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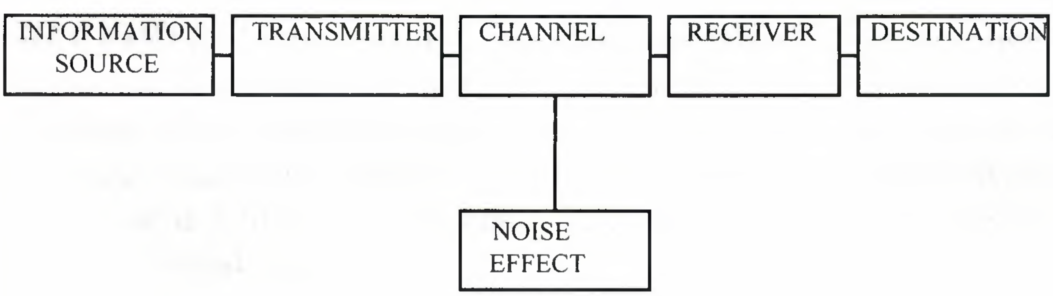
HISTORY OF COMMUNICATION
INTRODUCTION

Every day, we come in contact with communication systems. The most common being the telephone, radio, and television. We can communicate with people through these media. One of the earliest invention to communication was the invention of the electric battery, being demonstrated in 1837. Nearly 40 years later, in 1875, Emile baudot developed a code for telegraphy in which each latter was encoded into a fixed length binary code words of length 5. Telephony came into being the invention of the telephone in the 1870's Alexander Graham Bell patented his invention of the telephone in 1876's and in 1877 established Bell Telephone Company. Automatic switching was another important advance in the development of telephony.

The phenomenal growth in communication services over 50 years by the invention of electrical transistors in 1947 by Walter Brattain, John Bardeen, and William Shockley, the integrated circuit in 1958 by Jack Killby and Robert Noyce, and the laser by towns and Schawlow in 1958, have possible development of small size, low power low weight and high speed electronic circuits which are used in the construction of satellite communication systems, wideband microwave radio system, and lightwave communication systems using fiber optic cables.

MODEL OF COMMUNICATION SYSTEM



INFORMATION: The messages originated may be human voice, a television picture , a teletype message or data. If the desired transmit data is not in electrical form than it is changed into electrical waveform by an input transducer that is a device which change physical message into electrical message.

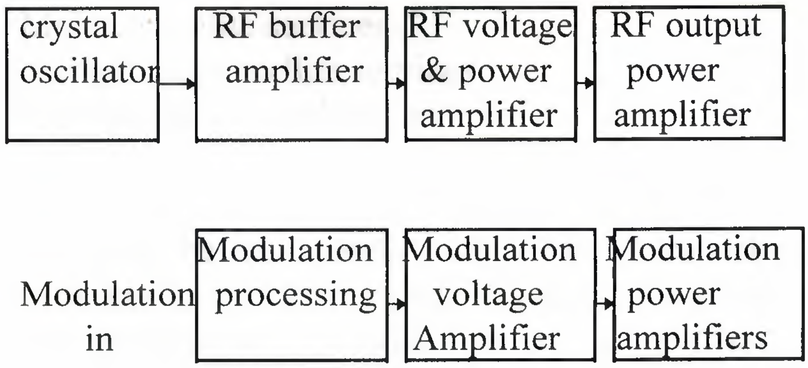
TRANSMITTER: Transmitter modifies baseband signal for efficient transmission. Consists of subsystems, a preemphasizer, sampler, a quantizer a coder and modulator. A transmitter modulates information into the message carrier waves that is superimposed on high frequency sine wave. Modulation may be high level or low level. And the system may be A.M, F.M, P.M or combination of these modulation techniques depending on the requirement.

CHANNEL: In communication we can receive the electrical signal from a frequency band. May be a wire, coaxial cable, a wave guide, an optical fibber or a radio link.

NOISE: Unwanted signal is called noise. A noise signal not only destroy the data signals also contaminates the whole path of transmission. Basically there are two types of noise signals, an external noise signal and internal noise signal. The external noise includes interface from *signal transmitted on nearby channels, men made noise is due to the* faulty connections of switches for electrical equipment, due to auto mobile ignition radiation, due to overhead power transmission lines passing over the communication channels, natural noise from lightning, electrical storms. The other source of noise is due to thermal motion of electrons, conductor random emission and diffusion or recombination of charge carriers in electronic devices.

RECEIVER: It converts the transmitted signals wave into desired form of output. The range of frequencies over which a receiver operates with a selected performance known as sensitivity. The receiver output is fed to the output transducer which converts the electrical signal into its original form that is a message. The basic function of receiver is to demodulate the transmitted signal.

RADIO TRANSMITTER



OWERVIEW OF SATELLITE SYSTEMS

The use of satellites in communications system is very much a fact of everyday life, as evidenced by the many homes which are equipped with antennas, used for reception of satellite television. What may not be so well known is that satellites form an essential part of telecommunications systems, worldwide, carrying large amounts of data and telephone traffic in addition to television signals.

Satellites offer a number of features not readily available with other means of communications. Because very large areas of the earth are visible from a satellite , the satellite can form the star point of communications net linking together many users simultaneously, users who may be widely separated geographically. The same feature enables satellites to provide communications link to remote communities in sparsely populated areas which are difficult to access by other means.

Frequency Allocations For Satellite Systems:

Allocating frequencies to satellite services is a complicated process which requires international coordination and planning. This is carried out under the auspices of International Telecommunication. To facilitate frequency planning, the world is divided into three regions:

region 1: Europe, Africa, what was formerly the Soviet Union, and Mongolia
region 2: north and south America and Greenland
region 3:Asia, Australia, and the Southwest pacific

within these regions, frequency bands are allocated to various satellite services, although a given service may be allocated different frequency bands in different regions. Some of the services provided by satellite services are:

Fixed satellite service (FSS)
Broadcasting satellite service (BSS)
Mobile satellite services
Navigational satellite services
Meteorological satellite services

There are many subdivisions within these broad classifications, for example, the fixed satellite service provides links for existing telephone networks as well as for transmitting television signals to cable companies for distribution over cable systems. Broadcasting satellite services are intended mainly for direct broadcast to the home, some times referred to as the direct broadcast satellite (DBS). Mobile satellite services would include land mobile, maritime mobile, and aeronautical mobile. Navigational satellite services include global positioning systems, and satellites intended for the meteorological services often provide a search and rescue service.

Table lists the frequency band designations in common use for satellite services. The Ku band signifies the band under the K band, and the Ka is the band above the K band. The Ku band is the one used at present for direct broadcast satellites, and it is also used for certain fixes satellite services. The C band is used for fixed satellite services , and no direct broadcast services are allowed in this band. The VHF band is used for certain mobile and navigational services.

The L band is used for satellite services and navigation systems. For the fixed satellite service in the C band, the most widely used subrange is 4 to 6 GHz.

Table: Frequency band designations:

Frequency range, GHz	Band designation
0.1-0.3	VHF
0.3-1.0	UHF
1.0-2.0	L
2.0-4.0	S
4.0-8.0	C
8.0-12.0	X
12.0-18.0	Ku
18.0-24.0	K
24.0-40.0	Ka
40.0-100.0	mm

ORBITING METHODS

Satellites which orbit the earth follow the same laws that govern the motion of the planets around the sun. From early times much has been learned about planetary motion through careful observations johannes Kepler was able to derive three laws describing planetary motion. Later, in 1665, Sir Isaac Newton was able to derive Kepler's laws from his own law of mechanics and develop the theory of gravitation.

Kepler' First Law:

Kepler's first law states the path followed by the satellite around the primary will be an ellipse. An ellipse has two focal points shown as F_1 and F_2 shown in figure. The center of mass of the two body system termed the barycenter, is always centered on one of the foci. In our specific case, because of the enormous difference between the masses of the earth, which is therefore always at one of the foci

The semimajor axis of the ellipse is denoted by a, and the semiminor axis by b. The eccentricity e is given by

$$e = \left(\frac{a^2-b^2}{b^2}\right)^{1/2}$$

The eccentricity and the semimajor axis are two of theorbital parameters specified for satellites orbiting the earth. For an elliptical orbit 0<e<1. When e=0, the orbit becomes circular.

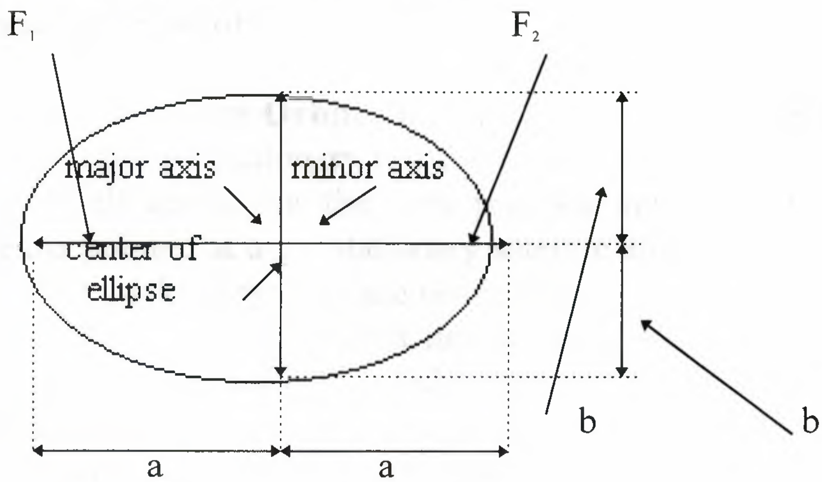


Figure: Showing the foci ,the semimajor axis a, and the semiminor axis b, of an ellipse

Kepler's Second Law: Kepler's second law states that for equal time intervals, the satellite will sweep out equal areas in its orbital plane, focused at the barycenter. The satellite travels distances in 1 s, then the areas will be equal. The average velocity in each case is $S_1$and S_2 meters per second, and because of the equal area law. An important consequence of this is that the satellite takes longer to travel a given distance when it is farther away from earth.

Kepler's Third law: States that the square of the periodic time of orbit is proportional to the cube of the mean distance between the two bodies.

The mean distance is equal to the semimajor axis a. For the artificial satellites orbiting the earth, Kepler's third law can be written in the form

$$a^3=\mu/n^2$$

where n is the mean motion of the satellite in radians per second and μ is the earth's gravitational constant. With a in meters its value is
$\mu=3.986005*10^{14}\ m^3/s^2$

With n in radians per second the orbital period in seconds is given by

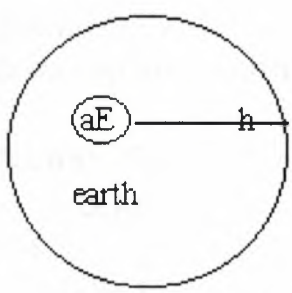
$$P=2\pi/n$$

the importance of the Kepler's third law is that it shows there is a fixed relationship between the period and size. One very important orbit in particular, known as the geostationary orbit, is determined by the rotational period of the earth.

The Geostationary Orbit: The geostationary orbit is the orbit in which a satellite appears stationary relative to the earth. This is the most widely used of all orbits, for the very practical reason that an earth station antenna pointed at a geostationary satellite automatically follows it, and elaborate tracking systems are not required.

The geostationary orbit lies in the equatorial plane, meaning that the inclination is zero. This follows, since any finite inclination implies that there are ascending and descending nodes, meaning that satellite would have to cross lines of latitude and by definition would not be geostationary. Another requirement is that the satellite must orbit the earth in the same direction as the earth spins, and at the same speed.

Kepler's third law may be used to find the required altitude of the geostationary orbit above the equator. The geostationary orbit lies in the earth's equatorial plane and is circular.



Geostationary orbit

Figure: The geostationary orbit is circular and lies in the earth's equatorial plane at an altitude h=35,786 km above the equator.

The geostationary orbit, being in the equatorial plane, has zero inclination. Although it is possible to have a geosynchronous orbit, that is, one which has the same orbital period as the earth's spin period, at some inclination , this will not be Geostationary.

Antenna Look Angles: The user must be able to determine the azimuth and elevation angles of the ground station antenna.
For large commercially operated ground stations, the look angle settings will be controlled by computer.

Figure shows the geometry involved in determining the look angles. Here a_E is the earth's equatorial radius, and R is the radius at the earth station, λ_E is the latitude of the earth station:

$$l=(a_E/(1-e_E^2\sin^2\lambda_E)^{1/2}+H)\cos\lambda_E$$

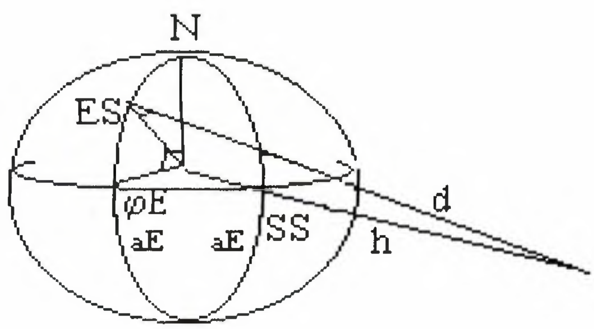


Figure: The geometry used in determining the look angles for a Geostationary satellite.

