Q.1. *Attempt any four of the fallowings-*

a. *Explain an elecronic communiaction system with the help of a block diagram.* 05

Ans: Block diagram of communication system

In this article of block diagram of communication system we learn different blocks in a communication system and their use.

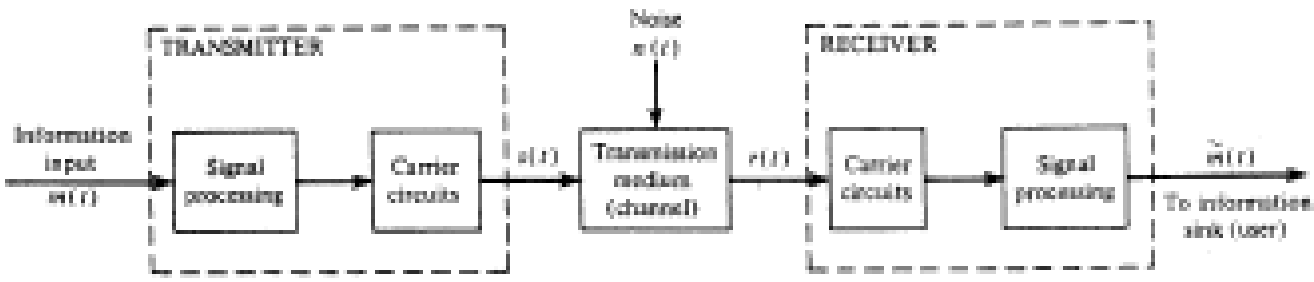


Figure 1-1 Communication system.

Block Diagram of a Communication System.

a. Transmitter & Encoder

Transmitter is the first block in the block diagram of communication system. Here the message which is to be sent is generated. This message is sent to encoder.

Encoder is the second block in the block diagram of communication system. This block performs encoding. it is the process in which the actual message is converted in to symbols for transmission. In this a sequence of characters are put in a specialized format for efficient transmission. If you transmit a message with out encoding then there may be a chance of other persons trapping the message by knowing the frequency of transmission. But if you transmit it through encoding even though they trap it they cant decode.Only the person who know how it is encoded can decode it. This is done to increase the security.

b. Noisy Channel

This is the third block in the block diagram of communication system. Noisy channel is nothing but the medium through which the message is transmitted. Messages are conveyed through this channel. Different channels have different strengths and weaknesses. Each channel has its own frequency and different applications have different operating frequencies. For example mobile communications operate in the vicinity of 800 MHZ and FM radio operates in the vicinity of 100 MHZ.

c. Decoder

This is the fourth block in the block diagram of communication system. Decoder is used to decode the encoded message and retrieve the actual message. Decoding must be done correctly . If this part is not performed well then the message which is received might not be correct. As i said earlier the person who knows how to encode can only do the decoding.

This encoding and decoding will be very help full in military and mobile communications. For example in a war if some message is sent by American army to one of its Tanker saying that to destroy the terrorists building. If this message is trapped by the terrorists they are going to escape. And this happens when no encoding is done. Now if the message is sent through encoding then though the message is trapped by terrorists they cannot decode it. Thus the tanker could successfully bomb the target.

d. Receiver

This is the final block in block diagram of communication system. This can be said as the target to which the information need to be delivered. In the above example American army is the transmitter and the tanker is the receiver.

b. *compare Analog and Digital communication.* 05

Ans:

Analog communication systems, amplitude modulation (AM) radio being a typifying example, can inexpensively communicate a bandlimited analog signal from one location to another (point-to-point the largest possible signal-to-noise ratio for the demodulated message. An analysis of this receiver thus indicates that some residual error will always be present in an analog system's output.

Although analog systems are less expensive in many cases than digital ones for the same application, \ digital systems offer much more efficiency, better performance, and much greater flexibility.

, Efficiency: The Source Coding Theorem allows quantification of just how complex a given message source is and allows us to exploit that complexity by source coding (compression). In analog communication, the only parameters of interest are message bandwidth and amplitude. We cannot exploit signal structure to achieve a more efficient communication system.Performance: Because of the Noisy Channel Coding Theorem, we have a specific criterion by which to formulate error-correcting codes that can bring us as close to error-free transmission as we might want. Even though we may send information by way of a noisy channel, digital schemes are capable of error-free transmission while analog ones cannot overcome channel disturbances; see this problem fo comparison.

Flexibility: Digital communication systems can transmit real-valued discrete-time signals, which could be analog ones obtained by analog-to-digital conversion, and symbolic-valued ones (compute) data, for example). Any signal that can be transmitted analog means can be sent by digital means, with the

only issue being the number of bits used in A/D conversion (how accurately do we need to represent signal amplitude). Images can be sent by analog means (commercial television), but better communication performance occurs when we use digital systems (HDTV). In addition to digital communication's ability to transmit a wider variety of signals than analog systems, point-to-point digital systems can be organized into global (and beyond as well) systems that provide efficient and flexible information transmission. Computer networks, explored in the next section, are what we call such system today. Even analog-based networks, such as the telephone system, employ modern computer networking ideas rather than the purely analog systems of the past.

c . *compare TDM and FDM.* 05

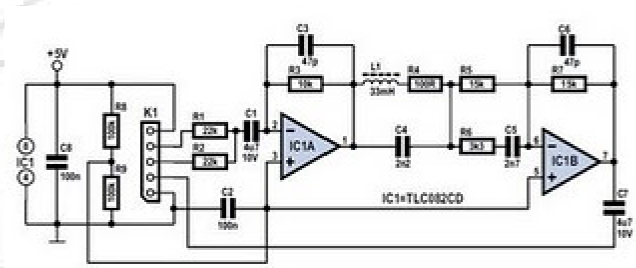
Ans:

Sr. No.	FDM	TDM
I.	The signals which are to be multiplexed are added in the time domain. But they occupy different slots in the frequency domain.	The signals which are to be multiplexed can occupy the entire bandwidth but they are isolated in the time domain.
2.	FDM is usually preferred for the analog signals.	TDM is preferred for the digital signals.
3.	Synchronization is not required.	Synchronization is required.
4.	The FDM requires a complex circuitry at the transmitter and receiver.	TDM circuitry is not very complex.
5.	FDM suffers from the problem of crosstalk due to imperfect band pass filters.	In TDM the problem of crosstalk is not severe.

d. *Explain Pre-emphasis in FM.* 05

Ans:

Pre-emphasis for FM Transmitter
This Pre-emphasis circuit was specially designed to be used with the FM Audio Transmitter. The *circuit* uses a dual opamp. The first opamp (IC1A) functions as a *mixer* and a buffer for the following correction network. The input sensitivity can be adjusted with the help of R3 (a lower value reduces the sensitivity). The 50 µs correction for the pre-emphasis is carried out by C5 and R6. IC1B buffers the signal before it is fed to the *transmitter* via K1.



