Unit 3
Chapter 2

3-10. Describe *time-division multiplexing.*
Answer:-

Multiplexing is the transmission of information (in any form) from more than one source to more than one destination over the same transmission medium (facility). **Time-Division-Multiplexing:-** With *time-division multiplexing* (TDM), transmissions from multiple sources occur on the same facility but not at the same time. Transmissions from various sources are *interleaved* in the time domain. PCM is the most prevalent encoding technique used for TDM digital signals. With a PCM-TDM system, two or more voice channels are sampled, converted to PCM codes, and then time-division multiplexed onto a single metallic or optical fiber cab le. The fundamental building block for most TDM systems in the United States begins with a DS-0 channel (digital signal level 0).

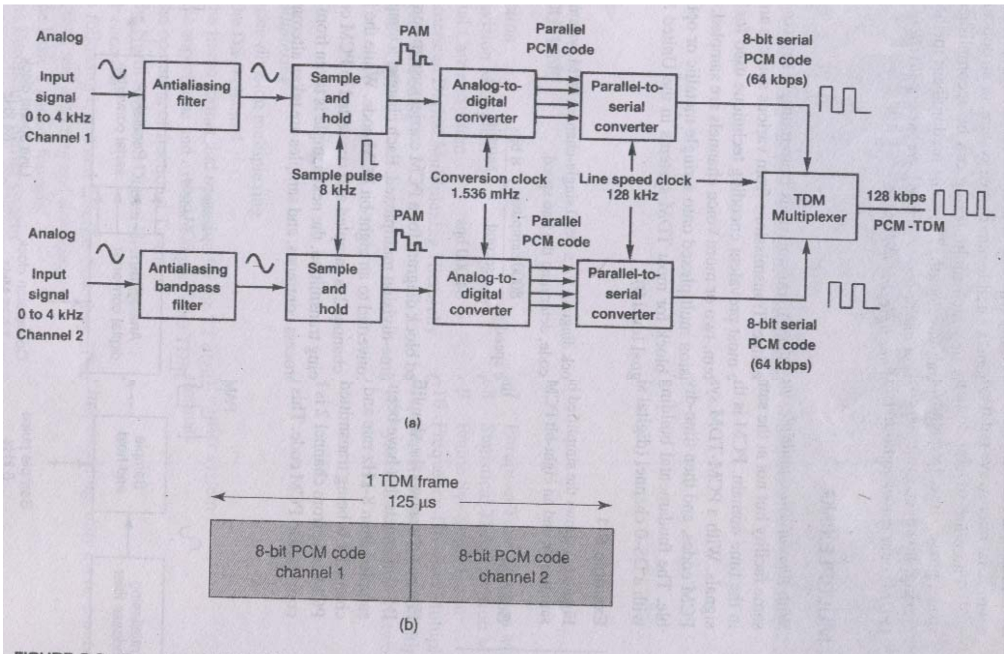

FIGURE 6-2 Two-channel PCM-TDM system: (a) block diagram; (b) TDM frame

Figure 6-2a shows the simplified block diagram for a PCM carrier system comprised of two DS-0 channels that have been time-division multiplexed. Each channel's input is alternately sampled at an 8-kHz rate and converted to an eight-bit PCM code. While the PCM code for channel 1 is being transmitted, channel 2 is sampled and converted to PCM code. While the PCM code from channel 2 is being transmitted, the next sample is taken from channel 1 and converted to PCM code. This process continues, and samples are taken alternately from each channel, converted to PCM code, and transmitted. The multiplexer is simply an electronic ally controlled digital switch with two inputs and one output. Channel 1 and channel 2 are alternately selected and connected to the transmission line through the multiplexer. One eight-bit PCM code from each channel (16 total bits) is called a TDM frame, and the time it takes to transmit one TDM frame is called *the frame time.* The frame time is equal to the reciprocal of the sample rate ($1/f_s$, or 1/8000 = *125* µs). Figure 6-2b shows the TDM frame allocation for a two-channel PCM system with an 8-kHz sample rate. The PCM code for each channel occupies a fixed time slot (epoch) within the total TDM frame.

3-11. Describe a *T1digital carrier system.*

Answer:-

A digital carrier system is a communications system that uses digital pulse rather than analog signals to encode information.

