

CHAPTER 8

MULTIPLEXING

8.1 Frequency-Division Multiplexing

- Characteristics
- Analog Carrier Systems
- Wavelength Division Multiplexing

8.2 Synchronous Time-Division Multiplexing

- Characteristics
- TDM Link Control
- Digital Carrier Systems
- SONET/SDH

8.3 Cable Modems

8.4 Asymmetric Digital Subscriber Line

- ADSL Design
- Discrete Multitone
- Broadband Access Configuration

8.5 xDSL

- High Data Rate Digital Subscriber Line
- Single-Line Digital Subscriber Line
- Very High Data Rate Digital Subscriber Line

8.6 Multiple Channel Access

- Frequency-Division Duplex (FDD)
- Time-Division Duplex (TDD)
- Frequency-Division Multiple Access (FDMA)
- Time-Division Multiple Access (TDMA)

8.7 Recommended Reading and Animations



8.8 Key Terms, Review Questions, and Problems

LEARNING OBJECTIVES

After reading this chapter, you should be able to:

- ◆ Explain the need for transmission efficiency and list the two major approaches used to achieve efficiency.
- ◆ Discuss the use of frequency-division multiplexing in voice networks.
- ◆ Describe the use of multiplexing in digital carrier systems.
- ◆ Discuss the T-1 service and describe its importance and the applications that are using it.
- ◆ Discuss the SONET standard and its significance for wide area networking.

In Chapter 7, we described efficient techniques for utilizing a data link under heavy load. Specifically, with two devices connected by a point-to-point link, it is generally desirable to have multiple frames outstanding so that the data link does not become a bottleneck between the stations. Now consider the opposite problem. Typically, two communicating stations will not utilize the full capacity of a data link. For efficiency, it should be possible to share that capacity. A generic term for such sharing is **multiplexing**.

A common application of multiplexing is in long-haul communications. Trunks on long-haul networks are high-capacity fiber, coaxial, or microwave links. These links can carry large numbers of voice and data transmissions simultaneously using multiplexing.

Figure 8.1 depicts the multiplexing function in its simplest form. There are n inputs to a multiplexer. The multiplexer is connected by a single data link to a **demultiplexer**. The link is able to carry n separate channels of data. The multiplexer combines (multiplexes) data from the n input lines and transmits over a higher-capacity data link. The demultiplexer accepts the multiplexed

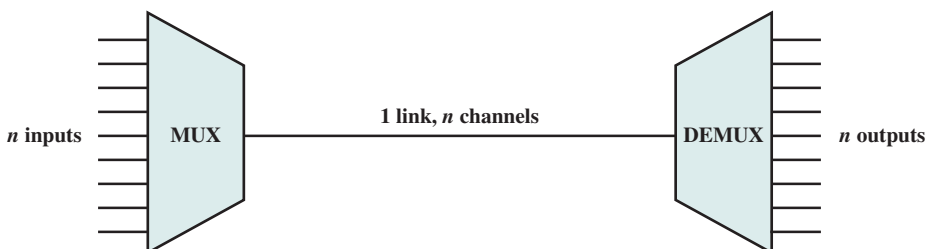


Figure 8.1 Multiplexing

data stream, separates (demultiplexes) the data according to channel, and delivers data to the appropriate output lines.

The widespread use of multiplexing in data communications can be explained by the following:

- The higher the data rate, the more cost-effective the transmission facility. That is, for a given application and over a given distance, the cost per kbps declines with an increase in the data rate of the transmission facility. Similarly, the cost of transmission and receiving equipment, per kbps, declines with increasing data rate.
- Most individual data communicating devices require relatively modest data rate support. For example, for many terminal and personal computer applications that do not involve Web access or intensive graphics, a data rate of between 9600 bps and 64 kbps is generally adequate.

The preceding statements were phrased in terms of data communicating devices. Similar statements apply to voice communications. That is, the greater the capacity of a transmission facility, in terms of voice channels, the less the cost per individual voice channel, and the capacity required for a single voice channel is modest.

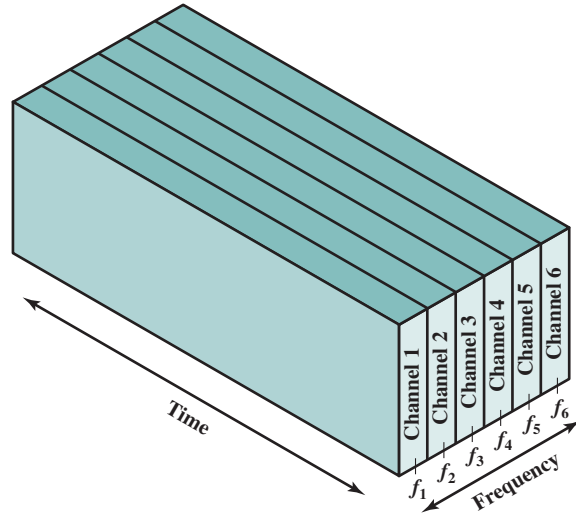
This chapter presents two types of multiplexing techniques. The first, frequency-division multiplexing (FDM), is the most heavily used and is familiar to anyone who has ever used a radio or television set. The second is a particular case of time-division multiplexing (TDM) known as synchronous TDM. This is commonly used for multiplexing digitized voice streams and data streams. We then look at two schemes that combine both types of multiplexing, cable modems and digital subscriber lines.

8.1 FREQUENCY-DIVISION MULTIPLEXING

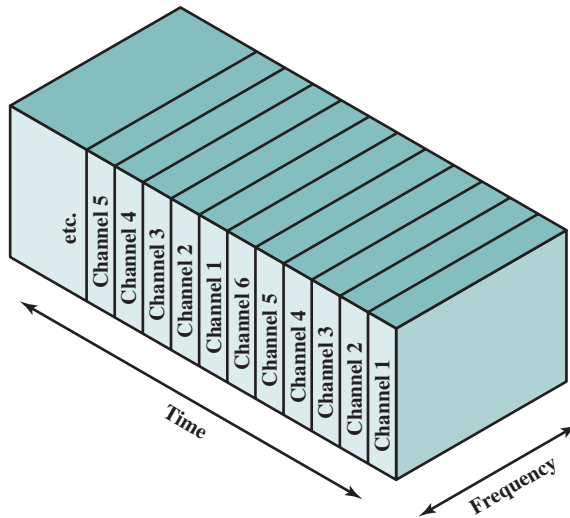
Characteristics

FDM is possible when the useful bandwidth of the transmission medium exceeds the required bandwidth of signals to be transmitted. A number of signals can be carried simultaneously if each signal is modulated onto a different carrier frequency and the carrier frequencies are sufficiently separated that the bandwidths of the signals do not significantly overlap. A general case of FDM is shown in Figure 8.2a. Six signal sources are fed into a multiplexer, which modulates each signal onto a different frequency (f_1, \dots, f_6). Each modulated signal requires a certain bandwidth centered on its carrier frequency, referred to as a **channel**. To prevent interference, the channels are separated by guard bands, which are unused portions of the spectrum.

The composite signal transmitted across the medium is analog. Note, however, that the input signals may be either digital or analog. In the case of digital input, the input signals must be passed through modems to be converted to analog. In either case, each input analog signal must then be modulated to move it to the appropriate frequency band.



(a) Frequency-division multiplexing



(b) Time-division multiplexing

Figure 8.2 FDM and TDM

A generic depiction of an FDM system is shown in Figure 8.3. A number of analog or digital signals $[m_i(t), i = 1, n]$ are to be multiplexed onto the same transmission medium. Each signal $m_i(t)$ is modulated onto a carrier f_i ; because multiple carriers are to be used, each is referred to as a **subcarrier**. Any type of modulation may be used. The resulting analog, modulated signals are then summed to produce a composite **baseband**¹ signal $m_b(t)$. Figure 8.3b shows the result. The spectrum of

¹The term *baseband* is used to designate the band of frequencies of the signal delivered by the source and potentially used as a modulating signal. Typically, the spectrum of a baseband signal is significant in a band that includes or is in the vicinity of $f=0$.

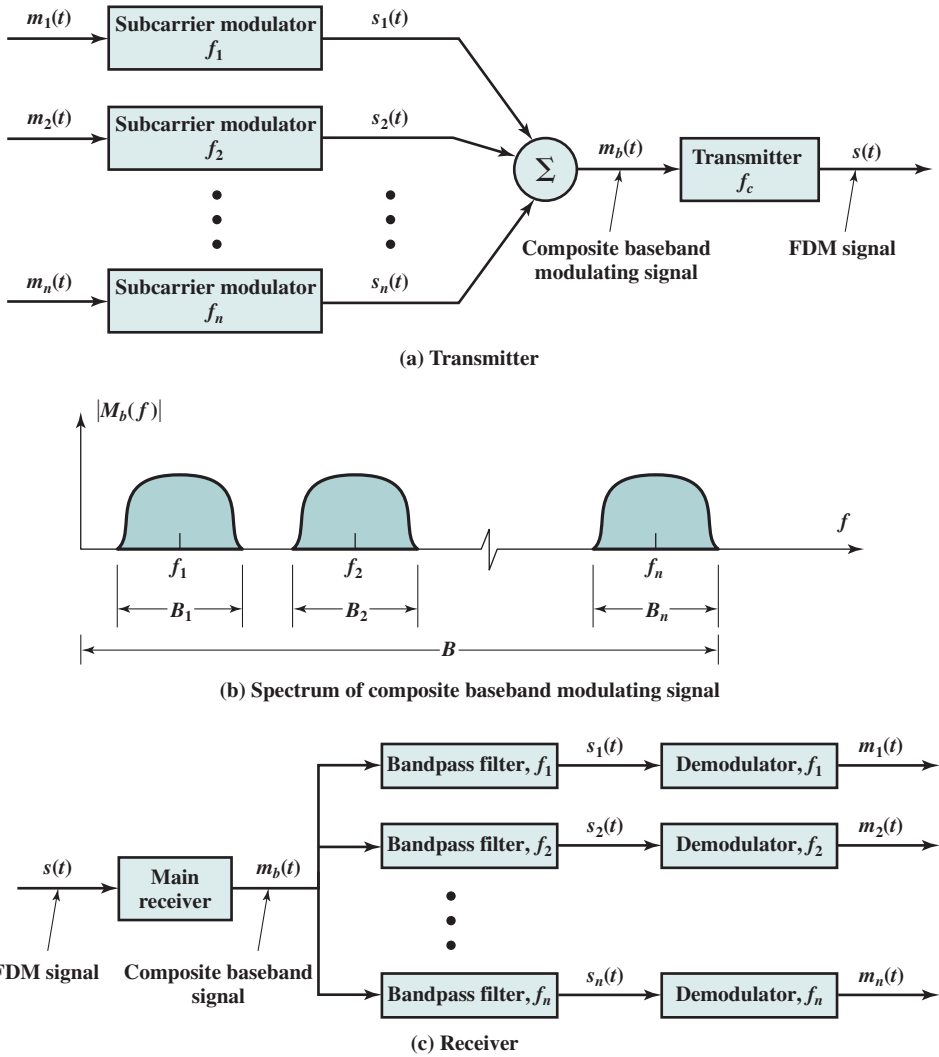


Figure 8.3 FDM System

signal $m_i(t)$ is shifted to be centered on f_i . For this scheme to work, f_i must be chosen so that the bandwidths of the various signals do not significantly overlap. Otherwise, it will be impossible to recover the original signals.

