SONET

➔The SONET standard was developed by the ANSI T1X1 committee with first publicationin 1988. The standard defines the features and functionality of a transport system based on the principles of synchronous multiplexing. This means that individual tributary signals may be multiplexed directly into a higher rate SONET signal without intermediate stages of multiplexing.

➔SONET is capable of transporting all the tributary signals that have been defined for the digital networks in existence today. SONET has the flexibility to readily accommodate the new types of customer service signals such as SMDS (switched multimegabit data service) and ATM (asynchronous transfer mode). Actually, it can carry any octet-based binary format such as TCP/ IP, SNA, OSI regimes, X.25, frame relay, and various LAN formats, which have been packaged for long-distance transmission.

➔SONET is based on a synchronous signal comprised of eight-bit octets, which are organized into a frame structure. The frame can be represented by a two-dimensional map comprising N rows and M columns, where each box so derived contains one octet (or byte). The upper left-hand corner of the rectangular map representing a frame contains an identifiable marker to tell the receiver it is the start of frame.

➔SONET consists of a basic, first-level, structure called STS-1

➔ The definition of the first level also defines the entire hierarchy of SONET signals because higher-level SONET signals are obtained by synchronously multiplexing the lower-level modules. When lower-level modules are multiplexed together, the result is denoted as STS-N (STS stands for synchronous transport signal), where N is an integer. The resulting format then can be converted to an OC-N (OC stands for optical carrier) or STS-N electrical signal. There is an integer multiple relationship between the rate of the basic module STS-1 and the OC-N electrical signals (i.e., the rate of an OC-N is equal to *N* times the rate of an STS-1). Only OC-1, -3, -12, -24, -48, and -192 are supported by today's SONET.

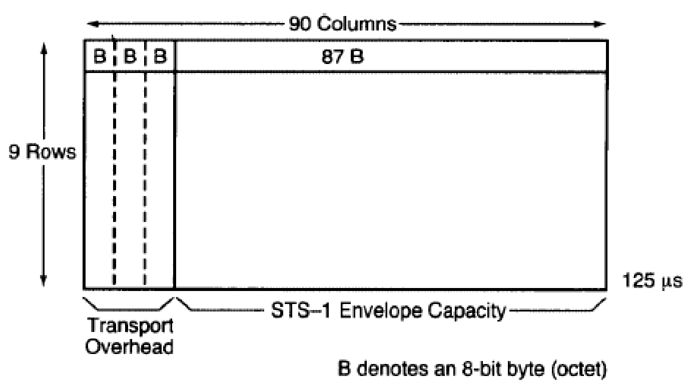

Fig:STS-1 frame

➔ STS-1 is the basic module and building block of SONET. It is a specific sequence of 810 octets (6480 bits), which includes various overhead octets and an envelope capacity for transporting payloads. STS-1 is depicted as a 90-column, 9-row structure. With a frame period of 125 ms (i.e., 8000 frames per second). STS-1 has a bit rate of 51.840 Mbps. The order of transmission of octets is row-by-row, from left to right. In each octet of TS-1 the most significant bit is transmitted first. As illustrated in Figure, the first three columns of the STS-1 frame contain the transport overhead. These three columns have 27 octets (i.e., 9 ∗ 3) of which 9 are used for the *section overhead* and 18 octets contain the *line overhead*. The remaining 87 columns make up the STS-1 envelope capacity.

Table Line Rates for Standard SONET Interface Signals

OC-N Level	STS-N Electrical Level	Line Rate (Mbps)
OC-1	STS-1 electrical	51.84
OC-3	STS-3 electrical	155.52
OC-12	STS-12 electrical	622.08
OC-24	STS-24 electrical	1244.16
OC-48	STS-48 electrical	2488.32
OC-192	STS-192 electrical	9953.28

SWITCHING (remaining portion …….)

a) Strowger Switching System

Strowger Switching System

- first automatic switching system.
- developed by A.B Strowger in 1889.
- two types of selector which form the building block of for the switching system i.e,

(i) Uniselector : It is one which has single rotary switch with a bank of contact. Depending upon the number of switching contacts uniselector are identified as 10 or 34 outlet uniselectors.

