

UNIT IV

SATELLITE ACCESS AND CODING METHODS

Modulation and Multiplexing: Voice, Data, Video, and Analog – Digital Transmission System, Digital Video Broadcast, Multiple Access: FDMA, TDMA, CDMA, DAMA Assignment Methods, Compression – Encryption, Coding Schemes.

4.1 INTRODUCTION TO MODULATION AND MULTIPLEXING

Communications satellites are used to carry telephone, video, and data signals, and can use both analog and digital modulation techniques.

Modulation:

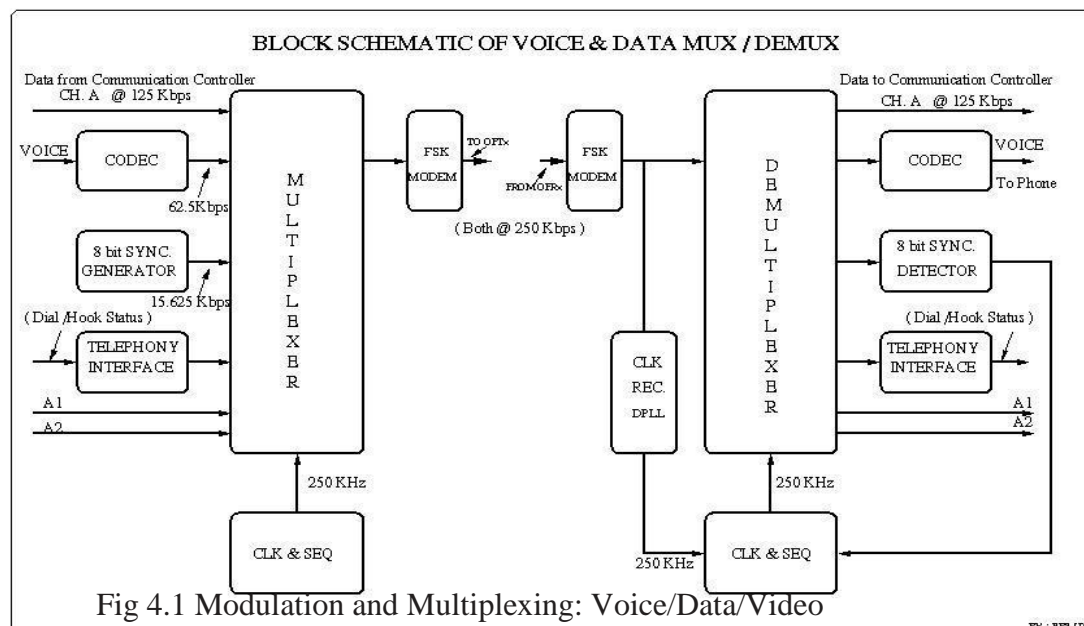
Modification of a carrier's parameters (amplitude, frequency, phase, or a combination of them) in dependence on the symbol to be sent.

Multiplexing:

Task of multiplexing is to assign space, time, frequency, and code to each communication channel with a minimum of interference and a maximum of medium utilization. Communication channel refers to an association of sender(s) and receiver(s) that want to exchange data one of several constellations of a carrier's parameters defined by the used modulation scheme.

Voice, Data, Video:

The modulation and multiplexing techniques that were used at this time were analog, adapted from the technology developed for the change to digital voice signals made it easier for long-distance.



Communication carriers to mix digital data and telephone Fiber-optic Cable Transmission Standards System Bit rate (Mbps) 64 - kbps Voice channel capacity Stuffing bits and words are added to the satellite data stream as needed to fill empty bit and word spaces.

Primarily for video provided that a satellite link's overall carrier-to-noise but in to older receiving equipment at System and Satellite Specification Ku-band satellite parameters.

Modulation and Multiplexing:

In analog television (TV) transmission by satellite, the baseband video signal and one or two audio subcarriers constitute a composite video signal.

Digital modulation is obviously the modulation of choice for transmitting digital data. Digitized analog signals may conveniently share a channel with digital data, allowing a link to carry a varying mix of voice and data traffic.

Digital signals from different channels are interleaved for transmission through time division multiplexing (TDM). TDM can carry any type of traffic—the bent pipe transponder that can carry voice, video, or data as the marketplace demands.

Hybrid multiple access schemes can use time division multiplexing of baseband channels which are then modulated.

4.1.1 ANALOG – DIGITAL TRANSMISSION SYSTEM:

Analog vs. Digital Transmission:

Compare at two levels:

1. Data—continuous (audio) vs. discrete (text)
2. Signaling—continuously varying electromagnetic wave vs. sequence of voltage pulses.

Also Transmission—transmits without regard to signal content vs. being concerned with signal content. Difference in how attenuation is handled, but not focus on this.

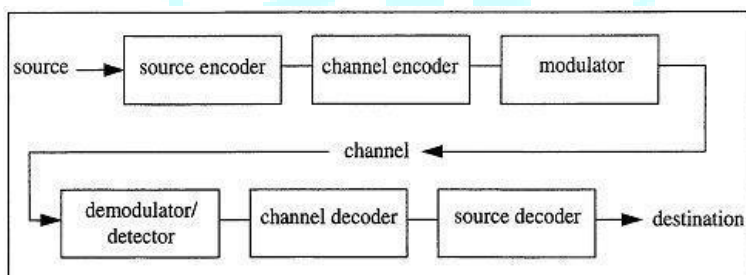


Fig 4.2 Basic Communication Systems

Seeing a shift towards digital transmission despite large analog base. Because of

- Improving digital technology
- Data integrity. Repeaters take out cumulative problems in transmission. Can thus transmit longer distances.
- Easier to multiplex large channel capacities with digital
- Easy to apply encryption to digital data
- Better integration if all signals are in one form. Can integrate voice, video and digital data.

4.1.2 DIGITAL DATA/ANALOG SIGNALS:

Must convert digital data to analog signal such device is a modem to translate between bit-serial and modulated carrier signals?

- To send digital data using analog technology, the sender generates a carrier signal at some continuous tone (e.g. 1-2 kHz in phone circuits) that looks like a sine wave. The following techniques are used to encode digital data into analog signals.

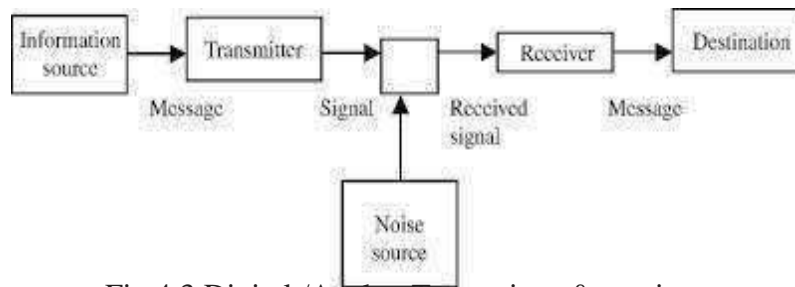


Fig 4.3 Digital /Analog Transmitter & receiver

Resulting bandwidth is centered on the carrier frequency.

- Amplitude-shift modulation (keying): vary the amplitude (e.g. voltage) of the signal. Used to transmit digital data over optical fiber.
- Frequency-shift modulation: two (or more tones) are used, which are near the carrier frequency. Used in a full-duplex modem (signals in both directions).
- Phase-shift modulation: systematically shift the carrier wave at uniformly spaced intervals.

For instance, the wave could be shifted by 45, 135, 225, 315 degree at each timing mark. In this case, each timing interval carries 2 bits of information.

Why not shift by 0, 90, 180, 270? Shifting zero degrees means no shift, and an extended set of no shifts leads to clock synchronization difficulties.

Frequency division multiplexing (FDM): Divide the frequency spectrum into smaller sub channels, giving each user exclusive use of a sub channel (e.g., radio and TV). One problem with FDM is that a user is given all of the frequency to use, and if the user has no data to send, bandwidth is wasted — it cannot be used by another user.

Time division multiplexing (TDM): Use time slicing to give each user the full bandwidth, but for only a fraction of a second at a time (analogous to time sharing in operating systems). Again, if the user doesn't have data to sent during his time slice, the bandwidth is not used (e.g., wasted).

Statistical multiplexing: Allocate bandwidth to arriving packets on demand. This leads to the most efficient use of channel bandwidth because it only carries useful data. That is, channel bandwidth is allocated to packets that are waiting for transmission, and a user generating no packets doesn't use any of the channel resources.

Digital Video Broadcasting (DVB):

Digital Video Broadcasting (DVB) has become the synonym for digital television and for data broadcasting world-wide.

DVB services have recently been introduced in Europe, in North- and South America, in Asia, Africa and Australia.

This article aims at describing what DVB is all about and at introducing some of the technical background of a technology that makes possible the broadcasting.