The composite signal may then be shifted as a whole to another carrier frequency by an additional modulation step. We will see examples of this later. This second modulation step need not use the same modulation technique as the first.

The FDM signal $s(t)$ has a total bandwidth B , where $B > \sum_{i=1}^n B_i$. This analog signal may be transmitted over a suitable medium. At the receiving end, the FDM signal is demodulated to retrieve $m_b(t)$, which is then passed through n bandpass filters, each filter centered on f_i and having a bandwidth B_i , for $1 \leq i \leq n$. In

this way, the signal is again split into its component parts. Each component is then demodulated to recover the original signal.

EXAMPLE 8.1 Let us consider a simple example of transmitting three voice signals simultaneously over a medium. As was mentioned, the bandwidth of a voice signal is generally taken to be 4 kHz, with an effective spectrum of 300 to 3400 Hz (Figure 8.4a). If such a signal is used to amplitude-modulate a 64-kHz carrier, the spectrum of Figure 8.4b results. The modulated signal has a bandwidth of 8 kHz, extending from 60 to 68 kHz. To make efficient use of bandwidth, we elect to transmit only the lower sideband. If three voice signals are used to modulate carriers at 64 kHz, 68 kHz, and 72 kHz, and only the lower sideband of each is taken, the spectrum of Figure 8.4c results.

Figure 8.4 points out two problems that an FDM system must cope with. The first is crosstalk, which may occur if the spectra of adjacent component signals overlap significantly. In the case of voice signals, with an effective bandwidth of

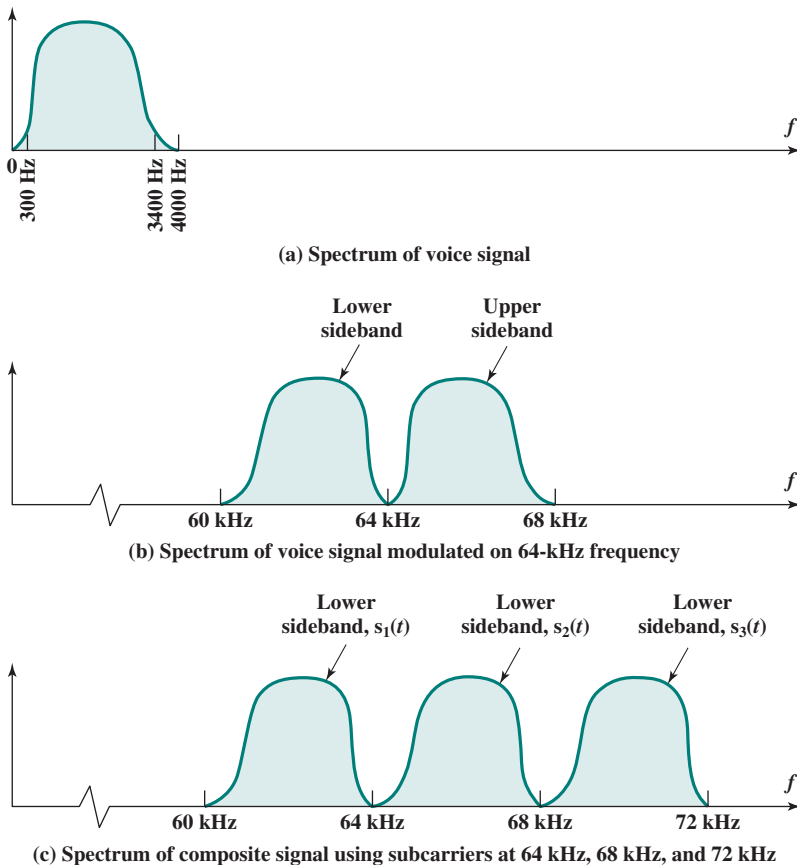


Figure 8.4 FDM of Three Voiceband Signals

only 3100 Hz (300 to 3400), a 4-kHz bandwidth is adequate. The spectra of signals produced by modems for voiceband transmission also fit well in this bandwidth. Another potential problem is intermodulation noise, which was discussed in Chapter 3. On a long link, the nonlinear effects of amplifiers on a signal in one channel could produce frequency components in other channels.

Analog Carrier Systems

The long-distance carrier system provided in the United States and throughout the world is designed to transmit voiceband signals over high-capacity transmission links, such as coaxial cable and microwave systems. The earliest, and still a very common, technique for utilizing high-capacity links is FDM. In the United States, AT&T has designated a hierarchy of FDM schemes to accommodate transmission systems of various capacities. A similar, but unfortunately not identical, system has been adopted internationally under the auspices of ITU-T (Table 8.1).

At the first level of the AT&T hierarchy, 12 voice channels are combined to produce a group signal with a bandwidth of $12 \times 4 \text{ kHz} = 48 \text{ kHz}$, in the range 60 to 108 kHz. The signals are produced in a fashion similar to that described previously, using subcarrier frequencies of from 64 to 108 kHz in increments of 4 kHz. The next basic building block is the 60-channel supergroup, which is formed by frequency-division multiplexing five group signals. At this step, each group is treated as a single signal with a 48-kHz bandwidth and is modulated by a subcarrier. The subcarriers have frequencies from 420 to 612 kHz in increments of 48 kHz. The resulting signal occupies 312 to 552 kHz.

There are several variations to supergroup formation. Each of the five inputs to the supergroup multiplexer may be a group channel containing 12 multiplexed voice signals. In addition, any signal up to 48 kHz wide whose bandwidth is contained within 60 to 108 kHz may be used as input to the supergroup multiplexer. As another variation, it is possible to combine 60 voiceband channels into a supergroup. This may reduce multiplexing costs where an interface with existing group multiplexer is not required.

Table 8.1 North American and International FDM Carrier Standards

Number of Voice Channels	Bandwidth	Spectrum	AT&T	ITU-T
12	48 kHz	60–108 kHz	Group	Group
60	240 kHz	312–552 kHz	Supergroup	Supergroup
300	1.232 MHz	812–2044 kHz		Mastergroup
600	2.52 MHz	564–3084 kHz	Mastergroup	
900	3.872 MHz	8.516–12.388 MHz		Supermaster group
$N \times 600$			Mastergroup multiplex	
3,600	16.984 MHz	0.564–17.548 MHz	Jumbogroup	
10,800	57.442 MHz	3.124–60.566 MHz	Jumbogroup multiplex	

The next level of the hierarchy is the mastergroup, which combines 10 supergroup inputs. Again, any signal with a bandwidth of 240 kHz in the range 312 to 552 kHz can serve as input to the mastergroup multiplexer. The mastergroup has a bandwidth of 2.52 MHz and can support 600 voice frequency (VF) channels. Higher-level multiplexing is defined above the mastergroup, as shown in Table 8.1.

Note that the original voice or data signal may be modulated many times. For example, a data signal may be encoded using QPSK (quadrature phase shift keying) to form an analog voice signal. This signal could then be used to modulate a 76-kHz carrier to form a component of a group signal. This group signal could then be used to modulate a 516-kHz carrier to form a component of a supergroup signal. Each stage can distort the original data; this is so, for example, if the modulator/multiplexer contains nonlinearities or introduces noise.

Wavelength Division Multiplexing

The true potential of optical fiber is fully exploited when multiple beams of light at different frequencies are transmitted on the same fiber. This is a form of frequency-division multiplexing but is commonly called **wavelength division multiplexing (WDM)**. With WDM, the light streaming through the fiber consists of many colors, or wavelengths, each carrying a separate channel of data. In 1997, a landmark was reached when Bell Laboratories was able to demonstrate a WDM system with 100 beams each operating at 10 Gbps, for a total data rate of 1 trillion bits per second (also referred to as 1 terabit per second or 1 Tbps). Commercial systems with 160 channels of 10 Gbps are now available. In a lab environment, Alcatel has carried 256 channels at 39.8 Gbps each, a total of 10.1 Tbps, over a 100-km span.

A typical WDM system has the same general architecture as other FDM systems. A number of sources generate a laser beam at different wavelengths. These are sent to a multiplexer, which consolidates the sources for transmission over a single fiber line. Optical amplifiers, typically spaced tens of kilometers apart, amplify all of the wavelengths simultaneously. Finally, the composite signal arrives at a demultiplexer, where the component channels are separated and sent to receivers at the destination point (Figure 8.5).

Most WDM systems operate in the 1550-nm range. In early systems, 200 GHz was allocated to each channel, but today most WDM systems use 50-GHz spacing.

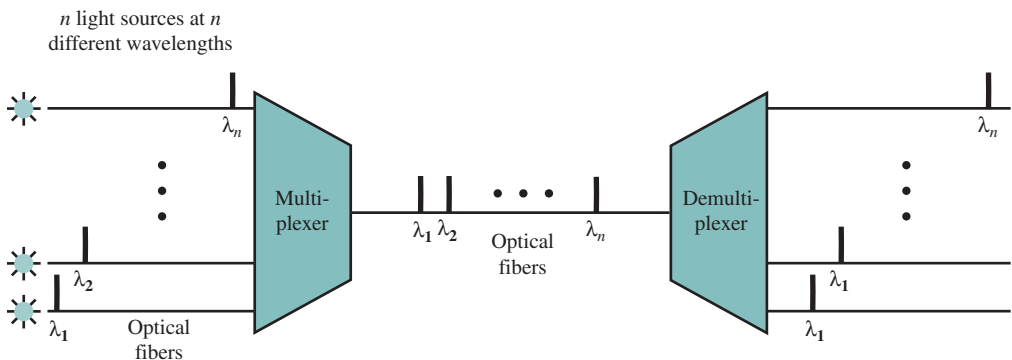


Figure 8.5 Wavelength Division Multiplexing

Table 8.2 ITU WDM Channel Spacing (G.692)

Frequency (THz)	Wavelength in Vacuum (nm)	50 GHz	100 GHz	200 GHz
196.10	1528.77	X	X	X
196.05	1529.16	X		
196.00	1529.55	X	X	
195.95	1529.94	X		
195.90	1530.33	X	X	X
195.85	1530.72	X		
195.80	1531.12	X	X	
195.75	1531.51	X		
195.70	1531.90	X	X	X
195.65	1532.29	X		
195.60	1532.68	X	X	
...	...			
192.10	1560.61	X	X	X

The channel spacing defined in ITU-T G.692, which accommodates 80 50-GHz channels, is summarized in Table 8.2.

The term **dense wavelength division multiplexing (DWDM)** is often seen in the literature. There is no official or standard definition of this term. The term connotes the use of more channels, more closely spaced, than ordinary WDM. In general, a channel spacing of 200 GHz or less could be considered dense.

8.2 SYNCHRONOUS TIME-DIVISION MULTIPLEXING

Characteristics

Synchronous time-division multiplexing is possible when the achievable data rate (sometimes, unfortunately, called bandwidth) of the medium exceeds the data rate of digital signals to be transmitted. Multiple digital signals (or analog signals carrying digital data) can be carried on a single transmission path by interleaving portions of each signal in time. The interleaving can be at the bit level or in blocks of bytes or larger quantities. For example, the multiplexer in Figure 8.2b has six inputs that might each be, say, 1 Mbps. A single line with a capacity of at least 6 Mbps (plus overhead capacity) could accommodate all six sources.

