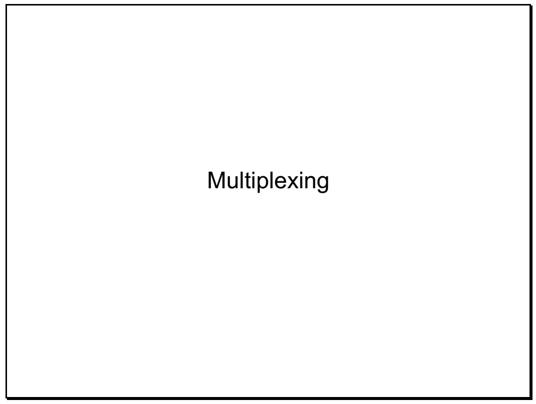
Slide 1



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.

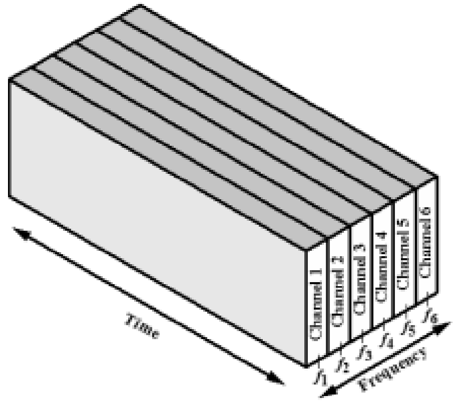
A common application of multiplexing is in long-haul communications. Trunks on long-haul networks are high-capacity fiber, coaxial, or microwave links. These links can carry large numbers of voice and data transmissions simultaneously using multiplexing.

Slide 2

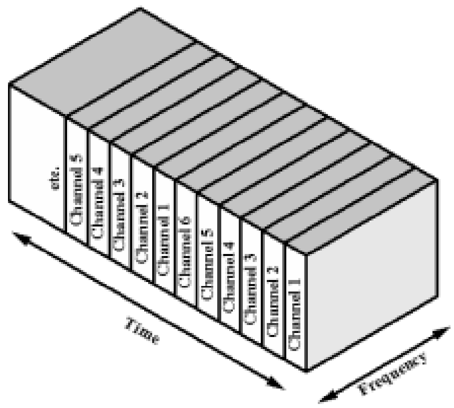
Figure 8.1 Multiplexing

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 over a higher-capacity data link. The demultiplexer accepts the multiplexed data stream, separates (demultiplexes) the data according to channel, and delivers data to the appropriate output lines.combines (multiplexes) data from the n input lines and transmits over a higher-capacity data link. The demultiplexer accepts the multiplexed data stream, separates (demultiplexes) the data according to channel, and delivers data to the appropriate output lines.

Slide 3



(a) Frequency division multiplexing



(b) Time division multiplexing

Figure 8.2 FDM and TDM

What is multiplexing?

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 (f1 , . . . , f6). 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.

Slide 4

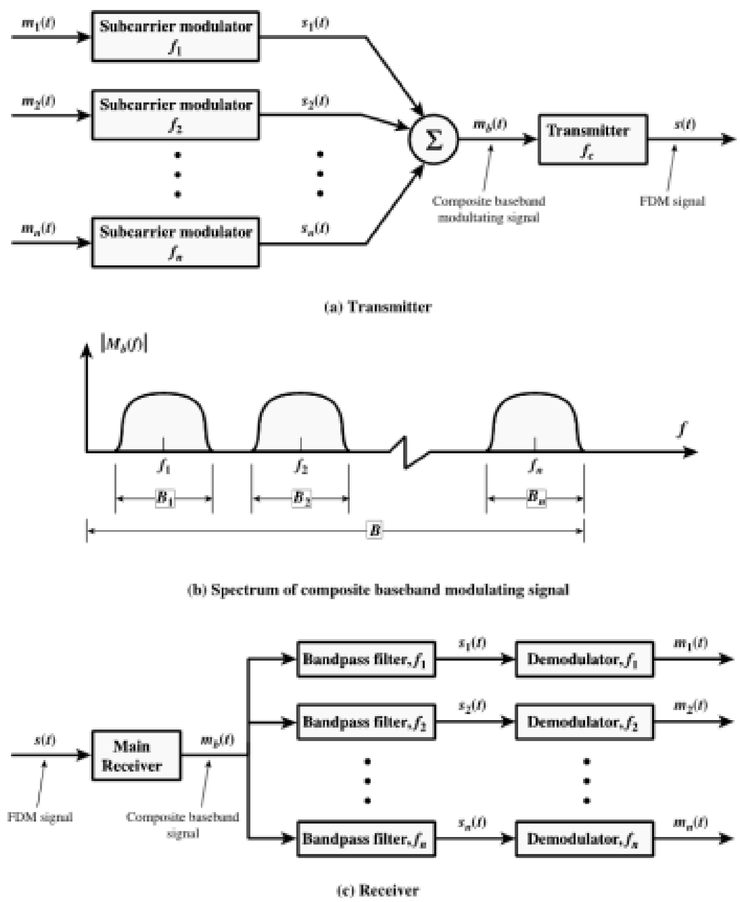


Figure 8.3 FDM System

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. Modulation equipment is needed to move each signal to the required frequency band, and multiplexing equipment is needed to combine the modulated signals. 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 signal $m_b(t)$. Figure 8.3b shows the result. The spectrum of 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.

The FDM signal $s(t)$ has a total bandwidth B = Sum 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 $1 \le i \le n$. In this way, the signal is again split into its component parts. Each component is then demodulated to recover the original signal.

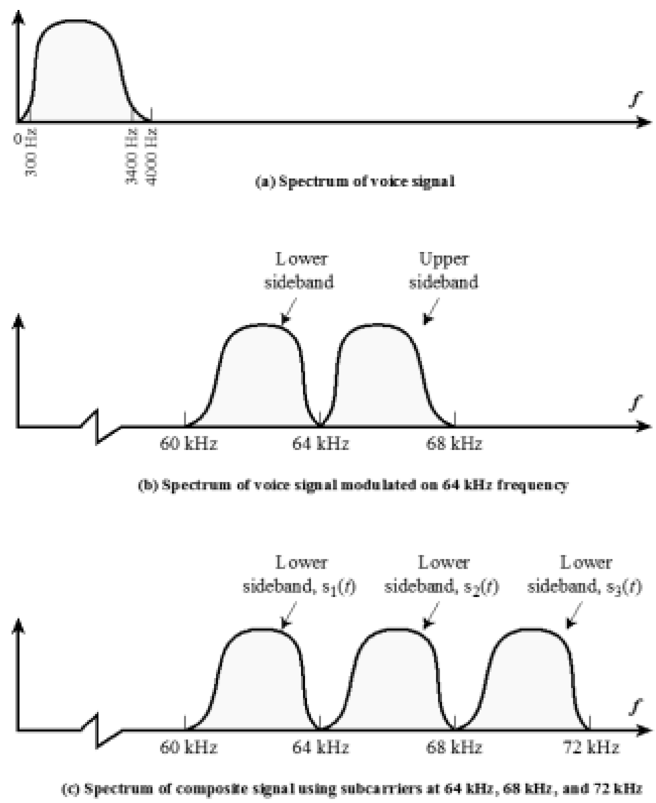


Figure 8.4 FDM of Three Voiceband Signals

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 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

- Long-distance links use an FDM hierarchy
- AT&T (USA) and ITU-T (International) variants

Group	Supergroup	Mastergroup
• 12 voice channels (4kHz each) = 48kHz • Range 60kHz to 108kHz	• FDM of 5 group signals supports 60 channels • Carriers between 420kHz and 612 kHz	• FDM of 10 supergroups supports 600 channels

- Original signal can be modulated many times

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.

Slide 7

Table 8.1
North American and International FDM Carrier Standards

?