Since the **FM transmitter** is a mono version, a *19 kHz filter* has been included to prevent a *stereo FM receiver* from mistakenly switching to stereo mode due to the presence of 19 kHz components in the received signal. Any signals around 19 kHz are blocked with the help of a simple tuned circuit (L1/C4). R4 ensures that the Q isn't too large. Due to tolerances you may find that the frequency can deviate from 19 kHz (in our prototype the resonance frequency was closer to 20 kHz). In view of thevalue of the inductor, a through-hole version has been used for this (see component list).Without the parallel circuit the crossover point of the correction network is about16.7 kHz. This is more than enough for audio via **VHF FM**. The addition of the parallel circuit causes the amplitude around 10 kHz to increase a little, and the –3 dB point is then reached at 13.5 kHz. In the prototype this cutoff point was about 1 kHz higher due to component tolerances. The board designed for this *circuit* has been kept as small as possible through the use of SMDs for most components. The dimensions of the **FM transmitter** board also played a part here. To make it easier to connect this *circuit* to the **transmitter board**, a connector was included on this board. The supply voltage and audio signals are carried via this connector. The board has been designed in such a way that it can either be mounted behind the **FM transmitter** or alongside it.

e. *What is the quantisation process in PCM? Define quantisation error.* 05

Ans. Quantization Process **:**

Quantization is a process of approximate rounding Off.Sampled signal in PCM transmitted is applied to the quantizer block.

Quantizer converts the sampled signal into an approximate quantized signal which consists of only a finite number of predecided voltage levels. Each sampled value at the input of the quantizer is approximated or rounded off to the nearest standard predecided voltage level.

These standard levels are known as the "quantization levels". Refer to Fig. to understand the process of"quantization.

The quantization process takes place as follows:

The input signal x (t) is assumed to have a peak to peak swing of VL to VHvolts.This entire voltage range has been divided into "Q" equal intervals each of size "s" "s" is called as the step size and its value is given as,

S=(VH-VL)/Q

In Fig., the value of Q = 8

At the center of these ranges, the quantization levels q0, q1,….q7 are placed. Thus the number of quantization levels is Q = 8. These are also called as decision thresholds.

xq(t) represents the quantized version of x (t). We obtain xq (t) at the output of the quantizer.

When x (t) is in the range ▲0, then corresponding loany value of x (t), the quantizer output will be equal to "q0" Similarly for all the values of x (t) in the range ai, the quantizer output is constant equal to "q1".Thus in each range from ▲0 to ▲7, the signal x (t) is rounded off to the nearest quantization level.

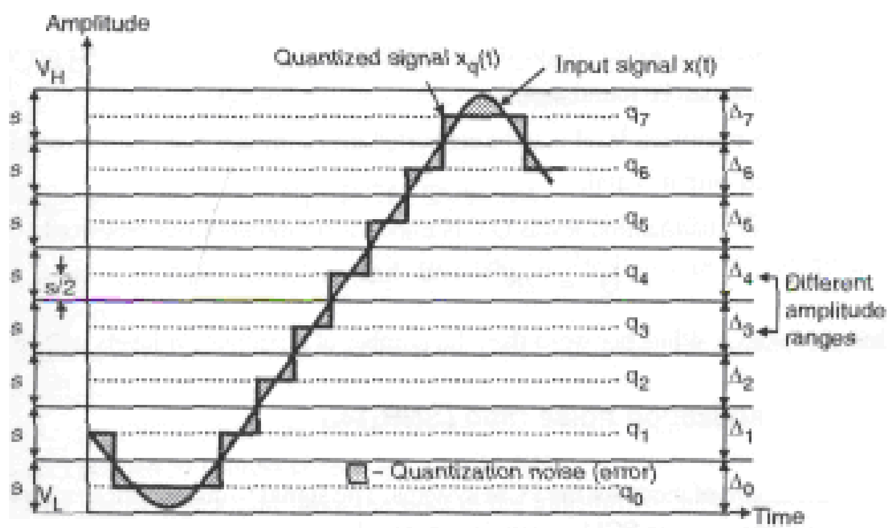


Fig. 10.2.6 : Process of quantization

Quantization error or quantization noise Є :

- The difference between Ihe instantaneous values of the quantized signal and input signal is called as quantization error or quantization noise.

Є = xq(t)-x(t)

- The quanttization error is shown by shaded portions of the waveform in Fig.
- The maximum value of quantization error is ± s / 2 where s is step size.
- Therefore to reduce the quantization error we have to reduce the step size by increasing the number of quantization levels i.e. Q.
- The mean square value of the quantization is given by Mean square value of quantization error = s^2/12 ...

- The relation between the number of quantization levels Q and the number of hits per word (N) in the transmitted signal can be found as follows:
- Because each quantized level is to be converted into a unique N bit digital word, assuming a binary coded output signal,
- The number of quantisation levels Q = Number of combinations of bits/word.

  Q = 2N
- Thus if N = 4 i.e. 4 hits per word then the number of quantization levels will be 2^4 i.e. 16,

Q .2 a. (i) *Define noise factor & noise figure.* 05

Ans. Noise figure:

It is factor of an amplifier or any network is defined in terms ofsignal to noise ratio at input and output of system.

F=(S/N ratio at i/p)/(S/N ratio at o/p).

Noise Figure :

- Sometimes the noise factor is expressed in decibels. When noise factor is expressed in decibels it is called noise figure.