A generic depiction of a synchronous TDM system is provided in Figure 8.6. A number of signals $[m_i(t), i = 1, n]$ are to be multiplexed onto the same transmission medium. The signals carry digital data and are generally digital signals. The incoming data from each source are briefly buffered. Each buffer is typically one bit or one character in length. The buffers are scanned sequentially to form a composite digital data stream $m_c(t)$. The scan operation is sufficiently rapid so that each buffer is emptied before more data can arrive. Thus, the data rate of $m_c(t)$ must at

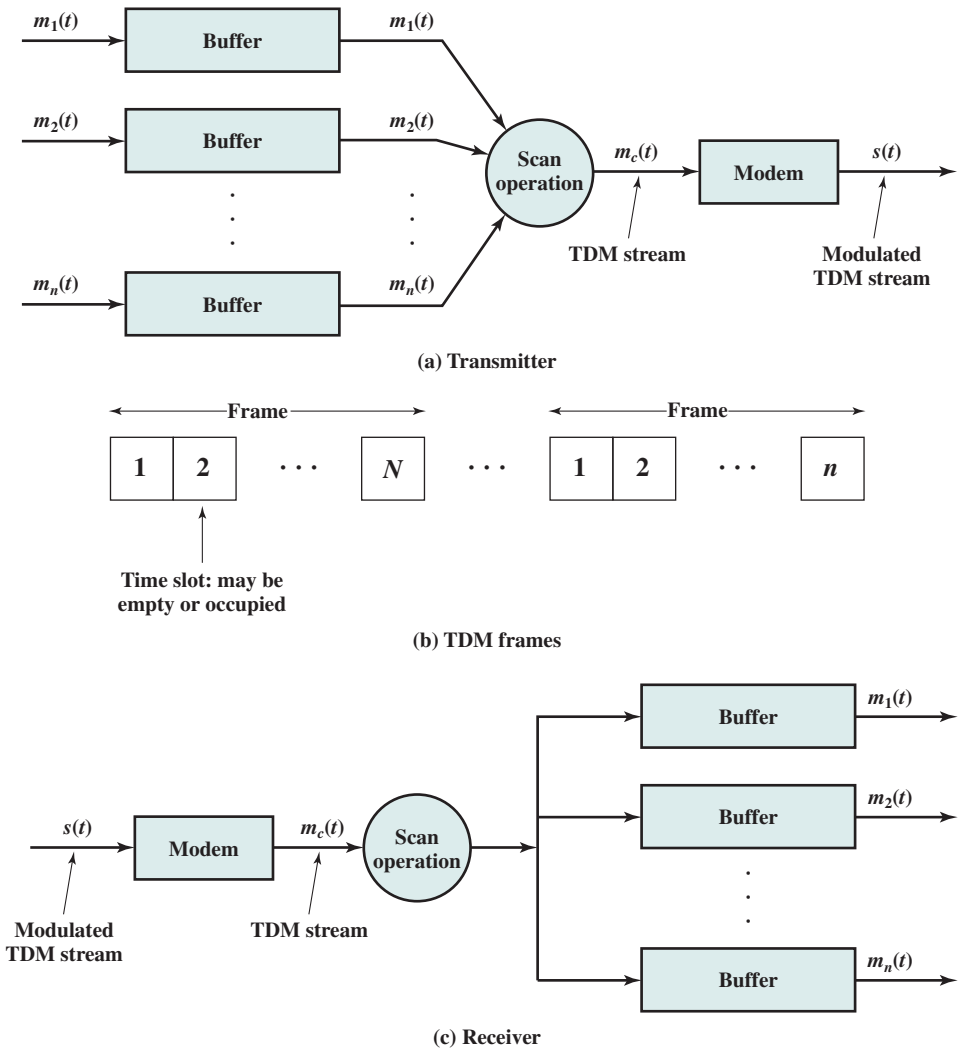


Figure 8.6 Synchronous TDM System

least equal the sum of the data rates of the $m_i(t)$. The digital signal $m_c(t)$ may be transmitted directly, or passed through a modem so that an analog signal is transmitted. In either case, transmission is typically synchronous.

The transmitted data may have a format something like Figure 8.6b. The data are organized into **frames**. Each frame contains a cycle of time slots. In each frame, one or more slots are dedicated to each data source. The sequence of slots dedicated to one source, from frame to frame, is called a **channel**. The slot length equals the transmitter buffer length, typically a bit or a byte (character).

The byte-interleaving technique is used with asynchronous and synchronous sources. Each time slot contains one character of data. Typically, the start and stop bits of each character are eliminated before transmission and reinserted by the receiver, thus improving efficiency. The bit-interleaving technique is used with

synchronous sources and may also be used with asynchronous sources. Each time slot contains just one bit.

At the receiver, the interleaved data are demultiplexed and routed to the appropriate destination buffer. For each input source $m_i(t)$, there is an identical output destination that will receive the output data at the same rate at which it was generated.

Synchronous TDM is called synchronous not because synchronous transmission is used, but because the time slots are preassigned to sources and fixed. The time slots for each source are transmitted whether or not the source has data to send. This is, of course, also the case with FDM. In both cases, capacity is wasted to achieve simplicity of implementation. Even when fixed assignment is used, however, it is possible for a synchronous TDM device to handle sources of different data rates. For example, the slowest input device could be assigned one slot per cycle, while faster devices are assigned multiple slots per cycle.

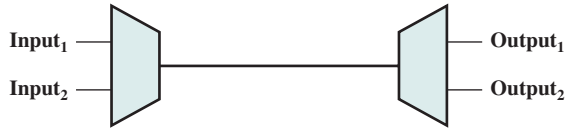
An alternative to synchronous TDM is **statistical TDM**. The statistical multiplexer dynamically allocates time slots on demand. As with a synchronous TDM, the statistical multiplexer has a number of I/O lines on one side and a higher speed multiplexed line on the other. Each I/O line has a buffer associated with it. In the case of the statistical multiplexer, there are n I/O lines, but only k , where $k < n$, time slots available on the TDM frame. For input, the function of the multiplexer is to scan the input buffers, collecting data until a frame is filled, and then send the frame. On output, the multiplexer receives a frame and distributes the slots of data to the appropriate output buffers. Packet switching is, in effect, a form of statistical TDM. For a further discussion of statistical TDM, see Appendix I.

TDM Link Control

The reader will note that the transmitted data stream depicted in Figure 8.6b does not contain the headers and trailers that we have come to associate with synchronous transmission. The reason is that the control mechanisms provided by a data link protocol are not needed. It is instructive to ponder this point, and we do so by considering two key data link control mechanisms: flow control and error control. It should be clear that, as far as the multiplexer and demultiplexer (Figure 8.1) are concerned, flow control is not needed. The data rate on the multiplexed line is fixed, and the multiplexer and demultiplexer are designed to operate at that rate. But suppose that one of the individual output lines attaches to a device that is temporarily unable to accept data. Should the transmission of TDM frames cease? Clearly not, because the remaining output lines are expecting to receive data at predetermined times. The solution is for the saturated output device to cause the flow of data from the corresponding input device to cease. Thus, for a while, the channel in question will carry empty slots, but the frames as a whole will maintain the same transmission rate.

The reasoning for error control is the same. It would not do to request retransmission of an entire TDM frame because an error occurs on one channel. The devices using the other channels do not want a retransmission nor would they know that a retransmission has been requested by some other device on another channel. Again, the solution is to apply error control on a per-channel basis.

Flow control and error control can be provided on a per-channel basis by using a data link control protocol such as HDLC on a per-channel basis.



(a) Configuration

Input₁..... F₁ f₁ f₁ d₁ d₁ d₁ C₁ A₁ F₁ f₁ f₁ d₁ d₁ d₁ C₁ A₁ F₁
 Input₂... F₂ f₂ f₂ d₂ d₂ d₂ d₂ C₂ A₂ F₂ f₂ f₂ d₂ d₂ d₂ d₂ C₂ A₂ F₂

(b) Input data streams

... f₂ F₁ d₂ f₁ d₂ f₁ d₂ d₁ d₂ d₁ C₂ d₁ A₂ C₁ F₂ A₁ f₂ F₁ f₂ f₁ d₂ f₁ d₂ d₁ d₂ d₁ d₂ d₁ C₂ C₁ A₂ A₁ F₂ F₁

(c) Multiplexed data stream

Legend: F = flag field d = one octet of data field
 A = address field f = one octet of FCS field
 C = control field

Figure 8.7 Use of Data Link Control on TDM Channels

EXAMPLE 8.2 Figure 8.7 provides a simplified example. We assume two data sources, each using HDLC. One is transmitting a stream of HDLC frames containing three octets of data each, and the other is transmitting HDLC frames containing four octets of data. For clarity, we assume that character-interleaved multiplexing is used, although bit interleaving is more typical. Notice what is happening. The octets of the HDLC frames from the two sources are shuffled together for transmission over the multiplexed line. The reader may initially be uncomfortable with this diagram, because the HDLC frames have lost their integrity in some sense. For example, each frame check sequence (FCS) on the line applies to a disjointed set of bits. Even the FCS is not in one piece. However, the pieces are reassembled correctly before they are seen by the device on the other end of the HDLC protocol. In this sense, the multiplexing/demultiplexing operation is transparent to the attached stations; to each communicating pair of stations, it appears that they have a dedicated link.

One refinement is needed in Figure 8.7. Both ends of the line need to be a combination multiplexer/demultiplexer with a full-duplex line in between. Then each channel consists of two sets of slots, one traveling in each direction. The individual devices attached at each end can, in pairs, use HDLC to control their own channel. The multiplexer/demultiplexers need not be concerned with these matters.

FRAMING We have seen that a link control protocol is not needed to manage the overall TDM link. There is, however, a basic requirement for framing. Because we are not providing flag or SYNC characters to bracket TDM frames, some means is needed to assure frame synchronization. It is clearly important to maintain framing synchronization because, if the source and destination are out of step, data on all channels are lost.

Perhaps the most common mechanism for framing is known as added-digit framing. In this scheme, typically, one control bit is added to each TDM frame. An

identifiable pattern of bits, from frame to frame, is used as a “control channel.” A typical example is the alternating bit pattern, 101010... This is a pattern unlikely to be sustained on a data channel. Thus, to synchronize, a receiver compares the incoming bits of one frame position to the expected pattern. If the pattern does not match, successive bit positions are searched until the pattern persists over multiple frames. Once framing synchronization is established, the receiver continues to monitor the framing bit channel. If the pattern breaks down, the receiver must again enter a framing search mode.

PULSE STUFFING Perhaps the most difficult problem in the design of a synchronous time-division multiplexer is that of synchronizing the various data sources. If each source has a separate clock, any variation among clocks could cause loss of synchronization. Also, in some cases, the data rates of the input data streams are not related by a simple rational number. For both these problems, a technique known as **pulse stuffing** is an effective remedy. With pulse stuffing, the outgoing data rate of the multiplexer, excluding framing bits, is higher than the sum of the maximum instantaneous incoming rates. The extra capacity is used by stuffing extra dummy bits or pulses into each incoming signal until its rate is raised to that of a locally generated clock signal. The stuffed pulses are inserted at fixed locations in the multiplexer frame format so that they may be identified and removed at the demultiplexer.