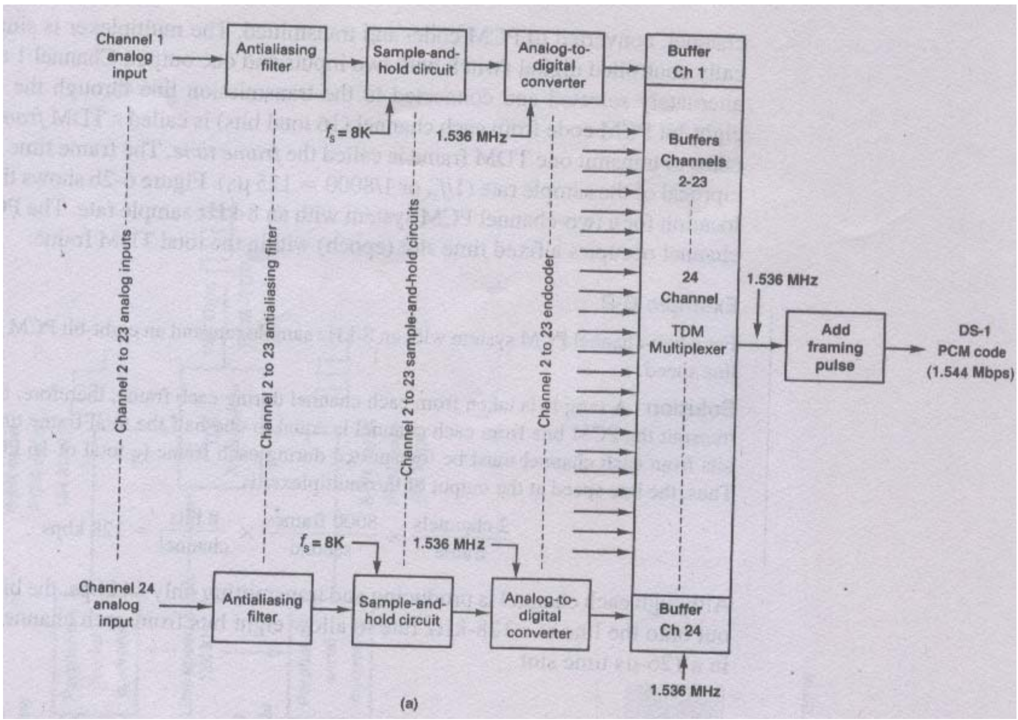

Figure 6-3a shows the block diagram for the Bell System T1 digital carrier system. A T1 carrier system time-division multiplexes PCM-encoded samples from 24 voice-band channels for transmission over a single metallic wire pair or optical fiber transmission line. Each voice-band channel has a bandwidth of approximately 300 Hz to 3000 Hz. Again, the multiplexer is simply a digital switch with 24 independent inputs and one time-division multiplexed output. The PCM output signals from the 24 voice-band channels are sequentially selected and connected through the multiplexer to the transmission line. The system does not become a TI carrier until it is line encoded and placed on special conditioned cables called *T1 lines.*

With a TI carrier system, D-type (digital) channel banks perform the sampling, encoding, and multiplexing of 24 voice-band channels. Each channel contains an eight-bit PCM code and is sampled 8000 times a second. Each channel is sampled at the same rate but not necessarily at the same time.

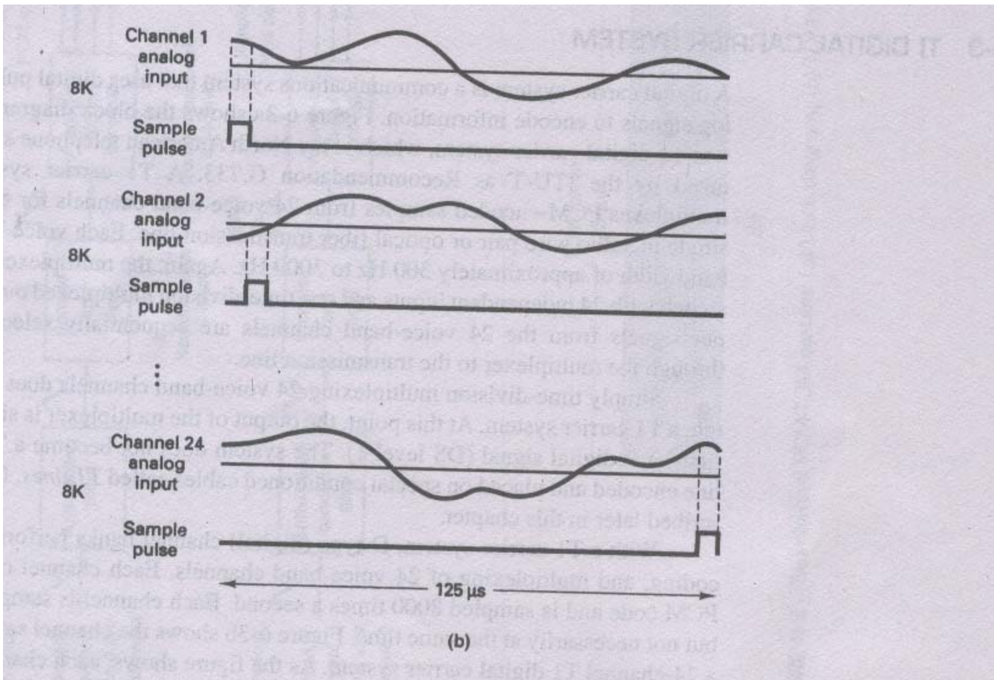

FIGURE 6-3 Bell system T1 digital carrier system: (a) block diagram; (b) sampling sequence

Figure 6-3b shows the channel sampling sequence for a 24-channel T1digital carrier system. As the figure shows, each channel is sampled once each frame but not at the same time. Each channel's sample is offset from the previous channel's sample by 1/24th of the total frame time. Therefore, one 64-kbps PCM-encoded sample is transmitted for each voice-band channel during each frame (a frame time of 1/8000 = 125 is). The line speed is 192 bits per frame.

3-12. Describe the *super frame TDM format.* Why is it used?

Answer:-

Super frame TDM Format :- The 8-kbps signaling rate used with the early digital channel banks was excessive for signaling on standard telephone voice circuits. Therefore, with modem channel banks, a signaling bit is substituted only into the least-significant bit (LSB) of every sixth frame. Hence, five of every six frames have eight-bit resolution, while one in every six frames (the signaling frame) has only seven-bit resolution. Consequently, the signaling rate on each channel is only 1.333 kbps (8000 bps/6), and the effective number of bits per sample is actually $7^{5/6}$ bits. Because only every sixth frame includes a signaling bit, it is necessary that all the frames be numbered so that the receiver knows when to extract the signaling bit. Also, because the signaling is accomplished with a two-bit binary word, it is necessary to identify the most- and least-significant bits (MSB and LSB, respectively) of the signaling word.

