prevent interference channels are separated by guard bands which are unused portion of the spectrum. The composite signal is transmitted across the medium. It could be either digital/analog. If ~~an~~ it is digital, the conversions to analog must be done using MODEM.



A number of analog signals which 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 and 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 bandwidth of the 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.



At the receiving end the FDM signal is demodul[ated] to retrieve the $m_b(t)$ which is passed through n bandpass filters each filters centred around f_i to produce the individual signals $m_1(t)$, m_2, ... $m_n(t)$. As it is evident this multiplex[ing] technique has two problems to face.

1> Crosstalk b/w adjacent frequency channels. Sufficiently could guard bands prevent it to a good extent.

2> Intermodulation noise

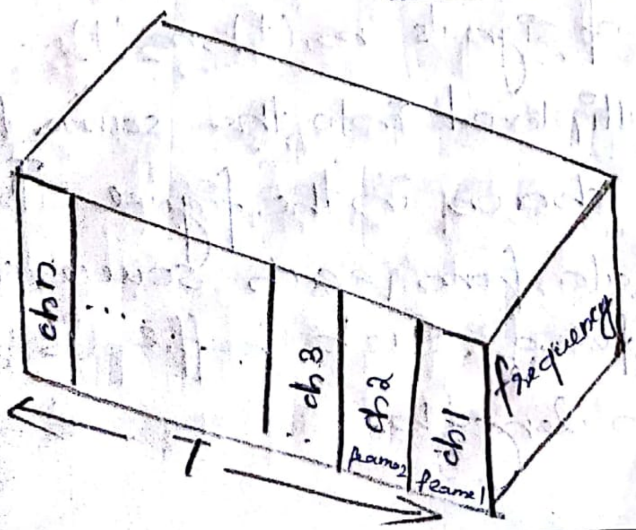
.../10/2017 Fri

→ Wave Division Multiplexing.

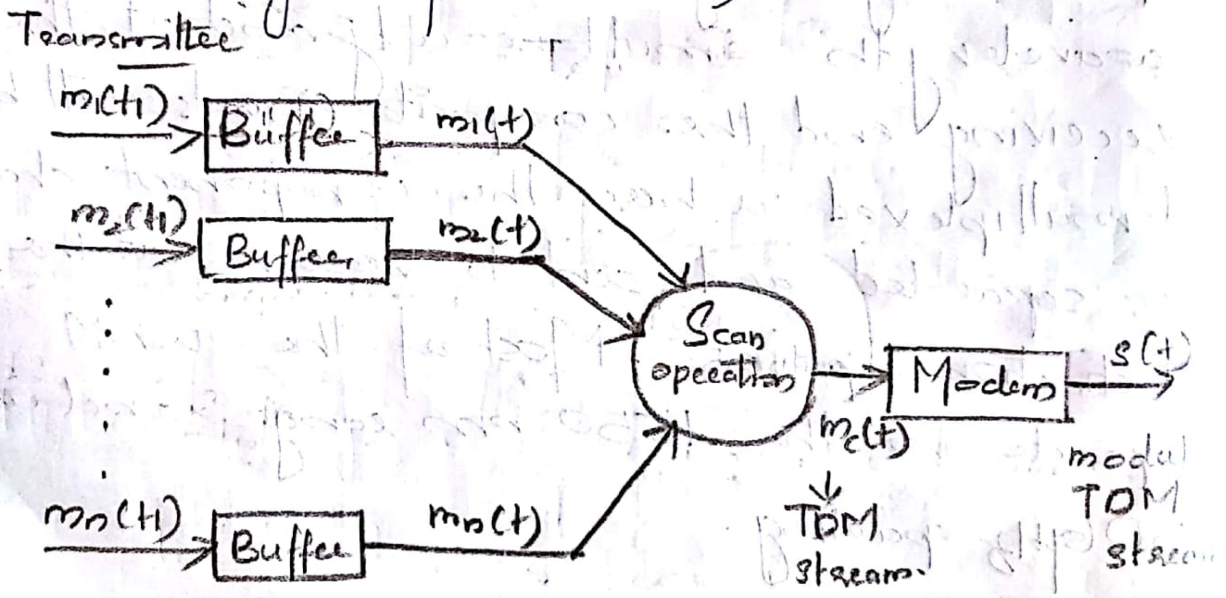
Wave division multiplexing is another form of frequency division multiplexing. The true potential of optical fibre was fully exploited only when multiple beams of light at different frequencies could be transmitted in the same fiber. This is commonly known as wavelength division multiplexing.

It was proved in 1997 at Bell Lab a typical WDM has the same general architecture as FDM s/ms. A number of sources will generate laser beams at different wavelengths are sent to a multiplexer which consolidates the sources for transmission over a single fibre line. Optical amplifiers which are spaced typically 10 kms apart amplify the wavelength simultaneously and at the receiving end the composite signals will be demultiplexed where the component channels are separated and sent to receivers at the destination point. Most of the WDM operate in the 1550 nm range and use a 50 GHz spacing.

Synchronous Time Division Multiplexing



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 at bit level or block level (as bytes of characters).

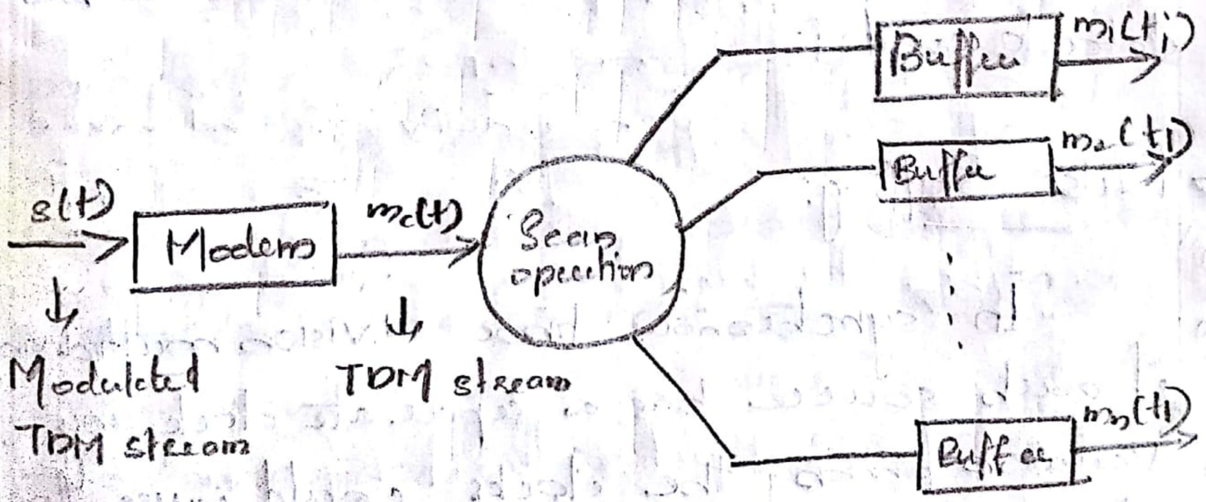


A number of signals $m_1(t), m_2(t), \ldots m_n(t)$ can be multiplexed into the same transmission medium as shown in the figure. The incoming data from each source are briefly buffered. Each buffer is 1 bit or 1 character in length.