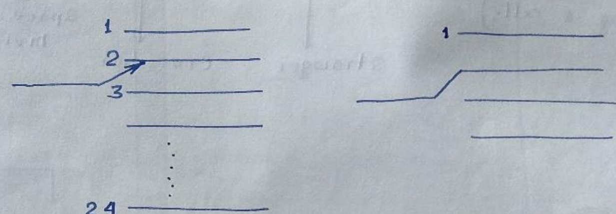

fig: Schematic representation of uniselector.

(ii) Two - motion Selector: It is capable of horizontal as well as vertical stepping movement. It has two rotary switches one providing vertical motion & other providing horizontal movements.

The horizontal & vertical motion in a two - motion selector may be effected directly using two pulse train from subscriber dialing. The first impulse train corresponding to the first digit operates the vertical magnet & second impulse train drives the horizontal rotary switch.

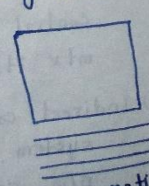

fig: two - motion selector

b) Step by Step switching

Step-by-step Switching

This may be constructed by using uniselectors or two motion selectors or both.

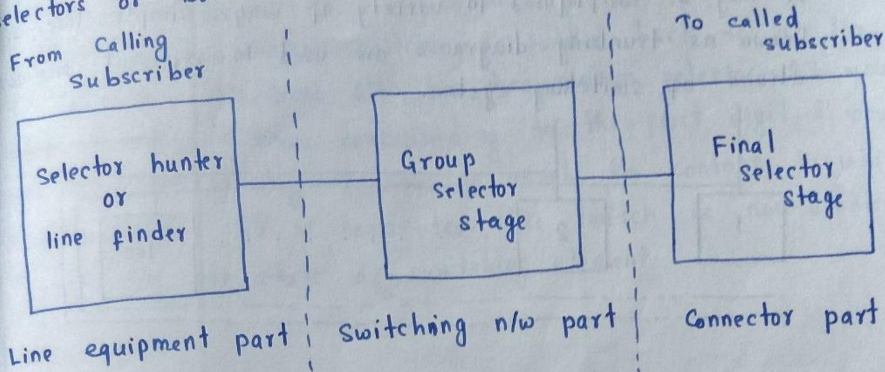

fig: Configuration of step-by-step switching system

It consists of three main parts ÷

(i) Line equipment part ÷ Consists of hunter or line finder. The main task of selector hunt is to search & seize a selector from the switching mtx part. There is only one selector hunter for each subscriber. Usually, 24 outlet uniselectors are used as selector hunters. Selector hunter scheme is sometimes called subscriber uniselector scheme as there is dedicated uniselector for each subscriber in the system. Line finder searches & finds the line of subscriber to be connected to the first selector associated with it. These are built using uniselector or two-motio selector. So, line finders & selector hunters are generally referred as pre-selectors.

(ii) Switching n/w or matrix part ÷ Consists of one or more sets of two motion selector known as first group selector, second group selector & so on.

(iii) Connector part ÷ It comprises one set of two motion selector known as final selectors.

