UNIT IV

TRANSMISSION NETWORKS

Subscriber loop – DSL – ADSL – FDM and TDM – PCM multiplex group – PDSH, SDH / SONET – cross talk – line equalizations – adaptive equalizers – single stage network – two, three, four stage networks – network synchronization.

4.1 Subscriber loop

Subscriber loop is the cables that connect the telephone handsets or other devices to the local switching office or end office. It is also called as local loop. Every subscriber has own pair of wires that is used to connect the subscriber with the local switching office.

Two types of cables are used as subscriber loop. They are

1. Twisted pair local loop
2. Fiber cable

4.1.1 Twisted pair local loop

Twisted pair local loop is an excellent transmission medium for analog voice signals. But it is limited to low frequency audio signals.

4.1.2 Fiber cable

The introduction of fiber cable needs a device at subscriber premises to convert electrical energy into light energy and this is the additional cost to the customer. But for high speed data transmission, switched cable TV, videophone, teleconferencing, the fiber optic local loop has become essential.

4.1.3 Limiting Factors of Subscriber Loop Design

While designing a Subscriber loop we have to consider two limiting factors. They are

i) Attenuation
ii) Voltage drop

The **attenuation** refers to the energy loss in the line, measured in decibels. If the length of the loop increases the attenuation also increases.

The second limiting factor is **voltage drop**. If the battery voltage is kept constant with increase in length, the effectiveness of the signalling and conversation will be limited. This is due to the reduction of current while it travels along the cable.

4.2 DSL (Digital Subscriber Line)

Digital information can also be transmitted over analog line. **The digital subscriber line (DSL) is the technology used between a customer premises and telephone companies, enabling more bandwidth over the already installed copper cabling that user have traditionally had.**

For years it has been believed that the upper limit for transmitting data on analog phone lines was 56kb/s. This limit is set using the maximum possible bandwidth and no compression. The reason for this limit is that POTS or Plain Old Telephone Service uses the lower 4 KHz only. The limit imposed by the POTS lines does not take advantage of all the bandwidth available on copper, which is on the order of 1 MHz. The DSL technologies take advantage of this difference and use the upper frequencies for data services. Previously this was not possible because of the interference that the data services would cause in the POTS band. Advances in digital signal processing have eliminated the near-end crosstalk that results from the use of the upper bandwidth for data. The new DSP technologies allow data and POTS to be transmitted on the same set of copper wires without interfering with each other.

A DSL requires two modems, one at the phone companies end and one at the subscribers end. The use of the term modem is not entirely correct because technically a DSL modem does not do modulation/demodulation as in a modem that uses the normal telephone network. DSL's also having the added benefit of transmitting telephone services on the same set of wire as data services. XDSL is a generic abbreviation for the many variations of digital subscriber line technology where x stands for specific type. Thus XDSL is a technology backed by telephone companies to provide next generation high bandwidth services to the home and business using the existing telephone cabling infrastructure. This technology accomplishes high speed delivery of data, voice, video and multimedia. Typical speed of XDSL technology starts at about 128 kbps and goes up to 8.192 Mbps for most home user and to 50 Mbps for some installations with high capacity networks. The speed variations depend on the equipment used, distance involved, cabling quality, encoding techniques and frequency spectrum available.

4.3 ADSL- Asymmetric Digital Subscriber Line

In transmission, the bit rate may differ for upstream and downstream. The upstream is transfer of information from subscriber to local telephone exchange and the downstream is transfer of information from local telephone exchange to the customer premises. If the transfer of information in bit rate is same for upstream as well as downstream, it is referred as symmetric. **If the bit rate varies for upstream and downstream, it is referred as asymmetric or rate adaptive or uneven.** As the subscriber usually want to receive high volume files quickly, but usually have small files to send, XDSL one mostly asymmetric, that is downstream bit rates are higher than upstream.

Asymmetric Digital Subscriber Line (ADSL) is the most popular form of XDSL technology. Its upstream and downstream bandwidth is asymmetric or uneven. ADSL can transmit up to 6 Mbps to a subscriber, and as much as 832 kbps or more in both directions. In practice, the bandwidth of downstream is high and is the high speed path.

4.3.1 ADSL Frequency spectrum

ADSL divides the bandwidth of a twisted pair cable into three bands. They are

i) POTS
ii) Upstream
iii) Downstream

Figure 4.1 ADSL Bandwith

The frequency spectrum from 26 kHz to 138 kHz is used for upstream transmission, and the frequency spectrum from 1.38 kHz to 1.1 MHz is used for downstream transmission. The lower 4 kHz channel is separated by an analog circuit and used in POTS. The frequency spectrum above 26 kHz is divided into 249 independent sub channels, each containing 4.3 kHz bandwidth. 25 channels are used for upstream transmissions and 224 channels are used for downstream transmissions.

4.3.2 Advantages of ADSL

1. It provides a simple, affordable mechanism to get more bandwidth to end users, both residential and small to medium businesses.

2. The high speed downstream is increasingly important for internet access, remote access to corporate server, integrated voice/data access and transparent LAN interconnection.

3. It enables carrier to offer value added, high speed networking services.

4.4 MULTIPLEXING

When the bandwidth of a medium is greater than individual signals to be transmitted through the channel, a medium can be shared by more than one channel of signals. The process of making the most effective use of the available channel capacity is called Multiplexing. For efficiency, the channel capacity can be shared among a number of communicating stations just like a large water pipe can carry water to several separate houses at once. Most common use of multiplexing is in long-haul communication using coaxial cable, microwave and optical fibre.

In telecommunication, the multiplexing means the use of one telecommunication line to handle several channels of voice or data. The best example of multiplexing is our TV cable. We select a particular channel using remote control from the cable, which carries many channels.

The primary use of multiplexing is to save communication line costs.