Scanned by CamScanner

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 $m_i(t)$ where $i = 1$ to m. The digital signal $m_c(t)$ may be transmitted directly or passed through a modem if we require an analog signal to be transmitted.

Receiver



The transmitted data may have a format that is the data are organised into frames. Each frame contains a cycle of time slots, the sequence of slots dedicated to ~~each~~ one source from frame to frame is called a channel.

The byte interleaving technique can be used with asynchronous of synchronous sources. At the receiver, the interleaved data are demultiplexed and routed to the appropriate destination buffer. For each input source $m_i(t)$ there is an identical output destination which will receive the output data at the same rate at which it was generated. The word synchronous is used because the time slots are preassigned to sources and are fixed.

The time slots for each source are transmitted whether or not the source has data to send.

Pulse Stuffing:

1/10/2017 Mon

In synchronous time division multiplexing if each source has a separate clock any variation among the clocks could cause loss of synchronization. Also in certain cases the data rates of input data streams are not related by a single 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 the multiplexer excluding the frame bits will be higher than the sum of maximum instantaneous incoming rates. The extra capacity is used by stuffing dummy bits or pulses into each incoming signal until its rate is raised to that of a locally generated clock signal. The stuffed pulses are inserted at fixed locations in the multiplexer frame format so that they can be identified and removed at the demultiplexer.

Statistical Time Division Multiplexing

In the case of synchronous time division multiplexing the slots may be sometimes wasted when certain devices will not have data to send. To avoid this wastage there is an alternative scheme named as statistical time division multiplexing. This uses the property of data transmission by dynamically allocating the time slots on demand. Even if there are n input lines only k time slots are available on the TDM frame where k<n. For input the function of the multiplexer is to scan the input buffers, collect the data until a frame is filled and then send the frame. On the output, the multiplexer receives a frame and distributes the slots of data to the output buffers.

Since in this scheme all the attached dev.. are not transmitting all the time the data r.. of the multiplexer line is less than the sum of the data rates of the attached devices.

Thus ~~in~~ a statistical time division multiplex.. can use a lower data rate to support as m.. devices, as ~~single~~ synchronous multiplexer.

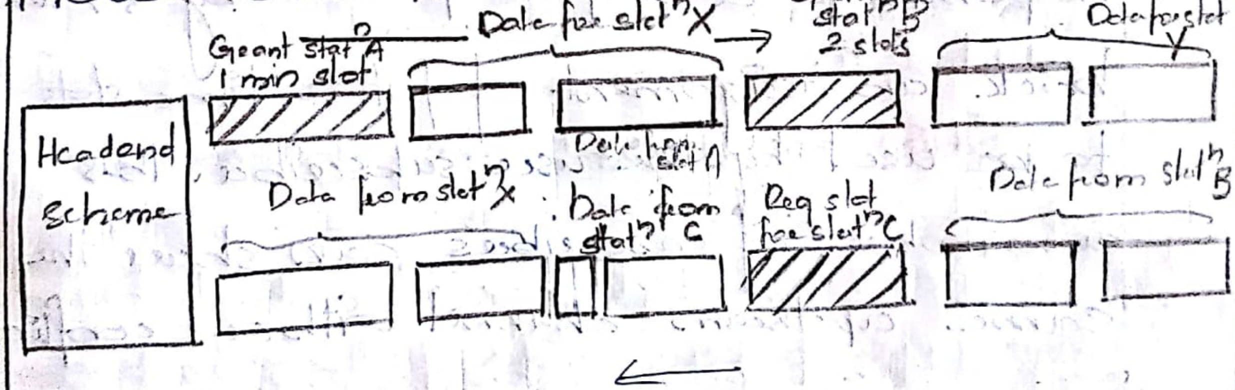
or in other words, statistical TDM can support more devices in the same data rate as compared to synchronous MUX.

Since data arrive from and are distribut.. in an unpredictable manner, address info is required for proper delivery and hence there is more overhead per slot for STDM. since the slot has to carry address as well as data.

The frame structure used by statistical M.. has impact on performance. Usually statistical TDM use synchronous protocol as SDFC fram.. In this the source is identified by the address. The length of data frame is available and its end is marked by 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 the length of data for each source. Thus statistical TDM subframe consists of a sequence of data fields each labelled with address and length.

(study frame formats)

How Cable modem works?



Cable modem:

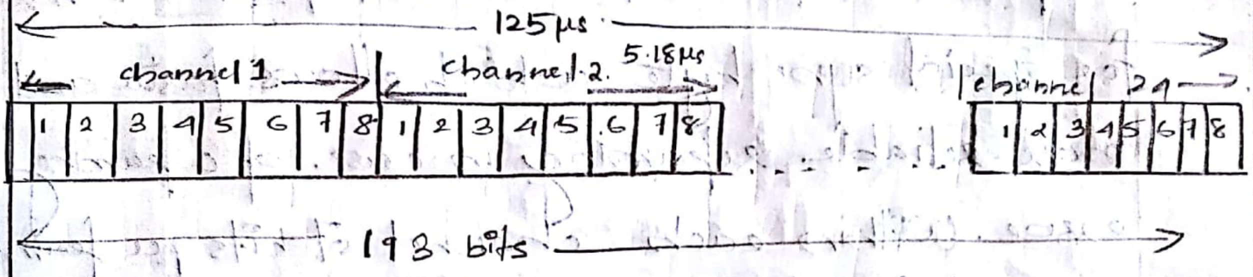
Eg of Statistical TDM

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 and there is an allocating scheme for transmission. A form of statistical TDM is used. From the headend to the subscriber the scheduler delivers data in the form of small packets. If more than one subscriber is active each subscriber gets only a fraction of the downstream capacity. The downstream is also used to grant the time slots to the subscriber. When subscriber has data to transmit it must first request the time slots on the shared upstream channel. Each subscriber is given dedicated time slots for the request purpose. The headend scheduler responds to a request packet by sending back an assignment of future time slots to be used by the user subscriber. Thus a number of subscribers can share the same upstream channel without conflict.

