## UNIT-IV

Circuit Switching : Simple switching Network, Circuit Switching Networks, Brief idea of following (detail working) not required.

Circuit Switching Concept : Space Division switching, Time Division Multiplexing, Routing in circuit switching Networks, Control Signalling, In channel & common channel signalling, Brief idea of SS7. Packet Switching: Packet switching principle, Routing, X.25 Data Encoding : Spread Spectrum. Asynchrous and Synchronous transmission, Full and Half duplex, Interfacing, Functional and Procedural aspects of V.24.

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Chapter

# 4

# Circuit Switching

## 4.1 Circuit Switched Networks

A circuit switched network consist of a set of switches connected by physical links. A connection between two stations is dedicated path made of one or more links how ever each connection uses only one dedicated channel on each link. Each link is hormally divided into h chonnels by using FDM or TDM as discussed.

A circuit switched network is made of a set of switches connected by physical link in which each link is divided in to h channels.

We have explicitly shown the multiplexing symbols to emphasize the division of the link into channels even though multiplexing can be implicity included in the switch fabric.

The end system, such as computers or telephones, are directly connected to a switch. We have shown only two end systems for simplicity. When end system & needs to communicaty, with end system M, system A needs to request a connection to M that must be accepted by all switches as well as by M itself. This called the **setup phase** a circuit (channel) is reserved on each link and the combination of circuits (channels) defines the dedicated path after the dedicated path made of connected circuits is established. data transfer can take place, after all data have been transferred, the circuits are torn down we need to emphasize several points here.

Circuit switching takes place at the physical layer.

Before starting the communication, the station must take a reservation for the resources to be used during the communication. These resources, such as channel (band width in FDM and time slote in TDM), switch buffers, switch processing time, and switch Input/Output ports, must remain dedicated during the entire duration of data transfer unit the teardown phase.

Data transferred between the two stations are not packetized (physical layer tansfer of the signal). The data are a continuous flow sent by the source station and received by the destination station, although there may be periods of silence.

There is no addressing involved during data transfer. The swithes route the data based on their occupied band (FDM) or time slot (TDM) of course, there is end to end addressing used during the setup phase as we will see shortly.

"In circuit switching, the resources need to be reserved during the setup phase. The resources remain dedicated fro the entire duration of data transfer untill the teardown phase."



**Three Phases :** The actual communication in a circuit switched network requires three phase : connection setup, data transfer, and connection teardown.

**Setup phase :** Before the two parties (or multiple parties in a conferetice call) can communicate, a dedicated circuit (combination of channels in link) need to established. The end system are normally connected through dedicated lines to the switches, so connection setup means creating dedicated channels between the switches for example when system a needs to connect to system M, it sends a setup request that includes the address of system M, to switch I. Switch I finds a channel between itself and switch IV that can be dedicaied for this purpose, switch I then sends the request to switch IV which finds a diedicated channel between itself and switch III, witch III informs system M of system A intention at this time.

In the next step to making a connection, an acknoledgement from system M needs to be sent in the opposite direction to system A only after system a receives this acknowledgment in the connection established.

Note that end-to-end addressing is required for creating a connection between the two end systems. These can be for example, the addresses of the computer assigned by the administrator in a TDM network, or telephone numbers in an FDM network.

**Data transfer phase :** After the establishment of the dedicated circuit (channel) are two parties can transfer data.

**Teardown phase :** When one of the parties needs to disconnect a signal sent to each switch to release the resources.

**Efficiency :** It can be argued that circuit switched networks are not as efficient as the other two types of networks because resources are allocated during the entire duration of the connection. These resources are unavailable to other connections. In a telephone network, people normally terminate the communication when they have finished their conversion. However, in computer networks, a computer can be connected to another computer even if there is no activity for a long time. In this case, allowing resources to be dedicated means that other connections are deprived.

**Delay :** Although a circuit switched network normally has low efficiency, the delay in this types of network is minimal during data transfer the data are not delayed at each switch; the resources are allocated for the duration of the connection. The idea of delay in a circuit-switched network when only two switches are involved. As figure 8.6 shows, there is no waiting time at each switch the total delay is due to the time needed to create the connection, transfer data, and disconnect the circuit. The delay caused by the setup is the sum of four parts; the propagation time of the souh computer request (slop of the first gray box), the request signal tranfer time (height of the first gray box), the propagation time o the acknowledgement from the destination computer (slop of the second gray box), and the signal transfer time of the acknowledgement (height of the second gray box). The delay due to data transfer is the sum of two parts the propagation time (slope of the colored box), which can be very long. The third box shows the time needed to teardown circuit. We have shown the case in which receiver rquest disconnection which creates the maximum delay.

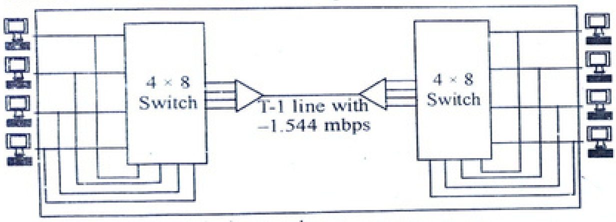
**Circuit Switched technology in telephone network**

The telephone companies have previously chosen the circuit switched approach to switching in the physical layer; today the tendency is moving toward other switching techniqe.

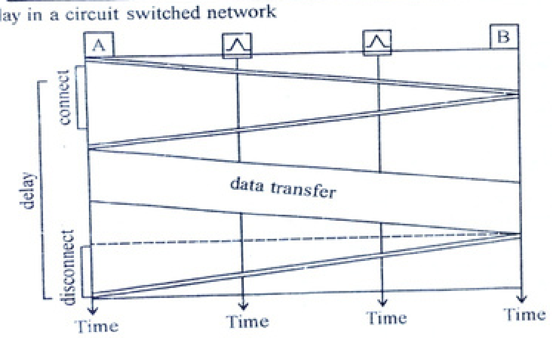
For example, the telephone number is used as the global address, and a signaling system (called 557) is used for the setup and teardown phases.

Switching at the physical layer in the traditional telephone network uses the circuit-switching approach.

Circuit switched network used in example




Delay in a circuit switched network






**Circuit switching concepts :** We use switches in circuit-switched and packet switches networks. In this section, we discuss the structures of the switches used in each type of network.

**Structure of Circuit Switches :** Circuit switching today can use either of two tecnologies : the space-division switch or time-division switch.

**Space Division Swritching :** In space division switching the paths in the circuit are separated from one another spatially. This technology was originally designed for use in analog networks but is used currently in both analog and digited network. It has envolved through of many design.

**Crossbar switch :** A crossbar switch connects n inputs to m outputs in a grid using electronic microswitches (transistors) at each corsspoint (see Fig. 1.1) the major limitation of this design is the number of crosspoints required. To connect n input to m outputs using a corssbar switch requires nxm crosspoints. For example, to connect 1000 inputs to 1000 outputs requires a switch with 1,00,000 crosspoints. A crossbar with this number of cross points is impractical. Such a switch is also inefficient because statistics show that, in practice, fewer than 25 percent of the crosspoints are in use at any given time. The rest are idle.

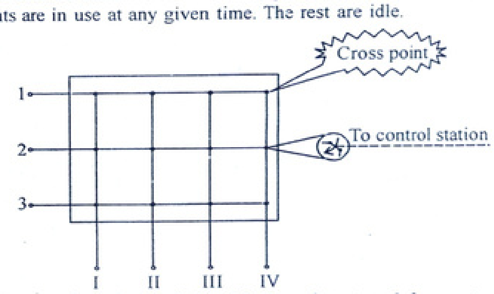



**Fig. 1 : Crossbar switch with three input and four outputs**

**Multistage switch :** The solution to the limitations of the crossbar switch is the multistage switch, which combines crossbar switches in several (normally three) stage as shown in figure 8.18. In a single crossbar switch, only one row or column (one path) is active for any connection. So we need N × N crosspoints. If we can allow multiple paths inside the switch, we can decrease the number of corsspoints. Each crosspoint in the middle stage can be accessed by multiple crosspoints in the first and third stage.

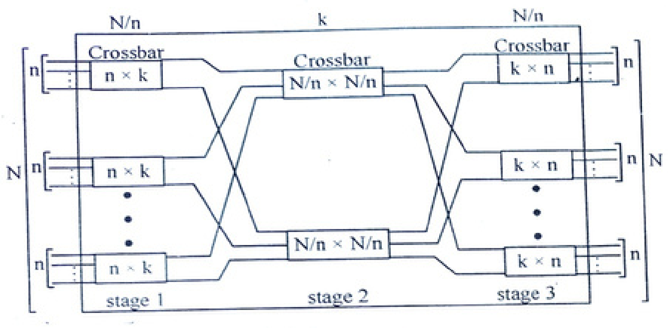



**Fig. 2 : Multistage switch**

To design a three stage switch we follow these steps :

1. We divide the N input lines into groups, each of n lines. For each group we use one crossbar of size n × k where k is the number of crossbar in the middle stage.

In other words, the first stage has N/n crossbars of n × k crosspoints.

2. We use k crossbars, each of size (N/n) × (N/n) in the middle stage.

3. We use N/n crossbars, each size k × n at the third stage.

We can calculate the total number of crosspoints as follows :

$$\left[\frac{N}{n}(n \times k) + k\left(\frac{N}{n} \times \frac{N}{n}\right) + \frac{N}{n}(k \times n) = 2KN + k\left(\frac{N}{n}\right)^2\right]$$

In a three stage switch, the total number of crosspoints is

$$2KN + K\left(\frac{N}{n}\right)^2$$

Which is much smaller than the number of crosspoints in a single stage switch ($N^2$)

■ **Example 1.1 Design a three stage, 200 × 200 switch (N = 200) with K = 4 and n = 20.**

**Solution :** In the first stage we have N/n or 10 crossbars, each of size 20 × 4. In the second stage, we have 4 corssbars, each of size 10 × 10. In the third stage, we have 10 crossbars each of size 4 × 20.



The total number of corsspoints is $2KN + K\,(N/n)^2$, or 2000 crosspoints This is 5 percent of the number of crosspoints in a single stage switch (200 × 200 = 40,000).

The multistage switch in ex.1.1. has one drawback blocking during periods of heavy traffic. The whole idea of multistage switching is to share the crosspoints in middle-stage crossbars. Sharing can cause a lack of availability if the resources are limited and all users want a connection at the same time. Blocking refers to times when one input cannot be connected to an output because there is no path available between them all the possible intermediate switches are occupied.

In a single stage switch blocking does not occur because every combination of input and output has it's own crosspoint; there is always a path. In the multistage switch described in ex. 1.1, however, only 4 of the first 20 inputs can USP the switch at a time, only 4 of the second 20 inputs can use the switch at a time, and so on. The small number of crossbars at the middle stage creates blocking.

In large system, such as those having 10,000 inputs and outputs, the number of stage can be increased to cut down on the number of crosspoints required. As the number of stages increases, however, possible blocking increases as well. Many people have experienced blocking on public telephone system in the wake of a natural disaster when the calls being made to check on or reassure relatives for outnumber the regular load of the system.

Close investigated the condition of nonblocking in multistage switches and came with the following formula. In a non-blocking switch, the number of middle-stage switches must be at least 2n-1. In other words, we need to have $K \geq 2n - 1$.

Note that the number of crosspoints is still smaller than that in a single-stage switch. Now we need to minimize the number of crosspoints with a fixed N by using the close criteria. We can take the derivate of the equation with respect to n (the only variable) and find the value of n that makes the result zero. This n must be equal to greater than $(N/2)^{1/2}$. In this case, the total number of crosspoints is greater than or equal to $4N[(2N)^{1/2} - 1]$.