EXAMPLE 8.3 An example, from [COUC13], illustrates the use of synchronous TDM to multiplex digital and analog sources (Figure 8.8). Consider that there are 11 sources to be multiplexed on a single link:

Source 1:	Analog, 2-kHz bandwidth
Source 2:	Analog, 4-kHz bandwidth
Source 3:	Analog, 2-kHz bandwidth
Sources 4–11:	Digital, 7200 bps synchronous

As a first step, the analog sources are converted to digital using pulse code modulation (PCM). Recall from Chapter 5 that PCM is based on the sampling theorem, which dictates that a signal be sampled at a rate equal to twice its bandwidth. Thus, the required sampling rate is 4000 samples per second for sources 1 and 3, and 8000 samples per second for source 2. These samples, which are analog (PAM), must then be quantized or digitized. Let us assume that 4 bits are used for each analog sample. For convenience, these three sources will be multiplexed first, as a unit. At a scan rate of 4 kHz, one PAM sample each is taken from sources 1 and 3, and two PAM samples are taken from source 2 per scan. These four samples are interleaved and converted to 4-bit PCM samples. Thus, a total of 16 bits is generated at a rate of 4000 times per second, for a composite bit rate of 64 kbps.

For the digital sources, pulse stuffing is used to raise each source to a rate of 8 kbps, for an aggregate data rate of 64 kbps. A frame can consist of multiple cycles of 32 bits, each containing 16 PCM bits and two bits from each of the eight digital sources.

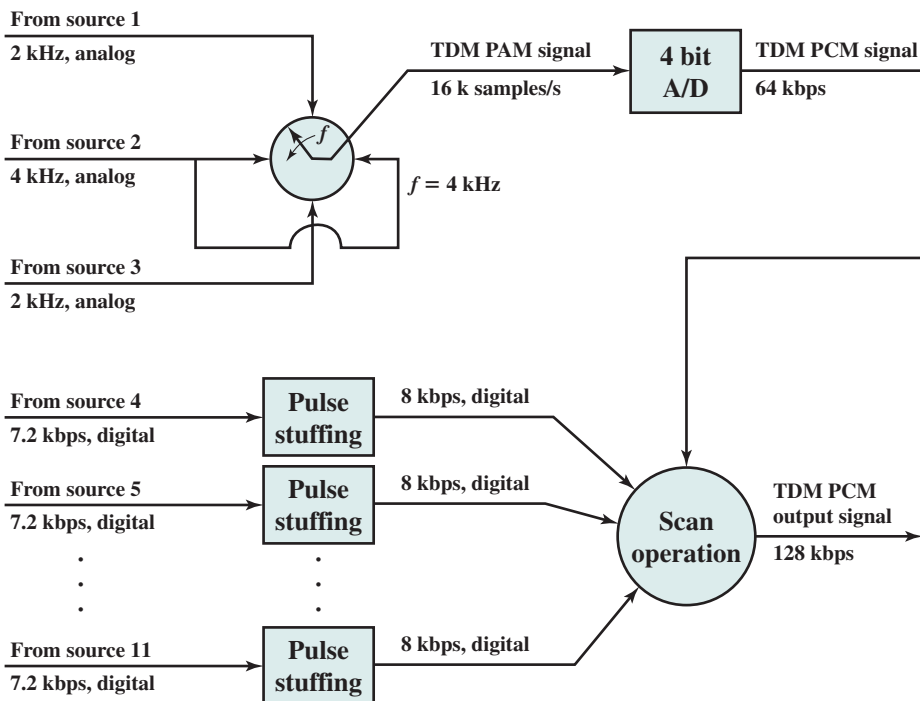


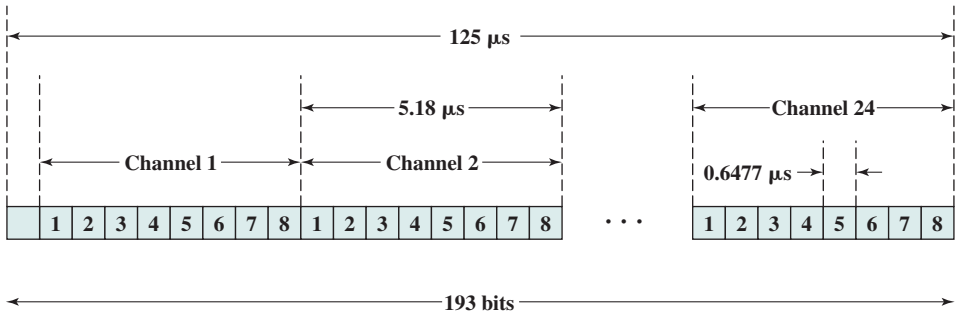
Figure 8.8 TDM of Analog and Digital Sources

Digital Carrier Systems

The long-distance carrier system provided in the United States and throughout the world was designed to transmit voice signals over high-capacity transmission links, such as optical fiber, coaxial cable, and microwave. Part of the evolution of these telecommunications networks to digital technology has been the adoption of synchronous TDM transmission structures. In the United States, AT&T developed a hierarchy of TDM structures of various capacities; this structure is used in Canada and Japan as well as the United States. A similar, but unfortunately not identical, hierarchy has been adopted internationally under the auspices of ITU-T (Table 8.3).

Table 8.3 North American and International TDM Carrier Standards

North American			International (ITU-T)		
Designation	Number of Voice Channels	Data Rate (Mbps)	Level	Number of Voice Channels	Data Rate (Mbps)
DS-1	24	1.544	1	30	2.048
DS-1C	48	3.152	2	120	8.448
DS-2	96	6.312	3	480	34.368
DS-3	672	44.736	4	1920	139.264
DS-4	4032	274.176	5	7680	565.148



Notes:

1. The first bit is a framing bit, used for synchronization.
2. Voice channels:
 - 8-bit PCM used on five of six frames.
 - 7-bit PCM used on every sixth frame; bit 8 of each channel is a signaling bit.
3. Data channels:
 - Channel 24 is used for signaling only in some schemes.
 - Bits 1–7 used for 56-kbps service.
 - Bits 2–7 used for 9.6-kbps, 4.8-kbps, and 2.4-kbps service.

Figure 8.9 DS-1 Transmission Format

The basis of the TDM hierarchy (in North America and Japan) is the DS-1 transmission format (Figure 8.9), which multiplexes 24 channels. Each frame contains 8 bits per channel plus a framing bit for $24 \times 8 + 1 = 193$ bits. For voice transmission, the following rules apply. Each channel contains one word of digitized voice data. The original analog voice signal is digitized using pulse code modulation at a rate of 8000 samples per second. Therefore, each channel slot and hence each frame must repeat 8000 times per second. With a frame length of 193 bits, we have a data rate of $8000 \times 193 = 1.544$ Mbps. For five of every six frames, 8-bit PCM samples are used. For every sixth frame, each channel contains a 7-bit PCM word plus a *signaling bit*. The signaling bits form a stream for each voice channel that contains network control and routing information. For example, control signals are used to establish a connection or terminate a call.

The same DS-1 format is used to provide digital data service. For compatibility with voice, the same 1.544-Mbps data rate is used. In this case, 23 channels of data are provided. The twenty-fourth channel position is reserved for a special sync byte, which allows faster and more reliable reframing following a framing error. Within each channel, 7 bits per frame are used for data, with the eighth bit used to indicate whether the channel, for that frame, contains user data or system control data. With 7 bits per channel, and because each frame is repeated 8000 times per second, a data rate of 56 kbps can be provided per channel. Lower data rates are provided using a technique known as subrate multiplexing. For this technique, an additional bit is robbed from each channel to indicate which subrate multiplexing rate is being provided. This leaves a total capacity per channel of $6 \times 8000 = 48$ kbps. This capacity is used to multiplex five 9.6-kbps channels, ten 4.8-kbps channels, or twenty 2.4-kbps channels. For example, if channel 2 is used to provide 9.6-kbps service, then up to five data subchannels share this channel. The data for each subchannel appear as six bits in channel 2 every fifth frame.

Finally, the DS-1 format can be used to carry a mixture of voice and data channels. In this case, all 24 channels are utilized; no sync byte is provided.

Above the DS-1 data rate of 1.544 Mbps, higher-level multiplexing is achieved by interleaving bits from DS-1 inputs. For example, the DS-2 transmission system combines four DS-1 inputs into a 6.312-Mbps stream. Data from the four sources are interleaved 12 bits at a time. Note that $1.544 \times 4 = 6.176$ Mbps. The remaining capacity is used for framing and control bits.

SONET/SDH

SONET (Synchronous Optical Network) is an optical transmission interface originally proposed by BellCore and standardized by ANSI. A compatible version, referred to as Synchronous Digital Hierarchy (SDH), has been published by ITU-T in Recommendation G.707.² SONET is intended to provide a specification for taking advantage of the high-speed digital transmission capability of optical fiber.

SIGNAL HIERARCHY The SONET specification defines a hierarchy of standardized digital data rates (Table 8.4). The lowest level, referred to as STS-1 (Synchronous Transport Signal level 1) or OC-1 (Optical Carrier level 1),³ is 51.84 Mbps. This rate can be used to carry a single DS-3 signal or a group of lower-rate signals, such as DS1, DS1C, DS2, plus ITU-T rates (e.g., 2.048 Mbps).

Multiple STS-1 signals can be combined to form an STS- N signal. The signal is created by interleaving bytes from N STS-1 signals that are mutually synchronized.

For the ITU-T Synchronous Digital Hierarchy, the lowest rate is 155.52 Mbps, which is designated STM-1. This corresponds to SONET STS-3.

FRAME FORMAT The basic SONET building block is the STS-1 frame, which consists of 810 octets and is transmitted once every 125 μ s, for an overall data rate of 51.84 Mbps (Figure 8.10a). The frame can logically be viewed as a matrix of 9 rows of 90 octets each, with transmission being one row at a time, from left to right and top to bottom.

Table 8.4 SONET/SDH Signal Hierarchy

SONET Designation	ITU-T Designation	Data Rate	Payload Rate (Mbps)
STS-1/OC-1		51.84 Mbps	50.112 Mbps
STS-3/OC-3	STM-1	155.52 Mbps	150.336 Mbps
STS-12/OC-12	STM-4	622.08 Mbps	601.344 Mbps
STS-48/OC-48	STM-16	2.48832 Gbps	2.405376 Gbps
STS-192/OC-192	STM-64	9.95328 Gbps	9.621504 Gbps
STS-768	STM-256	39.81312 Gbps	38.486016 Gbps
STS-3072		159.25248 Gbps	153.944064 Gbps

²In what follows, we will use the term *SONET* to refer to both specifications. Where differences exist, these will be addressed.

³An OC- N rate is the optical equivalent of an STS- N electrical signal. End-user devices transmit and receive electrical signals; these must be converted to and from optical signals for transmission over optical fiber.

