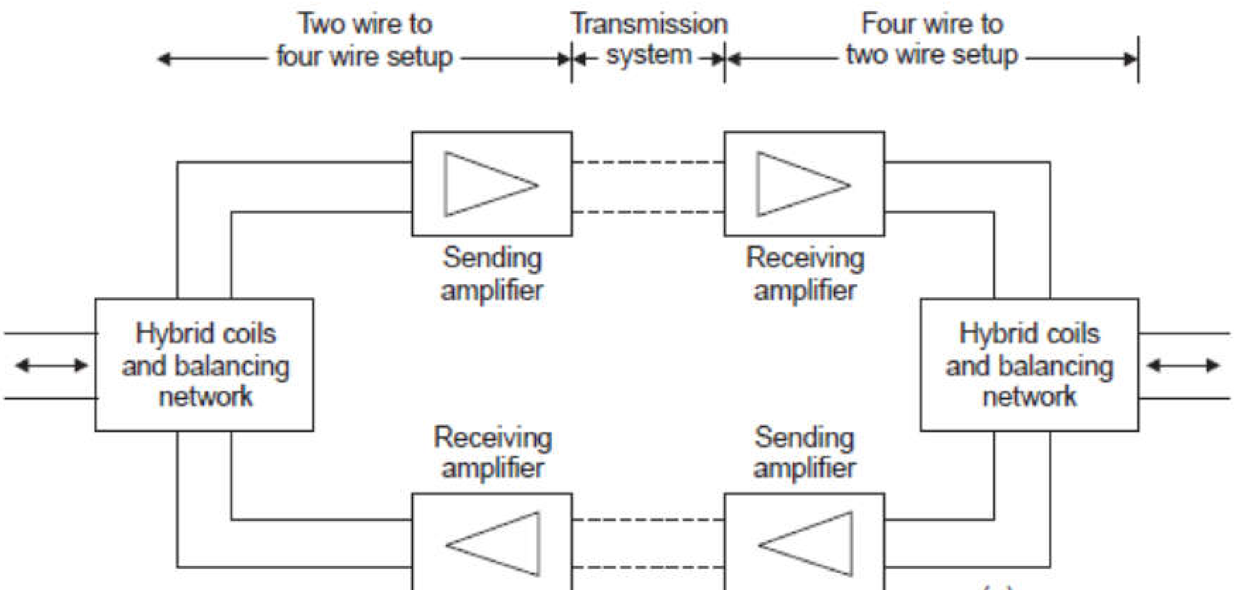
1) **Explain the operation of a four wire circuit connected to a two wire circuit through a hybrid transformer. Derive the Listener and talker echo attenuation**. [2.5+2.5+2+3 = 10M]

The term four wire implies that there are two wires carrying the signals in one direction and two wires carrying them in opposite direction. In normal telephone service, the local loops are two wire circuits, on which a single telephone call can be transmitted in both directions. If the distance between the subscribers is substantial, the amplifiers (repeaters) are necessary to compensate the attenuation. *As the amplifiers are unidirectional,* for two-way communication, four-wire transmission is necessary. The switching equipment in the local exchange and the line from subscriber to local office (local loops) are two wire operation. The local exchange will switch the subscriber loop to a toll connecting trunk. This is also a two-wire transmission. Telephone and Transmission Systems **35** The toll offices are interconnected with inter tool trunks (which connects towns and cities). These trunks are of four-wire transmission. Fig. a (below) shows the simple arrangement of the two wire and four wire transmission.



(a)

A four-wire circuit has amplifiers in its repeaters for each direction of transmission. The four wire circuits may be physical four wire or equivalent four wire. For short distances, actual four wires used for transmission is referred as physical four wire circuits. But for long distance trunks physical four wire is undesirable and usually equivalent four wire transmission is used, needing one pair of wires only. The two directions of transmission use different frequency bands so that they do not interfere with each other. The two directions are separated in frequency rather than space. At the toll office, the two wires are converted into four wire for long transmission. A hybrid coil accomplishes this conversion.

Echos and Singing:

Echoes and singing both occurs as a result of transmitted signals being coupled into a return path and fed back to the respective sources. Coupling will be zero only when perfect impedance matching occurs. Impedance matching between trunks and subscriber loop (two wire to four wire at hybrid) is difficult due to various subscriber loop lengths. A signal reflected to the speaker's end of the circuit is called **talker echo** and at the listener's end is called **listeners echo.** The talker echo is more troublesome. When the returning signal is repeatedly coupled back into the forward path to produce oscillations, **singing** occurs. Basically singing results if the loop gain at some frequency is greater than unity. An echo coming 0.5 msec after the speech is not much effect. The echoes with a round trip delay of more than 45 msec cannot be tolerated. Fig. 3.12 explains the path of the echo and the losses and gain of the signals at various parts of the system.