c) Cross bar switching

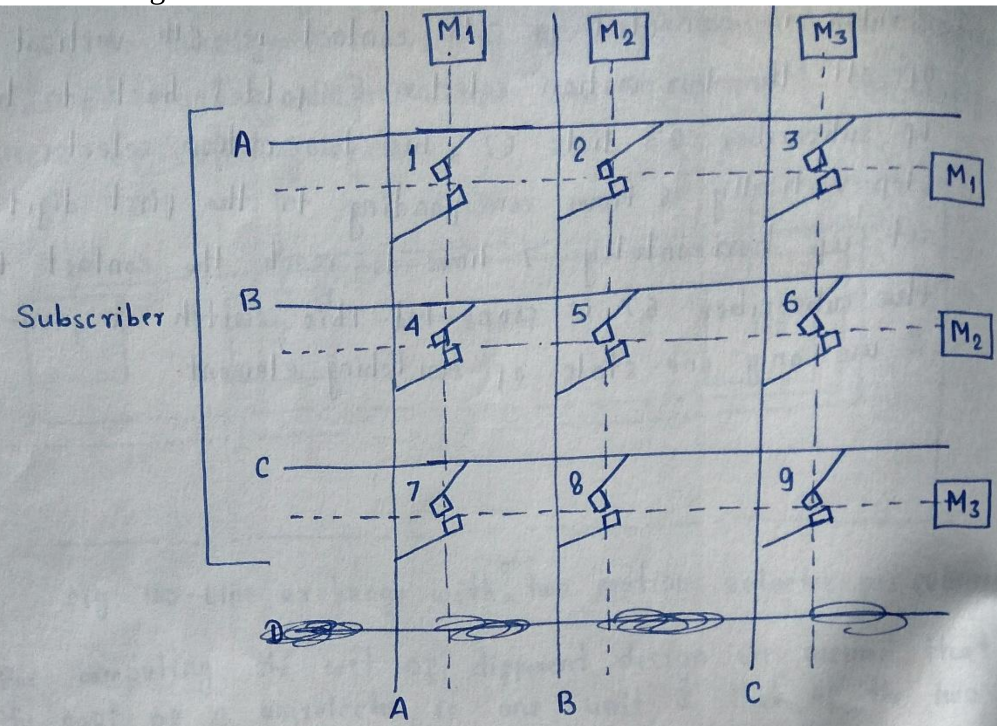

fig: 3 x 3 Crossbar switching

The major disadvantage of the strowger of the strowger switching system is it's dependence on moving parts & contacts that are subject to wear & tear. So, it was replaced by crossbar. Crossbar are designed using common control concept. The basic idea of this switching is to provide a mtx of n x m sets of contacts with only (n+m) activators or less to select one of the n x m sets of contact. This form of switching is also known as co-ordinate switching as the switching contacts are arranged in x-y plane.

The set of vertical & horizontal contacts points are connected to wires.

When an electromagnet say in horizontal direction is energized the bar attached to it slightly rotates in such a way that the contact points attached to the bar moves closer to it's facing contact points but do not actually make any contact

Now if an electromagnet in the vertical direction is energised, the corresponding bar rotates causing the contact point at the intersection of the two bar.

Since, the life of electromechanical switching is very short & depends on freq. of operation. So, it needs to develop the electronic switching.
Advances made in computer technology were incorporated & lead to store programme control.
This enable a digital computer to be used as a central control & perform different function with the same n/w by executing different programme.
The more additional functions are,
Call barring ; Repeat lost call ; Reminder Calls ; Call diversion ; Three way call ; charging Advice;

9

<<<< for SPACE DIVISION SWITCHING, TIME DIVISION SWITCHING refer switching(space, time).pdf>>>>>>>>>>>>>

***Time switch (also time division switching or TSI- time-slot interchanger)**

***PCM switching means- Time, space and combination of both switches**

Space switch(Time division space switch)

A typical time-division space switch (S) is shown in Figure(a,b) below. It consists of a cross-point matrix made up of logic gates that allow the switching of time slots in a spatial domain. These PCM time slot bit streams are organized by the switch into a pattern determined by the required network connectivity. The matrix consists of a number of input horizontals and output verticals with a logic gate at each cross point. The array, as shown in the figure, has M horizontals and N verticals, and we call it an M× N array. If M = N, the switch is non blocking; If M > N, the switch concentrates, and if M < N, the switch expands.

For a given time slot, the appropriate logic gate is enabled and the time slot passes from the input horizontal to the desired output vertical. The other horizontals, each serving a different serial stream of time slots, can have the same timeslot (e.g., a time slot from time slots number 1–24, 1–30, or 1–*n*; e.g., time slot 7 on each stream) switched into other verticals enabling their gates. In the next time-slot position (e.g., time slot 8), a completely different path configuration could occur, again allowing time slots from horizontals to be switched to selected verticals