11/10/2017 used

Digital Carrier Systems

Digital carrier s/ms were used for long distance communication for voice grade signals over high capacity transmission links such as optical fibre, coaxial cable and microwave. This happened in the time when telecommunication s/ms were passing onto digital technology by adopting synchronous TDM transmission structures. Later identical hierarchy has been adopted internationally under ITUT. The basis of TDM hierarchy DS-1 transmission format which multiplexes 24 channels.



Each frame contains 8 bits/channel plus the first bit, which is a framing bit used for synchronisation ie, $(24 \times 8) + 1 = 193$ bits.

For voice transmission the following rules apply:

→ Each channel contains one word of digitalised voice data, the analog voice signal is digitalised using pulse code modulation at a rate of 8000 samples/sec with a frame length of 193 bits. $8000 \times 193 = 1.544$ Mbps.

For five of every six frames 8 bit PCM samples are used.

For every sixth frame each channel contains a 7 bit PCM word plus a signalling bit. The signalling bits form a stream for each voice channel which contains network control and calling information.

→ The same DS-1 format is used to provide digital data service. In this case 23 data channels are provided. The 24th channel position is reserved for special sync byte which allows faster and more reliable 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 the channel contains user data or s/m control data.

A data rate of 56 ~~six~~ Kbps can be provided per channel. Lower data rates are provided by sub rate multiplexing. For this technique an additional bit is robbed from each channel to indicate which subrate multiplexing rate is being provided.

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

~~For higher~~

Higher level multiplexing is achieved by interleaving ~~inputs~~ bits from DS-1.

SONET/SDH:

- Synchronous Optical Network. (SONET)
- Synchronous Digital Hierarchy (SDH).

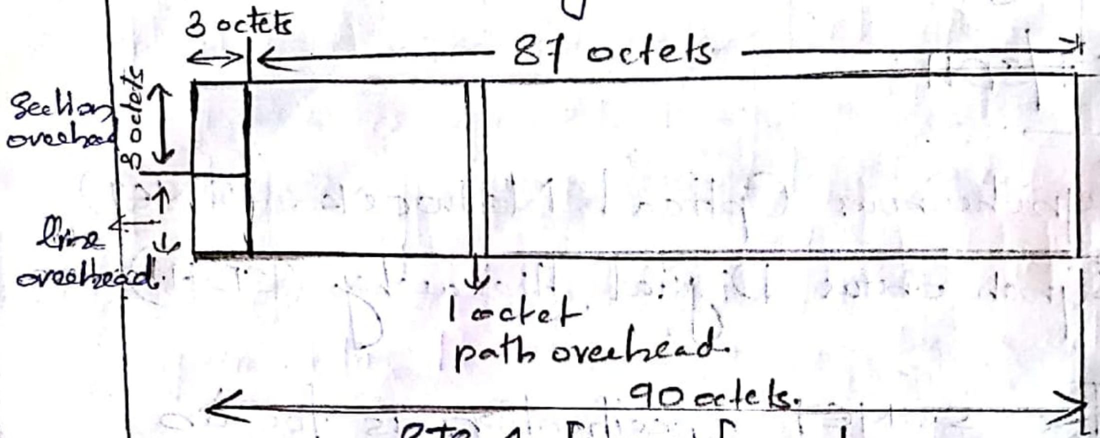
The SONET specification is for an optical transmission interface proposed by Bell Core and was later standardised by ANSI-A.

A compatible version referred to as synchronous digital hierarchy (SDH) has been published by ITUT. SONET is intended to provide specification for high speed digital transmission through optical fibre. ~~The~~

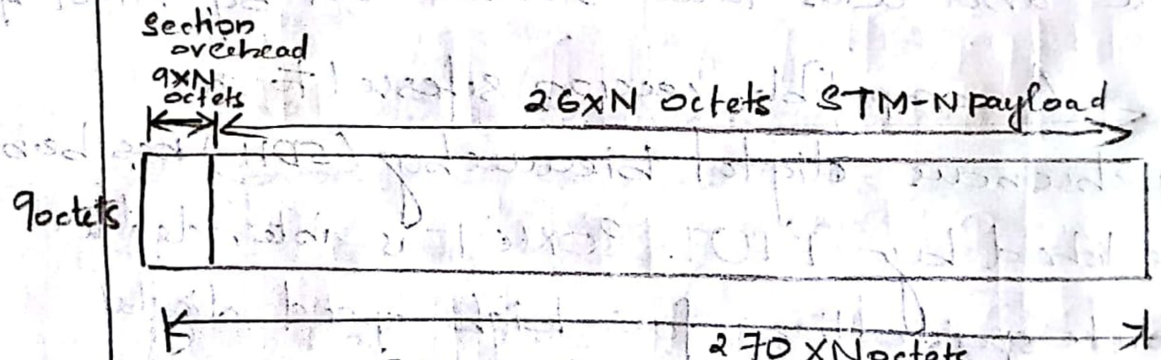
The SONET hierarchy

The SONET defines a hierarchy of standardized data rates at the lowest level referred as STS-1 (Synchronous Transport Signal) or OC-1 (Optical Carrier Level) is 51.84 Mbps. These are used for lower rate signals such as DS-1C, DS-2 etc.

The multiple STS-1 signals can be combined to form STM-N signals.



STS-1 frame format



STM-N format

→ The Frame format

The basic SONET ~~cell~~ building block is the SDH-SONET format which consist of 810 octets ~~and~~ which is transmitted every 1 with an overall data rate of 51.84 Mbps. The first 3 columns of the frame are devoted to overhead octets. Overhead octets are section overhead, line overhead and path overhead. The remainder of the frame is payload. The payload include a column of path overhead line overhead, which contains a pointer that indicates where the path overhead starts.

22/10/2017
Mon

Module 5

Data Communication Techniques.