Number of Voice Channels	Bandwidth	Spectrum (operate in.)	AT&T	ITU-T
4khz → 12 ≃	48 kHz	60–108 kHz	Group	Group
60	240 kHz	312–552 kHz	Supergroup = 5 groups	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	

At the first level of the AT&T hierarchy, 12 voice channels are combined to produce a group signal with a bandwidth of 12 × 4 kHz = 48 kHz, in the range 60 to 108 kHz. The next basic building block is the 60-channel supergroup, which is formed by frequency division multiplexing five group signals. The subcarriers have frequencies from 420 to 612 kHz in increments of 48 kHz. The resulting signal occupies 312 to 552 kHz. The next level of the hierarchy is the mastergroup, which combines 10 supergroup inputs. 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. Each stage can distort the original data; this is so, for example, if the modulator/multiplexer contains nonlinearities or introduces noise.

Wavelength Division Multiplexing (WDM)

Multiple beams of light at different frequencies

Carried over optical fiber links

- Commercial systems with 160 channels of 10 Gbps
- Lab demo of 256 channels 39.8 Gbps

Architecture similar to other FDM systems

- Multiplexer consolidates laser sources (1550nm) for transmission over single fiber
- Optical amplifiers amplify all wavelengths
- Demultiplexer separates channels at destination

Dense Wavelength Division Multiplexing (DWDM)

- Use of more channels more closely spaced

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 (FDM) 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. 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.

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.

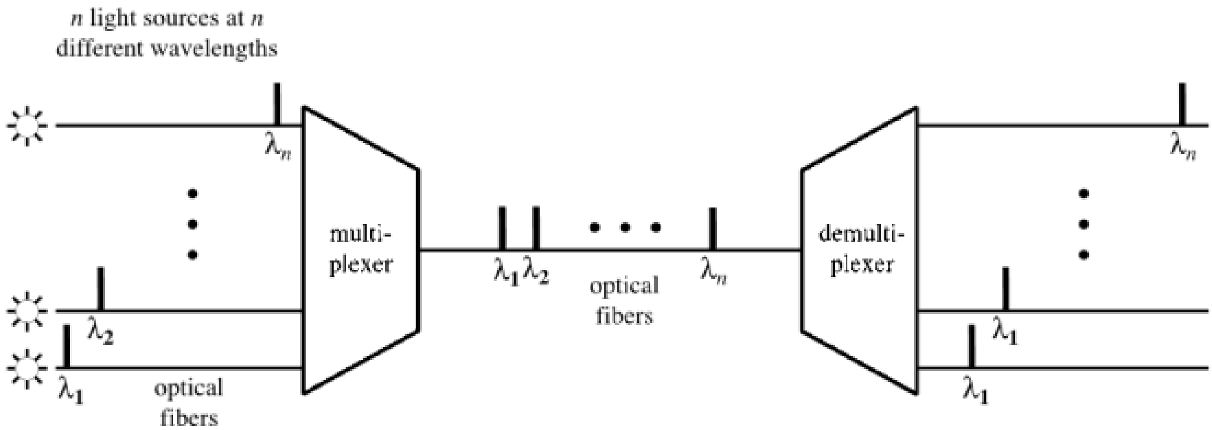


Figure 8.5 Wavelength Division Multiplexing

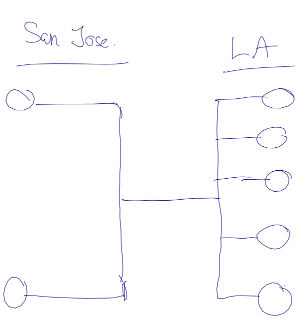
A typical WDM system has the same general architecture as other FDM systems. A number of sources generate a laser beam at different wavelengths. Most WDM systems operate in the 1550-nm range. These are sent to a multiplexer, which consolidates the sources for transmission over a single fiber line. Optical 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)

Table 8.2
ITU WDM Channel Spacing (G.692)

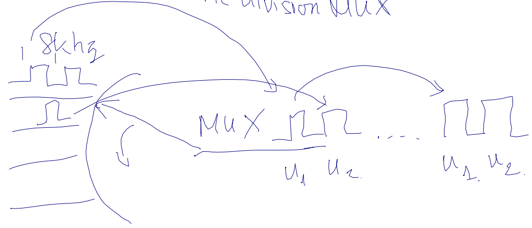
Frequency (THz)	Wavelength in Vacuum (nm)	50 GHz	100 GHz	200 GHz
196.10	1528.77	X	X	X
196.05	1529.16	X		
196.00	1529.55	X	X	
195.95	1529.94	X		
195.90	1530.33	X	X	X
195.85	1530.72	X		
195.80	1531,12	X	X	
195.75	1531.51	X		
195.70	1531.90	X	X	X
195.65	1532.29	X		
195.60	1532.68	X	X	
…	…			
192.10	1560.61	X	X	X

1) What is the freq of telephone signal? 4khz.

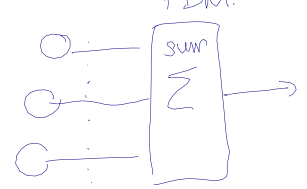
Most WDM systems operate in the 1550-nm range. In early systems, 200 GHz was allocated to each channel, but today most WDM systems use 50-GHz spacing. The channel spacing defined in ITU-T G.692, which accommodates 80 50-GHz channels, is summarized in Table 8.2.



time division MUX



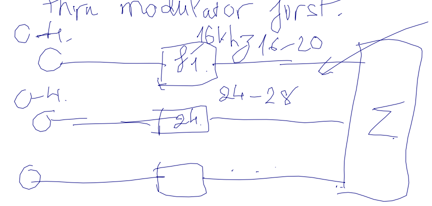
Freq Division MUX
FDM.



Signal should be sampled at 8 khz (prevent loss of signal).

Because of noise, we can't simply add as above, each signal has to go thru modulator first.

to avoid overlapping.



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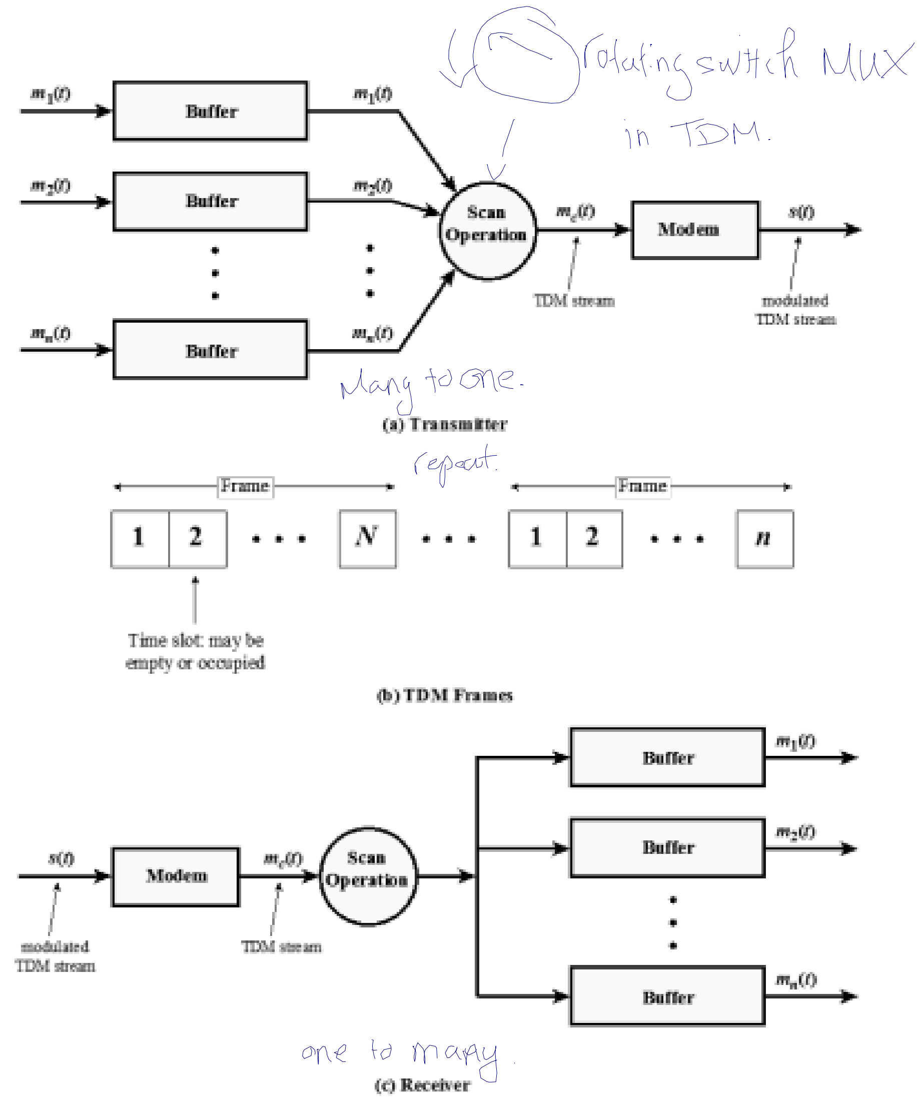

Figure 8.6 Synchronous TDM System

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 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.