In other words, the minimum number of crosspoints according to the close criteria is proportional to $N^{3/2}$.

According to dos criterion :

$n = (N/2)^{1/2}$

$K > 2n - 1$

Total number of crosspoints $\geq 4N\,[(2N)^{1/2} - 1]$

■ **Example 1.2 : Redesign the previous three stage 200 × 200 switch, using the clos criteria with a minimum number of crosspoints.**

**Solution :** We let n = $(200/2)^{1/2}$, or n = 10 we calculate K = 2n − 1 = 19. In the first stage, we have 200/2 or 20, crossbars, each with 10 × 19. In the second stage, we have 19 crossbars each with 10 × 10 crosspoints. In the third stage, we have 20 crossbars each with 19 × 10 crosspoints. The total number of crosspoints is 20(10 × 19) + 19 (10 × 10) + 20 (19 × 10) = 9500 if we use a single stage switch, we need 200 × 200 = 40,000 crosspoints. The number of crosspoints in this three stage switch is 24 percent that of single-stage switch.

Points are needed than in Example 1.1 (5 percent). The extra crosspoints are needed to present blocking.

A multistage switch that uses the dos criteria and a minimum number of crosspoints still reuires a have number of crosspoints for example, to have a 100,000 input/output switch, we need something close to 200 million crosspoints (instead of 10 billion). That means that if a telephone company needs to provide a switch to connect 100,000 telephones if accept blocking. Today, telephone companies use time-division switching or a combination of space and time division switches, as we will see shortly.

**Time Division Switching**

Time-division switching use time-division multiplexing (TDM) inside a switch. The most popular technology is called the time slot inter change (TSI).

**Time slot Interchange**

A system connects four inputs line to foor output lines. Imagine that each input line wants to send data to an output line according to the pattern.

The figure combines a TDM multiplexer a TDM demultiplexer, and a TSI consisting of random access memory (RAM) with several memory locations. The size of each location is the same as the size of a single time slot. The number of locations is the same as the number of inputs (in most cases, the numbers of inputs and outputs are equal).

The RAM fills up with incoming data from time slots in the order received slots are then sent out in an order based on the decisions of a control unit.

**Time and Space division Switch Combinations**

When we compare space-division and time-division switching, some interesting facts emerge. The advantage of space-division switching is that it is instantaneous. Its disadvantage is the number of crosspoints required to make space-division switching acceptable in term of blocking.

The advantage of time-division switching is that if needs no corsspoints. Its disadvantage, in the case of TSI, is that processing each connecting creates delays. Each time slot must be stored by the RAM, then retrieved and passed on.

In a third option, we combine space division and time-division technologies to take advantage of the best of both, combining the two result in switches that are optimized both physically (the number of crosspoints) and temporally (the amount of delay). Multistage switches of this sort can be designed as time-space time (TST) switch.



Figure shows a simple TST switch that consist of two time stages and one space stage and has 12 inputs and 12 outputs. Instead of one time-division switch, it divides the inputs into three groups (of four inputs each) and directs them to three time-slot interchange. The result is that the average delay is one-third of what would result from using one time slot interchange to handle all 12 inputs.

The last stage is a mirror image of the first stage. The middle stage is a space division switch (cossbar) that connects the TSI groups to allow connectivity between all possible input and output pairs (e.g. to connect input 3 of the first group to output 7 of the second group).

**Synchronous Time-division Multiplexing**

**Time division multiplexing (TDM)** is a digital process that allows several connections to share the high bandwidth of a ling. 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 3 gives a conceptual view of TDM. Note that the same link is used as in FDM; here, how ever, the link is shown sectioned by time rather than by frequency. In the figure, protions of signals 1, 2, 3 and 4 occupy the link sequentially.


**Fig. 3 : TDM**

Note that in figure 6.12 we are concerned with only multiplexing, not switching. This means that all the data in a message from source 1 always go to one specific destination be it 1, 2, 3 or 4. The delivery is fixed and unvarying, unlike switching.

We also need to remember that TDM is, in principle, a digital multiplexing technique. Digital data from different sources are combined into on timeshared link. However, this does not mean that the sources cannot produce analog data; analog data can be sampled changed to digital data, and than multiplexed by using TDM.

**TDM is a digital multiplexing technique for combining several low-rate channels into one high rate one.**

We can divide TDM into two different schems; synchronous and statistical.

We first discuss **synchronous TDM** and then show how **statistical TDM** differs. In synchronous TDM, each input connection has an allotment in the output even if it is not sending data.

**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 s, the output time slot is Tin 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 6.13 shows an example of synchronous TDM where n is 3.

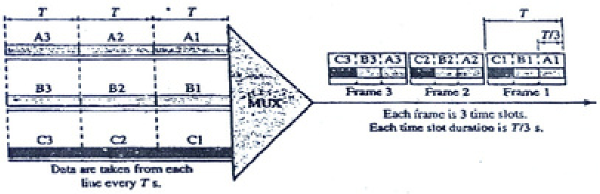



**Fig. 4 : Synchronous time division multiplexing**

In synchronous TDM, a round of data units from each input connection is collected into a frame (we will see the reason for this shortly). If we have n connections, a frame is divided into n time slots and one slot is allocated fro 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 (unless a frame carries some other information, as we will be shortly).

The data rate of the output link must be n times the data rate of a connection to guarante the flow of data. In Figure 6.13, the data rate of the link is 3 times the data rate of a connection; likewise, the duration of a unit on a connection is 3 times that of the time slot (duration of a unit on the link). In the figure we represent the data prior to multiplexing as 3 times the size of the data after multiplexing. This is just to convey the idea that each unit is 3 times longer in duration before multiplexing than after.

**In synchronous TDM, the data rate of the link is n times faster, and the unit duration is n times shorter.**

Time slots are grouped into frames. A frame consists of one complete cycle



of time slots, with one slot dedicated to each sending device. In a system with n input lines, each frame has n slots, with each slot allocated to carrying data from a specific input line.

**Example 6.5 : In figure 5, the data rate for each input connection is 3 kbps. If 1 bit at a time is multiplexed is unit is 1 bit, what is the duration of (a) each input slot, (b) each output slot, and (c) each frame.**

**Solution :** We can answer the questions as follows :

a. The data rate of each input connection is 1 kbps. This means that the bit duration is 1/1000 or 1 ms. The duration of the input time slot is 1 ms (same as bit duration).

b. The duration of each output time slot is one-third of the input time slot. This means that its duration of the output time slot is 1/3 ms.

c. Each frame carries three output time slots. So the duration of a frame is 3 × 1/3 ms, or 1. The duration of a frame is the same as the duration of an input unit.

**Example 6.6 Figure 6.14 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 (a) the input bit duration (b) the output bit duration, (c) the output bit rate, and (d) the output frame rate.**

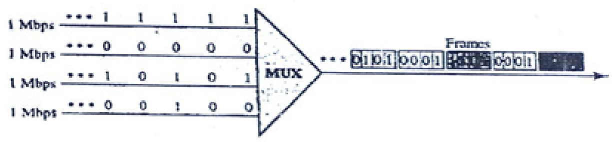



**Fig. 5 : Example 6.6**

**Solution :** We can answer the questions as follows :

a. The input bit duration isthe inverse of the bit rate : 1/1 Mbps = 1 μs.

b. The output bit duration is one-fourth of the input bit duration or 1/4 μs.

c. 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.

d. The frame rate is always the same as any input rate. So the frame rate is, 1,000,000 frames per second. Because we are sending 4 bits in each frame, we can verify the result of the previous question by multiplying the frame rate bythe number of bits per frame.

**Example 6.7 : Four 1-kbps connections are multiplexed together. A unit is 1 bit. Find (a) the duration of 1 bit before multiplexing, (b) the transmission rate of the link, (c) the duration of a time slot, and (d) the duration of a frame.**

**Solution :** We can answer the quaestions as follows :

a. The duration of 1 bit before multiplexing is 1/1 kbps, or 0.001 s (1ms).

b. The rate of the link is 4 times the rate of a connection, or 4 kbps.

c. The duration of each tie slot is one fourth ofthe 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.

d. The duration of a frame is always the same as the duration of a unit before multiplexing or 1 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. On the multiplexing side, as the switch opens in front of a connection, that connection has the opportunity to send a unit onto path. This process is called **interleaving.** On the demultiplexing side, as the switch opens in front of a connection, that connection ahs the opportunity to receive a from te path.

Figure 6.15 shows the interleaving process for the connection shown in figure 6.15. In this figure, we assume that no switching in involved and that the data from the connection at the multiplexer site go to te first connection at the demultiplexer discuss switching in Chapter 8.

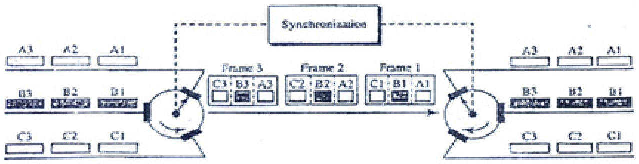



**Fig. 6 : Interleaving**

**Example 6.8. Four channels are multiplexed using TDM. If each channel sends 100 bytes and we multiplex 1 byte per channel, show the frame traveling on the link, the size of the frame, the duration frame, the**



**frame rate, and the bit rate for the link.**

**Solution :** The multiplexer is shown in Figure 7. Each frame carries 1 byte from each channel; the each frame, therefore, is 4 bytes, or 32 bits. Because each channel is sending 100 bytes/s frame carries 1 byte from each channel, the frame rate must be 100 frames per second. The duration of a frame is therefore 1/100 s. The link is carrying 100 frames per second, and since frame contains 32 bits, the bit rate is 100 × 32, or 32000 bps. this is actually 4 times the bit each channel, which is 100 × 8 = 800 bps.

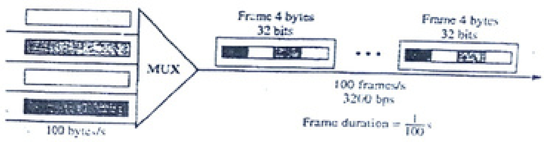



**Fig. 7 Example 6.8**

**Example 6.9 : Amultiplexer combines four 100 kbps channeis using a time slot of 2 bits. Show the output with four arbitrary inputs. What is the frame rate ? What is the frame duration ? What is the bit rate ? What i the bit duration ?**

**Solution :** Figure 8 shows the output fro four arbitrary inputs. The link carries 50,000 frames per second since each frame ontains 2 bits per channel. The frame duration is therefore 1/50,000 s or 20 μs. The frame rate is 50,000 frames per second, and each frame carriers 8 bits; the bit rate is 50,000 × 8 = 400,000 bits or 400 kbps. The bit duration is 1/400,000 s, or 2.5 μs. Note that the frame duration is 8 times the bit duration because each frame is carrying 8 bits.

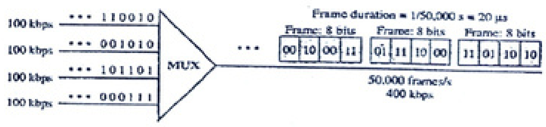



**Fig. 8 : Example 6.9**

**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 6.18 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.

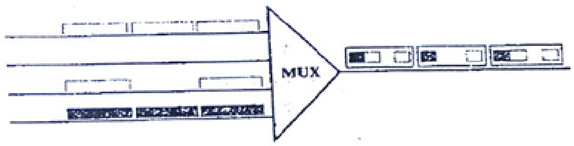



Fig. 9. Empty slots

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. We learn in the next section that statistical TDM can improve the efficiency by removing the empty slots from the frame.

**Data Rate Management :** One problem with TDM is how to handle a disparity in the input data rates. In all our discussion so far, we assumed that the data rates of all input lines were the same. However, if data rates are not the same, thre strategies, or a combination of them, can be used. We call these strategies multilevel multiplexing, multiple-slot allocation and pulse stuffing.

