MODULE IV

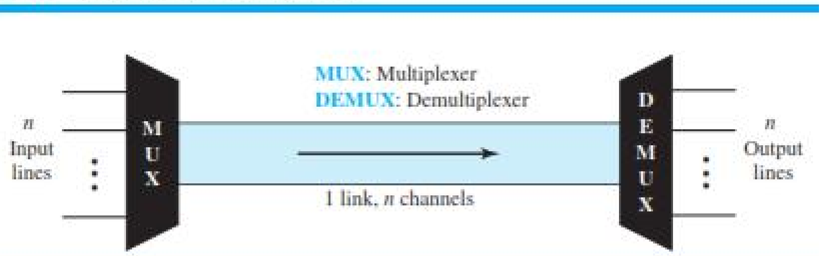
Syllabus	End Sem. Exam Marks
Multiplexing- Space Division Multiplexing, Frequency Division Multiplexing, Wave length Division Multiplexing - Time Division multiplexing: Characteristics, Digital Carrier system, SONET/SDH , Statistical time division multiplexing: Cable Modem, Code Division Multiplexing. Multiple Access– CDMA.	15 %

MULTIPLEXING

Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allow the simultaneous transmission of multiple signals across a single data link.

In a multiplexed system, n lines share the bandwidth of one link. Figure below shows the basic format of a multiplexed system. The lines on the left direct their transmission streams to a multiplexer (MUX), which combines them into a single stream (many-to-one). At the receiving end, that stream is fed into a demultiplexer (DEMUX), which separates the stream back into its component transmissions (one-to-many) and directs them to their corresponding lines. In the figure, the word link refers to the physical path. The word channel refers to the portion of a link that carries a transmission between a given pair of lines. One link can have many (n) channels.

Figure *Dividing a link into channels*



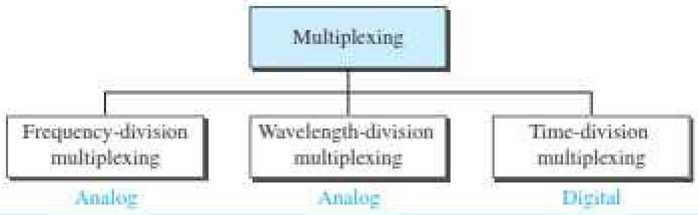
There are three basic multiplexing techniques:

a. frequency-division multiplexing
b. wavelength-division multiplexing

c. time-division multiplexing.

The first two are techniques designed for analog signals, the third, for digital signals.

Figure Categories of multiplexing



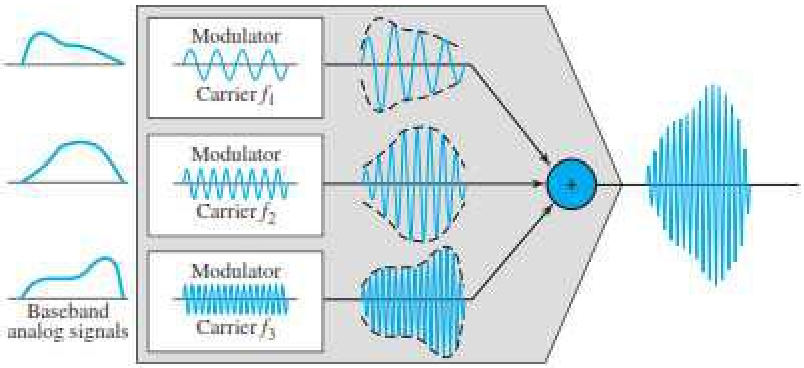
Frequency-Division Multiplexing

Frequency-division multiplexing (FDM) is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted. In FDM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. Channels can be separated by strips of unused bandwidth—guard bands—to prevent signals from overlapping.

Figure Frequency-division multiplexing

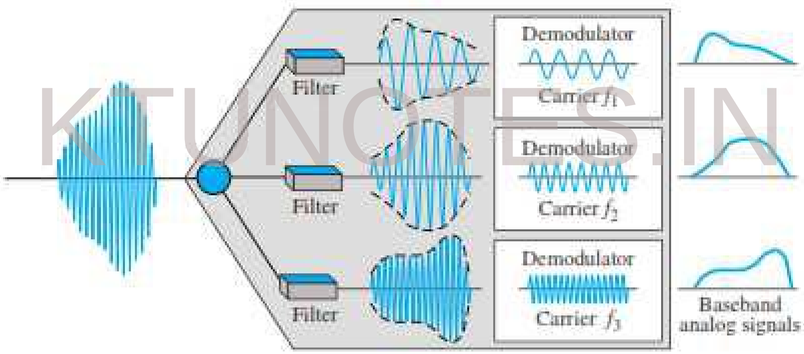
Each source generates a signal of a similar frequency range. Inside the multiplexer, these similar signals modulate different carrier frequencies (f_1, f_2, and f_3). The resulting modulated signals are then combined into a single composite signal that is sent out over a media link that has enough bandwidth to accommodate it.

Figure *FDM process*



The demultiplexer uses a series of filters to decompose the multiplexed signal into its constituent component signals. The individual signals are then passed to a demodulator that separates them from their carriers and passes them to the output lines.

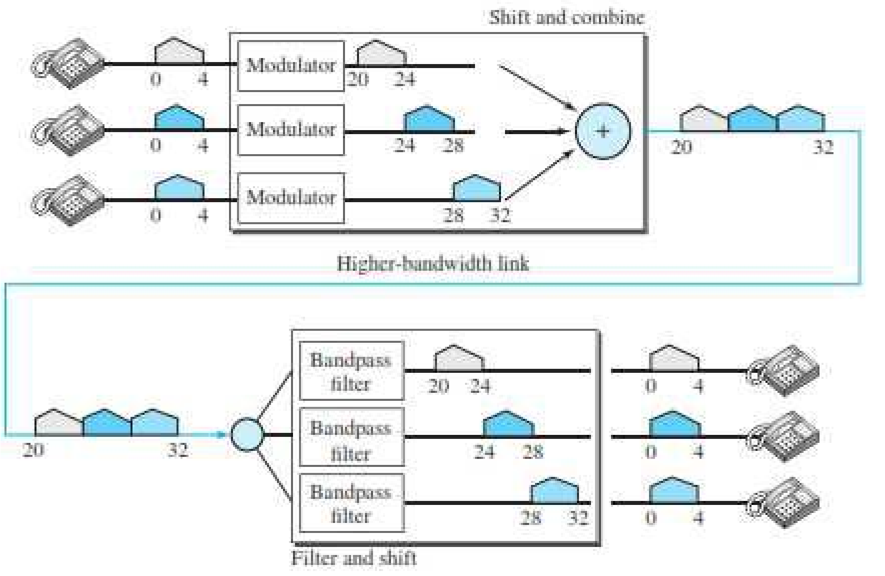
Figure *FDM demultiplexing example*



Example

Assume that a voice channel occupies a bandwidth of 4 kHz. We need to combine three voice channels into a link with a bandwidth of 12 kHz, from 20 to 32 kHz. Show the configuration, using the frequency domain. Assume there are no guard bands.

Figure Example



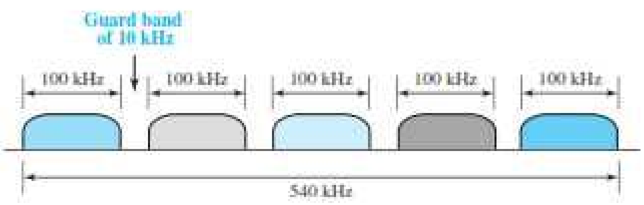
Example

Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference?

Solution

For five channels, we need at least four guard bands. This means that the required bandwidth is at least 5 × 100 + 4 × 10 = 540 kHz.

Figure Example



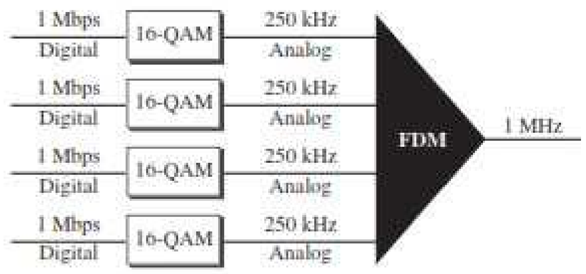
Example

Four data channels (digital), each transmitting at 1 Mbps, use a satellite channel of 1 MHz. Design an appropriate configuration, using FDM.