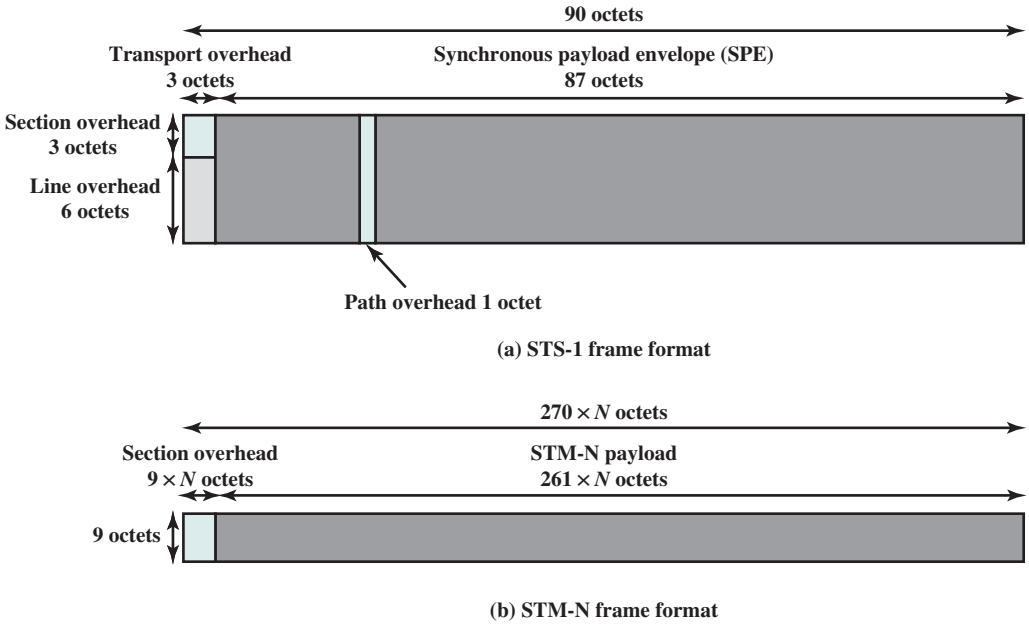


Figure 8.10 SONET/SDH Frame Formats

The first three columns (3 octets \times 9 rows = 27 octets) of the frame are devoted to overhead octets. Nine octets are devoted to section-related overhead and 18 octets are devoted to line overhead. Figure 8.11a shows the arrangement of overhead octets, and Table 8.5 defines the various fields.

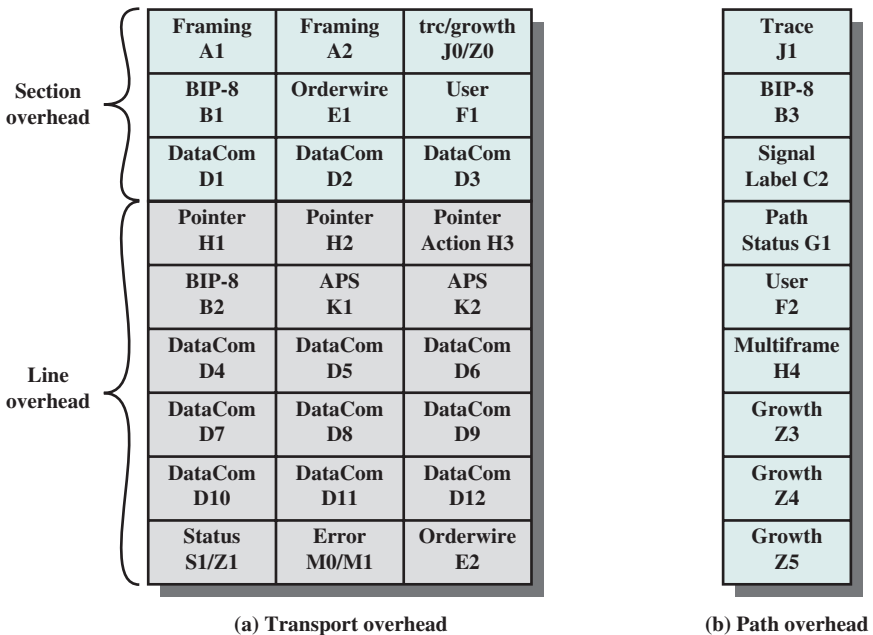


Figure 8.11 SONET STS-1 Overhead Octets

Table 8.5 STS-1 Overhead Bits

Section Overhead	
A1, A2:	Framing bytes = F6,28 hex; used to indicate the beginning of the frame.
J0/Z0:	Allows two connected sections to verify the connections between them by transmitting a 16-byte message. This message is transmitted in 16 consecutive frames with first byte (J0) carried in first frame, second byte in second frame, and so on (Z0).
B1:	Bit-interleaved parity byte providing even parity over previous STS- <i>N</i> frame after scrambling; the <i>i</i> th bit of this octet contains the even parity value calculated from the <i>i</i> th bit position of all octets in the previous frame.
E1:	Section level 64-kbps PCM orderwire; optional 64-kbps voice channel to be used between section terminating equipment, hubs, and remote terminals.
F1:	64-kbps channel set aside for user purposes.
D1-D3:	192-kbps data communications channel for alarms, maintenance, control, and administration between sections.
Line Overhead	
H1-H3:	Pointer bytes used in frame alignment and frequency adjustment of payload data.
B2:	Bit-interleaved parity for line level error monitoring.
K1, K2:	Two bytes allocated for signaling between line level automatic protection switching equipment; uses a bit-oriented protocol that provides for error protection and management of the SONET optical link.
D4-D12:	576-kbps data communications channel for alarms, maintenance, control, monitoring, and administration at the line level.
S1/Z1:	In the first STS-1 of an STS- <i>N</i> signal, used for transporting synchronization message (S1). Undefined in the second through <i>N</i> th STS-1 (Z1).
M0/M1:	Remote error indication in first STS-1 (M0) and third frames.
E2:	64-kbps PCM voice channel for line level orderwire.
Path Overhead	
J1:	64-kbps channel used to send repetitively a 64-octet fixed-length string so a receiving terminal can continuously verify the integrity of a path; the contents of the message are user programmable.
B3:	Bit-interleaved parity at the path level, calculated over all bits of the previous SPE.
C2:	STS path signal label to designate equipped versus unequipped STS signals. <i>Unequipped</i> means the line connection is complete but there is no path data to send. For equipped signals, the label can indicate the specific STS payload mapping that might be needed in receiving terminals to interpret the payloads.
G1:	Status byte sent from path terminating equipment back to path originating equipment to convey status of terminating equipment and path error performance.
F2:	64-kbps channel for path user.
H4:	Multiframe indicator for payloads needing frames that are longer than a single STS frame; multiframe indicators are used when packing lower rate channels (virtual tributaries) into the SPE.
Z3-Z5:	Reserved for future use.

The remainder of the frame is payload. The payload includes a column of path overhead, which is not necessarily in the first available column position; the line overhead contains a pointer that indicates where the path overhead starts. Figure 8.11b shows the arrangement of path overhead octets, and Table 8.5 defines these.

Figure 8.10b shows the general format for higher-rate frames, using the ITU-T designation.

8.3 CABLE MODEM

A **cable modem** is a device that allows a user to access the Internet and other online services through a cable TV network. To support data transfer to and from a cable modem, a cable TV provider dedicates two 6-MHz channels, one for transmission in each direction. Each channel is shared by a number of subscribers, and so some scheme is needed for allocating capacity on each channel for transmission. Typically, a form of statistical TDM is used, as illustrated in Figure 8.12. In the downstream direction, cable **headend** to subscriber, a cable scheduler delivers data in the form of small packets. Because the channel is shared by a number of subscribers, if more than one subscriber is active, each subscriber gets only a fraction of the downstream capacity. An individual cable modem subscriber may experience access speeds from 500 kbps to 1.5 Mbps or more, depending on the network architecture and traffic load. The downstream direction is also used to grant time slots to subscribers. When a subscriber has data to transmit, it must first request time slots on the shared upstream channel. Each subscriber is given dedicated time slots for this request purpose. The headend scheduler responds to a request packet by sending back an assignment of future time slots to be used by this subscriber. Thus, a number of subscribers can share the same upstream channel without conflict.

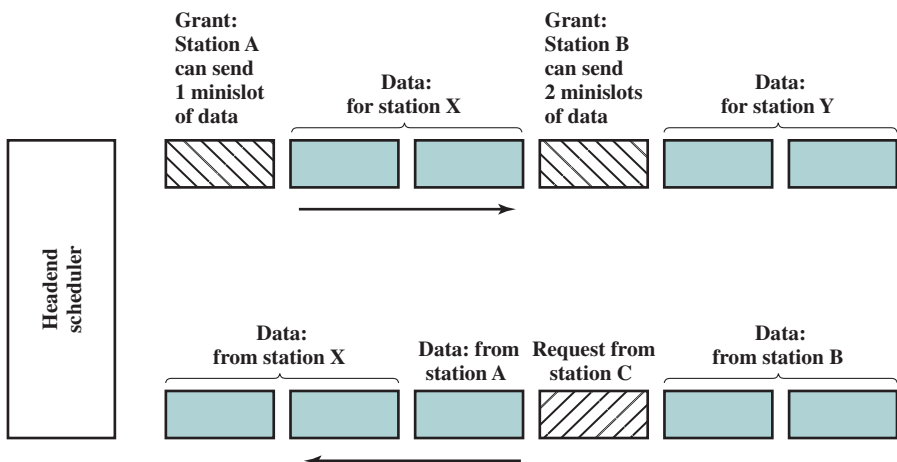


Figure 8.12 Cable Modem Scheme

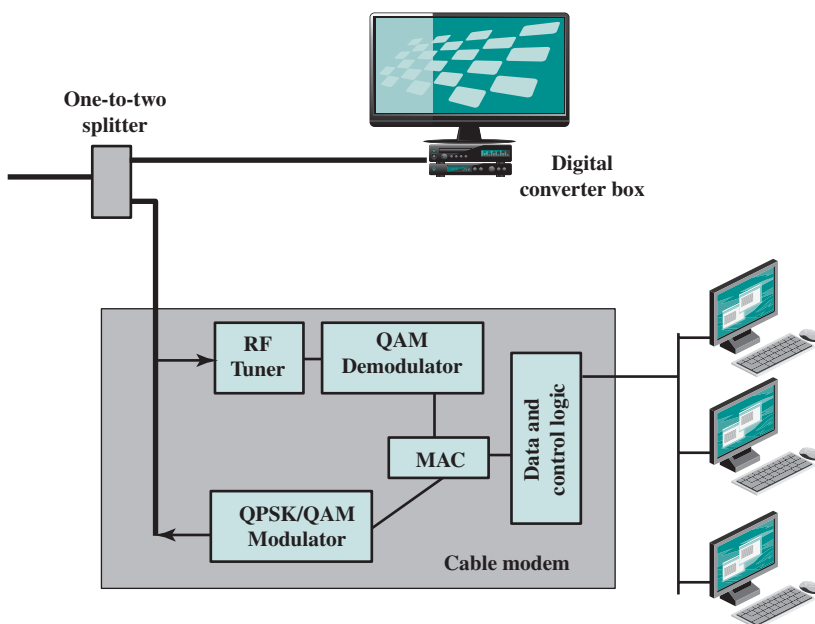


Figure 8.13 Cable Modem Configuration

To support both cable television programming and data channels, the cable spectrum is divided into three ranges, each of which is further divided into 6-MHz channels. In North America, the spectrum division is as follows:

- User-to-network data (upstream): 5–40 MHz
- Television delivery (downstream): 50–550 MHz
- Network to user data (downstream): 550–750 MHz

Figure 8.13 shows a typical cable modem configuration at a residential or office location. At the interface to the external cable, a one-to-two splitter enables the subscriber to continue to receive cable television service through numerous FDM 6-MHz channels, while simultaneously supporting data channels to one or more computers in a local area network. The inbound channel first goes through a radio frequency (RF) tuner that selects and demodulates the data channel down to a spectrum of 0 to 6 MHz. This channel provides a data stream encoded using 64-QAM (quadrature amplitude modulation) or 256-QAM. The QAM demodulator extracts the encoded data stream and converts it to a digital signal that it passes to the media access control (MAC) module. In the outbound direction, a data stream is modulated using either QPSK or 16-QAM.