**Multiplevel Multiplexing :** Multilevel multiplexing is a technique used when the data rate of an input line is a multiple of othrs. For example, in Figure 10, 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 dat rate equal to the last three. A second level of multilexing can create an output of 160 kbps.

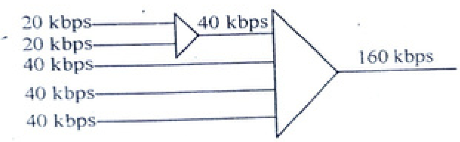



Fig. 10 Multilevel multiplexing

**Multipl-slot Allocation :** Sometimes it is more efficient to allot more than one slot in a frame to a single input line. For eample, we might have an input line that has a data rate that is a multiple of another input. In figure 11, the input line with a 50 kbps data rate can be given two slots inthe output. We insert a serial to parallel converter in this line to make two inputs out of one.



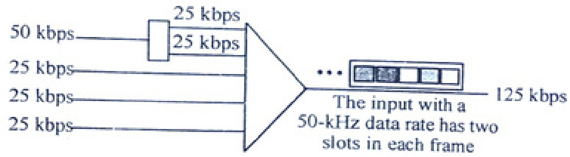



Figure 11 Multiple slot multiplexing

**Pulse Stuffing :** Sometimes that bit rates of sources are not multiple integers of each other. Therefore, neither of the above two techniques can be applied. One solution is make the highest input datarate the dominant data rate and ten 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. The idea is shown in Figure 12. The input with a data rate of 46 is pulse-stuffed to increase the rate to 50 kbps. Now multiplexing call take place.

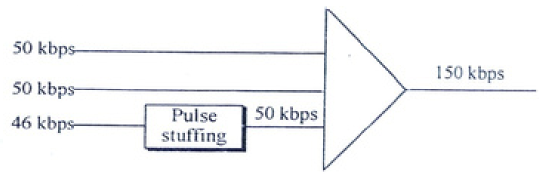



Fig. 13 Pulse stuffing

**Frame Synhronizaing :** 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 demultiplexr 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 demultiplxer to synchronize with the incoming stream so that is can separate the time slots accurately. In most cases, this synchronization information consists of 1 bit per frame, alternating between 0 and 1, as shown in Figure 6.22.

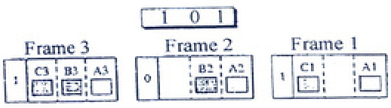



Figure 14 Framing bits

**Example 6.10. We hae four sources, each creating 250 characters per second. If the interleaved unit is a charater and 1 synchronizing bit is added to each frame, find (a) the data of each source, (b) the duration of each character in each source, (c) the frame rate, (d) the duration of each frame, (e) the number of bits in each frame, and (f) the data rate of the link.**

**Solution :** We can answer questions as follows :

a. The data rate of each source is 250 × 8 = 2000 bps = 2 kpbs.

b. Each source sends 250 chracters per second; therefore, the duration of a charater is 1/250 s, or 4 ms.

c. 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.

d. The duration of each frame is 1/250 s, or 4ms. Note that the duration of each frame is the same as the duration of each character coming from each source.

e. Each frame carries 4 characters and 1 extra synchronizing bit. This means that each frame is 4 × 8 + 1 = 33 bits.

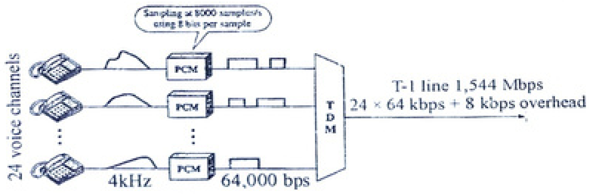



Fig. 15 : T-1 Line for multiplexing telephone lines



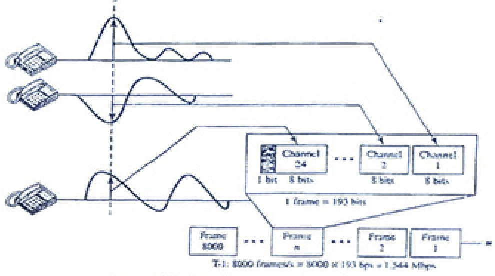



Fig. 16. T-1 Frame structure

Each slot contains one signal segment from each channel; 24 segments are interleated in on frame. If a T-1 line carries 8000 frames, the data rate is 1.544 mbps (193 × 8000 = 1.544 Mbps) the capaticy of the line.

**E lines :** Europeans use a version of T lines called E lines. The two systems are conceptually identical, but their capacities differ. Table 6.2 shows the E lines and their capacities.

**Table 6.2 E line rates**

| Line | Rate (Mbps) | Voice Channels |
|---|---|---|
| E-1 | 2.048 | 30 |
| E-2 | 8.448 | 120 |
| E-3 | 34.368 | 480 |
| E-4 | 139.264 | 1920 |

**More Synchronous TDM Applications :** Some second-generation cellular telephone companies use synchronous TDM. For example, the digital version of cellular telephony divides the available bandwidth into 30-kHz bands. For each band, TDM is applied sothat six users can share the band. This means that each 30-kHz band is now made of six time slots, and the digitized voice signals ofthe users are inserted in the slots. Using TDM, the number of telephone users in each area is now 6 times greater. We discuss second-generation cellular telephony in chapter 16.

**Statistical Time-division Multiplexing :** As we saw in the previous section, in synchronous TDM, each input has a reserved slot in the output frame. This an be inefficient if some input lines have no data to send. In statistical time-division multiplexing, slots are dynamically allocated to improve band width 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 allocated a slot for an input line if the line has data to send; otherwise, it skips the line and checks the next line.

Figure 6.26 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.

**Addressing :** Figure 6.26 also shows a major difference between slots in synchornous TDM and statistical TDM. 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 preassigned 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 relation ship between the inputs and outputs because there are no preassigned 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 from 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.

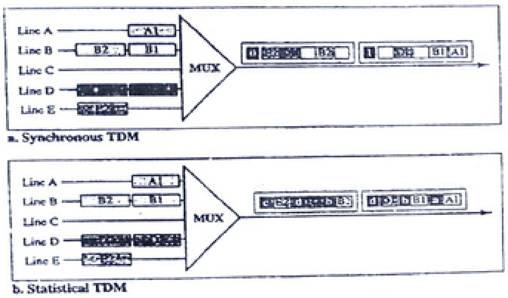



**Fig. 17 : TDM slot comparision**

**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 reasnable to make transmission efficient. For exaple, 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 man bytes while th address is just a few bytes.

**No synchronization bit :** There is another difference between synchronous and stastical 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.

**Bondwidth :** 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 lies 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 time some slots need to wait.

## 4.2 Routing in Circuit-Switching Networks

In a large circuit-switching network, such as the AT & T long distance telephone network, many of the circuit connections will require a path through more than one switch. When a call is placed, the network must devise a route through the network from calling subscriber to called subscriber that passes through some number of switches and trunks. There are two main requirements for the network architecture that bears on the routing strategy: efficient and resilience. First, it is desirable to minimize the amount of equipment (switches and trunks) in the network, subject to handle the expected loan. The load requirement is usually expected over the terms of a busy-hour of use during the course of a day.

From a functional point of view, it is necessary to handle that amount of load. From a cost point of view, we would like to handle that load with minimum equipment. Another requirement is resilience. Although the network may be sized for the busy hour load, it is possible for the traffic to surge temporarily above that level(for example, during a major storm). It will also be the case that, from time to time, switches and trunks will fail and be temporarily unavailable(unfortunately, maybe during the same storm). We will like the network to provide a reasonable level of service under such conditions.

The key design issue that determines the nature of the tradeoff between efficiency and resilience in the routing strategy. Traditionally the routing function in public telecommunications networks has been quite simple. In essence, the switches of a network were organized into a tree structure, or hierarchy. A path was constructed by starting all the calling subscriber, tracing up the tree to the first common node, and then tracing down the tree to the called subscriber. To add some resilience to the network, additional high-usage trunks were added that cut across the tree structure to connect exchanges with high volumes of traffic between them. In general, this is a static approach. The addition of high-usage trunks provides redundancy and extra capacity, but limitations remain both in terms of efficiency and resilience.

Because this routing scheme is not able to adapt to changing conditions, the network must be designed to meet some typical heavy demand. A dynamic routing approach is one in which routing decisions are influenced by current traffic conditions. Typically, the circuit-switching nodes have a peer relationship with each other rather than a hierarchical one. All nodes are capable of performing the same function. In such an architecture, routing is both more complex and more flexible. It is more complex because the architecture does not provide a "natural" path or set of paths based on hierarchical structure. But it is also more flexible, because more alternative routes are available.

As an example, we look at a form of routing in circuit-switching networks known as alternate routing. The essence of alternate routing schemes is that the possible routes to be used between two end offices are predefined. It is the responsibility of the originating switch to select the appropriate route of or each call. Each switch is given a set of pre planned routes for each destination, in order of preference. If a direct trunk connection exists between two switches, this is usually the preferred choice. If this trunk is unavailable, then the second choice is to be tried, and so on. The routing sequence (sequence in which the routes in set are tried) reflect an analysis bases as historical traffic patterns and are designed to optimize the use of network resources.

If there is only one routing sequence defined for each source-destination pair, the scheme is known as a fixed alternate routing a scheme. More commonly a dynamic alternate routing as scheme is used. In the later case , a different set of preplanned routes is used for different time periods, to take advantage of the differing traffic patterns and are designed to optimize the use of network traffic patterns(which determines the sequence of routes to be considered)

## Routing in Packet-Switching Networks

The primary function of a packet-switching network is to accept packets from a source station and deliver them to a destination station. To accomplish this, a path or route through the network must be determined: generally, more than one route is possible. Thus, a routing function must be performed.

**The requirements for this function include.**

1. Correctness
2. Simplicity
3. Fairness
4. Optimality
5. Robustness
6. Efficiency
7. Stability

The first two items on the list, correctness and simplicity, are self-explanatory, robustness has to do with the ability of the network to deliver packets via same route in the face of localized failures and overloads. Ideally, the network can react to such contingencies with the loss of packets or the breaking of virtual circuits. The designer who seeks to do business must cope with the competing requirements



for stability. A tradeoff also exists between fairness and optimality. Some performance criteria may give higher priority to the exchange of packets between nearby stations compared to an exchange between distant stations. This policy may maximize average throughput but will appear unfair to the station that primarily needs to communicate with distant stations.

Finally, any routing technique involves some processing overhead at each node and often a transmission over head needs to be less than the benefit accrued based on some reasonable metric, such as increased robustness or fairness. With these requirements in mind, we are in a position to assess the various design elements that contribute to a routing strategy. The table lists these elements.

## Performance Criteria

The selection of a route is generally based on some performance criterion. The simplest of this is to choose the minimum hop route(One that passes through the least number of nodes) through the network. This is an easily measured criterion which should minimize the consumption of the network resource. A generalization of the minimum hop criterion on is Least Cost Routing).In this, a cost is associated with each link, and for any pair of attached stations, the route through the network that accumulates the least cost is sought. For example, consider the following figure 27.

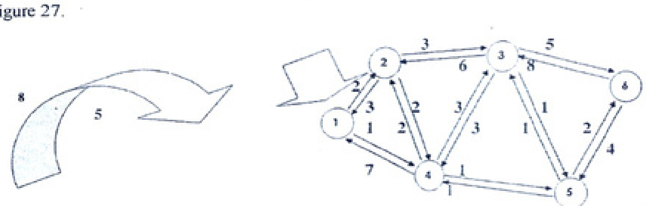


**Fig 18 : Example Packet switching network**

The figure 18 illustrates a network in which the two arrowed lines between a pair of nodes represent a link between these nodes, and the corresponding numbers represent the current link cost in each direction. The shortest path from node 1 to 6 is 1-3-6. (cost=5+5=10). But the least cost path is 1-4-5-6(cost=1+2+2+4=9). Costs are assigned to links to support one or more design objectives. The cost could be inversely related to the data rate or the current queuing delay on the link. In the first case, the least cost route should minimize delay.