Solution

The satellite channel is analog. We divide it into four channels, each channel having a 250-kHz bandwidth. Each digital channel of 1 Mbps is modulated so that each 4 bits is modulated to 1 Hz. One solution is 16-QAM modulation.

Figure *Example*



Applications of FDM

A very common application of FDM is AM and FM radio broadcasting. Radio uses the air as the transmission medium. A special band from 530 to 1700 kHz is assigned to AM radio. All radio stations need to share this band. Each AM station needs 10 kHz of bandwidth. Each station uses a different carrier frequency, which means it is shifting its signal and multiplexing. The signal that goes to the air is a combination of signals. A receiver receives all these signals, but filters (by tuning) only the one which is desired. Without multiplexing, only one AM station could broadcast to the common link, the air.

The situation is similar in FM broadcasting. FM has a wider band of 88 to 108 MHz because each station needs a bandwidth of 200 kHz.

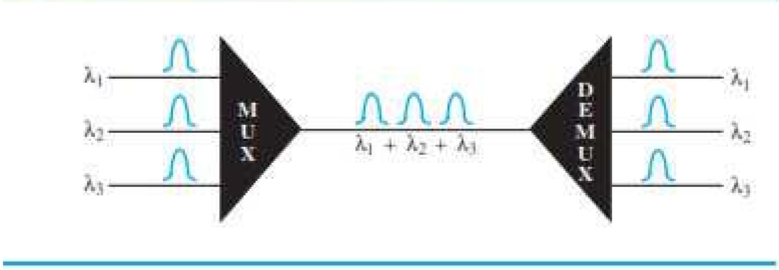
Another common use of FDM is in television broadcasting. Each TV channel has its own bandwidth of 6 MHz.

The first generation of cellular telephones also uses FDM. Each user is assigned two 30-kHz channels, one for sending voice and the other for receiving. The voice signal, which has a bandwidth of 3 kHz (from 300 to 3300 Hz), is modulated by using FM.

Wavelength-Division Multiplexing

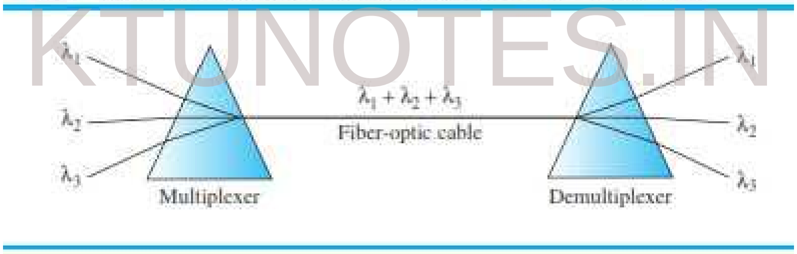
Wavelength-division multiplexing (WDM) is designed to use the high-data-rate capability of fiber-optic cable. The optical fiber data rate is higher than the data rate of metallic transmission cable, but using a fiber-optic cable for a single line wastes the available bandwidth. Multiplexing allows us to combine several lines into one. WDM is conceptually the same as FDM, except that the multiplexing and demultiplexing involve optical signals transmitted through fiber-optic channels. The idea is the same: We are combining different signals of different frequencies. The difference is that the frequencies are very high.

Figure *Wavelength-division multiplexing*



To combine multiple light sources into one single light at the multiplexer and do the reverse at the demultiplexer. The combining and splitting of light sources are easily handled by a prism.

Figure *Prisms in wavelength-division multiplexing and demultiplexing*

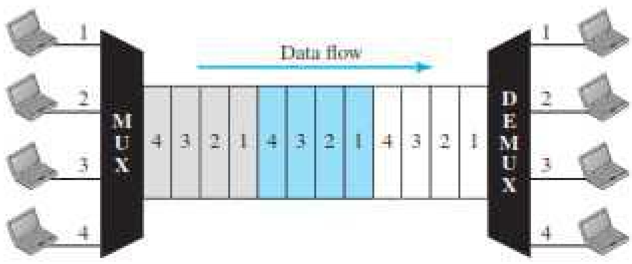


Time-Division Multiplexing

Time-division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a link. Instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link.

Figure below gives a conceptual view of TDM. The same link is used as in FDM; here, however, the link is shown sectioned by time rather than by frequency. In the figure, portions of signals 1, 2, 3, and 4 occupy the link sequentially.

Figure *TDM*



TDM is divided into two different schemes:

a. Synchronous TDM
b. Statistical TDM

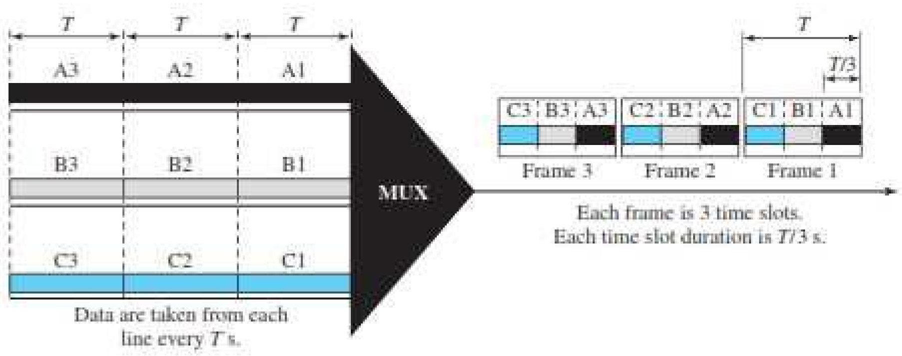
Synchronous TDM

In synchronous TDM, each input connection has an allotment in the output even if it is not sending data. TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

Time Slots and Frames

In synchronous TDM, the data flow of each input connection is divided into units, where each input occupies one input time slot. A unit can be 1 bit, one character, or one block of data. Each input unit becomes one output unit and occupies one output time slot. However, the duration of an output time slot is n times shorter than the duration of an input time slot. If an input time slot is T sec, the output time slot is T/n s, where n is the number of connections. In other words, a unit in the output connection has a shorter duration; it travels faster.

Figure *Synchronous time-division multiplexing*



In synchronous TDM, a round of data units from each input connection is collected into a frame. If we have n connections, a frame is divided into n time slots and one slot is allocated for each unit, one for each input line. If the duration of the input unit is T, the duration of each slot is T/n and the duration of each frame is T. The data rate of the output link must be n times the data rate of a connection to guarantee the flow of data.

Example

In a TDM system with n = 3, the data rate for each input connection is 1 kbps. If 1 bit at a time is multiplexed (a unit is 1 bit), what is the duration of

1. Each input slot,
2. Each output slot, and
3. Each frame?

Solution

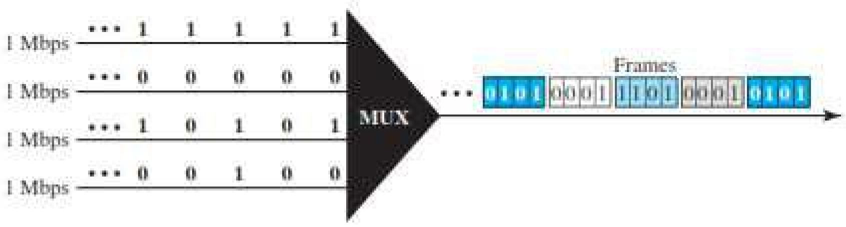
1. The data rate of each input connection is 1 kbps. This means that the bit duration is 1/1000s or 1 ms. .The duration of the input time slot is 1 ms (same as bit duration).
2. The duration of each output time slot is one-third of the input time slot. This means that the duration of the output time slot is 1/3 ms.
3. Each frame carries three output time slots. So the duration of a frame is 3 × 1/3 ms, or 1 ms .The duration of a frame is the same as the duration of an input unit.