Noise Figure = Fdb = 101og10F

- Substituting the expression for the noise factor we get

Noise figure=10 log10[(S/N at the i/p)/(S/N at the o/p)]

Noise figure Fdb=(S/N)idb-(S/N)db

Q.2.(ii) *if each stage of an amplifier has gain of 10 db and noise figure of 10 db determine the overall noise figure of two stage cascaded amplifier.* 05

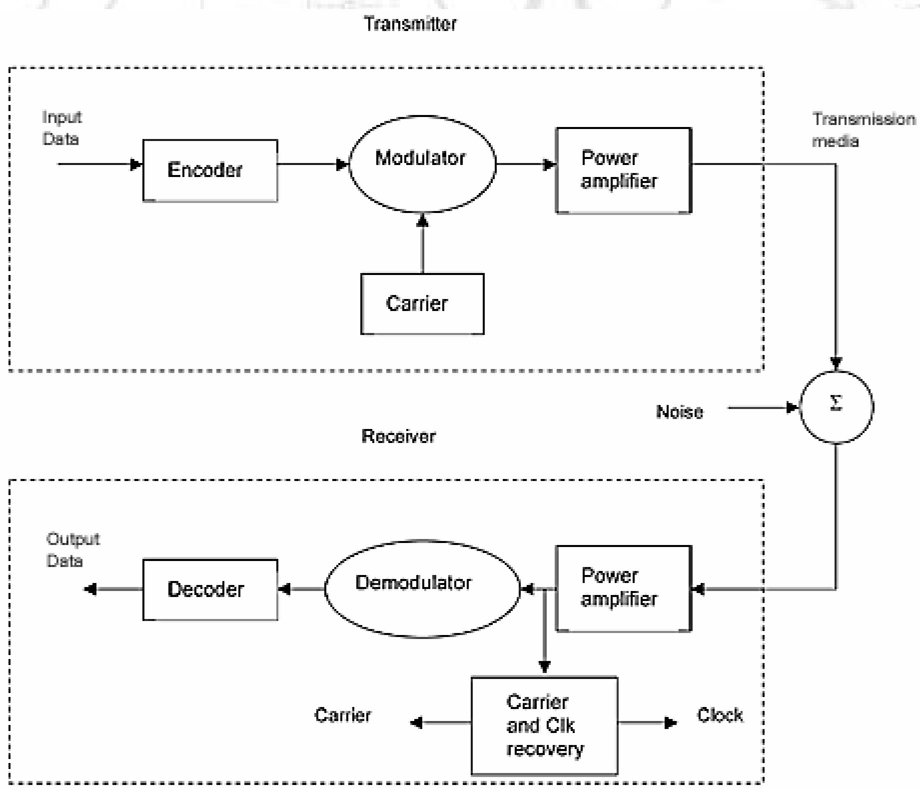
Ans:

G=10 log 10
Fdb=10 log F
Where F is noise factor
F=F1+((F2-1)/G1)
F=10+((10-1)/10)
F=10.9
Overall noise figure ,Fdb=10 log (10.9)
Fdb=10.37 db.

Q No.2b. *Using a block diagram explain a low level Am transmitter.* 10

Ans:

Low level modulation:



Various component of transmitter are described below:-

Master Oscillator:
The master oscillator generates a stable sub harmonic carrier frequency (i.e. the fraction of a Desire carrier frequency). This stable sub-harmonic oscillation is generated by using a crystal oscillator and then frequency is raised to the desired value by harmonic generator.

Buffer Amplifier:
This is a tuned amplifier providing high input impedance at the master oscillator frequency. Any variation in load current does not affect the master oscillator due to this high input impedance of buffer amplifier at the operating frequency of the master oscillator.

Harmonic Generator:
It is an electronic circuit that generates harmonics of its input frequency. The principle of harmonic generation is the same as that of a non-linear modulator. When a signal is applied to a non-linear circuit, it generates harmonics of input frequency. The desired harmonic is selected by a properly tuned circuit. The circuit uses a class C tuned amplifier.

Driver Amplifier or Intermediate Power Amplifier:
One or more stages of a class C tuned amplifier are used to increase the power level of a carrier signal to provide a large drive to the modulated class C amplifier.

Modulation system:
The collector modulation circuit is used for modulation in high power transmitters. The modulating
amplifier is a class A, or class B amplifier amplifying the base-band signal.

Feeder and Antenna:
The transmitter power is fed to a transmitting antenna for effective radiation. The length of the antenna (a conductor) should be of the order of the wavelength for effective radiation.

Q. 3 a. *Prove the following properties of the Fourier Transform-*
(i) Time shifting; 05

Ans.

Time shifting:

The time shifting property states that if x(t) and X(f) form fourier transform pair then,

$$x(t-td) \overset{F}{\leftrightarrow} e^{-j2\pi f td} X(f)$$

Proof:

$F[x(t-td)] = e^{-j2\pi f(t-\lambda)}$ x(t-td $e^{-j2\pi f td}$ dt

Let t-td =τ

T=td+ τ

Dt=d τ

Now we get,

$F[x(t-td)]=\int_{-\infty}^{\infty} x(\tau).\ e^{-j2\pi f(td-T)}\ d\tau$

$= e^{-j2\pi ftd} \int_{-\infty}^{\infty} x(\tau).\ e^{-j2\pi fT}\ d\tau$

$F[x(t-td)]= e^{-j2\pi ftd}\ X(f)$ ………. Proved

The time shifting property states that if x(t) & X(f) form fourier transform pair X(t-td)

(ii) *Convolution in time domain* 05

Ans.

Convolution in time domain:

This property states that the convolution of signals in time domain will transformed in to multiplication of their fourier transforms in frequency domain.

$[x1(t)*x2(t)] \leftrightarrow X1(f)\ X2(f)$

Proof:

So

$$x1(t)*x2(t) = \int_{-\infty}^{\infty} x1(\lambda)*x2(t-\lambda)\ d\lambda$$

taking fourier transform

$$F[x1(t)*x2(t)]= \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} x1(\lambda)*x2(t-\lambda)\ d\lambda]\ e^{-j2\pi ft}\ dt$$

Multiply and divide RHS of equ. By $e^{-j2f\lambda}$ to get,

$$F[x1(t)*x2(t)] = \int_{-\infty}^{\infty} x1(\lambda)\ e^{-j2f\lambda}\ d\lambda \int_{-\infty}^{\infty} x2(t-\lambda)\ e^{-j2\pi f(t-\lambda)}\ dt$$

Let $(t-\lambda)$=m in equ.

$$F[x1(t)*x2(t)]=\int_{-\infty}^{\infty} x1(\lambda)\, e^{-j2f\lambda}\, d\lambda \int_{-\infty}^{\infty} x2(m)\, e^{-j2\pi f(m)}\, dm$$

Using the definition of the fourier transform to RHS we get,

F[x1(t)*x2(t)]= X1(f) X2(f)

This is required result.

