EC504C: TELECOMMUNICATION SYSTEM

OCW

OCW For EC504C Consist of PART A,B,&C

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PART A:

OCW PART A COVERS THE UNDERMENTIONED MODULES:

- Module 1:Signal Characteristics,Introduction to Telephone Systems- Bandwidth Requirement of Various Applications,Components and Examples of Telecommunication systems; Carbon Microphone and Headphone, Tone dialing; Telephone Instruments - push button types.
- Module 2:Telecommunication Transmission Lines:- Copper, Co-axial, and Fiber optic cables; Transmission Bridge - Hybrid circuit for 2-wire to 4-wire conversion and vice versa. PCM Carriers; American and European standards of carrier channels.
- Module 3: Switching System: Electro-mechanical switching- Basic idea of Strowger, Crossbar(Multi Stage Switching);Circuit Switching & Packet Switching.

Signal Characteristics

A communication system conveys information, from its source to a destination which is located some distance away. The information may be in the form of voice telephone calls, data, text, images, or video. So, the basic function of communication is information transfer in form of signal. A signal is a detectable physical quantity that varies with time, space, or any other independent variable and carry some information.

Signals are classified into the following categories:

- Continuous Time and Discrete Time Signals: A signal is said to be continuous when it is defined for all instants of time, whereas, a signal is said to be discrete when it is defined at only discrete instants of time.

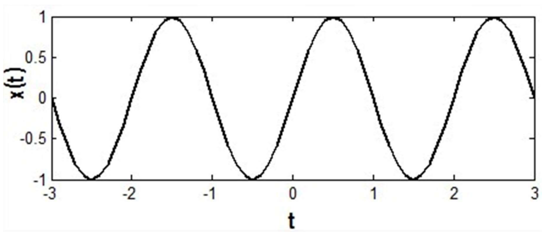

Fig 1. Continuous time signal

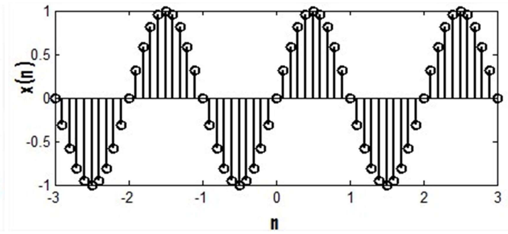

Fig 2. Discrete time signal

- Deterministic and Non-deterministic Signals: A signal is said to be deterministic if there is no uncertainty with respect to its value at any instant of time, and a signal is said to be non-deterministic if there is uncertainty with respect to its value at some instant of time. Non-deterministic signals are random in nature hence they are called random signals.

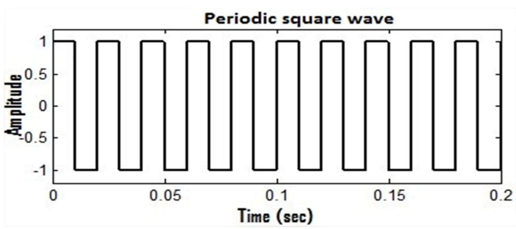

Fig 3. Deterministic signal

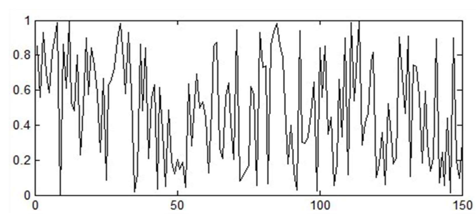

Fig 4. Non-deterministic signal

- Even and Odd Signals: A signal is said to be even when it satisfies the condition x(t) = x(-t), on the other hand, A signal is said to be odd when it satisfies the condition x(t) = x(-t).
- Periodic and Aperiodic Signals: A signal is said to be periodic if it satisfies the condition x(t) = x(t + T), Where, T = fundamental time period, 1/T = f = fundamental frequency. The signal which do not follow the above condition is called aperiodic signal.

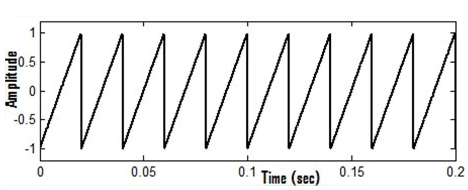

Fig 5. Periodic signal

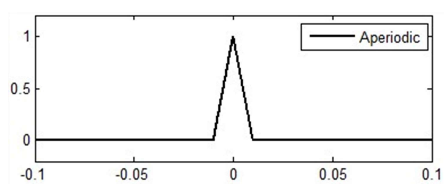

Fig 6. Aperiodic signal

- Energy and Power Signals: A signal is said to be energy signal when it has finite energy.

Energy □ = □ (□)□□

A signal is said to be power signal when it has finite power.

Power □ = lim 12□ □ (□)□□
→

- Real and Imaginary Signals: A signal is said to be real when it satisfies the condition x(t) = x*(t)

  A signal is said to be odd when it satisfies the condition x(t) = -x*(t)

The data traveling through a telecommunications system uses analog and digital electromagnetic signals. The analog signal is a continuous waveform used for voice communication that goes through a communication medium. Digital signals, on the other hand, transmit data coded as one bits and zero bits or on-off electric pulses. Computers communicate using digital signals. Whenever a computer needs to communicate over an analog line it needs a modem to translate the signals. A modem translates analog signals into digital and digital signals into analog.

Bandwidth Requirement in telecommunication System

The telephone system is one of the most marvelous inventions of the communications era. A telephone is a telecommunication device that permits two or more users to communicate when they are geographically far apart to be heard directly. In telephone conversation, one who initiates the call is referred to as the calling subscriber and the one for whom the call is destined is known as called subscriber. A telephone converts sound, mostly human voice, into electronic signals that are transmitted via cables and other communication channels to another telephone which reproduces the sound to the receiving user.

All signals that are transmitted consist of multiple frequencies. The range of frequencies a signal occupies is called the bandwidth of the signal. The bandwidth is measured in terms of Hertz (Hz).

The bandwidth of a signal depends on the amount of information contained in it and the quality of it. The range of frequencies necessary for an analogue voice signal, with a fixed telephone line quality (recognizable speaker), is 300 - 3400 Hz. This means that the bandwidth of the signal is 3,100 Hz. A human voice contains much higher frequencies, but this bandwidth gives a good compromise between the quality of the signal and the bandwidth. To transmit audio, a much wider bandwidth of about 20 kHz is needed. The bandwidth of a television signal is in the order of 5,000,000 Hz or 5 MHz.

Bandwidth, together with noise, is the major factor that determines the information-carrying capacity of a telecommunications channel. The term bandwidth is often used as data rate or bit rate to express the capacity of a digital channel. Although they are closely related, they are not the same.

The required bandwidth for various applications are listed below –

Band Name	Abbreviation	Frequency	Applications
Extremely Low	ELF	3–30 Hz	Communication with Submarines
Frequency			

Super Low Frequency	SLF	30- 300Hz	Communication with Submarines
Ultra Low Frequency	ULF	300-3000Hz	Submarine communication, communication within mines
Very Low Frequency	VLF	3-30KHz	Navigation, time signals, submarine communication, wireless heart rate monitors, geophysics
Low Frequency	LF	30-300KHz	Navigation, time signals, AM long wave broadcasting (Europe and parts of Asia), RFID, amateur radio
Medium Frequency	MF	300-3000KHz	AM (medium-wave) broadcasts, amateur radio, avalanche beacons
High Frequency	HF	3-30MHz	Shortwave broadcasts, citizens band radio, amateur radio and over-the-horizon aviation communications, RFID,over-the-horizon radar, automatic link establishment (ALE) / near-vertical incidence skywave (NVIS) radio communications, marine and mobile radio telephony
Very High Frequency	VHF	30-300MHz	FM, television broadcasts, line-of-sight groundto-aircraft and aircraft-to-aircraft communications, land mobile and maritime mobile communications, amateur radio, weather radio
Ultra High Frequency	UHF	300-3000MHz	Television broadcasts, microwave oven, microwave devices/communications, radi

			o astronomy, mobile phones,wireless LAN, Bluetooth, ZigBee, GPS and two-way radios such as land mobile, FRS and GMRS radios, amateur radio, satellite radio, Remote control Systems, ADSB
Super High Frequency	SHF	3-30GHz	Radio astronomy, microwave devices/communications, wireless LAN, DSRC, most modern radars, communications satellites, cable and satellite television broadcasting, DBS, amateur radio, satellite radio
Extremely high Frequency	EHF	30-300GHz	Radio astronomy, high-frequency microwave radio relay, microwave remote sensing, amateur radio, directed-energy weapon, millimeter wave scanner, wireless LAN (802.11ad)
Thz or Tremendously High Frequency	THF	300-3000GHz	Experimental medical imaging to replace Xrays, ultrafast molecular dynamics, condensedmatter physics, terahertz time-domain spectroscopy, terahertz computing/communications, remote sensing, amateur radio

Table 1. Bandwidth requirement in Various Applications

Carbon Microphone

Microphones are the devices which convert sound pressure waves into signal patterns. The carbon microphone was developed in the 1870s by Englishman David Edward Hughes. Now a days this type of microphone is not widely used. It was the first reliable form of microphone and it was widely used for many years before, but now replaced by other types of microphones that gives much superior levels of performance.

Carbon Microphone is also known as carbon button microphone or a carbon transmitter. It consists of two metal plates separated by carbon granules. One of the plates faces outwards and acts as a diaphragm. When sound waves strike these metal plates the granules change which in turn alters the electrical resistance between the plates.

Direct current passes through the two plates and the changing resistance results in changing current. The current then passes through a telephone system to change the sound into an electrical signal.

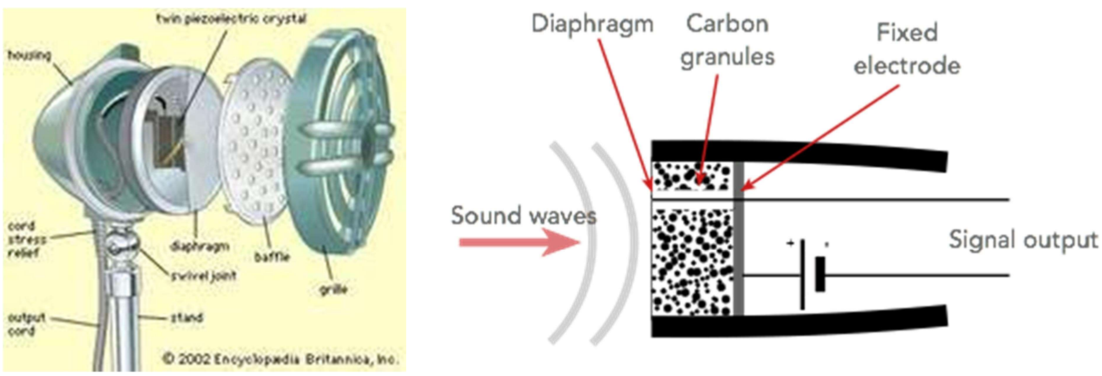

Fig 7. Carbon Microphone

Application

- Carbon microphones were an ideal choice of microphone in the early days of the telephone. They were widely used in telephone applications because they gave a high output which meant no amplification was used.
- As radio started to be used, the carbon microphone was initially used there as well – for broadcasting as well as communications purposes.

Advantage

- High output
- Simple principle & construction
- Cheap and simple to manufacture

Disadvantage

- Very noisy - high background noise and on occasions it would crackle
- Poor frequency response
- Requires battery or other supply for operation

Headphone

Headphones are a pair of small loudspeaker drivers worn on or around the head over a user's ears. They are basically transducers, which convert an electrical signal into a corresponding sound signal. Headphones is a single user audio source, in contrast to a loudspeaker, which emits sound into the open air for anyone nearby to hear. Headphones are also known as ear speakers, earphones .In the context of telecommunication, a headset is a combination of headphone and microphone. Headphones connect to a signal source such as an audio amplifier, radio, CD player, portable media player, mobile phone, video game console, or electronic musical instrument, either directly using a cord, or using wireless technology such as Bluetooth, DECT or FM radio.

Application

- Headphones may be used with stationary CD and DVD players, home theater, personal computers, or portable devices
- In the professional audio sector, headphones are used in live situations by disc jockeys with a DJ mixer, and sound engineers for monitoring signal sources.

Tone Dialing

Tone dialing is the modern way of dialing. It is a way of communicating with the central telephone relay to indicate the telephone number that we wish to call. In tone dialing, different tones are used to indicate different numbers. This dialing system sends only one signal for each digit. Generally, tone dialing comes along with telephones with numeric keypads. During using tone dialing phones with numeric key pads, the caller is capable of dialing the phone number, which they wish to call in a quicker time. Further, tone dialing is also called as Dual Tone Multi Frequency (DTMF).The touch tone dialing arrangement is as shown below:

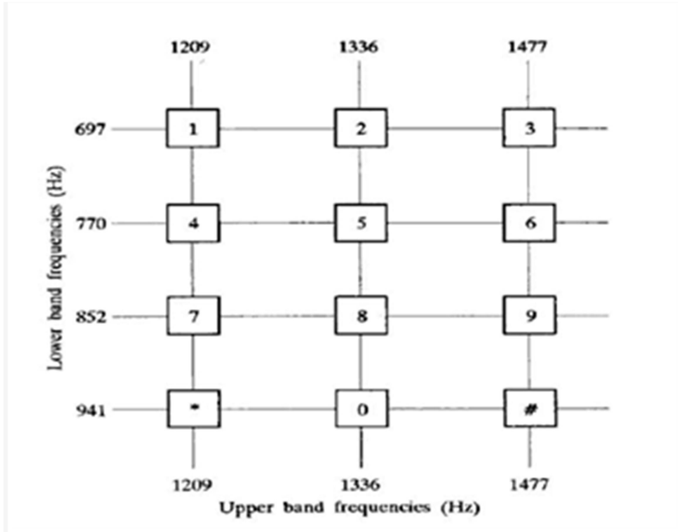

Fig 8. DTMF Technology

Each key in DTMF system represent two audio frequencies which are associated row and column wise as shown. For example if key 5 is pressed then audio frequency pair (770, 1336) in Hz is generated and transmitted to the telephone exchange. These tones are recognized at the telephone central office (CO). Tolerance on these frequencies should be within +/- 1.5 percent.

The following fig. shows a simplified block diagram of a touch tone receiver. The limiters attenuate differences in levels between components of an incoming multi-frequency signal.

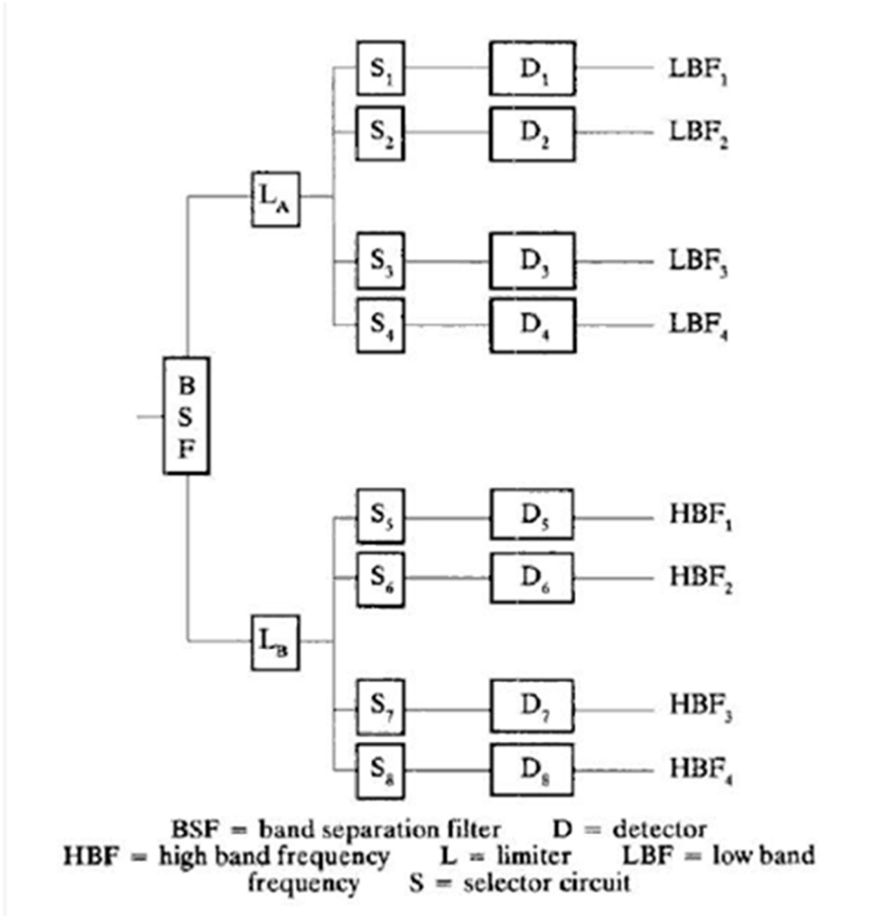

Fig 9. Block Diagram of touch tone receiver

If two frequencies reach the limiter with one of the being relatively stronger, the o/p of the limiter peaks with the stronger signal and weaker signal is further attenuated. If both reach with same strength, limiter o/p is much below full o/p and neither signal dominates. The selective circuitry is designed to recognize signal when it falls within the specified narrow-pass band and has an amplitude within about 2.5 db of full o/p of the limiter. The limiter and the selective circuitry together reduce the probability mistaking touch signal to audio or vice versa. Band elimination filters are sometimes used in place of band separation filters as they permit a wider spectrum of speech to pass through the filters.

Push Button Telephone

The push-button telephone is a telephone that has buttons or keys for dialing a telephone number. Western Electric experimented in 1941 with methods of using mechanically activated reeds to produce two tones for each of the ten digits and by the late 1940s such technology was fieldtested. But the technology proved unreliable and it was not until long after the invention of the transistor when push-button technology matured. On 18 November 1963, after approximately three years of customer testing, the Bell System in the United States officially introduced dualtone multi-frequency (DTMF) technology under its registered trademark Touch-Tone. Over the next few decades touch-tone service replaced traditional pulse dialing technology and it eventually became a world-wide standard for telecommunication signaling.

Transmission Line

In communications a transmission line is a specialized cable designed to that transfers electrical signals from one place to another. Different types of transmission lines that are used in telecommunication includes- Copper wire cable, Coaxial Cable, Optical Fiber Cable.

Copper Cable

One of the most commonly used conductor materials as a transmission line conductor is copper. Copper is the best conductor due to its high electrical conductivity and greater tensile strength. It is ideal conductor for transmission and distribution lines. But due to its high cost and nonavailability. It is rarely used for overhead transmission lines. Now days Aluminum conductor is used for overhead lines. The use of copper conductor only fount machine winding.

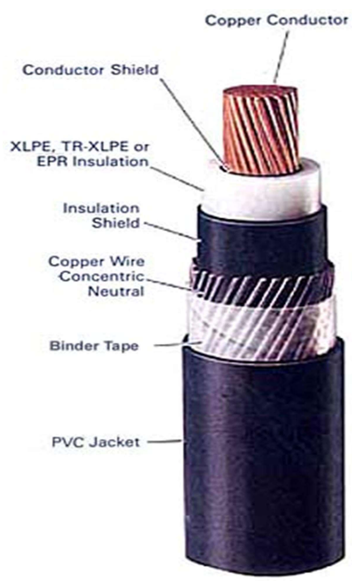

Fig 10. Copper Cable

Coaxial Cable

Coaxial cable is a type of copper cable specially built with a metal shield and other components to block signal interference. Coaxial cable is commonly used by cable operators, telephone companies, and internet providers around the world to convey data, video, and voice communications to customers. It has also been used extensively within homes. This cable has high bandwidths and greater transmission capacity.

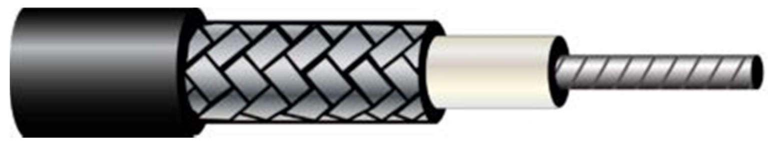

Fig 11. Structure of Coaxial Cable

Coaxial cable is made of two conductor electrical cable consisting of a center conductor and an outer conductor with an insulating spacer between the two. Primarily, this cable is used for the transmission of Radio Frequency energy. The system offers tight control over electrical impedance. This yields excellent performance at high frequencies and superior EMI control/shielding.