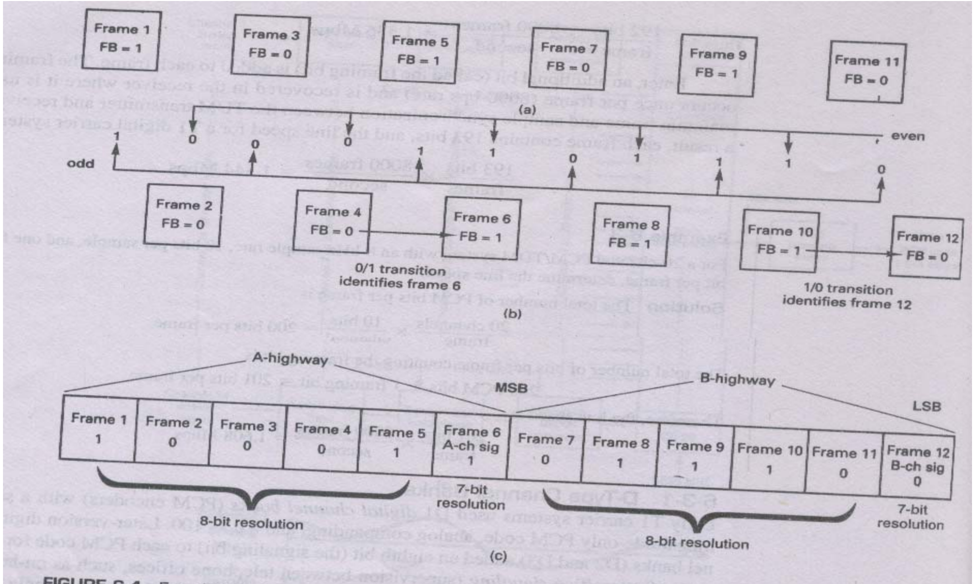

FIGURE 6-4 Framing bit sequence for the T1 superframe format using D2 and D3 channel banks: (a) frame synchronizing bits (odd-numbered frames); (b) signaling frame alignment bits (even-numbered frames); (c) composite frame alignment

Consequently, the super frame format shown in Figure 6-4 was devised. Within each super- frame are 12 consecutively numbered frames (1 to12). The signaling bits are substituted in frames 6 and 12, the MSB into frame 6, and the LSB into frame 12. Frames1to 6 are called the A highway, with frame 6 designated the A channel signaling frame. Frames 7 to 12 are called the B high way, with frame 12 designated the B channel signaling frame. Therefore, in addition to identifying the signaling frames, the sixth and twelfth frames must also be positively identified.
To identify frames 6 and 12, a different framing bit sequence is used for the odd- and even-numbered frames. The odd frames (frames 1, 3, 5, 7, 9, and 11) have an alternating 1/0) pattern, and the even frames (frames 2, 4, 6, 8, 10, and 12) have a 00 1110repetitive pattern. As a result, the combined framing bit pattern is 1000 11011100. The odd numbered frames are used for frame and sample Synchronization and the even-numbered frames are used to identify the A and B channel signaling frames (frames 6 and 12). Frame 6 is identified by a 0/1 transition in the framing bit between frames 4 and 6. Frame 12 is identified by a 1/0 transition in the framing bit between frames *10* and 12.

3-13. Describe a *fractional T carrier.*

Answer:-

Fractional T Carrier Service:- Fractional T carrier emerged because standard T1 earners provide a higher capacity (i.e., higher bit rate) than most users require. Fractional T1 systems distribute the channels (i.e., bits) in a standard T1 system among more than one user, allowing several subscribers to share one T1line. Bit rates offered with fractional Ti carrier systems are 64kbps (1 channel), 128 kbps (2 channels), *256* kbps (4 channels), 384 kbps (6 channels), 512 kbps (8 channels), and 768 kbps (12 channels), with 384 kbps (1/4 T1) and 768 kbps (1/2 T1)being the most common. The minimum data rate necessary to propagate video information is 384 kbps.

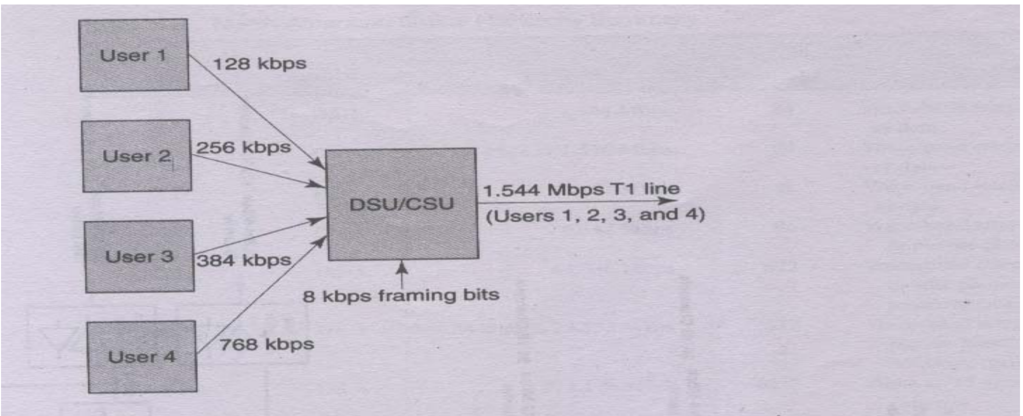

FIGURE 6-6 Fractional T1 carrier service

Figure 6-6 shows four subscribers combining their transmissions in a special unit called a *data service unit/channel service unit* (DSU/CSU). A DSU/CSU is a digital interface that provides the physical connection to a digital carrier network. User I is allocated 128 kbps, user 2,256 kbps, user 3,384 kbps, and user 4,768 kbps for a total of 1.536 kbps (8 kbps is reserved for the framing bit).