8.4 ASYMMETRIC DIGITAL SUBSCRIBER LINE

In the implementation and deployment of a high-speed wide area public digital network, the most challenging part is the link between subscriber and network: the digital subscriber line. With billions of potential endpoints worldwide, the prospect

of installing new cable for each new customer is daunting. Instead, network designers have sought ways of exploiting the installed base of twisted-pair wire that links virtually all residential and business customers to telephone networks. These links were installed to carry voice-grade signals in a bandwidth from 0 to 4 kHz. However, the wires are capable of transmitting signals over a far broader spectrum—1 MHz or more.

ADSL is the most widely publicized of a family of new modem technologies designed to provide high-speed digital data transmission over ordinary telephone wire. ADSL is now being offered by a number of carriers and is defined in an ANSI standard. In this following section, we first look at the overall design of ADSL and then examine the key underlying technology, known as DMT.

ADSL Design

The term *asymmetric* refers to the fact that ADSL provides more capacity downstream (from the carrier's central office to the customer's site) than upstream (from customer to carrier). ADSL was originally targeted at the expected need for video on demand and related services. This application has not materialized. However, since the introduction of ADSL technology, the demand for high-speed access to the Internet has grown. Typically, the user requires far higher capacity for downstream than for upstream transmission. Most user transmissions are in the form of keyboard strokes or transmission of short e-mail messages, whereas incoming traffic, especially Web traffic, can involve large amounts of data and include images or even video. Thus, ADSL provides a perfect fit for the Internet requirement.

ADSL uses frequency-division multiplexing in a novel way to exploit the 1-MHz capacity of twisted pair. There are three elements of the ADSL strategy (Figure 8.14):

- Reserve lowest 25 kHz for voice, known as POTS (plain old telephone service). The voice is carried only in the 0 to 4 kHz band; the additional bandwidth is to prevent crosstalk between the voice and data channels.
- Use either echo cancellation⁴ or FDM to allocate two bands, a smaller upstream band and a larger downstream band.
- Use FDM within the upstream and downstream bands. In this case, a single bit stream is split into multiple parallel bit streams and each portion is carried in a separate frequency band.

When echo cancellation is used, the entire frequency band for the upstream channel overlaps the lower portion of the downstream channel. This has two advantages compared to the use of distinct frequency bands for upstream and downstream.

⁴Echo cancellation is a signal-processing technique that allows transmission of digital signals in both directions on a single transmission line simultaneously. In essence, a transmitter must subtract the echo of its own transmission from the incoming signal to recover the signal sent by the other side.

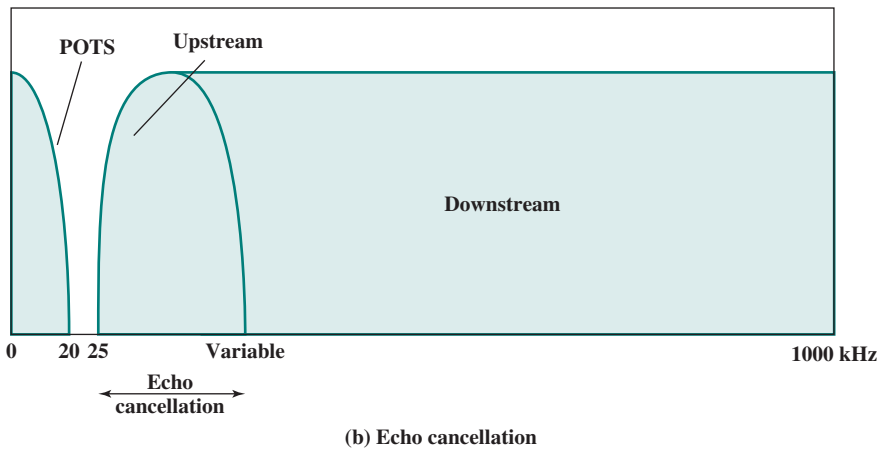
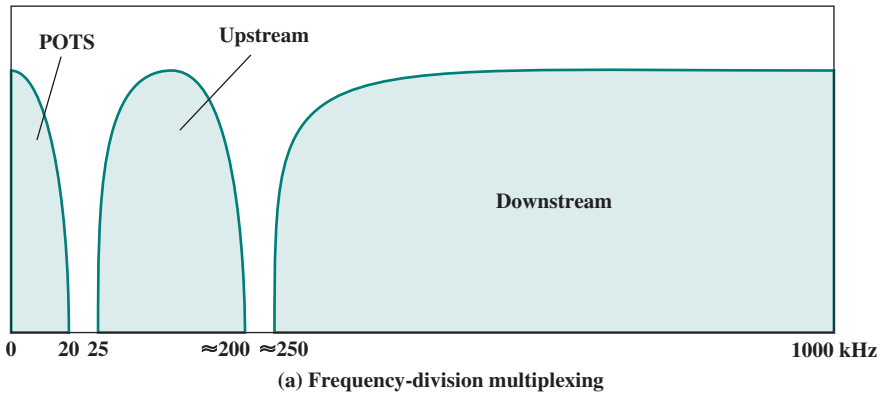


Figure 8.14 ADSL Channel Configuration

- The higher the frequency, the greater the attenuation. With the use of echo cancellation, more of the downstream bandwidth is in the “good” part of the spectrum.
- The echo cancellation design is more flexible for changing upstream capacity. The upstream channel can be extended upward without running into the downstream; instead, the area of overlap is extended.

The disadvantage of the use of echo cancellation is the need for echo cancellation logic on both ends of the line.

The ADSL scheme provides a range of up to 5.5 km, depending on the diameter of the cable and its quality. This is sufficient to cover about 95% of all U.S. subscriber lines and should provide comparable coverage in other nations.

Discrete Multitone

Discrete multitone (DMT) uses multiple carrier signals at different frequencies, sending some of the bits on each channel. The available transmission band (upstream or downstream) is divided into a number of 4-kHz subchannels. On initialization,

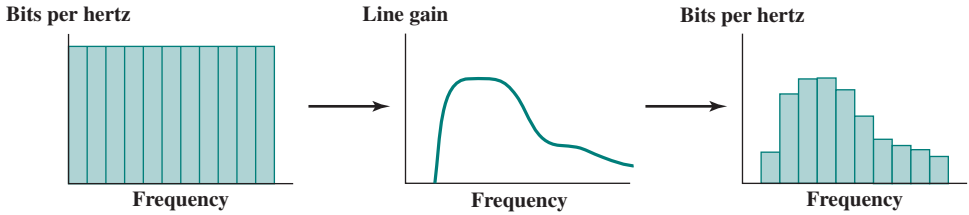


Figure 8.15 DMT Bits per Channel Allocation

the DMT modem sends out test signals on each subchannel to determine the signal-to-noise ratio. The modem then assigns more bits to channels with better signal transmission qualities and less bits to channels with poorer signal transmission qualities. Figure 8.15 illustrates this process. Each subchannel can carry a data rate of from 0 to 60 kbps. The figure shows a typical situation in which there is increasing attenuation and hence decreasing signal-to-noise ratio at higher frequencies. As a result, the higher-frequency subchannels carry less of the load.

Figure 8.16 provides a general block diagram for DMT transmission. After initialization, the bit stream to be transmitted is divided into a number of substreams, one for each subchannel that will carry data. The sum of the data rates of the substreams is equal to the total data rate. Each substream is then converted to an analog signal using quadrature amplitude modulation, described in Chapter 5. This scheme works easily because of QAM's ability to assign different numbers of bits per transmitted signal. Each QAM signal occupies a distinct frequency band, so these signals can be combined by simple addition to produce the composite signal for transmission.

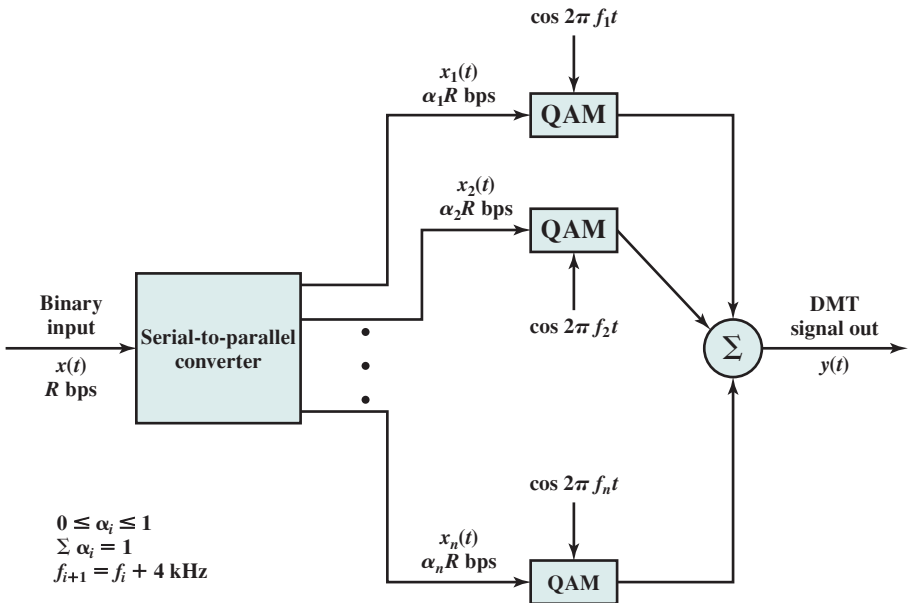
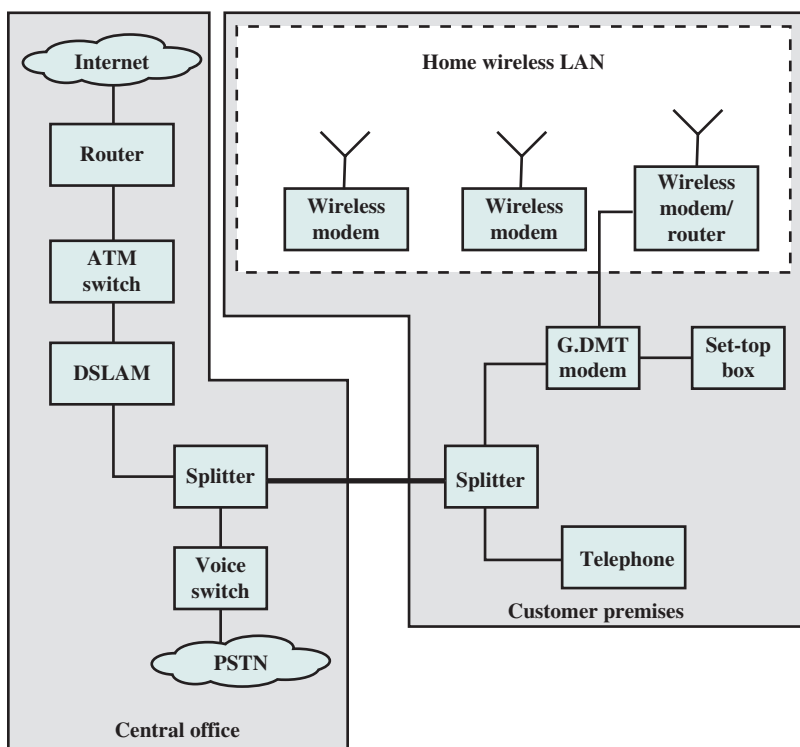


Figure 8.16 DMT Transmitter

Present ADSL/DMT designs employ 256 downstream subchannels. In theory, with each 4-kHz subchannel carrying 60 kbps, it would be possible to transmit at a rate of 15.36 Mbps. In practice, transmission impairments prevent attainment of this data rate. Current implementations operate at from 1.5 to 9 Mbps, depending on line distance and quality.