The earth station is denoted by ES. Point S denotes the satellite in Geostationary orbit, and point SS the subsatellite point, and distance d is the distance between earth station and satellite referred to as the range.

The Polar Mount Antenna: Where the home antenna has to be steerable, expense usually precludes the use of separate azimuth and elevation actuators. Instead, a single actuator is used which moves the antenna in a circular arc. This is known as a polar mount antenna. The antenna pointing can only be accurate for one satellite, and some pointing error must be accepted for satellites on either side of this.

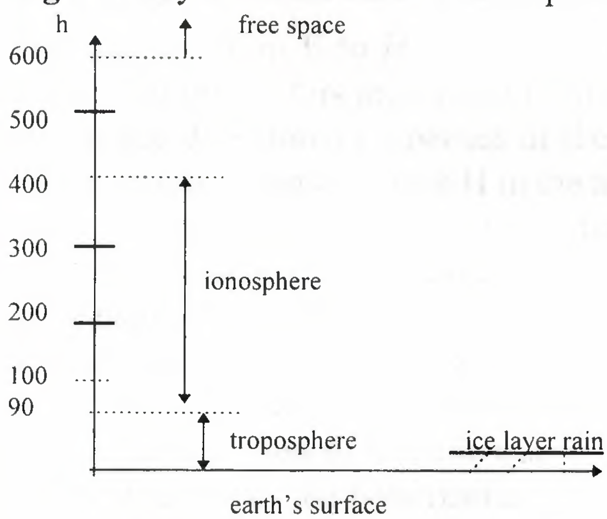
Earth Eclipse Of The Satellite: If the earth's equatorial plane coincided with the plane of the earth's orbit around the sun, geostationary satellites would be eclipsed by the earth once each day.

RADIO WAVE PROPOGATION

A signal traveling between an earth station and a satellite must pass through the earth's atmosphere including the ionosphere as will be shown in figure.

Atmospheric Losses: Losses occur in the earth's atmosphere as a result of energy absorption by the atmospheric gases. These losses are treated separately from those which result from adverse weather conditions, which are atmospheric losses. To distinguish between these, the weather related losses are referred to as atmospheric attenuation and the absorption losses simply as atmospheric absorption.

Figure: Layers in the earth's atmosphere



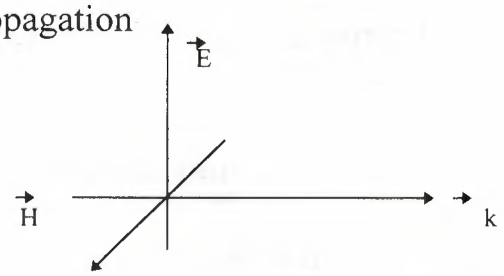
Ionospheric Effects: Radio waves traveling between satellites and earth stations must pass through the ionosphere. The ionosphere is the upper region of the earth's atmosphere, which has been ionized, mainly by solar radiation. The free electrons in the ionosphere are not uniformly distributed but form in layers. Furthermore, clouds of electrons may travel through the ionosphere and give rise to fluctuations in the signal that can be determined on a statistical basis.

Rain Attenuation: Rain attenuation is a function of rain rate. By rain rate is meant the rate at which rainwater would accumulate in a rain gauge situated at the ground. In calculations relating to radio wave attenuation , the rain rate is measured in millimeters per hour.

POLARIZATION

In the far field zone of a transmitting antenna, the radiated wave takes on the characteristics of a transverse electromagnetic wave. By the far field zone is meant at distances greater than $2D^2/\lambda$ from the antenna and λ is the wavelength. For a parabolic antenna of 3 m diameter transmitting a 6 GHz wave, the far field zone begins at approximately 360 m. The Tem designation is shown in figure, where it can be seen that both the magnetic field H and the electric field E are transverse to the direction of propagation, denoted by the propagation vector k

E, H, and k represent vector quantities and it is important to note their relative directions. When one looks along the direction of propagation



The rotation from E to H is in the direction of the right-hand-threaded screw, and the vectors are said to form a right hand set. The wave always retains the directional properties of the right-hand set appears is to note that the letter E comes before H in the alphabet.

At great distances from the transmitting antenna, such as are normally encountered in radio systems, the TEM wave can be considered to be plane. This means that the E and H vectors lie in the plane which is at right angles to the vector k. The vector k is said to be normal to the plane. The magnitudes are related by $E=HZ_0$, where $Z_0=120\pi$ ohms.

The direction of the line traced out by the tip of the electric field vector determines the polarization of the wave. As we know electric and magnetic fields are varying as functions of time. The magnetic fields varies exactly in phase with the electric, and its amplitude is proportional to the electric field amplitude.

In the early days of radio, there was little chance of ambiguity in specifying the direction of polarization in relation to the surface of the earth. Vertical polarization meant that the electric field was perpendicular to the earth's surface. Although the terms vertical and horizontal are used with the satellite transmissions. A linear polarized wave transmitted by a Geostationary satellite may be designated vertical if its electric field is parallel to the earth's polar axis.

A vertically polarized electric field is

$$E_v = a_v E_v \sin\omega t$$

A horizontally polarized wave is

$$E_h = a_h E_h \sin\omega t$$

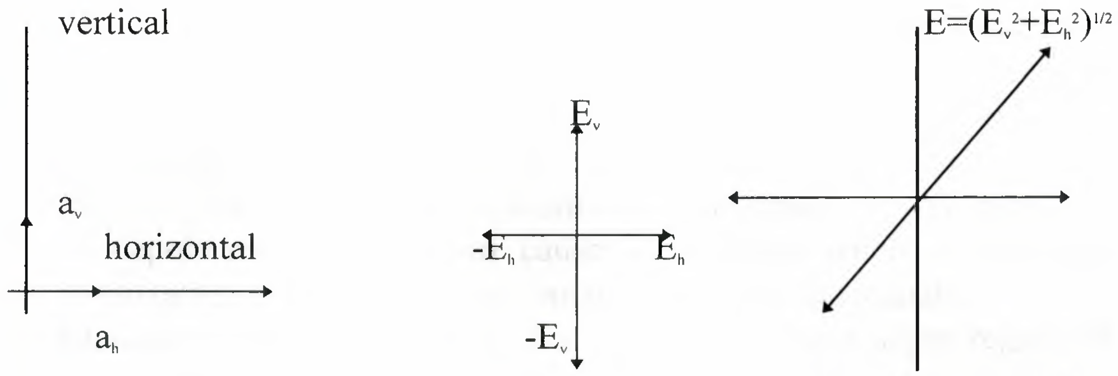


Figure: Horizontal and vertical components of linear polarization

Antenna Polarization: An antenna which transmits a wave with a given sense of polarization is said to be polarized.

A receiving antenna which transmits a wave to deliver maximum power is also polarized. The reciprocity theorem for antennas ensures that an antenna which is designed to transmit a given sense of polarization will also be matched to that polarization for reception. For example, in a terrestrial system, a horizontal half-wave dipole antenna will produce horizontally polarized waves when transmitting , and the same antenna will receive maximum power from a horizontally polarized wave. A combination of two half-wave dipoles, one horizontal and one vertical and fed with signals 90 degrees apart in phase, will produce a circularly polarized wave. The combined output from these two dipoles when receiving will be maximum when the receiving signal is circularly polarized.

The use of orthogonal polarization allows signals to overlap in frequency without interference. Unfortunately certain depolarization effects can occur in the transmission signals.

Polarization of satellite signals: The definition of horizontal polarization is where the electric field vector is parallel to the equatorial plane, and vertical polarization is where the electric field vector is parallel to the earth's polar axis. Subsatellite point on the equator, both polarization will result in electric fields that are parallel to the local horizontal plane.

Depolarization:
The propagation path between the satellite and earth station passes through the ionosphere, and possibly through layers of ice crystals in the upper atmosphere and rain, all of which are capable of altering the polarization of the wave being transmitted. An orthogonal component may be generated from the transmitted polarization, an effect referred to as depolarization. This can cause interference where the orthogonal polarization is used to provide isolation between the signals.

Ionospheric Depolarization: The ionosphere is the upper region of the earth's atmosphere that has been ionized, mainly by solar radiation. The free electrons in the ionosphere are not uniformly distributed but form layers. Furthermore, clouds of electrons may travel through the ionosphere and give rise to fluctuations in the signal. One of the effects of the ionosphere is to produce a rotation of the polarization of the signal, an effect known as Faraday rotation.

Rain Depolarization: The ideal shape of a raindrop is spherical, as this minimizes the energy required to hold the raindrop together. The shape of small raindrops is close to the spherical but the larger drops are better modeled as oblate spheroids with some flattening underneath.

Ice Depolarization: The experimental evidence suggest that the chief mechanism producing depolarization in ice is differential phase shift, with little differential attenuation present. This is because ice is good dielectric, unlike water which has considerable losses. Ice crystals tend to be needle-shaped or platelike, and if randomly oriented have little effect, but depolarization occurs when they become aligned.

THE SPACE SEGMENT

A satellite commucations system can be broadly divided into two segments, a ground segment and a space segment. The space segment include the satellites, but it also includes the ground facilities needed to keep the satellites operational, these being referred to as the tracking, telemetry, and command facilities.

The equipment carried aboard the satellite can also be classified according to the function. The payload refers to the equipment used to provide the service for which the satellite has been launched. The bus refers not only to the vehicle which carries the payload, but also to the various subsystems which provide the power, attitude control, orbital control, thermal control, and command and telemetry functions required to service the payload.

The Power Supply: The primary electrical power for operating the electronic equipment is obtained from solar cells. Individual cells can generate only small amounts of power, and therefore arrays of cells in series-parallel connection are required.

Attitude Control: The attitude of a satellite refers to its orientation in space. Much of the equipment carried aboard the satellite is therefore for the purpose of controlling its attitude. Attitude control is necessary, for example to ensure that directional antennas point in the proper directions. In the case of earth environmental satellites, the earth-sensing instruments must cover the required regions of the earth, which also requires attitude control. A number of forces, referred to as the disturbance torques, can alter the attitude, some examples being the gravitational fields of the earth and the moon, solar radiation, and meteorite impacts. Attitude control must not be confused with station keeping, which is the term used for maintaining a satellite in its correct orbital position.

a)Spin Stabilization: Spin stabilization is used with cylindrical satellites. The satellite is constructed so that it is mechanically balanced about one particular axis and is then set spinning around this axis. For Geostationary satellites the spin axis is adjusted to be parallel to the N-S axis of the earth. Spin rate is typically in the range of 50 to 100 rev/min.

b)Three Axis Stabilization: In three axis stabilization, as the name suggests, there are stabilizing elements for each of the three axis, roll, pitch, and yaw.

Station keeping: In addition to having its attitude controlled ,it is important that a Geostationary satellite be kept in its correct orbital slot. The equatorial ellipticity of the earth causes Geostationary satellites to drift slowly along the orbit.

Thermal Control: Satellites are subject to large thermal gradients, receiving the sun's radiation on one side while the other side faces into space. In addition, thermal radiation from the earth and the earth's albedo, which is the fraction of the radiation falling on earth which is reflected, can be significant for low-altitude earth-orbiting satellites, although its negligible for Geostationary satellites. Equipment in the satellite also generates heat which has to be removed. The main important consideration is that the satellite's equipment should operate as nearly as possible in a stable temperature environment.

TT&C Subsystem: The telemetry, tracking, and command subsystem performs several routine functions the spacecraft. The telemetry, or telemetering, function could be interpreted as measurement at a distance. Specifically it refers to the overall operation of generating an electrical signal proportional to the quantity being measured, and encoding and transmitting this to a distant station, which for the satellite is one of the earth stations.

Transponders: A transponder is the series of interconnected units which forms a single communications channel between the receive and transmit antennas in a communications satellite. Some of the units utilized by a transponder in a given channel may be common to a number of transponders.

The Antenna Subsystem: The antennas carried aboard a satellite provide the dual functions of receiving the uplink and transmitting the downing signals.

THE EARTH SEGMENT

The earth segment of a satellite communications system consist of the transmit and receive earth stations. The simplest of these are the home TV receive-only systems. And the most complex are terminal stations used for international communications networks. Also included in the earth segment are those stations which are on ships at sea, and commercial and military land and aeronautical mobile stations.