3-14. Describe *digital line encoding.*
Answer:-

Digital Line Encoding:-*Digital line encoding* involves converting standard logic levels (TTL, CMOS, and the like) to a form more suitable to telephone line transmission. Essentially, six primary factors must be considered when selecting a line-encoding format:

1. Transmission voltages and dc component

2. Duty cycle

3. Bandwidth considerations

4. Clock and framing bit recovery

5. Error detection

6. Ease of detection and decoding

Transmission voltages or levels can be categorized as being either *unipolar* (UP) or *bipolar* (BP). Unipolar transmission of binary data involves the transmission of only a single nonzero voltage level (e.g., either a positive or a negative voltage for a logic 1 and 0V for logic 0). In bipolar transmission, two nonzero voltages are involved (e.g., a positive voltage for logic 1 and an equal-magnitude negative voltage for logic 0 or vice versa). Unipolar and bipolar transmission voltages can be combined with either return-to- zero or nonreturn to zero in several ways to achieve a particular line-encoding scheme.

Figure 6-8 shows five line-encoding possibilities. In Figure 6-8a, there is only one nonzero voltage level (+V = logic 1); a zero voltage indicates a logic 0. Also, each logic 1 condition maintains the positive voltage for the entire bit time (100% duty cycle). Consequently, Figure 6-8a represents a unipolar nonreturn-to-zero signal (UPNRZ). Assuming an equal number of 1s and 0s, the average dc voltage of a UPNRZ waveform is equal to half the nonzero voltage (V/2).
In Figure 6-8b, there are two nonzero voltages (+V = logic 1 and-V = logic 0), and a 100% duty cycle is used. Therefore, Figure 6-8b represents a bipolar nonreturn-to zero signal (BPNRZ). When equal-

magnitude voltages are used for logic 1s and logic 0s and assuming an equal probability of logic 1s and Logic 0s occurring, the average dc voltage of a BPNRZ waveform is 0 V.

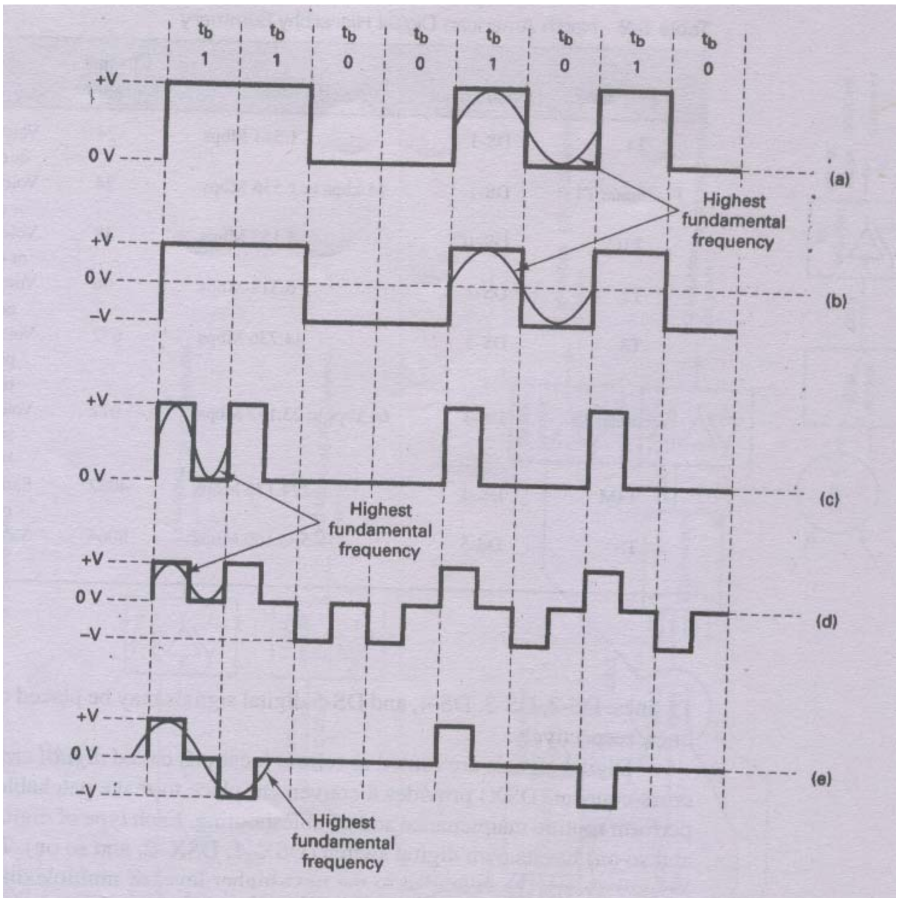

FIGURE 6-8 Line-encoding formats: (a) UPNRZ; (b) BPNRZ; (c) UPRZ; (d) BPRZ; (e) BPRZ-AMI

In Figure 6-8c, only one nonzero voltage is used, but each pulse is active for only *50%* of a bit time ($t_b/2$). Consequently, the waveform showed in Figure 6-8c represents a unipolar return-to-zero signal (UPRZ). Assuming an equal probability of 1s and 0s occurring, the average dc voltage of a UPRZ waveform is one-fourth the nonzero voltage (V/4).

Figure 6-8d shows a waveform where there are two nonzero voltages (+ V = logic 1 and-V = logic 0). Also, each pulse is active only 50% of a bit time. Consequently, the waveform shown in Figure 6-8d represents a bipolar return-to-zero (BPRZ) signal. Assuming equal-magnitude voltages for logic 1 s and logic 0s and an equal probability of 1 s and 0s occurring, the average dc voltage of a BPRZ waveform is 0 V.