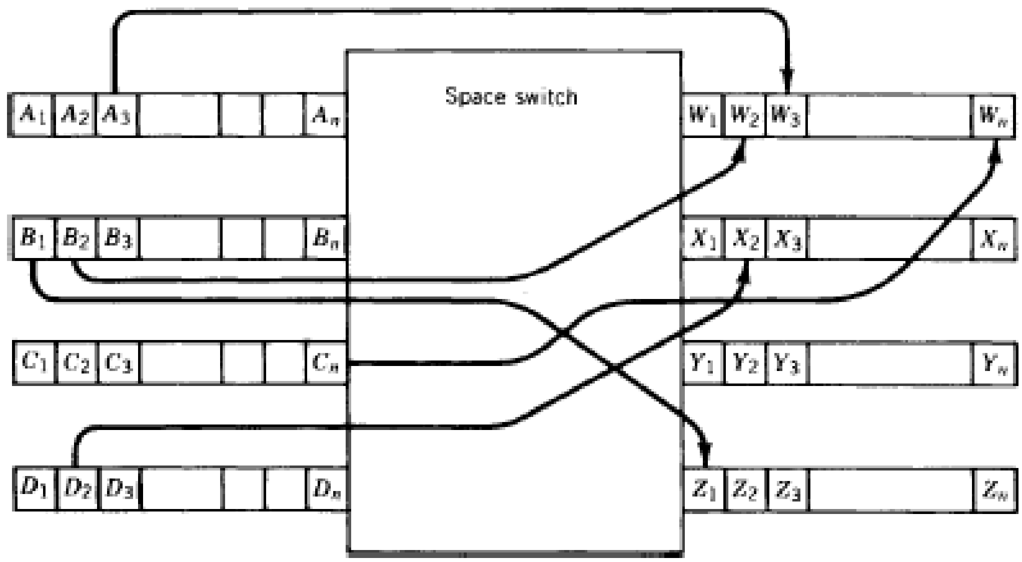

Figure a: Space switch connects time slots in a spatial configuration

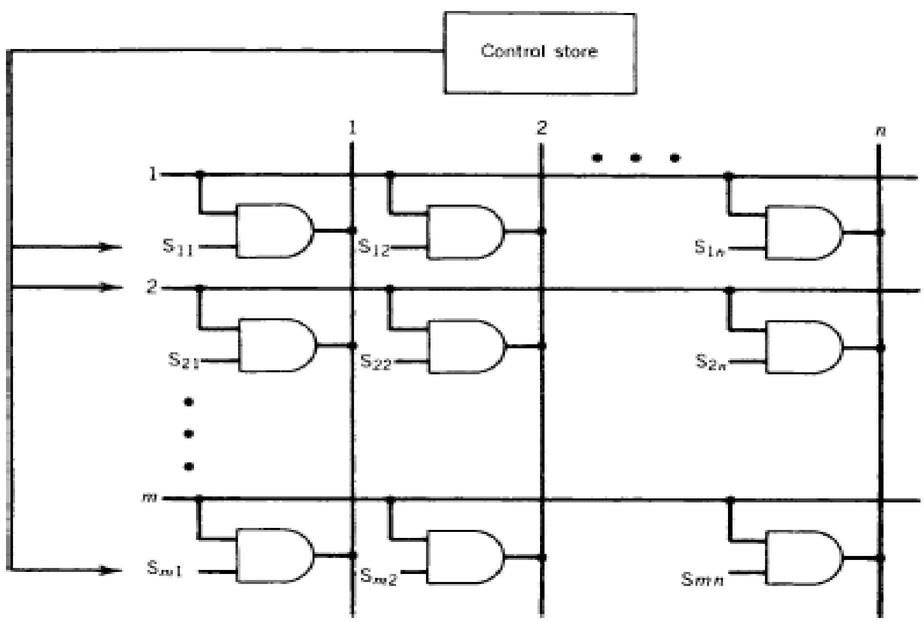

Figure b:Time-division space switch cross-point array showing enabling gates.

Combination switches (multi stage switch)

a)TST (time- space – time)

b)STS (space-time-space)

others TSSSST,…

a)TST (time- space – time) switch

➔ Digital switches are composed of time and space switches in any order

➔ a switch that consists of a sequence of a time-switching stage, a space-switching stage, and a time-switching stage is called a TST switch.