Receive-Only Home TV Systems: Planned broadcasting directly to home TV receivers is scheduled to take place in the Ku(12 GHz) band. This service is known as direct broadcast satellite(DBS) service. There is some variation in the frequency band assigned to different geographical regions.

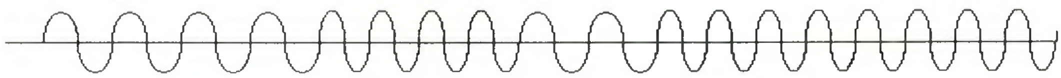
Master Antenna TV System: A master antenna system is used to provide the reception of DBS TV channels to a small group of users, for example to the tenants in an apartment building. It is basically similar to the home system, with each user having access to all of the channels independently of the other users.

Community Antenna TV System: The community antenna TV system employs a single outdoor unit, with separate feeds available for each sense of polarization, like the master antenna television system, so that all channels are made available simultaneously.

Transmit-Receive Earth Stations: A transmit -only station is only required, in relaying TV signals to the remote TV receive-only stations. Transmit-receive stations provide both functions, and are required for telecommunications traffic generally including network TV. There are considerable differences in detail between transmit-receive earth stations, depending on the types of service being provided.

ANALOG AND DIGITAL COMMUNICATION SYSTEM

Analog Communication System: In this system an analog signal is used for the transmission of message. This message is called analog message. An analog message is a message whose value varies with continuous time fashion. For example, the temperature or the atmospheric pressure of a certain location can vary over a continuous range and can assume infinite possible speech signal has the amplitude that vary over the continuous range. An analog signal is shown in fig:

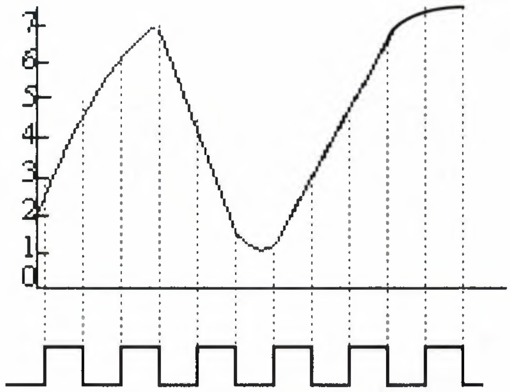


Digital communication system: By digital technology we can communicate through the discontinuous signals instead of continuous signals. The value of digital techniques derives from the ability to construct unique codes to represent different items of information. These are binary number system and other types of digital electronic equipment which have revolutionized modern society. In digital communication system , the message produced by the source are converted into sequence of binary digits.

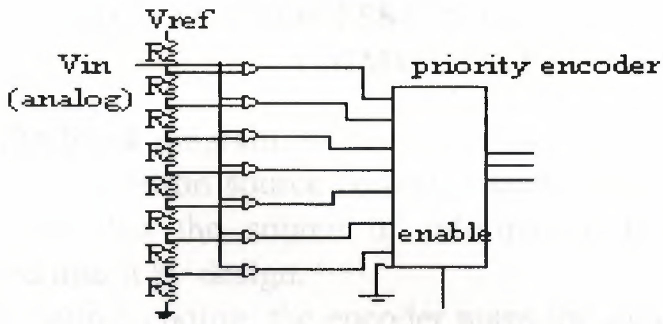
Digital Signals: Digital signals are coded representations of information. Keyboard characters, for example, are usually encoded in binary digital code. A binary code has two symbols, usually denoted as 0 and 1, and these are combined to form binary words to represent the characters.

Analog signals such as speech and video may be converted to a digital form through an analog to digital converter. A particular form of A/D conversion is employed, known as pulse code modulation.

In digital terminology, a binary symbol is known as a binit from the binary digit. The information carried by a binit is, in most practical situations, equal to a unit of information known as a bit.



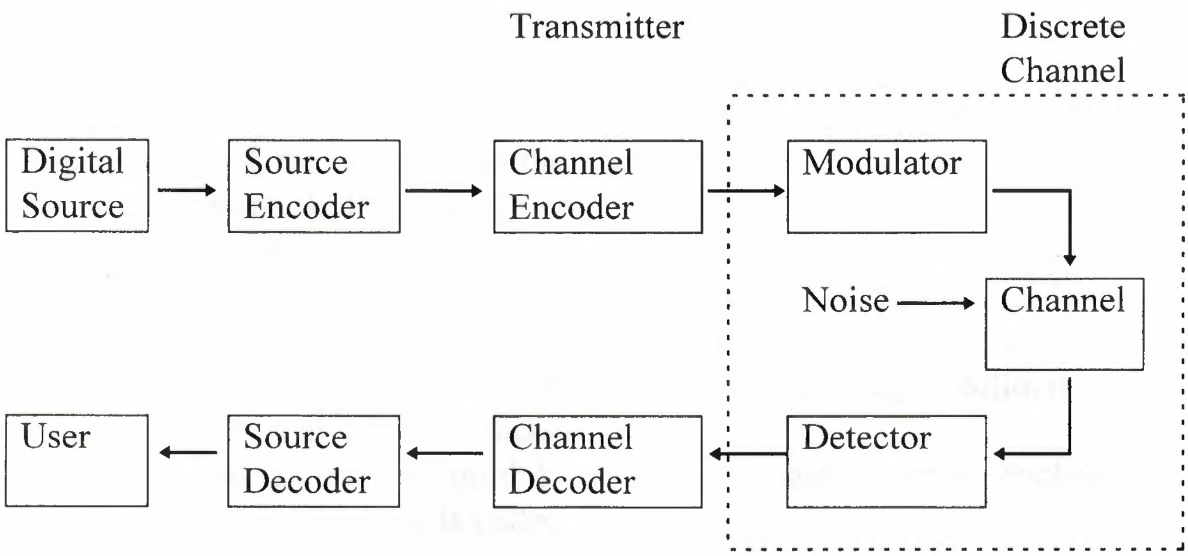
Encoder enable pulse



THE ADVANTAGES OF DIGITAL SYSTEM OVER ANALOG SYSTEM

The impact of the digital communication has experienced a phenomenal growth in both scoop and application. The growth of digital communication is largely due to the following factors:

- The impact of the computer, not only as a source of data but also as a tool for communication, and the demands of other digital services such as telex.
- The use of digital communication offers flexibility and compatibility in that adoption of a common digital formays makes it possible for a transmission system to sustain many different source of information in flexible manner.
- In contrast with analog system, digital communication system can transmit messages with great accuracy.
- The possibility of using regenerative using a regenerative repeaters is a further advantage in digital communication.



BASIC SIGNAL PROCESSING OPERATIONS IN DIGITAL COMMUNICATIONS

The block diagram, mentioned above shows the three basic signal processing operation source coding, channel coding, and modulation. It is assumed that the source of information is digital by nature or converted into it by design.

In source coding, the encoder maps the digital signal generated at the source output into another signal in digital form. The mapping is one to one, and the objective is to eliminate or reduce redundancy so as to provide an efficient representation of the source output. The primary benefit thus gained from the application of source coding is reduce bandwidth requirement. In channel coding, the objective is for the encoder to map the incoming digital into a channel input to map the channel output into an output digital signal in such a way that the effect of channel noise is minimized, that is the combined role of the channel encoder and decoder is to proved for reliable communication over a noisy channel.

In a prescribed fashion in the channel encoder, this provision is satisfied by introducing redundancy and exploiting it in the decoder to reconstruct the original encoder input as accurately as possible. We remove redundancy in channel coding and introduce controlled redundancy.

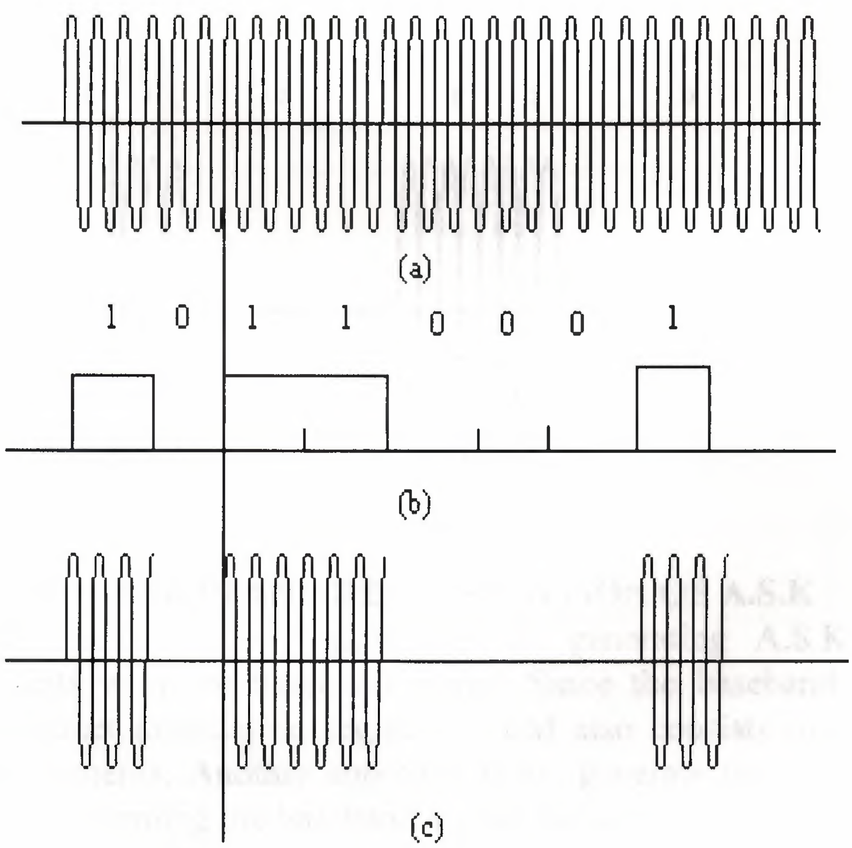
We may clearly source coding, channel coding alone or the two together. The source coding is naturally performed first followed by channel encoding in the transmitter. We produce in the receiver's order; channel decoding is performed first, followed by source decoding.

Whichever combination is used resulting improvement in system performance achieved at the cost of increased circuit complexity.

With the purpose of providing, it is performed for the efficient transmission of the signal over the channel as for the modulation. The modulator particularly operates by keying shifts in the amplitude, frequency, are phase of a sinusoidal carrier waves to the channel encoder output. Digital modulation techniques is referred to as amplitude - shift keying, frequency - shift keying or phase - shift keying respectively. The detector performs demodulation, thereby producing a signal follows the time variations in the channel encoder output.

The combination of modulator, channel, and detector, enclosed inside the dashed rectangle is called discrete channel.

AMPLITUDE SHIFT KEYING



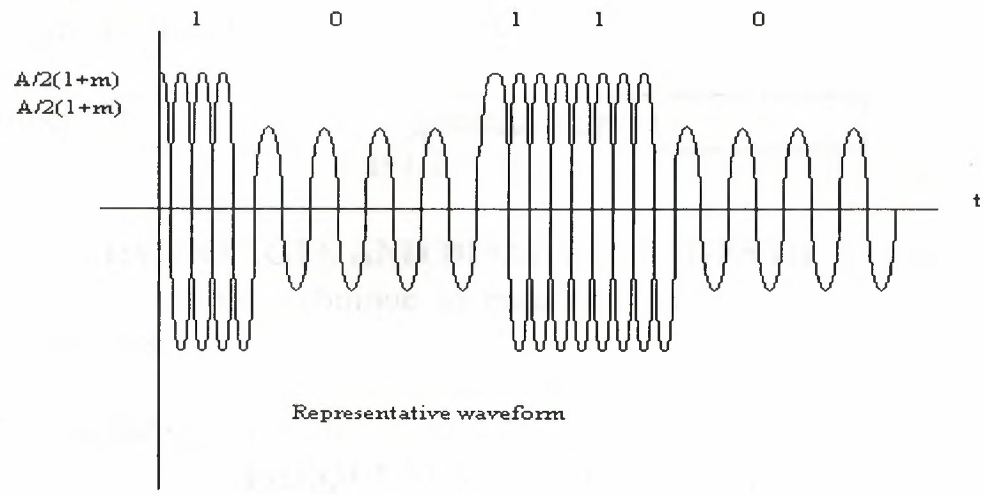
In the above figure, section a is the carrier cosωct, section b is the modulating signal and section c is the modulated signal m(t) cosωct.

Sinusoidal signal has two possible amplitude in each period. Therefore, we can use the following equation for the transmitted signal.