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TDM Link Control

- No headers and trailers
- Data link control protocols not needed
- Flow control ← layer 2
  - Data rate of multiplexed line is fixed
  - If one channel receiver can not receive data, the others must carry on
  - Corresponding source must be quenched
  - Leaving empty slots
- Error control
  - Errors detected and handled on individual channel

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.

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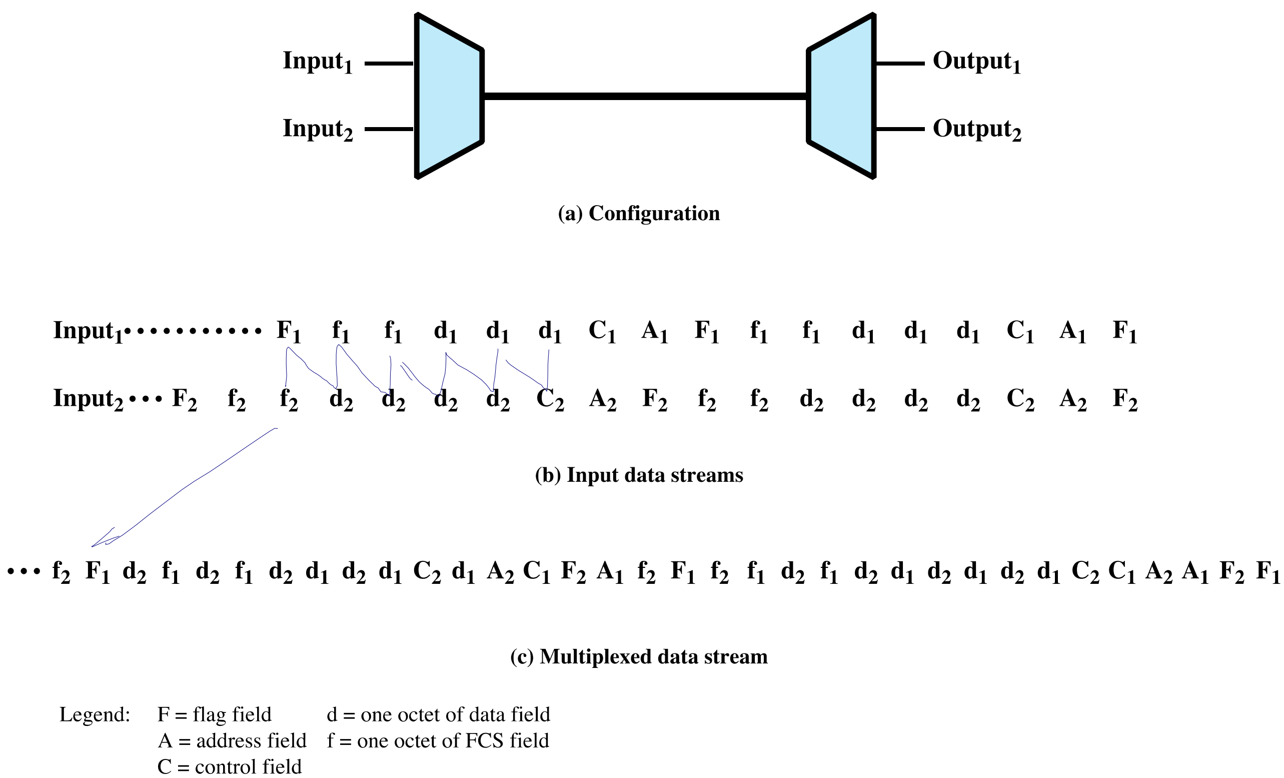

Figure 8.7 Use of Data Link Control on TDM Channels

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.

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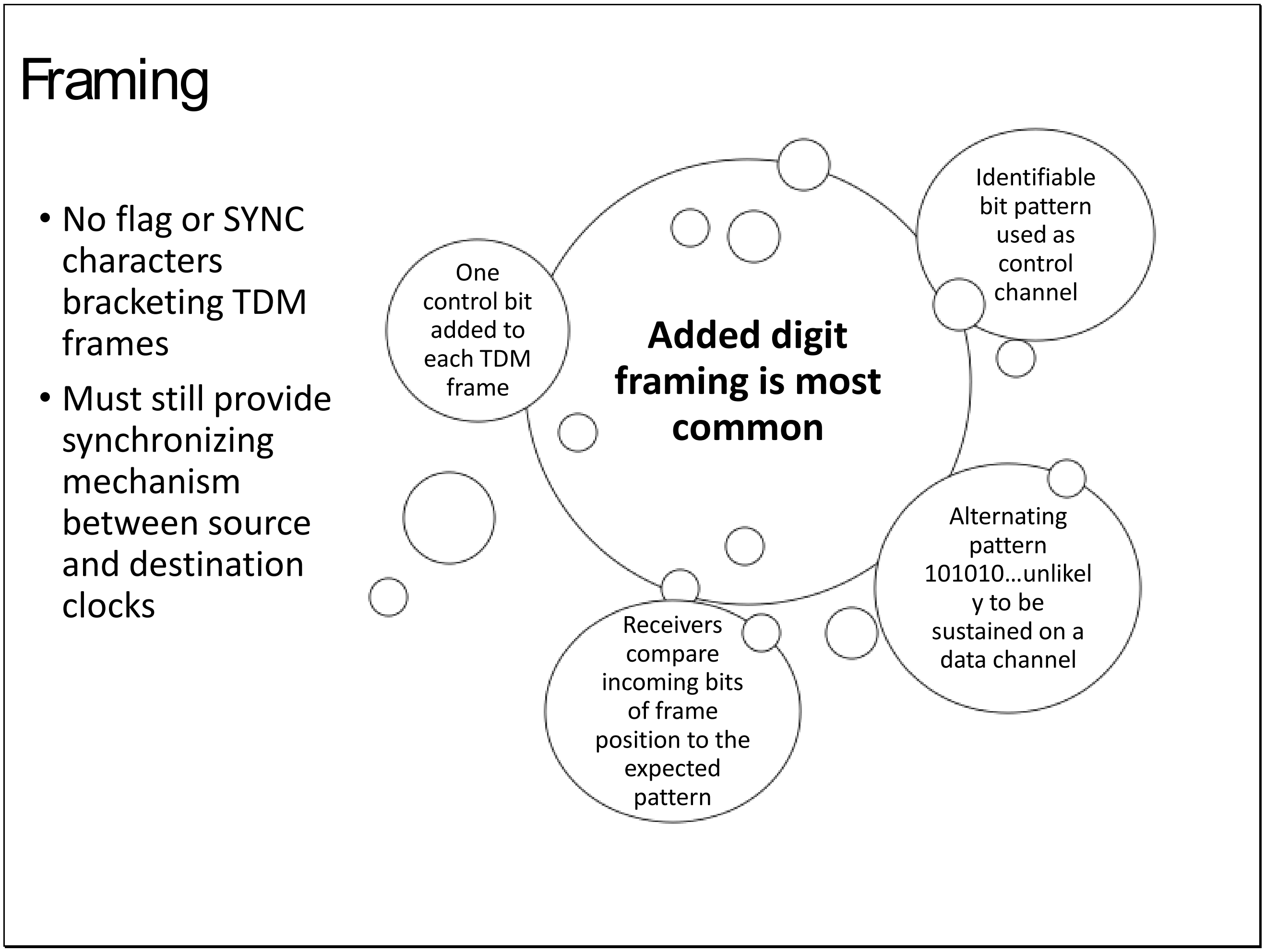

We have seen that a link control protocol is not needed to manage the overall TDM link. There is, however, a basic requirement for framing. Because we are not providing flag or SYNC characters to bracket TDM frames, some means is needed to assure frame synchronization. It is clearly important to maintain 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.