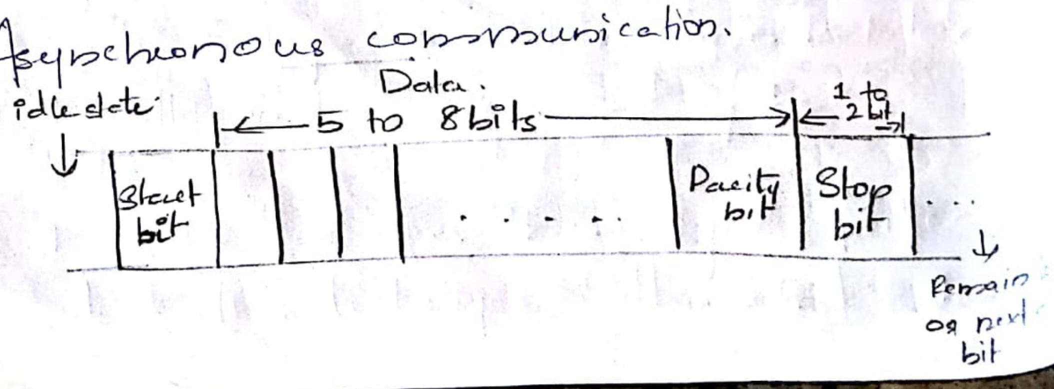
For any two devices to communicate using a transmission media, there should be agreement to exchange data. The timing i.e., the wait duration and spacing of bits must be the same for transmitter and receiver. The two common methods usually employed are:

Synchronous data communication

Asynchronous data communication

A number of errors that seeks in is also another problem. The error rate has to be limited so that there is effective communication b/w the transmitter and receiver.

1) Asynchronous communication.



In asynchronous transmission instead of sending uninterrupted strings of data, data is transmitted one character at a time, where each character is 5 to 8 bits in length.
Timing or synchronization need to be maintained only within each character and the receiver has an opportunity to synchronize at the beginning of each new character.

When no character is being send or transmitted the line remains in an idle state. Suppose we are using NRZ-L signalling method which is common for asynchronous transmission the idle state could be the presence of a negative voltage on the line.

The beginning of the character is signalled by a start bit with a value of binary zero. This is followed by 5 to 8 bits that actually mean 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 an odd parity is used.

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

Asynchronous transmission is cheap and simp but as an overhead of 20% due to the presence of start bit and end bit which doesn't contain any information.

2> Synchronous Communication

Preamble — Postamble

8 bit flag	Control fields	Data field //	Control field.	8 bit flag

info on framing

With synchronous transmission the bits are transmitted in a steady screen witho start or stop rules. The blocks may be man bits in length. To prevent timing drift between transmitter and receiver their clocks must be synchronised. One method is to provide a separate clock line between transmitter and receiver. One of the

Other side will use the regular pulses as a clock. This technique will work for short distances, but for long distances the clock pulses are also subject to the same impairments as data and timing errors can occur.

The other alternative is to embed the clocking information in the data signal itself.

With synchronous transmission there is another level of synchronization required to allow the receiver to determine the beginning & ending of data.

Each block of data begins with a data called preamble bit packet and ends with

In addition other bits are added to the block which carry control info used in the data link layer. The data alongwith preamble, postamble and control info is called a frame. The exact format of frame depends on the data link control procedures used.

Typically the frame starts with a preamble called flag (8 bit). The same flag is used as postamble.

The receiver looks for the occurrence of the flag pattern to signal the start of the frame which is followed by control fields, data fields ~~and~~, more control fields and is followed by postamble flag.

For sizeable blocks of data synchronous transmission is more effective than asynchronous transmission (20% or more overhead

25/10/2017 Wed.

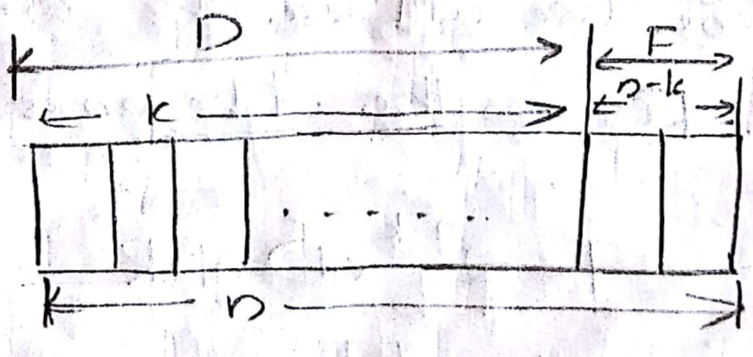
Parity Check

There is even parity check and odd parity check. This is the simplest error detection scheme. It can detect a single error. Even parity is usually used for synchronous transmission and odd parity for asynchronous transmission. The user of parity bit is not fool proof since noise impulses are often long enough to discard more than one bit especially at higher data rate

→ Cyclic Redundancy check: (CRC)

Given a k bit block of bits/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 predetermined number.

The receiver then divide the incoming frame by that number and if there is no remainder assumes that there is no error. Now we define T to be an n bit frame that is to be transmitted. Now D is the k bit block of data or message i.e, the first 'k' bits of T. F is the n-k bit of (FCS) T.



3
2351
1.
235×10
+1
↓
1-3

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

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

Effectively padding D with sufficient number of 0's and adding F effectively concatenating D and F. ~~Now~~ We want T to be exactly divisible by P. Suppose divide 2^{n-k} D by P ie, $\frac{2^{n-k} \cdot D}{P}$

$\frac{7}{2} = 3$

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

There is a quotient and a remainder, since the division is modulo 2. the remainder is always atleast one bit shorter than the divisor. We will use this remainder as our FCS so the T becomes

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

Let us see whether $\frac{T}{P}$ has no remainder

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

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

$$= Q + \underbrace{\frac{R}{P} + \frac{R}{P}}_{0 \Rightarrow \text{XOR same bits} \Rightarrow 0} \quad (\text{from}(1))$$

So, $\frac{T}{P} = Q$.

Hence there is no remainder and T is exactly divisible by P. Thus this FCS is easily generated by $\left(\frac{2^{n-k}D}{P}\right)$ and use the $(n-k)$ bit remainder as FCS. On reception the receiver will divide T by P. and if it gets no remainder conclude that there is no error else it shows that there is error. The pattern P is chosen to be 1 bit longer than the desired FCS and the exact bit pattern chosen depends on the type of errors detected.