Coaxial cables differ in design depending on their end use. Shorter cables often appear in household applications such as AV systems or personal Ethernet connections, while longer cables can connect entire radio and television networks or long-distance phones. Micro/mini cables are also frequently used in various consumer, military, aerospace, and medical devices.

Fiber optic cables

In recent years it is apparent that fiber-optics are steadily replacing copper wire as an appropriate means of communication signal transmission. They span the long distances between local phone systems as well as providing the backbone for many network systems. Fiber optics transmit data in the form of light particles -- or photons -- that pulse through a fiber optic cable. A fiber-optic cable is made up of very thin strands of glass or plastic known as optical fibers; one cable can have as few as two strands or as many as several hundred. Optical cables are used to transfer digital data signals in the form of light up to distances of hundreds of miles with higher throughput rates than those achievable via electrical communication cables. When light signals are sent through the fiber optic cable, they reflect off the core and cladding in a series of zig-zag bounces, adhering to a process called total internal reflection. All optical fibers use a core of transparent silicon covered with less refractive indexed cladding to avoid light leakage to the surroundings. Due to the extreme sensitivity of the optical fiber, it is normally covered with a high-strength, lightweight protective material like Kevlar. The glass fiber core and the cladding each have a different refractive index that bends incoming light at a certain angle. The light signals do not travel at the speed of light because of the denser glass layers, instead traveling about 30% slower than the speed of light. To renew, or boost, the signal throughout its journey, fiber optics transmission sometimes requires repeaters at distant intervals to regenerate the optical signal by converting it to an electrical signal, processing that electrical signal and retransmitting the optical signal.

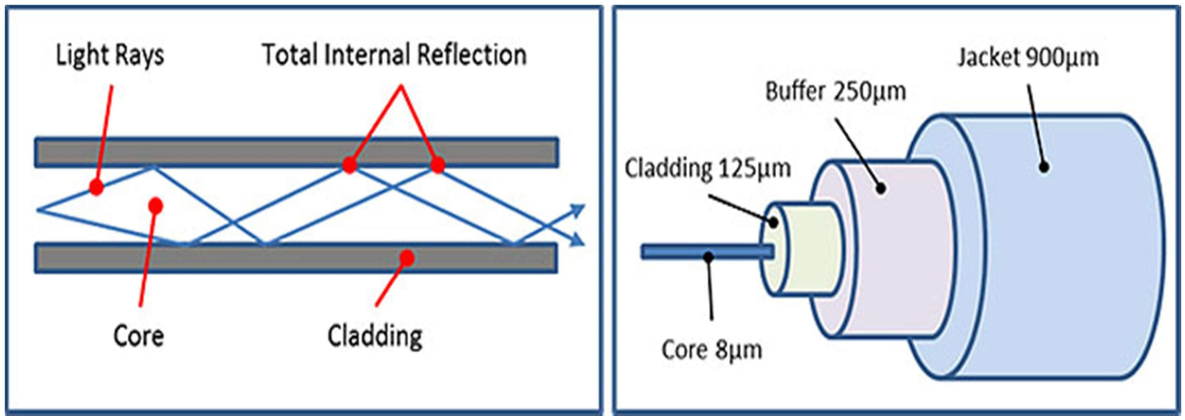

Fig 12. Structure and signal propagation through Fiber Optic Cable

Types of fiber optic cables

Multimode fiber and single-mode fiber are the two primary types of fiber optic cable. Singlemode fiber is used for longer distances due to the smaller diameter of the glass fiber core, which lessens the possibility for attenuation -- the reduction in signal strength. The smaller opening isolates the light into a single beam, which offers a more direct route and allows the signal to travel a longer distance. Single-mode fiber also has a considerably higher bandwidth than multimode fiber. Single-mode fiber is usually more expensive because it requires precise calculations to produce the laser light in a smaller opening.

Multimode fiber is used for shorter distances because the larger core opening allows light signals to bounce and reflect more along the way. The larger diameter permits multiple light pulses to be sent through the cable at one time, which results in more data transmission. This also means that

there is more possibility for signal loss, reduction or interference, however. Multimode fiber optics typically use an LED to create the light pulse.

While copper wire cables were the traditional choice for telecommunication, networking and cable connections for years, fiber optics has become a common alternative. Most telephone company long-distance lines are now made of fiber optic cables. Optical fiber carries more information than conventional copper wire, due to its higher bandwidth and faster speeds. Because glass does not conduct electricity, fiber optics is not subject to electromagnetic interference and signal losses are minimized.

In addition, fiber optic cables can be submerged in water and are used in more at-risk environments like undersea cable. Fiber optic cables are also stronger, thinner and lighter than copper wire cables and do not need to be maintained or replaced as frequently. Copper wire is often cheaper than fiber optics, however, and is already installed in many areas where fiber optic cable hasn't been deployed. Glass fiber also requires more protection within an outer cable than copper, and installing new cabling is labor-intensive, as it typically is with any cable installation.

Fiber optics uses

Computer networking is a common fiber optics use case, due to optical fiber's ability to transmit data and provide high bandwidth. Similarly, fiber optics is frequently used in broadcasting and electronics to provide better connections and performance.

Hybrid Circuit for 2 Wire to 4 Wire Conversion

A telephone conversation inherently requires transmission in both directions. When both directions are carried on the same pair of wires, it is called two-wire transmission. The telephones in our homes and offices are connected to a local switching center (exchange) by means of two-wire circuits. A more proper definition for transmitting and switching purposes is that when oppositely directed portions of a single telephone conversation occur over the same electrical transmission channel or path, we call this two-wire operation.

The need for hybrids comes from the nature of analog plain old telephone service home or small business telephone lines, where the two audio directions are combined on a single two-wire pair. Within the telephone network, switching and transmission are almost always four-wire circuits with the two signals being separated. Hybrids perform the necessary conversion. In older analog networks, conversion to four-wire was required so that repeater amplifiers could be inserted in long-distance links. In today's digital systems, each speech direction must be processed and transported independently. Four-Wire Transmission. Carrier and radio systems require that oppositely directed portions of a single conversation occur over separate transmission channels or paths (or use mutually exclusive time periods). Thus we have two wires for the transmit path and two wires for the receive path, or a total of four wires, for a full-duplex (two-way) telephone conversation. For almost all operational telephone systems, the end instrument (i.e., the telephone subset) is connected to its intervening network on a two-wire basis.

Nearly all long-distance telephone connections traverse four-wire links. From the near-end user the connection to the long-distance network is two-wire or via a two-wire link. Likewise, the farend user is also connected to the long-distance network via a two-wire link. Such a long-distance connection is shown in Figure 13. Schematically, the four-wire interconnection is shown as if it were a single-channel wire-line with amplifiers. However, it would more likely be a multichannel multiplexed configuration on fiber-optic cable or over radio. As illustrated in Figure 13, conversion from two-wire to four-wire operation is carried out by a hybrid, which is a four-port, four-winding transformer.

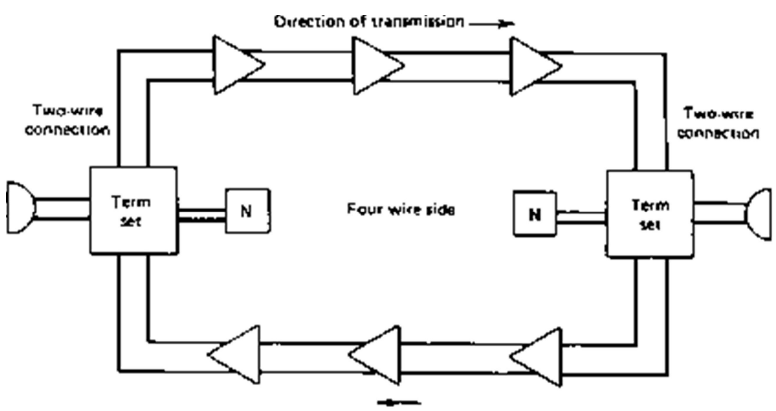

Fig 13. A typical long distance connection model

A hybrid, in terms of telephony, is a transformer with four separate windings. Based on a simplified description, a hybrid may be viewed as a power splitter with four sets of wire-pair connections. A functional block diagram of a hybrid device is shown in Figure 14. Two of the wire-pair connections belong to the four-wire path, which consists of a transmit pair and a receive pair. The third pair is a connection to the two-wire link, which is connected to the subscriber subset via one or more switches. The last pair of the four connects the hybrid to a resistance-capacitance balancing network, which electrically balances the hybrid with the twowire connection to the subscriber subset over the frequency range of the balancing network. Balancing, in this context, means matching impedances—that is, the impedance of the two-wire side to the hybrid two-wire port.

Signal energy entering from the two-wire subset connection divides equally. Half of it dissipates (as heat) in the impedance of the four-wire side receive path and the other half goes to the fourwire side transmit path, as illustrated in Figure 14. Here the ideal situation is that no energy is to be dissipated by the balancing network (i.e., there is a perfect balance or impedance match). The balancing network is supposed to display the characteristic impedance of the two-wire line (subscriber connection) to the hybrid. Half of the energy is dissipated by the balancing network (N) and half at the two-wire port (L).

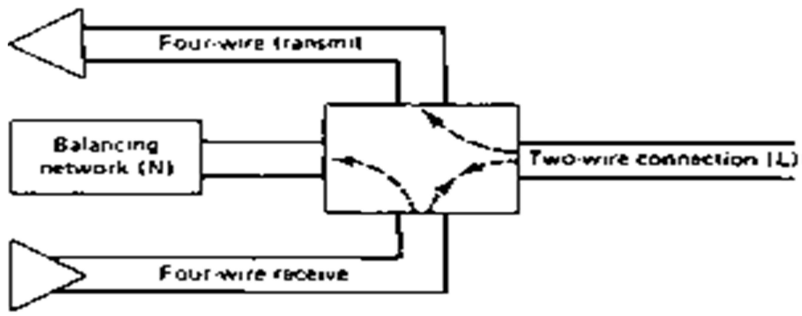

Fig 14. Operation of a hybrid transformer

The earliest hybrids consisted of a coil with multiple windings - it may be considered as a transformer or an inductor, but in many cases it's really an autotransformer, with all windings joined at some point. Figure15 shows one of the arrangements that were used - the hook switch and ringer circuits have been omitted for clarity. Here, in all cases, the resistance of the transformer windings must be considered when determining impedance matching.

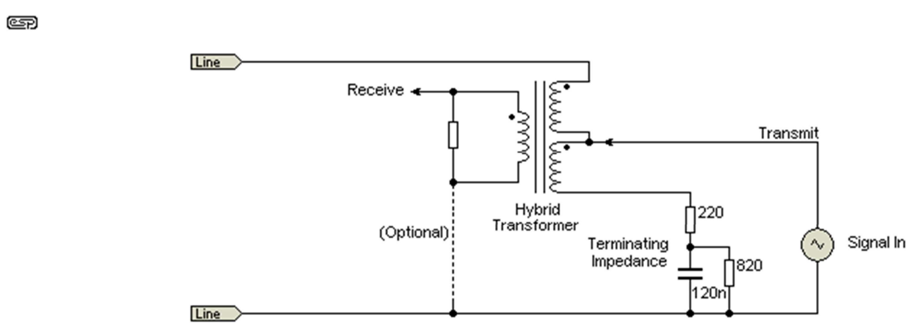

Fig 15. Single transformer hybrid

The dots indicate the start of winding. The single transformer hybrid is capable of extremely good performance. In exactly the same way as an active hybrid. The performance in both directions is degraded if the line impedance does not match the design value. The impedance matching network must be located in the position shown. Maximum return loss is achieved with the load impedance, at the receive port should be as high as possible.

The dual transformer version has the benefit that all ports (transmit line and receive line) are isolated from each other. This is of no consequence in a telephone, but may be important for some exchange equipment. While this version is still used (transformers are still available), it is uncommon in general phone circuits. Performance can be extremely good, but compared to IC replacements that are now very common, but the space and expense make it unattractive.

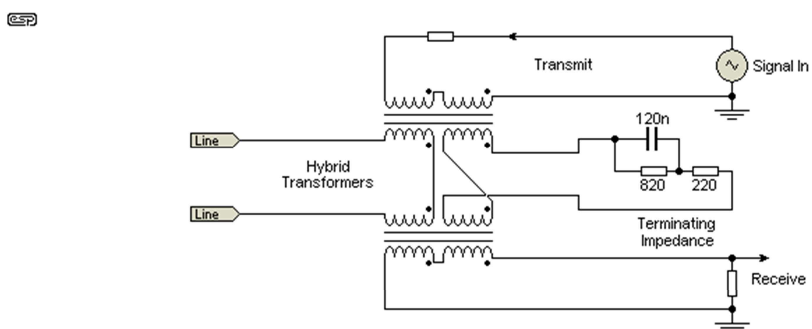

Fig 16. Dual transformer hybrid

Here, the terminating impedance has no influence at all on the impedance presented to the 2-wire port. The proper impedance is set by using a modified network in series with the transmit port, and in parallel with the receive port. In this respect, this hybrid is unique. However - both are affected by the actual external impedance, so an artificial line will mess up both. Also the echo is caused in the hybrid circuit by any impedance mismatch.

PCM Carriers: American and European standards of carrier channels

PCM system is a digital representation of an analog signal, in which the magnitude of the analog signal is sampled regularly at uniform intervals, with each sample being quantized to the nearest value within a range of digital steps. For digital communication, there are two quite different PCM standards.

- The North American standard, also called T1, the more standard name is DS1 PCM hierarchy.
- The European system E1 hierarchy.

PCM system was designed to convert analog voice communications into digital voice communications. To do that, voice characteristics were analyzed. Basically voice operates in a telephony world in a band-limited channel operation. The typical human speech is represented by a simple 1200 Hz tone, which is represented by a sine wave. The steps for developing an equivalent PCM digital signal from an analog signal are-

- Sampling
- Quantization
- Encoding

Sampling

Sampling is done over the analog wave at twice the highest range of frequencies that can be carried across the line. In North American T1 (DS1) system, 24 voice channels are sampled sequentially and are interleaved to form a PAM-multiplexed wave. The sampling gate is open about 5.2 µs (125 µs/24) for each voice. The full sequence of 24 channels is sampled successively from channel 1 through channel 24 in a 125 µs period. This 125 µs period is a called a frame, and inside the frame all 24 channels are successively sampled just once.

In European E1 system, there are 30 voice channels plus an additional two service channels, for a total of 32 channels. By definition, this system must sample 8000 times per second because it is also optimized for voice operation, and thus its frame period is also 125 µs. To accommodate the 32 channels, the sampling gate is open 125 µs/32 or about 3.906 µs.

Quantization

Assigning a binary sequence to each of those voltage samples, is called quantization. That means this process quantify the values using a logical pattern of 1s and 0s to represent the height of the signal at any point in time. This deals with the amplitude shifts only. Quantization is of two types- uniform and non uniform quantization. In uniform quantization levels are uniformly spaced, whereas in non-uniform quantization spacing between the levels will be unequal and dependent on the signal strength.

Non-uniform quantization uses a technique of companding. Companding derives from two words: compression and expansion. Compression takes place on the transmit side; expansion on the receive side. Compression reduces the dynamic range with little loss of fidelity, and expansion returns the signal to its normal condition.

Encoding

Older PCM systems used 7-bit code, and modern systems use an 8-bit code with its improved quantizing distortion performance. The companding and coding are carried out together, simultaneously. The compression and later expansion functions are logarithmic. The logarithmic curve follows one of two laws, the A-law and the μ-law.

The A-law is used with European E1 system. A law can be expressed as follows-

$$F(x) = \operatorname{sgn}(x) \begin{cases} \frac{A|x|}{1+\log(A)}, & |x| < \frac{1}{A} \\ \frac{1+\log(A|x|)}{1+\log(A)}, & \frac{1}{A} \leq |x| \leq 1, \end{cases}$$

Where A is the compression parameter. In Europe, A = 87.6

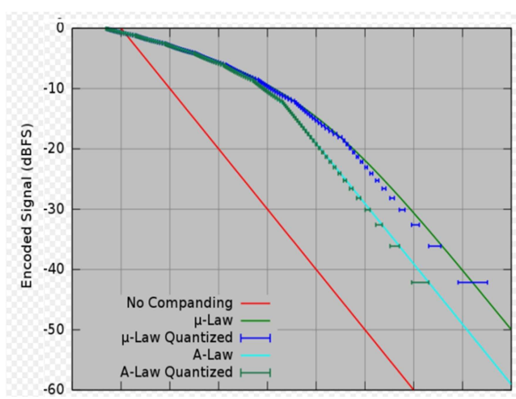

Fig 17. Companding of A-law and μ-law algorithms

North American T1 system μ-law algorithm is used. The μ-law algorithm may be described in an analog form and in a quantized digital form.

$$F(x) = \operatorname{sgn}(x)\frac{\ln(1+\mu|x|)}{\ln(1+\mu)} \quad -1 \leq x \leq 1$$

Where μ = 255 (8 bits) in the North American and Japanese standards.

PCM Frame

The final result of the sampling and subsequent quantization and coding is a series of electrical pulses, a serial bit stream of 1s and 0s that requires some identification or indication of the beginning of a sampling sequence. Once the receiver receives the “indication”, it knows a priori (in the case of DS1) that 24 eight-bit slots follow. It synchronizes the receiver. Such identification is carried out by a framing bit, and one full sequence or cycle of samples is called a frame in PCM terminology.

Considering the framing structure of North American DS1 (T1) system, It is a 24-channel PCM system using 8-level coding (e.g., 28 = 256 quantizing steps or distinct PCM code words). The DS1 frame shown in the following figure has one bit added as a framing bit (this is that indication to tell the distant end receiver where the frame starts.) It is called the “S” bit.

The DS1 frame then consists of (8×24) +1 = 193 bits.

By definition, 8000 frames are transmitted per second (i.e., 4000 x 2, the Nyquist sampling rate), so the bit rate of DS1 is

193 x 8000 = 1,544,000 bps or 1.544 Mbps

The E1 European PCM system is a 32-channel system. Of the 32 channels, 30 transmit speech (or data) derived from incoming telephone trunks and the remaining 2 channels transmit synchronization-alignment and signaling information. Each channel is allotted an 8-bit slot (TS), and this is shown in the following table.

TS	Type of Information
0	Synchronizing (framing)
1-15	Speech
16	Signaling
17-31	Speech

Again, E1 in its primary rate format transmits 32 channels of 8-bit time slots. An E1 frame therefore has 8×32=256 bits. There is no framing bit. Framing alignment is carried out in TS 0.

The E1 bit rate to the line is

$$256 \times 8000 = 2{,}048{,}000 \text{ bps} \qquad \text{or } 2.048 \text{ Mbps}$$

Electro-mechanical switching

In telecommunications, an electronic switching system (ESS) is a telephone switch that uses digital electronics and computerized control to interconnect telephone circuits for the purpose of establishing telephone calls.

The generations of telephone switches before the advent of electronic switching in the 1950s used purely electro-mechanical relay systems and analog voice paths. These early machines typically utilized the step-by-step technique. The first generation of electronic switching systems in the 1960s were not entirely digital in nature, but used reed relay-operated metallic paths or crossbar switches operated by stored program control (SPC) systems.

Step by step or Strowger switching system

Step by step or Strowger switching is the first automatic switching system introduced by Almon B Strowger. This system uses selectors for switching. The selectors used in Strowger exchange are mainly of types:

- Uniselector
- Two Motion selector

Both the selectors belong to the same types of switches called rotary switches.

Uniselector

This is called uniselector because the rotary motion of this switch is unidirectional, i.e. the wiper assembly moves only in one direction. The uniselector consists of moving contacts called wipers. These are used to make electrical connections with any one of the several contacts, called Bank Contact, in an arc around it. The arc in most cases consists of ten steps.