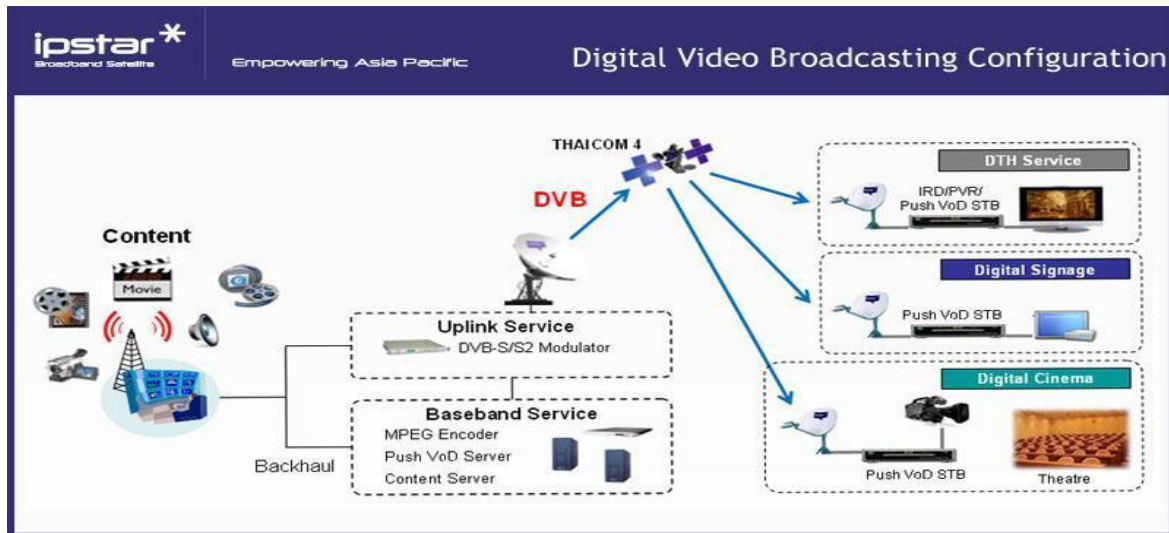


Fig 4.4 Digital Video Broadcasting systems

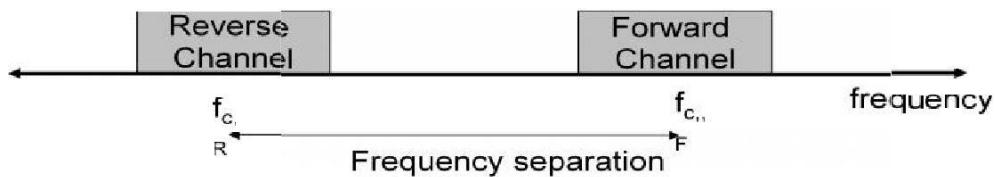
4.2 DUPLEXING

For voice or data communications, must assure two way communication (duplexing, it is possible to talk and listen simultaneously). Duplexing may be done using frequency or time domain techniques.

- Forward (downlink) band provides traffic from the BS to the mobile
- Reverse (uplink) band provides traffic from the mobile to the BS.

FREQUENCY DIVISION DUPLEXING (FDD):

Provides two distinct bands of frequencies for every user, one for downlink and one for uplink. A large interval between these frequency bands must be allowed so that interference is minimized.



Frequency separation should be carefully decided
Frequency separation is constant

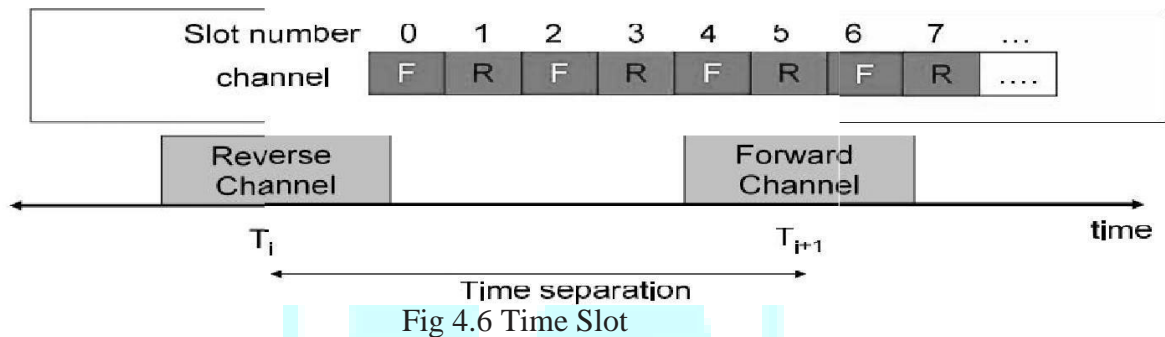
Fig 4.5 Frequency Separation

TIME DIVISION DUPLEXING (TDD):

In TDD communications, both directions of transmission use one contiguous frequency allocation, but two separate time slots to provide both a forward and reverse link.

Because transmission from mobile to BS and from BS to mobile alternates in time, this scheme is also known as “ping pong”.

As a consequence of the use of the same frequency band, the communication quality in both directions is the same. This is different from FDD.



4.3 SINGLE ACCESS

With single access, a single modulated carrier occupies the whole of the available bandwidth of a transponder. Single-access operation is used on heavy-traffic routes and requires large earth station antennas.

The earth station employs a 30-m-diameter antenna and a parametric amplifier, which together provide a minimum $[G/T]$ of 37.5 dB/K.

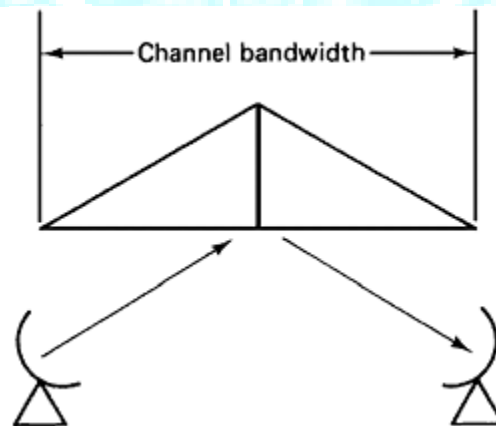


Fig 4.7 Frequency Modulation Single Access.

4.4 MULTIPLE ACCESS TECHNIQUES

With the increase of channel demands and the number of earth stations, efficient use of a satellite transponder in conjunction with many stations has resulted in the development of multiple access techniques. Multiple access is a technique in which the satellite resource (bandwidth or time) is divided into a number of nonoverlapping segments and each segment is allocated exclusively to each of the large number of earth stations who seek to communicate with each other.

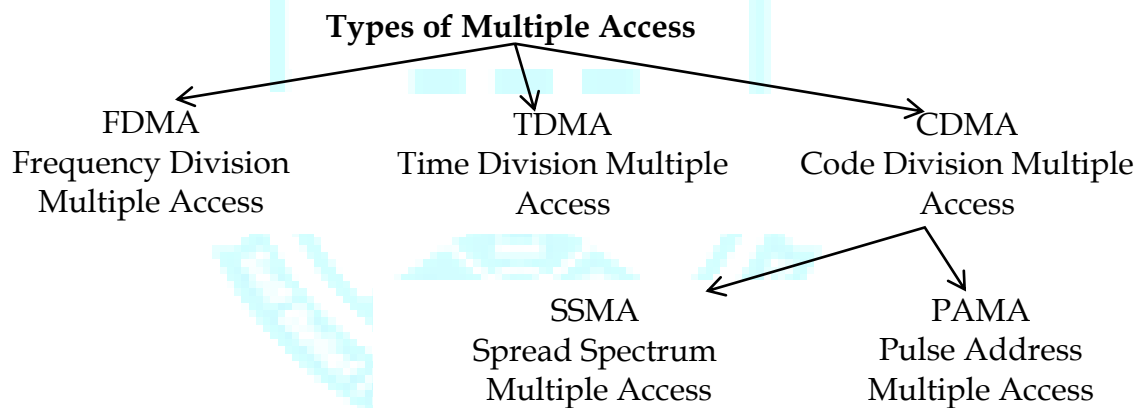
There are three known multiple access techniques. They are:

- Frequency Division Multiple Access (FDMA)
- Time Division Multiple Access (TDMA)
- Code Division Multiple Access (CDMA)

The transmission from the BS in the downlink can be heard by each and every mobile user in the cell, and is referred as broadcasting. Transmission from the mobile users in the uplink to the BS is many-to-one, and is referred to as multiple access.

Multiple access schemes to allow many users to share simultaneously a finite amount of radio spectrum resources. Should not result in severe degradation in the performance of the system as compared to a single user scenario. Approaches can be broadly grouped into two categories: narrowband and wideband.

Multiple Accessing Techniques : with possible conflict and conflict-free



Classification of Multiple Access based on the way in which circuits are assigned to the users

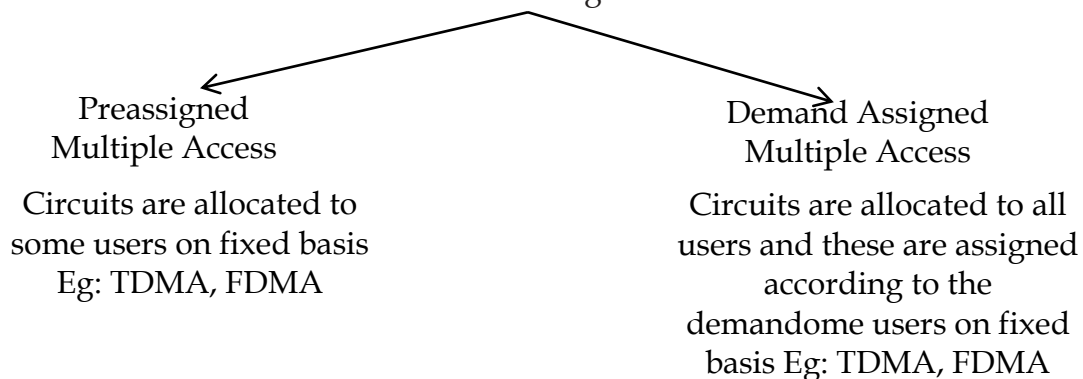


Fig 4.8 Types of Multiple Access

4.4.1 FREQUENCY DIVISION MULTIPLE ACCESS: (FDMA)

The most widely used of the multiple access techniques is FDMA. In FDMA, the available satellite bandwidth is divided into portions of non-overlapping frequency slots which are assigned exclusively to individual earth stations. A basic diagram of an FDMA satellite system is shown in Fig.