Q No.3 b. *Determine fourier transform for a rectangular pulse of amplitude 'A' and time period 'T'(range of t is from -T/2 to +T/2)* 10

Ans.

$$F[x(t)]=X(f)=\int_{-\infty}^{\infty} x(t)\, e^{-j2\pi ft}$$

$$=A/-j2\prod f\ [\, e^{-j2\pi ft}\,]$$

Where $-T/2<t<T/2$

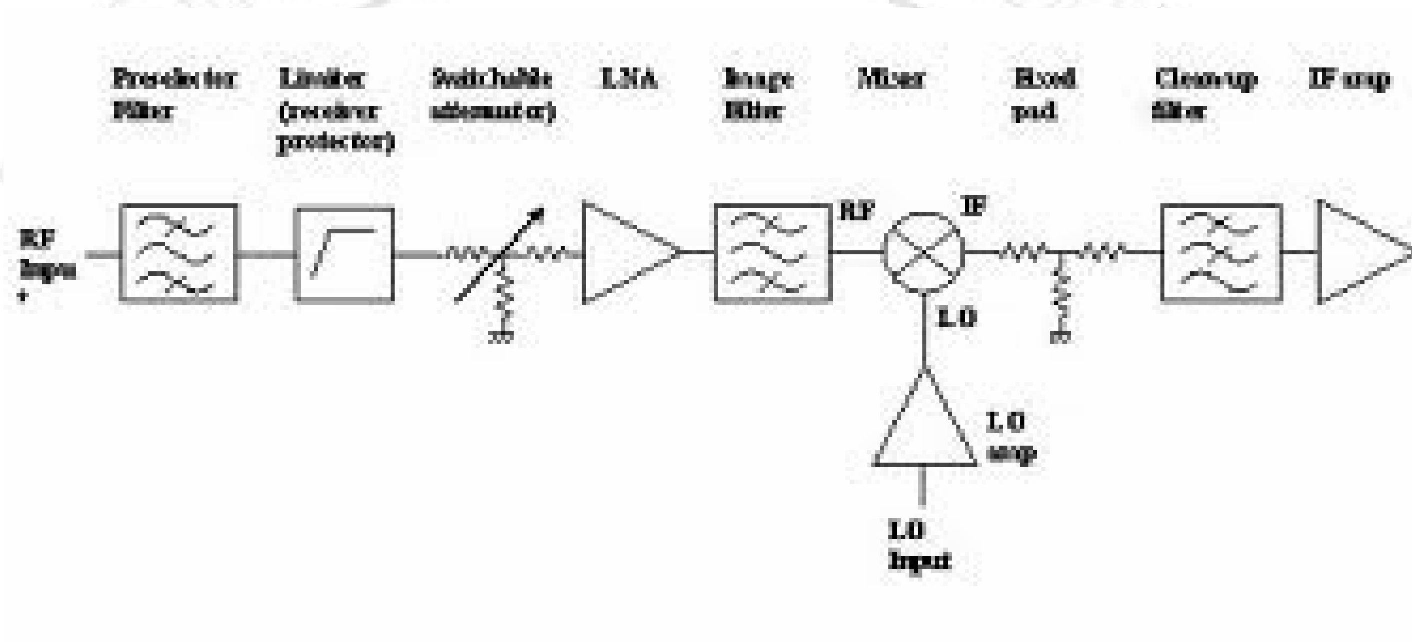
By applying Euler's theorm,

F[x(t)]=ATsinc(fT)

A rect(t/T)=ATsinc(fT)

Q .4. a. *what are the disadvantages of tuned RF receiver Draw the cercuit of superheterodyne receiver and explain the same.* 10

Ans.



Principle of a Superheterodyne Receiver.

The superheterodyne principle:

The principle of operation in the superheterodyne is illustrated by the diagram in Figure . In this system, the incoming signal is mixed with a local oscillator to produce sum and difference frequency components. The lower frequency difference component called the intermediate frequency (IF), is separated from the other components by fixed tuned amplifier stages set to the intermediate frequency. The tuning of the local oscillator is mechanically ganged to the tuning of the signal circuit or radio frequency (RF) stages so that the difference intermediate frequency is always the same fixed value. Detection takes place at intermediate frequency instead of at radio frequency as in the TRF receiver.

A superheterodyne receiver is a Radio Frequency receiver method that multiplies the received signal frequency with a local oscillator frequency to get frequencies that are the sum and difference of the 2 frequencies. For example, if the received signal is 5MHz and the local oscillator frequency is 4MHz, they are multiplied together. 1MHz and 9MHz frequencies would be gotten. Usually the 1MHz is the Intermediate Frequency (IF). It will be admitted (through a band pass filter) later passed through the required electronic circuits for proper processing

Use of the fixed lower IF channel gives the following advantages:

1. For a given Q factor in the tuned circuits, the bandwidth is lower making it easier to achieve the required selectivity.

2. At lower frequencies, circuit losses are often lower allowing higher Q factors to be achieved and hence, even greater selectivity and higher gain in the tuned circuits.

3. It is easier to control, or shape, the bandwidth characteristic at one fixed frequency. Filters can be easily designed with a desired bandpass characteristic and slope characteristic, an impossible task for circuits which tune over a range of frequencies.

4. Since the receiver selectivity and most of the receiver pre-detection gain, are both controlled by the fixed IF stages, the selectivity and gain of the superheterodyne receiver are more consistent over its tuning range than in the TRF receiver.

Q .4 b. *A sinusoidal carrier Vc=100cos (2∏10^5t) is amplitude modulated by sinusoidal voltage voltage Vm=50cos(2∏10^3t).modulation depth is 50%calculate the amplitude and frequency of each sideband. and rms voltage of modulated carrier.* 10

Ans:

Amplitude of each sideband=mEc/2
=(0.5*100)/2
=25v

Lowersideband frequency=fc-fm
=100000-1000
=99000Hz
=99 KHz.

Pt=Pc[1+m^2/2]
Where Pc=(10^4)/(2*R)
E(AM)^2/R=(10^4)/(2*R)[1+(0.5^2/2)]

Rms voltage E(AM)=75 volts

Q .5 a. *What is multiplexing in communication system? Draw the block diagram of TDM-PCM system. Explain each block.* 10

Ans.

PCM-TDM System (Multiplexing the PCM Signals:

- When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required.
- Fig. 11.11.1 shows ihe basic time division multiplexing scheme, called as IheT1, digital system.
- This system is used to convey multiple signals over telephone lines using wideband coaxial cable.