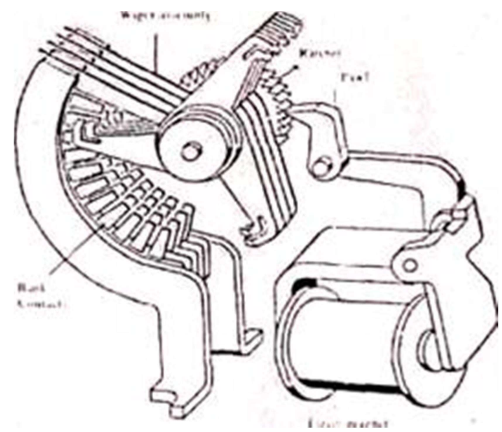

Fig: 18 Uniselector Strowger Switching system

The wiper assembly is divided into three sets of wipers so the switch has to turn through only one third of full cycle when operated. These wipers are operated by an electromagnet called, driving magnet called with the help of a ratchet and pawl mechanism. When the current flows through the windings of the driving magnet it is energized and attracts the armature; the pawl slips over one tooth of ratchet wheel. The ratchet is prevented from movement in the reverse direction by a detent. When the current stops through the windings of the driving magnet, it is de-energized and the armature comes back to its rest position. During the reverse movement of the armature and hence that of the pawl, the ratchet wheel stop ahead in the clockwise direction by one tooth and the wipers move to the next contact. The uniselector rotates as many steps as the electromagnet is energized and de-energized.

Two Motion selector

The two motion selector is a type of rotary switch in which the motion of the wiper assembly is in two directions, vertical as well as horizontal direction. In the vertical direction the wipers move upward to the desired level and make no connections with the bank contacts. While in the horizontal direction wiper make connection with bank contacts. The two motion selector has 10 levels each having 10 contacts, thus a total of 100 contacts are accessible.

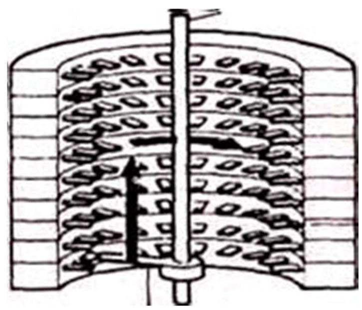

Fig: 19 Two motion selector Strowger Switching system

Each contact represents the terminal of one switch of the higher stage or of one telephone line in case of final selector. The dialing pulses cause the wiper assembly to step up or down to the desired level. If we take the example of a final selector, where up to 100 lines can be connected, the vertical and horizontal steeping of the selector are controlled by the digits dialed by the subscriber. When the first digit is dialed the dialing pulses energies and de-energies the vertical magnet. The vertical magnet with the help of ratchet and pawl mechanism step up the wiper assembly corresponding to the digit dialed. This is called vertical steeping.

When the second digit is dialed the dialing pulses are now diverted to another magnet, with the help of relay. These pulses energies and de-energies the horizontal magnet, which with the help of ratchet and pawl causes the wiper assembly to rotate the proper contact, corresponding to the second digit dialed. This is called horizontal steeping. Thus the wiper assembly makes connection with the required number dialed by the subscriber. After completion of the call the wiper assembly comes back to the home position. For this purpose the rotary magnet is operated by the current and thus wiper assembly moves through the remaining contacts of the level. A spring forces the wiper assembly to drop vertically and then to return horizontally to its normal position. Thus the two motion selector does not require additional magnet for its" homing process". This switch is also called a rectangular motion selector, because the wiper assembly moves along a rectangle.

Crossbar Switching System

In telecommunications, a crossbar switch (also known as cross-point switch, crosspoint switch, or matrix switch) is a switch connecting multiple inputs to multiple outputs in a matrix manner. Originally the term was used literally, for a matrix switch controlled by a grid of crossing metal bars, and later was broadened to matrix switches in general. It is one of the principal switch architectures, together with a rotating switch, memory switch and a crossover switch.

The basic idea of crossbar switching is to provide a matrix of nx m sets of contacts with only n x m activators or less to select one of the nxm sets of contacts as shown in Figure 1. This form of switching is also known as coordinate switching as the switching contacts are arranged in a x-yplane.

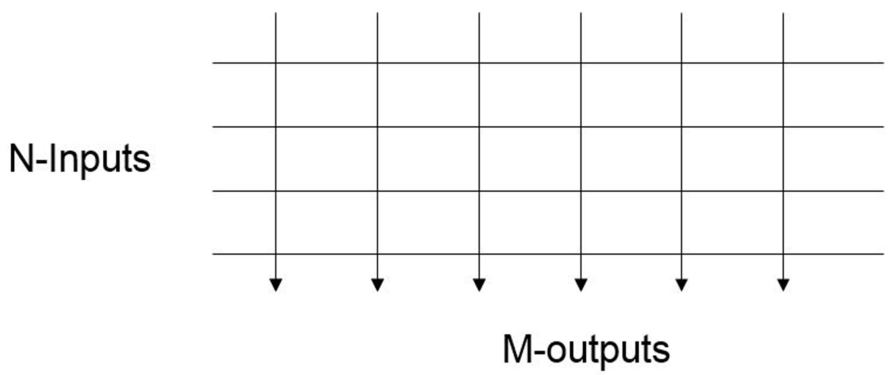

Figure 20 Cross-bar switch as an n x m matrix

A diagrammatic representation of a practical crosspoint switching matrix is shown in Figure 19. There is an array of horizontal and vertical wires shown by solid lines. A set of vertical and horizontal contact points are connected to these wires. The contact points form pairs, each pair consisting of a bank of three or four horizontal and a corresponding bank of vertical contact points. A contact point pair acts as a cross-point switch and remains separated or open when not in use. The contact points are mechanically mounted (and electrically insulated) on a set of horizontal and vertical bars shown as dotted lines. The bars, in turn, are attached to a set of electromagnets.

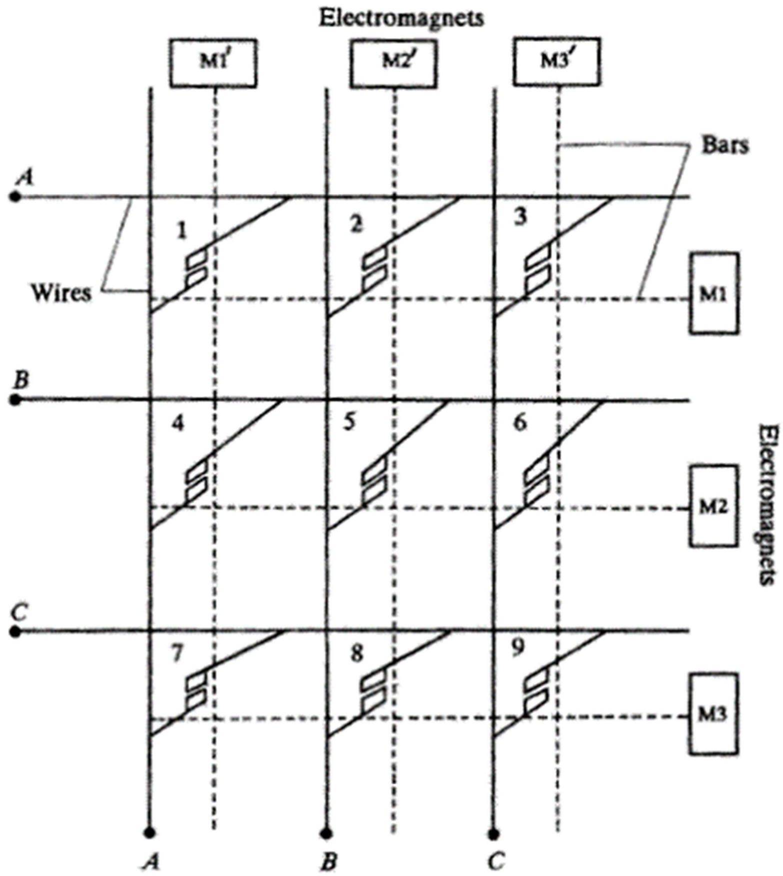

Figure 21 Implementation of cross-bar switching matrix

When an electromagnet, say in the horizontal direction, is energized, the bar attached to it slightly rotates in such a way that the contact points attached to the bar move closer to its facing contact points but do not actually make any contact. Now, if an electromagnet in the vertical direction is energized, the corresponding bar rotates causing the contact points at the intersection of the two bars to close. This happens because the contact points move towards each other. As an example, if electromagnets M2 and M3 are energized, a contact is established at the crosspoint 6 such that the subscriber B is connected to the subscriber C.

In order to fully understand the working of the crossbar switching, let us consider a 6 X 6 crossbar schematic shown in Figure 21.

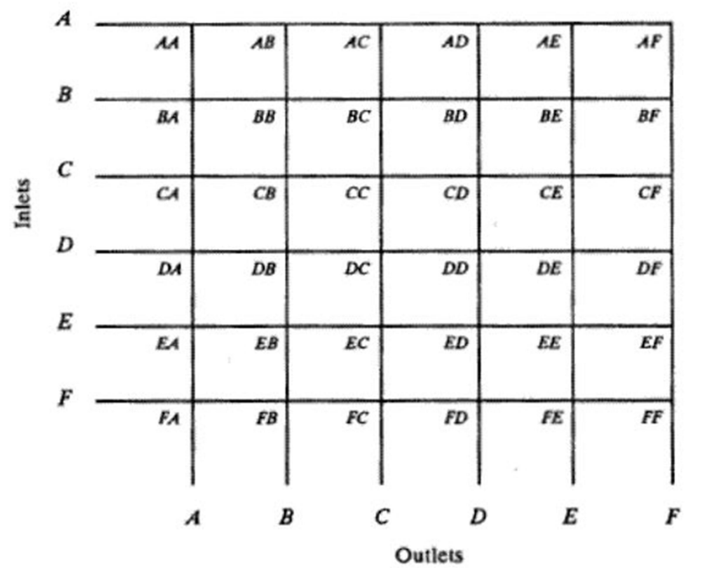

Fig 22. Cross-bar switch with six inlets and six outlets

The schematic shows six subscribers with the horizontal bars representing the inlets and the vertical bars the outlets. Now consider the establishment of the following connections in sequence: A to C and B to E. First the horizontal bar A is energized. Then the vertical bar C is energized. The crosspoint AC is latched and the conversation between A and C can now proceed. Suppose we now energize the horizontal bar of B to establish the connection B-E, the crosspoint BC may latch and B will be brought into the circuit of A-C. This is prevented by introducing an energizing sequence for latching the crosspoints. A crosspoint latches only if the horizontal bar is energized first and then the vertical bar. (The sequence may well be that the vertical bar is energized first and then the horizontal bar). Hence the crosspoint BC will not latch even though the vertical bar C is energized as the proper sequence is not maintained. In order to establish the connection B-E, the vertical bar E needs to be energized after the horizontal bar is energized. In this case, the crosspoint AE may latch as the horizontal bar A has already been energized for establishing the connection A-C. This case should also be avoided and is done by de-energizing the horizontal bar A after the crosspoint is latched and making a suitable arrangement such that the latch is maintained even though the energization in the horizontal direction. The crosspoint remains latched as long as the vertical bar E remains energized. As the horizontal bar A is deenergized immediately after the crosspoint AC is latched, the crosspoint AE does not latch when the vertical bar E is energized.

CROSSBAR SWITCH CONFIGURATIONS

Non-Blocking Cross-bar Switch

The switching matrices discussed in the preceding actions are referred to as non-blocking. In a non-blocking crossbar configuration, there are N2 switching elements for N subscribers. When all the subscribers are engaged, only N/2 switches are actually used to establish connections.

The following table shows the values of different design parameters for four non-blocking switches. Unit cost is assumed for each cross-point switching element. Providing N2 cross-points even for moderate number of users leads to impractical complex circuitry. A 1000-subscriber exchange would require 1 million cross-point switches. Therefore ways and means have to be found to reduce the number of switch contacts for a given number of subscribers.

Number of Subscribers	Switches	S/Capacity	EUF (Capacity/Sw)
4	16	2	0.125
16	256	8	0.03125
64	4,096	32	0.0078125
128	16,384	64	0.00390625

Non-blocking cross-bar switch design parameters

It may be observed in the switch matrix of Figure 20 that different switch points are used to establish a connection between two given subscribers, depending upon who initiates the call. For example, when the subscriber C wishes to call subscriber B, crosspoint CB is energized. On the other hand, when B initiates the call to contact C, the switch BC is used. By designing a suitable control mechanism, only one switch may be used to establish a connection between two subscribers, irrespective of which one of them initiates the call. In this case, the crosspoint matrix reduces to a diagonal matrix with N2/2 switches. A diagonal connection matrix using this principle for 4 subscribers is shown in Figure 22.

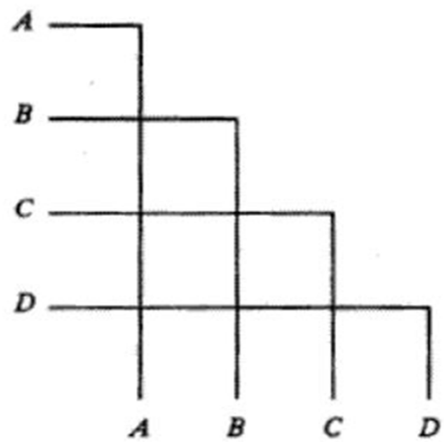

Fig 23. Diagonal Cross-point Matrix

The cross-points in the diagonal connection the inlets and the outlet of the same subscribe is not relevant. Hence, these are eliminated. The number of crosspoints then reduces to N(N-1)/2. It may be recalled that the quantity N(N-1)/2 represents the number of links in a fully connected network. So also, the diagonal crosspoint matrix is fully connected. The call establishment procedure here is dependent on the source and destination subscribers. When subscriber D initiates a call, his horizontal bar is energized first and then the appropriate vertical bar. If subscriber A initiates a call, the horizontal bar of the called party is activated first and then the vertical bar of A.

A diagonal cross-point matrix is a non-blocking configuration. Even N(N-1)/2 cross-point switches can be a very large number to handle practically. The number of cross-point switches can be reduced significantly by designing blocking configurations. These configurations may be single stage or multistage switching networks.

Use of Double-Swing Bars

The crossbar hardware may be reduced by connecting two subscribers to a single bar and letting the bar turn both in the clockwise and the anticlockwise directions, and thus closing two different cross-point contacts. With such an arrangement the number of crossbars reduces, but the number of crosspoint switches remains the same.

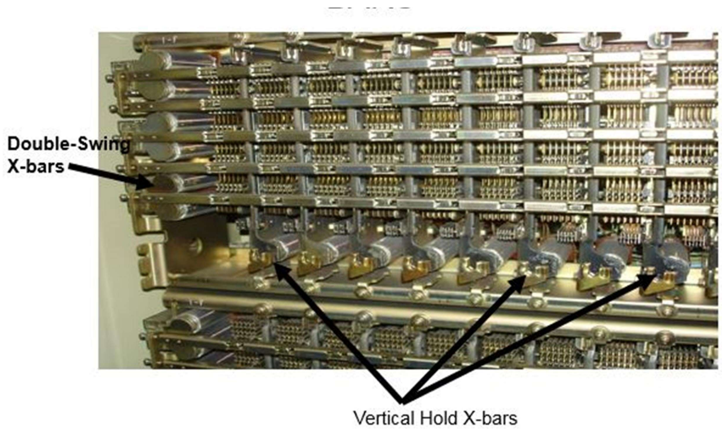

Fig 24. Photograph of a cross-bar switch using double swing horizontal bars.

Blocking Crossbar Switches

In blocking crossbar switches, the number of vertical bars is less than the number of subscribers and determines the number of simultaneous calls that can be put through the switch. Consider the 8 X 3 switch shown in Figure 25.

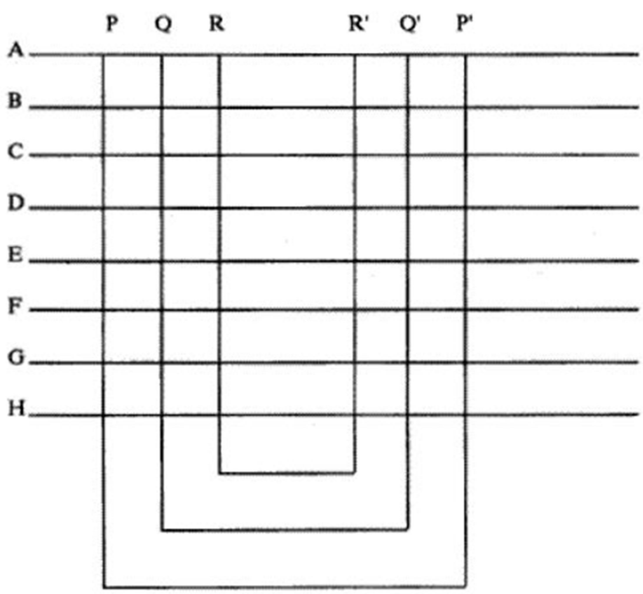

Figure 25. Blocking crossbar switch (8 - 3)

Let a connection be required to be established between the subscribers A and B. First the horizontal bar A is energized. Then one of the free vertical bars, say E: is energized. The crosspoint AP latches. Now if we energize the horizontal bar B, BP will not be latched, as the P vertical is energized before B was energized. In order to be able to connect A to B, we need another vertical crossbar which should electrically correspond to the vertical bar P. In this case, the bar P' is associated with the same electrical wire as the bar P. When P' is energized after B, the crosspoint BP' is latched and a connection between A and B is established. The sequence to be followed in establishing the A-B circuit may be summarized as:

(a) Energize horizontal A

(b) Energize free vertical P

ORGANIZATION OF CROSS-BAR EXCHANGE

The basic building blocks of a crossbar exchange are link frames, control markers and registers. Link frames consist of a number of crossbar switches arranged in two stages called primary and secondary with links between them as shown in Figure 26. The two-stage arrangement with links has the effect of increasing the number of outlets for a given number of inlets, thereby providing greater selectivity. The switch, in this case, is said to be expanding. Markers control the connections between the inlet and the outlets via the primary section, links and the secondary section.

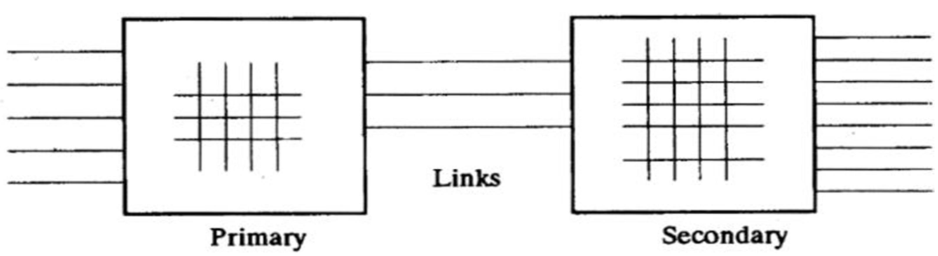

Figure 26. Primary and Secondary Links

A simplified organization of a crossbar exchange is shown in Figure 26. The line link frames along with the associated markers and registers are known as line unit, and the trunk link frame with its associated hardware as group unit. The trunk link frame may be subdivided into two or three link frames like local office link frame, incoming link frame, etc. Line units are two-way units, that is to say, they can be used for originating as well as terminating calls. It may be noted that this is a significant departure from the Strowger exchange designs where the originating and terminating units are separate and independent. Because of its two-way capability, the secondary section in the line link frame is sometimes called the terminal section. The subscriber lines are terminated on the outlets of the terminal section frames. The group unit is a unidirectional device. It receives the calls from the line unit or from distant exchanges. It routes the calls to the unit of the same exchange or to distant exchanges. It is capable of handling local, outgoing, incoming, terminating and transit calls.