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). This should be divided by P.

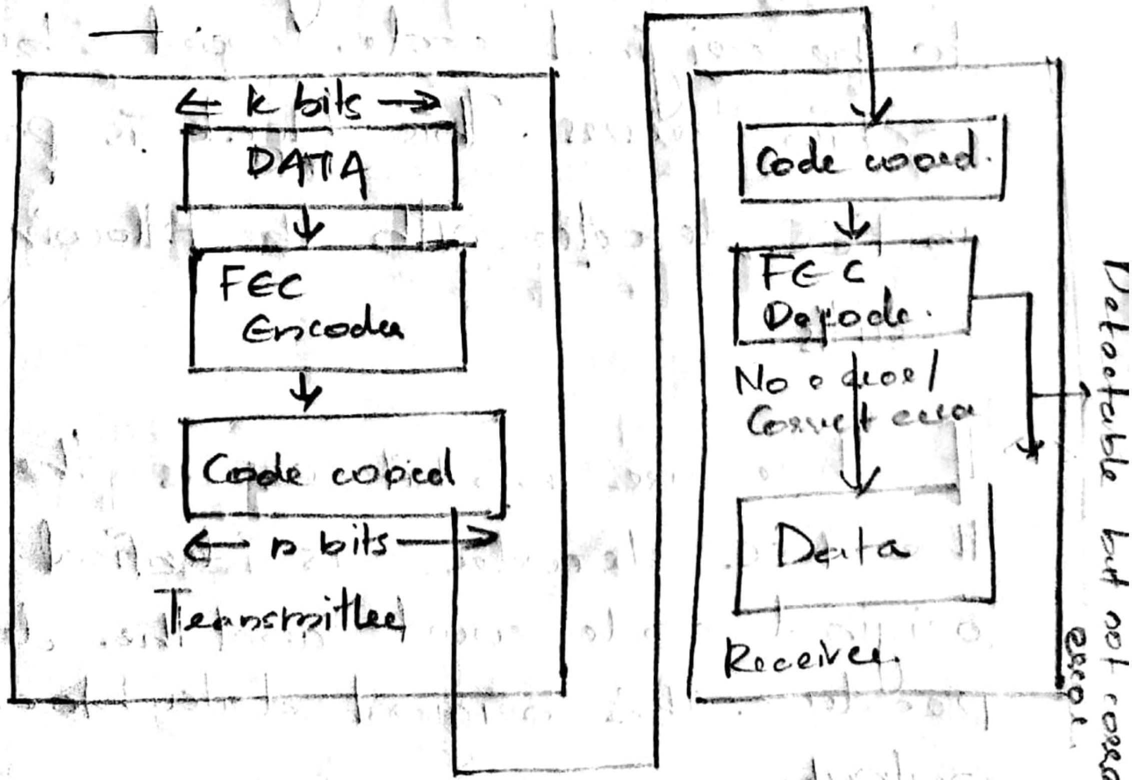
```
              110001010
        _______________
110101 | 101000110100000
         110101
         ------
         011101
          110101
          ------
          0011111
             110101
             ------
             0010110
               110101
               ------
               0011100
                 110101
                 ------
                 0001110
```

~~If there~~ is ~~no error~~

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 other transmitted bits. Typically for a block of K bits the error detection algorithm needs 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 and (n-k) bits of error detection code. The receiver performs the same error detecting calculation on the data bits and compares this value with the value of the error detection type code. There is an error detected only if there is a mismatch.

27/10/17
Fri.

ERROR CORRECTION



The figure shows a general method for correction of codes with forward error correction method. On the transmission end each k bit block of data is mapped into an n bit block (where $n > k$) ~~code~~ is called a code word using forward error correction encoder (FEC). 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 demodula 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 input to the FEC decoder is identical to the original code word and the decoder produce this original data block as output.

<2> For certain error patterns, the decoder detect and correct the errors and the FEC decoder is able to map this block into the original data block.

<3> For certain error patterns, the decoder can detect but not correct the errors.
In this case, decoder reports an uncorrectable error.

<4> For certain ~~erase~~ error patterns, the decoder do not detect any error and map 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 and adds (n-k) ~~b~~ check bits to that block and produce an n bit block. These extra bits helps to correct any bit error which might have come across during transmission.

→ Hamming Distance:

The hamming distance $d(V_1, V_2)$ between two n bit binary sequences v_1 and v_2 is the number of bits in which V_1 and v_2 disagree

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.

Eg: Data Block $(k=2)$	Code word $(n=5)$
00	00000
01	00111
10	11001
11	11110

$d(00100, 00000) = 1$ ✓ → hamming distance

$d(00100, 00111) = 2$

$d(00100, 11001) = 4$

$d(00100, 11110) = 3$

From the example, the transmitted message 00100 has a minimum distance with the first code word and it is corrected as the first code word. 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 $\left(\frac{n-k}{k}\right)$ is called the redundancy of the code and the ratio of data bits to the total bits $\left(\frac{k}{n}\right)$ is called the code rate.

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

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

It can be shown that for any given positive integers 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 $\leq (t-1)$ bits can be corrected. Errors of t bits can be detected but need not always be corrected.