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Pulse Stuffing is a common solution

Have outgoing data rate (excluding framing bits) higher than sum of incoming rates

sender

Stuff extra dummy bits or pulses into each incoming signal until it matches local clock

receiver.

Stuffed pulses inserted at fixed locations in frame and removed at demultiplexer

- Problem of synchronizing various data sources
- Variation among clocks could cause loss of synchronization
- Issue of data rates from different sources not related by a simple rational number

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.

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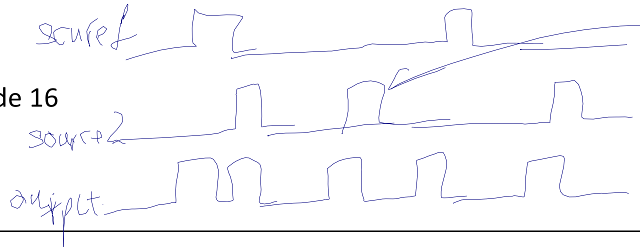
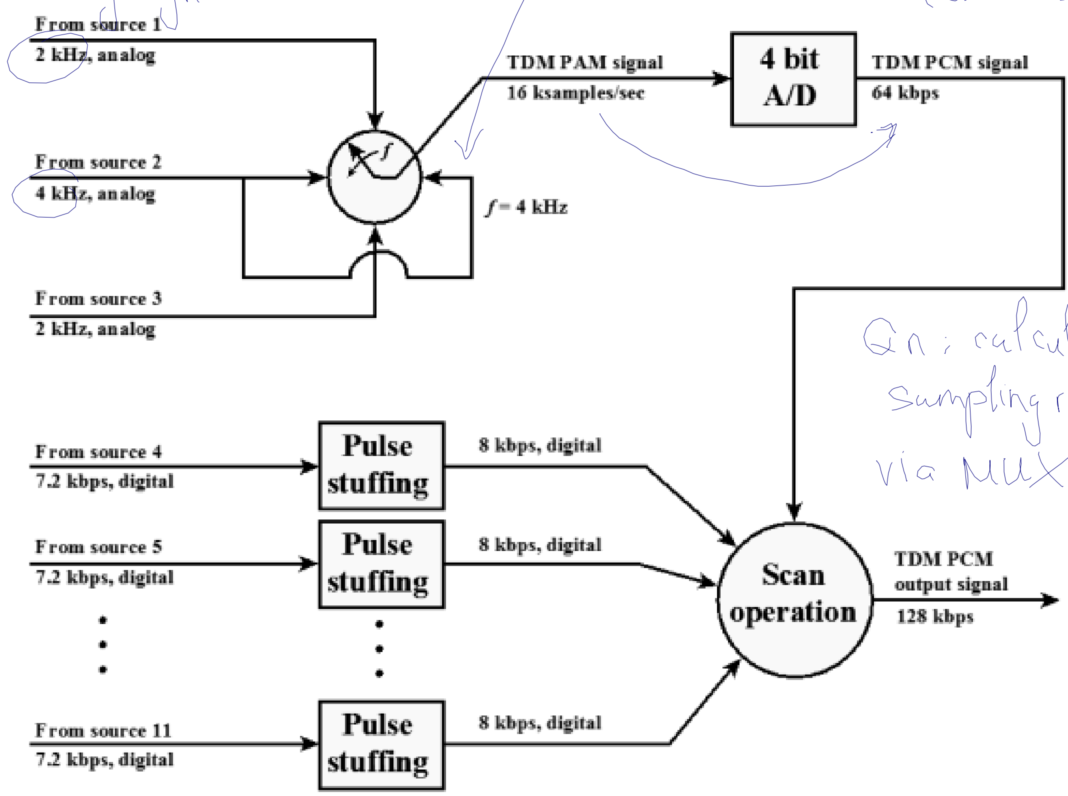


this require double sampling as freq of 2nd source is twice faster than every other sources, if we don't sample twice we will lose the signal.



Analog signal

4bit converter (double sampling: the 2nd time should be half the time split)

Qn: calculate sampling rate via MUX

Figure 8.8 TDM of Analog and Digital Sources

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.

DS1:

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U1 8khz DS = 64kHz = 8×8(bit). TDM

DS1-1

DS1-24

U24

SOF

24 channels: 24 × 8 = 192 bit + 1 ← SOF: start of frame (1 bit)

= 193 bits → $T = \frac{1}{8k} = 125\mu s$

⇒ 193 bits in 125 μs

$\frac{193}{125\times 10^{-6}} = \frac{193\times 10^6}{125} = 1544$ Khz.

U1 8kHz U1

DS2: DS1-1 8 1.544 M

DS1-2 8

DS1-3 8

DS1-4 8

6.312 DS-2.

7 DS2 → DS3

Qn how many bits are added?

$\frac{1.544 \times 4 + x}{125} = 6.312$.

OAM: Operater Admin Managemt.

Table 8.3

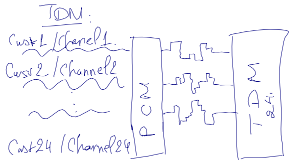
explains (QN in exam)

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

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).

TDM:



We have sampling rate = 2× voice freq = 8000 Hz

Bit rate = 8-bit

⇒ each customer b/w is 8 × 8000 = 64 kbps → DS-0

DS-1: Since there are 24 channels

⇒ Bit rate = 24 × 8 + 1 (framing bit) = 193 bit

⇒ TDM b/w = 193 × 8000 = 1.544 Mbps

Explanation: if only 1 subscriber, time for 1 signal is $\frac{1}{f} = \frac{1}{8000} = 125\mu s$.

What TDM did is compress 24 subscribers into 125 μs,

i.e each subscriber (channel) now has $\frac{125}{24} = 5.2\mu s$ for 1 signal

DS1

125 μs : 193 bit

1 s : ? bit.

? bit = $\frac{193}{125\mu s}$ = 193×8000

DS-2: 96 channels = 4×DS-1

125 μs : 193×4 + x

1 s : 6.312 Mbps

(193×4 + x) × 8000 = 6312000

772 + x = 789

x = 17 bits

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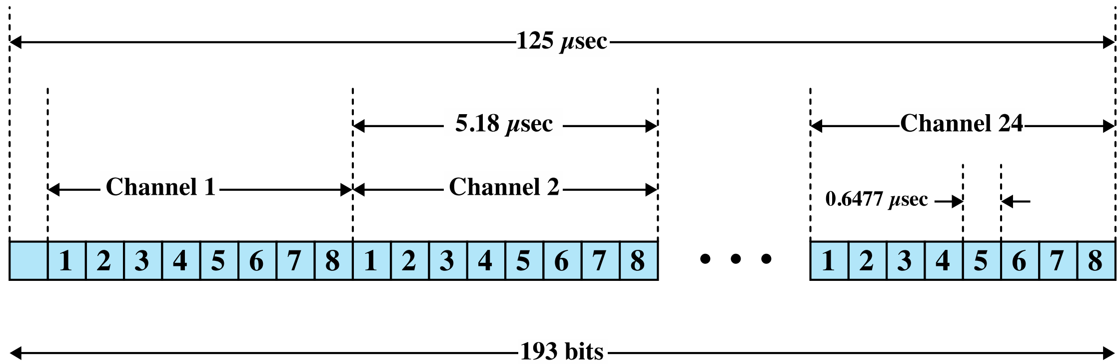

Notes:

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, 4.8, 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 $24 * 8 + 1 = 193$ 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 $8000 * 193 = 1.544$ 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 $6 * 8000 = 48$ 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 $1.544 * 4 = 6.176$ Mbps. The remaining capacity is used for framing and control bits.

SONET/SDH

similar one by EU, one by US.