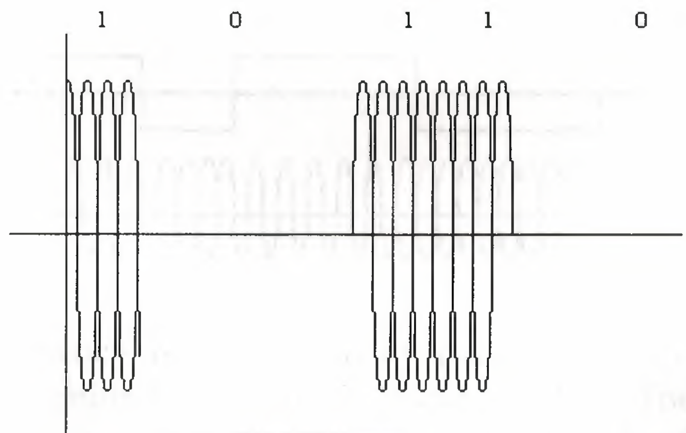
$$S_i(t) = A/2\ (1 + M d_i(t)) * \cos(2 * 3.14159 F_c t)$$

Here i=0 and i=1 to send a binary 0 and 1.

The value of D(t) is +1 or -1 to make the data bipolar. M is the index modulation. The common case used in amplitude shift keying is M=1 which creates on-off condition.



Representative waveform

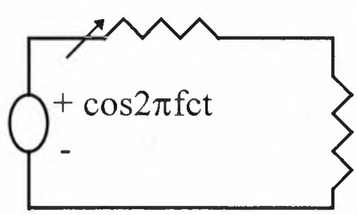


ON-OFF Keying

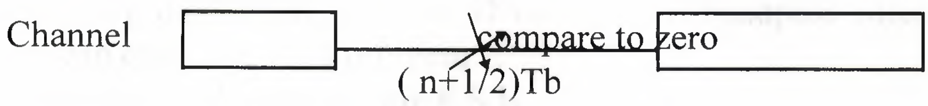
MODULATION AND DEMODULATION OF A.S.K

Modulator: There are two approaches of generating A.S.K. One technique starts with the baseband signal. Since the baseband signal consist of distinct modulation segments, A.M also consists of distinct modulation segments. Another approach is to generate the A.M wave directly without forming the baseband signal. For on-off keying we need simply switch an oscillator on and off.

Modulator for on off keying



Demodulator: There are two types of demodulators. In the first demodulation technique we recover baseband signal by demodulating A.M waveform, then we make the sample of the baseband signal at the mid point of each interval. After samples we compare the values of each sample with other one. If the value of the sample is positive than it is assigned binary 1 on the other hand if the value is found negative than it is assigned binary 0.

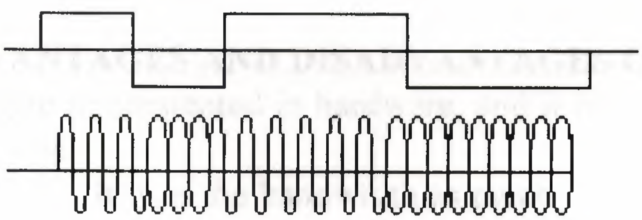


ADVANTAGES AND DISADVANTAGES OF A.S.K

It is a very simple technique in hardware realization. Has two main disadvantages:

- It is susceptible to noise interface.
- it has a wide bandwidth.

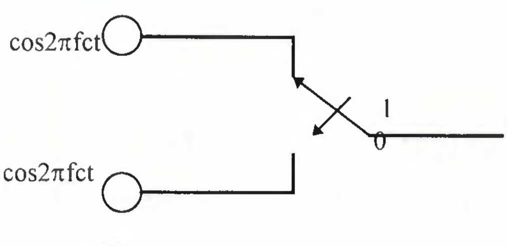
FREQUENCY SHIFT KEYING



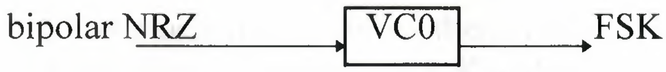
The pulse frequency is transmitted by pulse frequency Wc0 and the binary 1 is transmitted by the pulse frequency Wc1. The bandwidth of this modulation is higher than A.S.K or P.S.K. The frequency shift keying is the sum of the two amplitude shift keying signal.

MODULATION AND DEMODULATION OF F.S.K

Modulation: There are two methods for the modulation of F.S.K signal. First using simple modulator. Second method is using an F.M modulator

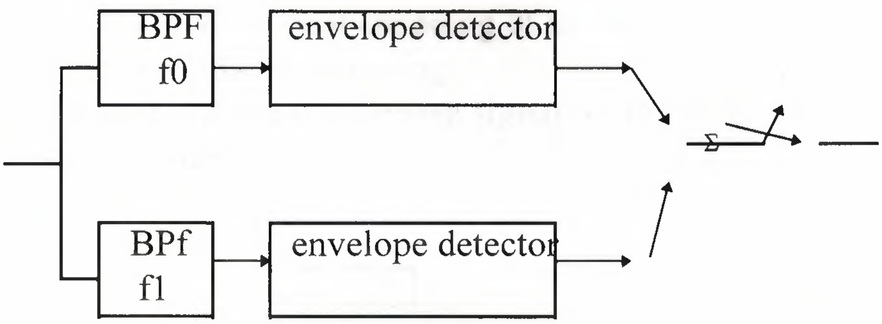


The F.M modulator :



Demodulator: There are two types of F.S.K demodulator. One is coherent detector that requires two carriers at receiver. We can eliminate by using incoherent detector. If we use two bandpass filter at the input we can overcome this problem.

Non-coherent detector for F.S.K



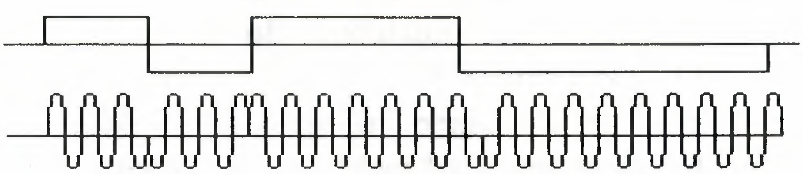
ADVANTAGES AND DISADVANTAGES OF F.S.K

It is very simple implemented in hardware, and it has less susceptibility to noise than A.M.

Disadvantages are: It uses the bandwidth as much as A.M, and the speed of F.S.K is much lower than A.M. As example F.S.K's rated speed is 300 b/s., and rated speed of A.S.K is 1500 b/s.

PHASE SHIFT KEYING

In the most modern modems we use phase shift modulation. This modulation is quite complex. Binary 0's and 1's are represented by changing the phase of the pulse by +90 for binary 1's and -90 for binary 0's. In four phase system this shifting can be +135, +45, -45, -135 so that 2 bits of information can be indicated instead of one as in two phase system.



The mathematical representation of P.S.K is shown below:

$$U(t)=b(t)*A\cos(wt+Q)$$

$$b(t)=1 \text{ if } 1, -1 \text{ if } 0$$

ADVANTAGES AND DISADVANTAGES OF P.S.K

- It has simple modulator realization
- It has much susceptibility then A.S.K and F.S.K
- Bandwidth is less then A.S.K and F.S.K

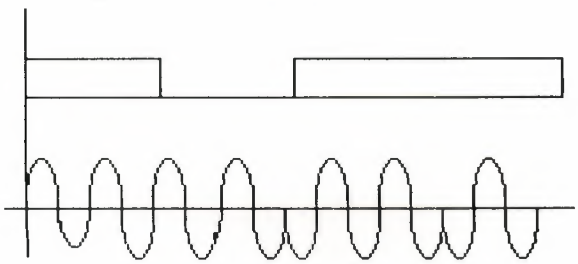
Disadvantage: More complex if compared to other demodulation techniques.

DIFFERENTIAL PHASE SHIFT KEYING

This scheme is noncoherent demodulation of P.S.K.. We don't need reference signal at the receiver by combining two basic operations at the transmitter.

- Differential encoding of the binary wave
- Phase shift keying

To send symbol 0 we phase signal by 180 degrees, and to send symbol 1 we leave the current signal unchanged.



Starting from the A.S.K we observe that the rated speed of A.S.K is 1500 b/s which has relatively better speed than F.S.K, which proved the rated speed of 1200 b/s, but on the other hand P.S.K has much higher speed than both of them. Comparing the noise susceptibility of the schemes, we can see that P.S.K is more stable to the noise effects then other schemes which minimize the probability of error during the data transmission. Another important consideration we must take into account during the designing of communication is channel width which should be as small as possible so that P.S.K provide this feature that requires minimum bandwidth as compare to A.S.K and F.S.K.

The simpler the circuitry the lesser the error rate. The realization of modulator and demodulator of P.S.K is much simpler than that of A.S.K and F.S.K. One main disadvantage is uncertain change of phase during data transmission, we can overcome this problem by using modified version of P.S.K that is D.P.S.K which covers all the future of P.S.K and eliminates the uncertain change of phase during data transmission.

DPSK COMMUNICATION SYSTEM

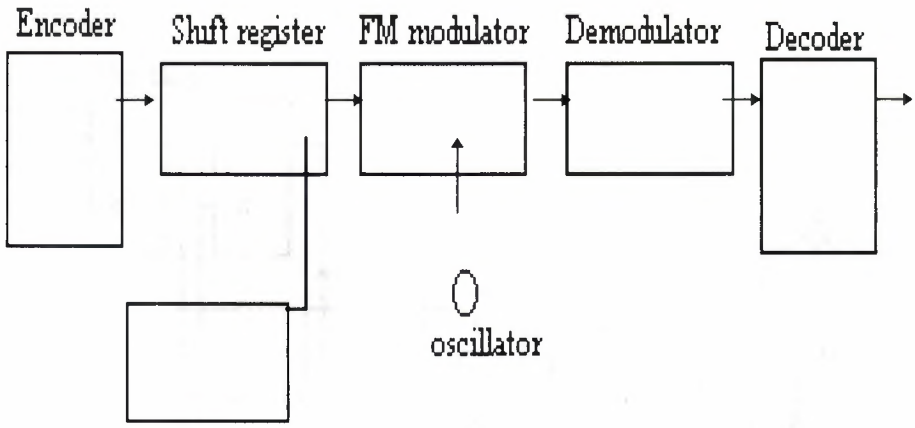
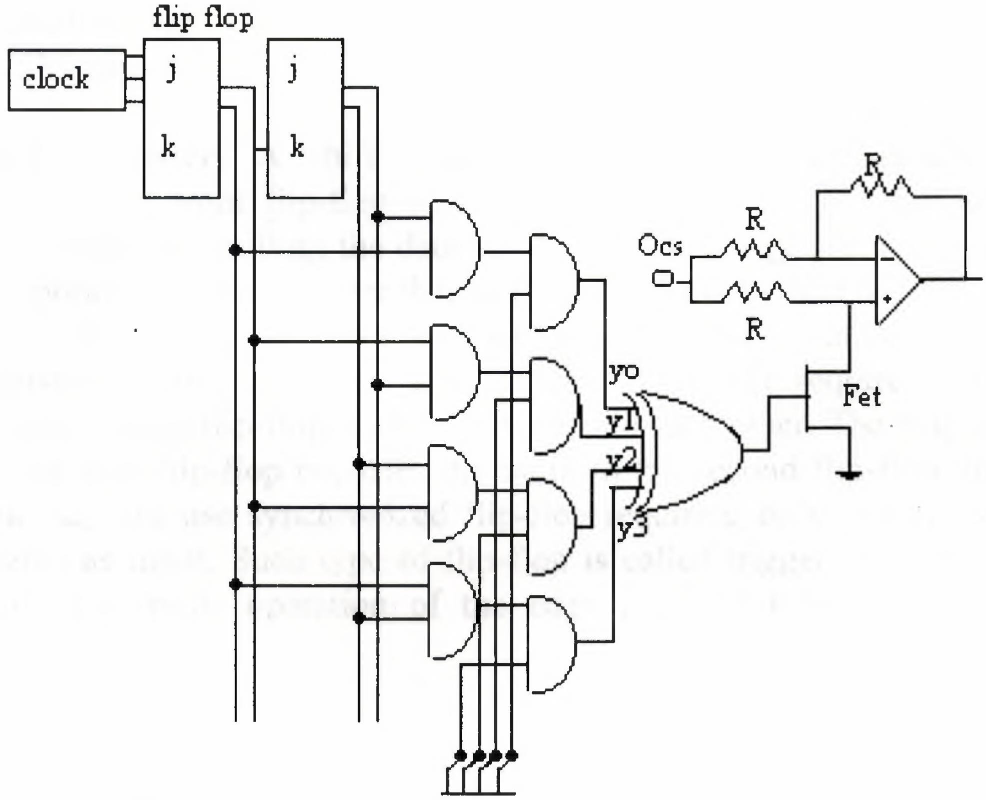