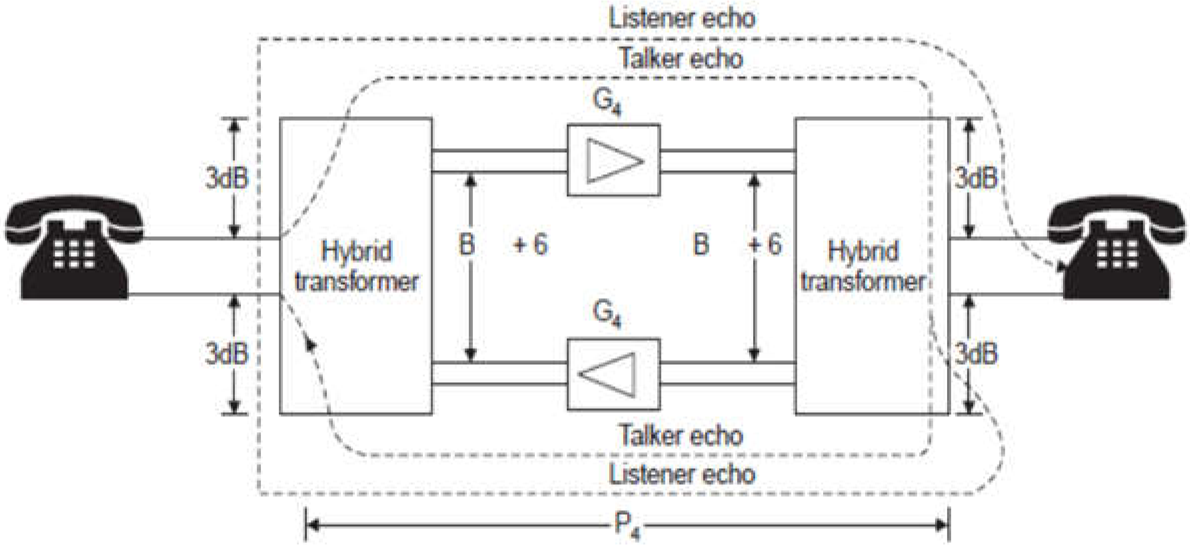


Fig. 3.12. Block diagram of echo path.

Total attenuation from one two wire circuit to the other is

$$L_2 = 6 - G_4$$

G4 is Net gain of one side of four wire circuit (i.e Total amplifier gain minus total line loss)

The attenuation through the hybrid transformer from one side of the four-wire circuit to the other is called the *transhybrid loss*. It can be shown[2] that this loss is 6 + B dB, where

$$B = 20 \log_{10} \left| \frac{N + Z}{N - Z} \right| \text{dB} \qquad (2.5)$$

Z is the impedance of the two-wire line, N is the impedance of the balance network.

The loss B represents that part of the transhybrid loss which is due to the impedance mismatch between the two-wire line and the balance network. It is known as the *balance-return loss* (BRL).

The attenuation L_t of the echo that reaches the talker's two-wire line, round the path shown in Figure 2.3, is

$$L_t = 3 - G_4 + (B + 6) - G_4 + 3 \text{ dB} = 2L_2 + B \text{ dB}$$

This echo is delayed by a time

$$D_t = 2T_4$$

where T_4 is the delay of the four-wire circuit (between its two-wire terminations). The attenuation L_1 of the echo that reaches the listener's two-wire line (relative to the signal received directly) is

$$L_1 = (B + 6) - G_4 + (B + 6) - G_4 \text{ dB} = 2L_2 + 2B \text{ dB}$$

and this is delayed by a time $2T_4$ relative to the signal received directly.

2) **A)Explain the following power levels in dBm and dBW [4M]**

(i) 1 mW, (ii) 1 W, (iii) 2 mW, (iv) 100 mW.

(i) 1 mW = 0 dBm = − 30 dBW.
(ii) 1 W = 0 dBW = + 30 dBm.
(iii) 2 mW = 0 dBm + 3 dB = + 3 dBm = − 30 dBW + 3 dB = − 27 dBW.
(iv) 100 mW = 0 dBm + 20 dB = + 20 dBm = − 30 dBW + 20 dB = − 10 dBW.

B) Explain with a neat diagram, the 30 channel PCM frame format with all calculations included. [2+3+1 = 6M]

PCM systems were first developed for telephone transmission over cables originally designed for audio-frequency transmission. It was found that these are satisfactory, using suitable bipolar coding,[11] for transmitting up to 2 Mbit/s. Consequently, telephone channels are combined by time-division multiplexing to form an assembly of 24 or 30 channels. This is known as the *primary multiplex group*. It is also used as a building block for assembling larger numbers of channels in higher-order multiplex systems, as described in Sections 2.6.3 and 2.6.4.

The operation of a primary multiplexer is shown in Figures 2.11 and 2.12. The length of the frame is 125 μs, corresponding to the sampling interval. It contains one speech sample from each channel, together with additional digits used for synchronization and signalling. Two frame structures are widely used: the European 30-channel system and the DS1 24-channel system used in North America and Japan. Both systems employ 8-bit coding: however, the 30-channel system uses A-law companding and the 24-channel system uses the mu law.

As shown in Figure 2.11, the frame of the 30-channel system is divided into 32 time-slots, each of 8 digits. Thus, the total bit rate is 8 kHz × 8 × 32 = 2.048 Mbit/s. Time-slots 1 to 15 and 17 to 31 are each allotted to a speech channel. Time-slot 0 is used for frame alignment. Time-slot 16 is used for signalling, as described in Section 8.5.