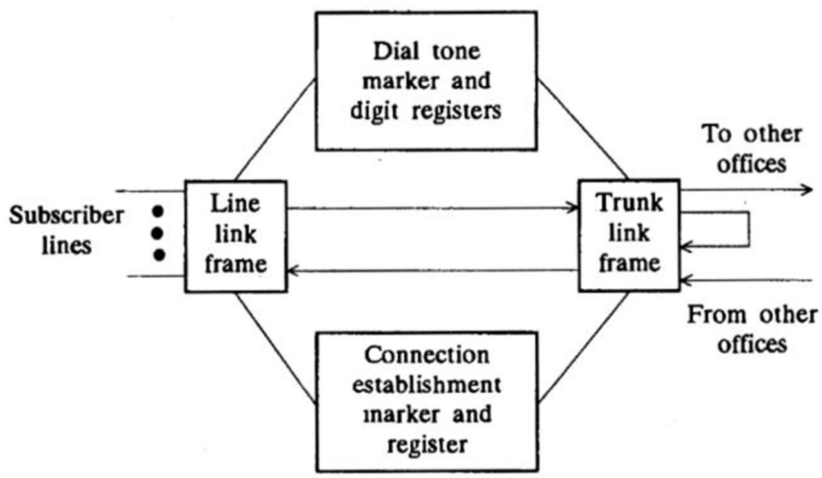

Figure 27. Organization of a cross-bar exchange

In a crossbar exchange, the call processing progresses in three stages, i.e.

(a) Pre-selection,

(b) Group selection, and (c) Line selection.

Pre-selection, which is performed by the originating marker, starts from the moment the subscriber lifts the handset of the telephone and ends when the dial tone is sent out to him by a register.

In group selection stage, the call is switched through to the desired direction. The direction is decided in accordance with the code given by the translator.

In the line selection stage, the calling subscriber is connected to the called subscriber by the terminating marker. The line of the called party is then monitored by the terminating marker until either party (calling or called) hang up.

Circuit Switching

Circuit switching is a switching method in which a dedicated communication path in physical form between two stations within a network is established, maintained and terminated for each communication session. It has basically three phases as circuit establishment, data transfer and circuit disconnect.

Once the connection is established, the data transfer is transparent. The main feature of such a connection is that it provides a fixed data rate channel and both subscribers must operate at this rate, It is considered inefficient compared to packet switching because channel capacity is completely dedicated for duration of connection. If there is no data at any moment of time, channel capacity goes wasted. Moreover, setting up of connection takes time.

Circuit switching has two types of transmissions.

Datagram transmissions - Datagram transmissions have individually addressed frames.

Data-stream transmissions - Data-stream transmissions have a stream of data for which address checking occurs only once. The routing in circuit switching may have either static routing or dynamic routing. In case of static routing, this methodology uses the same approach all the time while dynamic routing allows alternate routing depending on traffic.

The Key Point of Circuit Switching are.

- It is the simplest method of data communication in which a dedicated physical connection or path is established between the sending and receiving device.
- In circuit switched networks, a set of switches are connected by physical links. A connection between two stations is a dedicated path made of one or more links.
- Figure shows a circuit switched network in which computer A, B and C are connected to computer D, E, F and G via four switches. If these computers are to be connected with a point-to-point connections, 12 dedicated lines are required which will incur high line cost.

- The four switches connecting these computers thus provide dedicated links by reducing the line cost. Here I, II, III and IV are the circuit switches or nodes. Nodes I, III, IV are connected to computers while II is only routing node.
- In circuit switching the routing decision is made when path is set up across the network. After the link has been set between the sender and receiver, the information is forwarded continuously over the link.
- The dedicated path established between the sender and the receiver is maintained for entire duration of conversation
- This link or path is released only when data transmission between sender and receiver is over.
- Circuit switching takes place at the physical layer.

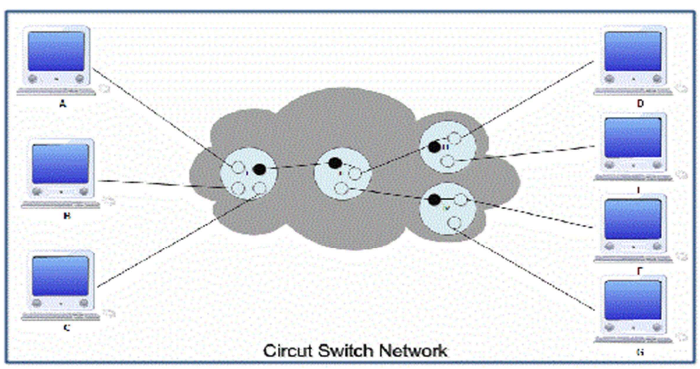

Fig 28. Circuit Switching Network

- Before starting communication, the stations must make a reservation of resources to be used during the communication. These resources can be switch buffers, switch processing time, switch input/output ports. These resources remain dedicated during the entire duration of data transfer.
- Data transferred between the two stations are not packetized (i.e. in form of packets). The data are a continuous flow· sent by the source station and received by the destination station and there may be periods of silence.

- There is no addressing involved in data transfer. The switches route the data based on their occupied band (FDM) or time slot (TDM). However, there is end-to-end addressing used during set up phase.
- In telephone systems circuit switching is used.
- The communication in a circuit switched network takes place in three phases:
  - Circuit establishment or setup phase.
  - Data transfer phase.
  - Circuit disconnects or tears down phase.

Circuit establishment or Setup Phase

- In circuit .switched network, before actual data transfer takes place, a dedicated circuit or path is established between the sender and receiver.
- For example, as shown in fig. if two communicating devices are A and D, then a dedicated path will be set up from A to I, I to II, II to III and III to D first.
- End-to-End addressing (i.e. source and destination address) is required for creating a connection between two end systems.

Data Transfer Phase

- Actual data transfer between the source and destination takes place after the dedicated path is set up between them.
- The data flows are continuous between sender and receiver. There may be periods of silence in between.
- Generally all the internal connections are duplex.

Circuit Disconnect or Teardown Phase

When one of the parties needs to disconnect, a signal is sent to each switch to release the resources.

Circuit Switch

- A circuit switch is a device that creates a temporary connection between an input link and output link.
- A circuit switch usually has n input lines and m output lines i.e. number of input lines and number of output lines may not be equal.

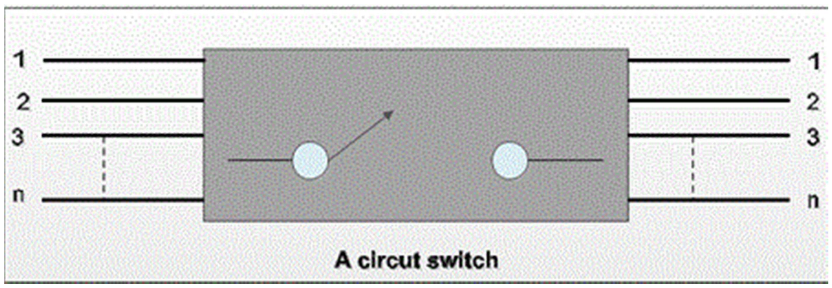

Fig 29. Circuit Switching

The advantages of circuit switching are

- The dedicated path/circuit established between sender and receiver provides a guaranteed data rate.
- Once the circuit is established, data is transmitted without any delay as there is no waiting time at each switch.
- Since a dedicated continuous transmission path is established, the method is suitable for long continuous transmission.

Disadvantages of Circuit Switching

The various disadvantages of circuit switching are:

- As the connection is dedicated it cannot be used to transmit any other data even if the channel is free.
- It is inefficient in terms of utilization of system resources. As resources are allocated for the entire duration of connection, these are not available to other connections.
- Dedicated channels require more bandwidth.

- Prior to actual data transfer, the time required to establish a physical link between the two stations is too long.

Packet Switching:

Packet-switched describes the type of network in which relatively small units of data called packets are routed through a network based on the destination address contained within each packet. Breaking communication down into packets allows the same data path to be shared among many users in the network. This type of communication between sender and receiver is known as connectionless (rather than dedicated). Most traffic over the Internet uses packet switching and the Internet is basically a connectionless network.

In order to transfer the file fast and efficient manner over the network and minimize the transmission latency, the data is broken into small pieces of variable length, called Packet. At the destination, all these small-parts (packets) has to be reassembled, belonging to the same file. A packet composes of payload and various control information. No pre-setup or reservation of resources are needed.

Packet Switching uses Store and Forward technique while switching the packets; while forwarding the packet each hop first store that packet then forward. This technique is very beneficial because packets may get discarded at any hop due to some reason. More than one path is possible between a pair of source and destination. Each packet contains Source and destination address using which they independently travel through the network. In other words, packets belonging to the same file may or may not travel through the same path. If there is congestion at some path, packets are allowed to choose different path possible over existing network.

Packet-Switched networks were designed to overcome the weaknesses of Circuit-Switched networks, since circuit-switched networks were not very effective for small messages.

Advantage of Packet Switching :

- More efficient in terms of bandwidth, since the concept of reserving circuit is not there.
- Minimal transmission latency.
- More reliable as destination can detect the missing packet.

- More fault tolerant because packets may follow different path in case any link is down, Unlike Circuit Switching.
- Cost effective and comparatively cheaper to implement.

Disadvantage of Packet Switching :

- Packet Switching don't give packets in order. Since the packets are unordered, it is needed to provide sequence numbers to each packet.
- Complexity is more at each node because of the facility to follow multiple path.
- Transmission delay is more because of rerouting.
- Packet Switching is beneficial only for small messages, but not suitable for large message.

Modes of Packet Switching :

Connection-oriented Packet Switching (Virtual Circuit) :

Before starting the transmission, it establishes a logical path or virtual connection using signaling protocol, between sender and receiver and all packets belongs to this flow will follow this predefined route. Virtual Circuit ID is provided by switches/routers to uniquely identify this virtual connection. Data is divided into small units and all these small units are appended with help of sequence number. Overall, three phases takes place here- Setup, data transfer and tear down phase.

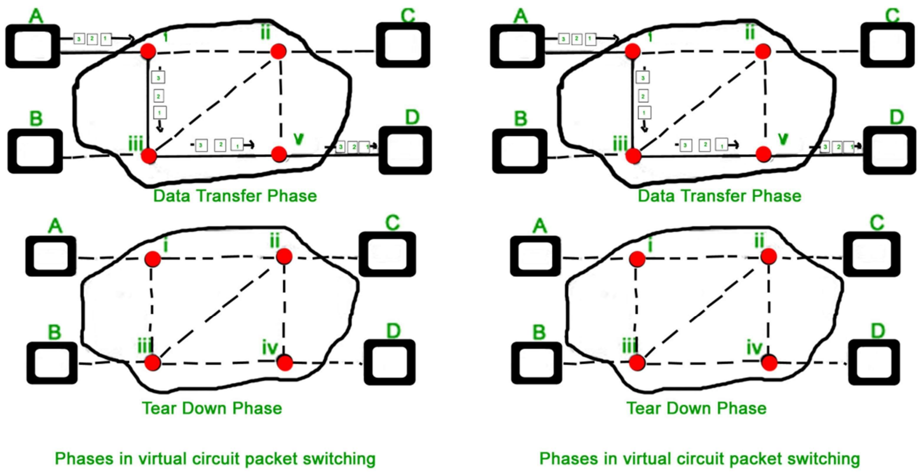

Figure 30. Phases of packet Switching

All address information is only transferred during setup phase. Once the route to destination is discovered, entry is added to switching table of each intermediate node. During data transfer, packet header (local header) may contain information such as length, timestamp, sequence number etc.

Connection-oriented switching is very useful in switched WAN. Some popular protocols which uses Virtual Circuit Switching approach are X.25, Frame-Relay, ATM and MPLS (Multi Protocol Label Switching).

Connectionless Packet Switching (Datagram) :

Unlike Connection-oriented packet switching, In Connectionless Packet Switching each packet contains all necessary addressing information such as source address, destination address and port numbers etc. In Datagram Packet Switching, each packet is treated independently. Packets belongs to one flow may take different routes because routing decisions are made dynamically, so the packets arrived to destination might be out of order. It has no connection setup and tear down phase, like Virtual Circuits.

Packet delivery is not guaranteed in connectionless packet switching, so the reliable delivery must be provided by end systems using additional protocols.

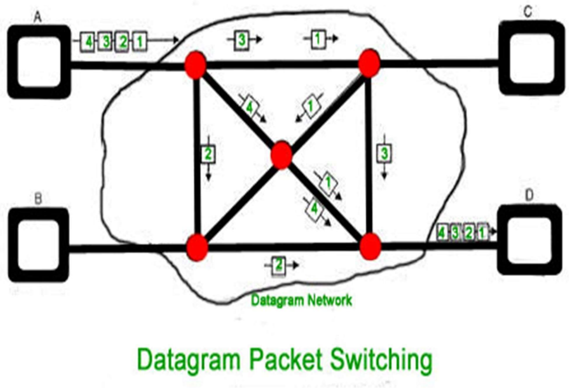

Figure 31. Phases of packet Switching

To send packet from A to B there are delays since this is a Store and Forward network.

Delays in Packet switching :

- Transmission Delay
- Propagation Delay
- Queuing Delay
- Processing Delay

Transmission Delay :

Time taken to put a packet onto link. In other words, it is simply time required to put data bits on the wire/communication medium. It depends on length of packet and bandwidth of network.

Queuing Delay :

Queuing delay is the time a job waits in a queue until it can be executed. It depends on congestion. It the time difference between when packet arrived Destination and when the packet data was processed or executed. It may be caused by mainly three reasons i.e. originating switches, intermediate switches or call receiver servicing switches.

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OCW

PART B(EC504C):

OCW PART B COVERS THE UNDERMENTIONED MODULES:

- Module 3:Digital Switching systems –Concept of Speech Digitisation & Transmission, Time division Time switch, Time multiplexed Space switch, Time multiplexed Time switch, Hybrid switching, ; TS, ST, STS, TST systems.

- Module 4:Telephone Network-Subscriber Loop Systems: BORSCHT Functions; Switching hierarchy & routing, signaling techniques-in channel & common channel signaling, SS7.(onlyBasic Idea) ,Numbering Plan

- Module 5: Stored Program Control: Software architecture, Application software;. Electronic Exchanges Digital PABX

Speech Digitisation & Transmission:

Features of speech signal

- Audio range: 16Hz-20kHz
- Speech frequency range:100Hz-10kHz
- Telephone quality speech:300-3400Hz
- Effect of Channel = Transmission loss = Noise + Interference
- Metric: signal to noise ratio (SNR)

The acceptable level of speech transmission is obtained in the frequency range of 400– 3400 Hz and our ear is sensible to frequencies that are around 3 kHz. In analog speech transmission, one of the shortcomings is noise and interference which is more significant during pauses. In digital transmission, speech and speech pauses are encoded with data pattern and transmitted at a constant power level which ensures better quality, provides higher capacity and deals with longer distance. The main drawback of digital transmission is the requirements of greater bandwidth. Nonetheless, the benefits of digital transmission outweigh the bandwidth consideration.

Digital Transmission :

What is a Digital signal?

A digital signal is a signal that is being used to represent data as a sequence of discrete values; at any given time it can only take on one of a finite number of values. This contrasts with an analog signal, which represents continuous values; at any given time it represents a real number within a continuous range of values.

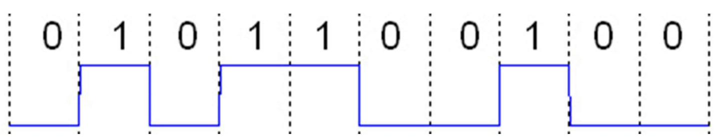


Fig 1:Digital Signal

Analog to digital conversion : Steps: Sampling + quantization + Coding

Sampling: In signal processing, sampling is the reduction of a continuous-time signal to a discrete-time signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal).A sample is a value or set of values at a point in time and/or space.A sampler is a subsystem or operation that extracts samples from a continuous signal.A theoretical ideal sampler produces samples equivalent to the instantaneous value of the continuous signal at the desired points.

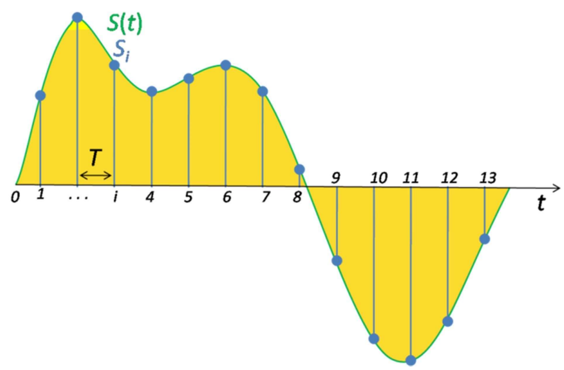


Fig 2 : Sampling

Sampling theorem (Nyquist criterion) „ The minimum sampling frequency required to reconstruct the original waveform from the sampled sequence is given by f s ≥ 2H. f s : sampling frequency or Nyquist rate. H : highest frequency component in the input analog waveform.

Quantization & Coding:

Quantization replaces each real number with an approximation from a finite set of discrete values. Most commonly, these discrete values are represented as fixed-point words. Though any number of quantization levels is possible, common word-lengths are 8-bit (256 levels), 16bit (65,536 levels) and 24-bit (16.8 million levels).

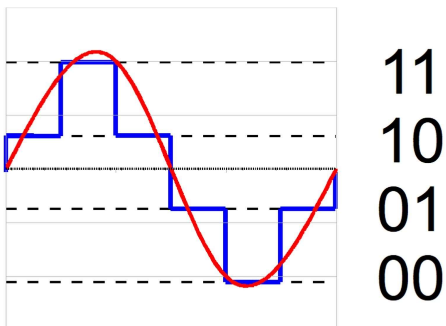


Fig 3: 2-bit resolution with four levels of quantization compared to analog

Time division multiplexing:

Time-division multiplexing (TDM) is a method of putting multiple data streams in a single signal by separating the signal into many segments, each having a very short duration. ... The circuit that combines signals at the source (transmitting) end of a communications link is known as a multiplexer.

1. TDM is the digital multiplexing technique.
2. In TDM, the channel/link is not divided on the basis of frequency but on the basis of time.
3. Total time available in the channel is divided between several users.
4. Each user is allotted a particular a time interval called time slot or time slice during which the data is transmitted by that user.

5. Thus each sending device takes control of entire bandwidth of the channel for fixed amount of time.
6. In TDM the data rate capacity of the transmission medium should be greater than the data rate required by sending or receiving devices.
7. In TDM all the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one-by-one.
8. Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel.
9. The TDM system can be used to multiplex analog or digital signals, however it is more suitable for the digital signal multiplexing.

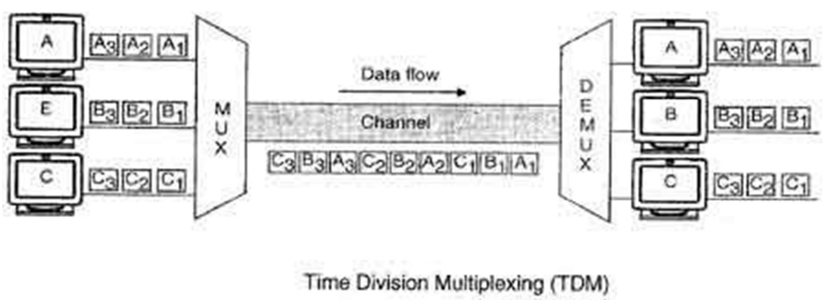


Fig:4 Time Division Multiplexing

TDM is the most commonly used means of subdividing the capacity of a digital transmission facility among a number of sources. Synchronous TDM: repeatedly assigns a portion of the transmission capacity to each source.Asynchronous TDM: assigns capacity as and when needed.

TIME DIVISION SWITCHING

Time division switching involves the sharing of cross points for shorter periods of time. This paves way for the reassign of cross points and its associated circuits for other needed connections. Therefore, in time division switching, greater savings in cross points can be achieved. Hence, by using dynamic control mechanisms, a switching element can be assigned to many inlet-outlet pairs for few microseconds. This is the principle of time division switching.