- Synchronous Optical Network (ANSI)
- Synchronous Digital Hierarchy (ITU-T)
- High speed capability of optical fiber
- Defines hierarchy of signal rates
    - Synchronous Transport Signal level 1 (STS-1) or Optical Carrier level 1 (OC-1) is 51.84Mbps
    - Carries one DS-3 or multiple (DS1 DS1C DS2) plus ITU-T rates (e.g., 2.048Mbps)
    - Multiple STS-1 combine into STS-N signal
    - ITU-T lowest rate is 155.52Mbps (STM-1)

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. SONET is intended to provide a specification for taking advantage of the high-speed digital transmission capability of optical fiber.

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

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), 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). An 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. 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.

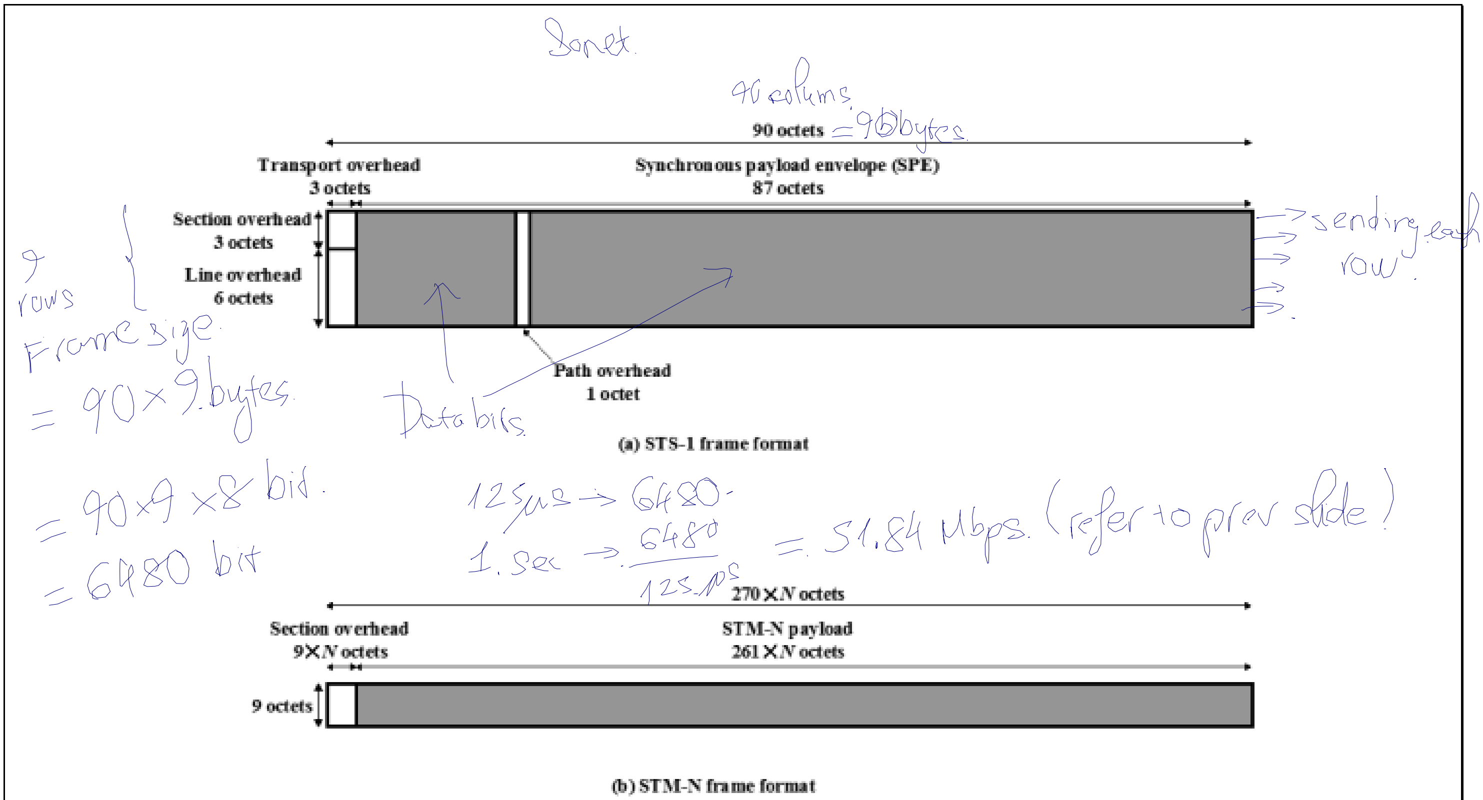

Figure 8.10 SONET/SDH Frame Formats

The basic SONET building block is the STS-1 frame, which consists of 810 octets and is transmitted once every 125 μs, 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.

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

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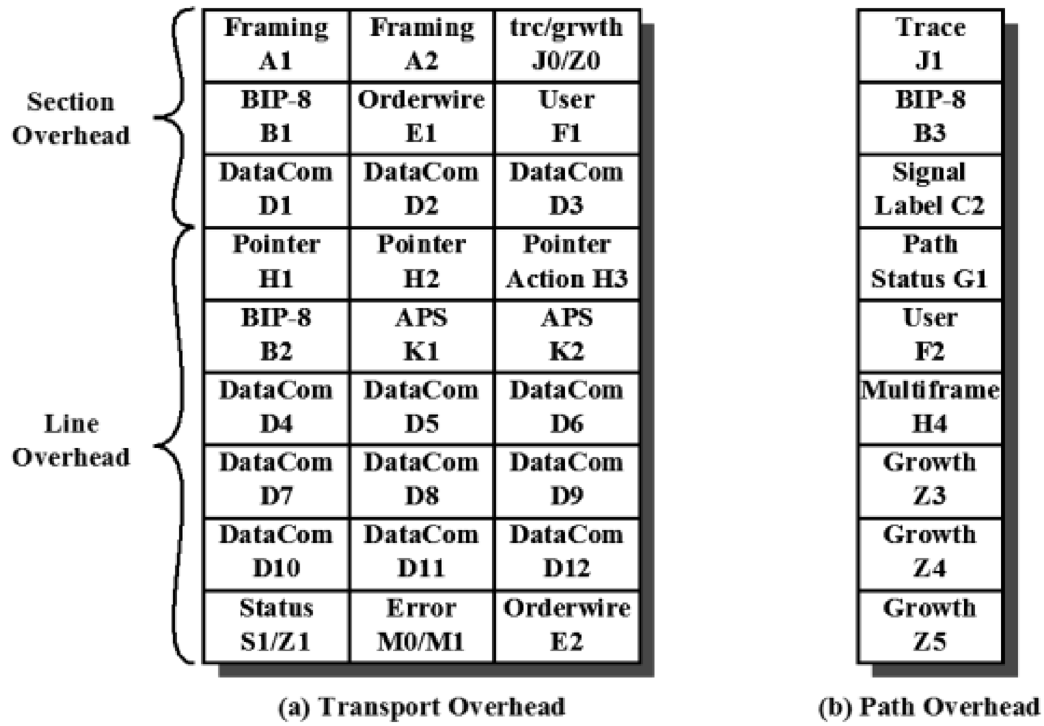

Figure 8.11 SONET STS-1 Overhead Octets

The first three columns (3 octets * 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.

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.

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 sixteen-byte message. This message is transmitted in sixteen 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-*N* frame after scrambling; the *i*th bit of this octet contains the even parity value calculated from the *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-N signal, used for transporting syncrhonization message (S1). Undefined in the second through *N*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. *Unequipped* means the 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.

(Table can be found on page 253 in textbook)

Table 8.5 defines the various fields.