In either the minimum-hop or least cost approach, the algorithm for determining the optimum route for any pair of station is relatively straight forward, and the processing time would be about the same for either computation. Because the least processing time would be about the same for either computation. The reason is that the least cost criterion is more flexible, this is more common than the minimum hop criterion.

**Table Optical fiber versus mechanic cable**

| Twisted pair cable | Co-axial cable | Optical fiber |
|---|---|---|
| 1. Transmission of signals takes place in the electrical form over the metallic conducting wires. | 1. Transmission of signals takes place in the electrical form over the inner conductor of the cable | 1. Signal transmission takes place in an optical forms over a glass fibre. |
| 2. In this medium the noise immunity is low | 2. Coaxial having higher noise immunity than twisted pair cable. | Optical fibre has highest noise immunity as the light rays are unaffected by the electrical noise. |
| 3. Twisted pair cable can be affected due to external magnetic field. | 3. Coaxial cable is less affected due to external magnetic field. | 3. Not affected by the external magnetic field. |
| 4. Cheapest medium | 4. Moderate expensive | Expensive |
| 5. Low Bandwidth | 5. Moderately high bandwidth. | Very high bandwidth. |
| 6. Attenuation is very high | 6. Attenuation is low. | Attenuation is very low. |
| 7. Installation is easy. | 7. Installation is fairly easy. | 8. Installation is difficult. |

## 4.3 Control Signalling System No.7(SS7)

Common Channel Signaling System No. 7 (i.e., SS7 or C7) is a global standard for telecommunications defined by the International Telecommunication Union (ITU) Telecommunication Standardization Sector (ITU-T). The standard defines the procedures and protocol by which network elements in the public switched telephone network (PSTN) exchange information over a digital signaling network to effect wireless (cellular) and wireline call setup, routing and control. The ITU definition of SS7 allows for national variants such as the American National Standards Institute (ANSI) and Bell Communications Research (Telcordia Technologies) standards used in North America and the European Telecommunications Standards Institute (ETSI) standard used in Europe.

The SS7 network and protocol are used for:



1. basic call setup, management, and tear down
2. wireless services such as personal communications services (PCS), wireless roaming, and mobile subscriber authentication
3. local number portability (LNP)
4. toll-free (800/888) and toll (900) wireline services
5. enhanced call features such as call forwarding, calling party name/number display, and three-way calling
6. efficient and secure worldwide telecommunications
7. Learn about our SS7/IP signaling products.

### Signaling Links

SS7 messages are exchanged between network elements over 56 or 64 kilobit per second (kbps) bidirectional channels called signaling links. Signaling occurs out-of-band on dedicated channels rather than in-band on voice channels. Compared to in-band signaling, out-of-band signaling provides:

1. faster call setup times (compared to in-band signaling using multi-frequency (MF) signaling tones)
2. more efficient use of voice circuits
3. support for Intelligent Network (IN) services which require signaling to network elements without voice trunks (e.g., database systems)
4. improved control over fraudulent network usage
5. Save network operating costs by reducing SS7 links.

### Signaling Points

Each signaling point in the SS7 network is uniquely identified by a numeric point code. Point codes are carried in signaling messages exchanged between signaling points to identify the source and destination of each message. Each signaling point uses a routing table to select the appropriate signaling path for each message.

There are three kinds of signaling points in the SS7 network.

1. SSP (Service Switching Point)
2. STP (Signal Transfer Point)
3. SCP (Service Control Point)

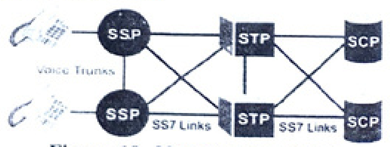



**Figure 19. SS7 Signaling Points**

SSPs are switches that originate, terminate, or tandem calls. An SSP sends signaling messages to other SSPs to setup, manage, and release voice circuits required to complete a call. An SSP may also send a query message to a centralized database (an SCP) to determine how to route a call (e.g., a toll-free 1-800/888 call in North America). An SCP sends a response to the originating SSP containing the routing number(s) associated with the dialed number. An alternate routing number may be used by the SSP if the primary number is busy or the call is unanswered within a specified time. Actual call features vary from network to network and from service to service.

Network traffic between signaling points may be routed via a packet switch called an STP. An STP routes each incoming message to an outgoing signaling link based on routing information contained in the SS7 message. Because it acts as a network hub, an STP provides improved utilization of the SS7 network by eliminating the need for direct links between signaling points. An STP may perform global title translation, a procedure by which the destination signaling point is determined from digits present in the signaling message (e.g., the dialed 800 number, calling card number, or mobile subscriber identification number). An STP can also act as a "firewall" to screen SS7 messages exchanged with other networks.

Because the SS7 network is critical to call processing, SCPs and STPs are usually deployed in mated pair configurations in separate physical locations to ensure network-wide service in the event of an isolated failure. Links between signaling points are also provisioned in pairs. Traffic is shared across all links in the linkset. If one of the links fails, the signaling traffic is rerouted over another link in the linkset. The SS7 protocol provides both error correction and retransmission capabilities to allow continued service in the event of signaling point or link failures.

### SS7 Signaling Link Types

Signaling gateways can be configured as an STP or SEP (Signaling End Point). Signaling links are logically organized by link type ("A" through "F") according to their use in the SS7 signaling network.

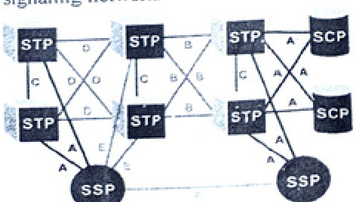



Figure 20 : SS7 Signaling Link Types



| | |
|---|---|
| A Link: | An "A" (access) link connects a signaling end point (e.g., an SCP or SSP) to an STP. Only messages originating from or destined to the signaling end point are transmitted on an "A" link. |
| B Link: | A "B" (bridge) link connects an STP to another STP. Typically, a quad of "B" links interconnect peer (or primary) STPs (e.g., the STPs from one network to the STPs of another network). The distinction between a "B" link and a "D" link is rather arbitrary. For this reason, such links may be referred to as "B/D" links. |
| C Link: | A "C" (cross) link connects STPs performing identical functions into a **mated pair**. A "C" link is used only when an STP has no other route available to a destination signaling point due to link failure(s). Note that SCPs may also be deployed in pairs to improve reliability; unlike STPs, however, mated SCPs are not interconnected by signaling links. |
| D Link: | A "D" (diagonal) link connects a secondary (e.g., local or regional) STP pair to a primary (e.g., inter-network gateway) STP pair in a quad-link configuration. Secondary STPs within the same network are connected via a quad of "D" links. The distinction between a "B" link and a "D" link is rather arbitrary. For this reason, such links may be referred to as "B/D" links. |
| E Link: | An "E" (extended) link connects an SSP to an alternate STP. "E" links provide an alternate signaling path if an SSP's "home" STP cannot be reached via an "A" link. "E" links are not usually provisioned unless the benefit of a marginally higher degree of reliability justifies the added expense. |
| F Link: | An "F" (fully associated) link connects two signaling end points (i.e., SSPs and SCPs). "F" links are not usually used in networks with STPs. In networks without STPs, "F" links directly connect signaling points. |

### Types of Signaling:

Signaling in Telecommunications Network

1. Channel Associated Signaling (CAS)
2. Common Channel Signaling (CCS)

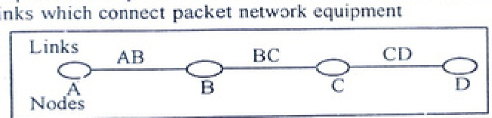
Signaling System Number (SS7) is a form of Common Channel Signaling.

1. Channel Associated Signaling (CAS): Used for In-Band Signaling
2. Signaling is transmitted in the same frequency band as used by voice.
3. Voice path is established when the call setup is complete, using the same path that the call setup signals used.

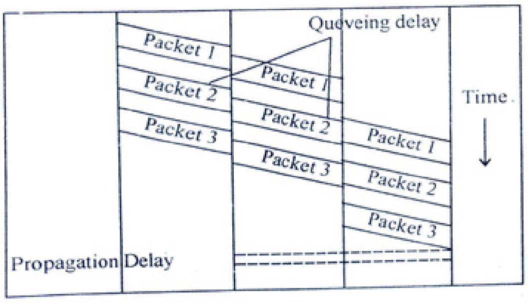
## 4.4 Packet Switching

Packet switching is similar to message switching using short messages. Any message exceeding a network-defined maximum length is broken up into shorter units, known as packets, for transmission; the packets, each with an associated header, are then transmitted individually through the network. The fundamental difference in packet communication is that the data is formed into packets with a pre-defined header format (i.e. PCI) and well-known "idle" patterns which are used to occupy the link when there is no data to be communicated.

Packet network equipment discards the "idle" patterns b/w packet and processes the entire packet as one piece of data. The equipment examines the packet header information (PCI) and then either removes the header (in an end system) or forwards the packet to another system if the out-going link is not available, then the packet is placed in a queve until the link become free. A packet network is formed by links which connect packet network equipment




Communication b/w A and D using circuits which are shared using packet switching.






Packet-switched communication b/w system A and D (The message in this case has been broken into thre parts) There an two important benefits from packet switching.

1. The first and most important benefit is that since packets are short, the communication links b/w the nedes are only allocated to transferring a single message for a short period of time while transmitting each packt. Longer messages require a series of packets to be sent, but do not require the link to be dedicated b/w the transmission of the each packet. The implication is that packets belonging to other message may be sent b/w the packets of the message being sent from A to D. This provides a much faires sharing of the resources of each of the links.

2. Another benefit of packet switching is knows as "pipelining". Pipelining is visible in the figure above. At the time packet 1 is sent from B to C, packet 2 is sent from A to B; packet 1 is sent from C to D while packet 2 is sent from B to C, and packet 3 is sent from A to B, and so forth. This simultaneous use of communication links repersent a gain in efficiency, the total delay for transmission across a packet network may be considerably less than for message switching despite the inclusion of a header in each packet rather than in each message.

**3. Packet switches principle :** A switch used in a packet switch network has a different structure from a switch used in a circuit switched network. We can say that a pocket switch has four components input ports, output ports, the routing processor and the switching fabric.

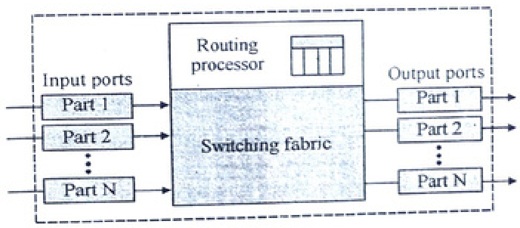



**Fig. 21 : Packet switch components**

**(A) Input ports :** An input port perfoms the physical and data link function of the pocket switch. The pis are constructed from the received signal. The packet is desapsulated from the frame errors are detected and corrected. The packet in now ready to be routed by the network layer.

In addition to a physical layer processor and a data link processor, the input port has buffers (queues) to hold the pocket before it is directed to the switching fabric.


**(B) Output port :** The output port performs the same functions as the input port, but in the reerse order. First the outgoing packets are queved, then the packet is encapsulated in a frame, and finally the physical layer function are applied to the frame to create the signal to be sent on the line.