Example

Figure below shows synchronous TDM with a data stream for each input and one data stream for the output. The unit of data is 1 bit. Find

(1) The input bit duration
(2) The output bit duration,
(3) The output bit rate
(4) The output frame rate.

Figure *Example*



Solution:

1. The input bit duration is the inverse of the bit rate: 1/1 Mbps = 1 μs.
2. The output bit duration is one-fourth of the input bit duration, or 1/4 μs.
3. The output bit rate is the inverse of the output bit duration, or 1/4 μs, or 4 Mbps. This can also be deduced from the fact that the output rate is 4 times as fast as any input rate; so the output rate = 4 × 1 Mbps = 4 Mbps.
4. The frame rate is always the same as any input rate. So the frame rate is 1,000,000 frames per second.

Example

Four 1-kbps connections are multiplexed together. A unit is 1 bit. Find (1) the duration of 1 bit before multiplexing, (2) the transmission rate of the link, (3) the duration of a time slot, and (4) the duration of a frame.

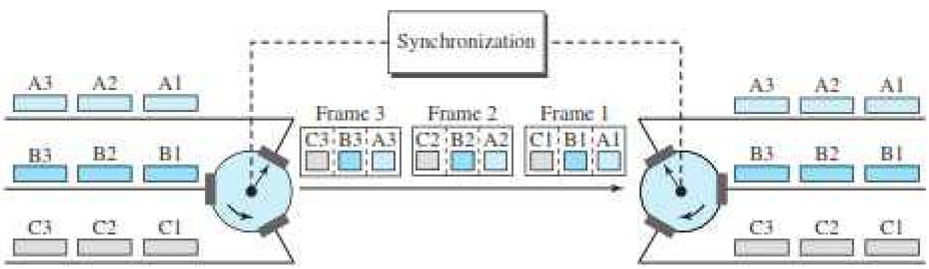
Solution

1. The duration of 1 bit before multiplexing is 1/1 kbps, or 0.001 s (1 ms).
2. The rate of the link is 4 times the rate of a connection, or 4 kbps.
3. The duration of each time slot is one-fourth of the duration of each bit before multiplexing, or 1/4 ms or 250 μs. Note that we can also calculate this from the data rate of the link, 4 kbps. The bit duration is the inverse of the data rate, or 1/4 kbps or 250 μs.
4. The duration of a frame is always the same as the duration of a unit before multiplexing, or1 ms. we can also calculate this in another way. Each frame in this case has four time slots. So the duration of a frame is 4 times 250 μs, or 1 ms.

Interleaving

TDM can be visualized as two fast-rotating switches, one on the multiplexing side and the other on the demultiplexing side. The switches are synchronized and rotate at the same speed, but in opposite directions. On the multiplexing side, as the switch opens in front of a connection, that connection has the opportunity to send a unit onto the path. This process is called interleaving. On the demultiplexing side, as the switch opens in front of a connection, that connection has the opportunity to receive a unit from the path.

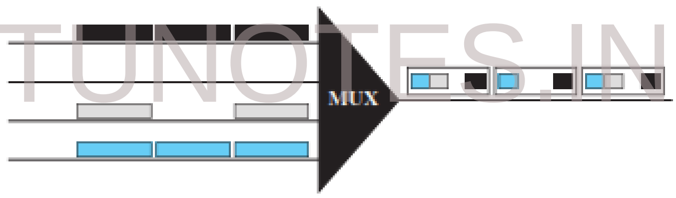
Figure *Interleaving*



Empty Slots

Synchronous TDM is not as efficient as it could be. If a source does not have data to send, the corresponding slot in the output frame is empty. Figure below shows a case in which one of the input lines has no data to send and one slot in another input line has discontinuous data. The first output frame has three slots filled, the second frame has two slots filled, and the third frame has three slots filled. No frame is full.

Figure *Empty slots*



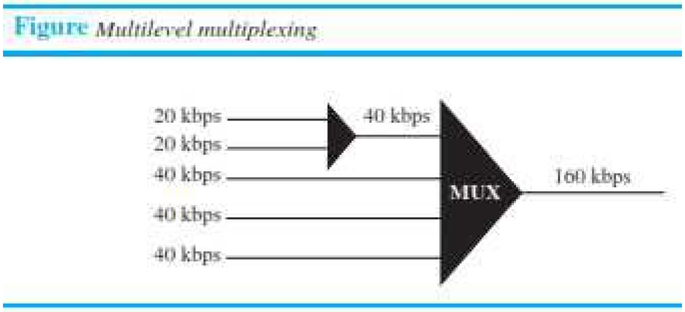
Data Rate Management

One problem with TDM is how to handle a disparity in the input data rates. So far, we assumed that the data rates of all input lines were the same. However, if data rates are not the same, three strategies, or a combination of them, can be used:-

- Multilevel multiplexing
- Multiple-slot allocation
- Pulse stuffing.

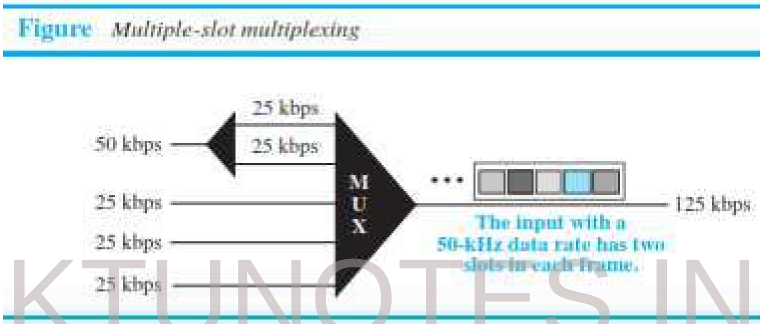
Multilevel Multiplexing

Multilevel multiplexing is a technique used when the data rate of an input line is a multiple of others. For example, in Figure below, we have two inputs of 20 kbps and three inputs of 40 kbps. The first two input lines can be multiplexed together to provide a data rate equal to the last three. A second level of multiplexing can create an output of 160 kbps.



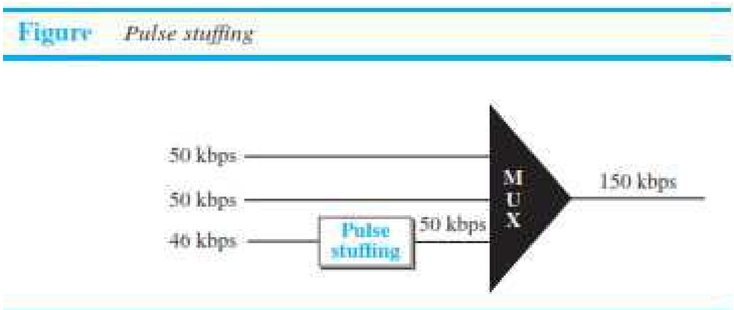
Multiple-Slot Allocation

Sometimes it is more efficient to allot more than one slot in a frame to a single input line. For example, we might have an input line that has a data rate that is a multiple of another input. In Figure below, the input line with a 50-kbps data rate can be given two slots in the output. We insert a demultiplexer in the line to make two inputs out of one.



Pulse Stuffing

Sometimes the bit rates of sources are not multiple integers of each other. Therefore, neither of the above two techniques can be applied. One solution is to make the highest input data rate the dominant data rate and then add dummy bits to the input lines with lower rates. This will increase their rates. This technique is called pulse stuffing, bit padding, or bit stuffing. In the figure below, the input with a data rate of 46 is pulse-stuffed to increase the rate to 50 kbps. Now multiplexing can take place.

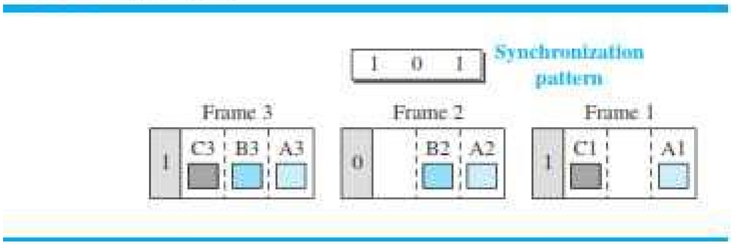


Frame Synchronizing

The implementation of TDM is not as simple as that of FDM. Synchronization between the multiplexer and demultiplexer is a major issue. If the multiplexer and the demultiplexer are not synchronized, a bit belonging to one channel may be received by the wrong channel.