Time division switching uses time division multiplexing to achieve switching. Two popular methods that are used in time division multiplexing are (a) the time slot interchange (TSI) and (b) the TDM bus. In ordinary time division multiplexing, the data reaches the output in the same order as they sent. But TSI changes the ordering of slots based on the desired connections. The de-multiplexer separates the slots and passes them to the proper outputs. The TDM uses a control unit. The control unit opens and closes the gates according to the switching need. The principle of time division switching can be equally applied to analog and digital signals. For interfacing sampled analog signals but not digitized, the analog time division switches are attractive. But for larger switches, there are some limitations due to noise, distortion and crosstalk which normally

occurs in PAM signals. Thus analog switching is now used only in smaller switching systems. In this section, digital time division switching are described briefly.

Time Division Time Switching

As shown below the Fig. 6 shows the general arrangement of the time division time switching.

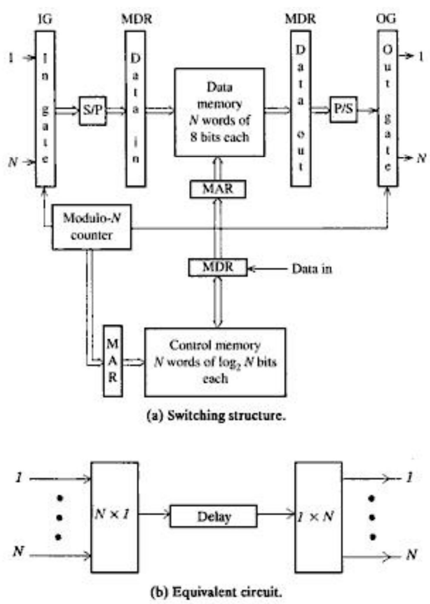


Fig 6: Basic Arrangement of Time Division Time Switching

1. **Sequential write/random read**
2. **Random write/sequential read**
3. **Random input/random output.**

The serial to parallel and parallel to serial converter are used to write the data into the memory and read the data out of memory. For convenience, two MDR are shown, but MDR is a single register. Gating mechanism is used to connect the inlet/outlet to MDR.

The input and output lines are connected to a high speed bus through input and output gates. Each input gate is closed during one of the four time slots. During the same time slot,only one output gate closed. This pair of gates allows a burst of data to be transferred from one input line to a specific output line through the bus. The contol unit opens and closes the gates according to switching need.

The time division time switch may be controlled by sequential write/random read or random write/ sequential read. Fig. 7 depicts both modes of operation and indicates how the memories are accessed to translate information from time slot 2 to time slot 16. Both methods use a cyclic control.

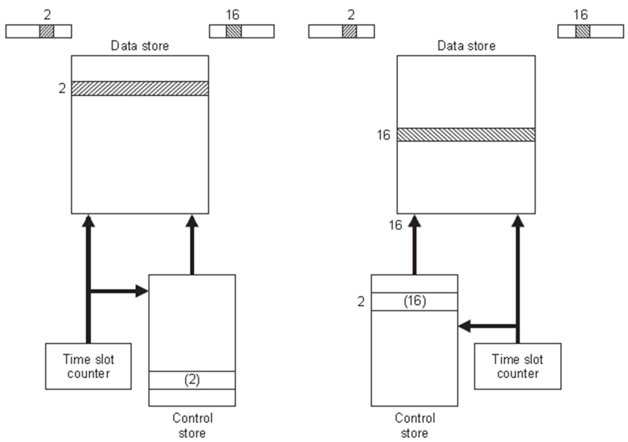


Fig 7: (a) Sequential Write/Random Read (b) Random Write/Sequential Read

Fig. 7 (a) implies that specific memory locations are dedicated to respective channels of the incoming TDM link. Data are stored in sequential locations in memory by incrementing modulo N counter with every time slot. Thus incoming data during time slot 2 is stored in the second location within the memory. On output, information retrieved from the control store specifies which address is to be accessed for that particular time slot. Thus sixteenth word of control store contains the number 2, implying that the contents of data store address 2 is transferred to the output link during outgoing slot 16.

Random write/sequential read mode of operation is opposite to that of sequential write/random read. Incoming data are written into the memory locations as specified by the control store, but outgoing data are retrieved sequentially under control of an outgoing time slot counter.The data received during time slot 2 is written directly into data store address 16 and it is retrieved during outgoing TDM channel number 16.

TIME MULTIPLEXED SPACE SWITCHING

When electronic switch is used, it is called as Electronic Switching System ESS. It became popular when the switch is TMSS.. These switches are required in transit exchanges. Here, the inlets and the outlets are trunks which carry time division multiplexed data streams. A time multiplexed space switch is shown in Fig.

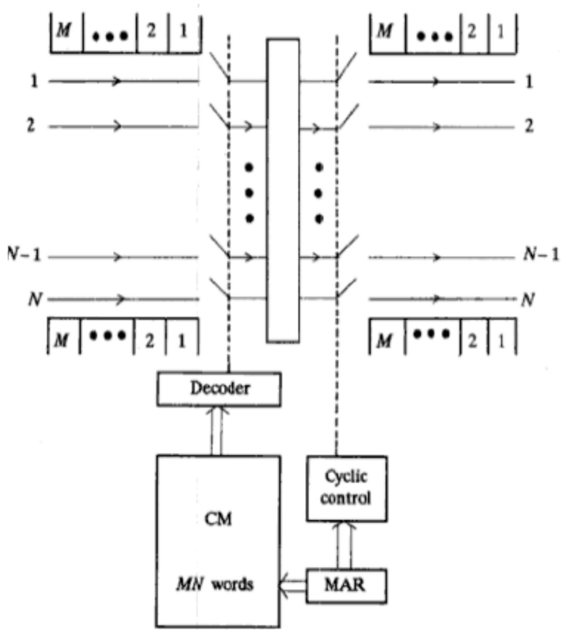


Fig 8: Time Multiplexed Space Switch

There are N incoming trunks and N outgoing traunks, each carrying a time division multiplexed

stream of M samples per frame. For normal telephony, each frame is of 125 us. In one frame duration, a total of MN speech samples have to be switched. One sample duration, usually, 125/M us, is referred to as a time slot. In one time slot N samples are switched.

Time multiplexed switches do not provide full availability. Each incoming trunk carries multiplexed samples form M different voice sources and each stream on the outgoing trunk is demultiplexed to M different destinations. Sources, trunks, destinations have one-to-one time relationship as follows:

1. sources and incomingtrunks time slots

2. outgoing trunk time slots and the destinations

3. time slots of incoming and outgoing trunks

The sample of the source is always carried in time slot I of the inlet and time slot j of the outlet is always demultiplexed to destination j. the time slot k of any incoming trunk is transferred to

time slot k on any outgoing trunk.As a result, a voice sample of slot Ifrom any inlet cannot be transferred to slot k of any outlet (i!= k). a sample from input slot I can only be transferred to destination I of one or more outlets. In other words, interchange of samples among different time slots is not possible. Thus,the switch doesnot provide full availability.

In every time slot, up to N or M samples are switched simultaneously. The control store has N addresses corresponding to N vertical outputs with each address selecting one gate in each vertical output. The size of the control memory is N and its width . Because of the parallel transfer of N or M data samples in each time slot, a large number of channels can be multiplexed per input line. The performance of this switch is similar to that of the time multiplexed space switch with N control memory modules.

TIME MULTIPLEXED TIME SWITCHING:

In the 1960s Bell Labs introduced the Time switch or Digital switch. This switch is nothing but a delay device. So a memory can be used as a switch.

Time multiplexed time switches permit time slot interchange (TSI) of sample values. In TSI, a speech sample input during one time slot maybe sent to the output during a different time slot. Such an operation necessarily implies a delay between the reception and the transmission of a

sample. The switch is organised in the sequential write/random read fashion. The time slot duration t_{TS} is given by t_{TS}= 125/M

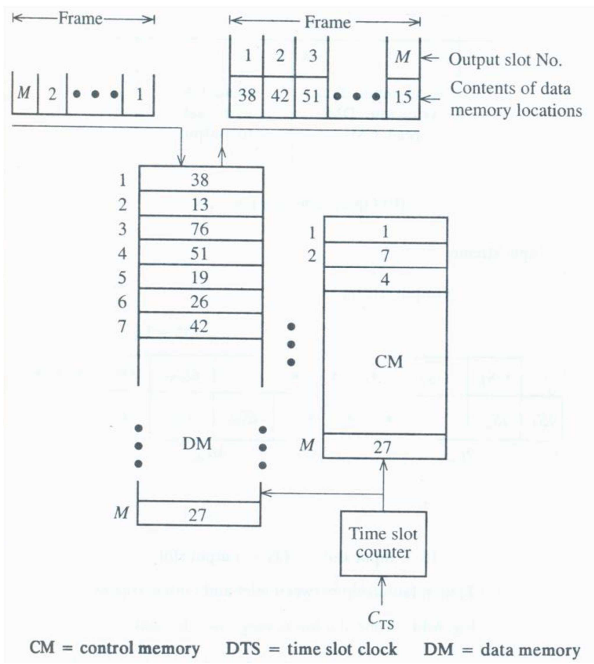


Fig 9: Principle of Time Slot Interchange

The time slot clock runs at the time slot rate, i.e. at the rate of one pulse every 125/M microseconds. The time slot counter is incremented by one at the end of each time slot. The contents of the counter provides location addresses for the data memory and the control memory; Data memory and control memory accesses take place simultaneously in the beginning of the time slot. Thereafter, the contents of the control memory are used as the address of the data memory and the data read out to the output trunk. The input sample is available for reading in at the beginning of the time slot and the sample is ready to be clocked in on the output stream at the end of the time slot. Even if there is no time slot interchange, a sample is delayed by a minimum of one time slot in passing from, the input stream to the output stream because of the storage action. In other words, a time-slot switch may be considered to have an inherent time delay of one time slot. In effect, the output stream is delayed by t_{TS} microseconds when compared to the incoming data stream. Depending on the output time slot to which an input slot contents are switched, the sample experiences a delay in the range of t_{TS} to M t_{TS} microseconds. Since there are no switching elements in this

configuration, the cost is equal to the number of memory locations. There are M locations each in the control and in the data memory. Therefore, the cost is given by C = 2M units.

A TSI switch may be designed to be expanding or concentrating. In such switches, the number of time slots (samples) per frame in the input stream and in the output stream are different. If we represent these numbers as M1and M2 respectively, the switch is expanding when M2 > M, vice-versa. The bit rates of the input and output streams are also different. For an expanding switch, the output bit rate is higher and for a concentrating switch, the input bit rate is higher.

In practical configurations of time multiplexed time switches, there are N time multiplexed input streams each multiplexing M subscribers, and there are Ntime multiplexed output streams each carrying M subscribers. The problem is to handle NM subscribers in the time duration of 125 microsecond. This can be done in four different ways:

1. Serial-in/serial-out
2. Parallel-in/serial-out
3. Serial-in/parallel-out
4. Parallel-in/parallel-out.

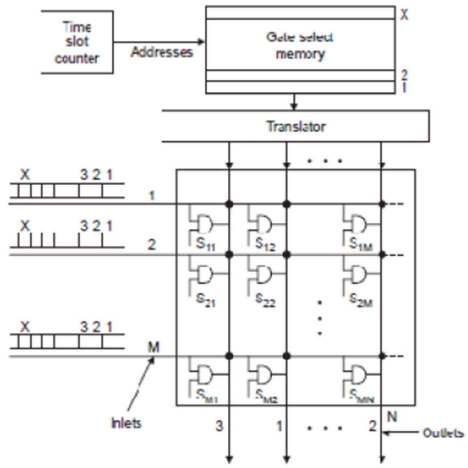
- DIGITAL OR TIME SWITCH IS LIKE A DIARY WHERE WE WRITE SOMETHING TODAY AND READ IT AGAIN ON A LATER TIME.

HYBRID SWITCHING:

Space and Time Switches

Space switch. Space switch: Fig. 12 shows a typical space switch. It uses a space array to provide switching generally the space switch consists of a matrix of M × N switching points where M is number of inlets and N is number of outlets. A connection between an inlet and an outlet is made by the simple logic gates (AND gates). As logic gates are unidirectional, two paths through switching matrix must be established to accommodate a two way conversation. The logic gate array can serve for concentration, expansion or distribution depending on M is larger, smaller or equal to N. Fig. 8 shows only one voice direction. However, the corresponding components are available for the opposite direction too. A number of M, of X slot multiplexers, provide the inputs and the outlets are connected to N, X slot de-multiplexers. The gate select memory has X locations. The word containing information about which cross point is to be enabled is decoded by the translator. During each internal time slot, one cross point is activated. In the shift to the

next interval time slot, the control memory is incremented by one step, and a new cross point



pattern is formed in the matrix

Fig 10. Space switch.

Time switch. The time-slot interchange (TSI) system is referred to as time switching (T-switching). Fig. 11 shows the block diagram of time switch.

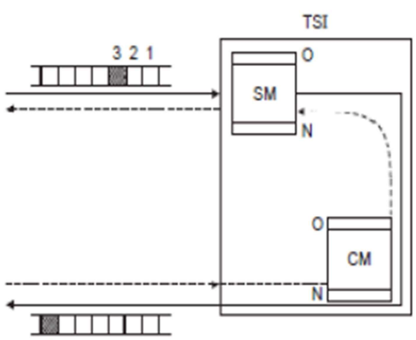


Fig 11: Time switch.

Each incoming time slot is stored in sequence in a speech memory (SM). The control memory (CM) determines in which order the time slots are to be read from SM. This means that a voice sample may be moved from say incoming time slot 3 to outgoing time slot 1.

Time-space (TS) Switching:This switch consists of only two stages. This structure

contains a time stage T followed by a space stage S as shown in Fig. . Thus this structure isreferred to as time-space (TS) switch. The space arrays have N inlets and N outlets. For eachinlet line, a time slot interchangerwith T slots is introduced. Each TSI is provided with a timeslot memories (not shown). Similarly a gate select memory needs to be provided for the space array (not shown).

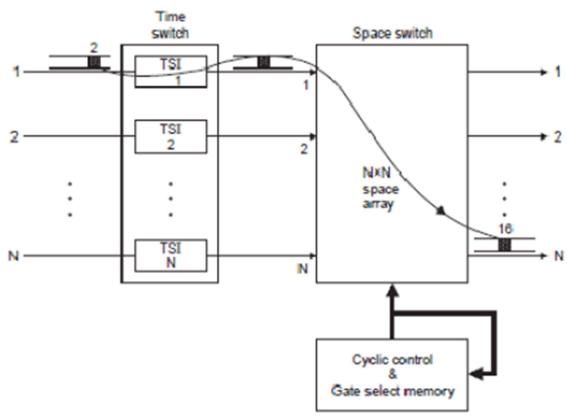


Fig 12: Time space (TS) switching.

The transmission of signals carried out from sender to receiver through multiplexer input and demultiplexer output. The reverse communication also similar. Thus a hybrid arrangement is needed to isolate the transmitted signal from the received signal. The basic function of the time switch is to delay information in arriving time slots until the desired output time slot occurs.Let the communication is to take place between subscriber A and B. Let A is assigned time slot 2 and line 7 and subscriber B is assigned time slot 16 and line 11. Then the signal moved from time slot 2 to time slot 16 by the time-slot exchanger and is transfered from line 7 to line 11 in the space array. Similarly, the signal originated by B is moved from slot 16 to slot 2 through line 11 to 8.

The cyclic control and gate select memory contains the information needed to specify the space stage configuration for each individual time slot of a frame. The time stage have to provide

decays ranging from one time slot to a full frame. During each outgoing time slot, control information is accessed that specifies interstage link number to output link. During other time slots, the space switch is completely reconfigured to support other connections.

Let each time slot interchanger have T slots. If the space array is a N × N, then the simultaneous connections possible is NT. If T = 128 and N = 16, 2048 connections can be supported. This structure is not free of blocking. The control store is a parallel end around shift resister. If space array is at the inlet side and time switch is at the output side, the structure is refered as space time (ST) switching. Both TS and ST arrangements are equally effective.

TS system is used in DMS 100 digital switching system developed in canada (1979). It handles

61000 trunks and accomodates 39000 trunks.

STS & TST Switching:

The TS structure is of blocking nature. Let A and B are the subscribers using different time slot on the same line want to connect to two subscribers C and D using same time slot on different lines. A and B can be moved to the same time slot but during that time slot, the inlet line can be connected to C's line or D's line but not both. This is the significant limitation of the structure. Moreover, time stage switching is generally less expensive than space stage switching as digital memory is much cheaper than digital cross points (AND gates). The multiple stages overcome the limitations of the individual switches and cost savings can also be achieved. TST, STS,TSST, TSSSST and TSTSTSTSTSTS are the switching system configurations used in digital switching system. However, the TST structure is the most common.

STS Switching:

In STS switching, the time stage is sandwiched between two space arrays. The digital switching system ITS 4/5 of USA (1976) uses the STS switching configuration. It handles 3000 trunks and accommodates 1500 Erlangs of traffic. Fig. 15 shows the space-time- space (S-T-S) switching network for M incoming and outgoing PCM highways. Establishing a path through an STS switch requires finding a time switch array with an available unit's access during the incoming time slot and an available read access during the desired outgoing time slot. The input side space stage as well as the output side space stage is free to utilize any free time switch modules. In the diagram shown in Fig. 11, the time slot 2 is connected to the TSM 2 where the time slot allotted

is 16 and passed to the (M – 1)th line of output space array. Thus the path is provided. This structure is of non-blocking nature.

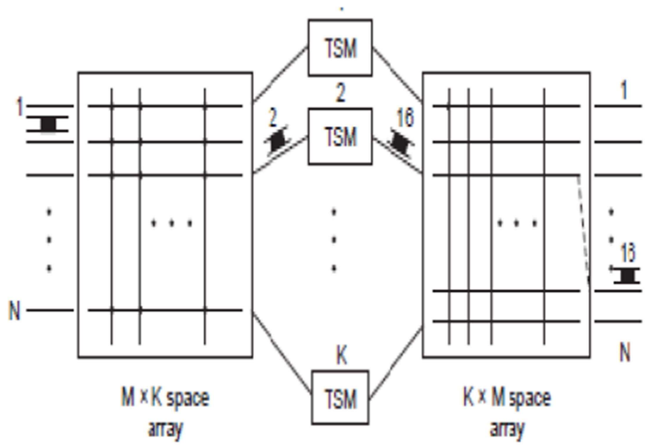


Fig 13: STS Switch

Blocking probability: The STS switch is identical to the probability graph of three stage space switches. Similar to that, the blocking probability of an STS switch is

$$\mathbf{B} = \left[1 - \left(1 - \frac{p}{\beta}\right)^2\right]^K$$

–

Where p = probability that a link is busy

N = is the factor by which the percentage of links that are busy is reduced. (_< 1)

K = number of center stage TSM.

TST SWITCHING:

In TST switching the space stage is sandwiched between two time stage switches. Of all the multistage switching, TST is a popular one.

Some important features of TST switches are

(i) Low blocking probability. An incoming channel time slot may be connected to an outgoing channel time slot using any possible space array time slot. Thus there are many alternative paths between two subscribers. This concept reduces the blocking probability of a three stage combination switch.

(ii) Stage independency. The space stage operates in a time-divided fashion, independently of the external TDM links. The number of space stage time slots L does not coincide with the number of external TDM time slots T.

(iii) Implementation advantage. The factors to be considered for switching design and implementation are traffic loads, modularity, testability, expandability and simple control requirements. For large switches with heavy traffic loads, the TST have good implementation advantage.

(iv) More cost effective. If the input channel loading is high, the time expansion of TST and

space expansion of STS are required. Time expansion of TST can be achieved at less cost than space expansion of STS. In comparison with STS, the TST have certain limitations. For small switches, the STS architectures are less complex to implement than TST. The control requirements of STS are simpler than TST.The principle of operation of TST switching is shown in Fig. below. In figure, two flows of time slots, one for each direction are connected together.

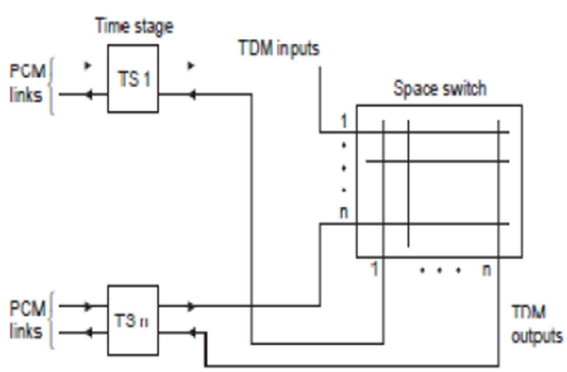


Fig 14: Principle of TST Switching

The functional block diagram which explains the transfer of signals from inlet to outlet is shown in above Fig. 14. The information arriving at the incoming link of TDM channel is delayed in the· inlet times stage until an appropriate path through the space stage is available. Then the information is transferred through the space stage to the appropriate outlet time stage. Here the information is held until the desired outgoing time slot occurs. Any space stage time slot can be used to establish a connection. The space stage operates in a time divided fashion, which is Independent of the external TDM links. There are many alternative paths between a prescribed input and output unlike a two stage network which has only one fixed path.

SUBSCRIBER LOOP SYSTEM:

In telephony, the local loop (also referred to as a local tail, subscriber line, or in the aggregate as the last mile) is the physical link or circuit that connects from the demarcation point of the customer premises to the edge of the common carrier or telecommunications service provider's network.

One end of each subscriber loop is terminated on a Main Distribution Frame (MDF) at the exchange. The drop wires (DW) from the telephones are connected to the distribution point (DP) which is located near the subscriber's premises. The distribution points at various locations are

connected together by a distribution cables (DC) and terminated to the feeder points (FD). The DC carries 10-500 pair of wires. Many feeder points related to a particular geographical area connected by a branch feeder (BF). From BF, through main feeder, all the subscriber loops are connected to MDF at the end office or local exchange. Typically, the MF carries 1002000 pair of wires to MDF. For the purpose of flexible interconnection such as transfer from location to other location or within the geographical area, the subscriber pair and exchange pairs are interconnected at the MDF by means of jumpers.

Fundamental Characteristics:

The subscriber loop is the most common interface in the network. The fundamental characteristic of this interface are.

Battery. To enable dc signalling and to provide bias current for carbon microphone, a battery of about 48 V is connected to subscriber loop at exchange.

Overvoltage protection. Protection of equipment and personal from lightning strikes and power line induction or shots.

Ringing. Application of a 20 Hz signal at 86 V rms for ringer excitation.

Supervision. Supervise the network by detecting the off hook/on hook and flow/no-flow dc current.

Coding. In the case of digital end office, analog to digital coding and digital to analog decoding functions necessary.

Hybrid. For two wire to four wire conversion, hybrid in necessary. Test. Line test toward the subscriber disconnection of the switch.

The first letter of the above characteristics are coined together which is commonly known as BORSCHT.

SWITCHING HIERARCHY & ROUTING:

Switching networks require some form of interconnections of switching exchanges to route traffic effectively and economically.Exchanges are interconnected by groups of trunk lines usually known as trunk groups.

The basic topologies for interconnecting exchanges are:

1. Mesh

2. Star

3. Hierarchy

Mesh topology is one which has all of the workstations connected to each other. This topology is typically only used when high availability is a requirement. It is expensive to maintain and troubleshoot.

Star topology is the most popular network topology in businesses today. It consists of nodes connected to a central switch or hub. If you have a home network, you are probably using the star topology.A Star connection utilizes an intermediate exchange called a TANDEM Exchange.

Hierarchical networks are capable of handling heavy traffic where required and use minimal number of trunk groups. As shown in the fig below , in a strictly hierarchical network, traffic from subscriber A to subscriber B and vice versa flows through the highest level of hierarchy.

A traffic route via the highest level of hierarchy is called final route.

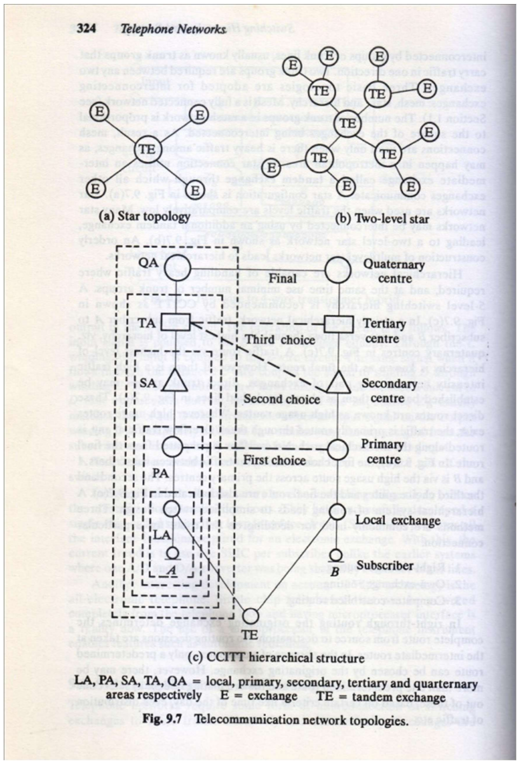


Fig 15: Telecommunication Network Topologies

IN CHANNEL SIGNALLING & COMMON CHANNEL SIGNALLING:

IN CHANNEL SIGNALLING:

- Uses same chaneel for traffic and call
- Requires no additional transmission facilities
- In Band Signalling
    - Uses same frequencies as voice call
    - Can go anywhere a voice signal can
    - Impossible to set up a call in a faulty speech path

COMMON CHANNEL SIGNALLING:

In telephony, common-channel signaling (CCS), in the US also common-channel interoffice signaling (CCIS), is the transmission of signaling information (control information) on a separate channel than the data, and, more specifically, where that signaling channel controls multiple data channels.

Common channel signaling (CCS) is signaling in which a group of voice-and-data channels share a separate channel that is used only for control signals. This arrangement is an alternative to channel associated signaling (CAS), in which control signals, such as those for synchronizing and bounding frames, are carried in the same channels as voice and data signals.

For example, in the public switched telephone network (PSTN) one channel of a communications link is typically used for the sole purpose of carrying signaling for establishment and tear down of telephone calls. The remaining channels are used entirely for the transmission of voice data. In most cases, a single 64kbit/s channel is sufficient to handle the call setup and call clear-down traffic for numerous voice and data channels.

The logical alternative to CCS is channel-associated signaling (CAS), in which each bearer channel has a signaling channel dedicated to it.

SS7

SS7 concepts:

SS7 defines a signaling network that is made of signaling points (network nodes) and signaling links. In practice most times a signaling link is a PCMtimeslot4.Among the signaling points from a call perspective are the originating point (OP) and the destination point (DP). Obviously, addresses of the OP and DP need to appear in signaling messages. These addresses are

called signaling point codes. OPC stands for Originating Point Code and DPC for Destination Point Code.

SS7 also specifies that intermediate signaling points that do not process the call itself may appear on the way from the OP to the DP. These intermediate points are called Signaling Transfer Points or STPs and work either on MTP (message transfer part) or SCCP levels. SCCP stands for Signaling Connection Control Part.

STPs can be seen as signaling message routers. They are useful e.g. in a large country for the purpose of concentrating or aggregating signaling from a large number of local or transit exchanges to a small set of Intelligent Network (IN) Nodes. We will come back to IN later on this course. Such transfer points are also useful for the purpose of implementing mobility management and global roaming.

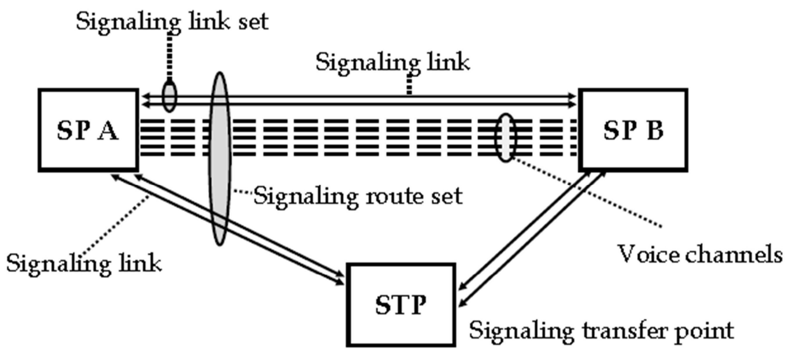


Fig 17: Principle of SS7

A signaling link is a pre-configured connection between two (adjacent) signaling points. A signaling link set is a set of signaling links such that all links have the same signaling end points.

A sequence of signaling link sets between two end points forms a signaling route. A signaling route differs from a link set because of the possible intermediate STPs. The set of all signaling routes connecting two signaling points is a signaling route set.

Numbering Plan:

A telephone numbering plan is a type of numbering scheme used in telecommunication to assign telephone numbers to subscriber telephones or other telephony endpoints. Telephone numbers are the addresses of participants in a telephone network, reachable by a system of destination code routing.

National Numbering Plan can be divided into area code,Exhange code and line number.

Area/Trunk Code	Exchange Code	Line Number
1-3 digits	1-3 digits	2-4/5 digits

- Mobile Station International Subscriber Directory Number (MSISDN) is a number used to identify a mobile phone number internationally. MSISDN is defined by the E.164 numbering plan. This number includes a country code and a National Destination Code which identifies the subscriber's operator.

Example: MISDN: +91-7381730900

CC(Country Code)- +91 (INDIA)

NDC(National Destination Code)-7381(Odissa)

SN (Subscriber Number)-730900-Prakash Das- Subcriber Code/Subscriber No.

STORED PROGRAM CONTROL

Organization of SPC:

CENTRALIZED SPC

It finds broadly application in early SPC switching systems.

DISTRIBUTED SPC

It is Gaining popularity in modern switching systems.

Early electronic switching systems are centralized SPC exchanges and used a single processor to perform the exchange functions. Presently centralized exchanges uses dual processor for high reliability.

Concept

- All the control equipment is replaced by a single powerful processor.

- Configuration of centralized SPC
- Typical organization
- Redundant configuration

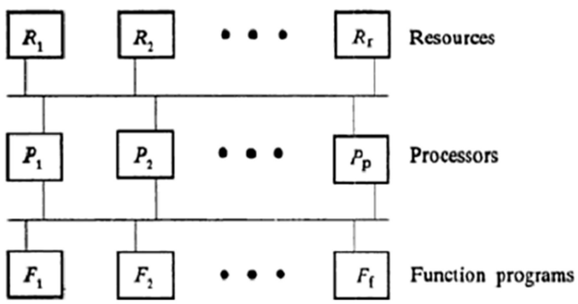


Fig 18 : A redundant Centralized SPC Control Structure

Operation modes in redundant configuration (e.g. dual processor)

- Standby mode
- Synchronous duplex mode
- Load sharing mode

STANDBY MODE
How does it work?

In this mode, any one of the processors will be active and the rest is standby. The standby

processor is brought online only when the active processor fail. This mode of exchange uses a secondary storage common to both processors. The active processor copies the status of the system periodically and stores in axis secondary storage. In this mode the processors are not connected directly. In secondary storage, programs and instructions related to the control functions, routine programs and other required information are stored.

All processors have the same capability to control the switching procedure.

One processor is active and the other is on standby, both hardware and software wise.

The standby processor is brought online only when the active processor fails.

How does the standby processor take over the control properly?

State of the exchange system should be clear to the standby processor as its starting point.

Reconstitution of the state:

Scanning:

- The standby processor scans all status signals as soon as it is brought into operation.
- Only the calls which are being established at the time of failure are disturbed.
- Only feasible for small exchanges.
- Shared secondary storage: popular. Shared secondary storage:

- The active processor copies system status into a secondary storage periodically, say every 5 seconds.
- As soon as a switchover occurs, the online processor loads the most recent update of the system status from the secondary storage and continues the operations. Only the calls which changed status between the last update and the failure are disturbed.
- Feasible for large exchanges

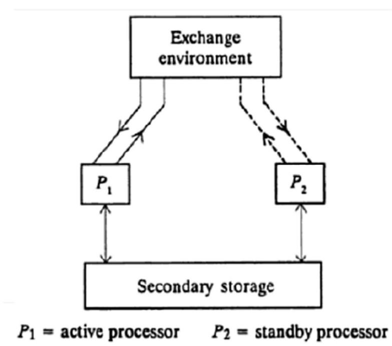


Fig 19 : Standby Dual Processor Configuration

Synchronous duplex mode:

How does it work?

In this mode, the processors p1 and p2 are connected together to exchange instructions and controls. Instead of a secondary storage common to P1 and P2, separate memory M1 and M2 are used. These processors are coupled to exchange stored data. This mode of operation also uses a comparator in between p1 and p2. The comparator compares the result of the processors. During normal operation, both processor receives all the information from the exchange and

receives related data from their memories. Although only one processor actually controls thee xchange and remaining is in synchronism with first one. If a mismatch occurs, the fault is identified by the comparator, and the faulty processor is identified by operating both individually. After the rectification of fault, the processor is brought into service.

- Both two processors execute the same set of instructions.
- One of the processor actually controls the exchange.
- The results from two processors are compared continuously by a comparator.
- If the results match, the system works normally. Otherwise, a fault occurs, a checkout program is run independently in both two processors to determine which one is faulty.
- The faulty processor is taken out of service, and the other one works independently.

Synchronous duplex mode

- In case of transient failure of comparator, there are three possibilities exist:

- Continue with both processors.
- Take out the active processor and continue with other processor.
- Continue with the active processor but remove the other processor from service.

Load sharing mode

How does it work?

- Both two processors have access to entire exchange environment. Each of them has independent memories for redundancy purpose.

- Both two processors are active simultaneously and share the load and the resources dynamically.

- An incoming call is assigned randomly or in a predefined order to one of the processors which then handles the call right through completion.

- Inter-processor links are configured for processors to exchange information needed for mutual coordination and verifying the 'state of health' of the other.

If a processor fails, the other processor takes over the entire load including the calls already set up by the failing processor.

Exclusion mechanism in resource sharing

The processors should not seek the same resource at the same time.

Implementation: hardware & software.

In this mode, the comparator is removed and alternatively an exclusion device (ED) is used.

The processors call for ED to share the resources, so that both the processors do not seek the same resource at the same time. In this mode, both the processor are active simultaneously and share the resources of exchange and the load dynamically. If one processor fails, with the help of ED, the other processor takes over the entire load of the exchange. Under normal operation, each processor handles one half of the calls on a statistical basis. However the exchange operator can vary the processor load for maintenance purpose.

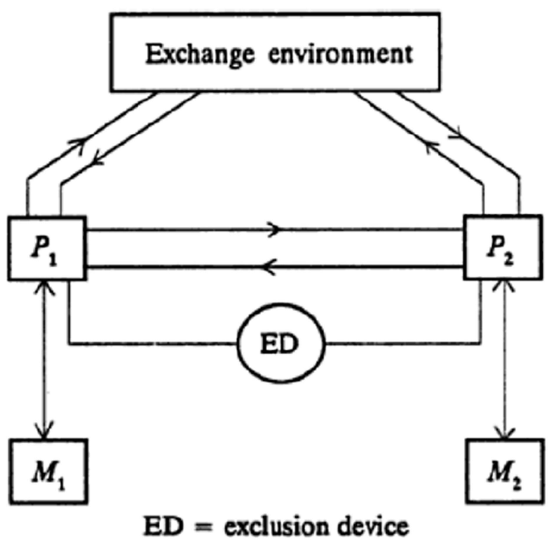


Fig 20 : Load sharing Mode

Distributed SPC:

CONCEPT OF DISTRIBUTED SPC

The control functions are shared by many processors within the exchanges.

Background

Low cost processors Advantages

• Better Availability

• Better Reliability

Decomposition of Control Functions

• Vertical decomposition

The exchange environment is divided into several blocks.

Each block is assigned to a processor.

Horizontal decomposition

- The control functions are divided into groups, e.g. event monitoring, call processing, and O&M functions.
- Each processor performs only one or some of the exchange control functions.
- A chain of processors are used to perform the entire control of the exchange.
- The entire chain may be duplicated to provide redundancy.

SOFTWARE ARCHITECTURE:

Software is basically two types:

i. System Software (Operating System) ii. Application Software

(Software based on Operating System)

Therefore a Special Design and Development is to be done for Switching Operating System

PROCESS: an instruction executed by the processor is commonly called as a "Process"

RUNNING PROCESS: an instruction is currently executing by the processor READY

PROCESS: next instruction of running process and an instruction timed out isnormally

called as a Ready Process

BLOCKED PROCESS: a Process or instruction is said to be blocked when it is condition alike “if”, “while” because the execution of this instructions are depending on the results of the conditional statements. The below diagram will depict the state transitions between the “Running Process”, “Ready Process” & “Blocked Process”

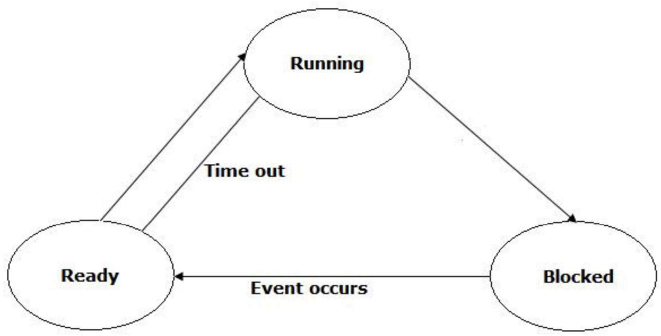


Fig 21: Precess States & Transition

Process Control Block: Each Control Process is represented by the operating system by a “Process Control Block (PCB)” which is a data structure containing the following information

about the process:

- Current State of the Process
- Process Priority & CPU Scheduling Parameter
- Memory Allocated To Process
- Status of events and I/O resources associated with the process.
- Program Status Word which contains the address of the next instruction to be executed.

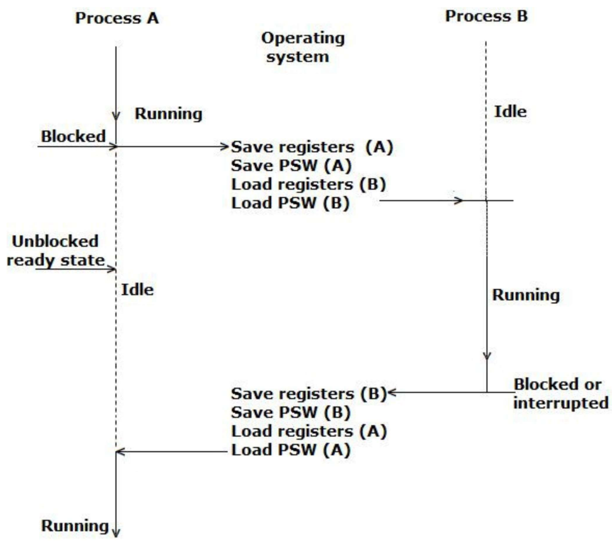


Fig 22: Process Switching

The above diagram depicts the process switching control of an operation system depending upon priority .

Critical Region:when numbers of parallel processes are running by the operating system,any time, any process may access common resources like memory space.

“When a process is accessing a common resource in any time of its execution, then the process is

said to be in “Critical Region”.

Semaphore:in order to avoid the problem of accessing any two or more processesare in critical state and to avoid “Deadlock” a variable “Semaphore” used.

Semaphore contain a number (which is equal to the number processes of accessing the common resources or be in critical state) by accessing this number operating system can manage between different processes in “Critical State” and by which, “Deadlock” is avoided.

DIGITAL PABX:

A private automatic branch exchange (PABX) is an automatic telephone switching system within a private enterprise. Originally, such systems - called private branch exchanges (PBX) - required the use of a live operator. Since almost all private branch exchanges today are automatic, the abbreviation "PBX" usually implies a "PABX."