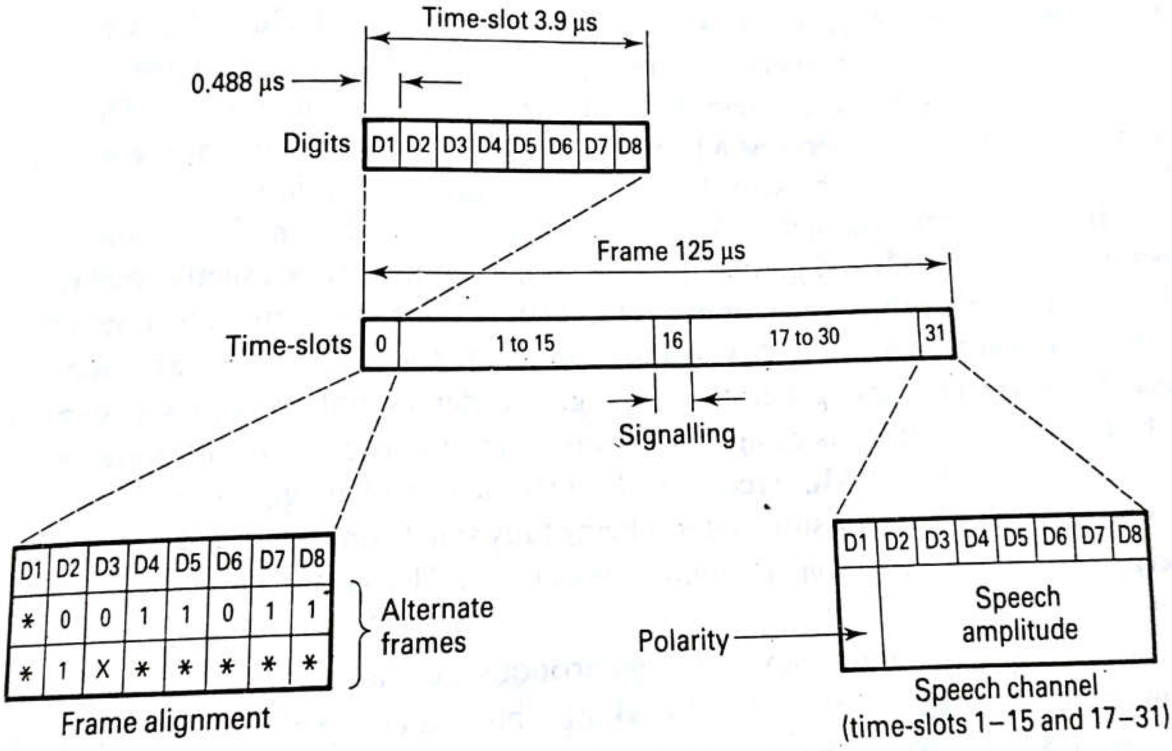


Figure 2.11 30-channel PCM frame format.

30 channel PCM:

world wide agreed Max voice frequency (f_m) = 4KHZ

Sampling rate $(f_s) = 2f_m = 2 \times 4K$

$= 8000$ samples/sec

Total bit rate $= 8K \times 8 \times 32$

$= 2.048$ Mbit/sec

3) Write in detail on Plesiochronous digital hierarchy.[5+1+2+2 = 10M]

In a PDH network you have different levels of Multiplexers.
Figure 1 shows three levels of multiplexing:-

· 2Mbit/s to 8Mbit/s
· 8Mbit/s to 34Mbit/s
· 34Mbit/s to 140Mbit/s

So to carry a 2Mbit/s data stream across the 140Mbit/s trunk requires it to be multiplexed up through the higher order multiplexers into the 140Mbit/s trunk and then to be multiplexed down through the lower order multiplexers.
Because Plesiochronous is not quite Synchronous each of the multiplexers need a little bit of overhead on their high speed trunks to cater for the slight differences in data rates of the streams on the low speed ports. Some of the data from low speed ports (that are running too fast) can be carried in the trunk overhead, and this can happen at all multiplexing levels. This is known as Justification or Bit Stuffing.

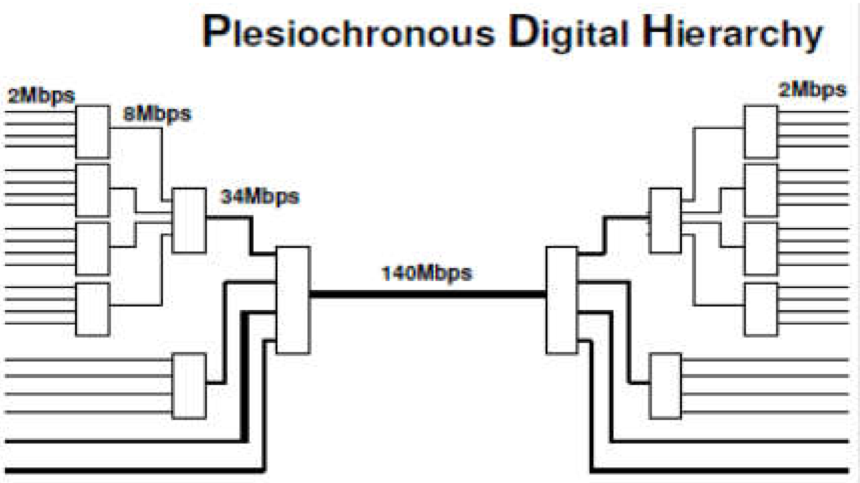


Figure 1

PDH Multiplexing Hierarchy

Figure 2 shows that there are two totally different hierarchies, one for the US and Japan and another for the rest of the world. The other thing to notice is that the different multiplexing levels are not multiples of each other. For example CEPT2 supports 120 Calls but it requires more than 4 times the bandwidth of CEPT1 to achieve this. This is because PDH is not exactly synchronous and each multiplexing level requires extra bandwidth to perform Bit Stuffing. So the Plesiochronous Hierarchy requires ―Bit Stuffing‖, at all levels, to cater for the differences in clocks. This makes it particularly difficult to locate a particular 2Mbit/s stream in the 140Mbit/s trunk unless you fully de-multiplex the 140Mbit/s stream all the way down to 2Mbit/s.