Broadband Access Configuration

Figure 8.17 shows a typical configuration for broadband service using DSL. The DSL link is between the provider central office and the residential or business premises. On the customer side, a splitter allows simultaneous telephone and data service. The data service makes use of a DSL modem, sometimes referred to as a G.DMT modem, because the modem conforms to the ITU-T G.992.1 recommendation for DMT over DSL. The DSL data signal can be further divided into a video stream and a data stream. The latter connects the modem to either a single local computer or to a wireless modem/router, which enables the customer to support a wireless local area network.



ATM = asynchronous transfer mode
 DSLAM = digital subscriber line access multiplexer
 PSTN = public switched telephone network
 G.DMT = G.992.1 discrete multitone

Figure 8.17 DSL Broadband Access

On the provider side, a splitter is also used to separate the telephone service from the Internet service. The voice traffic is connected to the public switched telephone network (PSTN), thus providing the same service as an ordinary telephone line to the subscriber. The data traffic connects to a DSL access multiplexer (DSLAM), which multiplexes multiple customer DSL connections on to a single high-speed asynchronous transfer mode line. The ATM line connects via one or more ATM switches to a router that provides an entry point to the Internet.

8.5 xDSL

ADSL is one of a number of recent schemes for providing high-speed digital transmission of the subscriber line. Table 8.6 summarizes and compares some of the most important of these new schemes, which collectively are referred to as xDSL.

High Data Rate Digital Subscriber Line

HDSL was developed in the late 1980s by BellCore to provide a more cost-effective means of delivering a T1 data rate (1.544 Mbps). The standard T1 line uses alternate mark inversion (AMI) coding, which occupies a bandwidth of about 1.5 MHz. Because such high frequencies are involved, the attenuation characteristics limit the use of T1 to a distance of about 1 km between repeaters. Thus, for many subscriber lines one or more repeaters are required, which adds to the installation and maintenance expense.

HDSL uses the 2B1Q coding scheme to provide a data rate of up to 2 Mbps over two twisted-pair lines within a bandwidth that extends only up to about 196 kHz. This enables a range of about 3.7 km to be achieved.

Table 8.6 Comparison of xDSL Alternatives

	ADSL	HDSL	SDSL	VDSL
Data Rate	1.5–9 Mbps downstream 16–640 kbps upstream	1.544 or 2.048 Mbps	1.544 or 2.048 Mbps	13–52 Mbps downstream 1.5–2.3 Mbps upstream
Mode	Asymmetric	Symmetric	Symmetric	Asymmetric
Copper Pairs	1	2	1	1
Range (24-Gauge UTP)	3.7–5.5 km	3.7 km	3.0 km	1.4 km
Signaling	Analog	Digital	Digital	Analog
Line Code	CAP/DMT	2B1Q	2B1Q	DMT
Frequency	1–5 MHz	196 kHz	196 kHz	≥ 10 MHz
Bits/Cycle	Varies	4	4	Varies

UTP = unshielded twisted pair

Single-Line Digital Subscriber Line

Although HDSL is attractive for replacing existing T1 lines, it is not suitable for residential subscribers because it requires two twisted pair, whereas the typical residential subscriber has a single twisted pair. SDSL was developed to provide the same type of service as HDSL but over a single twisted-pair line. As with HDSL, 2B1Q coding is used. Echo cancellation is used to achieve full-duplex transmission over a single pair.

Very High Data Rate Digital Subscriber Line

One of the newest xDSL schemes is VDSL. As of this writing, many of the details of this signaling specification remain to be worked out. The objective is to provide a scheme similar to ADSL at a much higher data rate by sacrificing distance. The likely signaling technique is DMT/QAM.

VDSL does not use echo cancellation but provides separate bands for different services, with the following tentative allocation:

- POTS: 0–4 kHz
- ISDN: 4–80 kHz
- Upstream: 300–700 kHz
- Downstream: ≥ 1 MHz

8.6 MULTIPLE CHANNEL ACCESS

In this section, we look at four multiplexing techniques used for sharing channel capacity among multiple transmitter/receiver stations. These techniques differ from the FDM and TDM techniques so far discussed, because no physical multiplexer is involved. Rather, individual stations are assigned a frequency band or a sequence of time slots and transmit directly on the channel and not through a multiplexer.

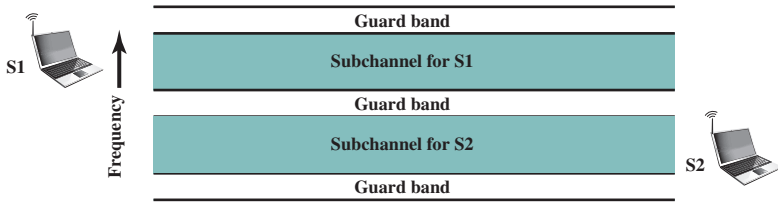
The techniques discussed in this section are used as building blocks in a number of wireless schemes, including wireless LANs such as Wi-Fi, cellular networks, satellite networks, and wireless broadband Internet access, such as WiMAX.

Frequency-Division Duplex (FDD)

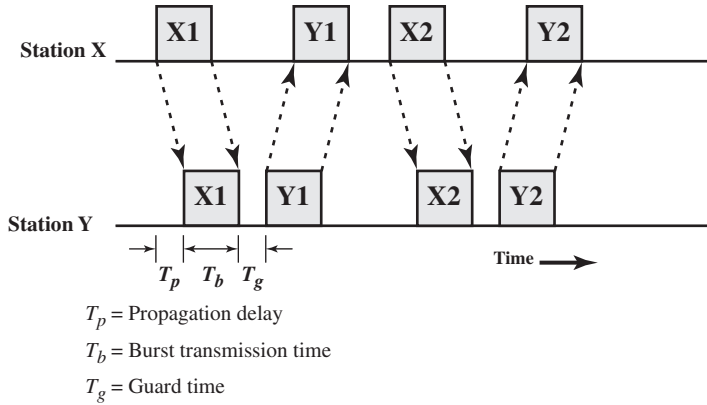
FDD, by itself, is not a particularly interesting case. FDD simply means that two stations have a full-duplex connection in which each station transmits on a separate frequency band. The two frequency bands are separated from each other and from other bands on the network by guard bands, to prevent interference (Figure 8.18a). The combination of the two frequency bands is often referred to as a **subchannel**, with the combination of the two subchannels viewed as a full-duplex channel between the stations.

Time-Division Duplex (TDD)

In TDD, also known as **time-compression multiplexing (TCM)**, data are transmitted in one direction at a time, with transmission alternating between the two directions.



(a) Frequency-division duplex (TDD)



(b) Time-division duplex (TDD)

Figure 8.18 Duplex Access Techniques

To achieve the desired subscriber data rate with simple TDD, the transmitter’s bit stream is divided into equal segments, compressed in time to a higher transmission rate, and transmitted in bursts, which are expanded at the other end to the original rate. A short quiescent period is used between bursts going in opposite directions to allow the channel to settle down. Thus, the actual data rate on the channel must be greater than twice the data rate required by the two end systems.

The timing implications are shown in Figure 8.19b. The two sides alternate in the transmission of data. Each side sends blocks of some fixed length, which take a time T_b to transmit; this time is a linear function of the number of bits in a block. In addition, a time T_p is required for the propagation of a signal from transmitter to receiver; this time is a linear function of the distance between transmitter and receiver. Finally, a guard time T_g is introduced to turn the channel around. Thus, the time to send one block is $(T_p + T_b + T_g)$. However, because the two sides must alternate transmissions, the rate at which blocks can be transmitted by either side is only $1/[2(T_p + T_b + T_g)]$. We can relate this to the effective data rate, R , as seen by the two endpoints. Let B be the size of a block in bits. Then the effective number of bits transmitted per second, or effective data rate, is

$$R = \frac{B}{2(T_p + T_b + T_g)}$$

The actual data rate, A , on the medium can easily be seen to be

$$A = B/T_b$$

Combining the two, we have

$$A = 2R \left(1 + \frac{T_p + T_g}{T_b} \right)$$

The choice of block size, B , is a compromise between competing requirements. For a larger block size, the value of T_b becomes larger compared to the values of T_p and T_g . Now consider that we have a fixed value of R , which is the data rate required for the link, and we need to determine the value of A . If B is increased, there is a decrease in the actual data rate, A . This makes the task of implementation easier. On the other hand, this is accompanied by an increase in the signal delay due to buffering, which is undesirable for voice traffic.

EXAMPLE 8.4 One of the standard interfaces defined for ISDN (Integrated Services Digital Network) is the basic interface, which provides a data rate of 192 kbps and uses a frame size of 48 bits. Suppose we use TDD with a block size equal to the frame size. Assume the distance between the subscriber and the network switch is 1 km and a guard time $10 \mu\text{s}$ is used. What is the actual data rate?

First, we need to determine the burst transmission time T_b . The propagation delay is $(1 \text{ km})/(3 \times 10^8 \text{ m/s}) = 3.33 \mu\text{s}$. We have $\frac{B}{2R} = (T_p + T_b + T_g)$, so that

$$T_b = \frac{B}{2R} - T_p - T_g = \frac{48}{0.384 \times 10^6} - (10 \times 10^{-6}) - (3.33 \times 10^{-6}) = 111.67 \mu\text{s}.$$