In Figure 6-8e, there are again two nonzero voltage levels (-V and +V), but now both polarities represent logic 1s, and 0 V represents a logic 0. This method of line encoding is called *alternate mark inversion* (AMI). With AMI transmissions, successive logic 1s is inverted in polarity from the previous logic 1. Because return to zero is used, the encoding technique is called *bipolar-return-to-zero alternate mark inversion* (BPRZ-AMI). The average dc voltage of a BPRZ AMI waveform is approximately 0 V regardless of the bit sequence.

3-15. Contrast the bandwidth considerations of *return-to-zero* and *nonreturn-to0-zero* transmission.
Answer:-

With UPNRZ, the worst-case condition is an alternating 1/0 sequence; the period of the highest fundamental frequency takes the time of two bits and, therefore, is equal to one-half the bit rate ($f_b/2$). With BPNRZ, again the worst-case condition is an alternating 1/0 sequence, and the highest fundamental frequency is one-half the bit rate *($f_b/2$)*. With UPRZ, the worst-case condition occurs when two successive logic 1s occur. Therefore, the minimum bandwidth is equal to the bit rate *(f_b)*. With BPRZ encoding, the worst-case condition occurs for successive logic 1s or successive logic 0s, and the minimum bandwidth is again equal to the bit rate *(f_b)*. With BPRZ-AMI, the worst-case condition is two or more consecutive logic 1s, and the minimum bandwidth is equal to one-half the bit rate *($f_b/2$)*.

3-16. Contrast error detection and decoding capabilities of *return-to-zero* and *nonreturn-to-zero* transmission.

Answer:- With UPNRZ, BPNRZ, UPRZ, and BPRZ encoding, there is no way to determine if the received data have errors. However, with BPRZ-AMJ encoding, an error in any bit will cause a bipolar violation (BPV—the reception of two or more consecutive logic is with the same polarity). Therefore, BPRZ-AMI has a built-in error-detection mechanism. T carriers use BPRZ-AMI, with +3 V and —3 V representing a logic 1 and 0 V representing a logic 0.

3-17.* What is *digital biphase?
Answer:- *Digital biphase* (sometimes called the *Manchester* code ot diphase) is a popular type of line encoding that produces a strong timing component for clock recovery and does not dc wandering. Biphase is a form of BPRZ encoding that uses one cycle of a square wave at 00 phase to represent a logic 1 and one cycle of a square wave at 180^0 phase to represent a logic 0. Digital biphase encoding is shown in Figure 6-10.

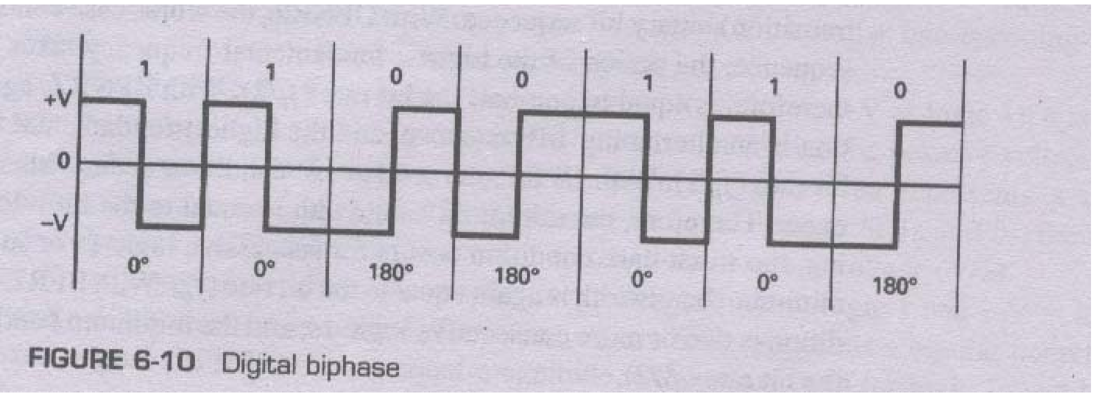

FIGURE 6-10 Digital biphase

Note that a transition occurs the center of every signaling element regardless of its logic condition or phase. Thus biphase produces a strong timing component for clock recovery. In addition, assuming equal probability of 1s and 0s, the average dc voltage is 0 V. and there is no dc wandering. A disadvantage of biphase is that it contains no means of error detection.

3-18. What is a *regenerative repeater?*
Answer:-

Figure 6-11 shows the block diagram for a regenerative repeater. Essentially, there are three functional blocks: an *amplifier/equalizer,* a *timing clock recovery circuit,* and the *regenerator* itself. The amplifier/equalizer filters and shapes the incoming digital signal and raises its power level so that the regenerator circuit can make a pulse-no pulse decision.

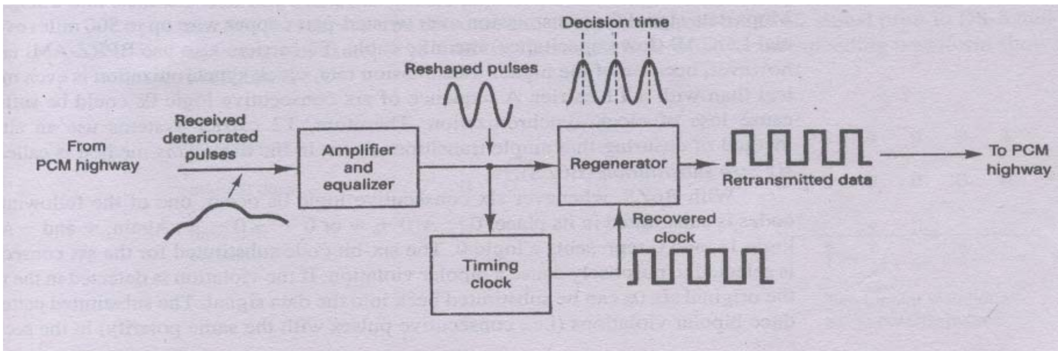

FIGURE 6-11 Regenerative repeater block diagram

The timing clock recovery circuit reproduces the clocking information from the received data and provides the proper timing information to the regenerator so that samples can be made at the optimum time, minimizing the chance of an error occurring. A regenerative rep eater is simply a threshold detector that compares the sampled voltage received to a reference level and determines whether the bit is logic 1 or logic 0.

Spacing of repeaters is designed to maintain an adequate signal-to-noise ratio for error-free performance. The signal-to-noise ratio at the output of a regenerative repeater is exactly what it was at the output of the transmit terminal or at the output of the previous regenerator (i.e., the signal-to-noise ratio does not deteriorate as a digital signal propagates through a regenerator; in fact, a regenerator reconstructs the original pulses with the origin al signal-to-noise ratio).

3-19. Briefly explain the following framing techniques: *added-digit framing, robbed -digit framing, added-channel framing, statistical framing,* and *unique-line code framing.*
Answer:-

Added-Digit Framing:- T1 carriers using Dl, D2, or D3 channel banks use *added-digit framing.* A special *framing digit* (framing pulse) is added to each frame. Consequently, for an 8-kHz sample rate, 8000 digits are added each second. To acquire frame synchronization, the digital terminal in the receiver searches through the incoming data until it finds the framing bit pattern. This encompasses testing a bit, counting off 193 more bits, and then testing again for the opposite logic condition. This process continues until a repetitive alternating 1/0 pattern is found. Initial frame Synchronization depends on the total frame time, the number of bits per frame, and the period of each bit. Searching through all possible bit positions requires *N* tests, where *N* is the number of bit positions in the frame.

Robbed-Digit Framing:- When a short frame is used, added-digit framing is inefficient. This occurs with single-channel PCM systems. An alternative solution is to replace the least significant bit of every nth frame with a framing bit. This process is called *robbed-digit framing.*

Added Channel Framing:- *Added-channel framing* is the same as added-digit framing except that digits at added in groups or words instead of as individual bits. One of the 32 times in each frame is dedicated to a unique synchronizing bit sequence.

Statistical Framing:- With *statistical framing,* it is not necessary to either rob or add digits. With the gray code, the second bit is logic 1 in the central half of the code range and logic 0 at the extremes. Therefore, a signal that has a centrally peaked amplitude distribution generates a high probability of logic 1 in the second digit. Hence, the second digit of a given channel can be used for the framing bit.

Unique-Line Code Framing:- With *unique-line code framing,* some property of the framing bit is different from the data bits. The framing bit is either made higher or lower in amplitude or with different time duration. With unique-line code framing, either added-digit or added-word framing can be used, or specified data bits can be used to simultaneously convey information and carry synchronizing signals. The advantage of unique- line code framing is that synchronization is immediate and automatic. The disadvantage is the additional processing requirements necessary to generate and recognize the unique bit.

3-20. Describe *frequency-division multiplexing.*

With *frequency-division multiplexing* (FDM), multiple sources that originally occupied the same frequency spectrum are each converted to a different frequency band and transmitted simultaneously over a single transmission medium, which can be a physical cable or the earth’s atmosphere (i.e., wireless). Thus, many relatively narrow-bandwidth channels can be transmitted over a single wide-bandwidth transmission system without interfering with each other. FDM is used for combining many relatively narrowband sources into a single wideband channel, such as in public telephone systems. Essentially, FDM is taking a given bandwidth and subdividing it into narrower segments with each segment carrying different information.

Figure 6-18a shows a simple FDM system where four 6-kHz channels are frequency division multiplexed into a single 20-kHz combined channel. As the figure shows, channel 1 signals amplitude modulates a 100 kHz carrier in a balanced modulator which inherently suppresses the 100 kHz carrier. The output of the balanced modulator is a double side band suppressed carrier waveform with a bandwidth of 10 kHz. The double sideband waveform passes through a band pass filter (BPF) where it is converted to a single side band signal. For this example, the lower sideband is blocked; thus, the output of the BPF occupies the frequency band between 100 kHz and 105 kHz (a bandwidth of *5* kHz).

Channel 2 signals amplitude modulate a *l05-kHz* carrier in a balanced modulator, again Producing a double sideband signal that is converted to single sideband *by* passing it through a band pass filter tuned to pass only the upper sideband.

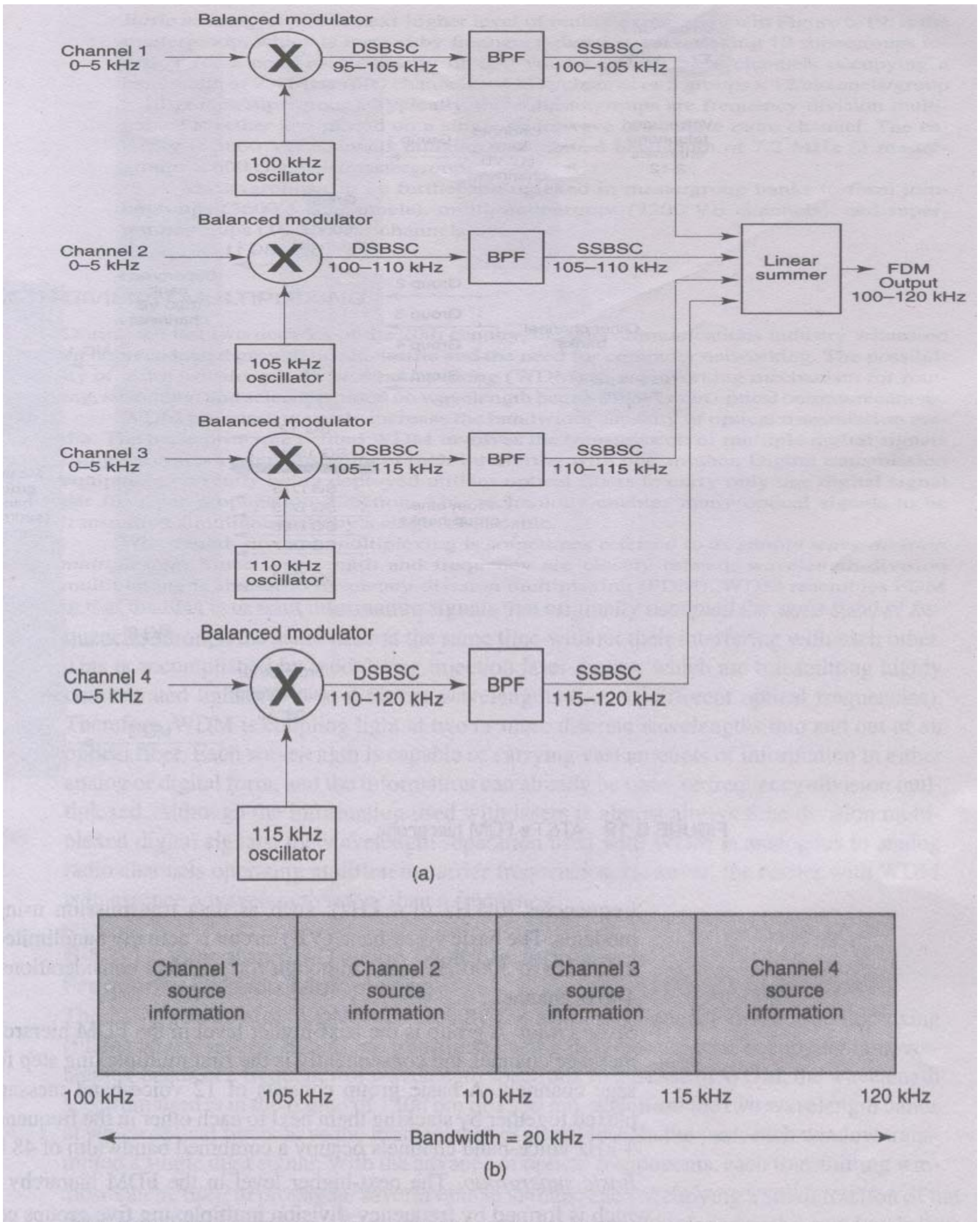

FIGURE 6-18 Frequency-division multiplexing: (a) block diagram; (b) frequency spectrum

Thus, the output from the BPF occupies a frequency band between 105 kHz and 110 kHz. The same process is used to convert signals from channels 3 and 4 to the frequency bands 110 *kHz* to 115 kHz and 115 kHz to 120 kHz, respectively The combined frequency spectrum produced by combining the outputs from the four band pass filters is shown in Figure 6-18b. As the figure shows, the total combined bandwidth is equal to 20 kHz, and each channel occupies a different 6kHz portion of the total 20-kHz bandwidth.

3-21. Briefly describe the AT&T *FDM hierarchy.*

Answer:-

Figure 6-19 shows AT&T's FDM hierarchy. As the figure shows, voice channels are combined to form groups, groups are combined to form super groups and super groups are combined to form master groups.

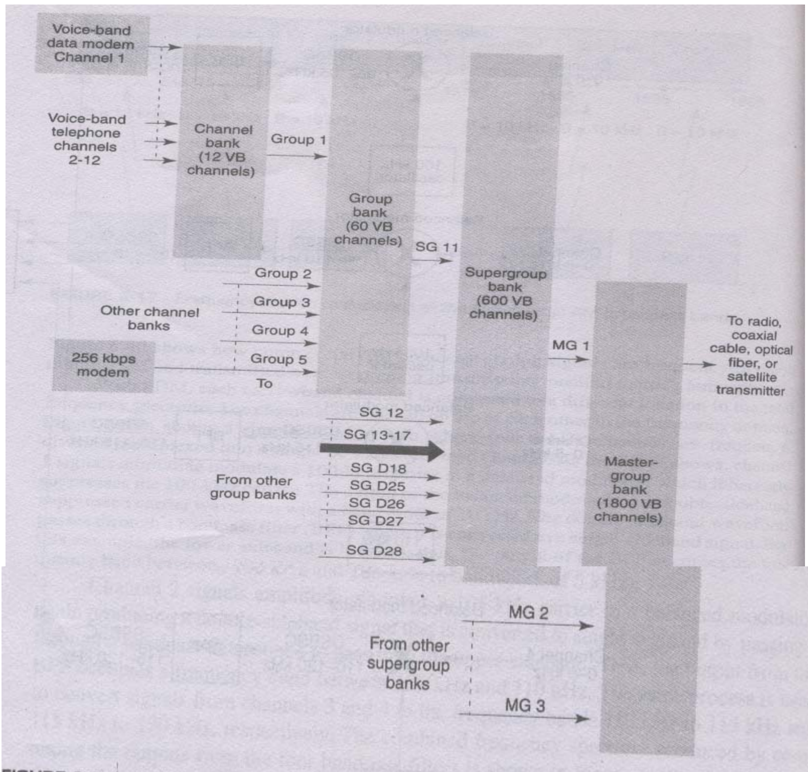

FIGURE 6-19 AT&T's FDM hierarchy

Message channel:- The message channel is the basic building block of the FDM hierarchy. The basic message channel was originally intended for the analog transmission of voice signals, although it now includes any transmissions, that utilize voice-band frequencies (0 kHz to 4 kHz), such as data transmission using voice-band data modems. The basic voice-band (VB) circuit is actually band1imted to approximately a 300-Hz to 3000-Hz band, although for practical considerations it is considered a 4-kHz channel.