**(C) Routing processor :** The Routing processor performs the functions of the network layer. The destination address is used to find the address of the next hop and at the same time, the output port number from which the packet is sent out. This activity is sometimes referred to as table 100 kup because the routing processor searches the routing table. In the newer packet switches, this function of the routing processor is being moved to the input ports to facilitate and expedite the process.

**(D) Switching fabrics :** The most difficult task in a pocket switch is to move the packet fro the input queue to the output queue. The speed with which this is done affects the size of the I/O queue and the overall delay in packet delivary. In the past, when a packet switch was actvally a dedicated computer, the memory of the computer or a bus a was used as the switching fabrics. The input port stored to pocket in memory; the output port refrieved th pocket from memory. Today packet switches are specialized machanisms that use a variety of switching fabrics. We briefly discuss some o these fabric here.

(i) Crossbar switch

(ii) Banyan switch

(iii) Batcher Banyan switch

(iv) Batcher-banyan switch

**(i) Cross switch :** The simplest type of switching fabric is the crossbar switch discussed in the previous section.

**(ii) Banyan Switch :** A more realistic approach than the corssbar switch is the banyen switch (named after the banyan tree). A banyan switch is a switch is a multistage switch with microswitchs at each stage that route the packet based



on the output port represented a binary string for n inputs and n outputs, we have $\log_2 n$ stages with n/2 microswitch at each stage. The first stage routes the packet based on the high order bit of the binary string. The second stage routes the packet based on the high orded bit.

**(iii) Batcher Banyan switch :** The problem with the banyan switch is the possibility of internal collision even when two pacekts are not heading for the same output port. We can solve this problem by sorting the arriving pacekts based on their desination port.

K.E. Batcher designed a switch that come before the banyan switch and sorts the incoming packets according to their final destinations.

The combination is called the Bacher-banyan switch.

**(iv) Batcher banyan switch :** The sorting switch uses hardware merging techniques but we do not discuss the details here. Normally, another hardware module called a trap is added between the batcher switch and the lanyan switch. The trap module prevents duplicate packets (packets with the some output distination) form passing to the banyan switch simultaneously. Only one packet for each destination is allowed at each tick; if there is more than one, they wait for the next tick.

**Routing X.25 Protocol :** This protocol is commonly used in wide area communication with multiple communicating devices. Though TCP/IP protocol tias grown much more popular as a wide area networking standard but X.25 protocol is much simpler for interfacing equipment or networks.

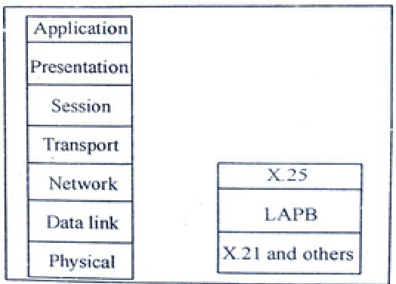



**Fig. 22 : The X.25 Protocol mapped to the OSI model**

As can seen from figure 6.26, X.25 function at the network layer. It normally interfaces with a protocol called LAPB (Link Access Procedures Balanced) at the data link layer, which inturn runs over X.21 or another physical layer CCITT protocol, such as X.21 bis or V.32.

At the network layer X.25 uses a types of addressing called channel addrsing, which is similar to logical network addressing except that the address is mantained for each connection.

X.25 packet switched networks allow remote devices to communicate with each other without the expenze of individual leased lines. One of the key design feature of packet switched networks is redundant error checking, which allows for the limitations of analog voice-grade lines when they are used for data communication. X.25 is a packet switching protocol that defines the interface between a synchronous packet switching most computer and analog dedicated circuits or dial up switched virtual circuits in the voice grade public data network. Used with X.25 are too slow to provide most LAN application services on a WAN. Transmission speeds of up to 64 Kbps are typical.

**X.25 An overview :** The X.25 communications standard is the interface approved by the CCITT plenary session in october 76. It definded the inter connetion of data terminal equipment (DTE) and data circuit termination equipment (DCE) for terminals operating in the packet mode on public data networks. The X.25 standard gains particular importance with its widespread support.

The X.25 protocol defines several levels of interface. At the physical level there is the electrical connection between DCE and DTE. This leel uses the X.21 standard for full duplex synchronous transmission. The X.21 standard is related to familiar American standard like the RS 232 standard.

The second level of X.25clarifies the links access proceduse the link access procedure manages the link between the DTE and the DCE. In dong so, the X.25 standard uses a subset of HDLC. The X.25 borrows a particular group of parameter setting from HDLC for error control, flow control and so on.

The third level of the X.25 standard describes a packet level procedure which controls virtual call through a public data network. This permits the establishment of end to end virtual circuits analogous to vice telephone call.

Finally, the X.25 standard elucidates the function of the packet assembler disassemble (PAD).

Therefore it implements the original design objectives of packet switching. If dominant features are :

(a) Virtual circuit switching and dynamic virtual routing to transport self contianed, self addressed message packets.

(b) Ability to use any available network channels or links.

(c) Ability to use redundent error checking at every node.

X.25 allows a variety of devices that are designeted as data ferminal equipement (DTE) to talk to the public data network (PDN). The PDN is designated as data communication equipment (DCE), as are devices such as modems, packets



switches and other ports. Hardware/Software devices, such as terminals, hosts and routers that deliver data to or from a networks I/O point are DTE. The X.25 Protocol is a DTE to DCE synchronous interface.

To begin communication one DTE device (for example, a router) calls another DTE to request a data exchange sessions. The DTE called can accept or refuse the connection. If the called DTE accepts the connection the two system begin full duplex data transfer Either side can terminate the connection at any time.

X.25 has been used since the mid 1970s and its it is stabl system. There are literally no data errors on modern X.25 netwroks. X.25 does have some drawbacks. The store and forward mechanism causes delays, On most single networks the turn-around delay is about 6 seconds. This does not affact large block transfers, but in transmissions that requires extensive "back the forth" communiation the delay can be very noticeable. Line speeds normally frequency. Although the modulation is done using one carier frequency at at ime, M frequencies are used in the long run. The bandwith occupied by a source after spreading is $B_{FHSS} >> B$.

Figure 6.28 shows the general layout for FHSS. A pseudorandom code generator called pseudorandom noise (PN), creates a k-bit pattern for every hopping period $T_h$. The frequency table uses the pattern to find the frequency to be used for this hopping period and passes it to the frequency synthesizer. The frequency synthesizer creates carrier signal of that frequency and the source signal modulates the carrier signal.

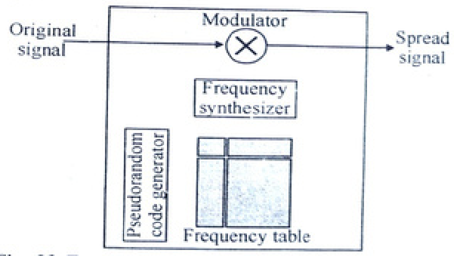



**Fig. 23 Frequency hopping spread spectrum (FHSS)**

Suppose we have decided to have hight hopping frequencies. This is extremely log for real applications and is just for illustration. In this case, M is 8 and k is 3. The psuedic random code generator will create eight different 3-bit patterns. These are mapped eight different frequencies in the frequency table (See figure 6.29).

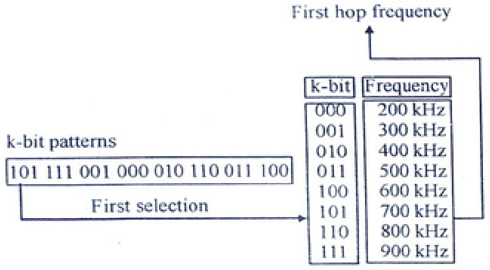



**Fig. 24 : Frequency selection in FHSS**

The pattern for this station is 101, 111, 001, 000, 010, 011, 100. Note thatthe pattern is pseudorandom it is repeated after eight hoppings. This means that at hopping period 1, the pattern is 101. The frequency selected is 700 kHz; the source signal modulates this carrier freuency. The second k-bit pattern selected is 111, which selects the 900-kHz carrier; the eighth pattern is 100, the frequency is 600 kHz. After eight hoppings, the pattern repeats, starting from 101 again, Figure 6.30 shows how the signal hops around from carrier to carroer. We assume the rquired bandwidth of the original signal is 100 kHz.

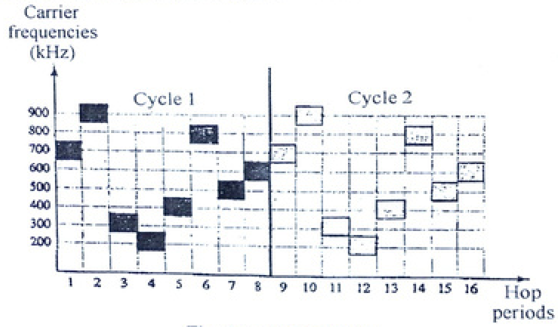



**Fig. 25 : FHSS cycles**

It can be shown that this scheme can accomplish the previously mentioned



goals. If there are many k-bit patterns and the hipping period is short, a sender and receiver can have privacy. If an intruder tries to intercept the transmitted signal, she can only access a small piece of data because she does not know the spreading sequence to quickly adapt herself to the next hop. The scheme has also an antijamming effect. A malicious sender may be able to send noise to jam the signal for one hopping period (randomly), but not for the whole period.

**Bandwidth Sharing :** If the number of hopping frequencies is M, we can multiplex M channels into one by using the same $B_{SS}$ bandwith. This is possible because a station uses just one frequency in each hopping perior; M-1 other frequencies can be used by other M-1 stations. In other words, M different stations can use the same $B_{SS}$ if an appropriate modulation technique such as multiple FSK (MFSK) is used. FHSS is simiiar ot FDM, as shown in Figure 6.31.

Figure 6.31 shows an example of four channels using FDM and four channels using FHSS. In FDM. Each station uses 1/M of the bandwidth, but the allocation is fixed; in FHSS each station uses 1/M of the bandwidth, but the allocation changes hop to hop.

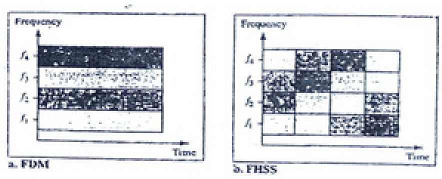



**Fig. 26 : Bandwidth sharing**

**Direct Sequence Spread Spectrum :** The direct sequence spread sepctrum (DSSS) technique also expands the band with of the original signal, but the process is different. In DSSS, we replace each data in with n bits using a spreading code. In other words, each bit is assigned a code of n called chips, where the chip rate is n times that of the data bit. Figure 6.32 show concept of DSSS.

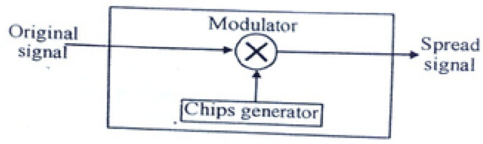



**Fig. 27 DSSS**

As an example, let us consider the sequence used in a wireless LAN, the famous Barker sequence where n is 11. We assume that the original signal and the chips in the chip generator use polar NRZ encoding. Figure 6.33 shows the chips and the result of multiplying the oroginal data by the chips ot get the spread signal.

In Figure 6.33, the spreading code is 11 chips having the pattern 10110111000 (in this case). If the original signal rate is N, the rate ofthe spread signal is 11N. This means that the required bandwith for the sprad signal is 11 times larger than the bandwidth of the original signal. The spread signal can provide privacy if the inctruder does not knw the code. It can also provide immunity against interference if each station uses a different code.