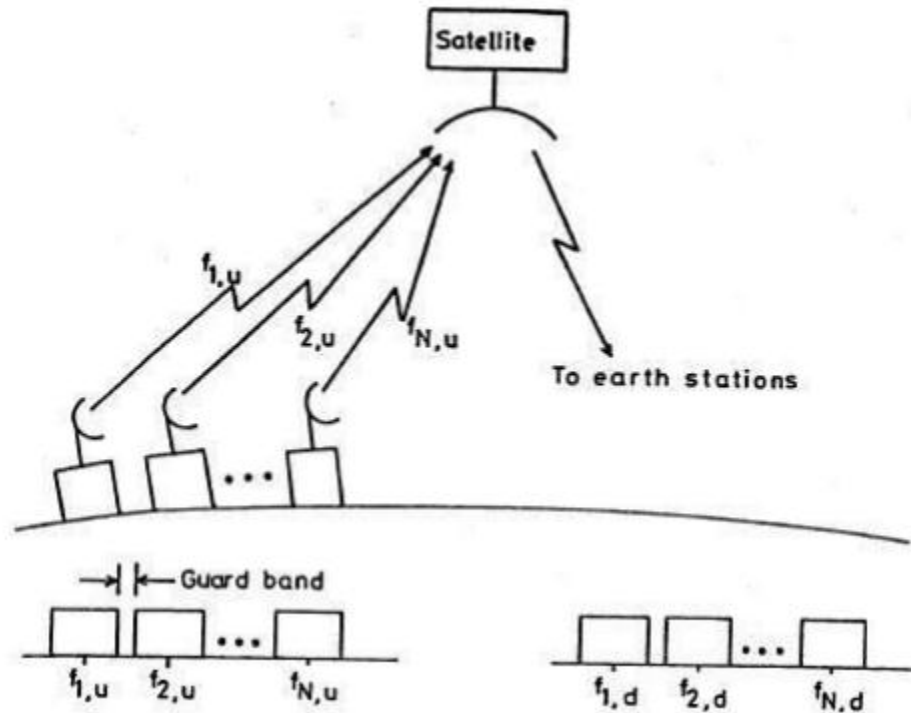


Fig 4.9 FDMA Satellite System

In FDMA, each user is allocated a unique frequency band or channel. During the period of the call, no other user can share the same frequency band.

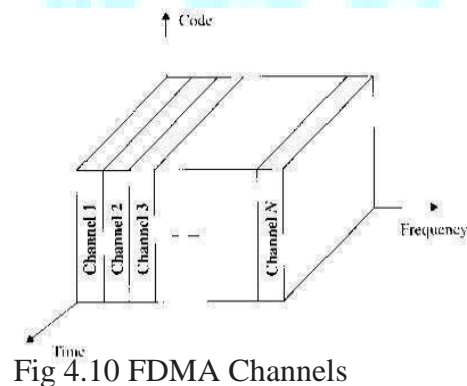


Fig 4.10 FDMA Channels

- All channels in a cell are available to all the mobiles. Channel assignment is carried out on a first-come first- served basis.
- The number of channels, given a frequency spectrum BT, depends on the modulation technique (hence B_w or B_c) and the guard bands between the channels guard.

- These guard bands allow for imperfect filters and oscillators and can be used to minimize adjacent channel interference.
- FDMA is usually implemented in narrowband systems.

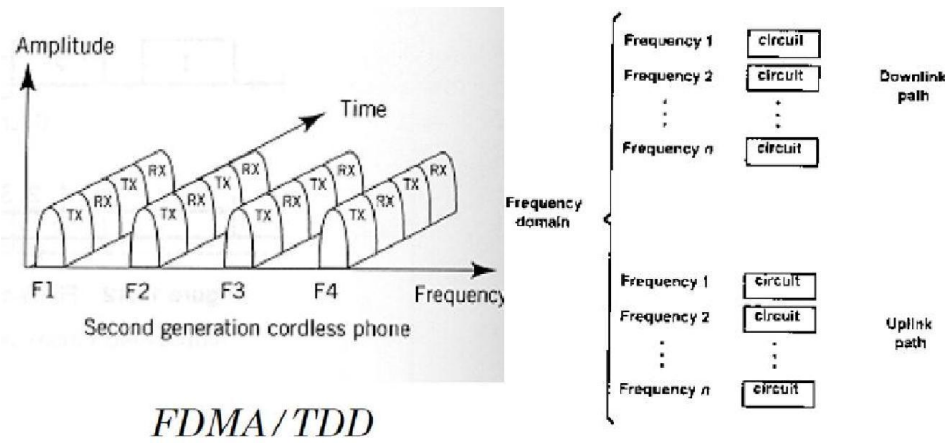


Fig 4.11 FDMA/FDD/TDD

Examples of this technique are

1. FDM/FM/FDMA used in INTELSAT II & III and SCPC satellite systems.
2. SPACE (signal-channel-per-carrier PCM multiple access demand assignment equipment) used in INTELSAT IV in which channels are assigned on demand to earth stations is considered as a FDMA system.

In FDMA systems, multiple signals from the same or different earth stations with different carrier frequencies are simultaneously passed through a satellite transponder. Because of the nonlinear mode of the transponder, FDMA signals interact with each other causing intermodulation products (intermodulation noise) which are signals at all combinations of sum and difference frequencies as shown in the example given in Fig.

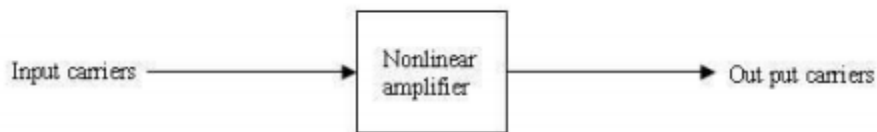


Fig 4.12 FDMA Non Linear Amplifier

The power of these intermodulation products represents a loss in the desired signal power. In addition, if these intermodulation products appear within the bandwidth of the other signals, they act as interference for these signals and as a result the BER performances will be degraded. The other major disadvantage of the FDMA system is the need for accurate uplink power control among network stations in order to mitigate the weak signal suppression effect caused by the disproportionate power sharing of the transponder power.

Nonlinear effects in FDMA:

- In a FDMA system, many channels share the same antenna at the BS. The power amplifiers or the power combiners, when operated at or near saturation are nonlinear.
- The nonlinear ties generate inter-modulation frequencies.
- Undesirable harmonics generated outside the mobile radio band cause interference to adjacent services.
- Undesirable harmonics present inside the band cause interference to other users in the mobile system.

Forms of DMA:

1. Fixed-assignment multiple access (FAMA)
 - The assignment of capacity is distributed in a fixed manner among multiple stations
 - Demand may fluctuate
 - Results in the significant underuse of capacity
2. Demand-assignment multiple access (DAMA)
 - Capacity assignment is changed as needed to respond optimally to demand changes among the multiple stations

Pre Assigned FDMA:

Frequency slots may be preassigned to analog and digital signals, and to illustrate the method, analog signals in the FDM/FM/FDMA format will be considered first. As the acronyms indicate, the signals are frequency-division multiplexed, frequency modulated (FM), with FDMA to the satellite.

For Ex, Consider each earth station will be assumed to transmit a 60-channel supergroup. Each 60-channel supergroup is then frequency modulated onto a carrier which is then upconverted to a frequency in the satellite uplink band.

Figure shows the situation for three earth stations:

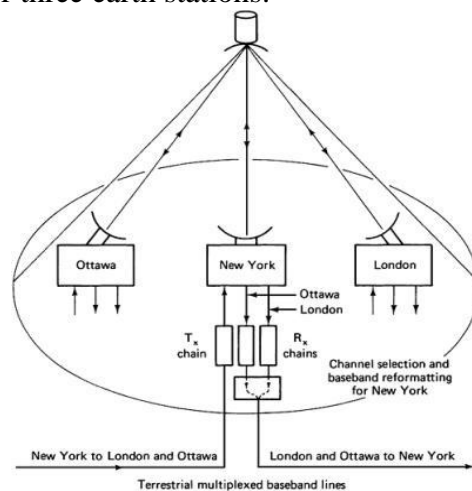


Fig 4.13 Preassigned FDMA System

one in Ottawa, one in New York, and one in London. All three earth stations access a single satellite transponder channel simultaneously, and each communicates with both of the others. Thus it is assumed that the satellite receive and transmit antenna beams are global, encompassing all three earth stations. Each earth station transmits one uplink carrier modulated with a 60-channel supergroup and receives two similar downlink carriers.

Demand Assigned FDMA:

In the demand-assigned mode of operation, the transponder frequency bandwidth is subdivided into a number of channels. A channel is assigned to each carrier in use, giving rise to the single-channel-per-carrier mode of operation discussed in the preceding section.

As in the preassigned access mode, carriers may be frequency modulated with analog information signals, these being designated FM/SCPC, or they may be phase modulated with digital information signals, these being designated as PSK/SCPC.

Demand assignment may be carried out in a number of ways.

1. In the polling method, a master earth station continuously polls all the earth stations in sequence, and if a call request is encountered, frequency slots are assigned from the pool of available frequencies. The polling delay with such a system tends to become excessive as the number of participating earth stations increases.
2. Instead of using a polling sequence, earth stations may request calls through the master earth station as the need arises. This is referred to as **centrally controlled random access**. The requests go over a digital orderwire, which is a narrowband digital radio link or a circuit through a satellite transponder reserved for this purpose. Frequencies are assigned, if available, by the master station, and when the call is completed, the frequencies are returned to the pool. If no frequencies are available, the blocked call requests may be placed in a queue, or a second call attempt may be initiated by the requesting station.
3. As an alternative to centrally controlled is **Distributed Control random Access** (Eg. **SPADE** system operated by INTELSAT on some of its satellite)

SPADE System:

- The word Spade is a loose acronym for **SCPC pulse-code-modulated multiple-access demand-assignment** equipment.
- Spade was developed by Comsat for use on the INTELSAT satellites (see, e.g., Martin, 1978) and is compatible with the INTELSAT SCPC preassigned system.
- However, the distributed-demand assignment facility requires a **common signaling channel (CSC)**. This is shown in Fig.