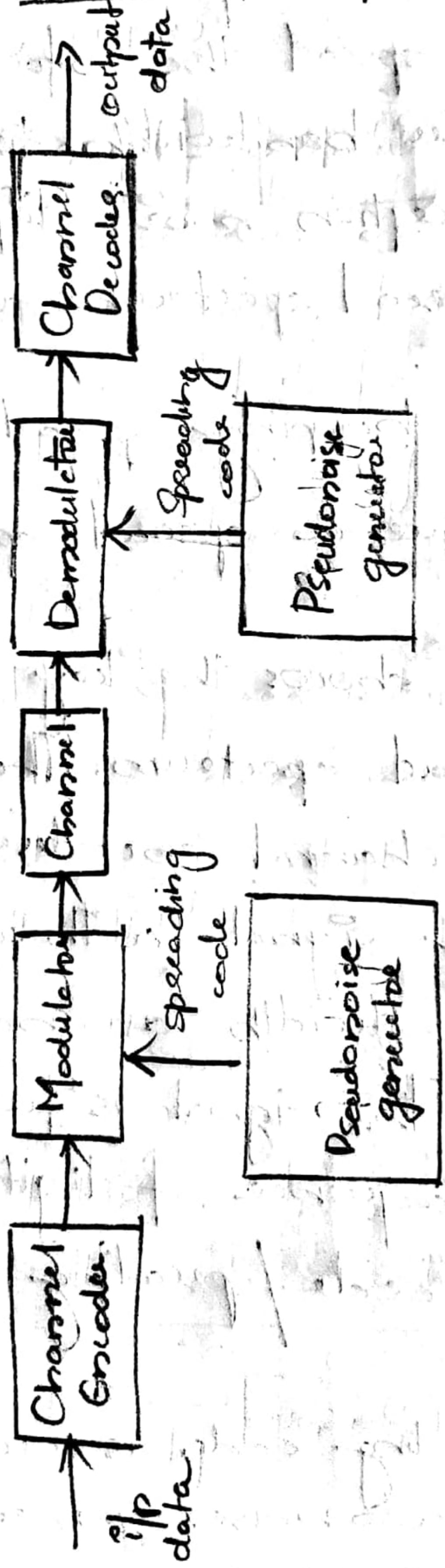
1/11/2017

1/11/2017. COed

MODULE 6.

ch 8

Spread Spectrum Techniques



The spread spectrum technique was developed initially for military and intelligence requirements. The basic idea is to spread the information signal over a wider bandwidth to make jamming and interception more difficult. Two types of spread spectrums are used:

1> Frequency hopping spread spectrum and
2> Direct sequence spread spectrum.

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

Spreading code is usually generated by a pseudonoise generator or a pseudo random number generator. The effect of this modulation is to spread the spectrum or to increase the bandwidth 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 and multipath distortion.

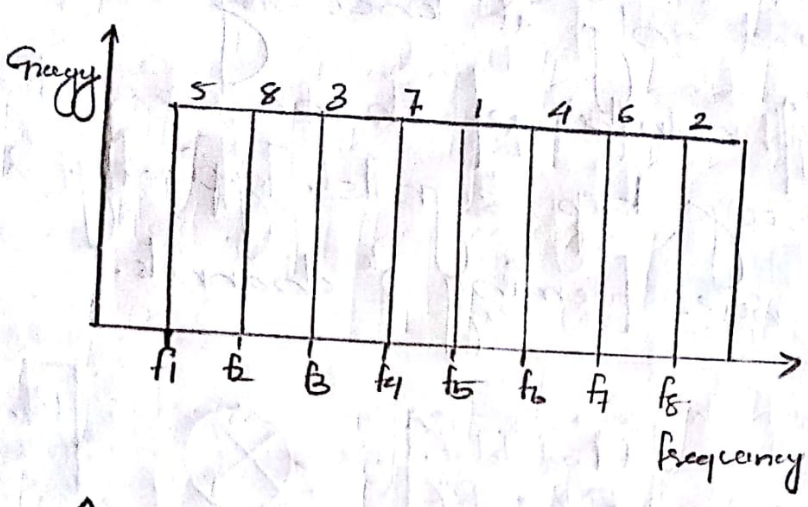
2> It can be used for hiding and encrypting signals. Only a recipient who knows the spreading code can recover the encoded information.

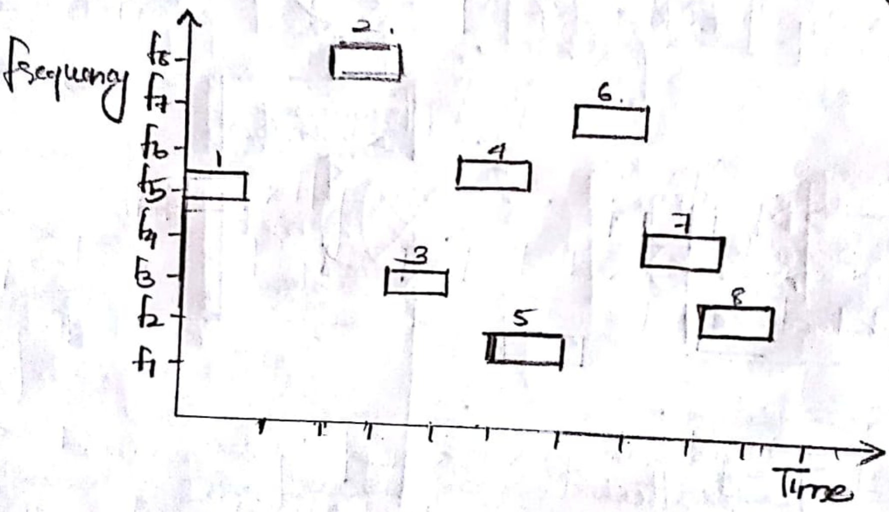
3> Several users can independently use the same higher bandwidth with very little interference. This property is used in cellphones with a technique known as Code Division Multiple Access (CDMA).

The pseudo random numbers are generated by an algorithm using some initial value known as seed value. The algorithm is a deterministic algorithm and produce a sequences of numbers which though not are statistically random is good enough as a random code and hence known as <u>pseudo random numbers</u>. Without knowing the algorithm and the seed it will be impractic to predict the sequence. Hence only a receiver which shares this information with a transmit will be able to decode the signal successfully.

3/11/2019 Fri

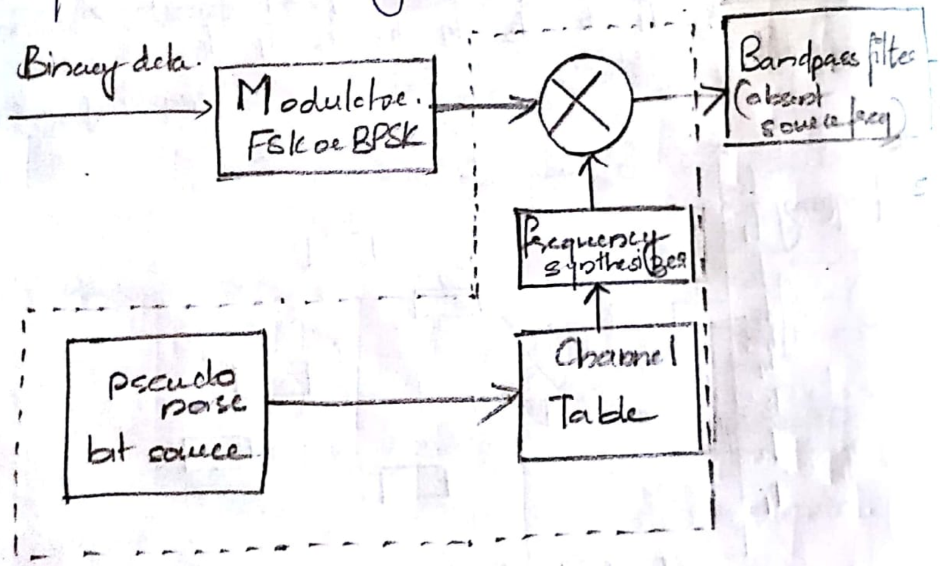
Frequency Hopping Spread Spectrum (FHSS).