2.2 Drop & Insert a 2Mbit/s stream To drop & insert a 2Mbit/s stream from a 140Mbit/s trunk you need to break the 140Mbit/s trunk and insert a couple of ―34Mbit/s to 140Mbit/s‖ multiplexers. You can then isolate the appropriate 34Mbit/s stream and multiplex the other 34Mbit/s streams back into the 140Mbit/s trunk. Then you de-multiplex the 34Mbit/s stream, isolate the appropriate 8Mbit/s Stream and multiplex the other 8Mbit/s streams through the higher layer multiplexer, into the 140Mbit/s trunk.

PDH Multiplexing Levels

	United States & Japan			Europe & Australia		
Multiplexing Level	Name	# calls	Rate (Mbps)	Name	# calls	Rate (Mbps)
1	DS1	24	1.544	CEPT1	30	2.048
2	DS2	96	6.312	CEPT2	120	8.448
3	DS3	672	44.736	CEPT3	480	34.368
4	DS4	4032	274.176	CEPT4	1920	139.264

Figure 2

→ able to manage large no. of clocks located at remote locations

Chain of equipment used in this N/w called as PDH.

→ The inputs to the digital multiplexer is not exactly ~~an~~ synchronous hence the name plesiochronous.

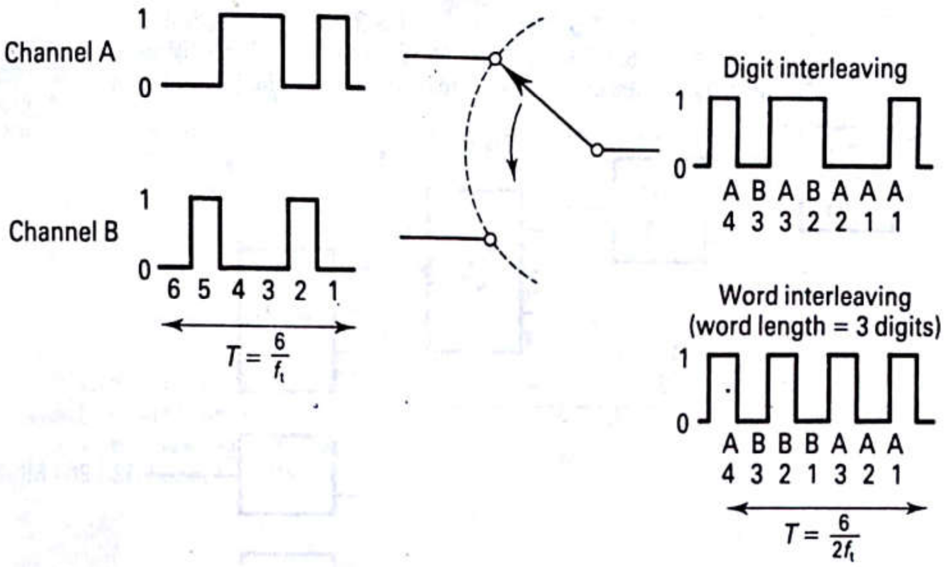


Figure 2.13 Interleaving digital signals. (a) Bit interleaving. (b) Word interleaving.

There are three incompatible sets of standards for plesiochronous digital multiplexing, centred on Europe, North America and Japan. The European standards are based on the 30-channel primary multiplex and the North American and Japanese standards on the 24-channel primary multiplex. The European and North American hierarchies are shown in Figures 2.14 and 2.15.

These systems all use bit interleaving. The frame length is the same as for the primary multiplex, i.e. 125 μs, since this is determined by the basic channel sampling rate of 8 kHz. However, when N tributaries are combined, the number of digits contained in the higher-order frame is greater than N times the number of digits in the

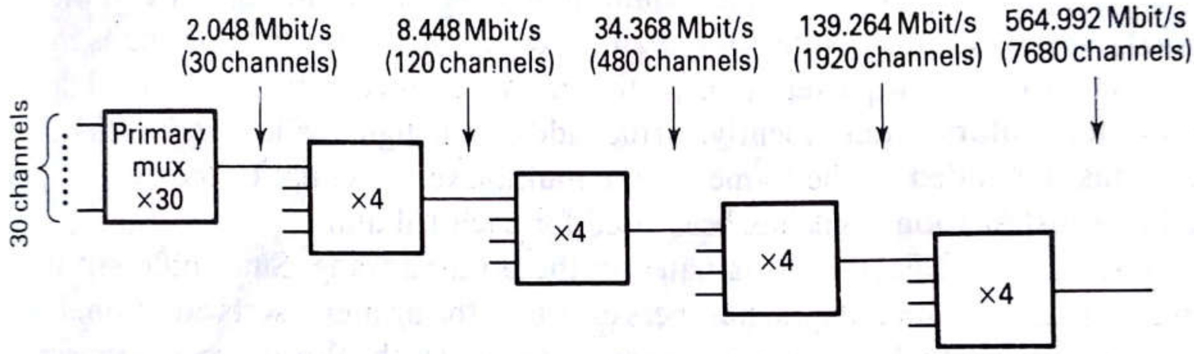


Figure 2.14 European plesiochronous digital hierarchy.