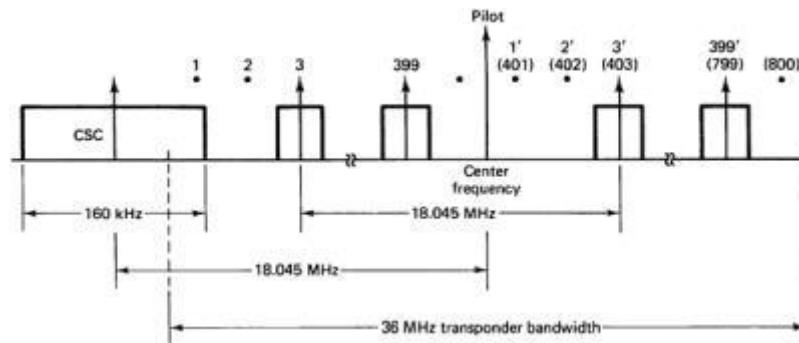


Fig 4.14 Channeling scheme for the Spade system.

- The CSC bandwidth is 160 kHz, and its center frequency is 18.045 MHz below the pilot frequency, as shown in Fig..
- Signaling information is routed through the CSC. Each earth station has DASS unit (Demand Assignment Signaling and Switching unit). It is used to perform the functions needed by the CSC.
- To avoid interference with the CSC, voice channels 1 and 2 are left vacant, and to maintain duplex matching, the corresponding channels 1 and 2 are also left vacant.
- Recalling from previous section that channel 400 also must be left vacant, this requires that channel 800 be left vacant for duplex matching.
- Thus six channels are removed from the total of 800, leaving a total of 794 one-way or 397 full-duplex voice circuits, the frequencies in any pair being separated by 18.045 MHz, as shown in Fig. . (An alternative arrangement is shown in Freeman, 1981.)

Channeling Scheme for Spade System:

- All the earth stations are permanently connected through the CSC.
- This is shown diagrammatically in Fig. for six earth stations A, B, C, D, E, and F.
- Each earth station has the facility for generating any one of the 794 carrier frequencies using frequency synthesizers.
- Furthermore, each earth station has a memory containing a list of frequencies available and the list is continuously updated through CSC.

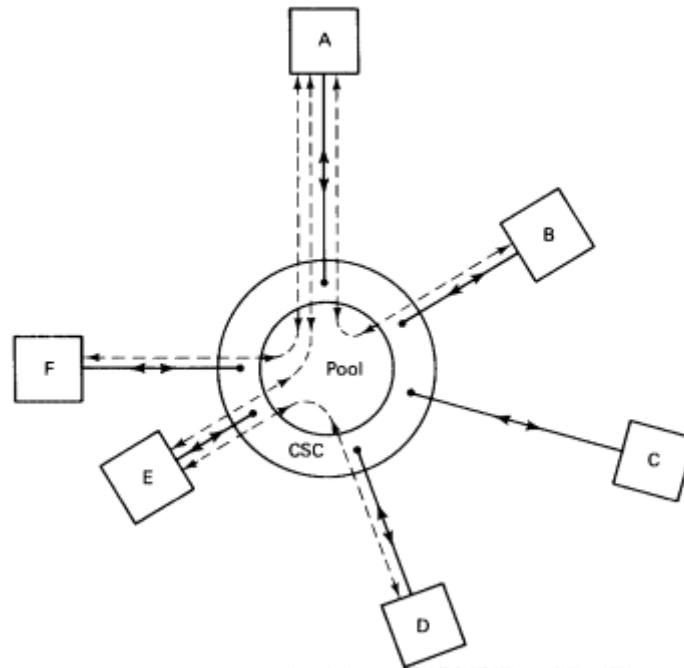


Fig 4.15 Diagrammatic representation of SPADE system

- Suppose that a call to station F is initiated from station C.
- Station C will first select a frequency pair at random from those currently available on the list, and signal this information to station F through the CSC.
- Station F must acknowledge through the CSC, that it can complete the circuit.
- Once the circuit is established, the other stations are instructed, through the CSC, to remove this frequency pair from the list.
- The round trip time between station C initiating the call and the station F acknowledging it is about 600 ms.
- During this time, the two frequencies chosen at station C may be assigned to another circuit.
- In this event, station C will receive the information on the CSC update and will immediately choose another pair at random even before hearing back from station F.
- Once a call has been completed and the circuit disconnected, the two frequencies are returned to the pool, the information again being transmitted through the CSC to all earth stations.
- Each earth station has equipment called the demand assignment signalling and switching unit which performs the function required by the CSC.
- The channel pair is automatically located to an incoming call by a DASS unit. Each SPADE terminal maintains a complete assignment map for the entire system.
- Before a station selects a channel, it first analysis the assignment map, chooses an unused channel and indicates its choices on the DASS network.
- The associated return channel completes the circuit.

4.4.2 TIME DIVISION MULTIPLE ACCESS: (TDMA)

- TDMA systems divide the channel time into frames. Each frame is further partitioned into time slots. In each slot only one user is allowed to either transmit or receive.
- Unlike FDMA, only digital data and digital modulation must be used.
- Each user occupies a cyclically repeating time slot, so a channel may be thought of as a particular time slot of every frame, where N time slots comprise a frame.

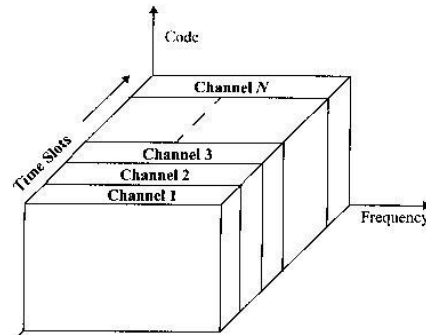


Fig 4.16 TDMA Channels

- With TDMA, 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 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 illustrates the basic TDMA concept, in which the stations transmit bursts in sequence.
- Burst synchronization is required, and in the system illustrated in Fig. 14.10, one station is assigned solely 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 termed a frame. A frame contains the reference burst R and the bursts from the other earth stations, these being shown as A , B , and C in Fig. 14.10.

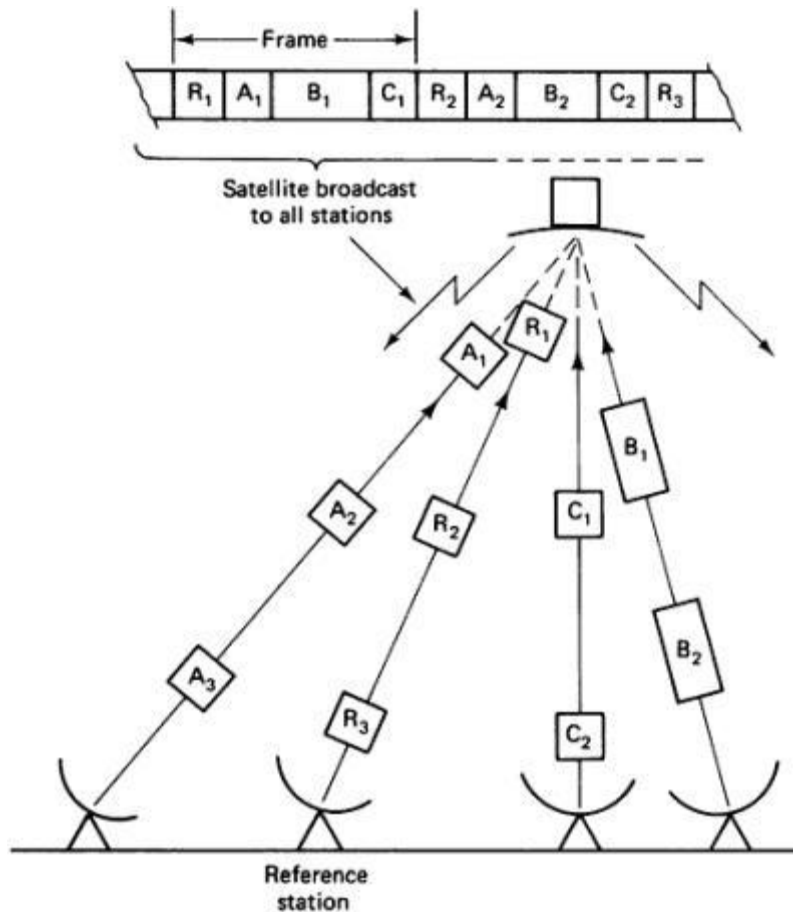


Fig 4.17 Time-division multiple access (TDMA) using a reference station for burst synchronization.

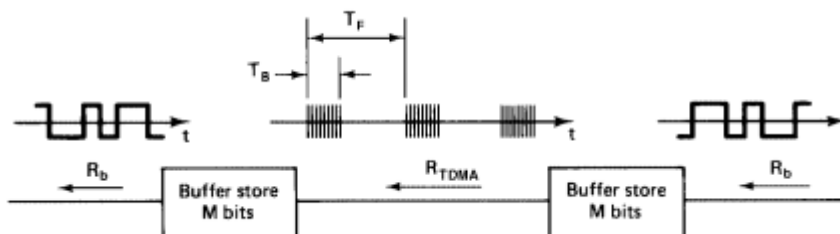
- Figure illustrates the basic principles of burst transmission for a single channel.
- Overall, the transmission appears continuous because the input and output bit rates are continuous and equal.
- However, within the transmission channel, input bits are temporarily stored and transmitted in bursts.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

$$M = R_b T_F$$

The M bits are transmitted in the burst time T_B , and the transmission rate (Burst Bit Rate) is

$$R_{TDMA} = M/T_b = M (T_F/T_b)$$

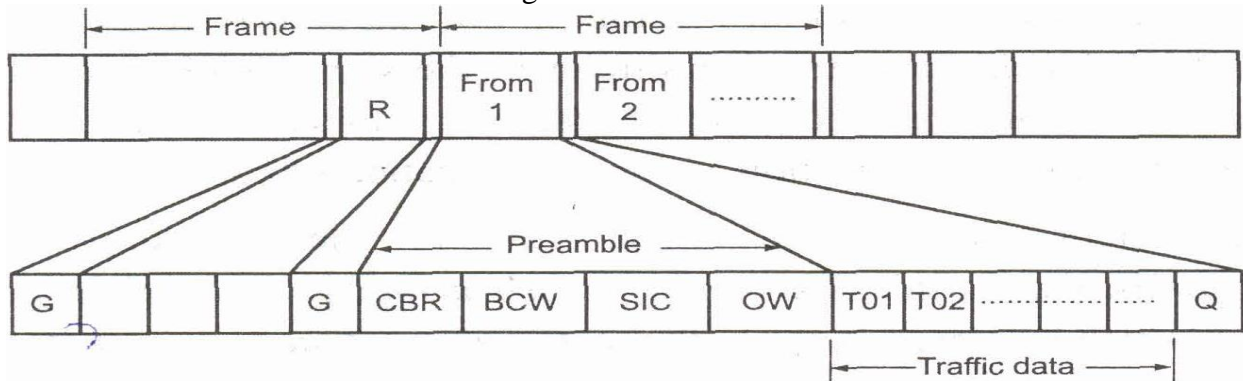
It is also referred to as Burst Rate.