For this reason, one or more synchronization bits are usually added to the beginning of each frame. These bits, called framing bits, follow a pattern, frame to frame, that allows the demultiplexer to synchronize with the incoming stream so that it can separate the time slots accurately. In most cases, this synchronization information consists of 1 bit per frame, alternating between 0 and 1,

Figure *Framing bits*



Example

We have four sources, each creating 250 characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (1) the data rate of each source, (2) the duration of each character in each source, (3) the frame rate, (4) the duration of each frame, (5) the number of bits in each frame, and (6) the data rate of the link.

Solution

1. The data rate of each source is 250 × 8 = 2000 bps = 2 kbps.
2. Each source sends 250 characters per second; therefore, the duration of a character is 1/250 s, or 4 ms.
3. Each frame has one character from each source, which means the link needs to send 250 frames per second to keep the transmission rate of each source.
4. The duration of each frame is 1/250 s, or 4 ms. Note that the duration of each frame is the same as the duration of each character coming from each source.
5. Each frame carries 4 characters and 1 extra synchronizing bit. This means that each frame is 4 × 8 + 1 = 33 bits.

6. The link sends 250 frames per second, and each frame contains 33 bits. This means that the data rate of the link is 250 × 33, or 8250 bps. Note that the bit rate of the link is greater than the combined bit rates of the four channels. If we add the bit rates of four channels, we get 8000 bps. Because 250 frames are traveling per second and each contains 1 extra bit for synchronizing, we need to add 250 to the sum to get 8250 bps.

Example

Two channels, one with a bit rate of 100 kbps and another with a bit rate of 200 kbps, are to be multiplexed. How this can be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?

Solution

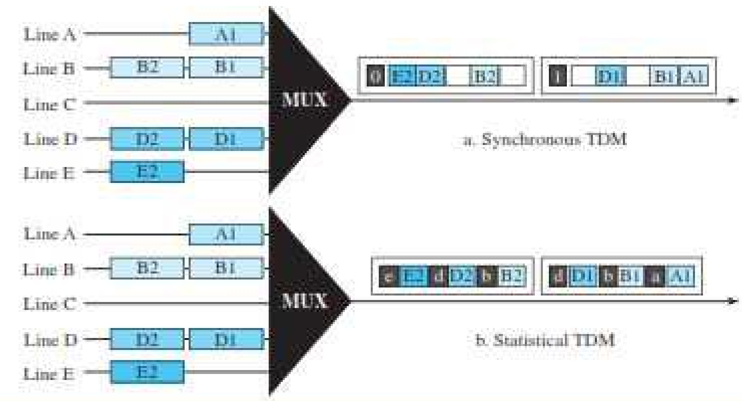
We can allocate one slot to the first channel and two slots to the second channel. Each frame carries 3 bits. The frame rate is 100,000 frames per second because it carries 1 bit from the first channel. The frame duration is 1/100,000 s, or 10 ms. The bit rate is 100,000 frames/s × 3 bits per frame, or 300 kbps. Note that because each frame carries 1 bit from the first channel, the bit rate for the first channel is preserved. The bit rate for the second channel is also preserved because each frame carries 2 bits from the second channel.

Statistical Time-Division Multiplexing

In synchronous TDM, each input has a reserved slot in the output frame. This can be inefficient if some input lines have no data to send. In statistical time-division multiplexing, slots are dynamically allocated to improve bandwidth efficiency. Only when an input line has a slot's worth of data to send is it given a slot in the output frame. In statistical multiplexing, the number of slots in each frame is less than the number of input lines. The multiplexer checks each input line in round robin fashion; it allocates a slot for an input line if the line has data to send; otherwise, it skips the line and checks the next line.

Figure below shows a synchronous and a statistical TDM example. In the former, some slots are empty because the corresponding line does not have data to send. In the latter, however, no slot is left empty as long as there are data to be sent by any input line.

Figure *TDM slot comparison*



Addressing

An output slot in synchronous TDM is totally occupied by data; in statistical TDM, a slot needs to carry data as well as the address of the destination. In synchronous TDM, there is no need for addressing; synchronization and pre-assigned relationships between the inputs and outputs serve as an address. We know, for example, that input 1 always goes to Input 2. If the multiplexer and the demultiplexer are synchronized, this is guaranteed.

In statistical multiplexing, there is no fixed relationship between the inputs and outputs because there are no pre-assigned or reserved slots. We need to include the address of the receiver inside each slot to show where it is to be delivered. The addressing in its simplest form can be n bits to define N different output lines with $n = \log_2 N$. For example, for eight different output lines, we need a 3-bit address.

Slot Size

Since a slot carries both data and an address in statistical TDM, the ratio of the data size to address size must be reasonable to make transmission efficient. For example, it would be inefficient to send 1 bit per slot as data when the address is 3 bits. This would mean an overhead of 300 percent. In statistical TDM, a block of data is usually many bytes while the address is just a few bytes.

No Synchronization Bit

There is another difference between synchronous and statistical TDM, but this time it is at the frame level. The frames in statistical TDM need not be synchronized, so we do not need synchronization bits.

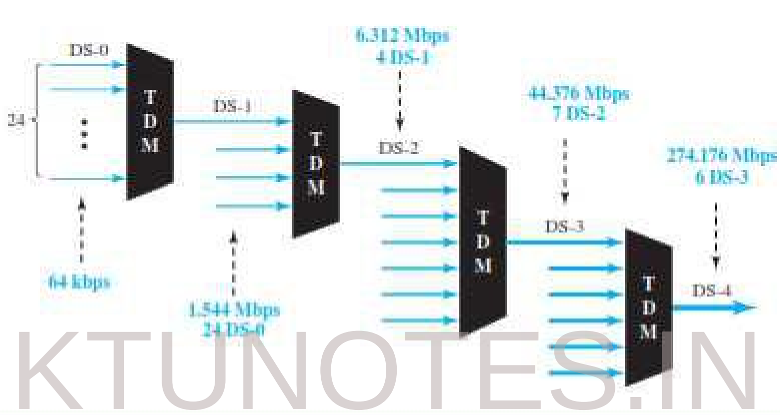
Bandwidth

In statistical TDM, the capacity of the link is normally less than the sum of the capacities of each channel. The designers of statistical TDM define the capacity of the link based on the statistics of the load for each channel. If on average only x percent of the input slots are filled, the capacity of the link reflects this. Of course, during peak times, some slots need to wait.

Digital Signal Service

Telephone companies implement TDM through a hierarchy of digital signals, called digital signal (DS) service or digital hierarchy. Figure below shows the data rates supported by each level.

Figure *Digital hierarchy*



- DS-0 is a single digital channel of 64 kbps.
- DS-1 is a 1.544-Mbps service; it can be used to multiplex 24 DS-0 channels
- DS-2 is a 6.312-Mbps service; it can be used to multiplex 4 DS-1 channels, 96 DS-0 channels, or a combination of these service types.
- DS-3 is a 44.376-Mbps service; it can be used to multiplex 7 DS-2 channels, 28 DS-1 channels, 672 DS-0 channels, or a combination of these service types.
- DS-4 is a 274.176-Mbps service; It can be used to multiplex 6 DS-3 channels, 42 DS-2 channels, 168 DS-1 channels, 4032 DS-0 channels, or a combination of these service types.