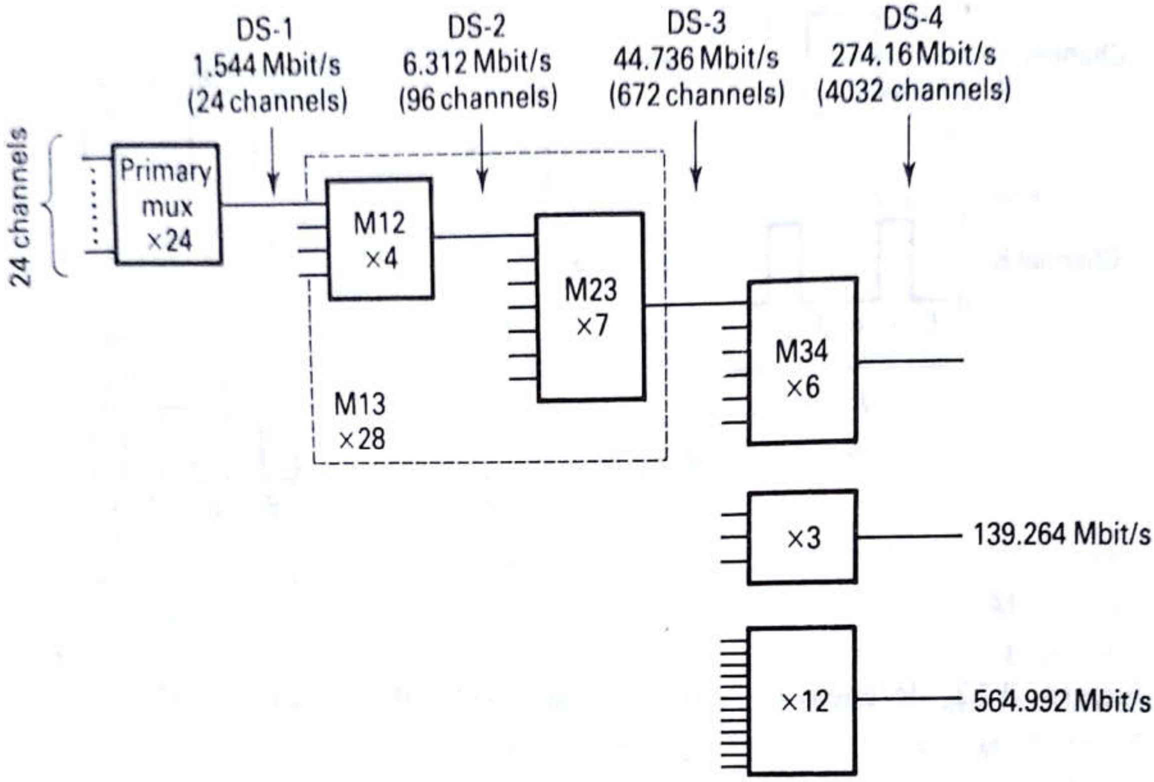


Figure 2.15 North American plesiochronous digital hierarchy.

* Draw backs in PDH:

1. No world wide standard → on Digital formate different regions have different speeds
   ↳ European
   ↳ N. America → Japan
2. No world Std for optical interface. , N/w is impossible at optical
3. Asynchronous multiplexing structure, so scalability totally absent.
4. hand limited management Compatability, no error monitoring & error Control is possible

4) A) four wire system has an overall loss(two wire to two wire)of 1 dB and the balance return loss at each end is 6dB.Find i) The singing point ii) The stability margin iii) the attenuation of talker and listener echo. [5M]

$$L_s = 2(B + 6 - G_4) \text{ dB} \tag{2.6}$$

ubstituting from equation (2.4) into equation (2.6) gives

$$L_s = 2(B + L_2) \text{ dB} \tag{2.7}$$

$M = B + L_2$ dB ....(2.9)

1. From equation (2.7), $L_s = 0$ when $L_2 = -B$
   $\therefore$ Singing point is + 6 dB.
2. From equation (2.9): $M = B + L_2 = 6 + 1 = 7$ dB.
3. $L_t = 2L_2 + B = 2 + 6 = 8$ dB.
   $L_l = 2L_2 + 2B = 2 + 12 = 14$ dB.

B) Explain how the telephone channels are multiplexed by Frequency division multiplexing method. [5M]

Frequency division multiplexing (FDM) means that the total bandwidth available to the system is divided into a series of non overlapping frequency sub-bands that are then assigned to each communicating source and user pair. Note that each transmitter modulates its source's information into a signal that lies in a different frequency sub-band (Transmitter 1 generates a signal in the frequency sub-band between 92.0 MHz and 92.2 MHz, Transmitter 2 generates a signal in the sub-band between 92.2 MHz and 92.4 MHz, and Transmitter 3 generates a signal in the sub-band between 92.4 MHz and 92.6 MHz). The signals are then transmitted across a common channel.

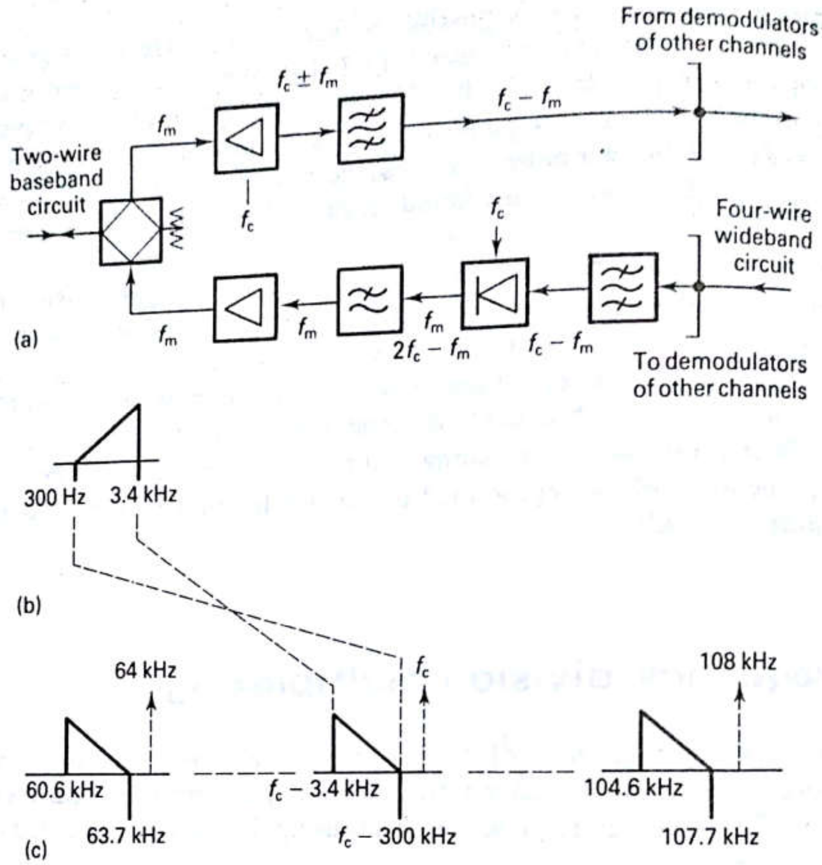


Figure 2.6 Principle of frequency-division multiplexing. (a) Channel translating equipment. (b) Frequency band of baseband signal. (c) Frequency band of wideband signal(CCITT basic group).

At the receiving end of the system, bandpass filters are used to pass the desired signal (the signal lying in the appropriate frequency sub-band) to the appropriate user and to block all the unwanted signals. To ensure that the transmitted signals do not stray outside their assigned subbands, it is also common to place appropriate passband filters at the output stage of each transmitter. It is also appropriate to design an FDM system so that the bandwidth allocated to each sub-band is slightly larger than the bandwidth needed by each source. This extra bandwidth, called a *guardband*,allows systems to use less expensive filters (i.e., filters with fewer poles and therefore less steep rolloffs).

FDM has both advantages and disadvantages relative to TDM. The main advantage is that unlike TDM, FDM is not sensitive to propagation delays. Channel equalization techniques needed for FDM systems are therefore not as complex as those for TDM systems. Disadvantages of FDM include the need for bandpass filters, which are relatively expensive and complicated to construct and design (remember that these filters are usually used in the transmitters as well as the receivers). TDM, on the other hand, uses relatively simple and less costly digital logic circuits. Another disadvantage of FDM is that in many practical communication systems, the power amplifier in the transmitter has nonlinear characteristics (linear amplifiers are more complex to build), and nonlinear amplification leads to the creation of out-of-band spectral components that may interfere with other FDM channels. Thus, it is necessary to use more complex linear amplifiers in FDM systems.

Example—FDM for commercial FM radio

The frequency band from 88 MHz to 108 MHz is reserved over the public airwaves for commercial FM broadcasting. The 88–108 MHz frequency band is divided into 200 kHz subbands. As we saw in Chapter 6, the 200 kHz bandwidth of each sub-band is sufficient for

highquality FM broadcast of music. The stations are identified by the center frequency within their
channel (e.g., 91.5 MHz, 103.7 MHz). This system can provide radio listeners with their choice of up to 100 different radio stations.

5) A) What are the differences between circuit switching and packet switching? [5M]

Message switching	Cirtuit switching
The source and destination do not interact in real time	The source and destination are connected temporarily during data transfer.
Message delivery is on delayed basis if destination node is busy or otherwise unable to accept traffic.	Before path setup delay, may be there due to busy destination node. Once the connection is made, the data transfer takes place with negligible propagation time.
Destination node status is not required before sending message.	Destination node status is necessary before setting up a path for data transfer.
Message switching network normally accepts all traffic but provides longer delivery time because of increased queue length.	A circuit switching network rejects excess traffic, if all the lines are busy.
In message switching network, the transmission links are never idle.	In circuit switching, after path setup, if the users denied service, the line will be idle. Thus, the transmission capacity will be less, if the lines are idle.

Example of Delay System or queuing Example of Lost call system

B) Explain the functions of a Switching system. [5M]

The basic functions that all switching systems must perform are as follows,

1.Attending: The system must be continuously monitoring all lines to detect call requests. The calling signal is sometimes known as a _seize‘ signal because it obtains a resource from the exchange.

2.Information receiving: In addition to receiving calls and clearing signals, the system must receive information from the caller as to the called line (or other service) required. This is called the address signal.

3.Information processing: The system must process the information received in order to determine the actions to be performed and to control these actions. Since both originating and terminating calls are handled differently for different customers, class of service information must be processed in addition to the address information.

4.Busy testing: Having processed the received information to determine the required outgoing circuit, the system must make a busy test to determine whether it is free or already engaged on an other call. If a call is to a customer with a group of lines to PBX(private branch exchanges), or to an outgoing junction route, each line in the group is tested until a free one is found. In an automatic system, busy testing is also required on trunks between switches in the exchange.