4.18 TDMA Burst Bit Rate

TDMA – Transmission

- In TDMA, the transmit timing of the bursts is accurately synchronized so that the transponder receives one burst at a time.
- Each earth station receives an entire burst stream and extracts the bursts intended for it.
- A frame consists of a number of bursts originating from a community of earth stations in a network.
- A TDMA frame structure is shown in Fig.



Where,

G	- Guard Time
CBR	- Carrier and Bit Timing Recovery
BCW	- Burst Code Word
SIC	- Station Identification Code
Q	- Postamble

Fig 4.19 TDMA Frame Format

- It consists of two reference bursts RB1 and RB2, traffic bursts and the guard time between bursts.
- As can be seen, each TDMA frame has two reference bursts RB1 and RB2.
- The primary reference burst (PRB), which can be either RB1 or RB2, is transmitted by one of the earth stations in the network designated as the primary reference earth station.
- For reliability, a second reference burst (SRB) is transmitted by a secondary reference earth station.
- To ensure an uninterrupted service for the TDMA network, automatic switchover between these two reference stations is provided.
- The reference bursts carry no traffic information and are used to provide synchronization for all earth stations in the network.
- The traffic bursts carry information from the traffic earth station.
- Each earth station accessing a transponder may transmit one or two traffic bursts per TDMA frame and may position them anywhere in the frame according to a burst time plan that coordinates traffic between earth stations in the network.
- The Guard time between bursts ensures that the bursts never overlap at the input to the transponder.
- The TDMA bursts structure of the reference and traffic burst are given in Fig

Various sequences in the reference burst and traffic burst are as follows:

1. Carrier and bit timing recovery (CBTR)

The CBTR pattern provides information for carrier and timing recovery circuits of the earth station demodulator. The length of the CBTR sequence depends on the carrier-to-noise ratio at the input of the demodulator and the acquisition range. For example, the 120 Mb/s TDMA system of INTELSAT V has a 48 symbol pattern for carrier recovery and a 128 symbol pattern for bit timing recovery.

2. Unique word (UW)

The unique word sequence in the reference burst provides the receive frame timing that allows an earth station to locate the position of a traffic burst in the frame. The UW in the traffic burst marks the beginning of the traffic burst and provides information to an earth station so that it selects only those traffic bursts intended for it. The UW is a sequence of ones and zeros selected to exhibit good correlation properties to enhance detection. The UW of the INTELSAT V TDMA system has a length of 24 symbols.

3. Teletype (TTY) and voice order wire (VOW)

Teletype and voice order wire patterns carry instructions to and from earth stations. The number of symbols for each of the patterns is 8 symbols for the INTELSAT V TDMA.

4. Service channel (SC)

The service channel of the reference burst carries management instructions such as burst time plan which gives the coordination of traffic between earth stations, i.e. position, length, and source and destination earth stations corresponding to traffic bursts in the TDMA frame.

The channel also carries monitoring and control information to the traffic stations. The SC of the traffic burst carries the traffic station's status to the reference station (value of transmit delay used and reference station from which the delay is obtained). It also contains other information such as the high bit error rate and UW loss alarms to other traffic stations. The INTELSAT V TDMA has an 8-symbol SC for each of the bursts.

Carrier and bit timing recovery	UW	TTY	Service Channel	VOW	Control and delay channel
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a) Reference Burst

Carrier and bit timing recovery	UW	TTY	Service Channel	VOW	Traffic Data
← Preamble →					

b) Traffic Burst

Fig 4.20 Reference Burst and Traffic Burst

5. Control and delay channel (CDC)

The control and delay channel pattern carries acquisition and synchronization information to the traffic earth stations to enable them to adjust their transmit delays so that bursts arrive at the satellite transponder within the correct time slots in the frame. It also carries the reference station status code which enables them to identify the primary and secondary reference bursts. Eight symbols are allocated for this channel in the INTELSAT V TDMA.

6. Traffic data

This portion contains the information from a source traffic station to a destination traffic station. The informants can be voice, data, video or facsimile signals. The traffic data pattern is divided into blocks of data (referred to as subburst).

The size of each data block is given by:

Subburst size (symbols) = symbol rate (symbols/sec) X frame length (sec).

The INTELSAT TDMA with a frame length of $T_f = 2$ msec for PCM voice data has a subburst size of 64 symbols long.

TDMA- Features:

- Burst transmission since channels are used on a time sharing basis. Transmitter can be turned off during idle periods.
- Multiple channels per carrier or RF channels.
- Narrow or wide bandwidth – depends on factors such as modulation scheme, number of voice channels per carrier channel.
- High ISI – Higher transmission symbol rate, hence resulting in high ISI. Adaptive equalizer required.
- A guard time between the two time slots must be allowed in order to avoid interference, especially in the uplink direction. All mobiles should synchronize with BS to minimize interference.
- Efficient power utilization: FDMA systems require a 3- to 6-dB power back off in order to compensate for inter-modulation effects.
- Efficient handoff: TDMA systems can take advantage of the fact that the transmitter is switched off during idle time slots to improve the handoff procedure. An enhanced link control, such as that provided by mobile assisted handoff (MAHO) can be carried out by a subscriber by listening to neighboring base station during the idle slot of the TDMA frame.
- Efficiency of TDMA
- Efficiency of TDMA is a measure of the percentage of bits per frame which contain transmitted data. The transmitted data include source and channel coding bits.

$$\eta_f = \frac{b_T - b_{OH}}{b_T} * 100\%$$

b_{OH} includes all overhead bits such as preamble, guard bits, etc.

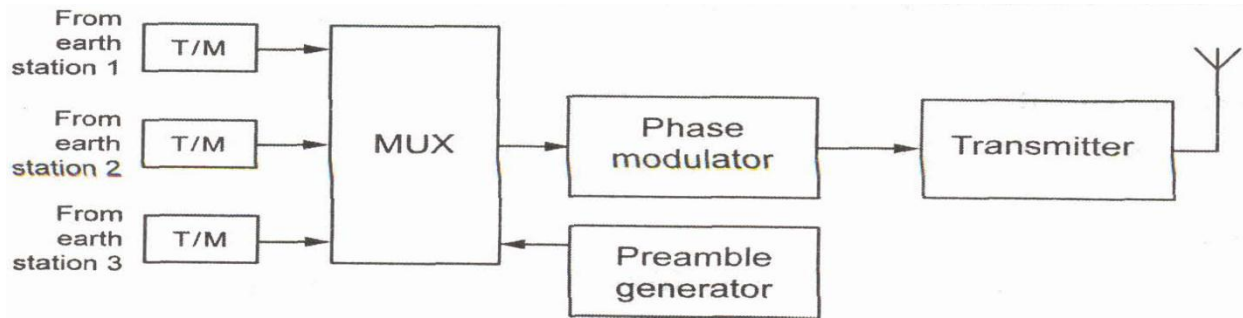


Fig 4.21 Block Diagram- TDMA - transmitter

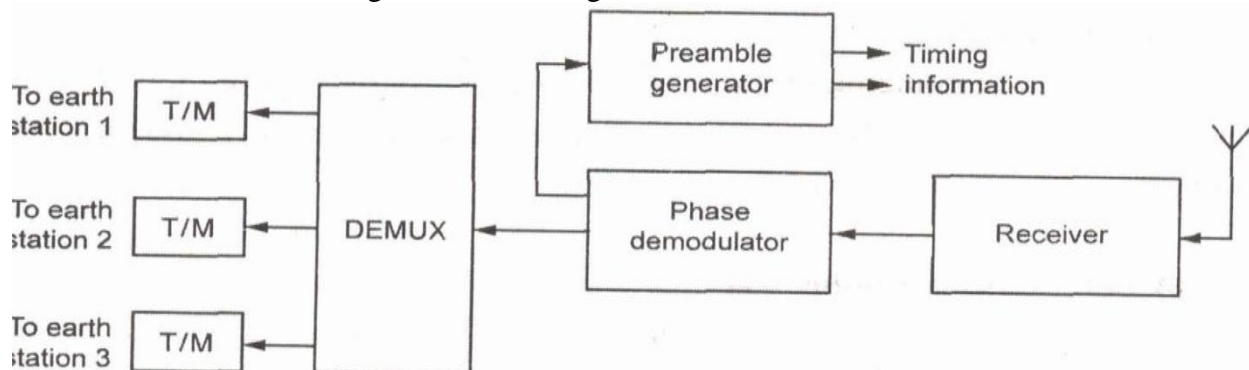


Fig 4.22 Block Diagram- TDMA – Receiver

4.4.3 CODE DIVISION MULTIPLE ACCESS (CDMA):

CDMA Features:

- Spreading signal (code) consists of chips
- Has Chip period and hence, chip rate
- Spreading signal use a pseudo-noise (PN) sequence (a pseudo-random sequence)
- PN sequence is called a code word
- Each user has its own cord word
- Code words are orthogonal. (low autocorrelation)
- Chip rate is order of magnitude larger than the symbol rate.
- The receiver correlator distinguishes the senders signal by examining the wideband signal with the same time-synchronized spreading code
- The sent signal is recovered by dispreading process at the receiver.