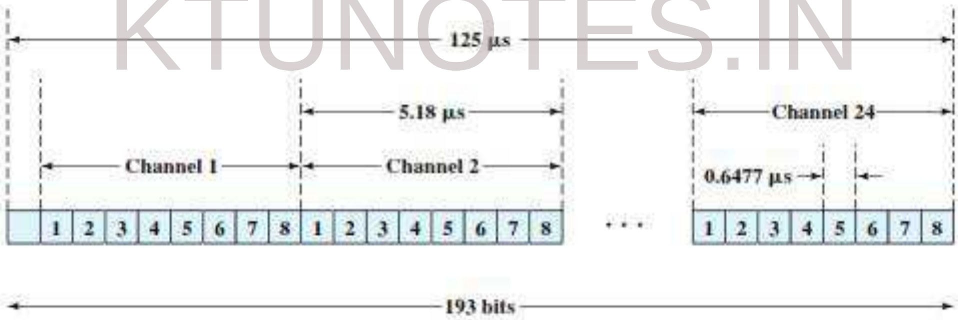
T Lines

DS-0, DS-1, and so on are the names of services. To implement those services, the telephone companies use T lines (T-1 to T-4). These are lines with capacities precisely matched to the data rates of the DS-1 to DS-4 services. So far only T-1 and T-3 lines are commercially available.

Table *DS and T line rates*

Service	*Line*	*Rate (Mbps)*	*Voice Channels*
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

The basis of the TDM hierarchy is the DS-1 transmission format, which multiplexes 24 channels. Each frame contains 8 bits per channel plus a framing bit for 24 X 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 (PCM) 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 X 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 signalling bit. The signalling 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.



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 DS-1 Transmission Format

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 X 8000 = 48kbps. 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 sub channels share this channel. The data for each sub channel 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 the remaining capacity is used for framing and control bits.

SONET/SDH

SONET (Synchronous Optical Network) is an optical transmission interface originally proposed by Bell Core and standardized by ANSI. A compatible version, referred to as Synchronous Digital Hierarchy (SDH), has been published by ITU-T. SONET is intended to provide a specification for taking advantage of the high-speed digital transmission capability of optical fiber. SONET/SDH is a synchronous network using synchronous TDM multiplexing. All clocks in the system are locked to a master clock.

Architecture

The architecture of a SONET system is divided into three portions: signals, devices, and connections.

Signals

SONET defines a hierarchy of electrical signalling levels called synchronous transport signals (STSs). Each STS level (STS-1 to STS-192) supports a certain data rate, specified in megabits per second. The corresponding optical signals are called optical carriers (OCs). SDH specifies a similar system called a synchronous transport module (STM). STM is intended to be compatible with existing European hierarchies, such as E lines, and with STS levels.

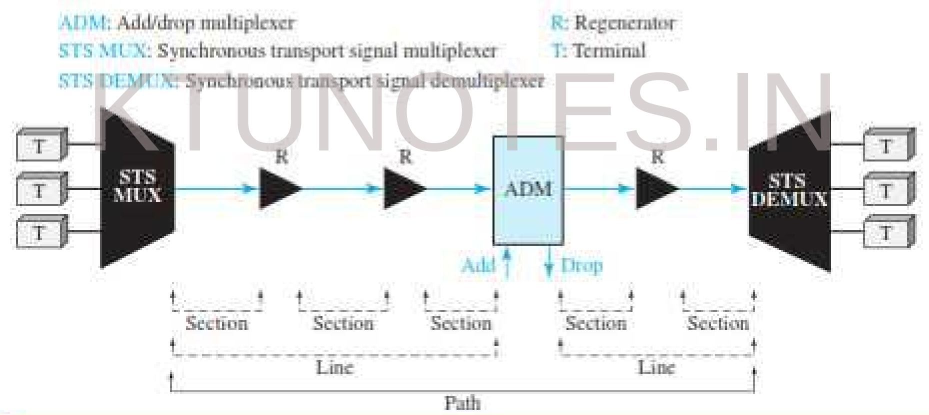
Table *SONET/SDH rates*

STS	*OC*	*Rate (Mbps)*	*STM*
STS-1	OC-1	51.840	
STS-3	OC-3	155.520	STM-1
STS-9	OC-9	466.560	STM-3
STS-12	OC-12	622.080	STM-4
STS-18	OC-18	933.120	STM-6
STS-24	OC-24	1244.160	STM-8
STS-36	OC-36	1866.230	STM-12
STS-48	OC-48	2488.320	STM-16
STS-96	OC-96	4976.640	STM-32
STS-192	OC-192	9953.280	STM-64

SONET Devices

SONET transmission relies on three basic devices: STS multiplexers/demultiplexer, regenerators, add/drop multiplexers and terminals.

Figure *A simple network using SONET equipment*



STS Multiplexer/Demultiplexer

STS multiplexers/demultiplexers mark the beginning points and endpoints of a SONET link. They provide the interface between an electrical network and the optical network. An STS multiplexer multiplexes signals from multiple electrical sources and creates the corresponding optical signal. An STS demultiplexer demultiplexes an optical signal into corresponding electric signals.

Regenerator

Regenerators extend the length of the links. A regenerator is a repeater that takes a received optical signal, demodulates it into the corresponding electric signal, regenerates the electric signal, and finally modulates the electric signal into its correspondent optical signal. A SONET regenerator replaces some of the existing overhead information (header information) with new information.

Add/drop Multiplexer

Add/drop multiplexers allow insertion and extraction of signals. An add/drop multiplexer (ADM) can add STSs coming from different sources into a given path or can remove a desired signal from a path and redirect it without demultiplexing the entire signal. Instead of relying on timing and bit positions, add/drop multiplexers use header information such as addresses and pointers to identify individual streams.

A number of incoming electronic signals are fed into an STS multiplexer, where they are combined into a single optical signal. The optical signal is transmitted to a regenerator, where it is recreated without the noise it has picked up in transit. The regenerated signals from a number of sources are then fed into an add/drop multiplexer. The add/drop multiplexer reorganizes these signals, if necessary, and sends them out as directed by information in the data frames. These remultiplexed signals are sent to another regenerator and from there to the receiving STS demultiplexer, where they are returned to a format usable by the receiving links.

Terminals

A terminal is a device that uses the services of a SONET network. For example, in the Internet, a terminal can be a router that needs to send packets to another router at the other side of a SONET network.

Connections

The devices defined in the previous section are connected using sections, lines, and paths.

Sections

A section is the optical link connecting two neighbouring devices: multiplexer to multiplexer, multiplexer to regenerator, or regenerator to regenerator.

Lines

A line is the portion of the network between two multiplexers: STS multiplexer to add/drop multiplexer, two add/drop multiplexers, or two STS multiplexers.

Paths

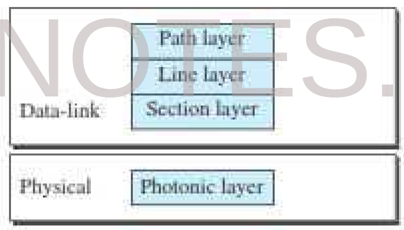
A path is the end-to-end portion of the network between two STS multiplexers. In a simple SONET of two STS multiplexers linked directly to each other, the section, line, and path are the same.

SONET Layers

The SONET standard includes four functional layers: the photonic, the section, the line, and the path layer. They correspond to both the physical and the data-link layers.

SONET defines four layers: path, line, section, and photonic.

Figure *SONET layers compared with OSI or Internet layers*



Path Layer

The path layer is responsible for the movement of a signal from its optical source to its optical destination. At the optical source, the signal is changed from an electronic form into an optical form, multiplexed with other signals, and encapsulated in a frame. At the optical destination, the received frame is demultiplexed, and the individual optical signals are changed back into their electronic forms. Path layer overhead is added at this layer. STS multiplexers provide path layer functions.

Line Layer

The line layer is responsible for the movement of a signal across a physical line. Line layer overhead is added to the frame at this layer. STS multiplexers and add/drop multiplexers provide line layer functions.

Section Layer

The section layer is responsible for the movement of a signal across a physical section. It handles framing, scrambling, and error control. Section layer overhead is added to the frame at this layer.

Photonic Layer

The photonic layer corresponds to the physical layer of the OSI model. It includes physical specifications for the optical fiber channel, the sensitivity of the receiver, multiplexing functions, and so on. SONET uses NRZ encoding, with the presence of light representing 1 and the absence of light representing 0.