TABLE FOR GENERATION OF D.P.S.K SIGNAL

bk	1 0 0 1 0 0 1 1
dk-1	1 1 0 1 1 0 1 1
dk	1 1 0 1 1 0 1 1 1
transmitted phase cradious	0 0 pi 0 0 pi 0 0 0

DPSK MODULATOR



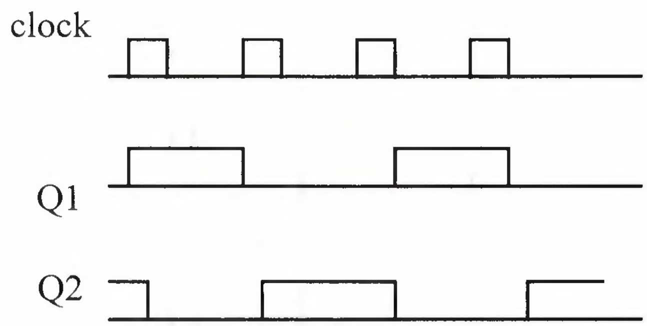
CONTENTS OF DPSK MODULATOR

We may split the DPSK modulator into three of operations. The first two parts are logical circuits that produced logic '1' and '0' at the output line according t the input signal. The three parts of the DPSK modulator are:

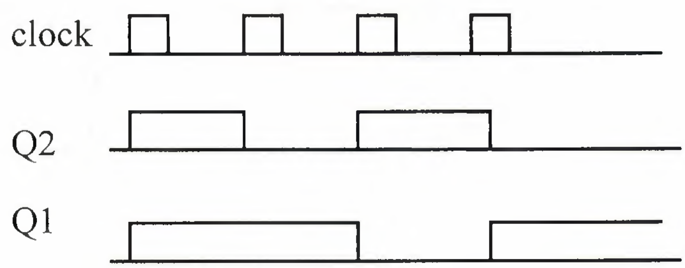
- Shift register
- Multiplexer
- FM modulator

Shift Register: A shift register is sequential logic module constructed from flip-flop that manipulates that bit position of binary data by shifting the data bits to the left or right or this is a temporary memory register that shifts its contents to right or left.

According to this case we can construct simple shift registers according to our aim. This shift register requires two negative-edge flip-flops parallel attached to each other. The output of the first flip-flop becomes the input to the second flip-flop. In our case we use synchronized flip-flop requiring only one clock signal as input. Such type of flip-flop is called trigger edge flip-flop. The basic operation of the edge-triggered flip-flop is as follows:



After the procession the same function of flip-flop starts again as shown:



After this the outputs of flip-flop are connected to the 3-input, 4 output AND gate with the following configuration.

$$Y0=\overline{Q1}*Q2$$

$$Y1=Q1*\overline{Q2}$$

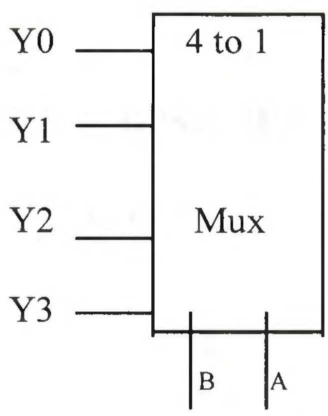
$$Y2=\overline{Q1}*\overline{Q2}$$

$$Y3=Q1*Q2$$

The truth table of the flip-flop are shown below:

C	Q1	Q2	Q2	Q2
0	0	1	0	1
1	1	0	0	1
2	0	1	1	0
3	1	0	1	0

Multiplexer: The task is to select only the one output at the same time by using data selector or multiplexer. A multiplexer is a modular device that selects one of the many input lines to appear on a single output line. The functional diagram and truth table of multiplexer is as follows:



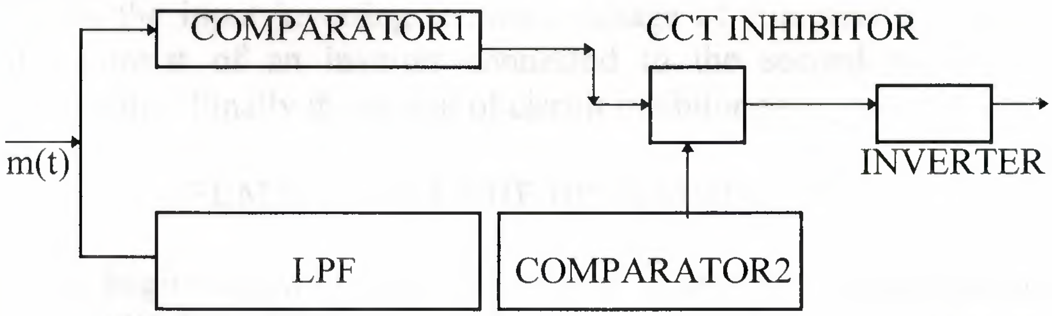
TRUTH TABLE

B	A	Y
0	**0**	**Y0**
0	**1**	**Y1**
1	**0**	**Y2**
1	**1**	**Y3**

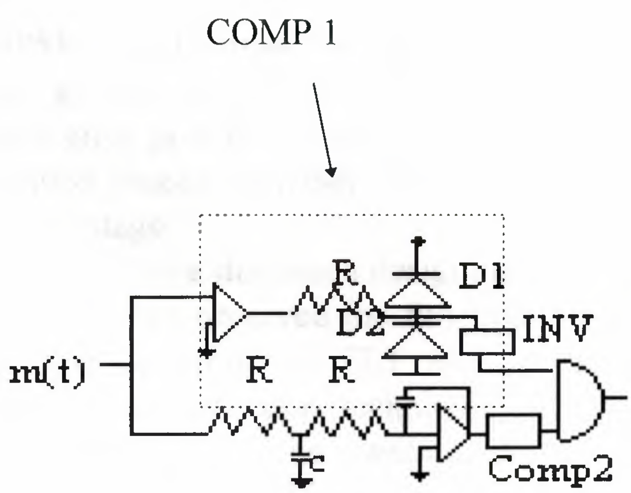
F.M Modulator: The field effect transistor works as switching operator for the amplifier. This is called analog switching operation. It received the input from the output of the multiplexer and performs the switching operation according to the gate voltage level. The reason to chose the Fet is that it has very high input impedance, from 1 megaohm to several megaohms, an other advantage of field effect transistor is that it has very low susceptibility to noise that is the very important consideration in any communication system this make it superior to BJT. It is important to note that a BJT is an analog operating device that works on negative and positive response of the sinusoidal signal. The operation of Fet is in such a way that when it gets logic 1 from multiplexer, short circuit to the ground, so that amplifier sinusoidal signal at its positive terminal. A non-inverting output appears at the output of the amplifier.

DPSK DEMODULATOR

When the mission of the transmission of the modulated signal, the next duty is to demodulate the received signal. The block diagram of DPSK demodulator is:



DPSK DEMODULATOR



The received DPSK modulated signal at the receiver passes to two branches of the modulator. In the first branch the signal goes into the amplitude limiter or comparator. When the comperator change the input signal into the digital signal by rejecting the negative part of the message signal.

In the second branch the message signal passes to another comperator via low pass filter where the low pass filter stable the signal for certain value of amplitude. A comparator circuit is a linear input voltage and compare to another reference voltage, the output being a digital condition representing the input voltage exceeded the reference voltage. The output values of comparator can be changed to require maximum and minimum level of signal according to the need of operation. Comparator is LM339 comparator, that provided maximum 5 volt and minimum 0 an output and it is also a suitable choice for our purpose.

The value of reference voltage is as follows:

$$Rf=R2/R1+R2*Vcc$$

Vcc is the input inverting terminal voltage of comparator. The circuit also consist of an inverter connected to the second output of the comparator. Finally the output of circuit inhibitor again inverted.

SUMMARY OF THE DPSK MODULATION

At the beginning, we talked about digital modulation techniques such as ASK, PSK, FSK, DPSK. And then we gave our attention to DPSK, which the phase remained unchanged for the binary 1 and changes by 180 degrees.

DPSK modulation is the most comprehensive modulation technique among the others due to its high rate of data transmission speed, low error probability and simple designing of modulator even the demodulation process is rather complicated and sensitive and this is the only disadvantage.

Later we have discussed designing the scheme and it required few circuitry parts. We observed the functions of shift register, multiplexer, and combine operation of FET and operational amplifier. We also designed the demodulator technique, which is little difficult than modulation, because of the transmitted signal at the reception of the DPSK demodulator is difficult to control according to the change of phase in the received signal. The demodulation scheme required low pass filter, amplitude limiter or comparator and one simple logic circuit called circuit inhabited.

Finally we see that DPSK gives the probability to avoid the coherent reception. DPSK eliminates the use of local oscillator at the receiver.

In modern communication system the use of Differential phase shift keying has become wide because of its advantageous features. They have high speed of data transmission, low probability of error, simple realization of DPSK modems.

I want to continue giving information about the modems, which have many applications and relations with modulation, because we know that without modulation there would be no modem.

MODEMS

MODEMS: A modem performs two main functions, modulation and demodulation. Inside a modem comprises three sections: circuitry associated with the transmit function, circuitry associated with the receive function, and circuitry for timing and power supplies which serves the other two.

MODEM COMPONENTS: A modem consists of a power supply, a transmitter and a receiver. The power supply converts AC to DC, which provides the voltage necessary to operate the modem's circuitry. In the transmitter section, a modulator, amplifier, filter and waveshaping and signal control circuitry convert digital DC pulses to an analog, wave-shaped signal that can be transmitted over a telephone line. The receiver section contains a demodulator and associated circuitry that reverse the process by converting the analog telephone signal back into a series of digital pulses that the computer or terminal device can use. Fig 2 diagrams the basic components of modem. Those components associated with the receiver are located in the lower portion. Fig 2 represents a general modem indicated by dashed lines are applicable only to synchronous devices. In addition, other components a microprocessor, ROM, and RAM that can be included to provide a modem with intelligence have been purposely omitted from this figure to enable us to focus on the modulation and demodulation of data by the transmitter and receiver in each device.

MODEM TRANSMITTER SECTION: The key components of a modem's transmitter section include a data encoder, a scrambler, a modulator, an amplifier, a filter, a timing

Fig 1: Signal conversion performed by modems. A modem converts a digital signal to analog tone and reconverts the analog tone into its original digital signal.

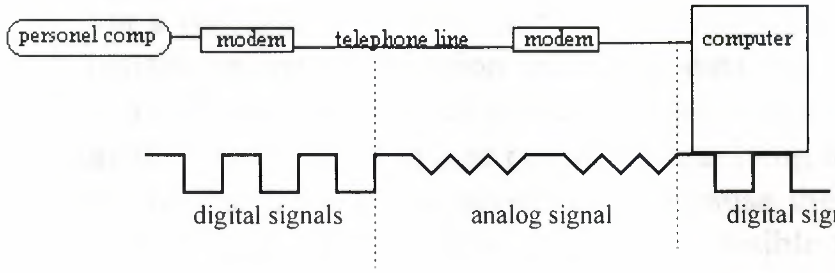
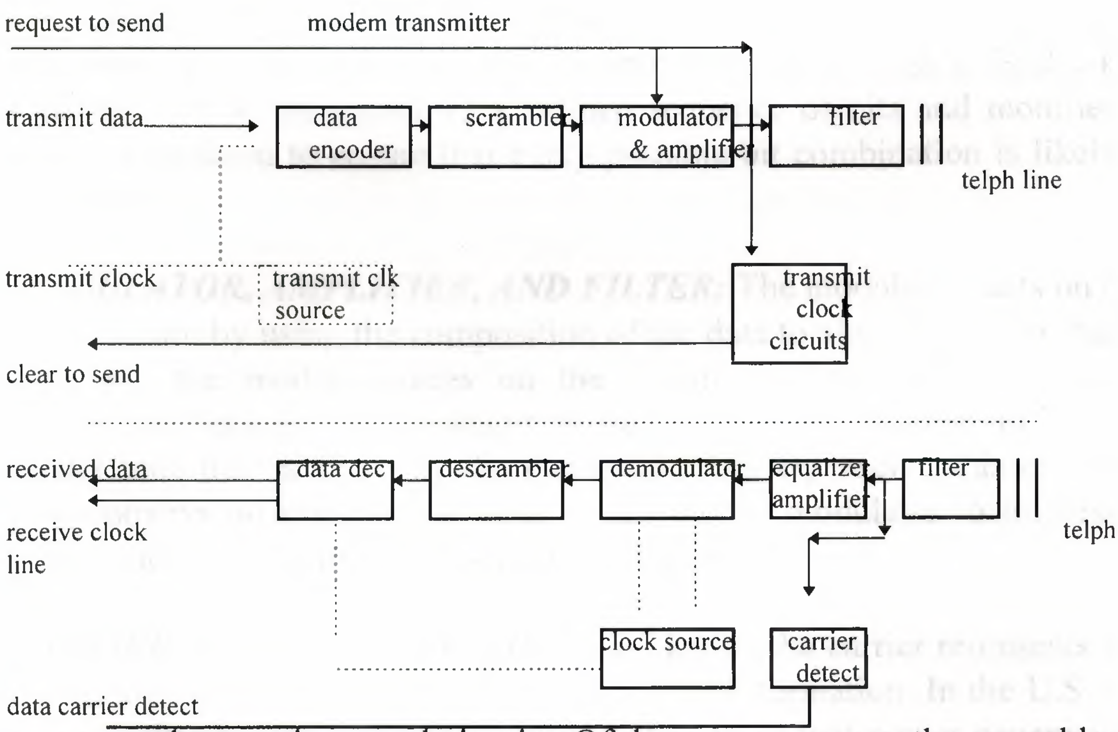


Fig 2: Basic components of a modem



source, and transmit control circuits. Of these components, the scrambler and transmit clock provided by the timing source are used only in synchronous modems.