CDMA Advantages:

- Low power spectral density.
Signal is spread over a larger frequency band
Other systems suffer less from the transmitter

- Interference limited operation
All frequency spectrum is used
- Privacy
The code word is known only between the sender and receiver. Hence other users cannot decode the messages that are in transit
- Reduction of multipath affects by using a larger spectrum

CDMA Data:

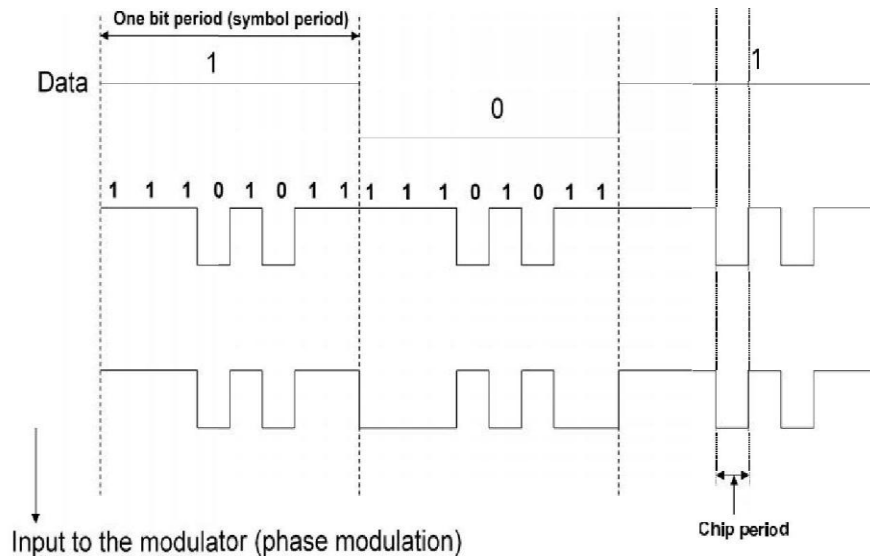
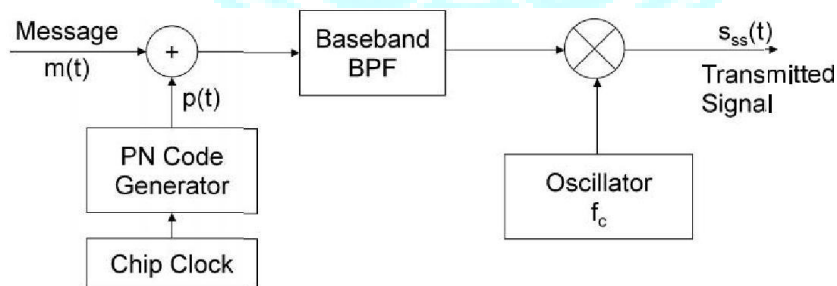


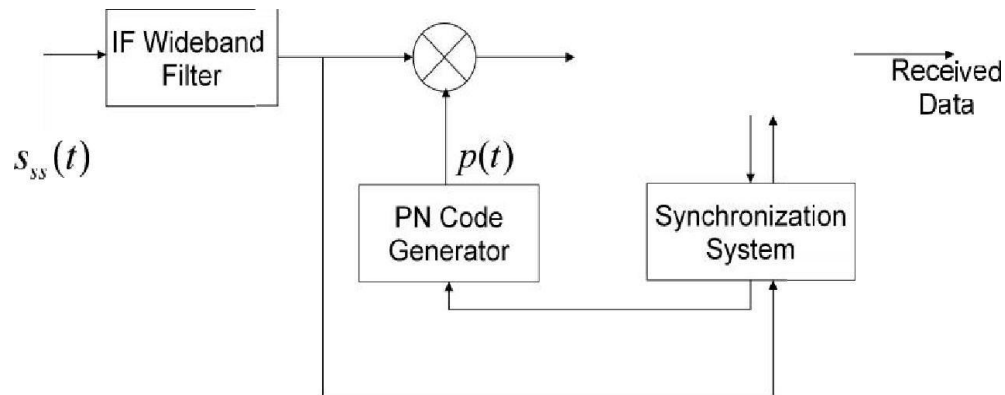
Fig 4.23 CDMA Channels transmission

DSSS Transmitter:



$$s_{ss}(t) = \sqrt{\frac{2E_s}{T_s}} m(t)p(t) \cos(2\pi f_c t + \theta)$$

Fig 4.24 CDMA Transmitter

DSSS Receiver

$$s_1(t) = \sqrt{\frac{2E_s}{T_s}} m(t) \cos(2\pi f_c t + \theta)$$

Fig 4.25 CDMA Receiver

FDMA/CDMA

Available wideband spectrum is frequency divided into number narrowband radio channels. CDMA is employed inside each channel.

DS/FHMA

The signals are spread using spreading codes (direct sequence signals are obtained), but these signal are not transmitted over a constant carrier frequency; they are transmitted over a frequency hopping carrier frequency.

Time Division CDMA (TCDMA)

Each cell is using a different spreading code (C DMA employed between cellss) that is conveyed to the mobiles in its range. Inside each cell (inside a CDMA channel), TDMA is employed to multiplex multiple users.

Time Division Frequency Hopping

- At each time slot, the user is hopped to a new frequency according to a pseudo-random hopping sequence.
- Employed in severe co-interference and multi-path environments. Bluetooth and GSM are using this technique
- A large number of independently steered high-gain beams can be formed without any resulting degradation in SNR ratio.
- Beams can be assigned to individual users, thereby assuring that all links operate with maximum gain.
- Adaptive beam forming can be easily implemented to improve the system capacity by suppressing co channel interference.

Advantage of CDMA

- It is recognized that CDMA's capacity gains over TDMA
- FDMA are entirely due to Its tighter, dynamic control over the use of the power domain.
- Choosing a new non-orthogonal PN sequence a CDMA system does not encounter the difficulties of choosing a spare carrier frequency or time slot to carry a Traffic Channel
- Ensure that interference will not be too great if it begins to transmit -that there is still enough space left in the power domain.

Disadvantages of CDMA:

- Satellite transponders are channelized too narrowly for roadband CDMA, which is the most attractive form of CDMA.
- Power control cannot be as tight as it is in a terrestrial system because of long round-trip delay.

Channel Allocation Schemes:

In radio resource management for wireless and cellular network, channel allocation schemes are required to allocate bandwidth and communication channels to base stations, access points and terminal equipment.

The objective is to achieve maximum system spectral efficiency in bit/s/Hz/site by means of frequency reuse, but still assure a certain grade of service by avoiding co-channel interference and adjacent channel interference among nearby cells or networks that share the bandwidth. There are two types of strategies that are followed:-

Fixed: FCA, fixed channel allocation: Manually assigned by the network operator

Dynamic:

DCA, dynamic channel allocation,
DFS, dynamic frequency selection
Spread spectrum

FCA:

In **Fixed Channel Allocation** or **Fixed Channel Assignment** (FCA) each cell is given a predetermined set of frequency channels.

FCA requires manual frequency planning, which is an arduous task in TDMA and FDMA based systems, since such systems are highly sensitive to co-channel interference from nearby cells that are reusing the same channel. This result in traffic congestion and some calls being lost when traffic gets heavy in some cells, and idle capacity in other cells.

DCA and DFS:

Dynamic Frequency Selection (DFS) may be applied in wireless networks with several adjacent non-centrally controlled access points.

A more efficient way of channel allocation would be **Dynamic Channel Allocation** or **Dynamic Channel Assignment** (DCA) in which voice channels are not allocated to cells permanently, instead for every call request base station requests a channel from MSC.

Spread Spectrum:

Spread spectrum can be considered as an alternative to complex DCA algorithms. Spread spectrum avoids co-channel interference between adjacent cells, since the probability that users in nearby cells use the same spreading code is insignificant.

Thus the frequency channel allocation problem is relaxed in cellular networks based on a combination of spread spectrum and FDMA, for example IS95 and 3G systems.

In packet-based data communication services, the communication is bursty and the traffic load rapidly changing. For high system spectrum efficiency, DCA should be performed on a packet-by-packet basis.

Examples of algorithms for packet-by-packet DCA are **Dynamic Packet Assignment** (DPA), **Dynamic Single Frequency Networks** (DSFN) and **Packet and resource plan scheduling** (PARPS).

Spread Spectrum Techniques:

In telecommunication and radio communication, spread-spectrum techniques are methods by which a signal (e.g. an electrical, electromagnetic, or acoustic signal) generated with a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth.

These techniques are used for a variety of reasons, including the establishment of secure communications, increasing resistance to natural interference, noise and jamming, to prevent detection, and to limit power flux density (e.g. in satellite downlinks).

Spread-spectrum telecommunications is a technique in which a telecommunication signal is transmitted on a bandwidth considerably larger than the frequency content of the original information.

Spread-spectrum telecommunications is a signal structuring technique that employs direct sequence, frequency hopping, or a hybrid of these, which can be used for multiple access and/or multiple functions.

Frequency-hopping spread spectrum (FHSS), direct-sequence spread spectrum (DSSS), time-hopping spread spectrum (THSS), chirp spread spectrum (CSS).

Techniques known since the 1940s and used in military communication systems since the 1950s

"spread" a radio signal over a wide frequency range several magnitudes higher than minimum requirement.

Resistance to jamming (interference). DS (direct sequence) is good at resisting continuous-time narrowband jamming, while FH (frequency hopping) is better at resisting pulse jamming.

Resistance to fading. The high bandwidth occupied by spread-spectrum signals offer some frequency diversity, i.e. it is unlikely that the signal will encounter severe multipath fading over its whole bandwidth, and in other cases the signal can be detected using e.g. a Rake receiver.

Multiple access capability, known as code-division multiple access (CDMA) or code-division multiplexing (CDM). Multiple users can transmit simultaneously in the same frequency band as long as they use different spreading codes.

4.5 COMPRESSION

Compression basically employs redundancy in the data.

Type of Compression	Applications
Temporal	1D data, 1D signals, Audio etc.,
Spatial	Correlation between neighboring pixels or data items
Spectral	correlation between color or luminescence components. This uses the frequency domain to exploit relationships between frequency of change in data
Psycho Visual	Exploit perceptual properties of the human visual system.