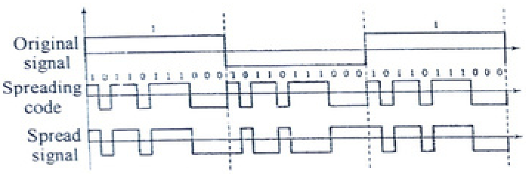



**Fig. 28 : DSSS example**

**Bandwidth Sharing :** Can we share a bandwidth in DSSS as we did in FHSS? The answer is no and yes. If we use a spreading code that spreads signals (from different stations) that cannot be combined and separated, we cannot share a bandwidth. For example, as we will see in chaper 14, some wireless LANs use DSSS and the spread bandwidth cannot be shared. However, if we use a special type of sequence code that allows the combining and separating of spread signals, we can share the bandwidth. As we will see in chapter 16, a special spreding code allows us to use DSSS in cellular telephony and share a bandwidth between several users.

**Spread Spectrum**

Mutliplexing combines signals from several sources to achieve bandwidth efficiency, the available bandwidth of a link is divided between the sources. In spread spectrum (SS), also combine signals from different sources to fit into a larger bandwith but our good are somewhat different spread spectrum is designed to be used in wireless applications (LANs and WANs). In these types of applications, we hae some concerns that outweight bandwidth efficiency. In wireless applications, all stations use air (or a vacuum) as the medium for communication Stations must be able to share this medium without interception by an eavesdropper



and without being subjet to jamming from a malicious intruder (in military operations, for example).

To achieve these goals wspread spectrum techniques ad redundancy; they spread the original spectrum needed for each station. If the required bandwidth for each station is B, spread spectrum expands it to $B_{SS}$, such that $B_{SS} >> B$. The expanded bandwidth allows the source to wrap its message in a protective envelope for a more secure transmission. An analogy isthe sending o a delicate, expensive gift. We can insert the gift in a special box to preent it from being damaged during transportation, and we can used a superior delivery service to guarantee the safety o the package.

Figure 6.27 shows the idea of spread spectrum. Spread spectrum achieves its goals through two principles ;

1. The bandwidth allocated to each stations needs to be, by far, larger tha what is needed. This allows redundancy.

2. The expanding of the original bandwidth B to the bandwidth $B_{SS}$ must be done by a process that is independent of the original signal. In other words, the spreading process occurs after the signal is created by the source.

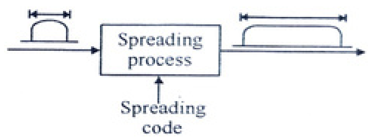



**Fig. 29 : Spread spectrum**

After the signal is created by the source, the spreading process uses a spreading code and spreads the bandwidth. The figure shows the original bandwidth B and the spreaded bandwidth $B_{SS}$. The spreading code is a series of numbers that look random, but are actually a pattern.

There are two techniques to spread the bandwidth: frequency hopping spread spectrum (FHSS) and direct sequence spread spectrum (DSSS).

**Frequency Hopping Spread Spectrum (FHSS)**

The frequency hopping spread spectrum (FHSS) techniques uses M different carrier frequencies that are modulated by the source signal. At one moment, the signal modulated one carrier frequency; atthe next moment, the signal modulates another carrier.

**Asynchronous And Synchronous Data Transimission**

**Asynchronous :** Asynchronous refers to a series of events that takes place which are not synchronized one after the other for example, the time interval between event A and B is not the same as B and C.

**Asynchronous Communications :** Asynchronous communication is often referred to as start up transmission because of the nature. That is the sender can send a character at any time convenient and the receiver will accept it. Asynchronous communication lines remain in an idle state until the hardware on the line is ready to transmit. Since the line is idle, a series of bits has to be sent to the receiving node to notify it that there is data coming. When data is finished, the node has to be notified that the transmission is complete and to go back to an idle state, hence the stop bits are to be sent. This pattern continuous for the duration of the time the link is operative. This is the characteristic of many ferminals when on a terminal, the time spent between successive key stokes would vary. Thus in asynchronous transmission, data is transmitted character by character at irregular intervals.

Synchronous devices usually do not use start and stop bit, so coordination between the two nodes i.e. the sender and the receiver is handled differently. In synchronous communication there are two "channels" one for data and another for link synchronization. The channel for synchronization uses the integral clock in the hardware for link synchronization between the two nodes when one of the nodes is ready to transmit data, a unique combination of bits called a sync. character is sent to the receiver. Since the first character will probably get transhed, a second one usually follows to ensure that synchronization is complete.

Sunchronous mode of data transmission involves blocking a group of chataters in some what the same way records are blocked on magnetic tape. Each block is then framed by header and trailer information which is used by the receiving device to set its clock in synchronism with the sending end clock. The header also contains information to identify sender and receiver. Following the header is a block of characters that contains the actual message to be transmitted. The number of characters may be variable and may consist of hundreds of characters. The message characters in the block are terminated by a trailer. The trailer contain an end of message character followed by a check character to aid detection of any transmission error. Thus with synchronous transmission entire blocks of characters are framed and transmitted together.

Asynchronous transmission is well suited to many key board type terminals. The advantage of this method is that it does not require any local storage at the terminal or the computer as transmission takes place character by character.

Hence it is cheaper to implement. The main disadvantage of asynchronous tansmission is that the transmission line is idle during the time intervals between transmitting character. If there are short, this is not bad because line cost would be low and idle time not expensive. Even through less efficient tan synchronous transmission, it is also used with devices such as card reader and printer to reduce cost.



**Efficiency of data transmission in synchronous and Asynchronous modes**

Asynchronous data incorporates the use of extra framing bits to establish the start and ending (stop) of a data character word. A receiver responds to the data stream when it detects a start bit. A data character is decoded and defind after the stop bit is receive and confirmed. Asynchronous data are easier to detect and synchronized, but the efficiency of data transmission is reduced by the addition of framing bits as overhead (no message data) bit. A comparison of a single character using the two data type is as follows.

**Comparison between Asynchronous and synchronous transmission**

Synchronous transmision is well suited to remote communication between a computer and such devices as buffered card readers and printers. It is also used for computer to computer communications.

The primry advantage of synchronous transmission is its efficiency. Not only does it diminate the need for individual start stop bit an each character, but much higher data rates can be used than with asynchronous transmission. The period between blocks is kept small and the block itself is sent at nearly the maximum line speed. This ensures efficient utilization of the transmission line. The main disadvantage is the need for local buffer storage at the two ends of the line to assemble blocks and also the need for accurately synchronized clocks at both ends. Therefore synchronous equipment usually cost more.

For this purpose, the singnificant Bit (LSB) first. The number of framing bits used for asynchronous data varies depending on the stations in the communication link.

For examle, suppose we use 1 start and 2 stop bits. This adds 3 addition bits to the character word. Hence total 10 bits are required to send the letter E using asynchronous data. However, in the case of synchronous transmission only 7 bits are required for transmission of the character E.

The efficiency to transmission is defined as the ratio of the number of message bits to the total number of transmitted bit.

$$\text{or efficiency} = \frac{\text{Data bits}}{\text{Total bits}} \times 100\%$$

As seen in the above example, for the letter E. i.e. 1000101, the synchronous mode, all bits carry message, so 100% efficiency is there.

However, in asynchronous mode, E is transmited by using 7 bits as message and another three bits (one start bit and two stop bits) totaling 10 bits so the efficiency is calculated as :

$$\text{efficiency} = \frac{\text{data bits i.e. } 7}{\text{total bits i.e. } 10} \times 100\% = 70\%$$

**Data Transmission Modes**

There are three modes of data transmission that correspond to the three types of circuits available. (See Figure 1.9). These are :

(a) Simplex

(b) Half-duplex

(c) Full-duplex

**(a) Simplex :** Simplex communications imply a simple method of communiating, which they are. In simplex communications mode, there is a one-way communiction transmission. Television transmission is a good example of simplex communications. The main transmitter sends out a signal (broadcast), but it does not expect a reply as the receiving units cannot issue a reply back to the transmitter. A data collection terminal on a factory floor (send only) or a line printer (receiver only). Another example of simplex communication is a keyboard attached to a computer because the keyboard can only send data to the computer.

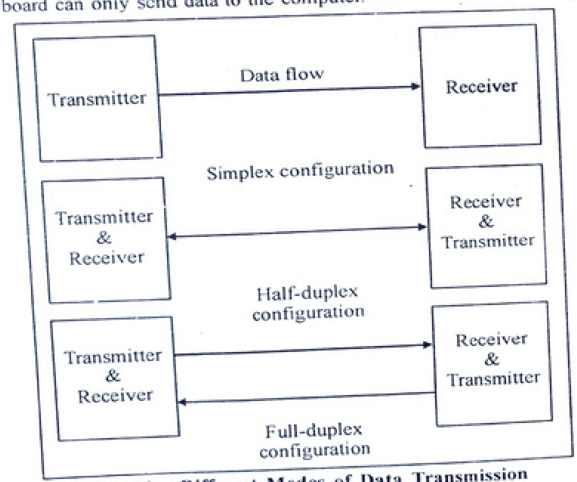



**Fig. 30 : Different Modes of Data Transmission**

At first though it might appear adequate for many types of application



which flow of information is unidirectional. However, in almost all data processing application, communication in both directions is required. Even for a "one-way" flow of information from a terminal to a computer, the system will be designed to allow the computer to signal the terminal that data has been received. Without this capacity, the remote user might enter data and never know that it was not received by the other terminal. Hence, simple circuits are seldom used because a return path is generally needed to send acknowledgement control or error signals.

**Half duplex :** In half-duple mode, both units communicate over the same medium but only one unit can send at a time. While one is in send mode, the other unit is in receive mode. It is like two polite people talking to each other-one talks, the other listens but neither one talks at the same time. Thus a half duplex line can alternately send and receive data. It requires two wires. This is the most common type of transmission for voice communications because only one persion is supposed to speak at a time. It is also used to connect a terminal with a computer. The terminal might transmit data and then the computer responds with an ancknowledgement. The transmission of data to and from a hard disk is also done in half duplex mode.

**Full duplex :** In a half duplex system the line must be "turned around" each time the direction is reversed. This involves a special switching circuit and requires a small amount of time. With high speed capabilities of the computer, this turn-around time is unacceptable in many instances. Also, some applications require simultaneous transmission in both directions. In such cases, a full-duplex system is used that allows information to flow simultaneously in both direction on the transmission path. Use of a full duplex line improves efficiency as the line turn around time required in a half-duplex arrangment is eliminated. It requires four wires.

**Data Transmission Modes**

There are three modes of data transmission that correspond to the three types of circuits available (see fig.) these are :

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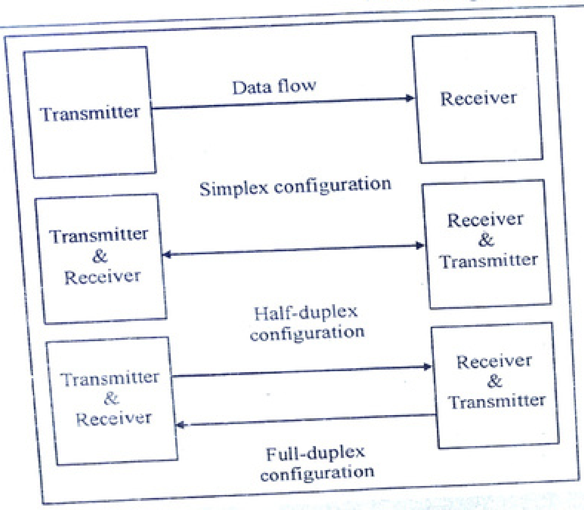


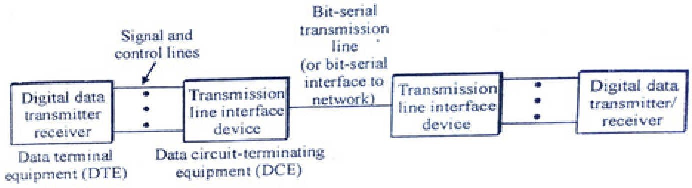

## 4.5 Interfacing

Most digital data processing devices have limited data transmission capability. Typically, they generate a simple digital signal, such as NRZ-L, and the distance across which they can transmit data is limited. Consequently, it is rare for such a device (terminal, computer) to attach directly to a transmission or networking facility. The more common situation is depicted in Figure 6.10. The devices we are discussing, which include terminals and computers, are generically referred to as **data terminal equipment (DTE).** A DTE makes use of the transmission system through the mediation of **data circuit-terminating equipment (DCE).** A example of the latter is a modem.