The data encoder is an option built into many modems. The encoder is used in conjunction with some modulation schemes, enabling each signal to represent more than one bit of information.

SCRAMBLERS: As we know, synchronous modems provide clocking signals on RS-232 interface. When a modem receives a modulated synchronous data stream and passes the demodulated data to an attached terminal device, it also provides a clocking signal to the data terminal.

This clocking signal tells the terminal device when to sample pin3, the received data circuit, and it is produced by the modem from the received data. Thus the received clock signal is commonly referred to as a derived clocking signal.

For a synchronous modem's received clock to function correctly, it must remain in synchronization with the data being received. This requires a sufficient number of changes in the composition of the data. For example, 0 to 1 and 1 to 0 to permit the receiving modem's circuitry to derive timing from the received data. Because the data stream can consist of an arbitrary bit pattern, it is quite possible that the data will contain long sequences of 0's and 1's. when these sequences occur, the data will not provide the modem's receiver with a sufficient number of transitions for clock recovery. This condition is responsible for the incorporation of scramblers into synchronous modems.

A scrambler modifies the data to be modulated based on a defined algorithm. This algorithm is normally implemented through a feedback shift register, which examines a defined sequence of bits and modifies their composition to ensure that every possible bit combination is likely to occur.

MODULATOR, AMPLIFIER, AND FILTER: The modulators acts on a serial stream by using the composition of the data to alter the carrier that tone that the modem places on the communications line. When a connection between two modems is established, one modem raises a carrier tone that is heard by the distant modem. By itself , this carrier tone conveys no information. It is varied by the modulator to impress information that the distant modem demodulates.

CARRIER SIGNAL STRENGTH: As we know, the carrier represents a continuous frequency that is altered to convey information. In the U.S.A and by other bodies in foreign countries, to ensure that power generated by the modem does not harm the telephone network

The transmit level of a modem used on the switched telephone network must be set to -9.0 dBm. Understanding what the modem transmit level means and the importance of the sensitivity level of a modem's receiver requires of power ratios and use of logarithms.

A common method used to categorize the quality of transmission on a circuit ,is to state the ratio of power received to power transmitted. This is called power ratio, expressed by a signal flowing down a transmission path and the gain resulting from the operation of amplifiers used to rebuild analog signals. The gain or loss of power expressed by

$$dB = 10\log_{10} p_0/p_I$$

where db= power ratio in decibels

p_0=output of received power

p_I=input power

the term decibel is used to express the ratio of two amounts of power, also used to provide measurement of the relative loudness of sound. The logarithm to the base 10 is known as the common logarithm. The value of the common logarithm of a number indicates how many times 10 must be multiplied by itself to yield that number.

In the prior equation, a situation where the output power is less than the input power results in a power loss . Thus a negative number must be placed before answer. If the output power exceeds the input power, there is power gain.

As an example, assume that the measured input and output powers in this equation are 1mw and 100 mw. The power ration than becomes

$dB=10\log_{10}100/1=10\log_{10}100=20$

In this example, a power gain occurred, due to use of amplifier. Suppose input and output powers are reversed, with 100 mw input and 1mw output. The power ration then becomes

$dB=10\log_{10}1/100=-20$

here, we have a power loss, indicated by the negative sign, meaning that there is less output power than input power.

Establishing standardized testing of telephone circuits required the use of a signal defined in terms of power and frequency. The signal selected as a test tone uses .001 watt of power at a frequency of 1004 Hz in the US and 800 Hz in most European countries. This represents the average amount of power generated in the transmitter of a telephone during a typical voice conversation. This signal also provides telephone company with a standardized for determining gains and losses. To ensure that no one forgets 1 mw is the reference level of power transmitted, the letter m is attached to the power level, and the measurement is in decibelmiliwatts.

$$dBm = 10\log_{10} \frac{\text{signal power}}{\text{1 miliwatt}}$$

In comparing dB and dBm, note that dB is used to express the amount of gain or loss, while dBm is used to indicate the new power level due to gain or loss.

In communications, an alternating voltage is used to define carrier. The maximum permissible voltage level that can be placed on a telephone line is 2.2 volt peak to peak signal. That signal level is used to represent a zero reference point. Since a higher voltage level is not permissible , all other permissible voltage levels are less than 2.2 volts, and are represented by negative dBm numbers. Most modems designed for use in the US transmit a carrier signal at -9 dBm or 800 mv. The signal level may varry, due to a variable distance between subscribers and telephone company service. Normally the telephone company corrects for line loss by installing a fixed loss loop resistor on the subscriber line and adding resistance. This enables a modem's carrier signal to arrive correctly at their office when the transmit signal is at -9 dBm. Thus, although many modems include a feature that enables the transmit level to be adjusted, its default of -9 dBm is normally the correct level to use.

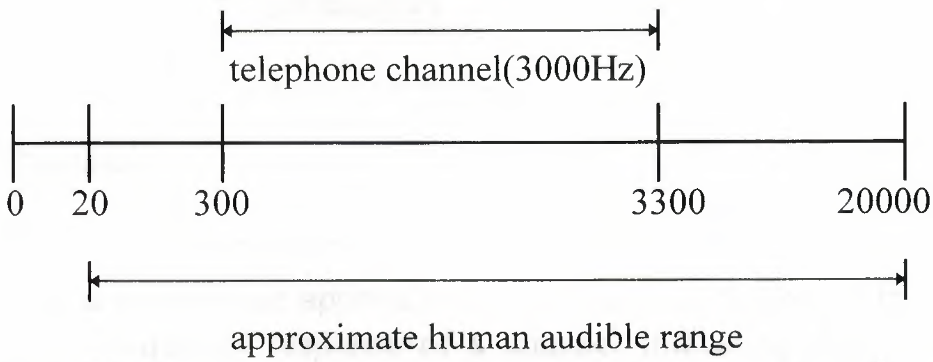
At least 1/40 th of a transmitted signal must appear at its destination. This means that the power loss between the transmitter of one modem and the receiver of a second modem should not exceed 16 dB. Since the use of logarithms enables gains and losses to be added algebraically, this means that the transmission of a carrier at -9 dBm must result in the other modem's receiving the signal at -25 dBm.
Although a received signal at -25 dBm represents a 45 mV peak to peak voltage that is significantly less than the original 800 mV peak to peak signal, that signal can be easily received by modems. In fact, most modems have a sensitivity level in excess of -40 dBm, while some modems have a sensitivity level as low as -50 dBm. However, once a signal is below -30 dBm, impulse noise can easily adversely affect the signal, which makes a modem's receiver sensitivity level below -30 dBm.

EQUALIZER: The equalizer illustrated for the modem receiver in the lower portion is designed to measure the characteristics of a received analog signal and to adjust itself to that signal. In doing so, the equalizer minimizes the effect of attenuation and delay on the various components of a transmitted signal. To accomplish this task, the modem's transmitter prefixes each transmission with a short training signal whenever the direction of transmission changes. This training signal represents a pre-defined modulation of the carrier whose ideal reception characteristics are known by the equalizer in the receiver of the distant modem thus the equalizer will be adjusted by the receiving modem until the best possible signal is received.

To appreciate the operation and use of equalizers, we first focus on several basic data channel parameters and the method by which communications carriers create a telephone channel.

BANDWITH: Bandwidth is a measurement of the width of a range of frequencies such that: $B=f_2-f_1$ where b is the bandwidth, f_2 is the highest frequency, and f_1 is the lowest frequency in a range of frequencies. Figure 3 illustrates the bandwidth channel compared with the audio spectrum heard by the human ear. As we know the term Hertz(Hz) is used to represent one cycle per second.

Fig 3: Bandwidth of a telephone channel



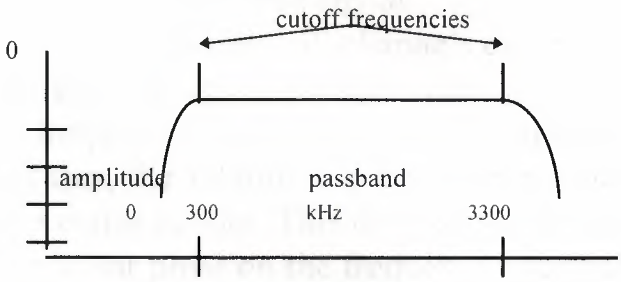
The 3000 Hz that forms a telephone channel is commonly referred to as the passband of the channel. The term refers to a contiguous portion of an area in the frequency spectrum that permits a predefined area in the frequency spectrum that permits a predefined range of frequencies to pass. Thus the passband of a telephone channel permits frequencies between 300 and 3300 Hz to pass.

Frequencies under 300 Hz and above 3300 Hz are essentially not required to the understanding of a telephone conversation, although their absence precludes a soprano from being fully appreciated at the other end of the telephone connection. By transmitting only 3000Hz instead of the full 20000 Hz that the human ear can hear, the bandwidth required for each call is reduced by a factor of six. The bandwidth reduction enables telephone companies to more efficiently employ frequency division multiplexing, a technique that allows many voice calls to be carried simultaneously on a common circuit routed between telephone company offices.

To construct a telephone channel passband, the telephone company uses low- and high pass filters designed to permit either all signals up to a predefined frequency or all signals under a predefined frequency to pass through the channel. As a result of the use of filters, the amplitude-frequency response becomes rounded at the cutoff frequencies at which the filters operate, and then begins to approach large negative values as the filters attenuation becomes more pronounced. Figure 4 illustrates how filters create a passband on a telephone channel.

Ideally, all frequencies across the passband of a telephone channel should undergo the same amount of attenuation as illustrated by the straight line between the cutoff frequencies shown in figure 4. High frequencies lose their strength more rapidly than low frequencies, so attenuation increases as frequencies increase toward the end of the passband. In addition, attenuation increases as the edges of the operating frequencies of passband

Fig 4: Telephone channel passband creation



filters on a channel are approached. As a result of these two factors, the amplitude-frequency response of a channel indicating the attenuation distortion that signals experience resembles the response in fig 5

To minimize the effect of attenuation distortion, some modems include an attenuation equalizer. This type of equalizer introduces a variable gain at frequencies within the passband that compensate for the differences in attenuation between high and low frequencies and the increased attenuation at the edges of the passband

DELAY DISTORTION: A second type of distortion that affects the recovery of information from a received signal is delay distortion. In a distortionless channel, all frequencies pass through the channel at the same speed. This results in the frequency and phase of the signal having a constant relationship with respect to time, and ensures that the

Fig 5: Typical amplitude- frequency response across a voice channel

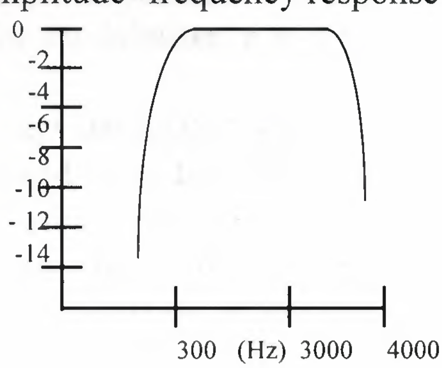
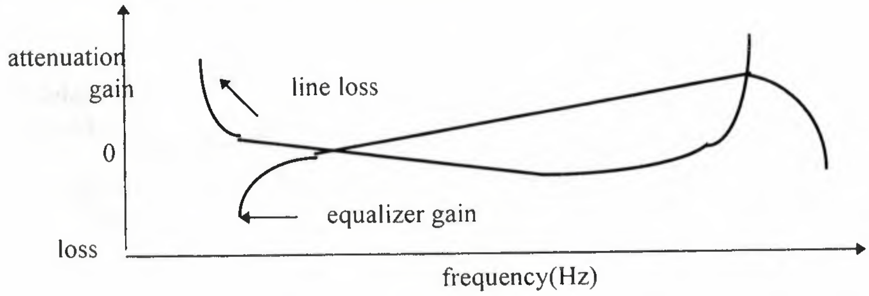


Fig 6: Using an equalizer to correct attenuation



transmission of one signal does not interface with the reception of previously transmitted signal.