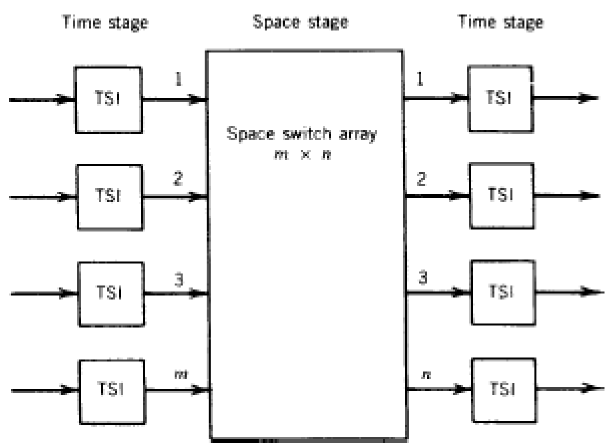

Fig: TST switch (TSI= time slot interchanger i.e. time switch)

➔ The first stage of the switch is the TSI or time stages that interchange time slots (in the time domain) between external incoming digital channels and the subsequent space stage. The space stage provides connectivity between time stages at the input and output. It is a multiplier of call-handling capacity. The multiplier is either the value for M or value for N, whichever is smaller. We also saw earlier that space-stage time slots need not have any relation to either external incoming or outgoing time slots regarding number, numbering, or position. For instance, incoming time slot 4 can be connected to outgoing time slot 19 via space network time slot 8.

➔ A TST switch is strictly nonblocking if $l = 2c - 1$,

where l is the number of space-stage time slots

c is the number of external TDM time slots

b) STS (space-time-space) switch

➔ A space–time–space switch reverses the order architecture of a TST switch. The STS switch consists of a space cross–point matrix at the input followed by an array of time-slot interchangers whose ports feed another cross-point matrix at the output

➔ Consider this operational example with an STS. Suppose that an incoming time slot 5 on port No. 1 must be connected to an output slot 12 at outgoing port 4. This can be accomplished by time-slot interchanger No. 1, which would switch it to time slot 12; then the outgoing space stage would place that on outgoing trunk No. 4. Alternatively, time slot 5 could be placed at the input of TSI No. 4 by the incoming space switch, where it would be switched to time slot 12, and then out port No. 4.

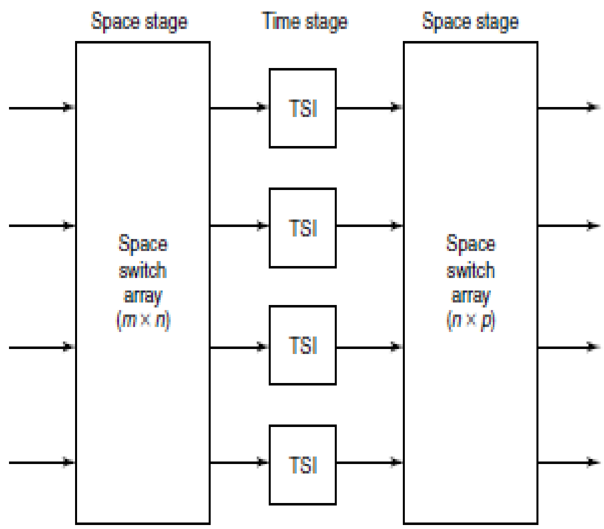

Figure STS switch

<u> comparision between TST and STS switch</u>**

➔ The architecture of TST switching is more complex than STS switching with space concentration.

➔The TST switch becomes more cost-effective because time expansion can be achieved at less cost than space expansion. Such expansion is required as link utilization increases because less concentration is acceptable as utilization increases.

➔It would follow, then, that TST switches have a distinct implementation advantage overSTS switches when a large amount of traffic must be handled.

➔ for small switches STS is favored due to reduced implementation complexities

➔ The choice of a particular switch architecture may be more dependent on such factors as modularity, testability, and expandability.