Operation of the T1, system :

The operation of the PCM-TDM system shown in Fig. 11.11.1 is as follows :

- This system has been designed to accommodate 24 voice channels marked S, to Sia. Each signal is bandlimited to 3.3 kHz, and the sampling is done at a standard rate of 8 kHz. This is higher than the Nyquist rate. The sampling is done by the commutator switch SW1.
- These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW1.
- Each sampled signal is then applied to the PCM transmitter which converts il into a digital signal by the process of A to D conversion and companding, as explained earlier.
- The resulting digital waveform is transmitted over a co-axial cable.

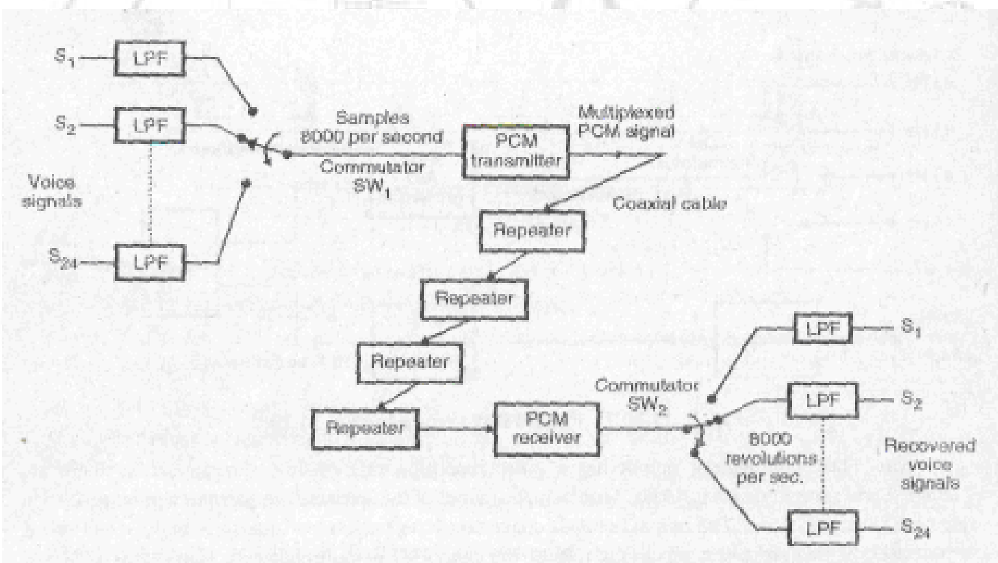
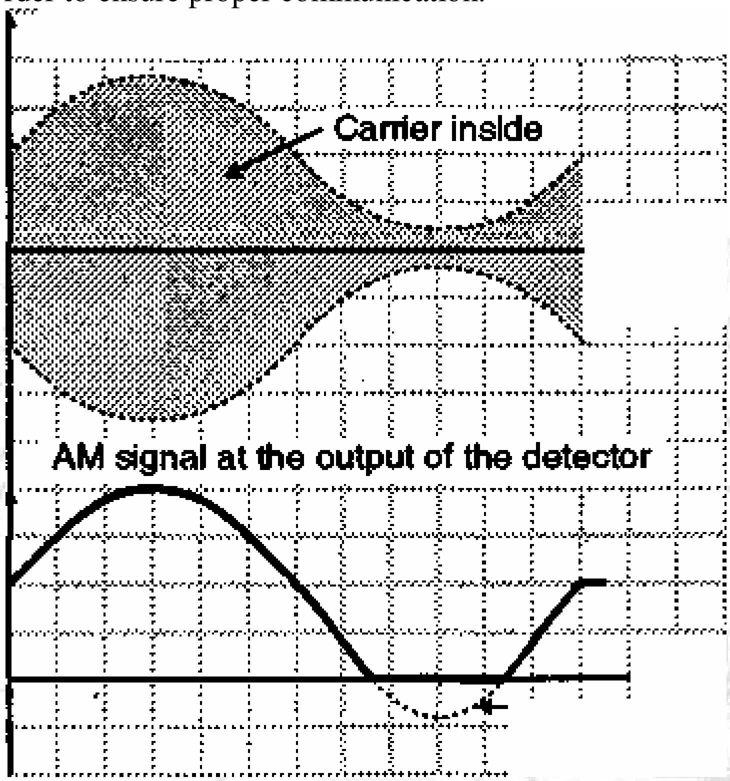


Fig. : Block diagram of a basic I'CM-TDM system

- Periodically, after every 6000 ft, the PCM-TDM signal is regenerated by amplifiers called "Repeaters". They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean PCM-TDM signal at their output. This ensures that the received signal is free from the distortions and noise.
- Al the destination the signal is companded, decoded and demultiplexed, using a PCM receiver. The PCM receiver output is connected to different low pass filters via the commutator switch SW2.
- Synchronization between the transmitter and receiver commutators SW, and SW2 is

essential in order to ensure proper communication.



Q5.b *State and prove Sampling theorem.* 10

Ans:

Sampling Theorem for Low Pass Signals :

- In order to represent the original message signal "faithfully' (without loss of information), it is necessary to take as many samples of the original signal as possible.
- Higher the number of samples, closer is the representation.
- The number of samples depends on the "sampling rate" and the maximum frequency of the signal to be sampled.
- Sampling theorem was introduced to the communication theory in 1949 by Shannon. Therefore this theorem is also called as "Shannon's sampling theorem". The statement of sampling theorem in time domain, for the bandlimited signals of finite energy is as follows :

Statement:

1 If a finite energy signal x (t) contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2W) seconds apart.

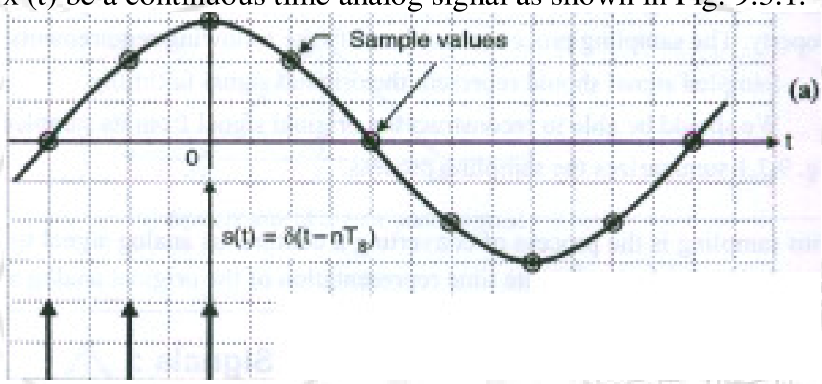
2 If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (I /2W) seconds .apart

Proof of Sampling Theorem :

Let us now prove the sampling theorem in lime domain. The assumptions made for this proof are as follows :

Assumptions:

- Let x (t) be a continuous time analog signal as shown in Fig. 9.3.1.