SONET Frames

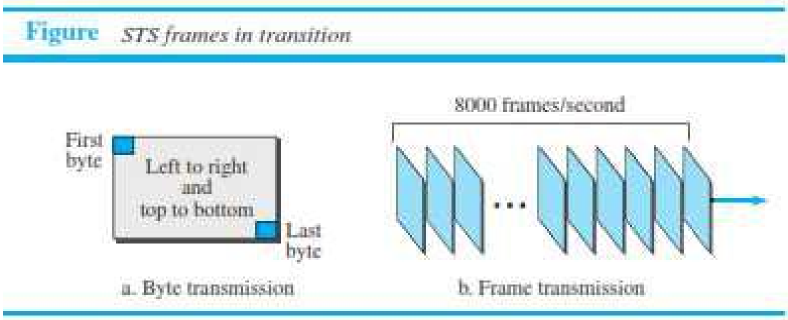
Each synchronous transfer signal STS-n is composed of 8000 frames. Each frame is a two-dimensional matrix of bytes with 9 rows by 90 × n columns. For example, an STS-1 frame is 9 rows by 90 columns (810 bytes), and an STS-3 is 9 rows by 270 columns (2430 bytes).

Figure *STS-1 and STS-n frames*

Frame, Byte, and Bit Transmission

One of the interesting points about SONET is that each STS-n signal is transmitted at a fixed rate of 8000 frames per second. This is the rate at which voice is digitized. For each frame the bytes are transmitted from the left to the right, top to the bottom. For each byte, the bits are transmitted from the most significant to the least significant (left to right).

If we sample a voice signal and use 8 bits (1 byte) for each sample, we can say that each byte in a SONET frame can carry information from a digitized voice channel. In other words, an STS-1 signal can carry 774 voice channels simultaneously



- **Each byte in a SONET frame can carry a digitized voice channel.**
- **In SONET, the data rate of an STS-n signal is n times the data rate of an STS-1 signal.**
- **In SONET, the duration of any frame is 125 μs.**

Example

Find the data rate of an STS-1 signal.

Solution

STS-1, like other STS signals, sends 8000 frames per second. Each STS-1 frame is made of 9 by (1 × 90) bytes. Each byte is made of 8 bits. The data rate is

$$\text{STS-1 data rate} = 8000 \times 9 \times (1 \times 90) \times 8 = 51.840 \text{ Mbps}$$

Example

Find the data rate of an STS-3 signal.

Solution

STS-3, like other STS signals, sends 8000 frames per second. Each STS-3 frame is made of 9 by (3 × 90) bytes. Each byte is made of 8 bits. The data rate is

$$\text{STS-3 data rate} = 8000 \times 9 \times (3 \times 90) \times 8 = 155.52 \text{ Mbps}$$

Example

What is the duration of an STS-1 frame? STS-3 frame? STS-n frame?

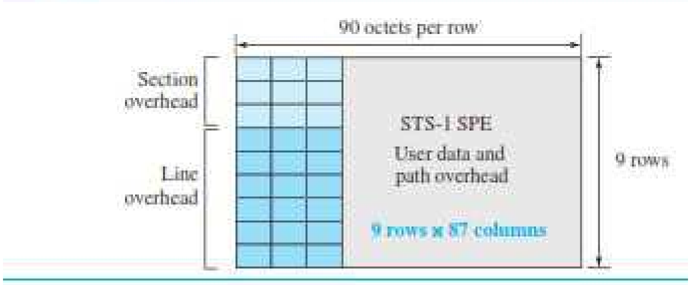
Solution

In SONET, 8000 frames are sent per second. This means that the duration of an STS-1, STS-3, or STS-n frame is the same and equal to 1/8000 s, or 125 μs.

STS-1 Frame Format

A SONET frame is a matrix of 9 rows of 90 bytes (octets) each, for a total of 810 bytes. The first three columns of the frame are used for section and line overhead. The upper three rows of the first three columns are used for section overhead (SOH). The lower six are line overhead (LOH). The rest of the frame is called the synchronous payload envelope (SPE). It contains user data and path overhead (POH) needed at the user data level.

Figure *STS-1 frame overheads*



Section Overhead

The section overhead consists of nine octets. The labels, functions, and organization of these octets are shown in Figure below.

Figure *STS-1 frame: section overhead*

- **Alignment bytes (A1 and A2)**:- Bytes A1 and A2 are used for framing and synchronization and are called alignment bytes. These bytes alert a receiver that a frame is arriving and give the receiver a predetermined bit pattern on which to synchronize. The bit patterns for these two bytes in hexadecimal are 0xF628. The bytes serve as a flag.
- **Section parity byte (B1):-** Byte B1 is for bit interleaved parity. Its value is calculated over all bytes of the previous frame. In other words, the ith bit of this byte is the parity bit calculated over all ith bits of the previous STS-n frame. The value of this byte is filled only for the first STS-1 in an STS-n frame. In other words, although an STS-n frame has n B1 bytes, only the first byte has this value; the rest are filled with 0s.
- **Identification byte (C1):-** Byte C1 carries the identity of the STS-1 frame. This byte is necessary when multiple STS-1s are multiplexed to create a higher-rate STS (STS-3, STS-9, STS-12, etc.). Information in this byte allows the various signals to be

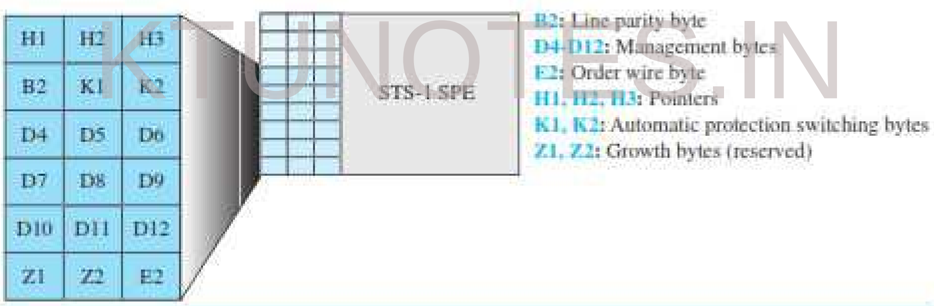
recognized easily upon demultiplexing. For example, in an STS-3 signal, the value of the C1 byte is 1 for the first STS-1; it is 2 for the second; and it is 3 for the third.

- **Management bytes (D1, D2, and D3):-** Bytes D1, D2, and D3 together form a 192-kbps channel (3 × 8000 × 8) called the data communication channel. This channel is required for operation, administration, and maintenance (OA&M) signalling.
- **Order wire byte (E1):-** Byte E1 is the order wire byte. Order wire bytes in consecutive frames form a channel of 64kbps (8000 frames per second times 8 bits per frame). This channel is used for communication between regenerators, or between terminals and regenerators.
- **User's byte (F1):-** The F1 bytes in consecutive frames form a 64-kbps channel that is reserved for user needs at the section level.

Line Overhead

Line overhead consists of 18 bytes. The labels, functions, and arrangement of these bytes are shown in Figure below.

Figure *STS-1 frame: line overhead*



- **Line parity byte (B2):-** Byte B2 is for bit interleaved parity. It is for error checking of the frame over a line (between two multiplexers). In an STS-n frame, B2 is calculated for all bytes in the previous STS-1 frame and inserted at the B2 byte for that frame. In other words, in a STS-3 frame, there are three B2 bytes, each calculated for one STS-1 frame.
- **Data communication channel bytes (D4 to D12)**. The line overhead D bytes (D4 to D12) in consecutive frames form a 576-kbps channel that provides the same service as the D1–D3 bytes (OA&M), but at the line rather than the section level (between multiplexers).
- **Order wire byte (E2).** The E2 bytes in consecutive frames form a 64-kbps channel that provides the same functions as the E1 order wire byte, but at the line level.

➢ **Pointer bytes (H1, H2, and H3)**. Bytes H1, H2, and H3 are pointers. The first two bytes are used to show the offset of the SPE in the frame; the third is used for justification.