On one side, the DCE is responsible for transmitting and receiving bits, one at a time, over a transmission medium or network. On the other side, the DCE must interact with the DTE. In general, this requires both data and control information to be exchanged. This is done over a set of wires referred to as **interchange circuits.** For this scheme to work, a high degree of cooperation is required. The two DCEs that exchange signals over the transmission line or network must understand each other. That is, the receiver of each must use the same encoding scheme (e.g., Manchester, PSK) and data rate as the transmitter of the other. In addition, each DTE-DCE pair must be designed to interact cooperatively. To ease



the burden on data processing equipment manufactures and users, standards have been developed that specify the exact nature of the interface between the DTE and the DCE. Such an interface has four important characteristics :




**(a) Generic interface to transmission medium**


**(b) Typical configuration**

**Fig. 31 : Data Communications Interfacing**

- Mechanical
- Electrical
- Functional
- Prcedural

The mechanical characteristics pertain to the actual physical connection of the DTE to the DCE. Typically, the signal and control interchange circuits are bundled into a cable with a terminator connector, male or female, at each end. The DTE and DCE must present connectors of opposite genders at one end of the cable, effecting the physical connection. This is analogous to the situation for residential electrical power. Power is provided via a socket or wall outlet, and the device to be attached must have the appropriate male connector (two-pronged, two-pronged polarized, or three-pronged) to match the socket.

The electrical characteristics have to do with the voltage levels and timing of voltage changes, Both DTE and DCE must use the same code (e.g. NRZ-L), must use the same voltage levels to mean the same things, and must use the same duration of signal elements. These characteristics determine the data rates and distances that can be achieved.

Functional characteristics specify the functions that are performed by assigning meanings to each of the interchange circuits. Functions can be classified into the broad categories of data, control, timing, and electrical ground.

Procedural characteristics specify the sequence of events for transmitting data, based on the functional characteristics of the interface. The examples that follow should clarify this point.

A variety of standards for interfacing exists. This section presents two of the most important : V.24/EIA-232-F and the ISDN Physical Interface.

### V.24/EIA-232-F

One of the most widely used interfaces is specified in the ITU-T standard, V.24. In fact, this standard specifies only the functional and procedural aspects of the interface; V.24 references other standards for the electrical and mechanical aspects. In the United States, there is a corresponding specification, virtually identical, that covers all four aspects: EIA-232 E. The correspondence is as follows:

- Mechanical : ISO 2110
- Electrical : V.28
- Functional : V.24
- Procedural V.24

EIA-232 was first issued by the Electronic Industries Alliance in 1962, as RS-232. It is currently in its sixth revision, EIA-232-F, issued in 1997. The current V.24 and V.28 specifications were issued in 1996 and 1993, respectively. This interface is used to connect DTE devices to voice-grade modems for use on public analog telecommunications systems. It is also widely used for many other interconnection applications.

### Mechanical Specification

The mechanical specification for EIA-232-F is illustrated in Figure 6.11. It calls for a 25-pin connector, defined in ISO 2110, with a specific arrangment of leads. This connector is the terminating plug or socket on a cable running from a DTE (e.g., terminal) or DCE (e.g., modem). Though a 25 wire cable could be used to connect the DTE to the DCE, many applications require far fewer wires.

### Electrical Specification

The electrical specification defines the signaling between DTE and DCE. Digital signaling is used on all interchange circuits. Depending on the function of the interchange circuit, the electrical values are interpreted either as binary data or as control signals. The convention specifies that, with respect to a common ground, a voltage more negative than-3 volts is interpreted as binary 1 and a voltage more positive than +3 volts is interpreted as binary 0. This is the NRZ-L code illustrated in Figure 5.2. The interface is rated at a signal rate of <20 kbps and a distance of < 15 meters. Greater distances and data rates are possible with



good design, but it is prudent to assume that these limits apply in practice as well as in theory.

The same voltage levels apply to control signals : a voltage more negative than -3 volts is interpreted as an OFF condition and a voltage more positive than +3 volts is interpreted as an ON condition.

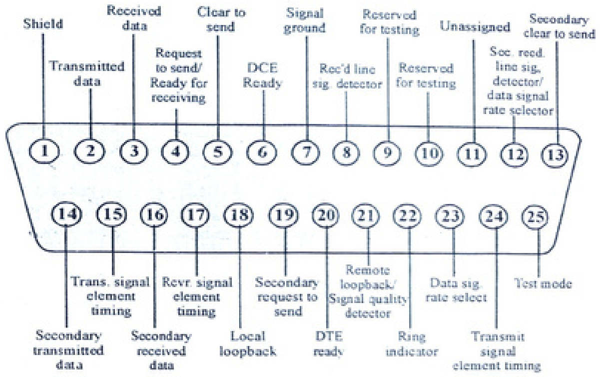



**Fig. 32 : In Assignments for V.24/EIA-232 (DTE Connector Face)**

### Functional Specification

Table 6.1 summarizes the functional specification of the interchange circuits, and Figure 6.11 illustrates the placement of these circuits on the connector. The circuits can be grouped into the categories of data, control, timing, and ground. There is one data circuit in each direction, so full-duplex operation is possible . In addition, there are two secondary data circuits that are useful when the device operates in a half-duplex fashion. In the case of half-duplex operation, data exchange between two DTEs (via their DCEs and the intervening communications link) is only conducted in one direction at a time. However, there may be a need to send a halt or flow control message to a transmitting device. To accommodate this, the communication link is equipped with a reverse channel, usually at a much lower data rate than the primary channel. At the DTE-DCE interface the reverse channel is carried on a separate pair of data circuits.

**Table 6.1 : V.24/EIA-232-F Interchange Circuits**

| V.24 | EIA-232 | Name | Direction to | Function |
|---|---|---|---|---|
| | | DATA SIGNALS | | |
| 103 | BA | Transmitted data | DCE | Transmitted by DTE |
| 104 | BB | Received data | DTE | Received by DTE |
| 118 | SBA | Secondary transmitted data | DCE | Transmitted by DTE |
| 119 | SBB | Secondary received data | DTE | Received by DTE |
| | | CONTROL SIGNAL | | |
| 105 | CA | Request to send | DCE | DTE wishes to transmit |
| 106 | CB | Clear to send | DTE | DCE is ready to receive response to Request to send |
| 107 | CC | DCE ready | DTE | DCE is ready to operate |
| 108.2 | CD | DTE ready | DCT | DTE is ready to operate |
| 125 | CE | Ring indicator | DTE | DCE is receiving a ringing signal on the channel line |
| 109 | CF | Received line signal detector | DTE | DCE is receiving a signal within appropriate limits on the channel line |
| 110 | CG | Signal quality detector | DTE | Indicates whether there is a high probability of error in the received |
| 111 | CH | Data signal rate selector | DCE | Selects one of two data rates |
| 112 | CI | Data signal rate selector | DTE | Selects one of two data rates |
| 133 | CI | Ready for receiving | DCE | On/off flow control |
| 120 | SCA | Secondary request to send | DCE | DTE Wishes to transmit on reverse channel |
| 121 | SCB | Secondary clear to send | DTE | DCE is ready to receive on reverse channel |
| 122 | SCF | Secondary received line signal detector | DTE | Same as 109, for reverse channel |
| 140 | RL | Remove loopback | DCE | Instructs remote DCE to loop back signals |
| 141 | LL | Local loopback | DCE | Instructs DCE to loop back signals |
| 142 | TM | Test mode | DTE | Local DCE is in a test condition |
| | | TIMING SIGNALS | | |
| 113 | DA | Transmitter signal element timing | DCE | Clocking signal transition to ON and OFF occur at center of each signal element |
| 114 | DB | Transmitter signal element timing | DTE | Clocking signal both 113 and 114 relate to signals on circuit 103 |
| 115 | DD | Receiver signal element timing | DTE | Clocking signal for circuit 104 |
| | | GROUND | | |
| 102 | AB | Signal ground/common return | | Common ground reference for all circuits |

There are 16 control circuits. The first 10 of these listed in Table 6.1 relate to the transmission of data over the primary channel. For asynchronous transmission, six of thse circuits are used (105, 106, 107, 108.2, 125, 109). The use of these

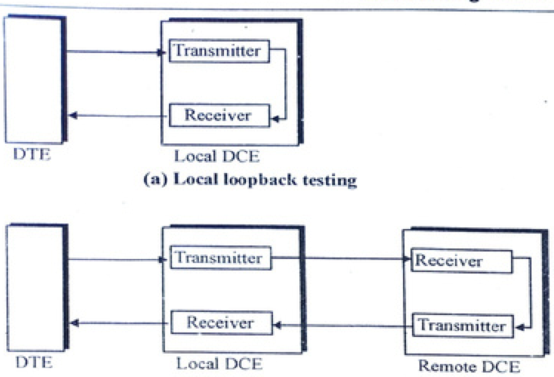


circuits is explained in the subsection on procedural specification. In addition to these six circuits, three other control circuits are used in synchronous transmission. The Signal Quality Detector circuit is turned ON by the DCE to indicate that the quility of the incoming signal over the telephone line has deteriorated beyond some defined threshold. Most high-speed modems support more than one transmission rate so that they can fall back to a lower spped if the telephone line becomes noisy. The Data Signal Rate Selectro circuits are used to change speeds; either the DTE or DCE may initiate the change. Circuit 133 enables a receiver to turn the flow of data on circuit 104 on and off. The next three control circuits (120, 121, 122) are used to control the use of the secondary channel, which may be used as a reverse channel or for some other auxiliary purpose.

The last group of control signals relates to loopback testing. These circuits allow the DTE to cause the DCE to perform a loopback test. These circuits are only valid if the modem or other DCE supports loopback control; this is now a common modem feature. In the local loopback function, the transmitter output of the modem is connected to the receiver input, disconnecting the modem from the transmission line. A stream of data generated by the user device is sent to the modem and loopedback to the user device. For remote loopback, the local modem is connected to the transmission facility in the usual fashion, and the receiver output of the remote modem is connected to the modem's transmitter input. During either form of test, the DCE turns ON the Test Mode circuit. Table 6.2 show the setting for all of the circuits related to loopback testing, and Figure 6.12 illustrates the use.

Loopback control is a useful fault isolation tool. For example, suppose that a user at a personal computer is communicating with a server by means of a modem connection and communication suddenly ceases. The problemk could be with the local modem, the communications facility, the remote modem, or the remote server. A network manager can use loopback tests to isolate the fault. Local loopback checks the functioning of the local interface and the local DCE. Remote loopback tests the operation of the transmission channel and the remote DCE.

| Local Loopback | | Remote Loopback | | |
|---|---|---|---|---|
| Circuit | Condition | Circuit | Local Interface | Remote Interface |
| DCE Ready | ON | DCE Ready | ON | OFF |
| Local Loopback | ON | Local Loopback | OFF | OFF |
| Remote Loopback | OFF | Remote Loopback | ON | OFF |
| Test Mode | ON | Test Mode | ON | ON |




(b) Remote loopback testing

**Fig. 33 : Local and Remote Loopback**

The timing signals provide clock pulses for synchronous transmission. When the DCE is sending synchronous data over the Received Data circuit (104), it also sends 1-0 and 0-1 transitions on receiver Element Signal Timing (115), with transitions timed to the middle of each Received Data signal element. When the DTE is sending synchronous data, either the DTE or DCE can provide timing pulses, depending on the circumstances.