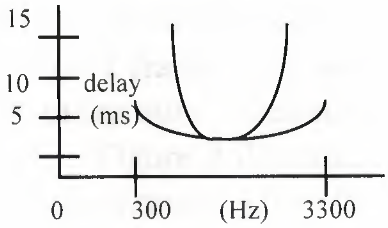
Unfortunately, all channels except those in the lab have a degree of distortion. When distortion occurs, the relationship between the phase and frequency of a signal becomes nonlinear. As the level of distortion increases, the relationship between the phase and frequency of a signal degenerates further. This degeneration, called phase delay, is measured at a particular point on the frequency spectrum by dividing the phase of the signal by its frequency. The direct measurement of phase relay is not practical, because it would require an absolute phase reference to keep track of phase changes over multiples of 360 degrees. Instead, electrical engineers use the slope of the phase versus frequency, which is known as envelope delay.

Envelope delay is the first derivative of phase delay. The shape of the envelope delay curve, obtained by measuring delays at different frequencies, reflects the degree of change in the slope of the phase-versus-frequency curve. This delay change varies based on the transmission distance. Fig 7 illustrates two typical delay curves for signals transmitted on a telephone channel, the steeper curve represents the envelope delay on a longer distance circuit than the flatter curve.

To see the potential effect of envelope delay on communications, assume that a modem transmits one of two tones. We know that this is one of the methods used for modulation, a technique referred as frequency-shift-keying.

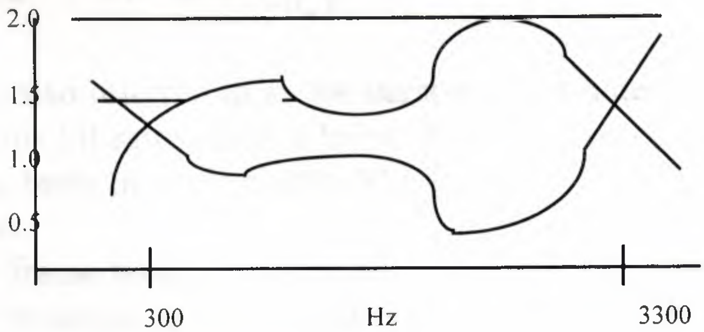
Due to delays associated with different frequencies, a delay in f_1 being received could mean that tone reaches the receiving modem as the dame time as f_2, which represents a different binary value. This could cause one received signal to be superimposed on the second signal, causing one tone to distort the other.

Fig 7: Typical envelope delay curves:



Although all communications circuits exhibit a degree of delay, it is important to flatten the delay time across the passband to minimize the potential for one tone of a signal to be superimposed on another tone. Some modems are designed with delay equalizers that introduce a delay approximately inverse to that exhibited by the telephone channel. Through a delay equalizer, the delay time associated with frequencies within the passband can be made relatively flat as shown in fig 8. This flattening reduces the potential of one tone interfering with another, a condition formally referred as intersymbol interference.

Fig 8: Using a delay equalizer:



TIME DIVISION MULTIPLE ACCESS

With time-division multiple access, only one carrier uses the transponder at any one time, and therefore intermodulation products, which result from the nonlinear amplification of multiple carriers, are absent. This leads to one of the most significant advantages of TDMA, which is that the transponder traveling wave tube(TWT) can be operated at maximum power output or saturation level.

Because the signal information is transmitted in bursts, TDMA is only suited to digital signals. Digital data can be assembled into burst format for transmission and reassembled from the received bursts, through the use of digital buffer memories.

Figure 1 illustrates the basic concept, in which the stations transmit bursts in sequence. Burst synchronization is required, and in the system illustrated in figure 1, one station is assigned for the purpose of transmitting reference bursts to which the others can be synchronized. The time interval from the start of one reference burst to the next is called a frame. A frame contains the reference bursts are and the bursts from the other earth stations, these being shown as A, B, C in figure 1.

Figure 2 illustrates the basic principles of burst transmission for a single channel. Overall, the transmission appears continues because the input and output bit rates are continuous and equal. However, within the transmission channel, input bits are temporarily stored and transmitted in bursts. Since the time interval between burst is the frame time T_F , the required buffer capacity is

$$M = R_b T_F$$

The buffer memory fills up at the input bit rate R_b during the frame time interval. These M bits are transmitted as a burst in the next frame without any break in continuity of the input. The M bits are transmitted in the burst time T_B, and the transmission rate, which is equal to the burst bit rate, is

$$R_T = M / T_B$$
$$= R_b T_F / T_B$$

This also referred to as the burst rate but note that this means the instantaneous bit rate within a burst. It will be seen that the average bit rate for the burst mode is simply M / T_F, which is equal to the input and output rates.

The frame time T_F will be seen to add to the overall propagation delay. For example in the simple system illustrated in figure 2, even if the actual propagation delay between transmit and receive buffers is assumed to be zero, the receiving side would still have to wait a time T_F before receiving the first transmitted burst. In a Geostationary satellite system the actual propagation delay is a significant fraction of a second, and excessive delays from other causes must be avoided. This sets an upper limit to the frame time, although with current technology, other factors restrict the frame time to well below this limit. The frame period is usually chosen to be a multiple of 125μs, which is the standard sampling period used in pulse-code modulation. Telephony systems, as this ensures that the PCM samples can be distributed across successive frames at the PCM sampling rate.

Figure 3 shows some of the basic units in a TDMA ground station, which for discussion purposes is labeled earth station A. Terrestrial links coming into earth station A carry digital traffic addressed to destination stations, labeled B, C, X. It is assumed that the bit rate is the same for the digital traffic on each terrestrial link. In the units labeled terrestrial interface modules, the incoming continuous bit rate signals are converted into the intermittent burst rate mode. These individual burst-mode signals are time-division multiplexed in the time division multiplexer, so that the traffic for each destination station appears in its assigned time slot within a burst.

Certain time slots at the beginning of each burst are used to carry timing and synchronizing information. These time slots collectively are collectively referred to as the preamble. The complete burst containing the preamble and the traffic data is used to phase-modulate the radio-frequency carrier. Thus the composite burst which is transmitted at RF consists of a number of time slots as shown in figure 4

The received signal at an earth station consists of bursts from all transmitting stations arranged in the frame format as shown in figure 4 The RF carrier is converted to intermediate frequency carrier(IF), which is then demodulated. A separate preamble detector provides timing information for transmitter and receiver along with a carrier synchronizing signal for the phase demodulator. In many systems a station receives its own transmission along with the others in the frame, which then be used for burst timing purposes.

A reference burst is required at the beginning of each frame to provide timing information for the acquisition and synchronization of bursts. In the INTELSAT international network at least two reference stations are used, one in the east and one in the west. These are designated primary reference stations, one of which is selected as the master primary. Each primary station is duplicated by a secondary reference station, making four reference stations in all. The fact that all the reference stations are identical means that any one can become the master primary. All the system timing is derived from the high stability clock in the master primary, which is accurate to 1 part in 10^{11}. A clock on the satellite is locked to the master primary, and this acts as the clock for the other participating earth stations. The satellite clock will provide a constant frame time, but the participating earth stations must make corrections for variations in the satellite range, since the transmitted bursts from all the participating earth stations must reach the satellite in synchronism.

In the INTELSAT system two reference bursts are transmitted in each frame. The first reference burst, which marks the beginning of a frame, is transmitted by a master primary reference station, and contains the timing information needed for the acquisition and synchronization of bursts. The second reference burst, which is transmitted by a secondary reference station, provides synchronization but not acquisition information. The secondary reference burst is ignored by the receiving earth stations unless the primary or master station fails.

REFERENCE BURST

The reference burst that marks the beginning of a frame is subdivided into time slots or channels used for various functions. These will differ in detail for different networks, but fig 4 shows some of the basic channels that are usually provided. These can be summarized as follows:

Guard time: A guard time is necessary between bursts to prevent the bursts from overlapping. The guard time will vary from burst to burst depending on the accuracy with which the various bursts can be positioned within each frame.

Carrier and bit-timing recovery(CBR): To perform coherent demodulation of the phase-modulated carrier, a coherent carrier signal must be first recovered from the burst. An unmodulated carrier wave is provided during the first part of the CBR time slot. This is used as synchronizing signal for a local oscillator at the detector, which then produces an output coherent with the carrier wave. The carrier in the subsequent part of the CBR time slot is modulated by a known phase-change sequence which enables the bit timing to be recovered. Accurate bit timing is needed for the operation of the sample-hold function in the detector circuit.

Burst code word(BCW): This is a binary word, a copy of which is stored at each earth station. By comparing the incoming bits in a burst with the stored version of the BCW, and this in turn provides an accurate time reference for the burst position in the frame. A known bit sequence is also carried in the BCW, which enables the phase ambiguity associated with the coherent detection.

Station identification code(SIC): This identifies the transmitting station. Figure 5 shows the makeup of the reference bursts used in certain of the INTELSAT networks. The numbers of symbols and the corresponding time intervals allocated to the various functions are shown. In addition to the channels already described, a coordination and delay channel is provided. This channel varies the identification number of the earth station being adressed, and various codes used in connection with the acquisition and synchronization of bursts at the adressed earth station. It is also necessary for an earth station to be necessary to know the propogation delay time to the satellite, to implement burst acquisition and synchronization. In the INTELSAT system the propogation delay is computed from measurements made at the reference station and transmitted to the earth station through the coordination and delay channel.The other channels in the INTELSAT reference burst are the following:

TTY:telegraph order-wire channel, used to provide telegraph communications between earth stations.

SC: Service channel which carries various network protocol and alarm messages.

VOW: Voice-order-wire channel used to provide voice communications between earth stations. Two VOW channels are provided.

PREAMBLE AND POSTAMBLE

The preamble is the initial portion of a traffic burst which carries information similar to that carried in the reference burst. In some systems the channel allocations in the reference bursts and the preambles are identical. No traffic is carried in the preamble. In figure 4, the only difference between the preamble and the reference burst is that the preamble provides an order-wire channel.

For the INTELSAT format shown in the figure 5, the preamble differs from the reference burst in that it does not provide a coordination and delay channel. Otherwise, the two are identical.

As with the reference burst, the preamble provides a carrier and bit-timing recovery channel and also a burst code word channel for burst timing purposes. The burst code word in the preamble of a traffic burst is different from the burst code word in the reference bursts, which enables the two types of bursts to be identified.

CARRIER RECOVERY

A factor which must be taken into account with TDMA is that the various bursts in a frame lack coherence so that carrier recovery must be repeated for each bursts. This applies to the traffic as well as the reference bursts. Where the carrier recovery circuit employs a phase locked loop. A problem known as hang-up can occur. This arises when the loop moves to an unstable region of its operating characteristic. The loop operation is such that it eventually returns to a stable operating point, but the time required to do this may be unacceptably long for burst type signals.

NETWORK SYNCHRONIZATION

Network synchronization is required to ensure that all bursts arrive at the satellite in their correct time slots. Timing markers are provided by the reference bursts, which are tied to a highly stable clock at the reference station and transmitted through the satellite link to the traffic stations. At any given traffic station, burst code word in the reference burst signals the start of receiving frame(SORF), the marker coinciding with the last bit in the burst code word.

It would be desirable to have the tightly stable clock located aboard the satellite as this would eliminate the variations in propagation delay arising from the uplink for the reference station, but this is not practical because of weight and space limitations. However, the reference bursts retransmitted from the satellite can be treated for timing purposes.

The network operates what is termed a burst time plan, a copy of which is stored at each earth station. The burst time plan shows each earth station where the receive bursts intended for it are relative to the SORF marker. This is illustrated in figure 6. At earth station A the SORF marker is received after some propagation delay, and the burst time plan tells station A that a burst intended for it and follows at time T_A after the SORF marker received by it. Likewise for station B the propagation delay is t_B , and the received bursts start at T_B after the SORF markers received at station B. The propagation delays for each station will differ, but typically they are in the region of 120 ms.

The burst time plan also shows a station when it must transmit its bursts in order to reach the satellite in the correct time slots. A major advantage of the TDMA mode of operation is that the burst time plan is essentially under software control, so that changes in traffic patterns can be accommodated much more readily than FDMA , here modifications to hardware are required. Against this, implementation of the synchronization is a complicated process. Corrections must be included for changes in propagation delay which result from the slowly varying position of the satellite. In general the procedure for transmit timing control has two stages. First there is the need for a station just entering, or reentering after a long delay, to acquire its correct slot position, this being referred to as burst position acquisition. Once the time slot has been acquired, the traffic station must maintain the correct position, this being known as burst position synchronization.