➔One consideration that generally favors an STS implementation is the relatively simpler control requirements. However, for large switches with heavy traffic loads, the implementation advantage of the TST switch and its derivatives is dominant.

Digital cross-connect

For a telephone call, a connection is made through a digital switching network at the start of a call and cleared down as soon as the call ends. However, a similar digital switching network may be used for semi-permanent connections. It is controlled manually from an operating terminal instead of automatically by the processor of an exchange. Such a switching network is called a *digital cross-connect unit*. It performs a function for digital circuits similar to that of a distribution frame for analog circuits. It is sometimes called a 'slow switch', in contrast to a 'fast switch' used to connect telephone calls and the connections made by a digital cross-connect unit are sometimes called 'nailed-up time-slots'.

Two functions that can be performed by digital cross-connect units are *grooming* and *consolidation*. In grooming, 64 kbit/s channels on a common PCM bearer are separated for routing to different destinations. For example, a line from a customer's PBX may carry a mixture of PCM channels, some to the public exchange and some to other PBXs in the customer's private network. In consolidation, channels on several PCM bearers that are not fully loaded are combined onto a smaller number of bearers, thereby improving the utilization of the PCM systems.

Private Branch Exchange (PBX)

Below the hierarchy of national public network, some customers have internal lines serving extension telephone. Still the PBX network is similar to the public exchange. However a small PBX may only generate sufficient traffic for all its connections to be made over a single highway. All the ports for the extension lines, exchange lines and the operator's position, have codecs connected to a common highway. The codecs are operated on the required time slots by a connection store.

The line capacity of the PBX is increased by using 8-bit parallel transmission instead of serial transmission to increase the number of time slots of a common highway. Each connection uses two time slots so as to provide full duplex transmission over the same highway.

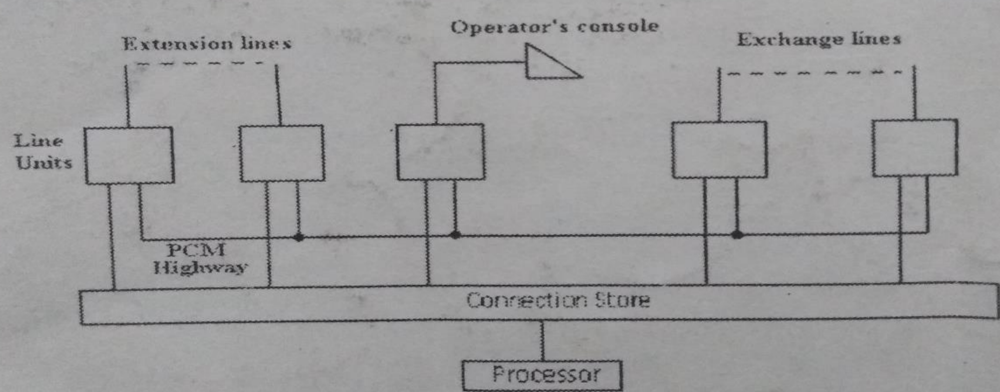

Fig: Trunking of a digital PBX

NGN (Next Generation Network)

NGN (Next Generation Network)

→ ITU-T Definition.

"A NGN is a packet-based network able to provide services including Telecommunication services & able to make use of multiple broadband Quality of Service (QoS) enabled transport technologies & in which the services-related functions are independent from underlying transport-related technologies."

→ General idea behind NGN is that one network transports all information & services (voice, data & all sorts of media such as video) by encapsulating these into packets.

→ NGN is ~~term~~ commonly built around IP so "all IP" is used to describe ~~term~~ transformation towards NGN.

Characteristicss/Features

a) All IP or Packet-based Network.

→ NGN is an integrated IP n/w for wired or wireless communications, and it could eventually handle all types of traffic or application over packet networks.

b) Separation of Application functions from the transport view.

→ In existing n/ws applications are vertically integrated with transport layer & certain n/ws are dedicated to specific application. On the other hand, NGN provides an open architecture by uncoupling applications & n/ws & allowing them to be offered separately.

c) Converged or Integrated n/ws.