5.Interconnection: For a call between two customers, three connections are made in the

following
sequence;
A connection to the calling terminal
A connection to the called terminal
A connection between the two terminals
In the manual system connections, a and b are made at the two ends of the cord circuit and connection c merely joins them in the cord circuit. Many automatic systems also complete connection c by joining a and b at the transmission bridge. However some modern systems release the initial connections a and b and establish connection c over a separate path through the switching network. This is known as *call-back* or *crank-back.* The calling line is called back and the connection to the called line is cranked back.

6.Alerting: Having made the connection, the system sends a signal to alert the called subscriber. E.g. by sending ringing current to a customers telephone.

7.Supervision: After the called terminal has answered, the system continues to monitor the connection in order to be able to clear it down when the call has ended. When a charge for the call is made by metering, the supervising circuit sends pulses over the private wire to operate a meter in the line circuit of the calling customer. When automatic ticketing is employed, the system must send the number of the caller to the supervisory circuit when the connection is setup. This process is called *calling line identification (CLI)* or *automatic number identification (ANI).* In SPC system, the data for call charging can be generated by a central processor as it sets up and clears down connections.

8.Information sending: If the called customer‘s line is located on another exchange, the additional function of information sending is required. The originating exchange must signal the required address to the terminating exchange (and possibly to intermediate exchanges if the call is to be routed through them).

6) With a neat diagrams explain how Two stage crossbar link network inter connect 100 incoming and 100 outgoing trunks using switches of size 10X10, also explain operation of 3X3 crossbar switch. [5+5 = 10M]

3.9 Crossbar systems

Strowger switches require regular maintenance. The banks need cleaning, mechanisms need lubrication and adjustment and wipers and cords wear out. This disadvantage led to the development of several other forms of switch.[1,9] One idea was to replace the manually operated switch of Figure 3.2 by a matrix of telephone relays, with their contacts multipled together horizontally and vertically as shown in Figure 3.11. Since a switch with N inlets and N outlets requires N^2 relays for its crosspoints, this was uneconomic except for small private exchanges requiring small switches.

A more economic solution was provided by the invention of the *crossbar switch* by G. A. Betulander[1] in 1917. This is shown in Figure 3.12. The crossbar switch retains a set of contacts at each crosspoint, but these are operated through horizontal and vertical bars by magnets at the sides of the switch. Thus, a switch with N inlets and N outlets only needs $2N$ operating magnets and armatures, instead of N^2. The magnets which operate the horizontal bars are called *select magnets* and those operating the vertical bars are called *hold magnets* or *bridge magnets*.

The mechanism of a crossbar switch is shown in more detail in Figure 3.13. Operation of a select magnet tilts one of the horizontal bars up or down. This causes flexible fingers to engage with the contact assemblies of one row of crosspoints and provides the link which was missing from their operating mechanisms. One of the bridge magnets is then operated and this closes the contacts of the crosspoint at the coordinates corresponding to the horizontal and vertical magnets. The select magnet is then released, but the finger remains trapped and the crosspoint contacts remain closed for as long as the bridge magnet is energized. Current flows in this magnet for as long as