Open-loop timing control: This is the simplest method of transmit timing. A station transmits at a fixed interval following reception of the timing markers, according to the burst time plan, and sufficient guard time is allowed to absorb the variations in propagation delay. The burst position error can be large with this method longer guard times are necessary, which reduces frame efficiency. However, for frame times longer than about 45ms, the loss of efficiency is less than 10 percent. In a modified version of the open-loop method known as adaptive open-loop essentially under software control, so that changes in traffic patterns can be accommodated much more readily than FDMA , here modifications to hardware are required. Against this, implementation of the synchronization is a complicated process. Corrections must be included for changes in propagation delay which result from the slowly varying position of the satellite. In general the procedure for transmit timing control has two stages. First there is the need for a station just entering, or reentering after a long delay, to acquire its correct slot position, this being referred to as burst position acquisition. Once the time slot has been acquired, the traffic station must maintain the correct position, this being known as burst position synchronization.

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Loopback timing control: Loopback refers to the fact that an earth station receives its own transmission from which it can determine range. It follows that the Loopback method can only be used where the satellite transmits a global beam encompassing all the earth stations in the network. A number of methods are available for the acquisition process, but basically these all require some form of ranging to be carried out so that a close estimate of the slot position can be acquired. In one method, the traffic station transmits a low level burst consisting of the preamble only. The power level is 20 to 25 dB below the normal operating level to prevent interference with other burst, and the short burst is swept through the frame until it is observed to fall within the assigned time slot for the station. The short burst is then increased to full power, and fine adjustments in timing are made to bring it to the beginning of the time slot. Acquisition can take up to about 3 s in some cases

Feedback timing control: Where a traffic station lies outside the satellite beam containing its own transmission, Loopback of the transmission does not occur, and some other method must be used for the station to receive ranging information. Where the synchronization information is transmitted back to earth station from a distant station, this is termed feedback closed-loop control. The distant station may be reference station, as in the INTELSAT network, or it may be another traffic station which is designated 'partner.' During the acquisition stage, the distant station can feedback information to guide the positioning of the burst, and once the correct time slot is acquired, the necessary synchronizing information can be fed back on continuous basis.

Figure 7 illustrates the feedback closed-loop control method for two earth stations A and B. The SORF marker is used as a reference point for the burst transmissions. However, the reference point which denotes the start of transmission frame(SOTF) has to be delayed by a certain amount, shown as D_A for earth station a and D_B for earth station B. This is necessary so that the SOTF reference points for each earth station coincide at the satellite transponder.

TRAFFIC DATA

The traffic data is immediately follow the preamble in a burst. As shown in figure 4, the traffic data subburst is subdivided into time slots addressed to the individual destination stations Any given destination station selects only the data in the time slots intended for that station. As with FDMA networks, TDMA networks cab be operated with both preassigned and demand assigned channels.

COMPARISON OF FDMA AND TDMA

With frequency-division multiple access, the modulated carriers at the input to the satellite are retransmitted from the satellite as a combined frequency-division multiplexed signal. Each carrier retains its modulation, which may be analog or digital. For this comparison, digital modulation will be assumed. The modulation bit rate for each carrier is equal to the input bit rate.

With time-division multiple access, the uplink bursts which are displaced in time from one another are retransmitted from the satellite as a combined time-division-multiplexed signal. The uplink bit rate is equal to the downlink bit rate.

RELATED FIGURES

Figure 1:

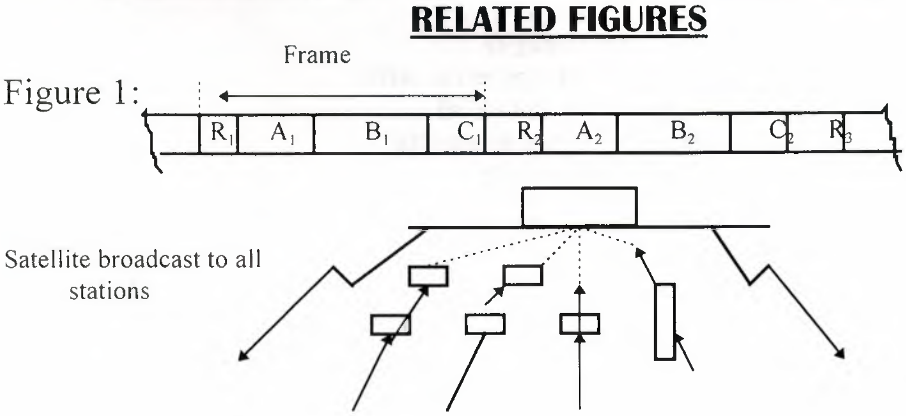


Figure 3:

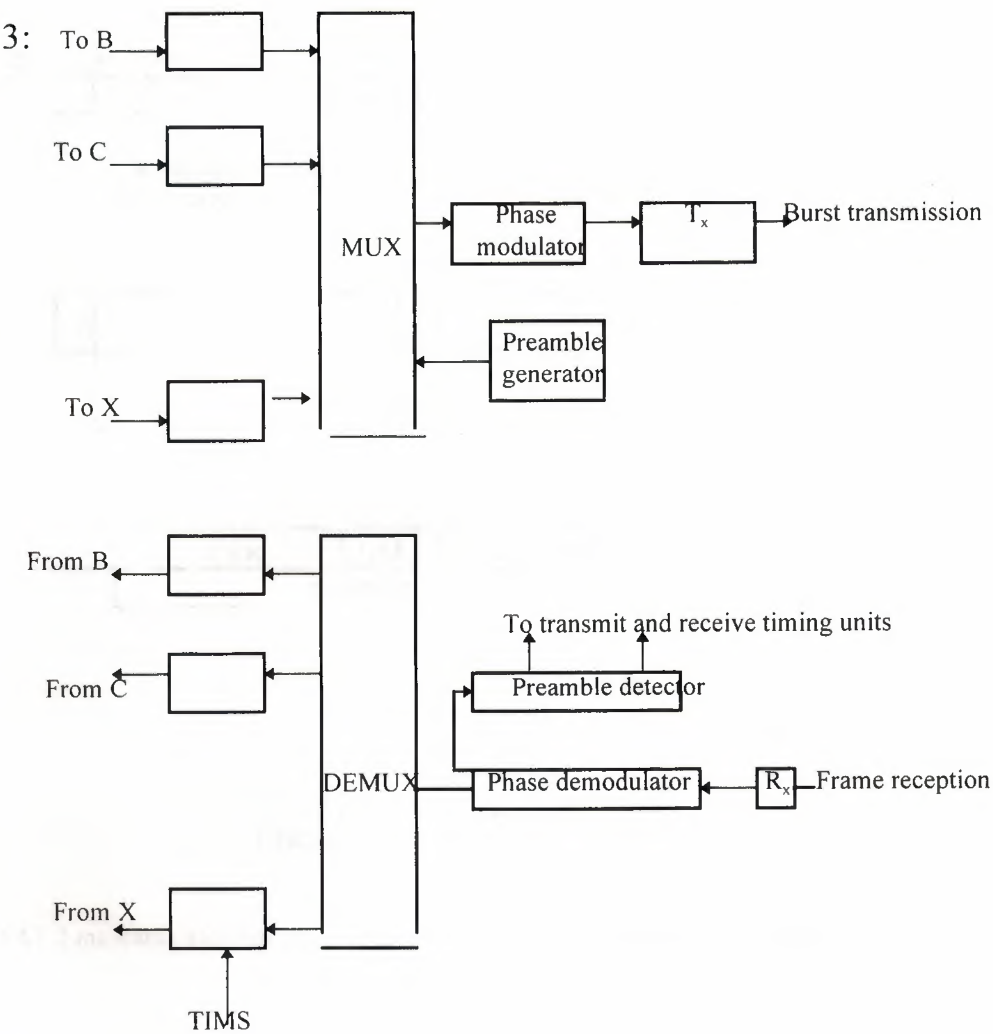
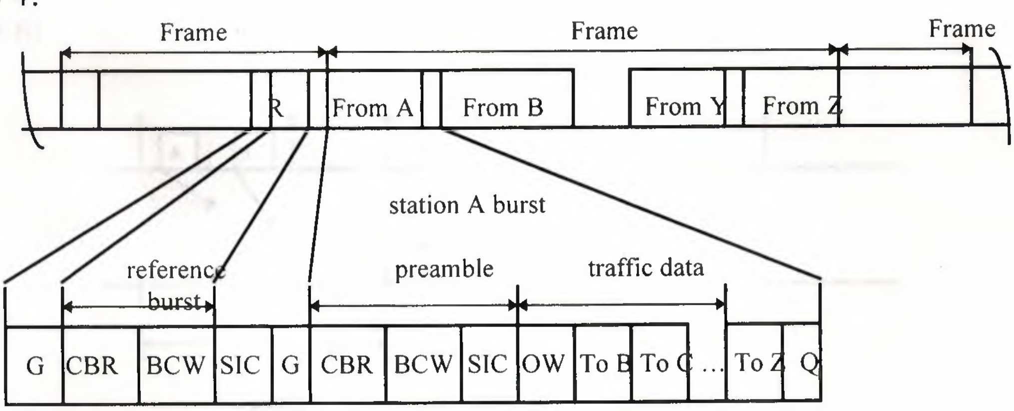
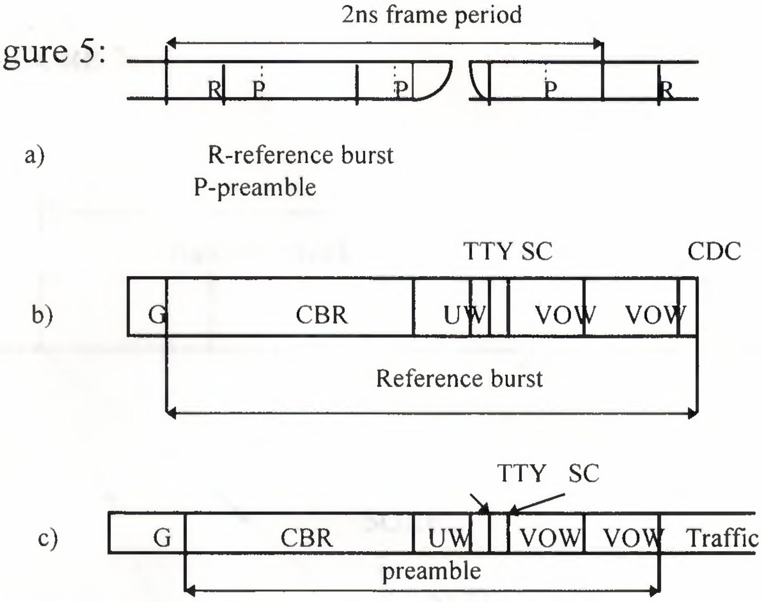


Figure 4:



Legend:
G-guard time
CBR-carrier and bit timing recovery
BCW-burst code word
SIC-station identification code
Q-postamble

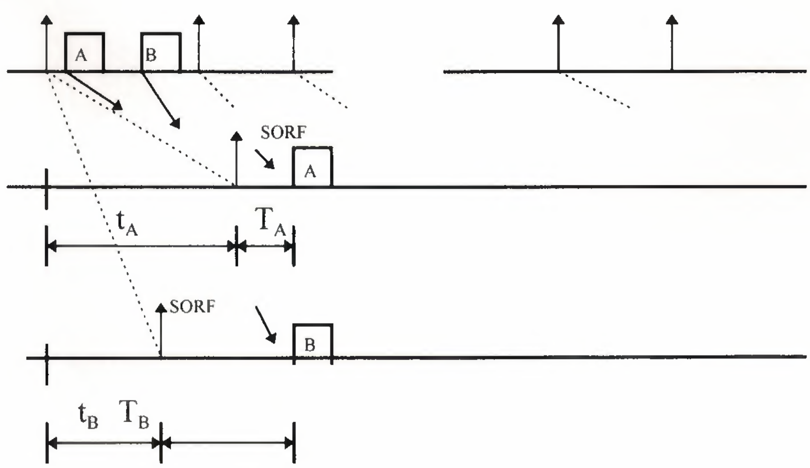
Figure 5:



G-guard time
CBR-carrier and bit-timing recovery
UW-unique word
TTY-teletype
SC-service channel
VOW-voice order wire
CDC-coordination and delay channel

a)INTELSAT 2 ms frame b)composition of the reference burst R c)composition of the preamble P

Figure 6:



Start of receive frame (SORF) marker in a time burst plane.

FIGURE 7: