6-1 INTRODUCTION

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). Although transmissions occur on the same facility, they do not necessarily occur at the same time. The transmission medium may be a metallic wire pair, a coaxial cable, a PCS mobile telephone, a terrestrial microwave radio system, a satellite microwave system, or an optical fiber cable.

There are several domains in which multiplexing can be accomplished, including space, phase, time, frequency, and wavelength. The most predominant methods of multiplexing, however, are *time-division multiplexing* (TDM), *frequency-division multiplexing* (FDM), and *wavelength-division multiplexing* (WDM).

6-2 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 cable. The fundamental building block for most TDM systems in the United States begins with a DS-0 channel (digital signal level 0).

Example 6-1

Figure 6-1 shows the simplified block diagram for a DS-0 single-channel PCM system. For an 8-kHz sample rate and an eight-bit PCM code, determine the line speed.

Solution

$$\text{line speed} = \frac{8000 \text{ samples}}{\text{second}} \times \frac{8 \text{ bits}}{\text{sample}}$$

$$= 64{,}000 \text{ bps}$$

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

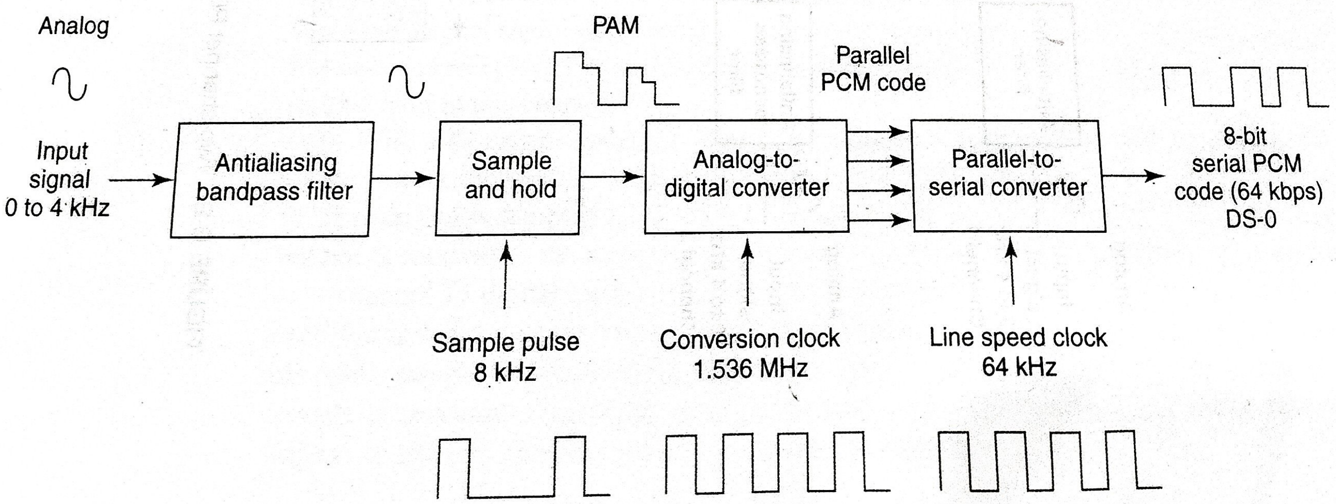


FIGURE 6-1 Single-channel (DS-0-level) PCM transmission system

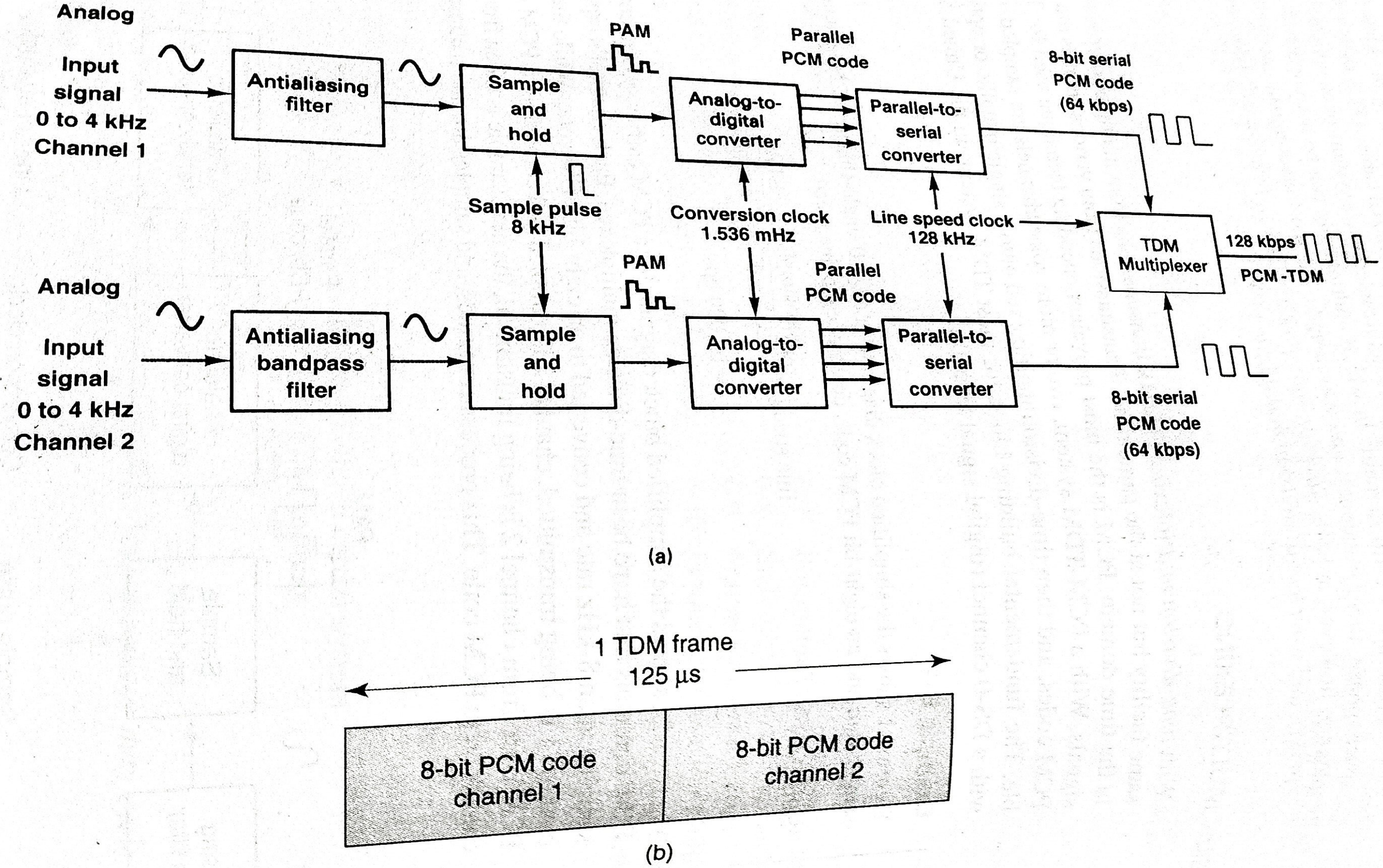


FIGURE 6-2 Two-channel PCM-TDM system: *(a)* block diagram; *(b)* TDM frame

6-3 TI DIGITAL CA

channel, converted to PCM code, and transmitted. The multiplexer is simply an electronically 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 $1/8000 = 125\ \mu s$). 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.

Example 6-2

For a two-channel PCM system with an 8-kHz sample rate and an eight-bit PCM code, determine the line speed.

Solution A sample is taken from each channel during each frame; therefore, the time allocated to transmit the PCM bits from each channel is equal to one-half the total frame time. Therefore, eight bits from each channel must be transmitted during each frame (a total of 16 PCM bits per frame). Thus, the line speed at the output of the multiplexer is

$$\frac{2\text{ channels}}{\text{frame}} \times \frac{8000\text{ frames}}{\text{second}} \times \frac{8\text{ bits}}{\text{channel}} = 128\text{ kbps}$$

Although each channel is producing and transmitting only 64 kbps, the bits must be clocked out onto the line at a 128-kHz rate to allow eight bits from each channel to be transmitted in a 126-μs time slot.

T1 DIGITAL CARRIER SYSTEM

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, which is the North American telephone standard and recognized by the ITU-T as Recommendation G.733. 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.

Simply time-division multiplexing 24 voice-band channels does not in itself constitute a T1 carrier system. At this point, the output of the multiplexer is simply a multiplexed first-level digital signal (DS level 1). The system does not become a T1 carrier until it is line encoded and placed on special conditioned cables called *T1 lines*. Line encoding is described later in this chapter.

With a T1 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 6-3b shows the channel sampling sequence for a 24-channel T1 digital 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 $1/8000 = 125\ \mu s$). The line speed is calculated as follows:

$$\frac{24\text{ channels}}{\text{frame}} \times \frac{8\text{ bits}}{\text{channel}} = 192\text{ bits per frame}$$

Biom. ① GET = +19% (standard) Put. delete

② DOSt

procurement

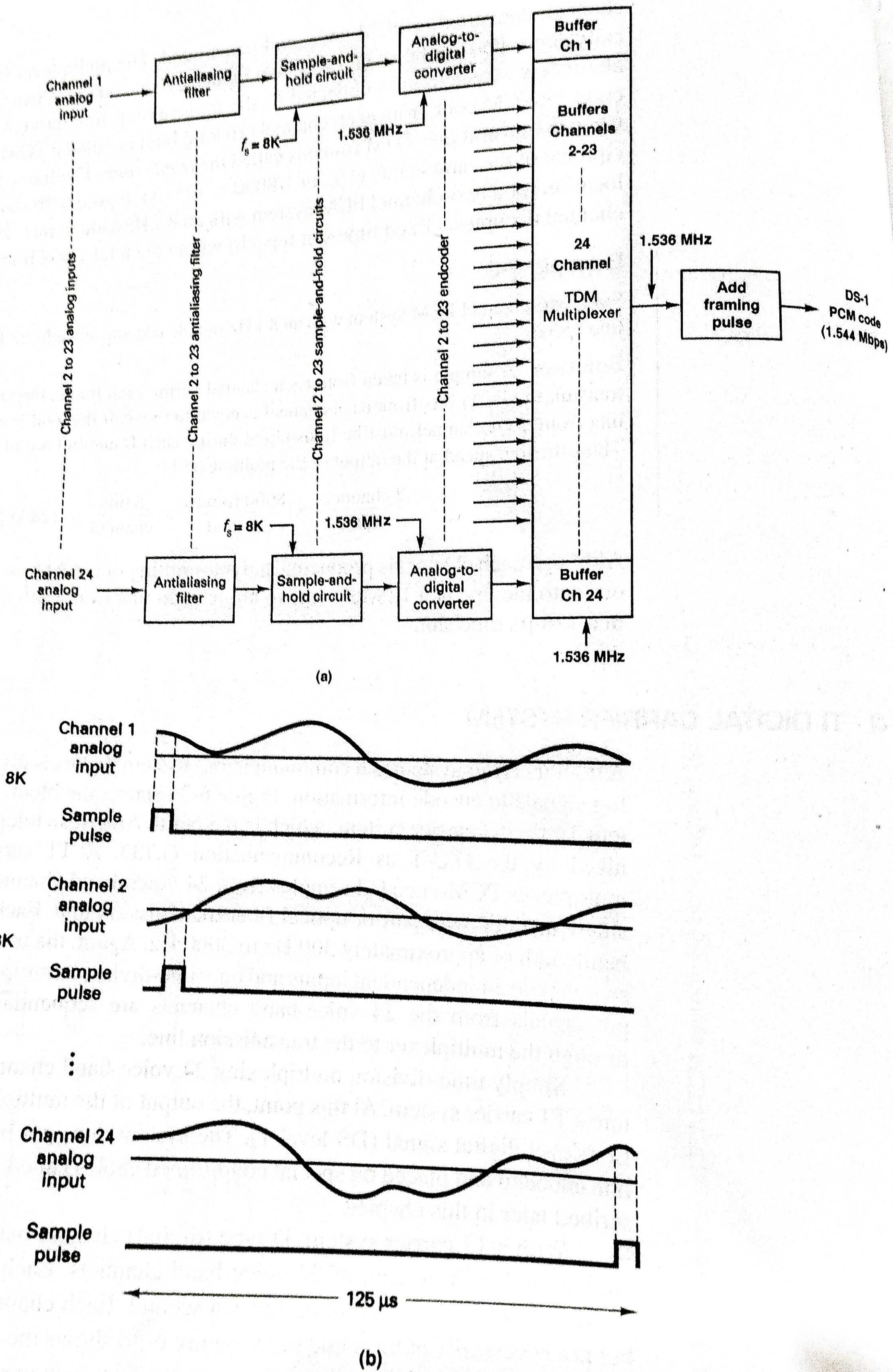


FIGURE 6-3 Bell system T1 digital carrier system: (a) block diagram; (b) sampling sequ

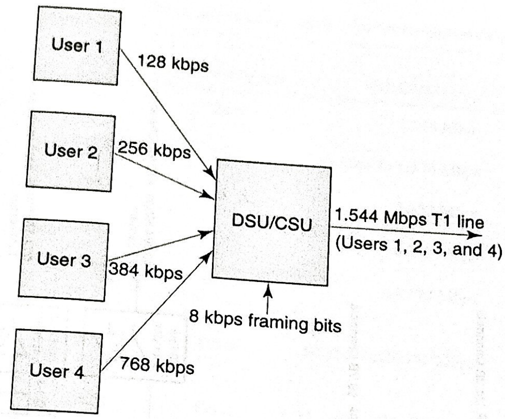


FIGURE 6-6 Fractional T1 carrier service

6-3-4 Fractional T Carrier Service

Fractional T carrier emerged because standard T1 carriers 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 T1 line. For example, several small businesses located in the same building can share one T1 line (both its capacity and its cost).

Bit rates offered with fractional T1 carrier systems are 64 kbps (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. Fractional T3 is essentially the same as fractional T1 except with higher channel capacities, higher bit rates, and more customer options.

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 1 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).

6-4 NORTH AMERICAN DIGITAL MULTIPLEXING HIERARCHY

Multiplexing signals in digital form lends itself easily to interconnecting digital transmission facilities with different transmission bit rates. Figure 6-7 shows the American Telephone and Telegraph Company's (AT&T's) North American Digital Hierarchy for multiplexing digital signals into a single higher-speed pulse stream suitable for transmission on the next-higher level of the hierarchy. To upgrade from one level in the hierarchy to the next-higher level, a special device called *muldem* (*mul*tiplexers/*dem*ultiplexer) is required. Muldems can handle bit-rate conversions in both directions. The muldem designations (M112, M23, and so on) identify the input and output digital signals associated with that muldem. For instance, an M12 muldem interfaces DS-1 and DS-2 digital signals. An M23 muldem interfaces DS-2 and DS-3 digital signals. As the figure shows, DS-1 signals may be further multiplexed or line encoded and placed on specially conditioned cables called

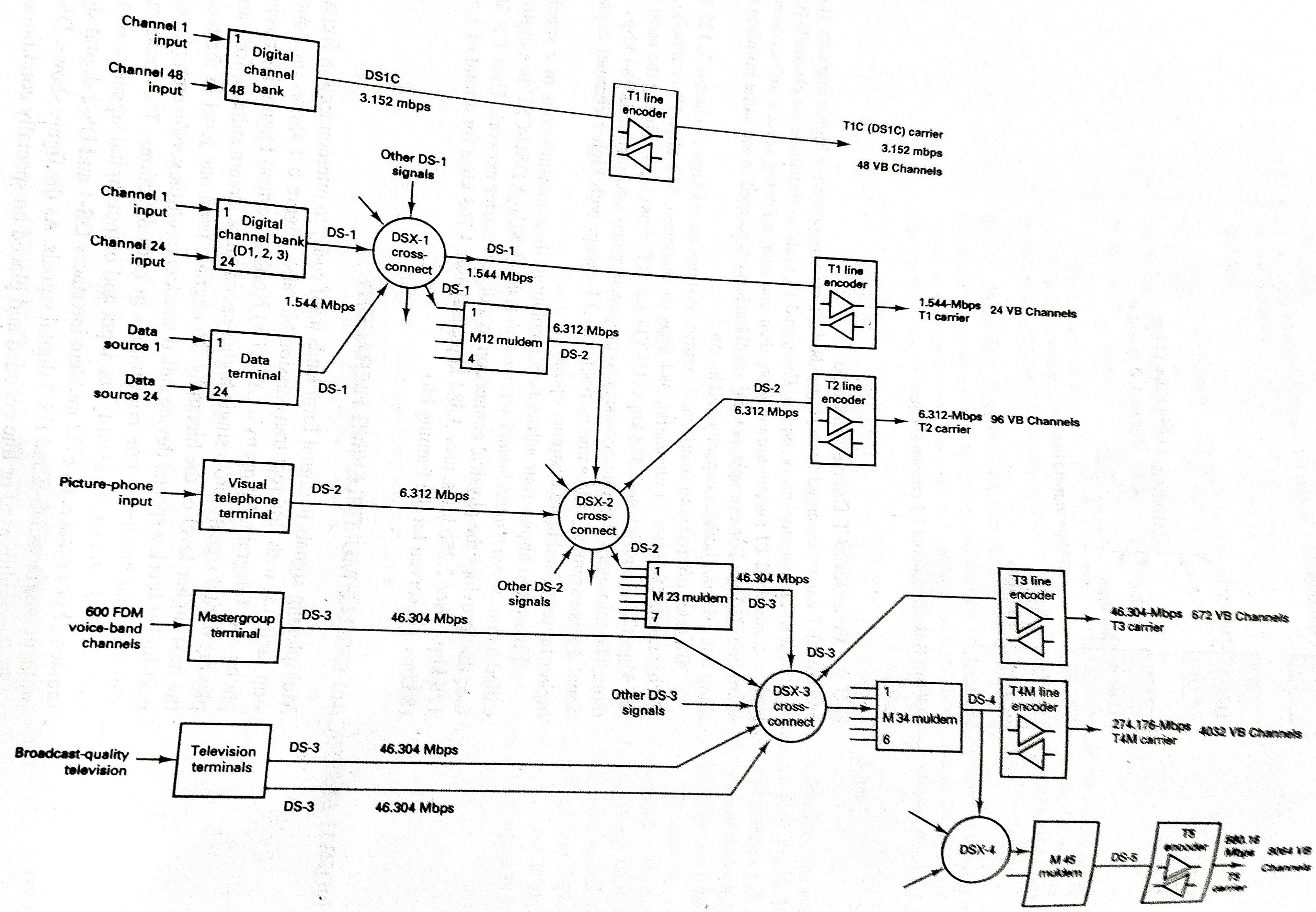


FIGURE 6-7 North American Digital Hierarchy

Table 6-2 North American Digital Hierarchy Summary

Line Type	Digital Signal	Bit Rate	Channel Capacities	Servi
T1	DS-1	1.544 Mbps	24	Voice-band or data
Fractional T1	DS-1	64 kbps to 1.536 Mbps	24	Voice-band or data
T1C	DS-1C	3.152 Mbps	48	Voice-band telephone or data
T2	DS-2	6.312 Mbps	96	Voice-band telephone, data, or picture phone
T3	DS-3	44.736 Mbps	672	Voice-band telephone, data, picture phone, and broadcast-quality television
Fractional T3	DS-3	64 kbps to 23.152 Mbps	672	Voice-band telephone, data, picture phone, and broadcast-quality television
T4M	DS-4	274.176 Mbps	4032	Same as T3 except more capacity
T5	DS-5	560.160 Mbps	8064	Same as T3 except more capacity

T1 lines. DS-2, DS-3, DS-4, and DS-5 digital signals may be placed on T2, T3, T4M, or T5 lines, respectively.

Digital signals are routed at central locations called *digital cross-connects*. A digital cross-connect (DSX) provides a convenient place to make patchable interconnects and to perform routine maintenance and troubleshooting. Each type of digital signal (DS-1, DS-2, and so on) has its own digital switch (DSX–1, DSX–2, and so on). The output from a digital switch may be upgraded to the next-higher level of multiplexing or line encoded and placed on its respective T lines (T1, T2, and so on).

Table 6-2 lists the digital signals, their bit rates, channel capacities, and services offered for the line types included in the North American Digital Hierarchy.

6-5 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

6-5-1 Transmission Voltages and DC Component

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 0 V [ground] for a logic 0). In bipolar transmission, two nonzero voltages are involved (e.g., a positive voltage for a logic 1 and an equal-magnitude negative voltage for a logic 0 or vice

6-6 T CARRIER SYSTEMS

T carriers are used for the transmission of PCM-encoded time-division multiplexed digital signals. In addition, T carriers utilize special line-encoded signals and metallic cables that have been conditioned to meet the relatively high bandwidths required for high-speed digital transmission. Digital signals deteriorate as they propagate along a cable because of power losses in the metallic conductors and the low-pass filtering inherent in parallel-wire transmission lines. Consequently, *regenerative repeaters* must be placed at periodic intervals. The distance between repeaters depends on the transmission bit rate and the line-encoding technique used.

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. 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 repeater is simply a threshold detector that compares the sampled voltage received to a reference level and determines whether the bit is a logic 1 or a 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 original signal-to-noise ratio).

6-6-1 T1 Carrier System

T1 carrier systems were designed to combine PCM and TDM techniques for the transmission of 24 64-kbps channels with each channel capable of carrying digitally encoded voice-band telephone signals or data. The transmission bit rate (*line speed*) for a T1 carrier is 1.544 Mbps, including an 8-kbps framing bit. The lengths of T1 carrier systems typically range from about 1 mile to over 50 miles.

T1 carriers use BPRZ-AMI encoding with regenerative repeaters placed every 3000, 6000, or 9000 feet. The transmission medium for T1 carriers is generally 19- to 22-gauge twisted-pair metallic cable. Because T1 carriers use BPRZ-AMI encoding, they are susceptible to losing clock synchronization on long strings of consecutive logic 0s. Ensuring

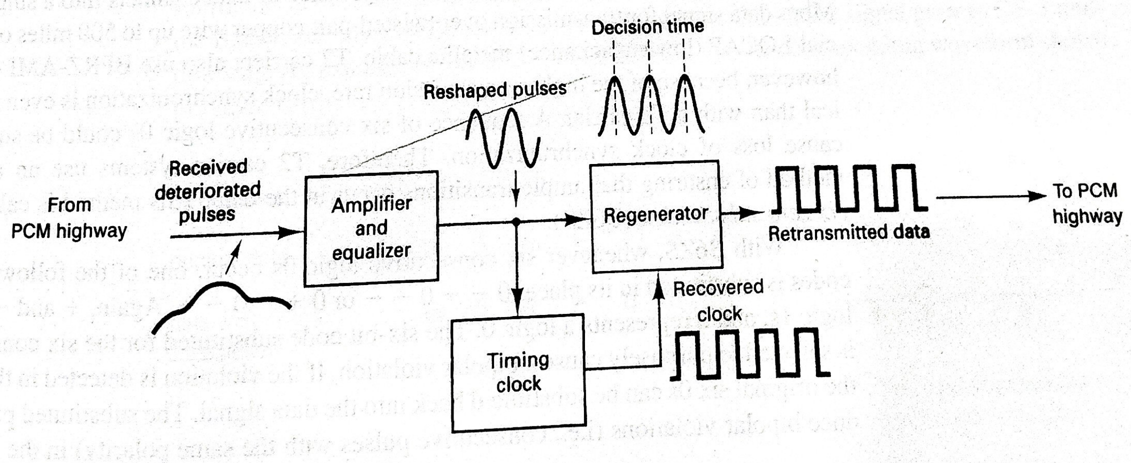
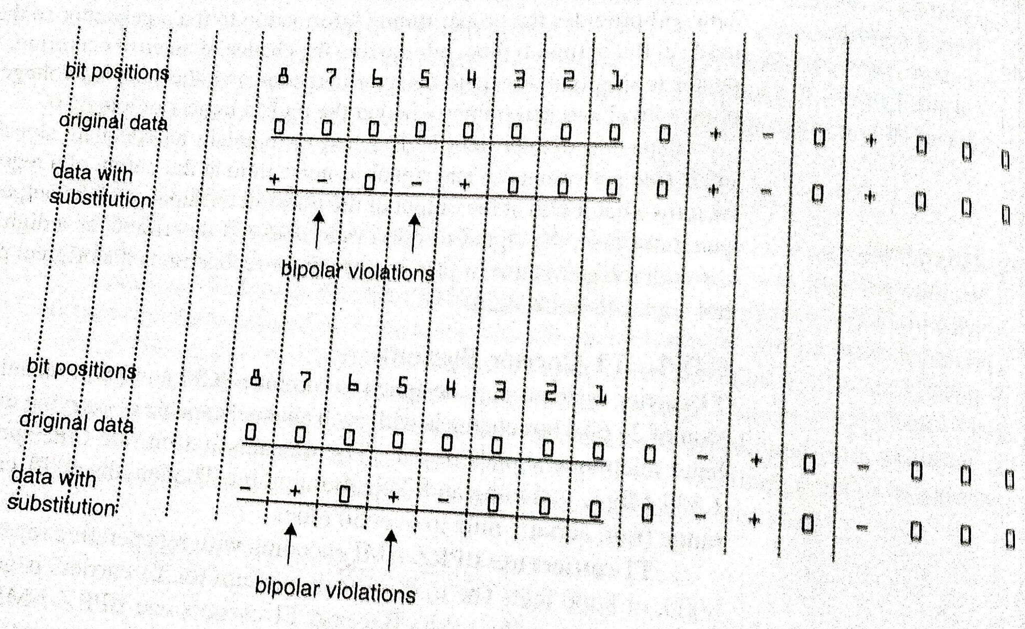


FIGURE 6-11 Regenerative repeater block diagram

sufficient transitions occur in the data stream is sometimes called *ones density*. Early T1 carrier systems provided measures to ensure that no single eight-bit byte was transmitted without at least one bit being a logic 1 or that 15 or more consecutive logic 0s were not transmitted. With modern T1 carriers, a technique called *binary eight zero substitution* (B8ZS) is used to ensure that sufficient transitions occur in the data to maintain clock synchronization. With B8SZ, whenever eight consecutive 0s are encountered, one of two special patterns is substituted for the eight 0s, either + − 0 − + 0 0 0 or − + 0 + − 0 0 0. The + (plus) and − (minus) represent bipolar logic 1 conditions, and a 0 (zero) indicates a logic 0 condition. The eight-bit pattern substituted for the eight consecutive 0s is the one that purposely induces bipolar violations in the fourth and seventh bit positions. Ideally, the receiver will detect the bipolar violations and the substituted pattern and then substitute the eight 0s back into the data signal. During periods of low usage, eight logic 1s are substituted into idle channels. Two examples of B8ZS are illustrated here with their corresponding waveforms shown in Figures 6–12a and 6–12b, respectively.



6-6-2 T2 Carrier System

T2 carriers time-division multiplex 96 64-kbps voice or data channels into a single 6.312-Mbps data signal for transmission over twisted-pair copper wire up to 500 miles over a special LOCAP (low capacitance) metallic cable. T2 carriers also use BPRZ-AMI encoding; however, because of the higher transmission rate, clock synchronization is even more critical than with a T1 carrier. A sequence of six consecutive logic 0s could be sufficient to cause loss of clock synchronization. Therefore, T2 carrier systems use an alternative method of ensuring that ample transitions occur in the data. This method is called *binary six zero substitution* (B6ZS).

With B6ZS, whenever six consecutive logic 0s occur, one of the following binary codes is substituted in its place: 0 − + 0 + − or 0 + − 0 − +. Again, + and − represent logic 1s, and 0 represents a logic 0. The six-bit code substituted for the six consecutive 0s is selected to purposely cause a bipolar violation. If the violation is detected in the receiver, the original six 0s can be substituted back into the data signal. The substituted patterns produce bipolar violations (i.e., consecutive pulses with the same polarity) in the second and

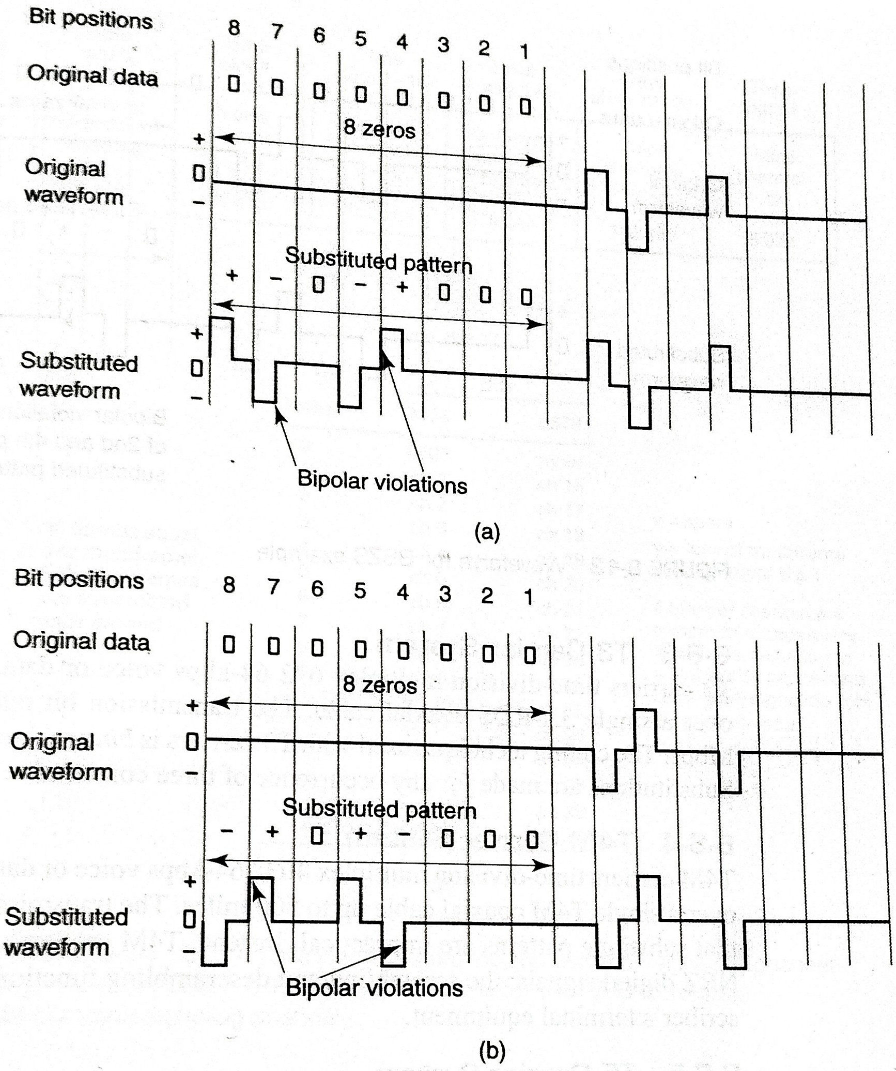
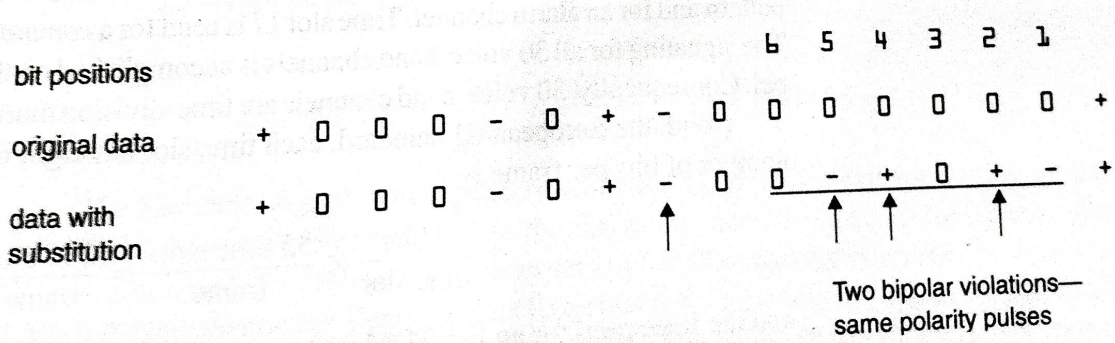


FIGURE 6-12 Waveforms for B8ZS example: (a) substitution pattern 1; (b) substitution pattern 2

fourth bits of the substituted patterns. If DS-2 signals are multiplexed to form DS-3 signals, the B6ZS code must be detected and stripped off from the DS-2 signal prior to DS-3 multiplexing. An example of B6ZS is illustrated here with its corresponding waveform shown in Figure 6-13.



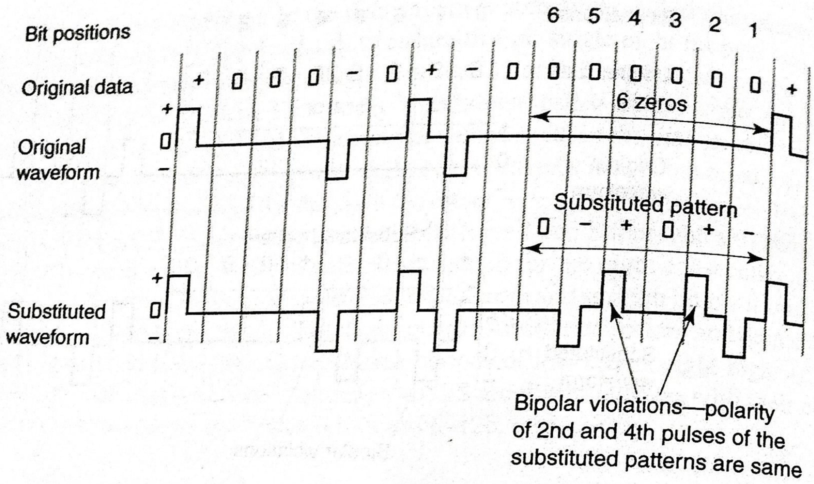


FIGURE 6-13 Waveform for B6ZS example

6-6-3 T3 Carrier System

T3 carriers time-division multiplex 672 64-kbps voice or data channels for transmission over a single 3A-RDS coaxial cable. The transmission bit rate for T3 signals is 44.736 Mbps. The coding technique used with T3 carriers is *binary three zero substitution* (B3ZS). Substitutions are made for any occurrence of three consecutive 0s.

6-6-4 T4M Carrier System

T4M carriers time-division multiplex 4032 64-kbps voice or data channels for transmission over a single T4M coaxial cable up to 500 miles. The transmission rate is sufficiently high that substitute patterns are impractical. Instead, T4M carriers transmit scrambled unipolar NRZ digital signals; the scrambling and descrambling functions are performed in the subscriber's terminal equipment.

6-6-5 T5 Carrier System

T5 carriers time-division multiplex 8064 64-kbps voice or data channels and transmits them at a 560.16-Mbps rate over a single coaxial cable.

6-7 EUROPEAN TIME-DIVISION MULTIPLEXING

In Europe, a different version of T carrier lines is used called *E lines*. Although the two systems are conceptually the same, they have different capabilities. Figure 6-14 shows the frame alignment for the E1 European standard PCM-TDM system. With the basic E1 system, a 125-μs frame is divided into 32 equal time slots. Time slot 0 is used for a frame alignment pattern and for an alarm channel. Time slot 17 is used for a *common signaling channel* (CSC). The signaling for all 30 voice-band channels is accomplished on the common signaling channel. Consequently, 30 voice-band channels are time-division multiplexed into each E1 frame.

With the European E1 standard, each time slot has eight bits. Consequently, the total number of bits per frame is

$$\frac{8 \text{ bits}}{\text{time slot}} \times \frac{32 \text{ time slots}}{\text{frame}} = \frac{256 \text{ bits}}{\text{frame}}$$

and the line speed for an E-1 TDM system is

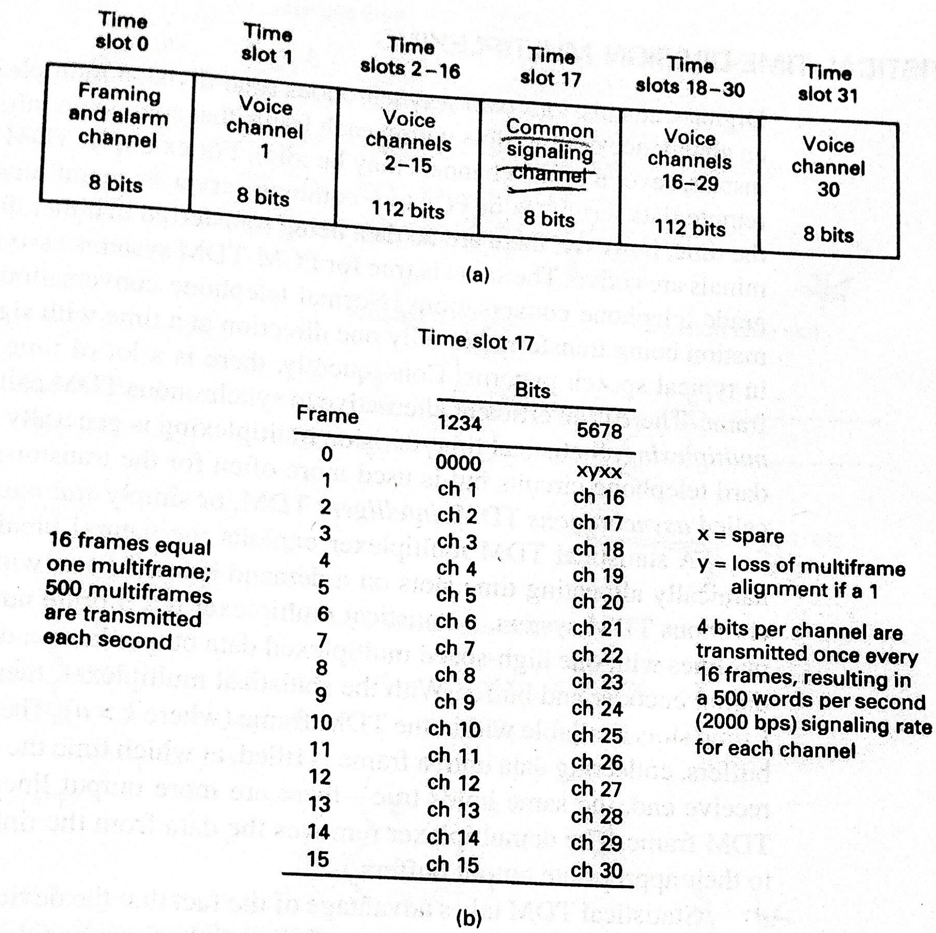


FIGURE 6-14 CCITT TDM frame alignment and common signaling channel alignment: (a) CCITT TDM frame (125 μs, 256 bits, 2.048 Mbps); (b) common signaling channel

Table 6-4 European Transmission Rates and Capacities

Line	Transmission Bit Rate (Mbps)	Channel Capacity
E1	2.048	30
E2	8.448	120
E3	34.368	480
E4	139.264	1920

$$\text{line speed} = \frac{256 \text{ bits}}{\text{frame}} \times \frac{8000 \text{ frames}}{\text{second}} = 2.048 \text{ Mbps}$$

The European digital transmission system has a TDM multiplexing hierarchy similar to the North American Hierarchy except the European system is based on the 32-time-slot (30-voice-channel) E1 system. The *European Digital Multiplexing Hierarchy* is shown in Table 6-4. Interconnecting T carriers with E carriers is not generally a problem because most multiplexers and demultiplexers are designed to perform the necessary bit rate conversions.

6-8 STATISTICAL TIME-DIVISION MULTIPLEXING

Digital transmissions over a synchronous time-division multiplexing system often contain an abundance of time slots within each frame that contain no information (i.e., at any given instant, several of the channels may be idle). For example, TDM is commonly used to link remote data terminals or PCs to a common server or mainframe computer. A majority of the time, however, there are no data being transferred in either direction, even if all the terminals are active. The same is true for PCM-TDM systems carrying digital-encoded voice-grade telephone conversations. Normal telephone conversations generally involve information being transferred in only one direction at a time with significant pauses embedded in typical speech patterns. Consequently, there is a lot of time wasted within each TDM frame. There is an efficient alternative to synchronous TDM called *statistical time-division multiplexing*. Statistical time-division multiplexing is generally not used for carrying standard telephone circuits but is used more often for the transmission of data when they are called *asynchronous* TDM, *intelligent* TDM, or simply *stat muxs*.

A statistical TDM multiplexer exploits the natural breaks in transmissions by dynamically allocating time slots on a demand basis. Just as with the multiplexer in a synchronous TDM system, a statistical multiplexer has a finite number of low-speed data input lines with one high-speed multiplexed data output line, and each input line has its own digital encoder and buffer. With the statistical multiplexer, there are n input lines but only k time slots available within the TDM frame (where $k > n$). The multiplexer scans the input buffers, collecting data until a frame is filled, at which time the frame is transmitted. On the receive end, the same holds true—there are more output lines than time slots within the TDM frame. The demultiplexer removes the data from the time slots and distributes them to their appropriate output buffers.

TDM

HDLC
otocol

Statistical TDM takes advantage of the fact that the devices attached to the inputs and outputs are not all transmitting or receiving all the time, and the data rate on the multiplexed line is lower than the combined data rates of the attached devices. In other words, statistical TDM multiplexers require a lower data rate than synchronous multiplexers need to support the same number of inputs. Alternately, a statistical TDM multiplexer operating at the same transmission rate as a synchronous TDM multiplexer can support more users.

Figure 6-15 shows a comparison between statistical and synchronous TDM. There are four data sources (A, B, C, and D) and four time slots, or epochs (t_1, t_2, t_3, and t_4). The synchronous multiplexer has an output data rate equal to four times the data rate of each of the input channels. During each sample time, data are collected from all four sources and transmitted regardless of whether there is any input. As the figure shows, during sample time t_1, channels C and D have no input data, resulting in a transmitted TDM frame void of information in time slots C_1 and D_1. With a statistical multiplexer, however, the empty time slots are not transmitted. A disadvantage of the statistical format is that the length of a frame varies and the positional significance of each time slot is lost. There is no way of knowing beforehand which channel's data will be in which time slot or how many time slots are included in each frame. Because data arrive and are distributed to receive buffers unpredictably, address information is required to ensure proper delivery. This necessitates more overhead per time slot for statistical TDM because each slot must carry an address as well as data.

The frame format used by a statistical TDM multiplexer has a direct impact on system performance. Obviously, it is desirable to minimize overhead to improve data throughput. Normally, a statistical TDM system will use a synchronous protocol such as HDLC (described in detail in a later chapter). With statistical multiplexing, control bits must be included within the frame. Figure 6-16a shows the overall frame format for a statistical TDM multiplexer. The frame includes a beginning flag and ending flag that indicate the beginning and end of the frame, an address field that identifies the transmitting device, a control field, a statistical TDM subframe, and a frame check sequence field (FCS) which

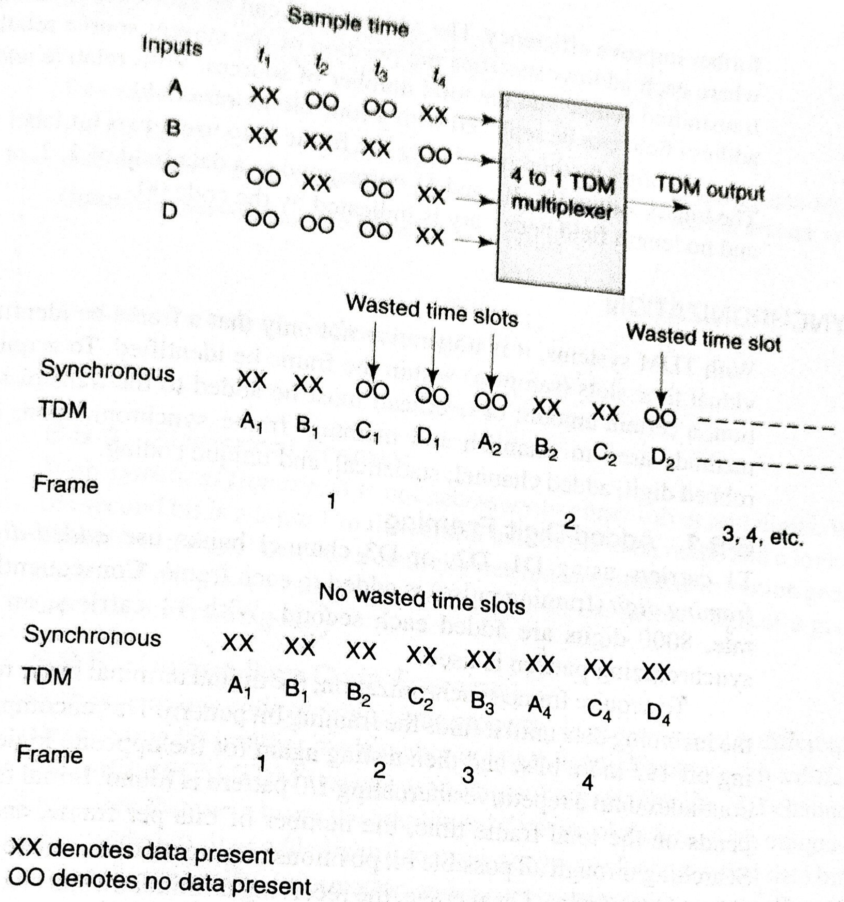


FIGURE 6-15 Comparison between synchronous and statistical TDM

(a)

Beginning flag	Address field	Control field	Statistical TDM subframe	FCS field	Ending flag

(b)

Address field	Data field

(c)

Address field	Length field	Data field	...	Address field	Length field	Data field

FIGURE 6-16 Statistical TDM frame format: (a) overall statistical TDM frame; (b) one source per frame; (c) multiple sources per frame

Figure 6-16b shows the frame when only one data source is transmitting. The transmitting device is identified in the address field. The data field length is variable and limited only by the maximum length of the frame. Such a scheme works well in times of light loads but rather inefficiently under heavy loads. Figure 6-16c shows one way to improve the efficiency by allowing more than one data source to be included within a single frame. With multiple sources, however, some means is necessary to specify the length of the data stream from each source. Hence, the statistical frame consists of sequences of data fields labeled with an address and a bit count. There are several techniques that can be used to further improve efficiency. The address field can be shortened by using relative addressing where each address specifies the position of the current source relative to the previously transmitted source and the total number of sources. With relative addressing, an eight-bit address field can be replaced with a four-bit address field.

Another method of refining the frame is to use a two-bit label with the length field. The binary values 01, 10, and 11 correspond to a data field of 1, 2, or 3 bytes, respectively, and no length field necessary is indicated by the code 00.

6-9 FRAME SYNCHRONIZATION

With TDM systems, it is imperative not only that a frame be identified but also that individual time slots (samples) within the frame be identified. To acquire frame synchronization, a certain amount of overhead must be added to the transmission. There are several methods used to establish and maintain frame synchronization, including added digit, robbed digit, added channel, statistical, and unique coding.

6-9-1 Added-Digit Framing

T1 carriers using D1, 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. With T1 carriers, an alternating 1/0 frame-synchronizing pattern is used.

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. On average, the receiving terminal dwells at a false framing position for two frame periods during a search; therefore, the maximum average synchronization time is

$$\text{synchronization time} = 2NT = 2N^2t_b \qquad \textbf{(6-13)}$$

where N = number of bits per frame
T = frame period of $N\,t_b$
t_b = bit time

For the T1 carrier, $N = 193$, $T = 125\ \mu s$, and $t_b = 0.648\ \mu s$; therefore, a maximum of 74,498 bits must be tested, and the maximum average synchronization time is 48.25 ms.

6-9-2 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*. The parameter n is chosen as a compromise between reframe time and signal impairment. For $n = 10$, the SQR is impaired by only 1 dB. Robbed-digit framing does not interrupt transmission but instead periodically replaces information bits with forced data errors to maintain frame synchronization.

6-9-3 Added-Channel Framing

Essentially, *added-channel framing* is the same as added-digit framing except that digits are added in groups or words instead of as individual bits. The European time-division multiplexing scheme previously discussed uses added-channel framing. One of the 32 time slots in each frame is dedicated to a unique synchronizing bit sequence. The average number of bits to acquire frame synchronization using added-channel framing is

$$\text{number of synchronization bits} = \frac{N^2}{\ldots}$$

where N = number of bits per frame
K = number of bits in the synchronizing word

Example 6-5

For the European E1 32-channel system with $N = 256$, $K = 8$, and a 2.048-Mbps transmission rate, determine the average number of bits needed to synchronize and the average synchronization time.

Solution Substituting into Equation 6-14 yields

$$\frac{256^2}{2(2^8 - 1)} = 128.5 \text{ bits}$$

Therefore, the average synchronization time is

$$\frac{128.5 \text{ bits}}{2.048 \text{ Mbps}} = 62.7 \ \mu\text{s}$$

6-9-4 Statistical Framing

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

6-9-5 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 a different time duration. The earliest PCM-TDM systems used unique-line code framing. D1 channel banks used framing pulses that were twice the amplitude of normal data bits. 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.

-10 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.

FDM is an analog multiplexing scheme; the information entering an FDM system must be analog, and it remains analog throughout transmission. If the original source information is digital, it must be converted to analog before being frequency-division multiplexed.

A familiar example of FDM is the commercial AM broadcast band, which occupies a frequency spectrum from 535 kHz to 1605 kHz. Each broadcast station carries an information signal (voice and music) that occupies a bandwidth between 0 Hz and 5 kHz. If the information from each station were transmitted with the original frequency spectrum, it would be impossible to differentiate or separate one station's transmissions from another. Instead, each station amplitude modulates a different carrier frequency and produces a 10-kHz signal. Because the carrier frequencies of adjacent stations are separated by 10 kHz, the total commercial AM broadcast band is divided into 107 10-kHz frequency slots stacked next to each other in the frequency domain. To receive a particular station, a receiver is simply tuned to the frequency band associated with that station's transmissions.

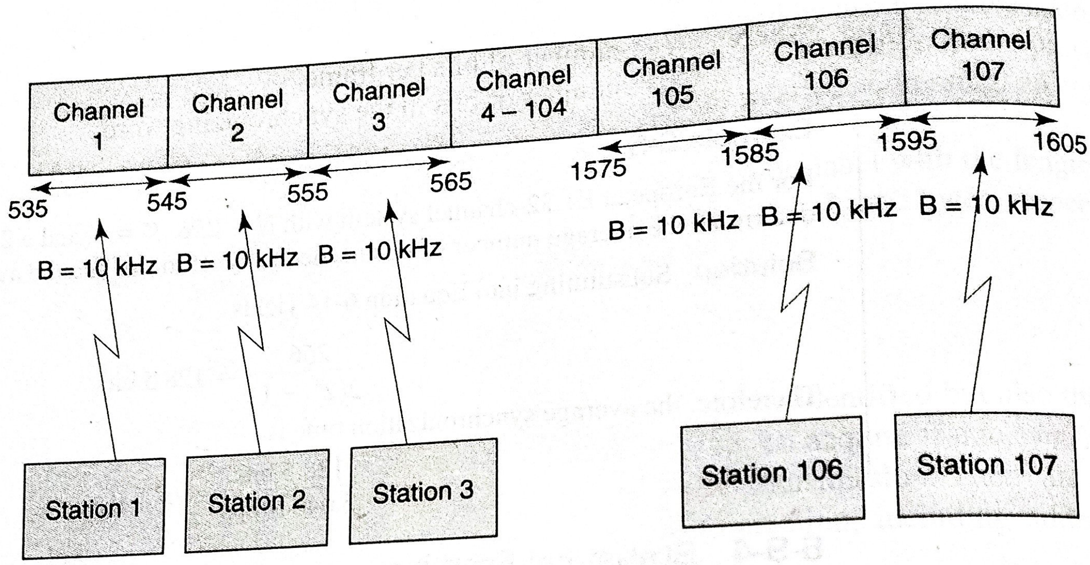


FIGURE 6-17 Frequency-division multiplexing of the commercial AM broadcast band

Figure 6-17 shows how commercial AM broadcast station signals are frequency-division multiplexed and transmitted over a common transmission medium (earth's atmosphere).

With FDM, each narrowband channel is converted to a different location in the total frequency spectrum. The channels are stacked on top of each other in the frequency domain. 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 modulate a 100-kHz carrier in a balanced modulator, which inherently suppresses the 100-kHz carrier. The output of the balanced modulator is a double-sideband suppressed carrier waveform with a bandwidth of 10 kHz. The double-sideband waveform passes through a bandpass filter (BPF) where it is converted to a single-sideband 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 105-kHz carrier in a balanced modulator, again producing a double-sideband signal that is converted to single sideband by passing it through a bandpass filter tuned to pass only the upper sideband. 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 bandpass 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 6-kHz portion of the total 20-kHz bandwidth.

There are many other applications for FDM, such as commercial FM and television broadcasting, high-volume telephone and data communications systems, and cable television and data distribution networks.

bandwidth of 2.4 MHz (600 channels × 4 kHz/channel or 5 groups × 12/c[...] × 10 groups/supergroup). Typically, three mastergroups are frequency-[...]plexed together and placed on a single microwave or satellite radio cha[...]pacity is 1800 VB channels utilizing a combined bandwidth of 7.2 MH[...] groups × 600 channels/mastergroup).

Mastergroups can be further multiplexed in mastergroup banks to form jumbogroups (3600 VB channels), multijumbogroups (7200 VB channels), and superjumbogroups (10,800 VB channels).

6-11 WAVELENGTH-DIVISION MULTIPLEXING

During the last two decades of the 20th century, the telecommunications industry witnessed an unprecedented growth in data traffic and the need for computer networking. The possibility of using *wavelength-division multiplexing* (WDM) as a networking mechanism for routing, switching, and selection based on wavelength began a new era in optical communications. WDM promises to vastly increase the bandwidth capacity of optical transmission media. The basic principle behind WDM involves the transmission of multiple digital signals using several wavelengths without their interfering with one another. Digital transmission equipment currently being deployed utilizes optical fibers to carry only one digital signal per fiber per propagation direction. This technology enables many optical signals to be transmitted simultaneously by a single fiber cable.

Wavelength-division multiplexing is sometimes referred to as simply *wave-division multiplexing*. Since wavelength and frequency are closely related, wavelength-division multiplexing is similar to frequency-division multiplexing (FDM). WDM resembles FDM in that the idea is to send information signals that originally occupied the same band of frequencies through the same fiber at the same time without their interfering with each other. This is accomplished by modulating injection laser diodes, which are transmitting highly concentrated light waves at different wavelengths (i.e., at different optical frequencies). Therefore, WDM is coupling light at two or more discrete wavelengths into and out of an optical fiber. Each wavelength is capable of carrying vast amounts of information in either analog or digital form, and the information can already be time- or frequency-division multiplexed. Although the information used with lasers is almost always time-division multiplexed digital signals, the wavelength separation used with WDM is analogous to analog radio channels operating at different carrier frequencies. However, the carrier with WDM is in essence a wavelength rather than a frequency.

6-11-1 Wavelength-Division Multiplexing versus Frequency-Division Multiplexing

The basic principle of WDM is essentially the same as frequency-division multiplexing (FDM) where several signals are transmitted using different carriers, occupying nonoverlapping 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. In the past, each window transmitted a single digit signal. With the advance of optical components, each transmitting window can be used to propagate several optical signals, each occupying a small fraction of the total wavelength window. The number of optical signals multiplexed with a window is limited only by the precision of the components used. Current technology allows over 100 optical channels to be multiplexed into a single optical fiber.

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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 GHz). 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). In standard optical fiber communications systems, as the light propagates down the cable, wavelength dispersion causes the light waves to spread out and distribute their energy over a longer period of time. Thus, in standard optical fiber systems, wavelength dispersion creates problems, which impose limitations on the system's performance. With WDM, however, wavelength dispersion is the essence of how the system operates.

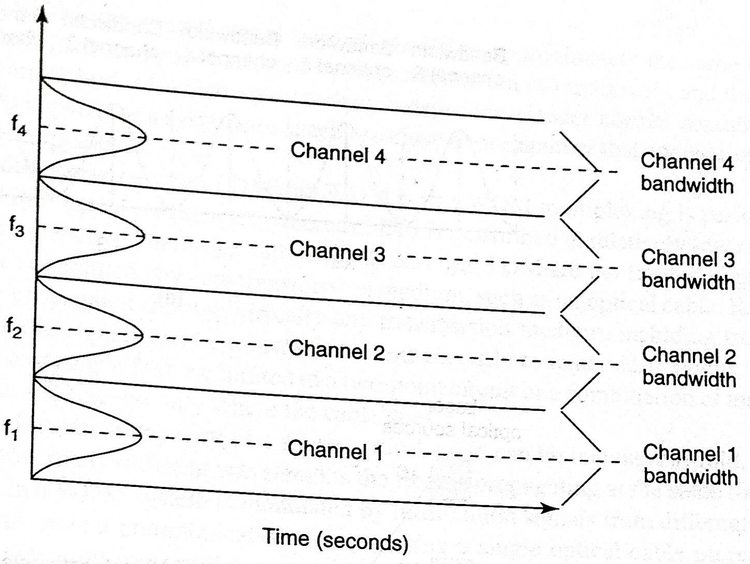
With WDM, information signals from multiple sources modulate lasers operating at different wavelengths. Hence, the signals enter the fiber at the same time and travel through the same medium. However, they do not take the same path down the fiber. Since each wavelength takes a different transmission path, each arrives at the receive end at slightly different times. The result is a series of rainbows made of different colors (wavelengths) each about 20 billionths of a second long, simultaneously propagating down the cable.

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.

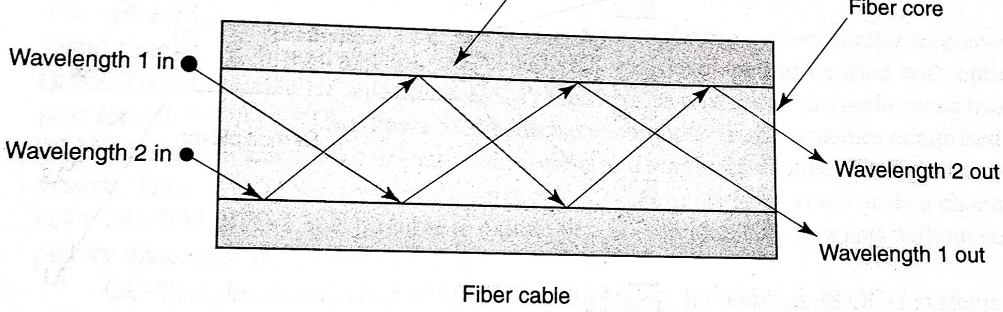
6-11-2 Dense-Wave-Division Multiplexing, Wavelengths, and Wavelength Channels

WDM is generally accomplished at approximate wavelengths of 1550 nm (1.55 μm) with successive frequencies spaced in multiples of 100 GHz (e.g., 100 GHz, 200 GHz, 300 GHz, and so on). At 1550-nm and 100-GHz frequency separation, the wavelength separation is approximately 0.8 nm. For example, three adjacent wavelengths each separated by 100 GHz correspond to wavelengths of 1550.0 nm, 1549.2 nm, and 1548.4 nm. Using a multiplexing technique called *dense-wave-division multiplexing* (D-WDM), the spacing between adjacent frequencies is considerably less. Unfortunately, there does not seem to be a standard definition of exactly what D-WDM means. Generally, optical systems carrying multiple optical signals spaced more than 200 GHz or 1.6 nm apart in the vicinity of 1550 nm are considered standard WDM. WDM systems carrying multiple optical signals in the vicinity of 1550 nm with less than 200 GHz of separation are considered D-WDM. Obviously, the more wavelengths used in a WDM system, the closer they are to each other and the denser the wavelength spectrum.

Light waves are comprised of many frequencies (wavelengths), and each frequency corresponds to a different color. Transmitters and receivers for optical fiber systems have been developed that transmit and receive only a specific color (i.e., a specific wavelength at a specific frequency with a fixed bandwidth). WDM is a process in which different sources of information (channels) are propagated down an optical fiber on different wavelengths where the different wavelengths do not interfere with each other. In essence, each wavelength adds an optical lane to the transmission superhighway, and the more lanes there are, the more traffic (voice, data, video, and so on) can be carried on a single optical fiber cable. In contrast, conventional optical fiber systems have only one channel per cable, which is used to carry information over a relatively narrow bandwidth. A Bell Laboratories



(a)



(b)

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

research team recently constructed a D-WDM transmitter using a single femtosecond, erbium-doped fiber-ring laser that can simultaneously carry 206 digitally modulated wavelengths of color over a single optical fiber cable. Each wavelength (channel) has a bit rate of 36.7 Mbps with a channel spacing of approximately 36 MHz.

Figure 6-21a shows the wavelength spectrum for a WDM system using six wavelengths, each modulated with equal-bandwidth information signals. Figure 6-21b shows how the output wavelengths from six lasers are combined (multiplexed) and then propagated over a single optical cable before being separated (demultiplexed) at the receiver with wavelength selective couplers. Although it has been proven that a single, ultrafast light source can generate hundreds of individual communications channels, standard WDM communications systems are generally limited to between 2 and 16 channels.

WDM enhances optical fiber performance by adding channels to existing cables. Each wavelength added corresponds to adding a different channel with its own information source and transmission bit rate. Thus, WDM can extend the information-carrying capacity of a fiber to hundreds of gigabits per second or higher.