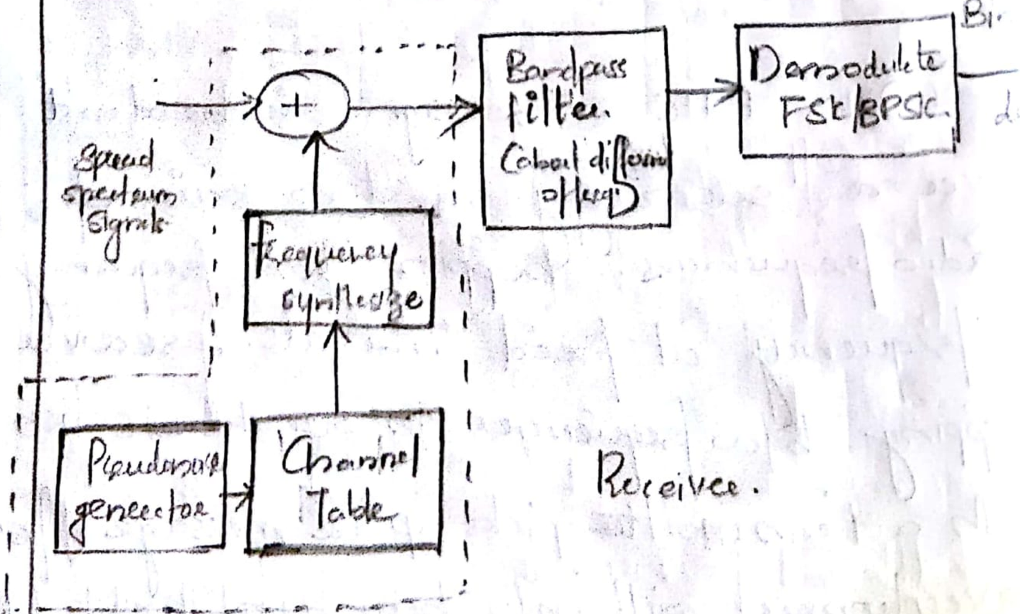


With FHSS the signal is broadcast over a seemingly random series of radio frequencies hopping from frequency to frequency at fixed intervals. A receiver hopping b/w frequencies in synchronisation with a transmitter picks up the message. An eavesdropper will only hear unintelligible blips.

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 signals. Typically 2^k carrier frequencies forming 2^k channels.



Transmitter



Receiver.

A typical FHS is shown in the figure. For transmission binary data are fed into a modulator using some digital to analog encoding scheme, such as FSK/BPSK. The resulting signal is centred on some base frequency or a pseudorandom number source serves as an index into a table of frequencies which is referred as the spreading code. Each k bits of the pseudo random number 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 the same shape but centred on the selected carrier frequency.

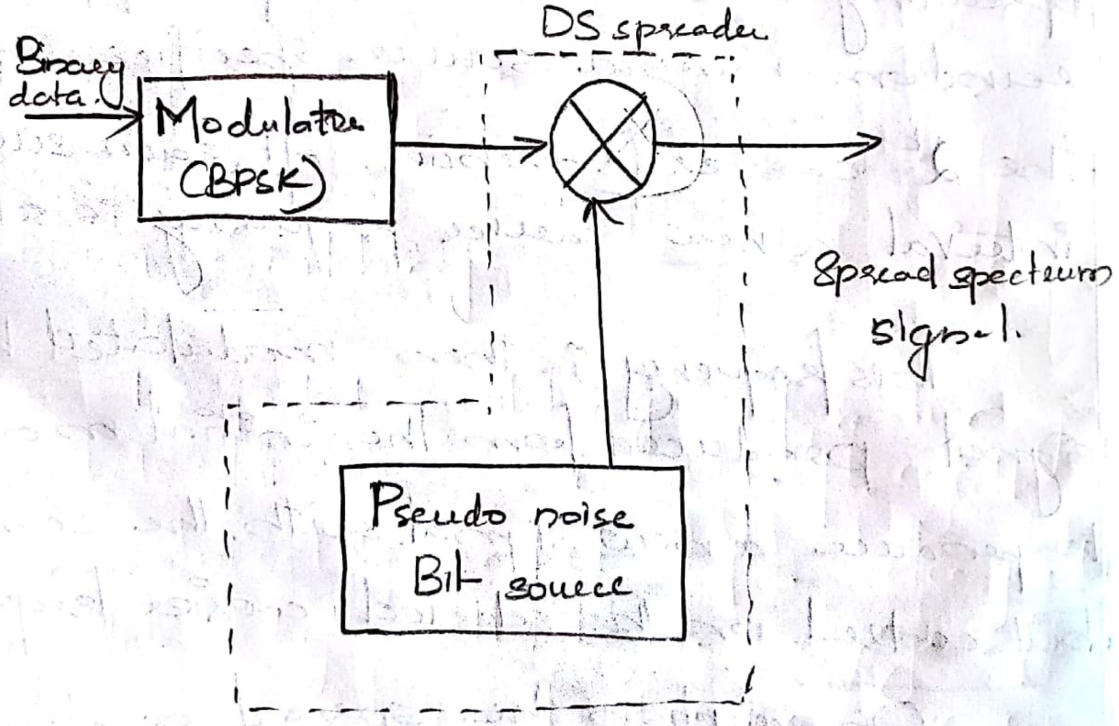
On reception the spread spectrum signal is demodulated using the same sequence of pseudo random number derived and demodulate it to produce the output data.