Compression can be categorised in two broad ways:

Lossless Compression :

where data is compressed and can be reconstituted (uncompressed) without loss of detail or information. These are referred to as bit-preserving or reversible compression systems also.

Lossy Compression :

where the aim is to obtain the best possible fidelity for a given bit-rate or minimizing the bit-rate to achieve a given fidelity measure. Video and audio compression techniques are most suited to this form of compression.

If an image is compressed it clearly needs to be uncompressed (decoded) before it can be viewed/listened to. Some processing of data may be possible in encoded form however. Lossless compression frequently involves some form of entropy encoding and are based in information theoretic techniques.

Lossy compression use source encoding techniques that may involve transform encoding, differential encoding or vector quantization.

4.5.2 MPEG COMPRESSION STANDARDS

MPEG is a group within the International Standards Organization and the International Electrochemical Commission (ISO/IEC) that undertook the job of defining standards for the transmission and storage of moving pictures and sound.

The standards are concerned only with the bit stream syntax and the decoding process, not with how encoding and decoding might be implemented. Syntax covers matters such as bit rate, picture resolution, time frames for audio, and the packet details for transmission.

The design of hardware for the encoding and decoding processes is left to the equipment manufacturer. The MPEG standards currently available are MPEG-1, MPEG-2, MPEG-4, MPEG-7 and MPEG-21.

Standard	Application
MPEG-1	The original standard for encoding and decoding streaming video and audio files.
MPEG-2	The standard for digital television, this compresses files for transmission of high-quality video.
MPEG-4	The standard for compressing high-definition video into smaller-scale files that stream to computers, cell phones and PDAs (personal digital assistants).
MPEG-7	It is a multimedia content description standard This description will be associate with the content itself, to allow fast and efficient searching for material that is of interest to the user.
MPEG-21	It is also referred to as the Multimedia Framework. The standard that interprets what digital content to provide to which individual user so that media plays flawlessly under any language, machine or user conditions

MPEG – 1

In Digital Broadcast Service (DBS) systems, MPEG-1 is used for audio compression, and MPEG-2 is used for video compression. Both of these MPEG standards cover audio and video, but MPEG-1 video is not designed for DBS transmissions.

MPEG-1 audio supports mono and two channel stereo only, which is considered adequate for DBS systems currently in use.

MPEG-2 audio supports multichannel audio in addition to mono and stereo. It is fully compatible with MPEG-1 audio.

The need for audio compression can be seen by considering the bit rate required for high-quality audio. The bit rate is equal to the number of samples per second (the sampling frequency f_s) multiplied by the number of bits per sample n :

$$R_b = f_s * n$$

For a stereo CD recording, the sampling frequency is 44.1 kHz, and the number of bits per sample is 16:

$$R_b = 44.1 * 10^3 * 16 * 2 = 1411.2 \text{ kb/s}$$

The factor 2 appears on the right-hand side because of the two channels in stereo. This bit rate, approximately 1.4 Mb/s, represents too high a fraction of the total bit rate allowance per channel, and hence the need for audio compression.

Audio compression in MPEG exploits certain perceptual phenomena in the human auditory system. In particular, it is known that a loud sound at one particular frequency will mask a less intense sound at a nearby frequency.

For example, consider a test conducted using two tones, one at 1000 Hz, which will be called the masking tone and the other at 1100 Hz, the test tone. Starting with both tones at the same level, say, 60 dB above the threshold of hearing, if now the level of the 1000Hz tone is held constant while reducing the level of the 1100-Hz tone, a point will be reached where the 1100-Hz tone becomes inaudible. The 1100-Hz tone is said to be masked by the 1000-Hz tone.

Assume for purposes of illustration that the test tone becomes inaudible when it is 18 dB below the level of the masking tone. This 18 dB is the masking threshold. It follows that any noise below the masking threshold also will be masked. For the moment, assuming that only these two tones are present, then it can be said that the noise floor is 18 dB below the masking tone. If the test-tone level is set at, say, 6 dB below the masking tone, then of course it is 12 dB above the noise floor. This means that the signal-to-noise ratio for the test tone need be no better than 12 dB. Now in a pulsecode modulated (PCM) system the main source of noise is that arising from the quantization process.

The signal-to-quantization noise ratio is given by

$$\left(\frac{S}{N}\right)_q = 2^{2n}$$

Where n is the number of bits per sample. In decibels this is

$$\left[\frac{S}{N}\right]_q = 10 \log 2^{2n} \cong 6n \text{ dB}$$

This shows that increasing n by 1 bit increases the signal-to-quantization noise ratio by 6 dB. Another way of looking at this is to say that a 1-bit decrease in n increases the quantization noise by 6 dB.

In the example above where 12 dB is an adequate signal-to-noise ratio, the above Equation shows that only 2 bits are needed to encode the 1100-Hz tone (i.e., the levels would be quantized in steps represented by 00, 01, 10, 11). By way of contrast, the CD samples taken at a sampling frequency of 44.1 kHz are quantized using 16 bits to give a signal-to-quantization noise ratio of 96 dB. Returning to the example of two tones, in reality, the audio signal will not consist of two single tones but will be a complex signal covering a wide spectrum of frequencies.

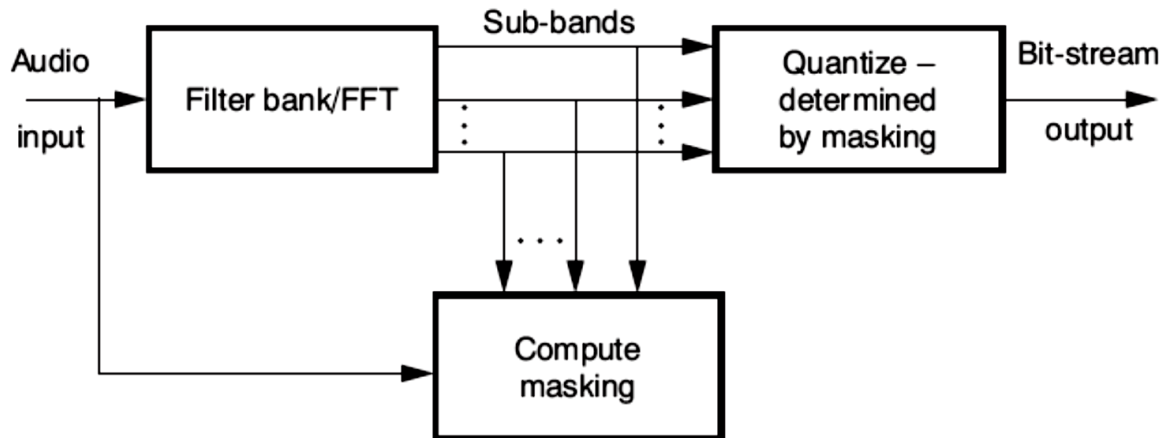


Fig 4.26 MPEG-1 Block Schematic

In MPEG-1, two processes take place in parallel, as illustrated in Fig. The filter bank divides the spectrum of the incoming signal into sub bands. In parallel with this the spectrum is analyzed to permit identification of the masking levels. The masking information is passed to the quantizer, which then quantizes the sub bands according to the noise floor.

The masking discussed so far is referred to as frequency masking for the reasons given earlier. It is also an observed phenomenon that the masking effect lasts for a short period after the masking signal is removed. This is termed temporal masking, and it allows further compression in that it extends the time for which the reduction in quantization applies.

The compressed bit rate for MPEG-1 audio used in DBS systems is 192 kb/s.

MPEG -2

In DBS systems, MPEG-2 is used for video compression. As a first or preprocessing step, the analog outputs from the red (R), green (G), and blue (B) color cameras are converted to a luminance component (Y) and two chrominance components (Cr) and (Cb). This is similar to the analog NTSC arrangement

In matrix notation, the equation relating the three primary colors to the Y, Cr, and Cb components is

$$\begin{bmatrix} Y \\ C_r \\ C_b \end{bmatrix} = \begin{bmatrix} 0.299 & 0.587 & 0.114 \\ -0.168736 & -0.331264 & 0.5 \\ 0.5 & -0.418688 & -0.081312 \end{bmatrix} \begin{bmatrix} R \\ G \\ B \end{bmatrix}$$

It is an observed fact that the human eye is less sensitive to resolution in the color components (Cr and Cb) than the luminance (Y) component. This allows a lower sampling rate to be used for the color components. This is referred to as chroma subsampling, and it represents one step in the compression process.

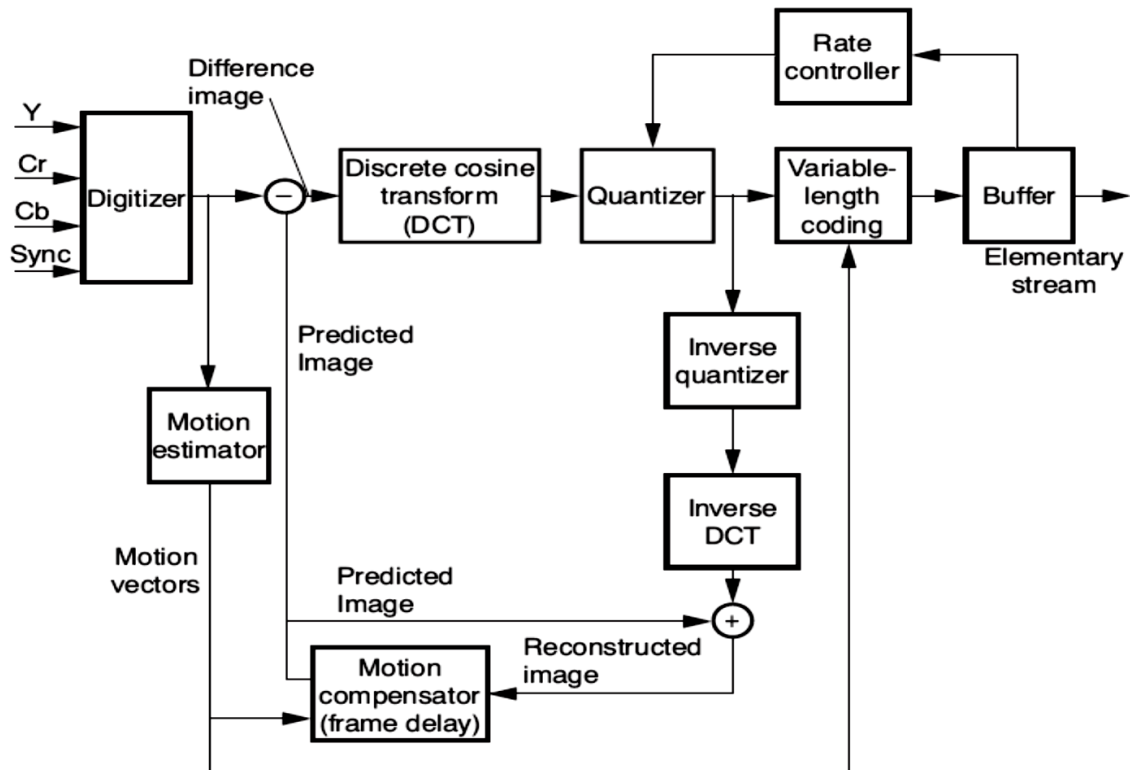


Fig 4.27 MPEG-2 encoder paths.