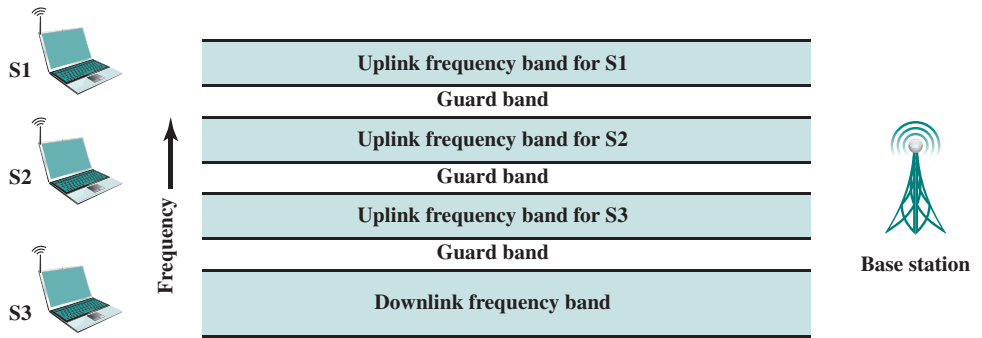
Thus,

$$A = 2 \times 192 \times [1 + (3.33 + 10)/111.67] \approx 430 \text{ kbps}$$

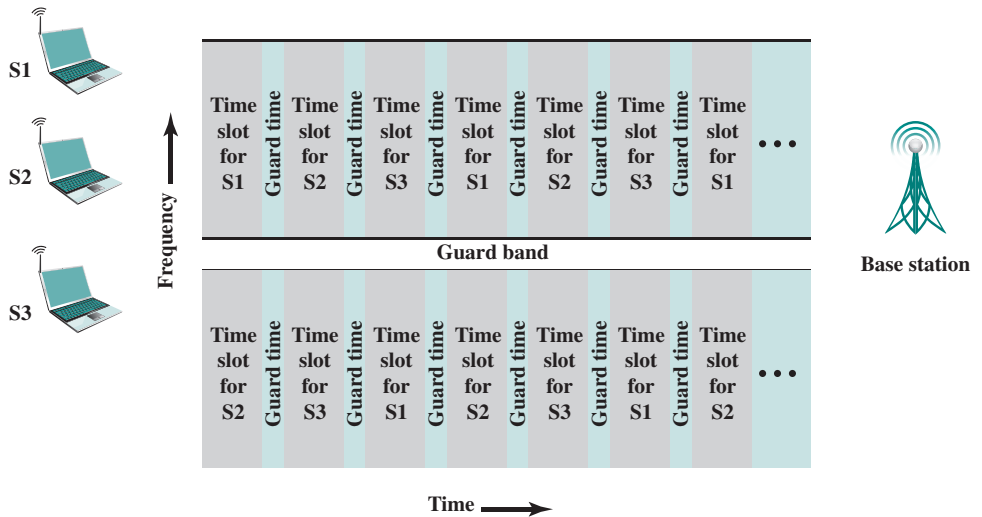
TDD is used in cordless telephones and is a building block for a number of wireless network systems.

Frequency-Division Multiple Access (FDMA)

FDMA is a technique used to share the spectrum among multiple stations. In a typical configuration, there is a base station that communicates with a number of subscriber stations. Such a configuration is found in satellite networks, cellular networks, Wi-Fi, and WiMAX. Typically, the base station assigns bandwidths to stations within the overall bandwidth available. Figure 8.19a is an example. Three stations are assigned separate frequency bands (subchannels) for transmission to the base station (uplink direction), with guard bands between the assigned transmission bands. Another frequency band, typically wider, is reserved for transmission from the base station to the other stations (downlink direction).



(a) Frequency-division multiple access (FDMA)



(b) Time-division multiple access (TDMA)

Figure 8.19 Multiple Channel Access Techniques

Key features of FDMA include the following:

- Each subchannel is dedicated to a single station; it is not shared.
- If a subchannel is not in use, it is idle; the capacity is wasted.
- FDMA is relatively less complex than TDMA and requires fewer overhead bits because each subchannel is dedicated.
- Individual subchannels must be separated by guard bands to minimize interference.

Time-Division Multiple Access (TDMA)

As with FDMA, TDMA is typically used in a configuration that consists of a base station and a number of subscriber stations. With TDMA there is a single, relatively large, uplink frequency band that is used to transmit a sequence of time slots. Repetitive time slots are assigned to an individual subscriber station to form a logical subchannel. Figure 8.19b is an example. In this example, each station gets an equal amount of the overall capacity of the uplink channel. Thus, each channel is assigned every third slot. Similarly, each subscriber station listens on designated time slots on the downlink channel, which may have the same slot assignment as the uplink channel, or a different one. In this example, the downlink channel is also equally distributed among the three stations.

Key features of TDMA include the following:

- Each subchannel is dedicated to a single station; it is not shared.
- For an individual station, data transmission occurs in bursts rather than continuously.
- Guard times are needed between time slots, to account for lack of perfect synchronization among the subscriber station.
- The downlink channel may be on a separate frequency band, as in our example. This is referred to as TDMA/FDD. With TDMA/FDD, the time slots assigned for subscriber station reception are typically nonoverlapping with that station's transmit time slots.
- The uplink and downlink transmission may be on the same frequency band, which is referred to as TDMA/TDD.

8.7 RECOMMENDED READING AND ANIMATIONS

A discussion of FDM and TDM carrier systems can be found in [FREE98b]. SONET is treated in greater depth in [STAL99] and in [TEKT01]. Useful articles on SONET are [BALL89] and [BOEH90]. A good overview of WDM is [MUKH00].

Two good articles on cable modems are [FELL01] and [CICI01].

[MAXW96] provides a useful a discussion of ADSL. Recommended treatments of xDSL are [HAWL97] and [HUMP97].

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Animations

An animation that illustrates multiplexing is available at the Premium Web site. The reader is encouraged to view the animation to reinforce concepts from this chapter.

8.8 KEY TERMS, REVIEW QUESTIONS, AND PROBLEMS

Key Terms

ADSL baseband cable modem channel demultiplexer dense wavelength division multiplexing (DWDM) dense WDM digital carrier system discrete multitone	downstream echo cancellation frame frequency-division multiplexing (FDM) headend multiplexer multiplexing pulse stuffing SDH	SONET statistical TDM subcarrier synchronous TDM time-division multiplexing (TDM) upstream wavelength division multiplexing (WDM)
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Review Questions

- 8.1** Why is multiplexing so cost-effective?
- 8.2** How is interference avoided by using frequency-division multiplexing?
- 8.3** What is echo cancellation?
- 8.4** Define *upstream* and *downstream* with respect to subscriber lines.
- 8.5** Explain how synchronous time-division multiplexing (TDM) works.

- 8.6** Why is a statistical time-division multiplexer more efficient than a synchronous time-division multiplexer?
- 8.7** Using Table 8.3 as a guide, indicate the major difference between North American and international TDM carrier standards.

Problems

- 8.1** The information in four analog signals is to be multiplexed and transmitted over a telephone channel that has a 400- to 3100-Hz bandpass. Each of the analog baseband signals is bandlimited to 500 Hz. Design a communication system (block diagram) that will allow the transmission of these four sources over the telephone channel using
- Frequency-division multiplexing with SSB (single sideband) subcarriers
 - Time-division multiplexing using PCM; assume 4-bit samples
- Show the block diagrams of the complete system, including the transmission, channel, and reception portions. Include the bandwidths of the signals at the various points in the systems.
- 8.2** To paraphrase Lincoln: "...all of the channel some of the time, some of the channel all of the time..." Refer to Figure 8.2 and relate the preceding to the figure.
- 8.3** Consider a transmission system using frequency-division multiplexing. What cost factors are involved in adding one more pair of stations to the system?
- 8.4** In synchronous TDM, it is possible to interleave bits, one bit from each channel participating in a cycle. If the channel is using a self-clocking code to assist synchronization, might this bit interleaving introduce problems because there is not a continuous stream of bits from one source?
- 8.5** Why is it that the start and stop bits can be eliminated when character interleaving is used in synchronous TDM?
- 8.6** Explain in terms of data link control and physical layer concepts how error and flow control are accomplished in synchronous time-division multiplexing.
- 8.7** One of the 193 bits in the DS-1 transmission format is used for frame synchronization. Explain its use.
- 8.8** In the DS-1 format, what is the control signal data rate for each voice channel?
- 8.9** Twenty-four voice signals are to be multiplexed and transmitted over twisted pair. What is the bandwidth required for FDM? Assuming a bandwidth efficiency (ratio of data rate to transmission bandwidth, as explained in Chapter 5) of 1 bps/Hz, what is the bandwidth required for TDM using PCM?
- 8.10** Draw a block diagram similar to Figure 8.8 for a TDM PCM system that will accommodate four 300-bps, synchronous, digital inputs and one analog input with a bandwidth of 500 Hz. Assume that the analog samples will be encoded into 4-bit PCM words.
- 8.11** A character-interleaved time-division multiplexer is used to combine the data streams of a number of 110-bps asynchronous terminals for data transmission over a 2400-bps digital line. Each terminal sends asynchronous characters consisting of 7 data bits, 1 parity bit, 1 start bit, and 2 stop bits. Assume that one synchronization character is sent every 19 data characters and, in addition, at least 3% of the line capacity is reserved for pulse stuffing to accommodate speed variations from the various terminals.
- Determine the number of bits per character.
 - Determine the number of terminals that can be accommodated by the multiplexer.
 - Sketch a possible framing pattern for the multiplexer.
- 8.12** Find the number of the following devices that could be accommodated by a T1-type TDM line if 1% of the T1 line capacity is reserved for synchronization purposes.
- 110-bps teleprinter terminals
 - 300-bps computer terminals

- c. 1200-bps computer terminals
- d. 9600-bps computer output ports
- e. 64-kbps PCM voice-frequency lines

How would these numbers change if each of the sources were transmitting an average of 10% of the time and a statistical multiplexer was used?

- 8.13** Ten 9600-bps lines are to be multiplexed using TDM. Ignoring overhead bits in the TDM frame, what is the total capacity required for synchronous TDM? Assuming that we wish to limit average TDM link utilization to 0.8, and assuming that each TDM link is busy 50% of the time, what is the capacity required for statistical TDM?
- 8.14** A synchronous nonstatistical TDM is to be used to combine four 4.8-kbps and one 9.6-kbps signals for transmission over a single leased line. For framing, a block of 7 bits (pattern 1011101) is inserted for each 48 data bits. The reframing algorithm (at the receiving demultiplex) is as follows:
1. Arbitrarily select a bit position.
 2. Consider the block of 7 contiguous bits starting with that position.
 3. Observe that block of 7 bits each frame for 12 consecutive frames.
 4. If 10 of the 12 blocks match the framing pattern the system is “in-frame”; if not advance one bit position and return to step 2.
 - a. Draw the multiplexed bit stream (note that the 9.6 kbps input may be treated as two 4.8-kbps inputs).
 - b. What is the percentage overhead in the multiplexed bit stream?
 - c. What is the multiplexed output bit rate?
 - d. What is the minimum reframe time? What is the maximum reframe time? What is the average reframe time?
- 8.15** A company has two locations: a headquarters and a factory about 25 km away. The factory has four 300-bps terminals that communicate with the central computer facilities over leased voice-grade lines. The company is considering installing TDM equipment so that only one line will be needed. What cost factors should be considered in the decision?
- 8.16** In synchronous TDM, the I/O lines serviced by the two multiplexers may be either synchronous or asynchronous although the channel between the two multiplexers must be synchronous. Is there any inconsistency in this? Why or why not?
- 8.17** Assume that you are to design a TDM carrier, say DS-489, to support 30 voice channels using 6-bit samples and a structure similar to DS-1. Determine the required bit rate.
- 8.18** For a statistical time-division multiplexer, define the following parameters:
- $$F = \text{frame length, bits}$$
- $$OH = \text{overhead in a frame, bits}$$
- $$L = \text{load of data in the frame, bps}$$
- $$C = \text{capacity of link, bps}$$
- a. Express F as a function of the other parameters. Explain why F can be viewed as a variable rather than a constant.
 - b. Plot F versus L for $C = 9.6$ kbps and values of $OH = 40, 80, 120$. Comment on the results and compare to Figure I.1 in Appendix I.
 - c. Plot F versus L for $OH = 40$ and values of $C = 9.6$ kbps and 8.2 kbps. Comment on the results and compare to Figure I.1 in Appendix I.
- 8.19** In statistical TDM, there may be a length field. What alternative could there be to the inclusion of a length field? What problem might this solution cause and how could it be solved?