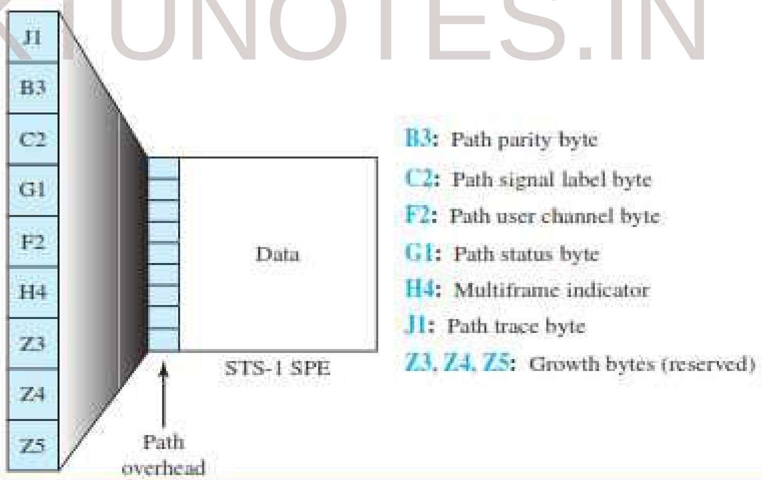
➢ **Automatic protection switching bytes (K1 and K2)**. The K1 and K2 bytes in consecutive frames form a 128-kbps channel used for automatic detection of problems in line-terminating equipment.

➢ **Growth bytes (Z1 and Z2)**. The Z1 and Z2 bytes are reserved for future use.

Synchronous Payload Envelope

The synchronous payload envelope (SPE) contains the user data and the overhead related to the user data (path overhead). One SPE does not necessarily fit it into one STS-1 frame; it may be split between two frames. This means that the path overhead, the leftmost column of an SPE, does not necessarily align with the section or line overhead. The path overhead must be added first to the user data to create an SPE, and then an SPE can be inserted into one or two frames. Path overhead consists of 9 bytes.

Figure *STS-1 frame: path overhead*



- **Path parity byte (B3)**. Byte B3 is for bit interleaved parity, like bytes B1 and B2, but calculated over SPE bits. It is actually calculated over the previous SPE in the stream.
- **Path signal label byte (C2)**. Byte C2 is the path identification byte. It is used to identify different protocols used at higher levels (such as IP or ATM) whose data are being carried in the SPE.
- **Path user channel byte (F2)**. The F2 bytes in consecutive frames, like the F1 bytes, form a 64-kbps channel that is reserved for user needs, but at the path level.

- Path status byte (G1). Byte G1 is sent by the receiver to communicate its status to the sender. It is sent on the reverse channel when the communication is duplex.
- **Multiframe indicator (H4)**. Byte H4 is the multiframe indicator. It indicates payloads that cannot fit into a single frame.
- **Path trace byte (J1).** The J1 bytes in consecutive frames form a 64-kbps channel used for tracking the path. The J1 byte sends a continuous 64-byte string to verify the connection. The choice of the string is left to the application program. The receiver compares each pattern with the previous one to ensure nothing is wrong with the communication at the path layer.
- **Growth bytes (Z3, Z4, and Z5)**. Bytes Z3, Z4, and Z5 are reserved for future use.

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 channels, one for transmission in each direction. Each channel is shared by a number of subscribers, and statistical TDM is used for allocating capacity on each channel for transmission.

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 below 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 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.

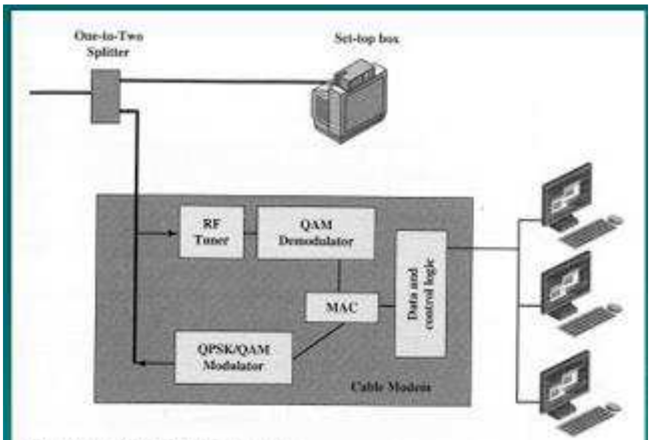


Figure 8.17 Cable Modem Configuration

CDMA

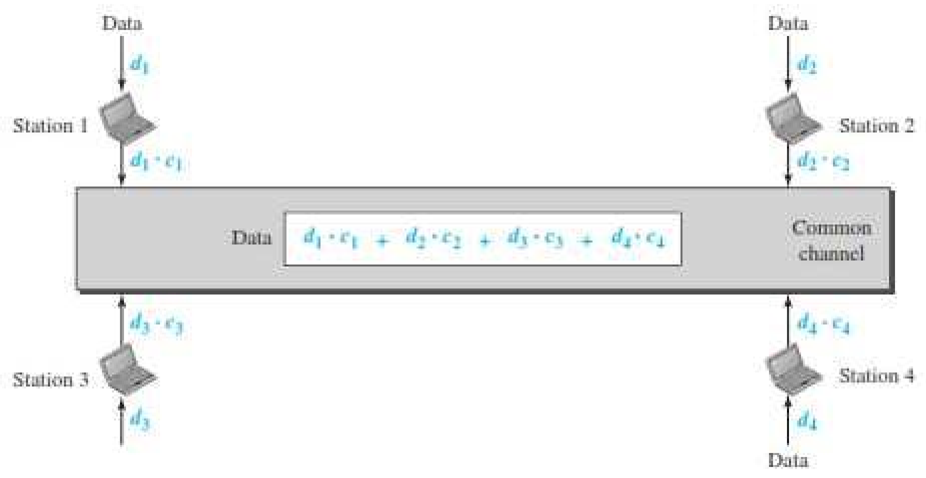
CDMA (Code Division Multiple Access) also called spread-spectrum and code division multiplexing, one of the competing transmission technologies for digital mobile phones. The transmitter mixes the packets constituting a message into the digital signal stream in an order determined by a pseudo-random number sequence that is also known to the intended receiver. Hence each different random sequence corresponds to a separate communication channel. CDMA is most used in the USA.CDMA differs from FDMA in that only one channel occupies the entire bandwidth of the link. It differs from TDMA in that all stations can send data simultaneously; there is no timesharing.

Let us assume we have four stations, 1, 2, 3, and 4, connected to the same channel. The data from station 1 are d1, from station 2 are d2, and so on. The code assigned to the first station is c1, to the second is c2 , and so on. The assigned codes have two properties.

1. If we multiply each code by another, we get 0.
2. If we multiply each code by itself, we get 4 (the number of stations).

Station 1 multiplies its data by its code to get d1· c1. Station 2 multiplies its data by its code to get d2· c2 and so on. The data that go on the channel are the sum of all these terms, as shown in the box. Any station that wants to receive data from one of the other three multiplies the data on the channel by the code of the sender. For example, suppose stations 1 and 2 are talking to each other. Station 2 wants to hear what station 1 is saying. It multiplies the data on the channel by c1, the code of station 1. Because (c1· c1) is 4, but (c2· c1), (c3· c1), and (c4· c) are all 0s, station 2 divides the result by 4 to get the data from station 1.

Figure *Simple idea of communication with code*



$$\text{data} = (d_1 \cdot c_1 + d_2 \cdot c_2 + d_3 \cdot c_3 + d_4 \cdot c_4) \cdot c_1$$
$$= d_1 \cdot c_1 \cdot c_1 + d_2 \cdot c_2 \cdot c_1 + d_3 \cdot c_3 \cdot c_1 + d_4 \cdot c_4 \cdot c_1 = 4 \times d_1$$

Chips

CDMA is based on coding theory. Each station is assigned a code, which is a sequence of numbers called chips.