$S(t) = \partial(t+2Ts)+ \partial(t+Ts)+ \partial(t)+ \partial(t-Ts)+ \partial(t-Ts)+\ldots$

œ

$S(t)= \sum \quad \partial(t-nTs)$

$x\partial(t)= \quad \sum \quad x(nTs) . \partial(t-nTs)$

N= -œ

Fourier Transform:

œ

$X(f) =fo \sum \quad \partial(f-nf0)$

N= -œ

œ

$S(f) =fs \sum \quad \partial(f-nfs)$

N= -œ

$X\partial(t)=x(t)S(t)$

$X\partial(f)=x(f)s(f)$

$$S(f) = x(f) \left[f_s \sum_{N=-\infty}^{\infty} \partial(f-nfs)\right]$$

Q 6.a . *Explain FM detection using PLL.* 10

Ans:

A PLL system can be used to implement an FM demodulator. When a PLL is locked on an FM signal, the VCO tracks the instantaneous frequency of that signal. Since the VCO output tracks the FM signal, and the VCO input voltage is proportional to the VCO output frequency, then the VCO input will be equal to the demodulated signal.

For this example, an FM signal consisting of a 10-kHz carrier frequency was modulated by a 400-Hz audio signal. The schematic diagram shows the connections of the CD4046B as an FM demodulator. The total FM signal amplitude is 500 mV, therefore, the signal must be ac coupled to the signal input (terminal 14). Phase comparator I is used for this application because a PLL system with a center frequency equal to the FM carrier frequency is needed.

The PLL FM demodulator has a number of key advantages:

Linearity: The linearity of the PLL FM demodulator is governed by the voltage to frequency characteristic of the VCO within the PLL. As the frequency deviation of the incoming signal normally only swings over a small portion of the PLL bandwidth, and the characteristic of the VCO can be made relatively linear, the distortion levels from phase locked loop demodulators are normally very low. Distortion levels are typically a tenth of a percent.

Manufacturing costs: The PLL FM demodulator lends itself to integrated circuit technology. Only a few external components are required, and in some instances it may not be necessary to use an inductor as part of the resonant circuit for the VCO. These facts make the PLL FM demodulator particularly attractive for modern applications.

Q .6 b. *What is disadvantage of of Delta modulation .Explain with neat diagram how is it removed in Adaptive Delta modulation.* 10

Ans.

Disadvantage of of Delta modulation:

1)The delta modulation transmits only one bit for one sample. Thus the signaling rate and transmission channel bandwidth is quite small for delta modulation.

2) The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter involved in delta modulation

Adaptive Delta Modulation (ADM):

- In the ADM system, the step size is not constant. Rather when (he slope overload occurs the step size- becomes progressive larger and therefore x' (t) will catch up with x (t) more rapidly.
- Whenever the slope of input signal is large, the slop size of the staircase approximated signal x' (t) is increased.
- On the other hand when the input signal is varying slowly the step size is reduced.
- Thus the step size is adapted as per the level of input signal.

Types of ADM :

- There are various types of ADM systems available depending on (he type of scheme used for adjusting the step size.
- In one type a discrete set of values is provided for the step size whereas in another type a continuous range of step size variation is provided.
- We will discuss the first type here.

Adaptive Delta Modulation :

- In the ADM system, the step size is not constant. Rather when the slope overload occurs the step size becomes progressive larger and therefore x' (t) will catch up with x(t) more rapidly.
- The ADM transmitter is as shown in Fig.

It" you compare this block diagram with that of" the linear delta modulator, then you will find that except for the counter being replaced by the digital processor, the remaining blocks are identity. Let us understand the operation of the digital processor. For that carefully see the waveforms of Fig.

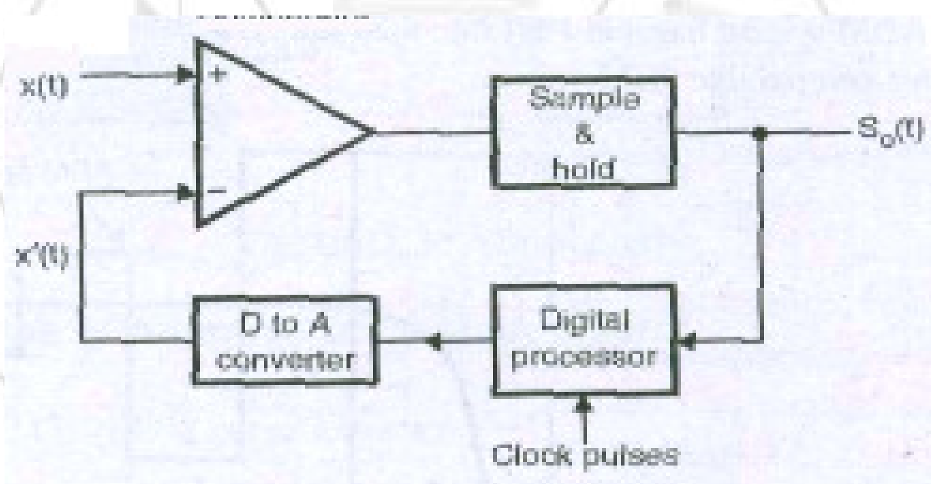


Fig. : ADM transmitter

Operation :

- In response to the kth clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the previous i.e. (k- 1))" clock edge.
- If the direction of both the steps is same, then the processor will increase the magnitude

of the present step by "8". If the directions are opposite then the processor will decrease the magnitude of the present step by "8",

- So(t)in the Fig. 10.11.1, i.e. the output of the ADM system is given as,

So (t) = + 1 if x (t) > x' (t) just before kth clock edge, and
So (t) = - 1 if x (t) < x' (t) just before the kth clock edge.

- Then the step size at the sampling instant k is given by,

δ (k) =[δ (k-1)] So(k)+ δ So(k-1)
let us take an example,
refer fig. below. Assume k=6

k-1=5,

δ(k-1) = δ(5)= δ
So(k)=So(6)=+1
So(k-1)=So(5)=+1
"Substitute in Equation
to get, δ(6)= δ+ δ=2 δ
Look at the Fig. , the step size at the 6th clock edge is 2 δ.
As shown in Fig. 10.11.2, due to variable step size, the slope overload error is reduced. But quantization error is increased. Due to the adjustable step size, the slope overload problem is solved. Hence ADM system has a low bit rale than the PCM system. Therefore the BW required is also less than a comparable PCM system.