Finally, the signal ground/common return (102) serves as the return circuit for all data leads. Hence, transmission is unbalanced, with only one active wire. Balanced and unbalanced transmission are discussed in the section on the ISDN interface.

### Procedural Specification

The procedural specification defines the sequence in which the various circuits are used for a particular application. We give afew examples.

The first example is a very common one for connecting two devices over a short distance within a building. It is known as an asynchronous private line modem or a limited distance modem. As the name suggests, the limited distance modem accepts digital signals from a DTE, such as a terminal or computer, converts these to analog signals, and then transmits these over a short length of medium, such as twisted pair. On the other end of the line is another limited distance modem, which accepts the incoming analog signals, converts them to digital, and passes them on to another terminal or computer. Of course, the exchange of data is two way. For this simple application, only the following interchange circuits are actually required :



- Signal Ground (102)
- Transmitted Data (103)
- Received Data (104)
- Request to Send (105)
- Clear to Send (106)
- DCE Ready (107)
- Received Line Signal Detector (109)

When the modem (DCE) is turned on and is ready to operate, it asserts (applies a constant negative voltage to) the DCE Ready line. When the DTE is ready to send data (e.g., the terminal user has entered a character), it asserts Request to Send. The modem responds, when ready, by asserting Clear to Send, indicating that data may be transmitted over the Transmitted Data line. If the arrangment is half duplex, then Request to Send also inhibits the receive mode. The DTE may now transmit data over the Transmitted Data line. When data arrive from the remote modem, the local modem asserts Received Line Signal Detector to indicate that the remote modem is transmitting and delivers the data on the Received Data line. Note that it is not necessary to use timing circuits, because this is asynchronous transmission.

The circuits just listed are sufficient for private line point-to-point modems, but additional circuits are required to use a modem to transmit data over the telephone network. In this case, the initiator of a connection must call the destination device over the network. Two additional leads are required :

- DTE Ready (108.2)
- Ring Indicator (125)

With the addition of these two lines, the DTE modem system can effectively use the telephone network in a way analogous to voice telephone usage. Figure 6.13 depicts the steps involved in dial-up half-duplex operation. When a call is made, either manually or automatically, the telephone system sends a ringing signal. A telephone set would respond by ringing its bell; a modem responds by asserting Ring Indicator. A person answers a call by lifting the handset; a DTE answers by as serting DTE Ready. A person who answers a call will listen for another's voice, and if nothing is heard, hang up. A DTE will listen for Received Line Signal Detector, which will be asserted by the modem when a signal is present if this circuit is not asserted, the DTE will drop DTE Ready. You might wonder how this last contingency might arise. One common way is if a person accidentally dials the number of a modem. This activates the modem's DTE, but when no carrier tone comes through, the problem is resolved.

It is instructive to consider situations in which the distances between devices are so close as to allow two DTEs to signal each other directly. In this case, the V.24/EIA-232 interchange circuits can still be used, but no DCE equipment is provided. For this scheme to work, a null modem is needed, which interconnects leads in such a way as to fool both DTEs into thinking that they are connected to modems. Figure 6.14 is an example of a null modem configuration; the reasons for the particular connections should be apparent to the reader who has grasped the preceding discussion.

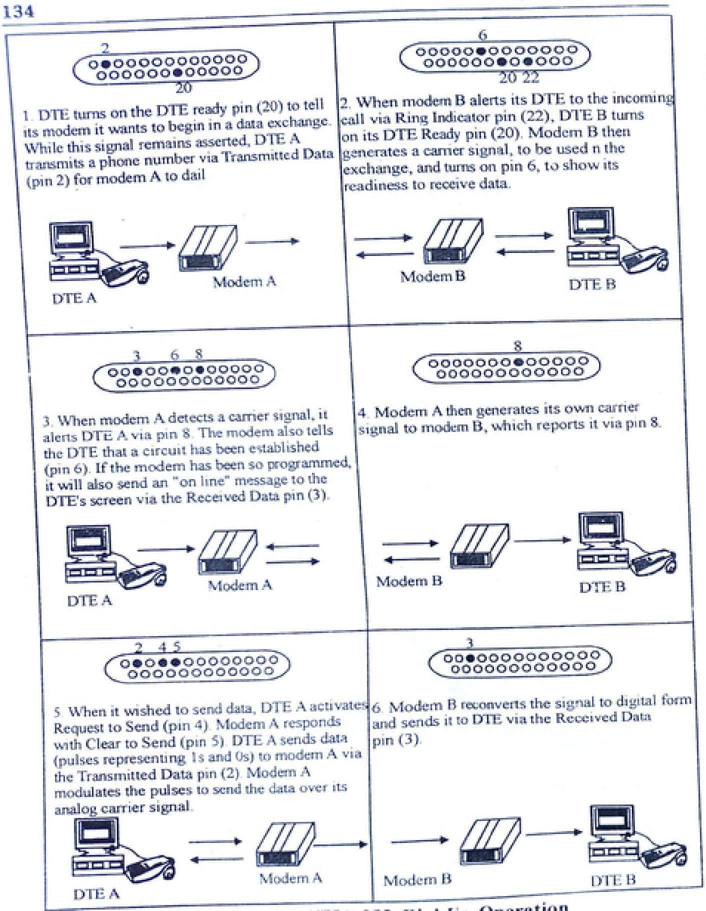



**Fig. 6.13 : V.24/EIA-232 Dial-Up Operation**



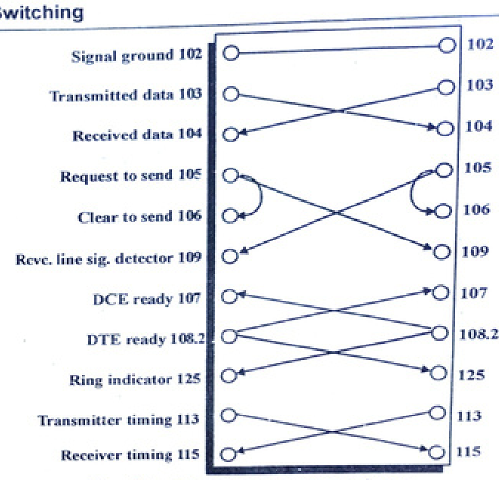



**Fig. 34 : Example of a Null Modem**

### ISDN Physical Interface

The wide variety of functions available with V.24/EIA-232 is provided by the use of a large number of interchange circuits. This is a rather expensive way to achieve results. An alternative would be to provide fewer circuits but to add more logic at the DTE and DCE interfaces. With the dropping costs of logic circuitry, this is an attractive approach. This approach was taken in the X.21 standard for interfacing to public circuit-switched network, specifying a 15-pin connector. More recently, the trend has been carried further with the specification of an 8-pin physical connector to an Integrated Services Digital Network (ISDN). ISDN is an all-digital replacement for existing public telephone and analog telecommunications networks. In this section, we look at the physical interface defined for ISDN.

#### Physical Connection

In ISDN terminology, a physical connection is made between terminal equipment (TE) and network-terminating equipment (NT). For purposes of our discussion, these terms correspond, rather closely, to DTE and DCE, respectively. The physical connection, defined in ISO 8877, specifies that the NT and TE cables shall terminate in matching connections that provide for eight contacts.

Figure 6.15 illustrates the contact assignments for each of the eight lines on both the NT and TE sides. Two pins are used to provide data transmission in each direction. These contact pins are used to connect twisted-pair leads coming from the NT and TE devices. Because there are no specific functional circuits, the transmit receive circuits are used to carry both data and control signals. The control information is transmitted in the form of messages.

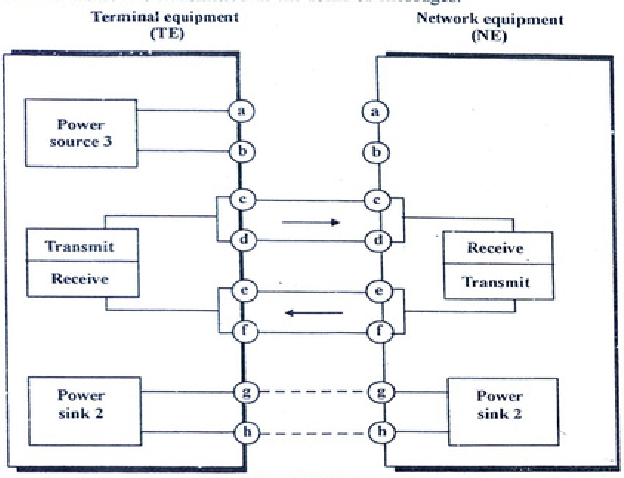



**Fig. 35 : ISDN Interface**

The specification provides for the capability to transfer power across the interface. The direction of power transfer depends on the application. In a typical application, it may be desirable to provide for power transfer from the network side toward the terminal to, for example, maintain a basic telephony service in the event of failure of the locally provided power. This power transfer can be accomplished using the same leads used for digital signal transmission (c, d, e, f), or on additional wires, using access leads g-h. The remaining two leads are not used in the ISDN con figuration but may be useful in other configurations.

### Electrical Specification

The ISDN electrical specification dictates the use of balanced transmission. With balanced transmission signals are carried on a line, such as twisted pair, consisting of two conductors. Signals are transmitted as a current that travels



down one conductor and returns on the other, the two conductors forming a complete circuit. For digital signals, this technique is known as differential signaling,[6] as the binary value depends on the direction of the voltage difference between the two conductors. Unbalanced transmission, which is used on older interfaces such as EIA-232, uses a single conductor to carry the signal, with ground providing the return path.

The balanced mode tolerates more, and produces less, noise than unbalanced mode. Ideally, interference on a balanced line will at equally on both conductors and not affect the voltage difference. Because unbalanced transmission does not posses these advantages, it is generally used only on coaxial cable. When it is used on interchange circuits, such as EIA-232, it is limited to very short distances.

The data encoding format used on the ISDN interface depends on the data rate. For the basic rate of 192 kbps, the standard specifies the use of pseudoternary coding (Figure 5.2). Binary one is represented by the absence of voltage, and binary zero is represented by a positive or negative pulse of 750 mV ± 10%. For the primary rate, there are two options : 1.544 Mbps using alternate mark inversion (AMI) with B8ZS and 2.048 Mbps using AMI with HDB3. The reason for the different schemes for the two different primary rates is simply historical; neither has a particular advantage.

## Exercises

**Very Short Questions** **[2 marks each]**

1. What is packet switching ?
2. Explain circuit switched network ?
3. What is time and space division switch ?
4. Whatis synchronous and asynchronous transmission.
5. What is X-25 ?
6. What is spread spectrum ?
7. What is transmission media ?
8. How many types of transmission media we use ?
9. What is synchronous time division multiplexing ?
10. What is telephone network ?

**Short Questions** **[4 marks each]**

1. Explain circuit switched network ?
2. What technology use of circuit switched in telephone network ?
3. Comparison between Synchronous and Asynchronous transmission media.

4. What is X.25 ? Explain

## Long Questions [12 marks each]

1. Explain circuit switched network with diagram.
2. Explain packet switched network and comparison with circuit switched network.
3. What is spread spectrum. Explain with diagram.
4. What is Data transmission mode ? Explain compare asynchronous and synchronous transmission.
5. Explain X.25 with diagram.