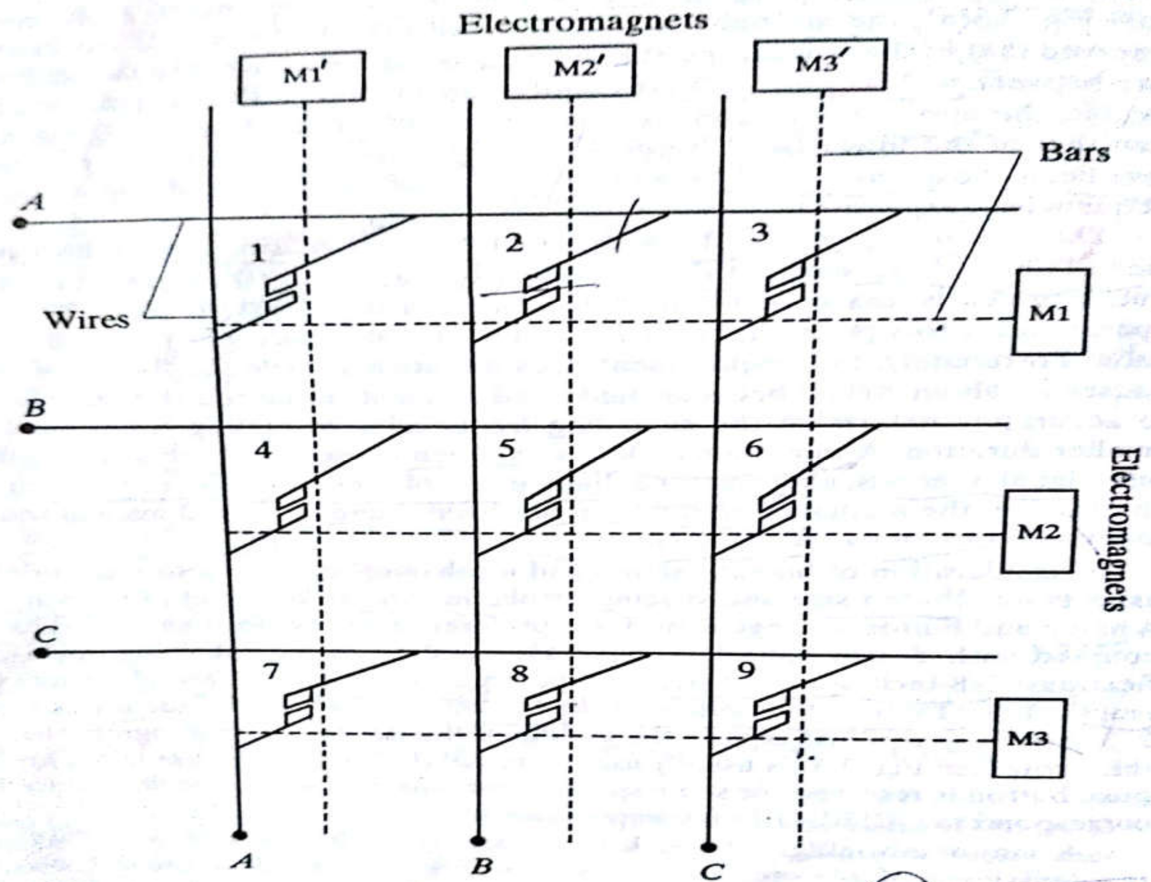


Fig. 3.6 3 × 3 crossbar switching.

the P wire is at earth potential. This persists until a 'clear' signal causes the earth to be removed at the end of a call.

Strowger selectors perform counting and searching. However, the crossbar switch has no 'intelligence'. Something external to the switch must decide which magnets to operate. This is called a *marker*. Since it takes less than a second to operate the switch, a marker can control many switches and serve many registers, as shown in

Unlike the two-motion selector, a crossbar switch can make more than one connection at a time. It can make as many connections as it has vertical bars. Thus, it can be used as if it were a group of uniselectors instead of a single two-motion selector. For example, a switch of size 10 × 10 can make up to ten simultaneous connections between ten incoming trunks and ten outgoing trunks.

In order to produce larger switches, a two-stage link system of primary and secondary switches is used, as shown in Figure 3.15. This is called a *link frame*. The figure shows twenty switches of size 10 × 10 used to connect 100 incoming trunks to 100 outgoing trunks. There is one link fom each primary switch to each secondary switch and these links are arranged systematically. The number of an outlet on a primary switch corresponds to the number of the secondary switch to which its link goes and the number of an inlet on a secondary switch corresponds to the number of the primary switch from which its link comes. For example, link 23 connects outlet 3 of primary switch 2 to inlet 2 of secondary switch 3.

When a marker is instructed to set up a connection from a given incoming trunk to a given outgoing trunk, this also defines the link to be used and the select and bridge magnets to be operated to make the connection. The marker does not make the

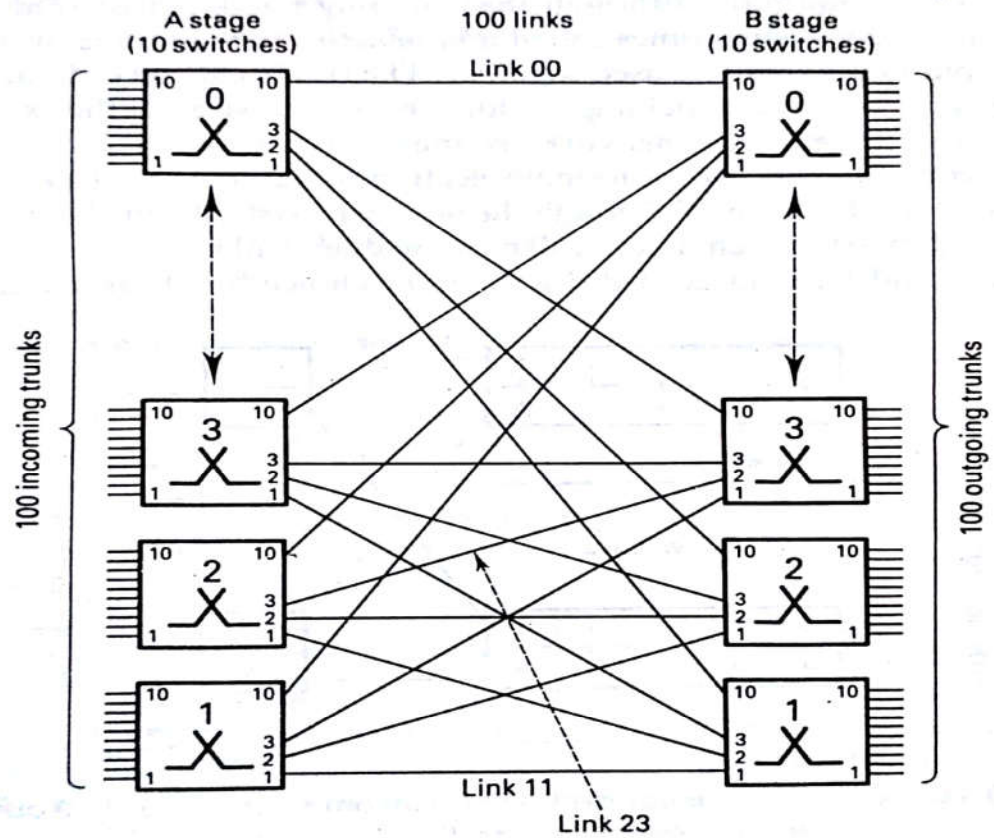


Figure 3.15 Two-stage link network (using switches of size 10 × 10 to interconnect 100 incoming and 100 outgoing trunks).

Manohar