Questions:

1. Deascibe different modes of Centralised SPC

2. Design of a subscriber loop system

3. Different types of signaling techniques that are used in a telecommunication network.

4. Routing protocol used in a telecommunication network.

5. Describe Switching hierarchy in a telecommunication network

EC504C: TELECOMMUNICATION ENGINEERING

OCW

PART C(EC504C):

OCW PART C COVERS THE UNDERMENTIONED MODULES:

- Module 6:Traffic Engineering: Blocking network, blocking probability, grade of service, traffic load, Erlang-B and C congestion formulas

- Module 7: Broad band transmission ISDN, DSL and ADSL, ISDN and B-ISDN

- Module 8:IP Telephony: Voice over IP, Session initiation protocol

- Module 9:Optical Network – SONET , SDH (Basic Idea , Transmission Media and Calculation of Speed)

TRAFFIC ENGINEERING:

Telecommunications traffic engineering, teletraffic engineering, or traffic engineering is the application of traffic engineering theory to telecommunications. Teletraffic engineers use their knowledge of statistics including queuing theory, the nature of traffic, their practical models, their measurements and simulations to make predictions and to plan telecommunication networks such as a telephone network or the Internet. These tools and knowledge help provide reliable service at lower cost.

The field was created by the work of A. K. Erlang for circuit-switched networks but is applicable to packet-switched networks, as they both exhibit Markovian properties, and can hence be modeled by e.g. a Poisson arrival process.

The crucial observation in traffic engineering is that in large systems the law of large numbers can be used to make the aggregate properties of a system over a long period of time much more predictable than the behaviour of individual parts of the system.

1. Busy Hour – A continuous one hour period during which the traffic volume or number of call attempts is greatest.

2. Peak Busy Hour – The busy hour observed over a 24-hour period. Peak busy hour varies from day to day.

3. Time Consistent Busy Hour – The 1-hour busy hour starting at the same time each day for which the traffic volume or number of call attempts is greatest over the number of days under observation.

4. Call Completion Rate (CCR) is the ratio of the number of successful calls to the number of call attempts. □□□ = □□□□□□□□□□□□□□□□□□□□□□□□□□□□□□□

CCR is used to dimension the network capacity. Most telecommunication networks are designed with a CCR of over 70%.

5. Busy Hour Calling Rate (BHCR) is the average number of calls originated by a subscriber during the busy hour. □□□□ = □□/ □□□□□□□□□□□□□□□□□□□□□□□□□□□□□□

Traffic Intensity = □□□□□□□□□□□*h*□□*h*□*h*□□□□□□□□□□□□□□□□□□□□□/□□□□□□□□□□□□□□□□□□□□□□□□□□□□□□

Blocking in telecommunication systems is when a circuit group is fully occupied and unable to accept further calls [1]. It also referred to as congestion. Due to blocking in telecommunications systems, calls are either queued (but not lost) or are lost (all calls made over congested group of circuits fail). Such systems are called queuing systems (delay systems) and lost-call systems respectively.

- An example of a queuing system: a message-switched exchange

- An example of a lost-call system: a circuit-switched exchange

The proportion of calls that are lost or delayed during blocking portray the measure of the grade of service which is basically the measure of the service provided. A large grade of service indicates a poor service offered to the customer. The grade of service is always specified at the busy hour.

The grade of service (B) in a lost-call system is defined as

B = Number of lost calls / Number of offered calls

B may also be defined as

- B = Lost traffic / Offered traffic
- B = Proportion of time in which congestion exists
- B = Probability that a call will be lost through congestion

There are acceptable grade of service standards for different telecommunication systems . Values which are lower than the stipulated values imply the systems offer poor service.

- 0.001 for cheap tie line circuits
- 0.002 for within building inter-exchange connections
- 0.01 for expensive international circuit groups
- 0.02 for cellular circuit groups

The grade of service is the blocking probability. A higher grade of service implies high probability of loss during the busy hour. Blocking probability is the chance that a customer will be denied service due to lack of resources. A blocking probability of 0.01 means 1% of customers will be denied service. It should be as low as possible and can be decreased by

1. Increasing resources in the system
2. Offering incentives and discounts during off-peak hours to encourage usage of resources outside the busy hour.

Two formulae are used for calculating the blocking probability: the Erlang-B and Erlang-C. The choise of formula is dependent upon the method of handling of customers when all resources are busy.

- Erlang-B: used for lost-call systems whereby calls are lost should all resources be busy.
- Erlang-C: used for queuing systems whereby calls are queued should all resources be busy.

The Erlang-B formula is:

$$P_B = \frac{\frac{A^N}{N!}}{\sum_{i=0}^{N} \frac{A^i}{i!}} \quad \text{...............(1)}$$

where

A is the total traffic offered in units of Erlangs
N is the number of circuits
P_B is the probability that a customer's request will be rejected due to lack of resources.

The Erlang-C formula is:

$$P_c = \frac{\frac{A^N}{N!}\frac{N}{N-A}}{\sum_{i=0}^{N-1} \frac{A^i}{i!} + \frac{A^N}{N!}\frac{N}{N-A}} \quad \text{...............(2)}$$

where:

A is the total traffic offered in units of Erlangs
N is the number of circuits
P_c is the probability that a customer has to wait for service

Grade of service

In telecommunication engineering, and in particular teletraffic engineering, the quality of voice service is specified by two measures: the grade of service (GoS) and the quality of service (QoS).

Grade of service is the probability of a call in a circuit group being blocked or delayed for more than a specified interval, expressed as a vulgar fraction or decimal fraction. This is always with reference to the busy hour when the traffic intensity is the greatest. Grade of service may be viewed independently from the perspective of incoming versus outgoing calls, and is not necessarily equal in each direction or between different source-destination pairs. "Grade of Service" sometimes means a measure of inbound call center traffic to verify adherence to conditions to measure the success of customers served.

On the other hand, the quality of service which a single circuit is designed or conditioned to provide, e.g. voice grade or program grade is called the quality of service. Quality criteria for such circuits may include equalization for amplitude over a specified band of frequencies, or in the case of digital data transported via analogue circuits, may include equalization for phase. Criteria for mobile quality of service in cellular telephone circuits include the probability of abnormal termination of the call.

When a user attempts to make a telephone call, the routing equipment handling the call has to determine whether to accept the call, reroute the call to alternative equipment, or reject the call entirely. Rejected calls occur as a result of heavy traffic loads (congestion) on the system and can result in the call either being delayed or lost. If a call is delayed, the user simply has to wait for the traffic to decrease, however if a call is lost then it is removed from the system.

The Grade of Service is one aspect of the quality a customer can expect to experience when making a telephone call. In a Loss System, the Grade of Service is described as that proportion of calls that are lost due to congestion in the busy hour. For a Lost Call system, the Grade of Service can be measured using Equation 1.

$$\text{Grade of Service} = \frac{\text{number of blocked calls}}{\text{number of offered calls}} \qquad (1)$$

For a delayed call system, the Grade of Service is measured using three separate terms:

- The mean delay – Describes the average time a user spends waiting for a connection if their call is delayed.
- The mean delay – Describes the average time a user spends waiting for a connection whether or not their call is delayed.
- The probability that a user may be delayed longer than time t while waiting for a connection. Time t is chosen by the telecommunications service provider so that they can measure whether their services conform to a set Grade of Service.

BLOCKING PROBABILITY :

Blocking probability definition as probability that all server on busy network system (dwelled). When all servers busy, then the system is no more to process incoming traffic. In this situation incoming traffic is believed to experience blocking.

Comprehension of GOS with blocking probability truly almost same that is to explain about call size that cannot serve by network system. Main different between both is, GOS is a size with point of view from network side or switching system. GOS measured based on observation total call from customers that cannot loaded, whereas blocking probability based on observation busy server (dwelled) on switching system. To distinguish between both clearly, GOS usually named as call congestion (call jam, because showing part from call that rejected or unloaded) or probability loss, whereas blocking probability as time congestion (time jam, because showing part from time which all server or busy line).

Grade of Service (GOS) is the ratio of lost traffic to offered traffic. Offered traffic is the product of the average number of calls generated by users and the average holding time per call. The actual traffic carried by the network is called the carried traffic. Accordingly, GOS is given by,

$GOS=(A-A_0)/A$ Where,

A= offered traffic

A_0= carried traffic

$A-A_0$= lost traffic

So, the smaller the value of grade of service, the better is the service.

GOS is called call congestion or loss probability and the blocking probability is called time congestion.

Delay probability is the probability that a call experiences delay. If the offered load or the input rate of traffic far exceeds the network capacity, then the queue lengths become very large and the calls experience undesirably long delays.

Subscriber viewpoint:

GOS = Call congestion = loss probability Network

viewpoint:

Blocking probability = time congestion

ERLANG

An Erlang is a unit of telecommunications traffic measurement. Strictly speaking, an Erlang represents the continuous use of one voice path. In practice, it is used to describe the total traffic volume of one hour.

For example, if a group of user made 30 calls in one hour, and each call had an average call duration of 5 minutes, then the number of Erlangs this represents is worked out as follows:

Minutes of traffic in the hour = number of calls x duration

Minutes of traffic in the hour = 30 x 5

Minutes of traffic in the hour = 150

Hours of traffic in the hour = 150 / 60

Hours of traffic in the hour = 2.5

Traffic figure = 2.5 Erlangs

Erlang traffic measurements are made in order to help telecommunications network designers understand traffic patterns within their voice networks. This is essential if they are to successfully design their network topology and establish the necessary trunk group sizes.

Erlang traffic measurements or estimates can be used to work out how many lines are required between a telephone system and a central office (PSTN exchange lines), or between multiple network locations.

OPTICAL NETWORKING

The high bandwidths of fiber-optic cable are suitable for today's high-data-rate technologies (such as video conferencing) and for carrying large numbers of lower-rate technologies at the same time. For this reason, the importance of fiber optics grows in conjunction with the development of technologies requiring high data rates or wide bandwidths for transmission. With their prominence came a need for standardization. The United States (ANSI) and Europe (ITUT) have responded by defining standards that, though independent, are fundamentally similar and ultimately compatible. The ANSI standard is called the Synchronous Optical Network (SONET). The ITU-T standard is called the Synchronous Digital Hierarchy (SDH).

ARCHITECTURE

Let us first introduce the architecture of a SONET system: signals, devices, and connections. Signals SONET defines a hierarchy of electrical signaling levels called synchronous transport signals (STSs). Each STS level (STS-l to STS-192) supports a certain data rate, specified in megabits per second (see Table). The corresponding optical signals are called optical carriers

(OCs). SDH specifies a similar system called a synchronous transport module (STM). STM is intended to be compatible with existing European 491 492 CHAPTER 17 SONETISDH hierarchies, such as E lines, and with STS levels. To this end, the lowest STM level, STM-l, is defined as 155.520 Mbps, which is exactly equal to STS-3.

STS	OC	*Rate (Mbps)*	*STM*
STS-1	OC-1	51.840	
STS-3	OC-3	155.520	STM-1
STS-9	OC-9	466.560	STM-3
STS-12	OC-12	622.080	STM-4
STS-18	OC-IS	933.120	STM-6
STS-24	OC-24	1244.160	STM-8
STS-36	OC-36	1866.230	STM-12
STS-48	OC-48	2488.320	STM-16
STS-96	OC-96	4976.640	STM-32
STS-I92	OC-I92	9953.280	STM-64

SONET Devices

Figure shows a simple link using SONET devices. SONET transmission relies on three basic devices: STS multiplexers/demultiplexers, regenerators, add/drop multiplexers and terminals.

STS Multiplexer/Demultiplexer :

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

Regenerator

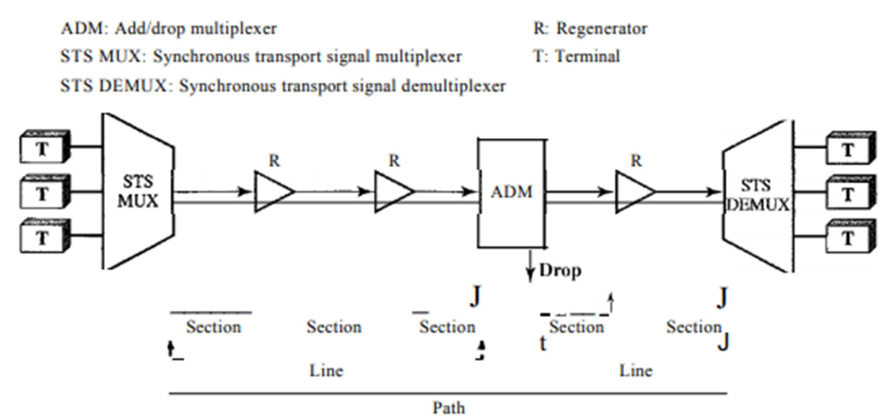
Regenerators extend the length of the links. A regenerator is a repeater that takes a received optical signal (OC-n), demodulates it into the corresponding electric signal (STS-n), regenerates the electric signal, and finally modulates the electric signal into its correspondent OC-n signal. A SONET regenerator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

Add/drop Multiplexer

Add/drop multiplexers allow insertion and extraction of signals. An add/drop multiplexer (ADM) can add STSs coming from different sources into a given path or can remove a desired signal from a path and redirect it without demultiplexing the entire signal. Instead of relying on timing and bit positions, add/drop multiplexers use header information such as addresses and pointers (described later in this section) to identify individual streams. In the simple configuration shown by Figure , a number of incoming electronic signals are fed into an STS multiplexer, where they are combined into a single optical signal. The optical signal is transmitted to a regenerator, where it is recreated without the noise it has picked up in transit. The regenerated signals from a number of sources are then fed into an add/drop multiplexer. The add/drop multiplexer reorganizes these signals, if necessary, and sends them out as directed by information in the data frames. These remultiplexed signals are sent to another regenerator and from there to the receiving STS demultiplexer, where they are returned to a format usable by the receiving links.

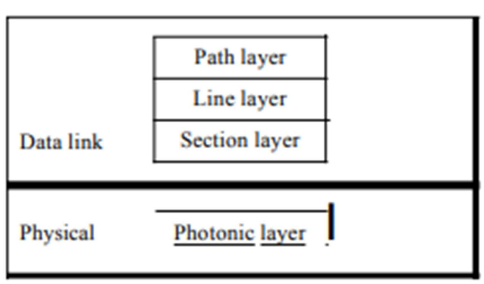
Terminals

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



SONET LAYERS

The SONET standard includes four functional layers: the photonic, the section, the line, and the path layer



Path Layer

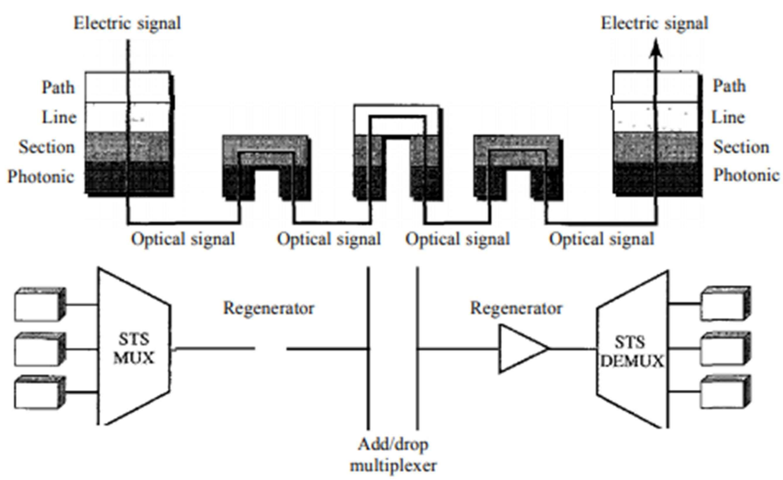
The path layer is responsible for the movement of a signal from its optical source to its optical destination. At the optical source, the signal is changed from an electronic form into an optical form, multiplexed with other signals, and encapsulated in a frame.

Line Layer

The line layer is responsible for the movement of a signal across a physical line. Line layer overhead is added to the frame at this layer. STS multiplexers and add/drop multiplexers provide line layer functions. Section Layer The section layer is responsible for the movement of a signal across a physical section. It handles framing, scrambling, and error control. Section layer overhead is added to the frame at this layer.

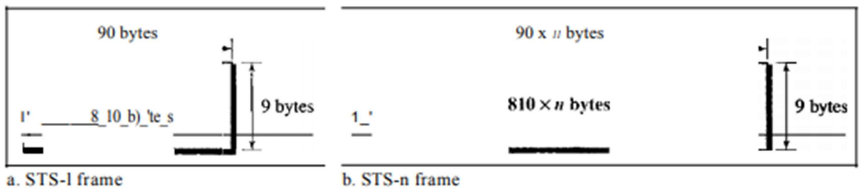
Photonic Layer

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



SONET FRAMES

Each synchronous transfer signal STS-n is composed of 8000 frames. Each frame is a twodimensional matrix of bytes with 9 rows by 90 x n columns. For example, STS-l frame is 9 rows by 90 columns (810 bytes), and an STS-3 is 9 rows by 270 columns (2430 bytes). Figure shows the general format of an STS-l and an STS-n.



Question :

What is the advantage of Optical networking ?

DIGITAL SUBSCRIBER LINE

After traditional modems reached their peak data rate, telephone companies developed another technology, DSL, to provide higher-speed access to the Internet. Digital subscriber line (DSL) technology is one ofthe most promising for supporting high-speed digital communication over the existing local loops. DSL technology is a set of technologies, each differing in the first letter

(ADSL, VDSL, HDSL, and SDSL). The set is often referred to as xDSL, where x can be replaced by A, V, H, or S.

ADSL MODEM

ADSL

The first technology in the set is asymmetric DSL (ADSL). ADSL, like a 56K modem, provides higher speed (bit rate) in the downstream direction (from the Internet to the resident) than in the upstream direction (from the resident to the Internet). That is the reason it is called asymmetric. Unlike the asymmetry in 56K modems, the designers of ADSL specifically divided the available bandwidth of the local loop unevenly for the residential customer. The service is not suitable for business customers who need a large bandwidth in both directions.

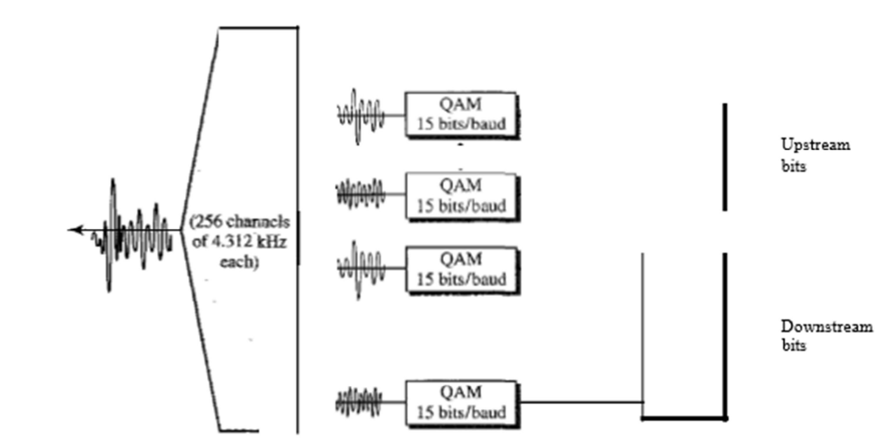
The modulation technique that has become standard for ADSL is called the discrete multitone technique (DMT) which combines QAM and FDM. There is no set way that the bandwidth of a system is divided. Each system can decide on its bandwidth division. Typically, an available bandwidth of 1.104 MHz is divided into 256 channels. Each channel uses a bandwidth of4.312 kHz, as shown in Figure.

Voice. Channel 0 is reserved for voice communication.

Idle. Channels 1 to 5 are not used and provide a gap between voice and data communication.

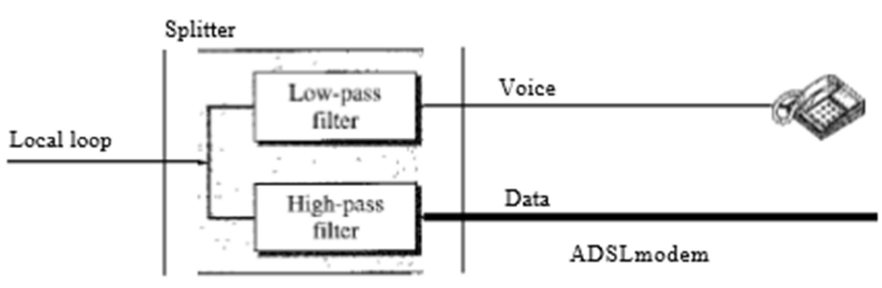
Upstream data and control. Channels 6 to 30 (25 channels) are used for upstream data transfer and control. One channel is for control, and 24 channels are for data transfer. If there are 24 channels, each using 4 kHz (out of 4.312 kHz available) with QAM modulation, we have 24 x 4000 x 15, or a 1.44-Mbps bandwidth, in the upstream direction. However, the data rate is normally below 500 kbps because some of the carriers are deleted at frequencies where the noise level is large. In other words, some of channels may be unused.