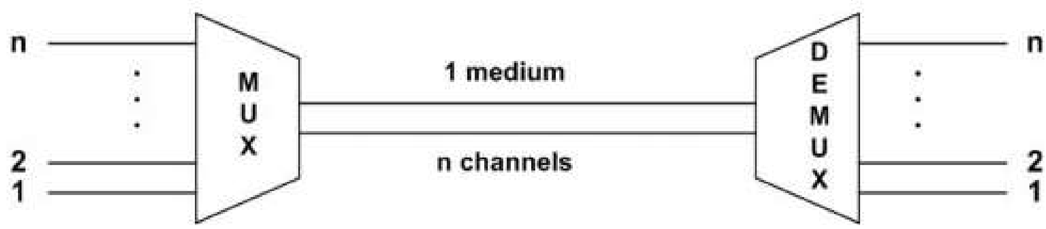

Figure 4.2 Basic concepts of Multiplexing

The above Figure 4.2 depicts the functioning of multiplexing functions in general. The multiplexer is connected to the demultiplexer by a single data link. The multiplexer combines (multiplexes) data from these ‘n’ input lines and transmits them through the high capacity data link, which is being demultiplexed at the other end and is delivered to the appropriate output lines. Thus, Multiplexing can also be defined as a technique that allows simultaneous transmission of multiple signals across a single data link.

Multiplexing techniques can be categorized into the following three types:

- Frequency-division multiplexing (FDM)
- Time-division Multiplexing (TDM)
- Space division multiplexing (SDM)

4.4.1 Frequency-division multiplexing (FDM)

FDM systems divide the available BW of the transmission medium into a number of narrow band or sub channels. The channels are sent over a common path by modulating each channel to different carrier frequency.

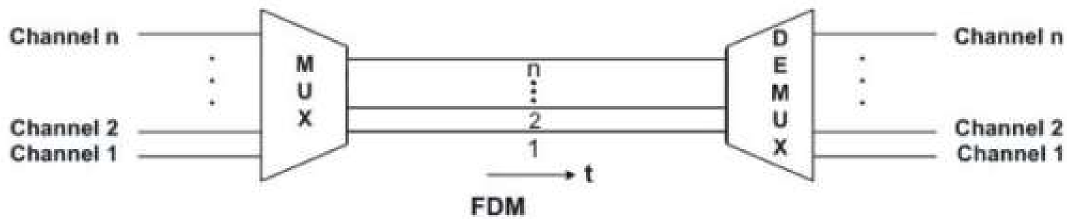

Figure 4.3 Basic Concepts of FDM

Basic approach is to divide the available bandwidth of a single physical medium into a number of smaller, independent frequency channels. Using modulation, independent message signals are translated into different frequency bands. All the modulated signals are combined in a linear summing circuit to form a composite signal for transmission. The carriers used to modulate the individual message signals are called sub-carriers, shown as f_1, $f_2 \ldots f_n$ in Fig.

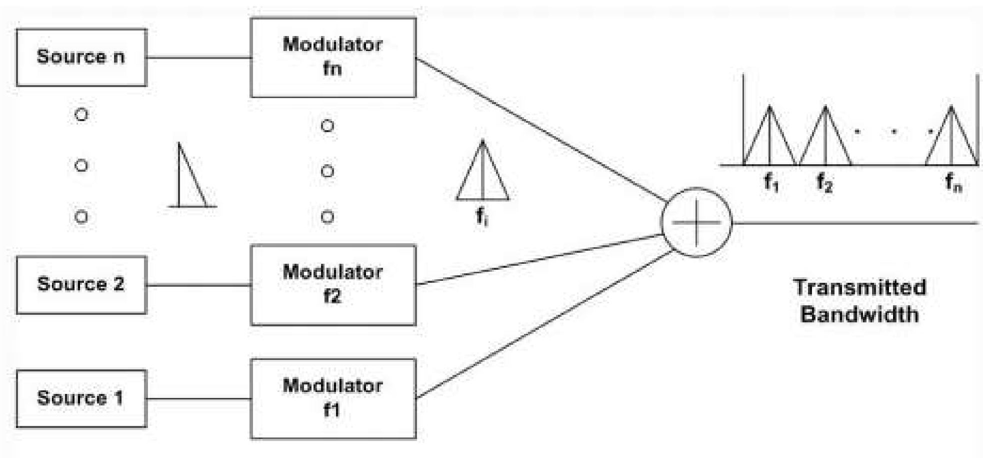

Figure 4.4 FDM multiplexing process

At the receiving end the signal is applied to a bank of band-pass filters, which separates individual frequency channels. The band pass filter outputs are then demodulated and distributed to different output channels as shown in Fig.

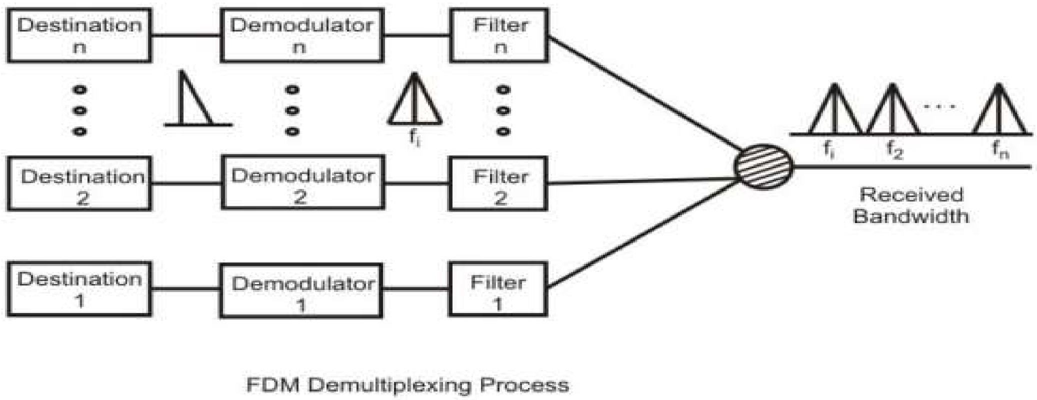

Figure 4.5 FDM Demultiplexing

If the channels are very close to one other, it leads to inter-channel cross talk. Channels must be separated by strips of unused bandwidth to prevent inter-channel cross talk. These unused channels between each successive channel are known as guard bands as shown in Fig

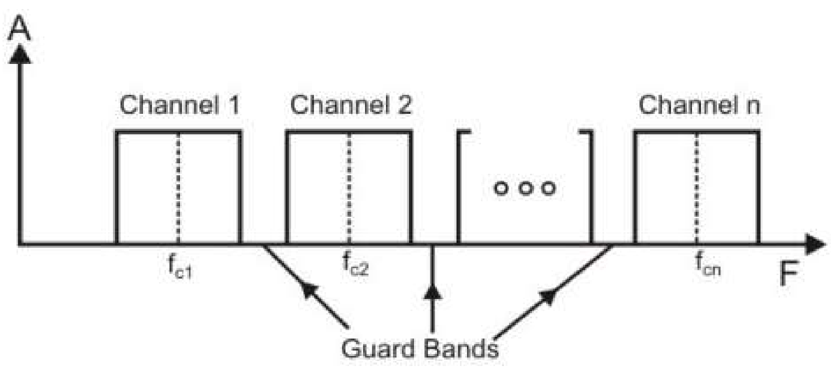

Figure 4.6 Use of Guard Bands

4.4.2 Time-Division Multiplexing (TDM)

In frequency division multiplexing, all signals operate at the same time with different frequencies, but in Time-division multiplexing all signals operate with same frequency at different times. This is a base band transmission system, where an electronic commutator sequentially samples all data source and combines them to form a composite base band signal, which travels through the media and is being demultiplexed into appropriate independent message signals by the corresponding commutator at the receiving end. The multiplexing operation is shown in Fig

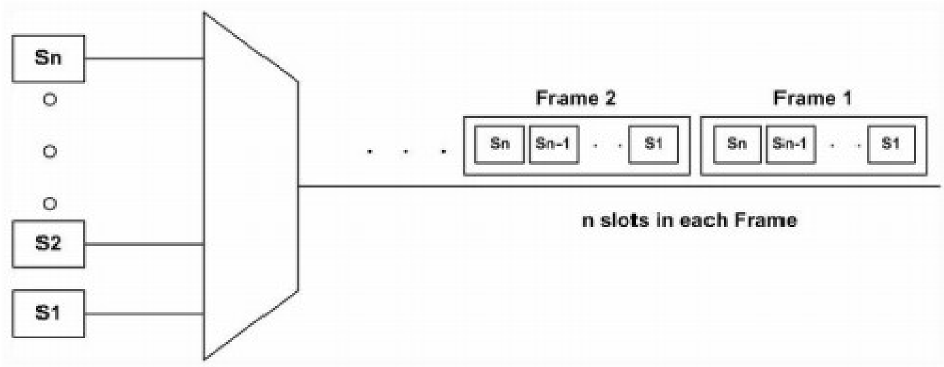

Figure 4.7 Time division multiplexing operation

As shown in the Fig, the composite signal has some dead space between the successive sampled pulses, which is essential to prevent interchannel cross talks. Along with the sampled pulses, one synchronizing pulse is sent in each cycle. These data pulses along with the control information form a frame. Each of these frames contain a cycle of time slots and in each frame, one or more slots are dedicated to each data source. The maximum bandwidth (data rate) of a TDM system should be at least equal to the same data rate of the sources.

Synchronous TDM is called synchronous mainly because each time slot is preassigned to a fixed source. The time slots are transmitted irrespective of whether the sources have any data to send or not. Hence, for the sake of simplicity of implementation, channel capacity is wasted. Although fixed assignment is used TDM, devices can handle sources of different data rates. This is done by assigning fewer slots per cycle to the slower input devices than the faster devices. Both multiplexing and demultiplexing operation for synchronous TDM are shown in Fig

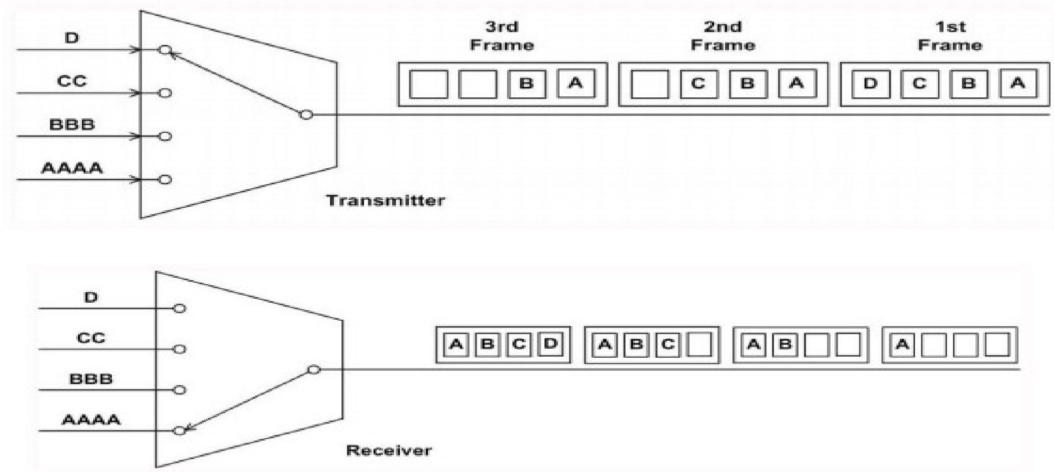

Figure 4.8 Multiplexing and demultiplexing in synchronous TDM

5. PCM multiplex group

Analog speech signals are converted to digital signals by using pulse code modulation (PCM), and multiplexed onto a common bearer by using TDM. In this way, telephone channels are combined to form an assembly of 24 or 30 channels. This is known as **primary multiplex group**. The primary multiplex group is also used as a building block for assembling larger numbers of channels in **higher - order multiplex systems**.

The digit stream on the transmission path is arranged in frames of 125μs duration (i.e. the interval corresponding to 8 kHz sampling). Each frame contains one coded speech sample from each channel, together with digits for signaling and synchronization.

Two frame structures are widely used:

1. European 30 - channel system
2. 24 - channel system (known as DS 1) used in North - America and Japan.

Both use 8 kHz sampling and 8 - bit samples. However, the 30 - channel system uses A - law companding and the 24 - channel system uses -μ law companding.

5.1 European 30 - channel system

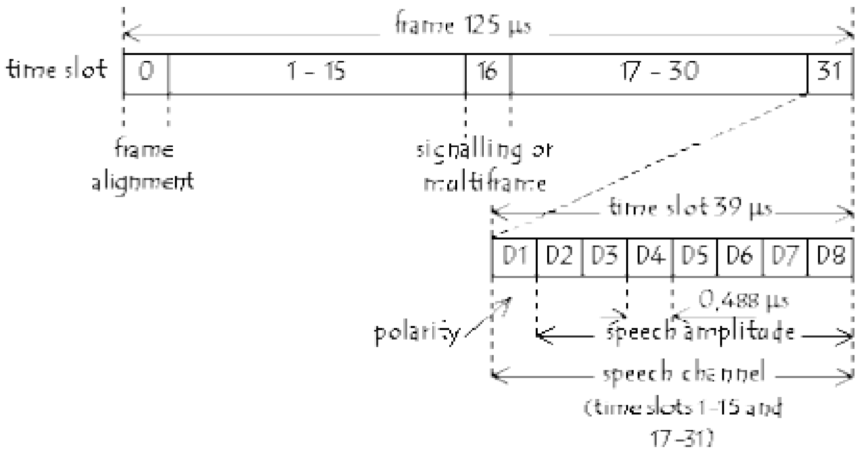

Figure 4.9 30 channel frame format

The 30 - channel frame is shown in Figure 1.23. It is divided into 32 time slots each of eight digits, so the total digit rate is 8 x 8x 32 kbits/s = 2,048 Mb/s.

Time slot 0 is allotted to the **frame - alignment word**. **Time slots 1 - 15 and 17 - 31** are each allotted to a **speech channel**. **Time slot 16** is allotted to **signalling**.

5.2 DS1 24 channel system

DS0 is a 64-kbps signal that makes up the basis for the DS1. Twenty four DS0s combined to produce DS1. Fig shows the DS-1 frame format. Incoming analog signals were time division multiplexed and digitized for transmission. Each individual TDM channel are assigned 8 bits per time slot. Each frame is made of 24 × 8 bits = 192 bits plus one additional bit added to each frame to identity the frame boundaries. Thus each frame contains 193 bits. The frame interval is 125 µ sec. Hence the basic T1 line rate is 1.544 Mbps.

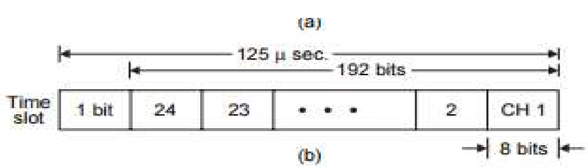

Figure 4.10 DS-1 frame Format

6. The plesiochronous digital hierarchy (PDH)

In PDH, inputs entering into the multiplexer are not synchronous. This is due to that the inputs are originate from different crystal oscillators. They are said to be **plesiochronous**.

The first-generation of higher-order digital multiplex systems was designed for this situation. It forms the plesiochronous digital hierarchy (PDH).

There are three sets of standards for plesiochronous digital multiplexing, centred on Europe, North America and Japan. These systems all use bit interleaving. In bit interleaving, one bit is taken from each tributary in turn. The frame length is the same as for primary multiplex, i.e. 125 µs, since this is determined by the basic channel-sampling rate of 8 kHz. When *N* inputs are combined, the digit rate of the higher-order frame is more than *N* times the digit rate of the input frames. This is because it is necessary to add extra "overhead" digits for two reasons.

The first reason is frame alignment. A higher-order demultiplexer must recognize the start of each frame in order to route subsequent received digits to the correct outgoing tributaries, just as a primary demultiplexer must route received digits to the correct outgoing channels.

The second reason for adding extra digits to the frame is to perform the process known as **justification**. This process is to enable the multiplexer and demultiplexer to maintain correct operation, although the input signals of the tributaries entering the multiplexer may drift relative to each other. In **positive justification** the transmitted digit rate per tributary is slightly higher than the nominal input rate. If an input tributary is slower, a dummy (fictive), i.e. a justification digit, is added to maintain the correct output digit rate. If the input tributary speeds up, no justification digit is added. The demultiplexer must remove the justification digits, in order to send the correct sequence of signal digits to the output tributary. In **negative justification**, instead of dummy digits being inserted when the digit rate of a tributary is too slow, a data digit is occasionally removed when a tributary is too fast and is transmitted in a spare time slot. Another option is to use both positive / zero / negative justification. The European PDH uses only positive justification.

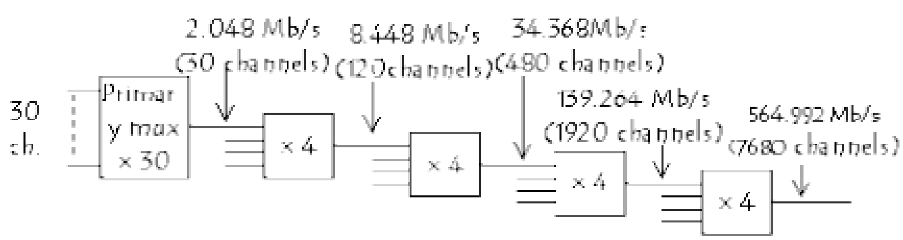

Figure 4.11 European plesiochronous digital hierarchy

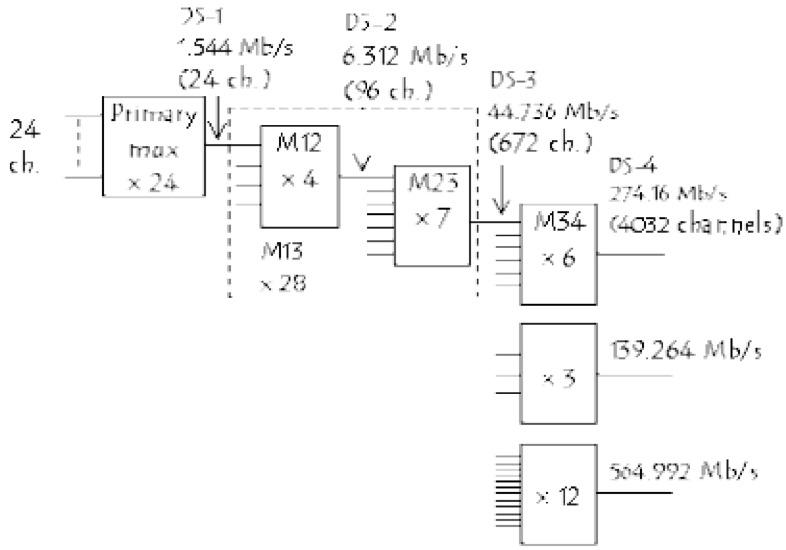

Figure 4.12 North America plesiochronous digital hierarchy

7. The synchronous digital hierarchy (SDH)

Networks are becoming fully digital, operating synchronously, using high-capacity optical-fibre transmission systems and time-division switching. It is advantageous for the multiplexers used in these networks to be compatible with the switches used at the network nodes, i.e. they should be synchronous rather plesiochronous.

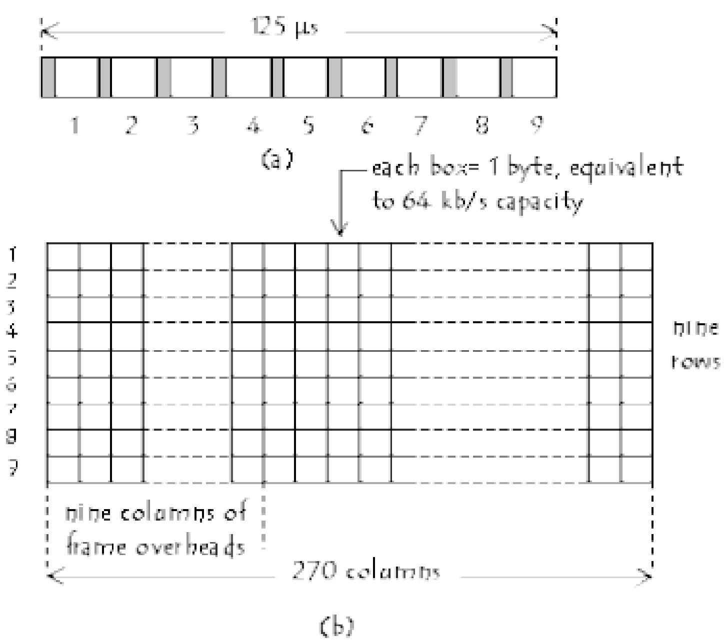

Figure 4.13 – SDH: frame structure of STM-1
(a) On line frame structure
(b) Frame structure shown in rows and columns

The basic SDH signal, called the **synchronous transport module at level 1** (STM-1) is shown in Figure

This has nine equal segments, with nine "overhead" bytes at the start of each. The remaining bytes (261) contain a mixture of traffic and overheads, depending on the type of traffic carried. The total length is (9+ 261)*9 = 2430 bytes, thus the overall bit rate is 2430 x 8 bits/125 μs = 155,520 kb/s, which is usually called "155 Mb/s"**.**

This frame is usually represented as nine rows and 270 columns of 8 - bit bytes, as shown in Figure. The first nine columns are for **section overheads**, such as frame alignment, error monitoring and data. The remaining 261 columns comprise the **payload**, into which a variety of signals can be mapped. Each column contain nine bytes (one from each row), with each byte having 64 kb/s capacity. Three columns (i.e. 27 bytes) can hold a 1.5 Mb/s PCM signal, with 24 channels and some overheads. Four columns (i.e. 36 bytes) can hold a 2 Mb/s PCM signal with 32 time slots. The STM-1 frame can also hold payloads at the European rates of 8, 34 and 140 Mb/s and the North America rates of 6, 45 and 140 Mb/s.

Because the SDH provides interfaces for network - management messages in a standard format, it can lead to a managed transmission - bearer network in which transport capacity can be allocated flexibly to various services. The network can be reconfigured under software control from remote terminals.

8. SONET

- Synchronous Optical Network (SONET) is a high speed optical carrier using fiber-optic cable.
- SONET is intended to provide a specification for taking advantage of the high speed digital transmission capability of optical fiber.
- It uses the SDH standard.

8.1 Characteristics or advantages of SONET

- SONET uses byte multiplexing at all levels.
- As SONET is a synchronous network, a single clock handles the timing of transmissions and equipment across the entire network.
- Establishes a standard multiplexing format using some number of STS-1 signals as building blocks.
- SONET/SDH contains recommendations for the standardization of fiber optic transmission system (FOTS) equipment sold by different manufacturers.

- The flexibility of SONET accomodates applications such as ISDN with variety of
- transmission rates.
- SONET provides extensive operation, administrations, maintenance and provisioning (OAM & P) functions for network managers.
- It has enhanced network reliability, availability and universal connectivity.

8.2 SONET Componets

Figure shows SONET's components. SONET transmission relies on three basic devices.

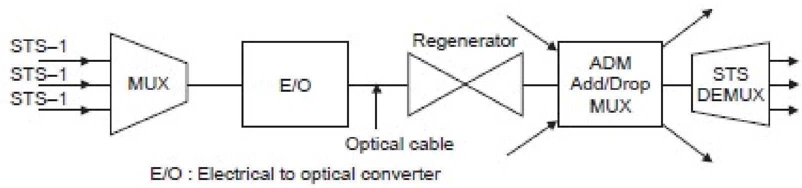

Figure 4.14 **SONET Components**

STS multiplexer/demultiplexer

The function of an STS MUX is to multiplex electrical input signals to a higher data rate and then convert the results to optical signals. The STS demultiplexers convert and demultiplex optical signals to electrical signals for the users.

Add/Drop multiplexer (ADM)

It adds or removes the lower rate signals from or into high rate multiplexed signals. It does this extraction or insertion and redirects it without demultiplexing the entire signal. These multiplexers use the header information such as addresses and pointers to identify individual streams

Regenerator

The regenerator performs the functions of a repeater. It receives the optical signal and regenerates it. If the optical cable is longer than standard, the regenerator will be used to receive the optical signal and then to regenerate the optical signal.

9. Cross Talk

Any sound heard in the telephone receiver associated with one channel resulting from the transmission of a signal over another channel is called Cross talk.

9.1 Major sources of cross talk

- Coupling between wire pairs in cable

- Inadequate filtering or carrier offsets in older frequency division multiplexing (FDM) equipments
- The effects of non-linear components on FDM signals.

Cross talk is one of the most disturbing and undesirable imperfections that can occur in a telephone network. In fact cross talk is more noticeable during speech pauses, where the power level of the desired signal is zero.The basic forms of cross talk concern to telecommunication engineers are near end cross talk (NEXT) and far end cross talk (FEXT).

9.2 NEXT (Near end Cross talk)

NEXT occurs near the transmitter and creates distortions that affect the signal on adjacent receive pairs. This type of noise can be generated when a transmission line carrying a strong signal is coupled with a transmission line carrying a weak signal. NEXT should be measured at both ends. NEXT is more troublesome because of a large difference in power levels between the transmitted and receive signals. Twisted wire pairs reduce this type of cross talk.

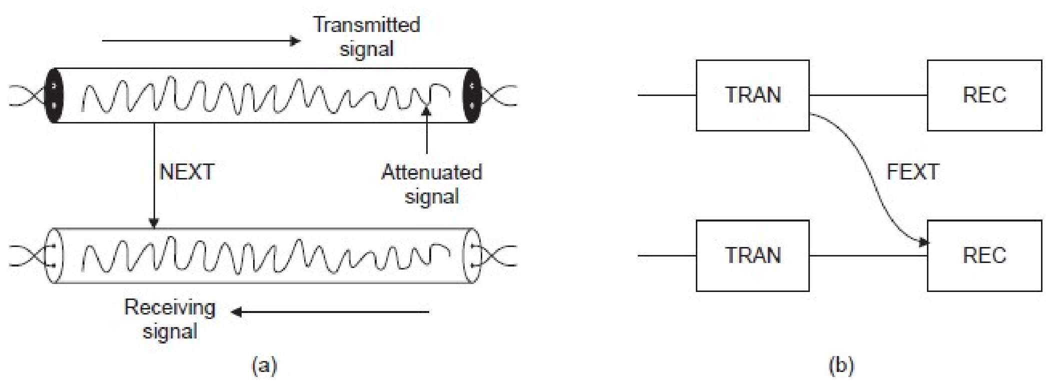

Figure 4.15 **(a) NEXT and (b) FEXT**

9.3 FEXT (Far end cross talk)

FEXT is a measure of the cross talk that exists at the receiver end of the cable. FEXT refers to unwanted coupling into a received signal from a transmitter at a distant location. This noise is relevant on networks that transmit on multiple pairs in the same direction in the same time.

10. EQUALIZATION

ISI has been recognized as the major obstacle to high speed data transmission over mobile radio channels. The goal of equalizers is to eliminate inter symbol interference (ISI) and the additive noise as much as possible.

Inter symbol interference of a transmitted pulse due to the dispersive nature of the channel, which results in overlap of adjacent pulses.

10.1 Categories of Equalization

Equalizers are used to overcome the negative effects of the channel. In general, equalization is partitioned into two broad categories

- **Maximum likelihood sequence estimation (MLSE)** which entails making measurement of channel impulse response and then providing a means for adjusting the receiver to the transmission environment.
- **Equalization with filters**, uses filters to compensate the distorted pulses. The general channel and equalizer pair is shown in Figure.

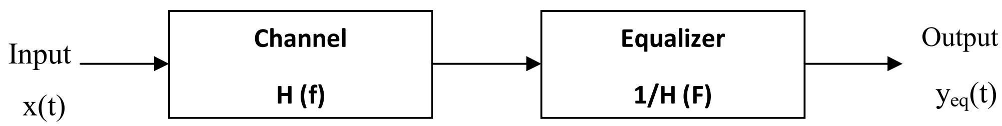

Figure 4.16 **Equalization with filters**

Depending on the time nature these types of equalizers can be grouped as preset or adaptive equalizers.

- **Preset equalizers** assume that the channel is time invariant and try to find H(f) and design equalizer depending on H(f).
- **Adaptive equalizers** assume channel is time varying channel and try to design equalizer filter whose filter coefficients are varying in time according to the change of channel, and try to eliminate ISI and additive noise at each time

10.2 Block diagram of Adaptive equalizer

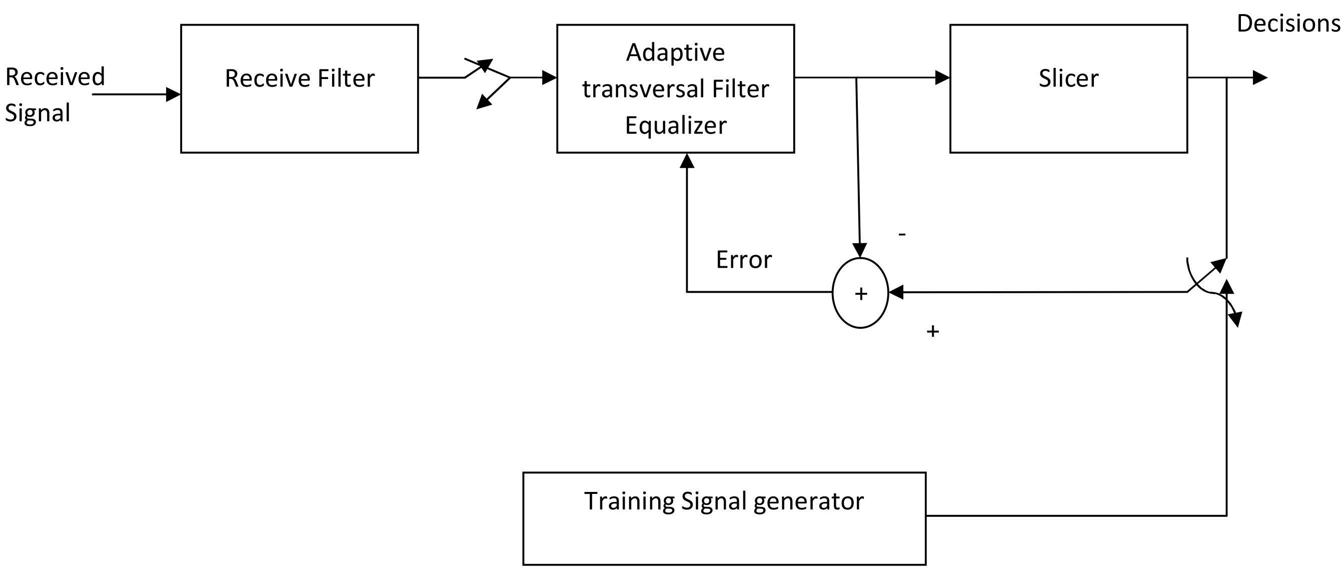

Figure 4.17 **Block diagram of Adaptive equalizer**

Decisions

10.3 Working principles of adaptive equalizers

- The received signal is applied to receive filter. In here, receive filter is not matched filter. Because we do not know the channel impulse response. The receive filter in here is just a low-pass filter that rejects all out of band noise.
- The output of the receiver filter is sampled at the symbol rate or twice the symbol rate.
- Sampled signal is applied to adaptive transversal filter equalizer. Transversal filters are actually FIR discrete time filters.
- The object is to adapt the coefficients to minimize the noise and intersymbol interference (depending on the type of equalizer) at the output.
- The adaptation of the equalizer is driven by an error signal.

10.4 Operation mode of adaptive equalizers

There are two modes that adaptive equalizers work;

1. **Decision Directed Mode**: This means that the receiver decisions are used to generate the error signal. Decision directed equalizer adjustment is effective in tracking slow variations in the channel response. However, this approach is not effective during initial acquisition .

2. **Training Mode**: To make equalizer suitable in the initial acquisition duration, a training signal is needed. In this mode of operation, the transmitter generates a data symbol sequence known to the receiver. The receiver therefore, substitutes this known training signal in place of the slicer output. Once an agreed time has elapsed, the slicer output is substituted and the actual data transmission begins.

The training sequence is designed to permit an equalizer at the receiver to acquire the proper filter coefficients in the worst possible channel conditions. Therefore filter coefficients are near their optimal values for reception of user data. An adaptive equalizer at the receiver uses a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate for the channel.

11. Switching Network

The basic function of an exchange is making a connection between calling subscriber and called subscriber.Based on inlet and outlets switching networks can be classified as

1. Single stage network
2. Two stage network

3. Three stage Network
4. Four Stage network

11.1 Single stage Network

A single stage switch block for ‘n’ subscribers basically consists of a n x n matrix as shown in fig. Each of the n-subscribers is to be connected to the remaining (n-1) subscribers using cross points.

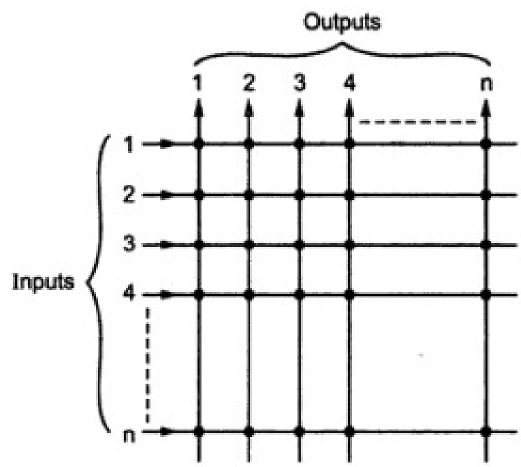

Figure 4.18 **Single stage network**

(Single stage Networks is same as Cross bar switching (Refer crossbar switching))

The following figure shows a single stage networks having M inlets and N outlets, consisting in a matrix of cross points.

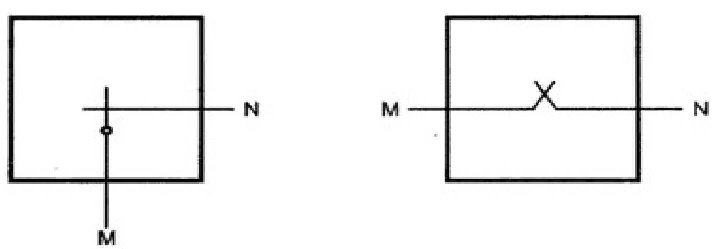

Figure 4.19 **: Electromechanical switches**

The number of simultaneous connections that can be made is either M (if M<N) or N (if N<M). The switch contains MN cross points. If M=N, the number of cross point is

$$\mathbf{C_1=N^2}$$

For the triangular cross point matrix, the redundant cross points are eliminated .No of croos point for triangular cross point matrix is

$$C_1=1/2\ N\ (N-1)$$

11.2 Two stage Networks

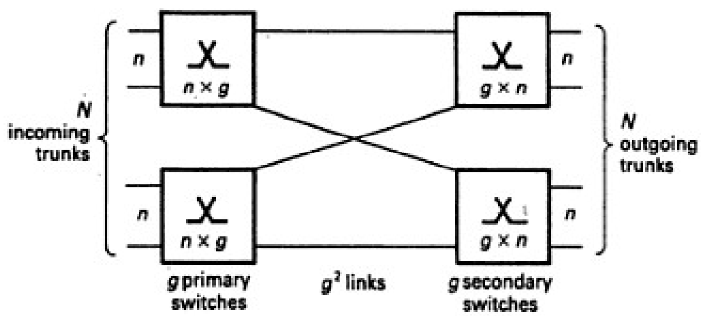

Figure 4.20 Two-stage switching network.

Two-stage network has N incoming and N-outgoing trunks and contains primary switches having n inlets and secondary switches having n outlets, then no of primary switches (g)= no of secondary switches=no of outlets per primary switch=no of inlets per secondary switch, where

$$g= N/n$$

The no. of cross point per primary switch = no. of cross points per secondary switch=gn=N.

The total no of cross point in the network= No of switches x Cross points per switch

$$C_2=2gN=2N^2/n$$

Since there is one link from each primary switch to each secondary switch , the no. of links is equal to the no.of primary switches x no. of secondary switches.

$$\text{No. of links}=g^2= (N/n)^2$$

The number of cross points thus varies as $1/n$, but the number of link varies as $1/n^2$. If n is made very large to reduce the number of cross points, there will be too few links to carry the traffic. Let the number of links be equal to the number of incoming and outgoing trunks then

$$g^2=N$$

Substitute this in the above result gives

$$\mathbf{N=\sqrt{n}}$$

Then the total no of cross points is

$$\mathbf{C_2= 2N^{3/2}}$$

11.3 Three Stage networks

In three stage networks, one link from each primary switches to secondary switch to each tertiary switch. A connection from inlet on a primary switch to a selected outlet on a tertiary switch may be made via secondary switch, unless its link to the primary switch or link to the secondary switch is busy.

Three stage network has N incoming trunks and N outgoing trunks and has primary switches with n inlets and tertiary switches with n outlets, then

No of primary switches (g_1)= No of tertiary switches(g_3)= N/n

The secondary switches have N/n inlets and N/n outlets.

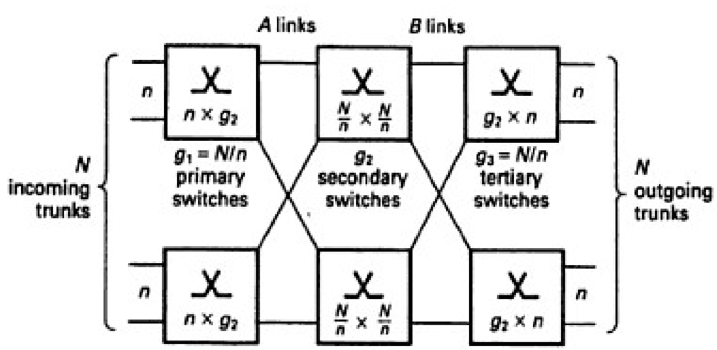

Fig 4.21 **- Three stage Networks**

If the number of primary-secondary links and secondary-tertiary links are each N, then the number of secondary switches is

g_2= N ÷ (N/n)=n

=no.of outlets per primary switch=no.of inlets per primary switch.

No.of crosspoints in primary stage= n^2 (N/n)=nN

No.of crosspoints in secondary stage= $n (N/n)^2 = N^2/n$

No.of crosspoints in tertiary stage= $n^2 (N/n) = nN$

The total no of cross point is

$$C_3 = N (2n+N/n)$$

Minimum no of crosspoint can be detrmined by differentiating the above equation with respect to n and equating to zero.

$$n = \sqrt{N/2}$$

and then

$$C3 = 2\sqrt{2}N^{3/2}$$

$$= \sqrt{2}\, C_2$$

$$= 2^{3/2} N^{-1/2} C_1$$

If the three-stage concentrator has M incoming trunks and N outgoing trunks (M>N), its primary switches have m inlets and its tertiary switches each have n outlets then

No of primary switches = M/m

No of tertiary switches= N/n

If there are g_2 secondary switches, then

Cross points per primary switch= mg_2

Cross points per secondary switch= M/m x N/n

Cross points per tertiary switch= $g_2 n$

The total number of cross pints is

$$C_3 = M/m \times mg_2 + g_2 \times M/m\ N/n + N/n \times g_2 n$$

$$= g_2 [M+N+M/m \times N/n]$$

Since m>n , let no. of primary links=N

$$N= g_2 M/m = g2 N/n$$

$$\text{Hence } g_2=n \text{ and } m= n M/N$$

Substituting it in C_3 equation

$$C_3= (M+N)n+N^2/n$$

Differentiating with respect to n to find a minimum gives

$$m= M/\sqrt{M+N}\ , \qquad n = N/ \sqrt{M+N}$$

$$\mathbf{C_3= 2N \sqrt{N+M}}$$

11.4 Four stage Networks

Four stage networks can be constructed by considering a complete two stage network as a single switch and then forming larger two-stage array from such switches.

If a four stage network with n incoming and n outgoing is constructed with switch size of n x n , then $N=n^3$ and the total no of switches is $4n^2$.Thus the total no of cross point is

$$C_4=4n^2.n^2$$

$$= 4N^{4/3}$$

11.5 The importance of and need for Synchronization

The synchronization network is a network that shall be able to provide all types of telecommunication traffic networks with reference timing signals of required quality. The objective for the traffic networks, for example switching, transport, signalling, mobile, is to not lose information. Loss of information is often caused by poor synchronization.

This can be avoided by properly connecting the traffic network to an adequate synchronization network (how to connect to a synchronization network is normally called network synchronization).

In the best case, poor synchronization causes only limited inconvenience to the traffic network. In the worst case, it can make the entire telecommunication network stop passing traffic.

Poor synchronization causes loss of information in varying degrees. The results for network operators providing poor synchronization to their networks are: reduced short and long term income, decreased customer satisfaction, low network availability and low traffic throughput.

Elements of a Synchronization Network

As pointed out earlier, the switches in digital communication networks in which time division multiplexing is applied, need a common timing reference. The requirements on the accuracy and stability of the reference result from the connection quality objectives of a digital connection, specified in ITU-T. Currently those requirements can only be met with atomic (Caesium-beam) clocks used as the network primary reference clock (PRC: Primary Reference Clock) or with use of GPS receivers; but by deploying different strategies on clock holdover, repair time and network planning, these objectives can also be met under failure condition for a limited period of time. The task of network synchronization is to distribute the reference signal from the PRC to all network elements requiring synchronization. The architecture of synchronization networks is described in clause 7. The method used for propagating the reference signal in the network is the master-slave method. Figure 5 shows a typical chain of master-slave-connected clocks.

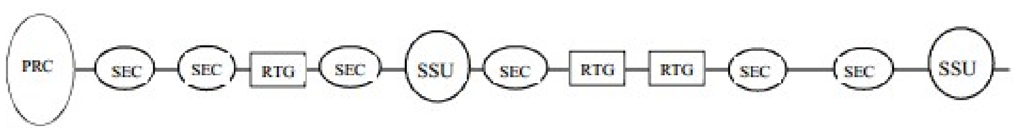

Figure 4.22 **Example of synchronization chain**

Several classes of slave clocks with different properties and different roles appear in the network:

• The PRC determines the long-term stability of the reference frequency.

• SSUs (Synchronization Supply Units) regenerate the clock signal after it has passed a chain of SECs and serve as

temporary references for parts of the network when the connection to the PRC is lost in failure situations.

Usually SSUs are located in network nodes where they distribute a timing signal to all network elements in the node.

• SECs (SDH Equipment Clocks) are the clocks incorporated in SDH network elements. SECs offer great flexibility in the selection of references and support automatic reconfiguration mechanisms in rings or chains of SDH network elements.

• RTGs (Regenerator Timing Generators) do not appear in the reference chain because due to their simple architecture and their relevant properties their influence in the synchronization chain is negligible.

The following clauses give more details on the properties and applications of the clocks in a synchronous network.

4.12.1 PRC (Primary Reference Clock)

The PRC represents a set of performance specifications for a clock generator function .These specifications aim at:

• providing a master clock source for a network;

• sufficient frequency accuracy for pseudo-synchronous operation of international (multi operator) 64 kbit/s switched networks

The network operator may run two (or more) PRC's at different locations in their network in order to achieve a very high availability of PRC reference signals to the network. In case of a fault of the currently active PRC the standby PRC will takeover the role of the failed PRC.

SSU

The SSU represents a set of performance specifications for a clock generator function. These specifications aim at:

• low bandwidth jitter filtering for removing efficiently jitter and short term wander from the synchronization reference signals;

• high degree of frequency accuracy in the holdover mode for providing a local synchronization backup and to allow for delayed maintenance response on synchronization faults;

• limited phase transient response on switching between input reference signals.