***Basic group**:-* A group is the next-higher level in the FDM hierarchy above the basic message channel and consequently is the first multiplexing step for combining message channels. A basic group consists of 12 voice-band message channels multiplexed together by stacking them next to each other in the frequency domain. Twelve4-kHz voice-band channels occupy a combined bandwidth of 48 kHz (4 x 12).

Basic super group:- The next-higher level in the FDM hierarchy is the super group, which is formed by frequency division multiplexing five groups containing 12 channels each, for a combined bandwidth of 240 kHz *(*5 groups x 48 kHz /group or 5 groups x 12 channels/group x 4 kHz/channel).

***Basic master group**:-* The next-higher level of multiplexing, shown in Figure 6-19, is the mastergroup, which is formed by frequency-division multiplexing 10 supergroups tog ether for a combined capacity of 600 voice-band message channels occupying a bandwidth of 2.4 MHz (600 channels x 4 kHz/channel or 5 groups x 12/channels/group x 10 groups/supergroup). Typically, three mastergroups are frequency-division multiplexed together and placed on a single microwave or satellite radio channel. The Capacity is 1800 VB channels utilizing a combined bandwidth of 7.2 MHz (3 master- groups x 600 channels/mastergroup).

3-22. Describe WDM versus FDM.
Answer:-

The basic principle of WDM is essentially the same as frequency-division multiplexing (FDM) where several signals are transmitted using different carriers, occupying non over lapping bands of a frequency or wavelength spectrum. In the case of WDM, the wavelength spectrum used is in the region of 1300 nm or 1500 nm, which are the two wavelength bands at which optical fibers have the least amount of signal loss.
Although FDM and WDM share similar principles, they are not the same. The most obvious difference is that optical frequencies (in THz) are much higher than radio frequencies (in MHz and 0Hz). Probably the most significant difference, however, is in the way the two signals propagate through their respective transmission media. With FDM, signals propagate at the same time and through the same medium and follow the same transmission path. The basic principle of WDM, however, is somewhat different. Different wavelengths in a light pulse travel through an optical fiber at different speeds (e.g., blue light propagates slower than red light). With WDM, information signals from multiple sources modulate lasers operating at different wavelengths. Hence, the signals enter the fiber at the same time and travel through the same medium. However, they do not take the same path down the fiber. Since each wavelength takes a different transmission path, each arrives at the receive end at slightly different times.

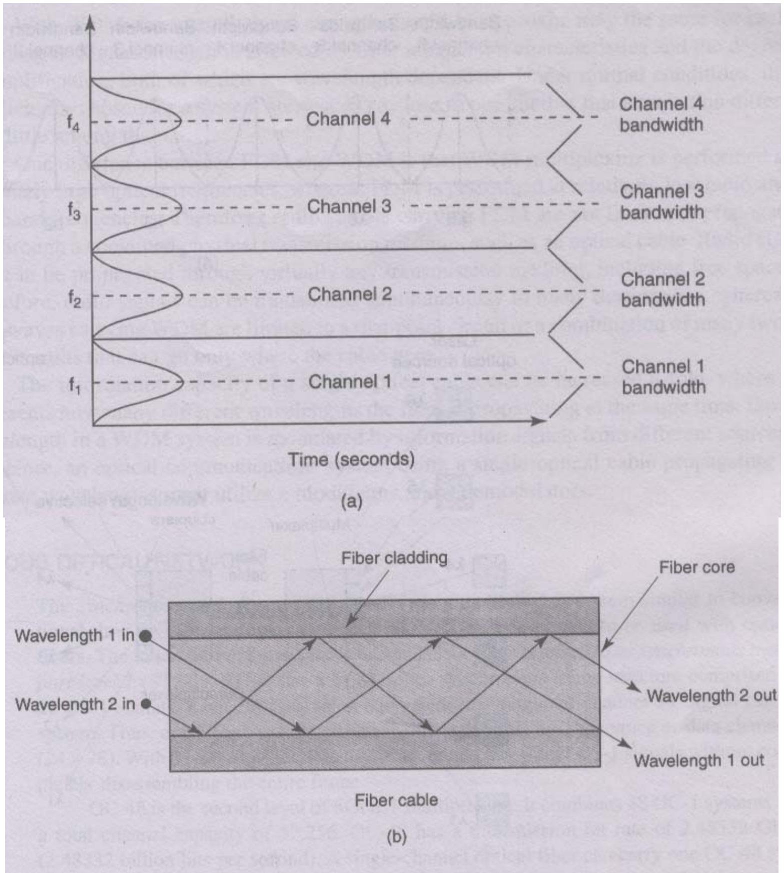

FIGURE 6-20 (a) Frequency-division multiplexing; (b) wave-length-division multiplexing

Figure 6-20 illustrates the basic principles of FDM and WDM signals propagating through their respective transmission media. As shown in Figure 6-20a, FDM channels all propagate at the same time and over the same transmission medium and take the same transmission path; however, they occupy different bandwidths. In Figure 6-20b, it can be seen that with WDM, each channel propagates down the same transmission medium at the same time; however, each channel occupies a different bandwidth (wavelength), and each wavelength takes a different transmission path.