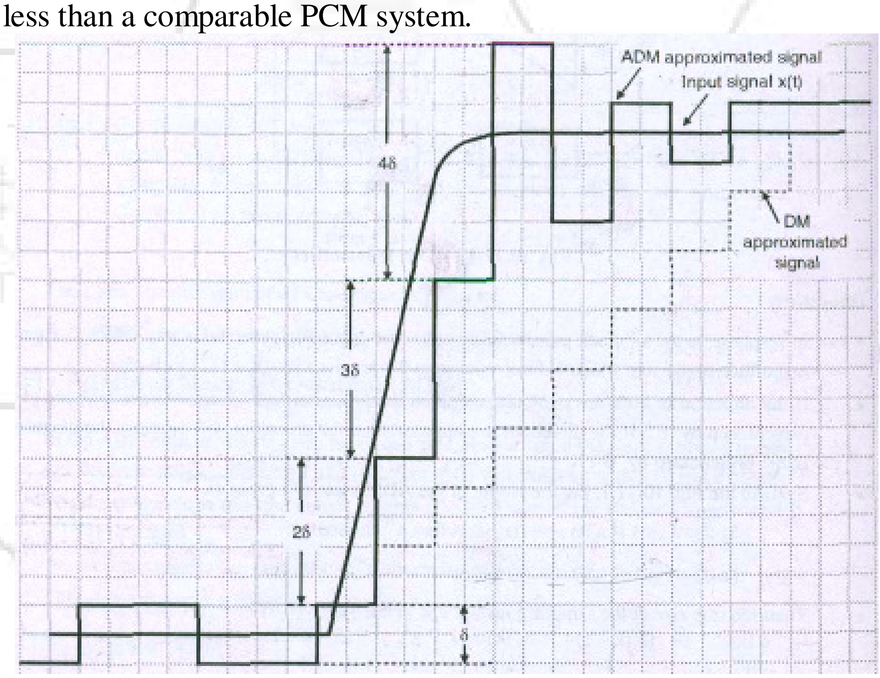


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Fig. : Waveforms of ADM

Q 7. *Write short notes on any four of the followings-*
(i) *satellite communication system:* 05

Ans.

- The use of satellites in communication systems has become very common now-a-days. This is because the satellites can "see" a very large area of the earth.
- Hence the satellite can form a star point of a communications net, to link many users together, simultaneously. This will include users widely separated geographically.
- The construction and launch costs of a satellite are extremely high.
- These costs are "distance insensitive", that means the cost of a short distance satellite link is approximately same as that of a long distance link.
- Therefore a satellite communication system is economical only where the system is used continuously and a large number of users use it.

Basic Communication Syster
Communication using Satellites :

- An artificial satellite orbits revolves around the earth in exactly the same manner as electrons revolve around the nucleus of an aloni,
- The paths in which satellites move arc called as orbits. The orbits are of different types such as synchronous orbits, polar orbits and inclined orbits, out of which (he synchronous or geostationary orbit is used by the geostationary satellites.
- The geostationary satellites take exactly 24 hours to complete one revolution around the earth, therefore they appear to be stationary.
- Depending on the type of application, the satellites are classified into the following categories :

Remote sensing satellites. Scientific satellites.
Weather:
Basic Principle of Satellite Communication :

- A geostationary communication satellite is basically a relay station in space.
- It receives signal from one earth station, amplifies it, improves the signal quality and radiate the signal back to other earth stations.
- Such a relay system allows us to communicate with any corner of the world.

Block Diagram of a Satellite Communication System :

- The block diagram of a satellite communication system is shown in Fig. 1.12.1, An earth station transmits information signal to the satellite using a highly directional dish antenna.
- The satellite receives (his signal, processes il and transmits it back at a reduced frequency,
- The receiving earth stations will receive this signal using parabolic dish antennas pointed towards the satellite.

Satellite

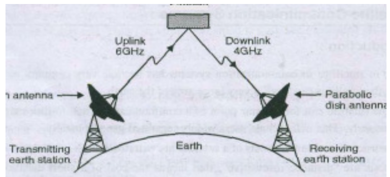

Fig. : Basic operation of satellite communication system

The signal which is being transmitted upwards to the satellite is called as the "uplink" and ii normally at a frequency of 6 GHz.

- The signal which is transmitted back to the receiving earth station is called as the "down link" and it is normally at a frequency of 4 GHz.
- Thus a satellite has to receive, process and transmit the signal. All these functions arc performed by a unit called satellite transponder. A communication satellite generally has two sets of transponders, each set having 12 transponders making it a total of 24 transponders. Each transponder has a bandwidth of 36 MHz which is sufficient to handle at least one TV channel.
- The uplink signal received by a transponder is weak and downlink signal transmitted by the transponder is strong. Therefore to avoid interference between them, the uplink and downlink frequencies are selected to be of different values.
- The operation of satellite takes place a(a very high signal frequencies in the microwave range. The typical band of signal frequencies used for (he communication satellites are as follows ;

1. Cband : 4/6 GHz
2. Kuband : 11/14 GHz
3. Ka band : 20/30 GHz

- The C band frequencies of 4/6 GHz indicate that the downlink frequency is 4 GHz while the uplink frequency is 6 GHz. One of (he advantages of operating at such a high frequency is reduction in the size of antennas and other components of (he system.
- K is extremely important to maintain the position of the satellite with respect to earth. Therefore control routines such as station keeping and altitude control are executed from the control room in the earth stations.
- Multiple access methods such as FDMA (frequency division multiple access), TDMA (time division multiple access) and CDMA (code division multiple access) are used to allow the access of a satellite to the maximum number of earth stations.
- The power requirement of a satellite is satisfied by solar panels and a set of nickel cadmium batteries, carried by the satellite itself.

ii. *satellite communication system:* 05

Ans.

- The use of satellites in communication systems has become very common now-a-days. This is because the satellites can "see" a very large area of the earth.
- Hence the satellite can form a star point of a communications net, to link many users together, simultaneously. This will include users widely separated geographically.

- The construction and launch costs of a satellite are extremely high.
- These costs are "distance insensitive", that means the cost of a short distance satellite link is approximately same as that of a long distance link.
- Therefore a satellite communication system is economical only where the system is used continuously and a large number of users use it.