[no study →

Cable Modems

Downstream

- Cable scheduler delivers data in small packets
- Active subscribers share downstream capacity
- Also allocates upstream time slots to subscribers

Upstream

- User requests timeslots on shared upstream channel
- Headend scheduler notifies subscriber of slots to use

-Dedicate two cable TV channels to data transfer
-Each channel shared by number of subscribers using statistical TDM

A cable modem is a device that allows a user to access the Internet and other online services through a cable television 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

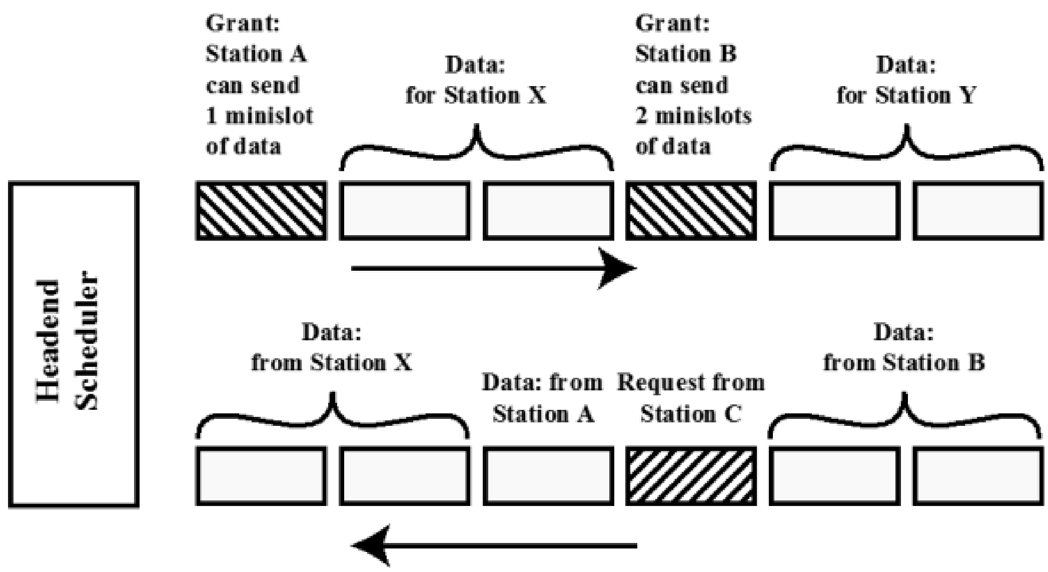

Figure 8.12 Cable Modem Scheme

Figure 8.12
Cable Modem Scheme

Cable Spectrum Division

- To support both cable television programming and data channels, the cable spectrum is divided in to three ranges:
  - User-to-network data (upstream): 5 - 40 MHz
  - Television delivery (downstream): 50 - 550 MHz
  - Network to user data (downstream): 550 - 750 MHz

To support both cable television programming and data channels, the cable spectrum is divided in to 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

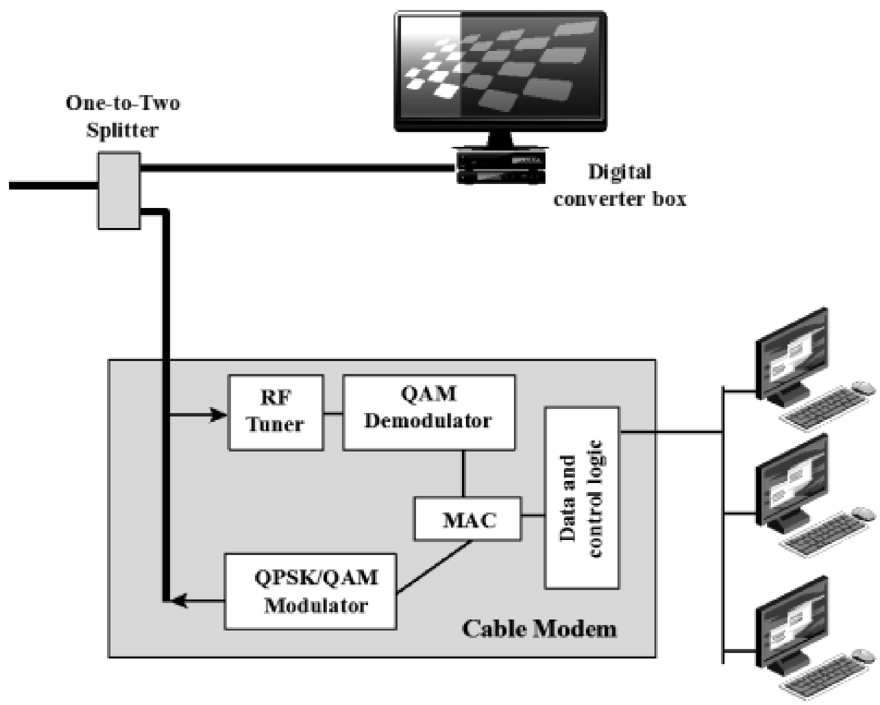

Figure 8.13 Cable Modem Configuration

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 (quadrature phase shift keying) or 16-QAM.

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Asymmetrical Digital Subscriber Line (ADSL)

- Link between subscriber and network
- Uses currently installed twisted pair cable
- Is Asymmetric - bigger downstream than up
- Uses Frequency Division Multiplexing
  - Reserve lowest 25kHz for voice (POTS)
  - Uses echo cancellation or FDM to give two bands
- Has a range of up to 5.5km

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. 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. 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), being a good fit to Internet requirements. ADSL uses frequency division multiplexing (FDM) in a novel way to exploit the 1-MHz capacity of twisted pair. When echo cancellation is used, the entire frequency band for the upstream channel overlaps the lower portion of the downstream channel. The ADSL scheme provides a range of up to 5.5 km, depending on the diameter of the cable and its quality.

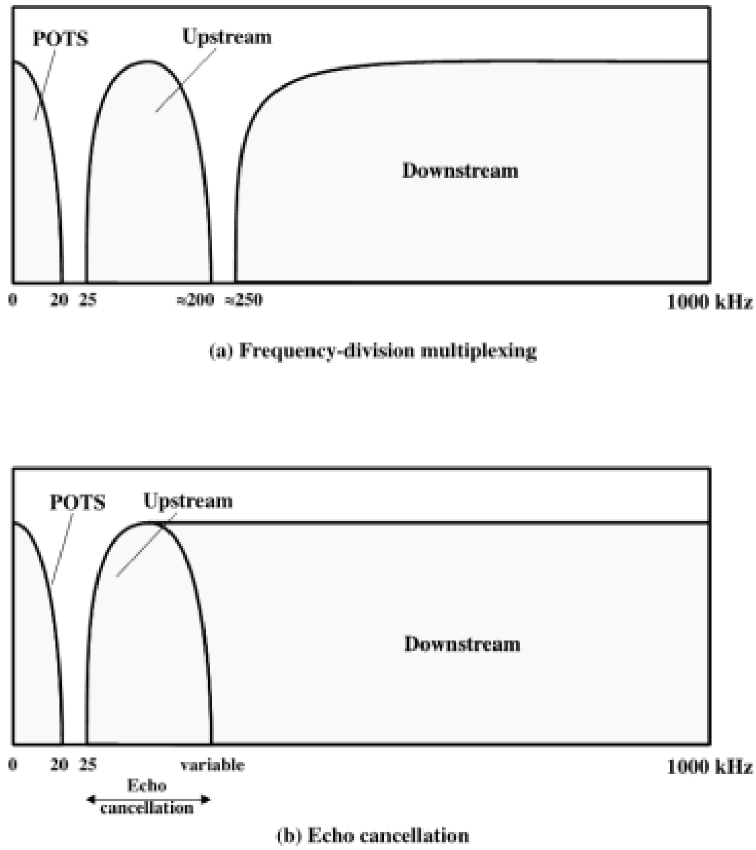

Figure 8.14 ADSL Channel Configuration

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 cancellation 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.

- 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 (DMT)

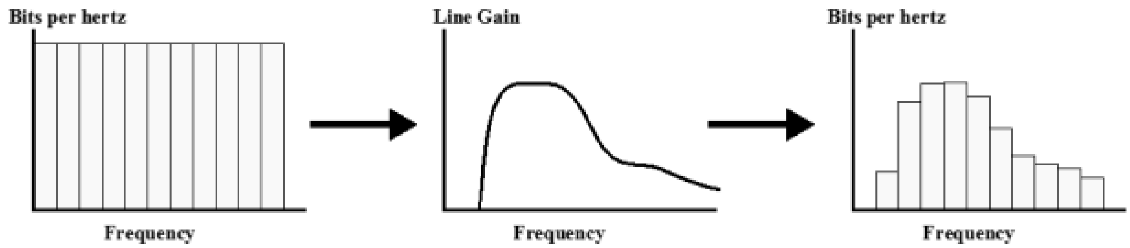

Figure 8.15 DMT Bits per Channel Allocation

- Multiple carrier signals at different frequencies
- Divide into 4kHz subchannels
- Test and use subchannels with better SNR
- 256 downstream subchannels at 4kHz (60kbps)
  - In theory 15.36Mbps, in practice 1.5-9Mbps

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, 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.