freq

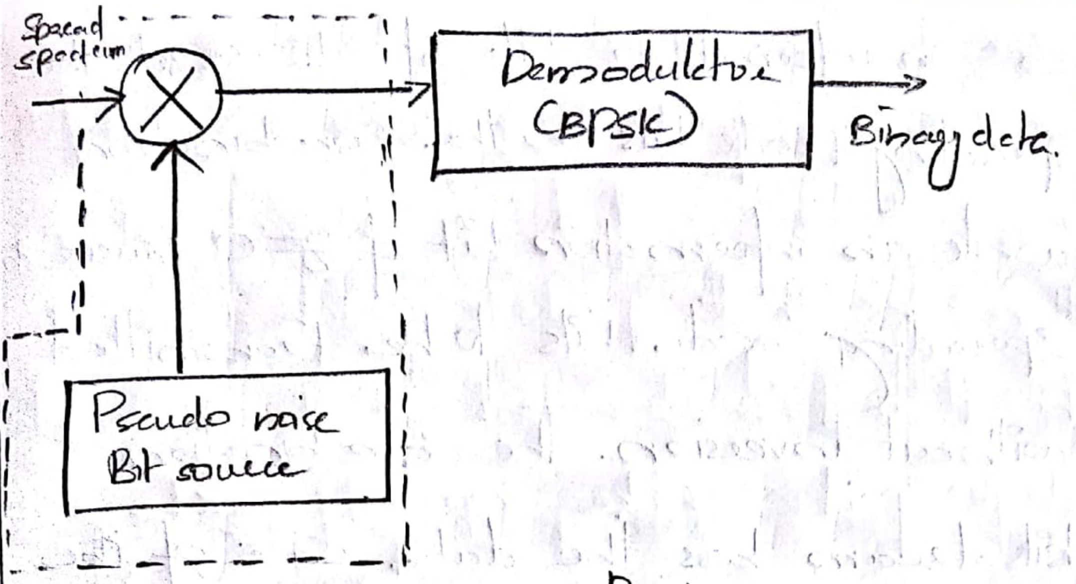
The frequency synthesis generator generates a constant frequency tone whose frequency hops among the set of 2^k freq. with the hopping pattern determined by the k bits from pseudonumber sequence.

6/11/2017
Mon

Direct Sequence Spread Spectrum (DSSS)



Transmitter



Receiver.

With direct sequence spread spectrum each ~~co~~ 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 use a 10 bit spreading code it will spread the signal across a frequency band 10 times greater than a 1 bit spreading code.

One technique with ~~which~~ the DSSS is to combine the digital information stream with a spreading code bit stream using XOR

So, an information bit of 1 inverts the spreading code bits in the combination while an information bit of ~~ze~~ 0 causes the spreading code bits to be transmitted without inversion. The combination bit stream has the data rate of the original spreading code sequence hence it has a wider bandwidth than the information stream.

8/11/2017 Wed

Packet Switching

In packet switching information goes as packets. This is used only for data communication and not for voice grade signals. Packets will have routing information depending on whether it follows a datagram approach or a virtual circuit approach. The packet size also varies. As compared to the circuit switching even when a load is busy packets can be send to the 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 the virtual circuit approach a routing table is made at the source from which the 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 and receive 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

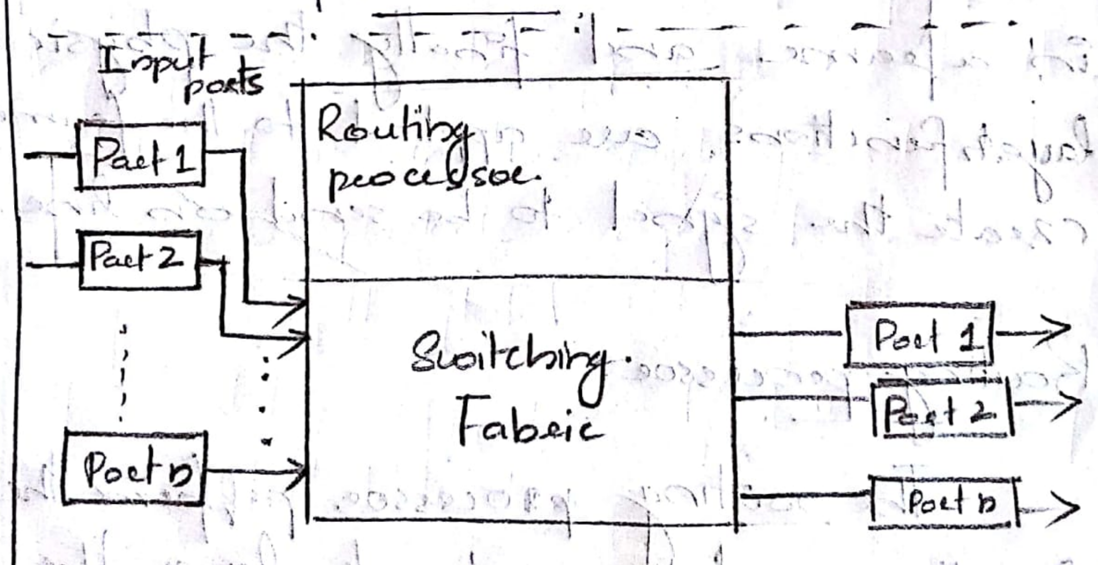
Datagram Approach:-

In this, datagram network the packet is treated independently of all other even when it is a part of multipacket transmission. Packets are known as datagrams in this approach. Datagrams circuit/datagram network is called a connectionless network. It means that packets which does not need information about the connection stage quite unlike circuit switching there is no connection phase/disconnecting phase. Packet switching is usually a buristy traffic. Most of the times the transmission line might be ideal and there might be busy phases and it is congested with packets.

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

In the 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 on the node etc. Due to this in the datagram approach, datagrams travels through multiple paths. They might reach the destination out of sequence. The packets may have to be reorganised according to their sequence numbers.

→ Structure of a Packet Switch:



Input port

An input port the physical & data link function of the packet switch. The bits are constructed from the received signal. The packet is decapsulated.

Errors are detected and corrected.

The port will be having buffers or queues to hold the packet before it is directed to the switching fabric.

Output port

The output port performs the same functions as the input port but in the reverse order. First the outgoing packets are queued, then the packet is encapsulated in a frame and finally the physical layer functions are applied to the frame to create the signal to be send on line

Routing processor.

The routing processor performs the functions of the network layer the destination address is used to find the address of the next port and the output port number to which the packet is send out.

Switching fabric.

The most difficult task in packet switching 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 the packet delivery. Packet switches are specialised mechanisms which uses a variety of switching fabric. The simplest type is a crossbar switch.