Downstream data and control. Channels 31 to 255 (225 channels) are used for downstream data transfer and control. One channel is for control, and 224 channels are for data. If there are 224 channels, we can achieve up to 224 x 4000 x 15, or 13.4 Mbps. However, the data rate is normally below 8 Mbps because some of the carriers are deleted at frequencies where the noise level is large. In other words, some of channels may be unused.



Customer Site: ADSLModem

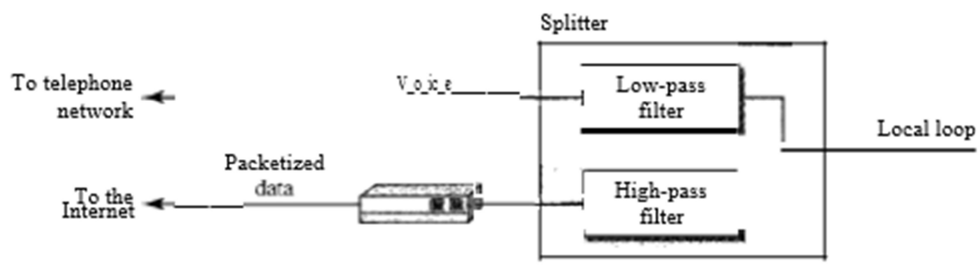
Figure shows an ADSL modem installed at a customer's site. The local loop connects to a splitter which separates voice and data communications. The ADSLmodem modulates and demodulates the data, using DMT, and creates downstream and upstream channels.



Note that the splitter needs to be installed at the customer's premises, normally by a technician from the telephone company. The voice line can use the existing telephone wiring in the house, but the data line needs to be installed by a professional. All this makes the ADSL line expensive.

Telephone Company Site: DSLAM

At the telephone company site, the situation is different. Instead ofan ADSL modem, a device called a digital subscriber line access multiplexer (DSLAM) is installed that functions similarly. In addition, it packetizes the data to be sent to the Internet (ISP server).



Question :

1. Calculate upload and download speed of ADSL modem.

IP TELEPHONY

IP telephony (Internet Protocol telephony) is a general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally have been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). Using the Internet, calls travel as packets of data on shared lines, avoiding the tolls of the PSTN. The challenge in IP telephony is to deliver the voice, fax, or video packets in a dependable flow to the user. Much of IP telephony focuses on that challenge.

IP telephony service providers include or soon will include local telephone companies, long distance providers such as AT&T, cable TV companies, Internet service providers (ISPs), and fixed service wireless operators. IP telephony services also affect vendors of traditional handheld devices.

Currently, unlike traditional phone service, IP telephony service is relatively unregulated by government. In the United States, the Federal Communications Commission (FCC) regulates phone-to-phone connections, but says they do not plan to regulate connections between a phone user and an IP telephony service provider.

VoIP is an organized effort to standardize IP telephony. IP telephony is an important part of the convergence of computers, telephones, and television into a single integrated information environment. Also see another general term, computer-telephony integration (CTI), which describes technologies for using computers to manage telephone calls.

IP telephony is designed to replace the telecommunications' infrastructure of circuit switched public data networks (CSPDN) and public switched telephone networks (PSTN) with packet switched IP communication networks.

In a consumer IP telephony solution, a soft IP phone application and backend Internet connection enable voice and data communication, such as calling and faxing. A user may call other softphone users, send or receive faxes and even communicate with circuit switched and cellular communication services.

In an enterprise environment, IP telephony is implemented through physical IP phones that work on top of an IP network infrastructure. An IP phone's built-in firmware provides the complete functionality for initiating and managing telephonic communications. Moreover, IP telephony also supports video communication between two or more users.

Voice over Internet Protocol (VoIP), a popular IP telephony implementation, only supports voice communication over IP.

VOICE OVER INTERNET PROTOCOL

Voice over Internet Protocol (also voice over IP, VoIP or IP telephony) is a methodology and group of technologies for the delivery of voice communications and multimediasessions over Internet Protocol (IP) networks, such as the Internet. The terms Internet telephony, broadband telephony, and broadband phone service specifically refer to the provisioning of communications services (voice, fax, SMS, voice-messaging) over the public Internet, rather than via the public switched telephone network (PSTN).

The steps and principles involved in originating VoIP telephone calls are similar to traditional digital telephony and involve signaling, channel setup, digitization of the analog voice signals, and encoding. Instead of being transmitted over a circuit-switched network, the digital information is packetized, and transmission occurs as IP packets over a packet-switched network. They transport media streams using special media delivery protocols that encode audio and video with audio codecs, and video codecs. Various codecs exist that optimize the media stream based on application requirements and network bandwidth; some implementations rely on narrowband and compressed speech, while others support high-fidelity stereo codecs. Some popular codecs include µ-law and a-law versions of G.711, G.722, an open source voice codec known as iLBC, a codec that only uses 8 kbit/s each way called G.729, and many others.

Early providers of voice-over-IP services offered business models and technical solutions that mirrored the architecture of the legacy telephone network. Second-generation providers, such as Skype, built closed networks for private user bases, offering the benefit of free calls and convenience while potentially charging for access to other communication networks, such as the PSTN. This limited the freedom of users to mix-and-match third-party hardware and software. Third-generation providers, such as Google Talk, adopted the concept of federated VoIP—which is a departure from the architecture of the legacy networks.[1] These solutions typically allow dynamic interconnection between users on any two domains on the Internet when a user wishes to place a call.

In addition to VoIP phones, VoIP is available on many personal computers and other Internet access devices. Calls and SMS text messages may be sent over mobile data or Wi-Fi.

PROTOCOL USED in VOIP

Voice over IP has been implemented in various ways using both proprietary protocols and protocols based on open standards. These protocols can be used by a VoIP phone, specialpurpose software, a mobile application or integrated into a web page. VoIP protocols include:

- Session Initiation Protocol (SIP), connection management protocol developed by the IETF
- H.323, one of the first VoIP call signaling and control protocols that found widespread implementation. Since the development of newer, less complex protocols such as MGCP and SIP, H.323 deployments are increasingly limited to carrying existing long-haul network traffic.
- Media Gateway Control Protocol (MGCP), connection management for media gateways
- H.248, control protocol for media gateways across a converged internetwork consisting of the traditional public switched telephone network (PSTN) and modern packet networks

- Real-time Transport Protocol (RTP), transport protocol for real-time audio and video data
- Real-time Transport Control Protocol (RTCP), sister protocol for RTP providing stream statistics and status information
- Secure Real-time Transport Protocol (SRTP), encrypted version of RTP
- Session Description Protocol (SDP), file format used principally by SIP to describe VoIP connections
- Inter-Asterisk eXchange (IAX), protocol used between VoIP servers
- Extensible Messaging and Presence Protocol (XMPP), instant messaging, presence information, and contact list maintenance
- Jingle, adds peer-to-peer session control to XMPP
- Skype protocol, proprietary Internet telephony protocol suite based on peer-to-peer architecture

SESSION INITIATION PROTOCOL

The Session Initiation Protocol (SIP) is a communications protocol for signaling and controlling multimedia sessions in applications of Internet telephony for voice and video calls, in private IP telephone systems, as well as in instant messaging over Internet Protocol (IP) networks.

The protocol defines the specific format of messages exchanged and the sequence of communications for cooperation of the participants. SIP is a text-based protocol, incorporating many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP).[1] A call established with SIP may consist of multiple media streams, but no separate streams are required for applications, such as text messaging, that exchange data as payload in the SIP message.

SIP works in conjunction with several other protocols that specify and carry the session media. Media type and parameter negotiation and media setup is performed with the Session Description Protocol (SDP), which is carried as payload in SIP messages. SIP is designed to be independent of the underlying transport layer protocol, and can be used with the User Datagram Protocol (UDP), the Transmission Control Protocol (TCP), and the Stream Control Transmission Protocol (SCTP). For the transmission of media streams (voice, video) SIP typically employs the Real-time Transport Protocol (RTP) or the Secure Real-time Transport Protocol (SRTP). For secure transmissions of SIP messages over insecure network links, the protocol may be encrypted with Transport Layer Security (TLS).

Protocol Operation :

SIP is only involved for the signaling operations of a media communication session and is primarily used to set up and terminate voice or video calls. SIP can be used to establish two-party (unicast) or multiparty (multicast) sessions. It also allows modification of existing calls. The modification can involve changing addresses or ports, inviting more participants, and adding or deleting media streams. SIP has also found applications in messaging applications, such as instant messaging, and event subscription and notification.

SIP works in conjunction with several other protocols that specify the media format and coding and that carry the media once the call is set up. For call setup, the body of a SIP message contains a Session Description Protocol (SDP) data unit, which specifies the media format, codec and media communication protocol. Voice and video media streams are typically carried between the terminals using the Real-time Transport Protocol (RTP) or Secure Real-time Transport Protocol (SRTP).

Under SIP, each resource, such as a user agent or a voicemail box, is identified by a Uniform Resource Identifier (URI), which follows the general standard syntax also used in Web services and e-mail. The URI scheme used for SIP is sip and a typical SIP URI has the form sip:username@domainname or sip:username@hostport, where domain name requires DNS SRV records to locate the servers for SIP domain while hostport can be an IP address or a fully qualified domain name of the host and port. If secure transmission is required, the scheme sips is used.

SIP employs design elements similar to the HTTP request/response transaction model. Each transaction consists of a client request that invokes a particular method or function on the server and at least one response. SIP reuses most of the header fields, encoding rules and status codes of HTTP, providing a readable text-based format.

SIP can be carried by several transport layer protocols including Transmission Control Protocol (TCP), User Datagram Protocol (UDP), and Stream Control Transmission Protocol(SCTP). SIP clients typically use TCP or UDP on port numbers 5060 or 5061 for SIP traffic to servers and other endpoints. Port 5060 is commonly used for non-encrypted signaling traffic whereas port 5061 is typically used for traffic encrypted with Transport Layer Security (TLS).

SIP-based telephony networks often implement call processing features of Signaling System 7 (SS7), for which special SIP protocol extensions exist, although the two protocols themselves are very different. SS7 is a centralized protocol, characterized by a complex central network architecture and dumb endpoints (traditional telephone handsets). SIP is a client-server protocol of equipotent peers. SIP features are implemented in the communicating endpoints, while the traditional SS7 architecture is in use only between switching centers.

Questions :

1. What is the importance of VOIP ?

2. Which protocols are used in VOIP ?

ISDN (Integrated Services Digital Network)

Integrated Services Digital Network (ISDN) is a set of communication standards for simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the public switched telephone network. It was first defined in 1988 in the CCITT red book. Prior to ISDN, the telephone system was viewed as a way to transport voice, with some special services available for data. The key feature of ISDN is that it integrates speech and data on the same lines, adding features that were not available in the classic telephone system. The ISDN standards define several kinds of access interfaces, such as Basic Rate Interface (BRI), Primary Rate Interface (PRI), Narrowband ISDN (N-ISDN), and Broadband ISDN (B-ISDN).

ISDN is a circuit-switched telephone network system, which also provides access to packet switched networks, designed to allow digital transmission of voice and data over ordinary telephone copper wires, resulting in potentially better voice quality than an analog phone can provide. It offers circuit-switched connections (for either voice or data), and packetswitched connections (for data), in increments of 64 kilobit/s. In some countries, ISDN found major market application for Internet access, in which ISDN typically provides a maximum of 128 kbit/s bandwidth in both upstream and downstream directions. Channel bonding can achieve a greater data rate; typically the ISDN B-channels of three or four BRIs (six to eight 64 kbit/s channels) are bonded.

ISDN is employed as the network, data-link and physical layers in the context of the OSI model. In common use, ISDN is often limited to usage to Q.931 and related protocols, which are a set of signaling protocols establishing and breaking circuit-switched connections, and for advanced calling features for the user. They were introduced in 1986.

In a videoconference, ISDN provides simultaneous voice, video, and text transmission between individual desktop videoconferencing systems and group (room) videoconferencing systems.

There are two levels of service: the Basic Rate Interface (BRI), intended for the home and small enterprise, and the Primary Rate Interface (PRI), for larger users. Both rates divide their capacity across a number of channels:

- B-channels carry payloads (e.g., data or voice streams)
- D-channels carry control and signaling information.

A BRI connection consists of two 64 Kbps B-channels and one 16 Kbps D-channel. Thus, a BRI delivers up to 128 Kbps of data.

In the United States, a PRI connection consists of 23 B-channels (1,472 Kbps total) and one 64 Kbps D-channel. In Europe a PRI connection consists of 30 B-channels (1,920 Kbps total) and 1 D-channel.

ISDN was slow to achieve standardization and was rapidly overtaken and surpassed in both speeds possible and breadth of deployment by packet-switched technologies.

Broadband ISDN (BISDN) extends the integration throughout the rest of an end-to-end path at higher data rates -- for example, using fiber optic or radio media. ISDN can combine both analog/voice data and digital data on the same network link. Most video conferencing services used in the 1990s and early 2000s were delivered primarily via ISDN services.

ISDN can deliver speeds up to 128 Kbps over home phone lines. Where it is still available for residential users, it -- like acoustic modems -- typically serves as a last-resort technique for lowend data connectivity, used only when DSL and cable modem services are not available.

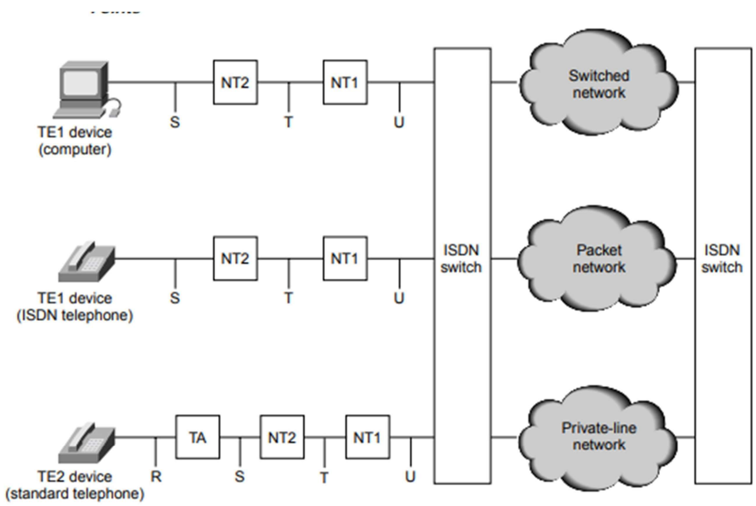
ISDN Devices

ISDN devices include terminals, terminal adapters (TAs), network-termination devices, linetermination equipment, and exchange-termination equipment. ISDN terminals come in two types. Specialized ISDN terminals are referred to as terminal equipment type 1 (TE1). NonISDN terminals, such as DTE, that predate the ISDN standards are referred to as terminal equipment type 2 (TE2). TE1s connect to the ISDN network through a four-wire, twisted-pair digital link. TE2s connect to the ISDN network through a TA. The ISDN TA can be either a standalone device or a board inside the TE2. If the TE2 is implemented as a standalone device, it connects to the TA via a standard physical-layer interface. Examples include EIA/TIA-232-C (formerly RS-232-C), V.24, and V.35. Beyond the TE1 and TE2 devices, the next connection point in the ISDN network is the network termination type 1 (NT1) or network termination type 2 (NT2) device. These are network-termination devices that connect the four-wire subscriber wiring to the conventional two-wire local loop. In North America, the NT1 is customer premises equipment (CPE) device. In most other parts of the world, the NT1 is part of the network provided by the carrier. The NT2 is a morecomplicated device that typically is found in digital private branch exchanges (PBXs) and that performs Layer 2 and 3 protocol functions and concentration services. An NT1/2 device also exists as a single device that combines the functions of an NT1 and an NT2. ISDN specifies a number of reference points that define logical interfaces between functional groups, such as TAs and NT1s. ISDN reference points include the following:

- R—The reference point between non-ISDN equipment and a TA. • S—The reference point between user terminals and the NT2.
- T—The reference point between NT1 and NT2 devices.
- U—The reference point between NT1 devices and line-termination equipment in the carrier network. The U reference point is relevant only in North America, where the NT1 function is not provided by the carrier network.

Figure illustrates a sample ISDN configuration and shows three devices attached to an ISDN switch at the central office. Two of these devices are ISDN-compatible, so they can be attached

through an S reference point to NT2 devices. The third device (a standard, non-ISDN telephone) attaches through the reference point to a TA. Any of these devices also could attach to an NT1/2 device, which would replace both the NT1 and the NT2.



Applications of ISDN:

1. Video Conferencing: ISDN circuits have been very useful for video conferencing. A video conferencing session can be established with a single ISDN line (128 Kbps) and a set-top video conferencing Codec (at both ends). The video conferencing can be done at a much lower bandwidth than a packet switched network (like Internet Leased Lines or high speed broadband) because, ISDN is a circuit switched network and hence there are no packet losses and ISDN can give equal throughput for both upstream and downstream transmissions (which is so critical for sending and receiving real time video in both the ends). Apart from the rental, the payment is only for usage (duration in minutes). So, if your company doesn't conduct video conferencing sessions frequently, ISDN can still be a good choice! Moreover, the data network would be free to do what it needs to do – carry data without any congestion and delays. Multiple ISDN (Up to four, in most video conferencing systems) can be terminated simultaneously in video conferencing systems to enable a conference at higher bandwidth (hence higher quality) or

conduct a multi-party video conference. Even if you have your entire video conferencing infrastructure on IP, but have to connect one customer on ISDN, you can still do so by using certain devices that convert ISDN to IP and vice versa.

2. Making Couple of Telephone Calls Simultaneously:
With an ISDN line, we can do a number of things simultaneously:

- We can dial out two telephone calls (by connecting two analog phones to the adapter)
- We can browse the Internet (128 Kbps) and if we get a call during browsing, the call will be connected in one channel (64 Kbps) and the Internet connectivity can continue in the other channel at a lower speed (64 Kbps)
- We can send a fax in one channel and simultaneously talk to someone in the other channel (while the fax is being transmitted)

3. ISDN Video Phones: There are certain video phones that connect to the ISDN network and can make a video call (voice and video) to any other similar ISDN phone connected to the ISDN network. The video phones have a bigger screen and built in camera to show the output video as well as capture the video.

4. Broadcasting Industry: The broadcasting industry (especially the radio broadcasting stations) use ISDN network for carrying the audio signals from one station to another as well as from the covering point (sports stadiumetc.) to the broadcasting point (radio station). But these days, IP networks are also popular for this application.

5. Digital PBX/ IP PBX: There are ISDN interface cards which allow ISDN lines to be terminated on a corporate PBX (both for Digital – Mixed PBX as well as IP PBX). So, the Digital ISDN lines can be terminated on the PBX and can be used for making outgoing calls as well as receiving incoming calls from any phone in the organization. This can be especially useful for companies that want to use ISDN for video conferencing, as the ISDN lines can be used for normal voice communications (through PBX) when they are not being used for video conferencing.

6. Back-up Internet Connectivity: The ISDN networks can be directly terminated on certain enterprise Routers that provide an automatic fail-over to the ISDN network for accessing Internet when the primary Internet Line (Internet Leased Lines, Broadband etc) is down. Sometimes, ISDN can also be the primary source of Internet connectivity for small companies in remote locations where the ADSL broadband networks are not supported.

7. Bank ATM/ Point of Sale Locations: ISDN networks make a good choice for connecting critical yet low bandwidth consuming applications like multiple Bank ATM's (Automated Teller Machines), Point of Sale locations like an exhibition which needs network access for accepting payment through credit cards, etc.

Question :

1. Calculate transmission speed in BRI mode.
2. Mention some application of ISDN.