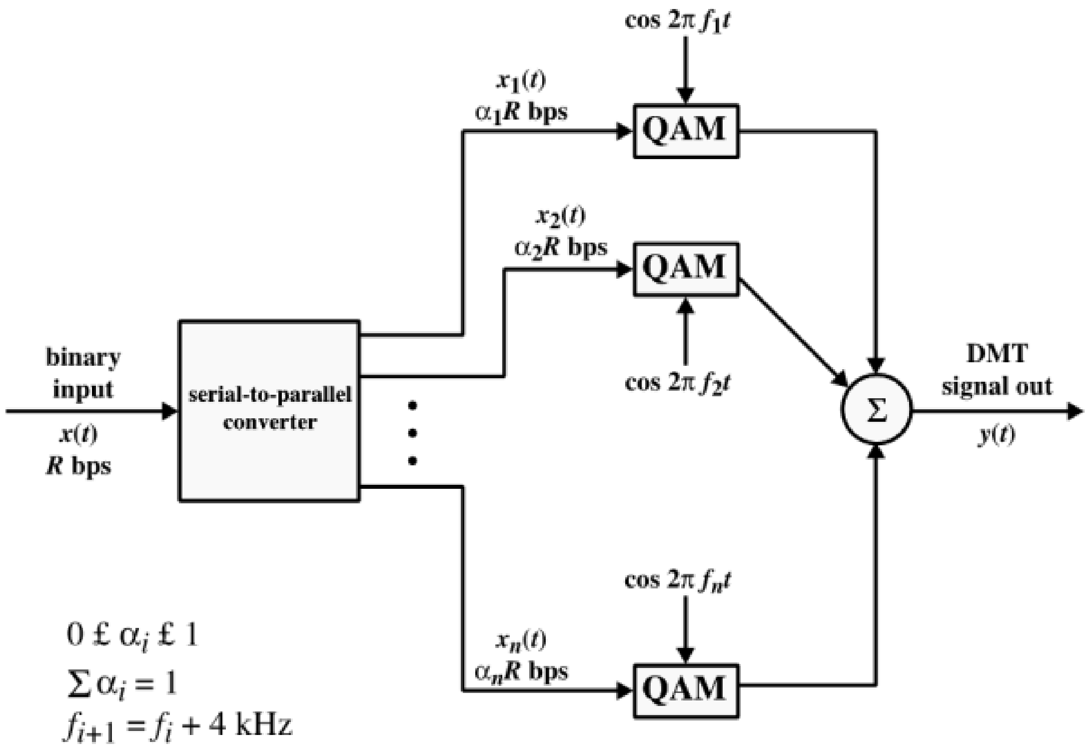

Figure 8.16 DMT Transmitter

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.

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.

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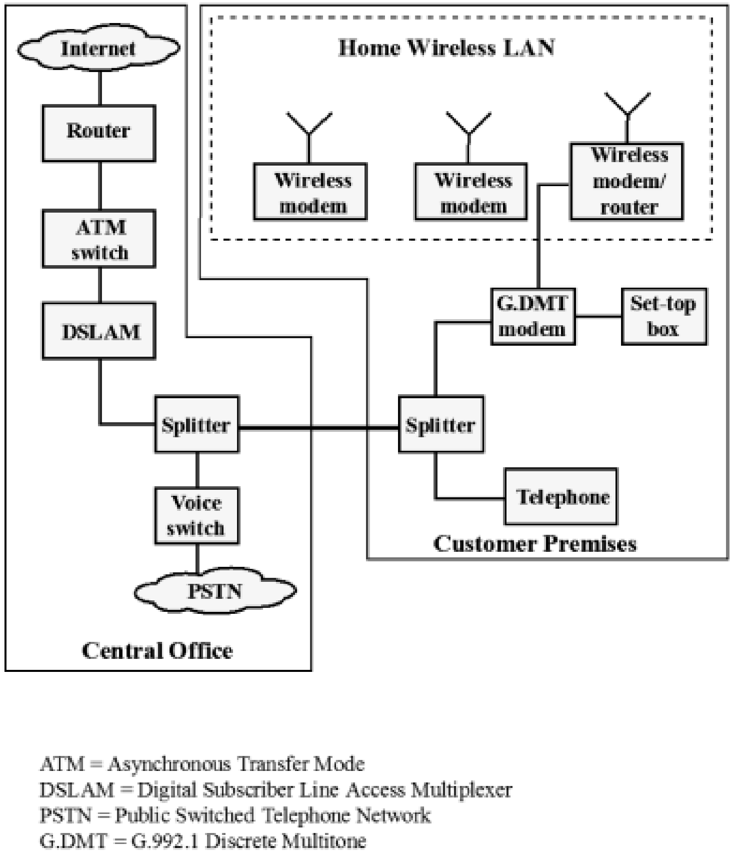

Figure 8.17 DSL Broadband Access

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.

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.

Table 8.6
Comparison of xDSL Alternatives

	ADSL	HDSL	SDSL	VDSL
Data rate	1.5 to 9 Mbps downstream 16 to 640 kbps upstream	1.544 or 2.048 Mbps	1.544 or 2.048 Mbps	13 to 52 Mbps downstream 1.5 to 2.3 Mbps upstream
Mode	Asymmetric	Symmetric	Symmetric	Asymmetric
Copper pairs	1	2	1	1
Range (24-gauge UTP)	3.7 to 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 to 5 MHz	196 kHz	196 kHz	≥ 10 MHz
Bits/cycle	Varies	4	4	Varies

UTP = unshielded twisted pair

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.

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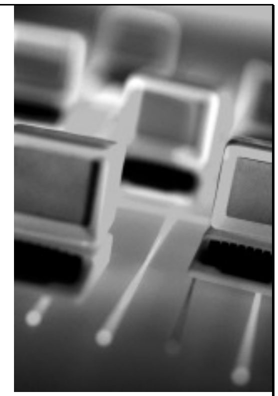
xDSL



- High data rate DSL (HDSL)
  - 2B1Q coding on dual twisted pairs
  - Up to 2Mbps over 3.7km
- Single line DSL
  - 2B1Q coding on single twisted pair (residential) with echo cancelling
  - Up to 2Mbps over 3.7km
- Very high data rate DSL
  - DMT/QAM for very high data rates
  - Separate bands for separate services

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.

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.

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

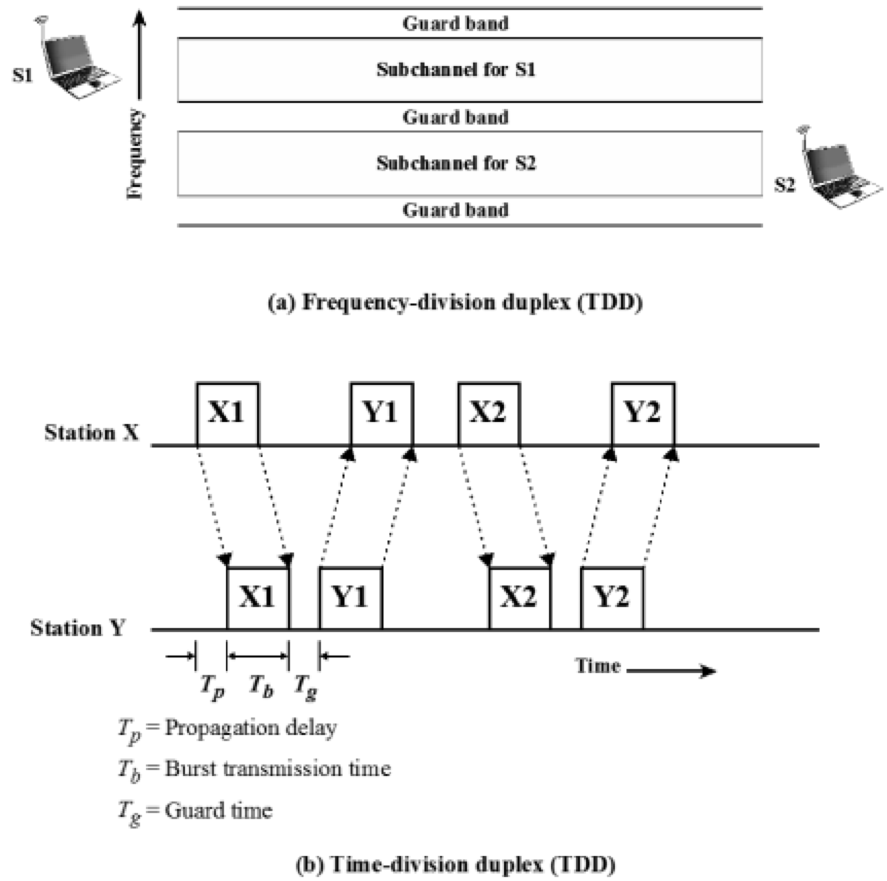


Figure 8.18 Duplex Access Techniques

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.