Figure *Chip sequences*

C_1	C_2	C_3	C_4
[+1 +1 +1 +1]	[+1 −1 +1 −1]	[+1 +1 −1 −1]	[+1 −1 −1 +1]

The chips are called orthogonal sequences and have the following properties:

1. Each sequence is made of N elements, where N is the number of stations.
2. If we multiply a sequence by a number, every element in the sequence is multiplied by that element. This is called multiplication of a sequence by a scalar. For example,

$$2 \bullet [+1 +1 -1 -1] = [+2 +2 -2 -2]$$

3. If we multiply two equal sequences, element by element, and add the results, we get N, where N is the number of elements in each sequence. This is called the inner product of two equal sequences. For example,

$$[+1 +1 -1 -1] \bullet [+1 +1 -1 -1] = 1 + 1 + 1 + 1 = 4$$

4. If we multiply two different sequences, element by element, and add the results, we get 0. This is called the inner product of two different sequences. For example,

$$[+1\ +1\ -1\ -1] \cdot [+1\ +1\ +1\ +1] = 1 + 1 - 1 - 1 = 0$$

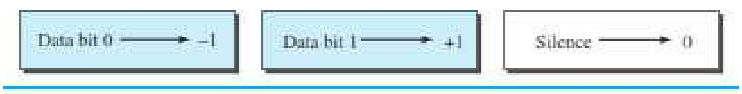
5. Adding two sequences means adding the corresponding elements. The result is another sequence. For example,

$$[+1\ +1\ -1\ -1] + [+1\ +1\ +1\ +1] = [+2\ +2\ \ 0\ \ 0]$$

Data Representation

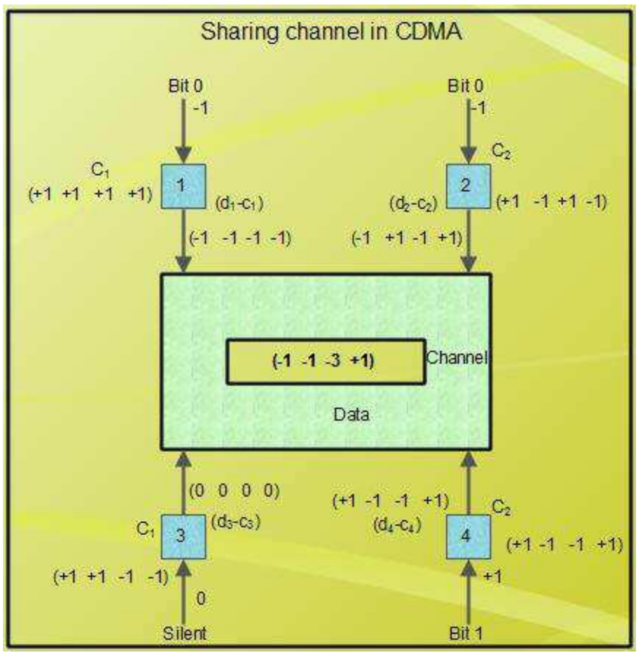
We follow these rules for encoding: If a station needs to send a 0 bit, it encodes it as −1; if it needs to send a 1 bit, it encodes it as +1. When a station is idle, it sends no signal, which is interpreted as a 0.

Figure *Data representation in CDMA*



Encoding and Decoding

Assume that stations 1 and 2 are sending a 0 bit and channel 4 is sending a 1 bit. Station 3 is silent. The data at the sender site are translated to −1, −1, 0, and +1. Each station multiplies the corresponding number by its chip (its orthogonal sequence), which is unique for each station. The result is a new sequence which is sent to the channel. The sequence on the channel is the sum of all four sequences.



Now imagine that station 3, which we said is silent, is listening to station 2. Station 3 multiplies the total data on the channel by the code for station 2, which is [+1 −1 +1 −1], to get

$$\mathbf{[-1\ -1\ -3\ +1] \cdot [+1\ -1\ +1\ -1] = -4/4 = -1 \rightarrow bit\ 1}$$

WORKSHEET

1. Assume that a voice channel occupies a bandwidth of 4 kHz. We need to multiplex 10 voice channels with guard bands of 500 Hz using FDM. Calculate the required bandwidth.

   Solution:-

   To multiplex 10 voice channels, we need nine guard bands. The required bandwidth is then B = (4 KHz) × 10 + (500 Hz) × 9 = 44.5 KHz

2. Four channels , two with a bit rate of 200kbps and two with a bit rate 150 kbps are to be multiplexed using multiple slots TDM with no synchronization bits.

   Answer the following questions: Assume 4 bits from the first 2 sources and 3 bits from the second 2 sources.

   i. What is the size of a frame in bits?

   ii. What is the frame rate?

   iii. What is the duration of a frame?

   iv. What is the date rate?

   Solution:-

   i. The frame carries 4 bits from each of the first two sources and 3 bits from each of the second two sources. Frame size = 4 × 2 + 3 × 2 = 14 bits.

   ii. Each frame carries 4 bit from each 200-kbps source or 3 bits from each 150 kbps. Frame rate = 200,000 / 4 = 150,000 /3 = 50,000 frames/s.

   iii. Frame duration = 1 /(frame rate) = 1 /50,000 = 20 μs.

   iv. Output data rate = (50,000 frames/s) × (14 bits/frame) = 700 kbps. We can also calculate the output data rate as the sum of input data rates because there are no synchronization bits. Output data rate = 2 × 200 + 2 × 150 = 700 kbps.

3. We need to use synchronous TDM and combine 20 digital sources, each of 100 Kbps. Each output slot carries 1 bit from each digital source, but one extra bit is added to each frame for synchronization.

   Answer the following questions:

   a. What is the size of an output frame in bits?
   b. What is the output frame rate?
   c. What is the duration of an output frame?
   d. What is the output data rate?
   e. What is the efficiency of the system (ratio of useful bits to the total bits)?

4. We have 14 sources, each creating 500 8-bit characters per second. Since only some of these sources are active at any moment, we use statistical TDM to combine these sources using character interleaving. Each frame carries 6 slots at a time, but we need to add 4-bit addresses to each slot.

   Answer the following questions:

   a. What is the size of an output frame in bits?

   b. What is the output frame rate?

   c. What is the duration of an output frame?

   d. What is the output data rate?

5. Ten sources, six with a bit rate of 200 kbps and four with a bit rate of 400 kbps, are to be combined using multilevel TDM with no synchronizing bits.

   Answer the following questions about the final stage of the multiplexing:

   a. What is the size of a frame in bits?

   b. What is the frame rate?

   c. What is the duration of a frame?

   d. What is the data rate?

6. Two channels, one with a bit rate of 190 kbps and another with a bit rate of 180 kbps, are to be multiplexed using pulse-stuffing TDM with no synchronization bits.

   Answer the following questions:

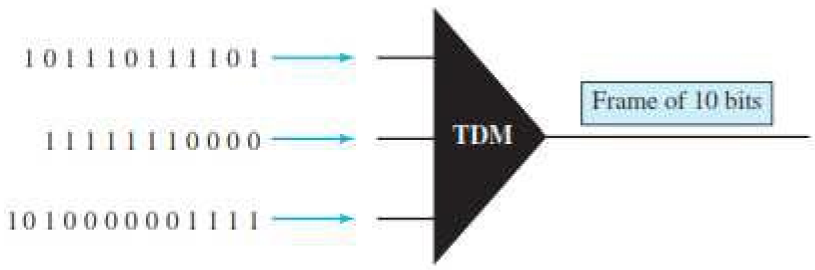
   a. What is the size of a frame in bits?

   b. What is the frame rate?

   c. What is the duration of a frame?

   d. What is the data rate?

7. Figure 6.64 shows a multiplexer in a synchronous TDM system. Each O/p slot is only 10 bits long (3 bits taken from each input plus 2 framing bit). What is the output stream? The bits arrive at the multiplexer as shown by the arrows.



QUESTIONS

1. Why is multiplexing so cost-effective?
2. How is interference avoided by using frequency division multiplexing?
3. Why is a statistical time division multiplexer more efficient than a synchronous time division multiplexer?
4. Which of the three multiplexing techniques is common for fiber-optic links? Explain the reason.
5. Distinguish between multilevel TDM, multiple-slot TDM, and pulse-stuffed TDM.
6. Distinguish between synchronous and statistical TDM.
7. What is CDMA?
8. What are the applications of FDM?
9. Write short notes on Digital Signal Service.
10. Why addressing is needed in statistical TDM?
11. What is the function of a SONET regenerator?
12. Discuss the functions of each SONET layer.
13. Why SONET is called a synchronous network?
14. What are the four layers of SONET?
15. What is the relationship between STS signals and OC signals?