Sampling is usually indicated by the ratios Y: U: V where Y represents the luminance (or luma) sampling rate, U the Cb sampling rate, and V the Cr sampling rate. The values for YUV are normalized to a value of 4 for Y, and ratios commonly encountered with digital TV are 4:4:4, 4:2:2 and 4:2:0.

4:4:4 means that the sampling rates of Y, Cb, and Cr are equal. Each pixel would get three digital words, one for each of the component signals. If the words are 8-bits then each pixel would be encoded in 3 bytes.

4:2:2 means that the Cb and Cr signals are sampled at half the rate of the Y signal component. Every two pixels would have two bytes for the Y signal, one byte for the Cb signal and one byte for the Cr signal, resulting in 4 bytes for the 2-pixel block.

4:2:0 means that Cb and Cr are sampled at half the Y sampling rate, but they are sampled only on alternate scan lines. Thus vertical as well as horizontal resolution is reduced by half. A 2 * 2 pixel block would have 6 bytes, 4 bytes for Y, 1 byte for Cb and 1 byte for Cr. MPEG-2 uses 4:2:0 sampling.

Following the digitizer, difference signals are formed, and the discrete cosine transform (DCT) block converts these to a “spatial frequency” domain. The familiar Fourier transform transforms a time signal $g(t)$ to a frequency domain representation $G(f)$, allowing the signal to be filtered in the frequency domain. Here, the variables are time t and frequency f . In the DCT situation, the input signals are functions of the x (horizontal) and y (vertical) space coordinates, $g(x, y)$. The

DCT transforms these into a domain of new variables u and v , $G(u, v)$. The variables are called spatial frequencies in analogy with the time-frequency transform. It should be noted that $g(x, y)$ and $G(u, v)$ are discrete functions.

In the quantizer following the DCT transform block, the discrete values of $G(u, v)$ are quantized to predetermined levels. This reduces the number of levels to be transmitted and therefore provides compression. The components of $G(u, v)$ at the higher spatial frequencies represent finer spatial resolution. The human eye is less sensitive to resolution at these high spatial frequencies; therefore, they can be quantized in much coarser steps. This results in further compression.

Compression is also achieved through motion estimation. Frames in MPEG-2 are designated I, P, and B frames, and motion prediction is achieved by comparing certain frames with other frames.

The I frame is an independent frame, meaning that it can be reconstructed without reference to any other frames. A P (for previous) frame is compared with the previous I frame, and only those parts which differ as a result of movement need to be encoded. The comparison is carried out in sections called macroblocks for the frames. A macroblock consists of $16 * 16$ pixels. A B (for bidirectional) frame is compared with the previous I or P frame and with the next P frame. This obviously means that frames must be stored in order for the forward comparison to take place. Only the changes resulting from motion are encoded, which provides further compression.

An estimate of the compression required can be made by assuming a value of 200 Mb/s for the uncompressed bit rate for SDTV, and taking 5 Mb/s as typical of that for a TV channel, the compression needed is on the order $200/5 = 40:1$. The 5 Mb/s would include audio and data, but these should not take more than about 200 kb/s.

The whole encoding process relies on digital decision-making circuitry and is computationally intensive and expensive. The decoding process is much simpler because the rules for decoding are part of the syntax of the bit stream. Decoding is carried out in the integrated receiver decoder (IRD) unit.

MPEG – 4:

MPEG-4 was developed jointly by the Video Coding Experts/Group (VCEG) of the International Telecommunication Union (ITU), Telecommunication Standardization Sector (ITU-T) which uses the designation H.264, and the MPEG of the ISO/IEC.

This version of MPEG is known by at least six different names (H.264, H.26L, ISO/IEC 14496-10, JVT, MPEG-4 AVC, and MPEG-4 Part 10) and the abbreviation AVC is commonly used to denote advanced video coding. Following the usage in Sullivan et al., it will be denoted here by H.264/AVC. Areas of application include video telephony, video storage and retrieval (DVD and hard disk), digital video broadcast, and others.

In general terms, MPEG-4 provides many features not present with other compression schemes, such as interactivity for viewers, where objects within a scene can be manipulated, but from the point of view of satellite television, the major advantage is the reduction in bit rate offered.

About a 2:1 reduction in bit rate, on average is achievable with H.264/AVC compared with MPEG-2. Fidelity Range Extensions (FRExt) was added to H.264/AVC that can provide a reduction of as much as 3:1 in certain situations.

FRExt supports 4:2:2 and 4:4:4 sampling. As with MPEG-2 the analog outputs from the red (R), green (G), and blue (B) color cameras are converted to a luminance component (Y) and two chrominance components (Cr) and (Cb) but with a different M matrix, this being:

$$\begin{bmatrix} Y \\ C_r \\ C_B \end{bmatrix} = \begin{bmatrix} 0.2126 & 0.587 & 0.0722 \\ -0.119977 & -0.331264 & 0.523589 \\ 0.561626 & -0.418688 & -0.051498 \end{bmatrix} \begin{bmatrix} R \\ G \\ B \end{bmatrix}$$

It follows that any format conversion would require a matrix recalculation. H.264/AVC takes advantage of the increases in processing power available from computer chips, but at the cost of more expensive equipment, both for the TV broadcaster and the consumer.

As with MPEG-2, frames are compared for changes through comparing macro blocks of 16 * 16 pixels, but H.264/AVC also allows for comparisons of sub macro blocks of pixel groups 16 * 8, 8 * 16, 8 * 8, 8 * 4, 4 * 8, and 4 * 4. At present it is not backward compatible with MPEG-2, which may present a problem with some high definition TV.

4.5 CODING METHODS

At the broadcast center, the high-quality digital stream of video goes through an MPEG encoder, which converts the programming to MPEG-4 video of the correct size and format for the satellite receiver in your house.

Encoding works in conjunction with compression to analyze each video frame and eliminate redundant or irrelevant data and extrapolate information from other frames. This process reduces the overall size of the file. Each frame can be encoded in one of three ways:

As an **intraframe**, which contains the complete image data for that frame. This method provides the least compression.

As a **predicted** frame, which contains just enough information to tell the satellite receiver how to display the frame based on the most recently displayed intraframe or predicted frame

As a **bidirectional** frame, which displays information from the surrounding intraframe or predicted frames. Using data from the closest surrounding frames, the receiver **interpolates** the position and color of each pixel.

This process occasionally produces **artifacts** -- glitches in the video image. One artifact is **macroblocking**, in which the fluid picture temporarily dissolves into blocks.

Macroblocking is often mistakenly called **pixilating**, a technically incorrect term which has been accepted as slang for this annoying artifact.

There really are pixels on your TV screen, but they're too small for your human eye to perceive them individually -- they're tiny squares of video data that make up the image you see. (For more information about pixels and perception,

The rate of compression depends on the nature of the programming. If the encoder is converting a newscast, it can use a lot more predicted frames because most of the scene stays the same from one frame to the next.

In more fast-paced programming, things change very quickly from one frame to the next, so the encoder has to create more intraframes. As a result, a newscast generally compresses to a smaller size than something like a car race.

4.5.1 Encryption and Transmission:

After the video is compressed, the provider encrypts it to keep people from accessing it for free. Encryption scrambles the digital data in such a way that it can only be **decrypted** (converted back into usable data) if the receiver has the correct decryption algorithm and security keys.

Once the signal is compressed and encrypted, the broadcast center beams it directly to one of its satellites. The satellite picks up the signal with an onboard dish, amplifies the signal and uses another dish to beam the signal back to Earth, where viewers can pick it up.

Video and Audio Compression:

Video and Audio files are very large beasts. Unless we develop and maintain very high bandwidth networks (Gigabytes per second or more) we have to compress to data.

Relying on higher bandwidths is not a good option -- M25 Syndrome: Traffic needs ever increases and will adapt to swamp current limit whatever this is.

As we will compression becomes part of the representation or *coding* scheme which have become popular audio, image and video formats. Some popular coding techniques are

1. Entrophy Encoding
 - Repetitive Sequence Supression
 - Zero Length Suppression
 - Run Length Encoding
 - Statistical Encoding
 - Pattern Substitution
 - Shannon Fano & Huffman Coding
2. Source Coding
 - Transform Coding
 - FFT Fast Fourier Transform

DECRYPTION:

It is the process of taking encoded or encrypted text or other data and converting it back into text that you or the computer are able to read and understand.

This term could be used to describe a method of un-encrypting the data manually or with un-encrypting the data using the proper codes or keys.

Data may be encrypted to make it difficult for someone to steal the information. Some companies also encrypt data for general protection of company data and trade secrets. If this data needs to be viewable, it may require decryption