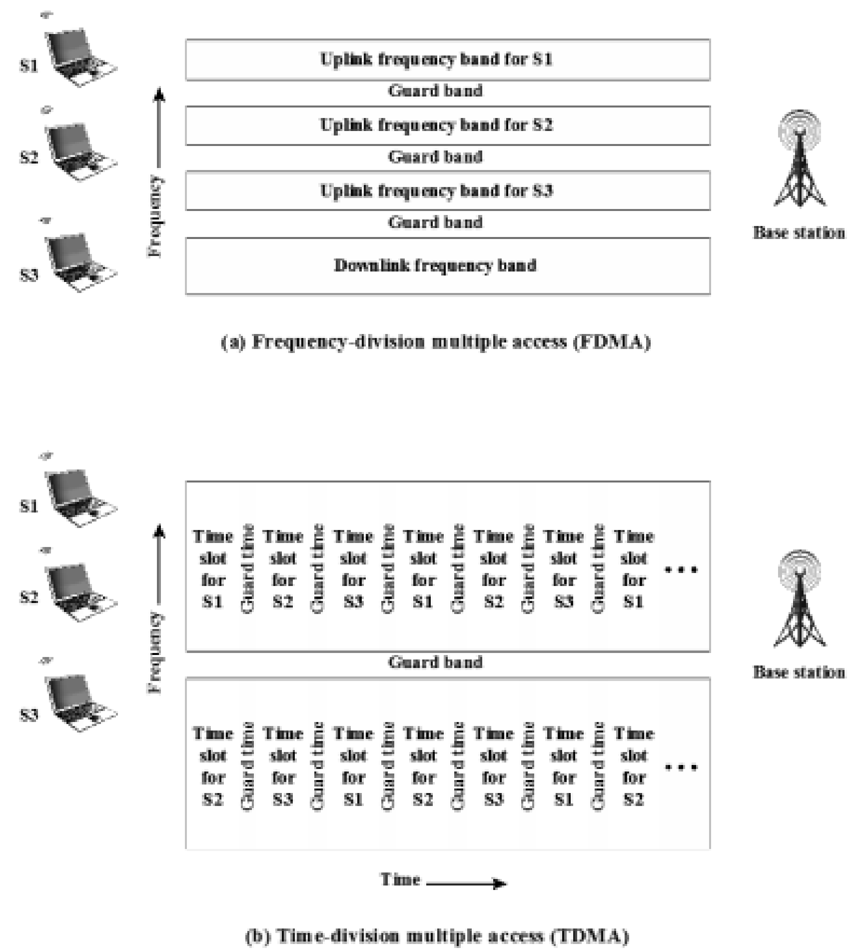


Figure 8.19 Multiple Channel Access Techniques

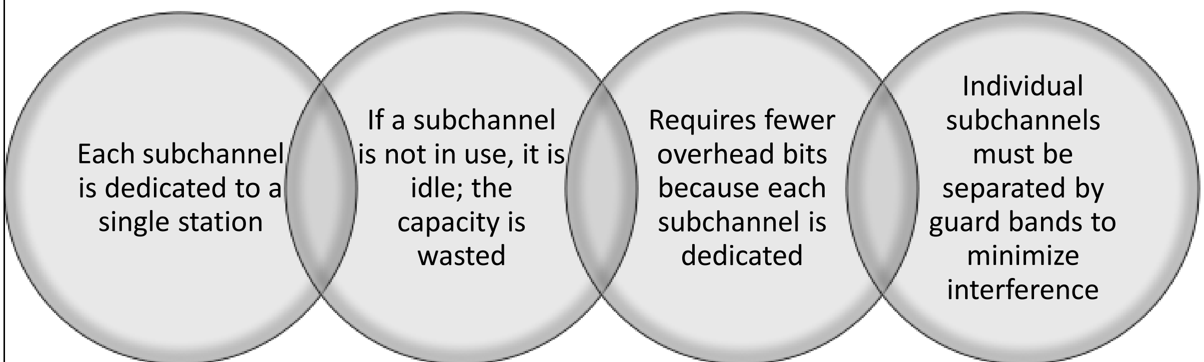
In TDD, also known as time-compression multiplexing (TCM) , data are transmitted in one direction at a time, with transmission alternating between the two directions. To achieve the desired subscriber data rate with simple TDD, the transmitter's bit stream is divided into equal segments, compressed in 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.

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

FDMA

- Frequency-Division Multiple Access
  - Technique used to share the spectrum among multiple stations
  - Base station assigns bandwidths to stations within the overall bandwidth available
  - Key features:



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).

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.

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. 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.

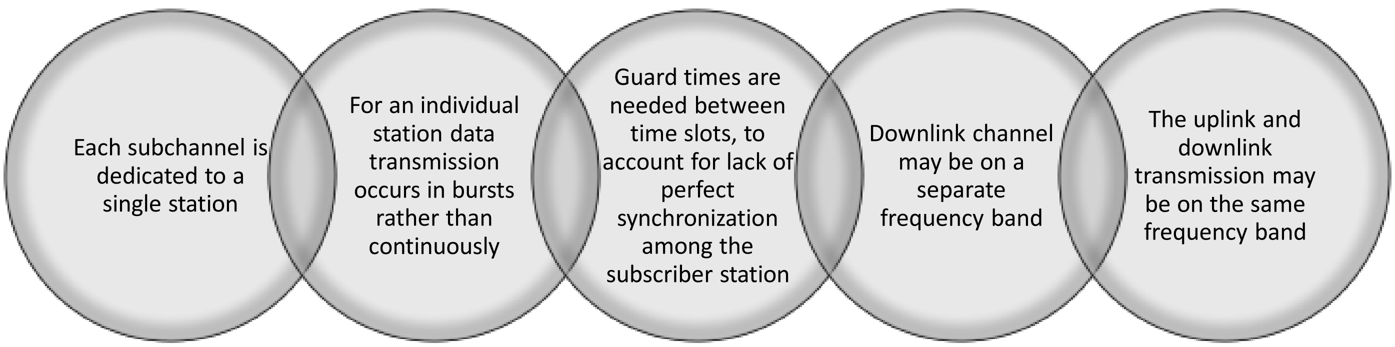
- Guard times are needed between time slots, to account for lack of perfect synchronization among the subscriber station.

- 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.

TDMA

- Time-Division Multiple Access
  - 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
  - Key features:



Summary

- Frequency-division multiplexing
  - Characteristics
  - Analog carrier systems
  - Wavelength division multiplexing
- Synchronous time-division multiplexing
  - Characteristics
  - TDM link control
  - Digital carrier systems
  - SONET/SDH
- Cable modems
- Asymmetric digital subscriber line
  - ADSL design
  - Discrete multitone
  - Broadband access configuration
- xDSL
  - High data rate digital subscriber line
  - Single-line digital subscriber line
  - Very high data rate digital subscriber line
- Multiple channel access
  - Frequency-division duplex (FDD)
  - Time-division duplex (TDD)
  - Frequency-division multiple access (FDMA)
  - Time-division multiple access (TDMA)

Chapter 8 summary.