Basic Communication Syster

Communication using Satellites :

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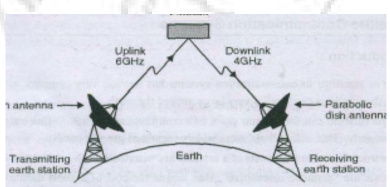

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iii . *FRISS formula:* 05

Noise Factor of Amplifiers in Cascade (Friiss Formula) :

In practice the filters or amplifiers are not used in isolated manner. They are used in the Cascade manner. • So in this section let us see the effect of cascading on the noise factor and noise temperature.The overall noise factor of such cascade connection can be determined as follows ;

Fig. shows two amplifiers connected in cascade.

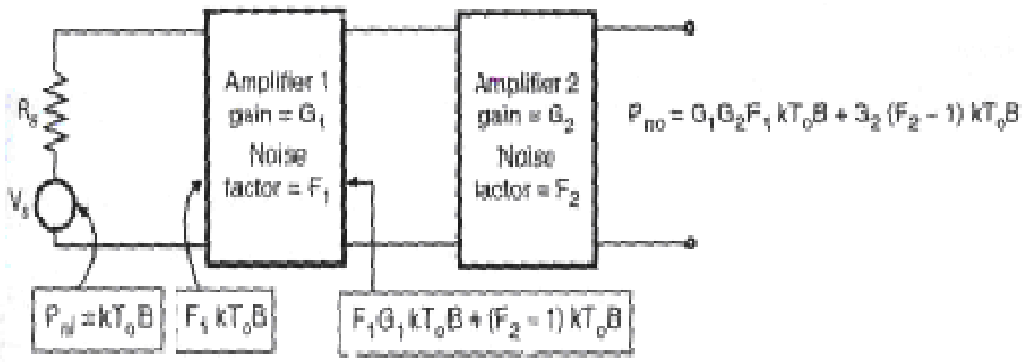

Amplifier 1
gain = G_1
Noise
factor = F_1
Amplifier 2
gain = G_2
Noise
factor = F_2
R_s
V_s
$P_{no} = G_1G_2F_1\,kT_0B + G_2(F_2 - 1)\,kT_0B$
$P_{ni} = kT_0B$
$F_1\,kT_0B$
$F_1G_1\,kT_0B + (F_2 - 1)\,kT_0B$

Fig. : Two amplifiers connected in cascade

Let the power gains of the two amplifiers be G1 and G2 respectively and let their noise factors be F1 and F2 respectively.

The total noise power at the input of the first amplifier is given as,

Pni(total) = F1kT0B

This is written by taking the reference of Equation .

The total noise power at the output of amplifier 1 will be the addition of two terms.

Noise input to amplifier 2 = G1 F1 kT0 B + (F2 - l)kT0 B .

The first term represents the amplified noise power (by G,) and the second term represents the noise contributed by the second amplifier.

The noise power at the output of the second amplifier is G2 times the input noise power to amplifiers2

Pno=G2*(noise input to amplifier 2)

Pno=G1G2F1kT0B+G2(F2-1)kT0B

So

G=G1G2

Overall noise factor F

F=Pno/G1G2Pni

And

Pni=KT0B

So

F=(G1G2F1kToB+G2(f2-1)kToB)/(G1G2kToB)

=F1+((f2-1)/G1)

So overall noise factor

F=F1+((F2-1)/G1)+((F3-1)/G1G2)+((F4-1)/G1G2G3)+…………

Is called friiss formula.

iv. *AGC Principle in receiver:* 05

Ans:

Gain control is necessary to adjust the receiver sensitivity for the best reception of signals of widely varying amplitudes. A complex form of automatic gain control (agc) or instantaneous automatic gain control (iagc) is used during normal operation. The simplest type of agc adjusts the IF amplifier bias (and gain) according to the average level of the received signal. With agc, gain is controlled by the largest received signals. When several radar signals are being received simultaneously, the weakest signal may be of greatest interest. Iagc is used more frequently because it adjusts receiver gain for each signal.

The agc circuit is essentially a wide-band, dc amplifier. It instantaneously controls the gain of the IF amplifier as the radar return signal changes in amplitude. The effect of iagc is to allow full amplification of weak signals and to decrease the amplification of strong signals. The range of iagc is limited, however, by the number of IF stages in which gain is controlled. When one IF stage is controlled, the range of iagc is limited to approximately 20 dB. When more than one IF stage is controlled, iagc range can be increased to approximately 40 dB.

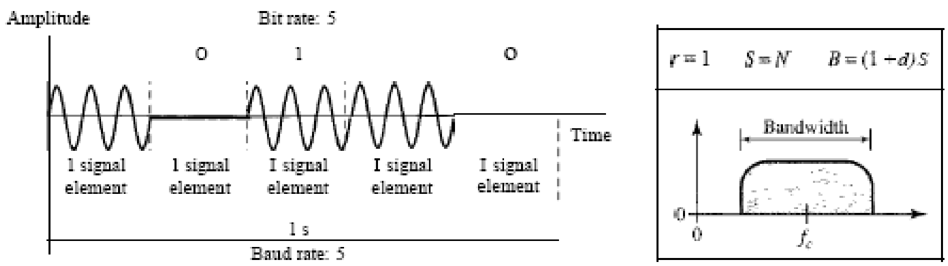
V. *Amplitude shift keying:* 05

Amplitude Shift Keying

In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes.

Binary ASK (BASK)

Although we can have several levels (kinds) of signal elements, each with a different amplitude, ASK is normally implemented using only two levels. This is referred to as binary amplitude shift keying or *on-off keying* (OOK). The peak amplitude of one signallevel is 0; the other is the same as the amplitude of the carrier frequency. Figure 5.3 gives a conceptual view of binary ASK.



Bandwidth for ASK Figure 5.3 also shows the bandwidth for ASK. Although the carrier signal is only one simple sine wave, the process of modulation produces a nonperiodic composite signal. This signal, as was discussed in Chapter 3, has a continuous set of frequencies. As we expect, the bandwidth is proportional to the signal rate (baud rate). However, there is normally another factor involved, called *d,* which depends on the modulation and filtering process. The value of *d* is between 0 and 1. This means that the bandwidth can be expressed as shown, where 5 is the signal rate and the *B* is the bandwidth.

$B = (1 + d) \times S$

The formula shows that the required bandwidth has a minimum value of 5 and a maximum value of 25. The most important point here is the location of the bandwidth. The middle of the bandwidth is where *Ie* the carrier frequency, is located. This means if we have a bandpass channel available, we can choose our *Ie* so that the modulated signal occupies that bandwidth. This is in fact the most important advantage of digitalto-*analog conversion. We can shift the resulting bandwidth to match what is available.*