→ Traditionally, separate n/ws have been used to provide voice, data & video applications each requiring separate access devices. NGN allows different kind of applications to be transformed into packets and delivered simultaneously. such as voice, data & video can be merged in same n/w.

d) Ubiquitous Network.

→ NGN allows for widespread mobility allowing users to seamlessly access all type of applications, at the same level of quality, in any geographical area.

e) Distribution of n/w intelligence.

→ Since the current PSTN is based on a smart n/w & dumb terminals, NGN have intelligence within the n/w & ~~small~~ Smart terminal within this n/w environment, it is possible for service providers to easily provide various types of services without the need of cumbersome n/w equipment.

* Transmission of real time traffic using packets

Many things can happen to pkts as they travel from origin to destination, resulting in the following problem as seen from the point of view of the sender & receiver:-

- Dropped packets.
- Delay.
- Jitter.
- Out-of-order delivery.
- Error
- Buffering

Dropped packets :- The router might fail to deliver (drop) some pkt if they arrive when their buffers are already full. Some, none, or all ~~pkt~~ of the pkts might be dropped, depending on the state of the n/w, & it is impossible to determine what will happen in advance. The receiving application may ask for this information to be retransmitted, possibly causing severe delays in the overall transmission.

Buffering :- By buffer we mean a routine or storage medium used to compensate for a difference in rate of flow of data betn. devices.

Delay :- It might take a long time for a pkt. to reach its destination, because it gets held up in long queues, or take a less direct route to avoid congestion. Alternatively, it might follow a fast, direct route. This delay is very unpredictable. In some cases, excessive delay can render an application, such as VOIP, unusable.

Jitter ÷ Pkts from source will reach the destination with different delays. This variation in delay is known as jitter & can seriously affect the quality of streaming audio &/or video.

Out-of-Order Delivery ÷ When a collection of related pkts. is routed through the internet, different pkts may take different routes, each resulting in a different delay. The result is that the pkt arrive in different order than they were sent. This problem necessitates special additional protocols responsible for rearranging out-of-order pkts to an ~~asyn~~isochronous state once they reach their destination. This is especially important for video & VOIP streams where quality is dramatically affected by both latency & lack of isochronicity.

Error ÷ Sometimes pkts are misdirected, or combined together, or corrupted, while en route. The receiver has to detect this & just as if the pkt was dropped, ask the sender to repeat itself.

Quality of Service (QoS)

In the fields of pkt - switched n/w & computer networking, the traffic engineering term Quality of Service refers to resource reservation control mechanism.

A defined Quality of Service may be reqd. for certain types of n/w traffic, for example:

- Streaming multimedia may require guaranteed throughput.
- IP telephony or voice over IP (VOIP) may require ~~st~~ strict limits of jitter & delay.
- Video ~~conferencing~~ teleconferencing (VTC) requires low jitter.
- Telemedicine.

* Integrated Service Architecture

It is an architecture that specifies the elements to guarantee of quality of service (QoS) on n/w. It can be used to allow video & sound to reach the receiver without interruption. It is a response to the growing variety & volume of traffic experienced in the Internet & Intranets. It provides a frame-work ~~dev~~ for the development of the protocols i.e, RSVP & RTP, to handle multimedia & sol; multicast traffic & provides guidance to router vendors on the development of efficient techniques for handling a varied load.

The integrated service architecture defines three classes of service based on applications' delay requirements (from highest perf-ormance to lowest):

- Guaranteed-service class: with BW, bounded delay & no-loss guarantees.

- Controlled-load service class: approximating best-effort service in a lightly loaded n/w, which provides for a form of statistical delay service agreement (nominal delay) that will not be violated more often than in an unloaded n/w.

- Best-effort service class: similar to that which the internet currently offers, which is further parti-tioned into three categories i.e, interactive burst (eg, web); interactive bulk (eg, FTP) & asynchronous (eg, e-mail);

The main point is that guaranteed service & controlled load classes are based on quantitative service requirements & both require signalling & admission control in n/w nodes. These services can be provided either per-flow or per-flow

④

aggregate, depending on flow concentration at different points in the n/w. Although the ISA need not be tied to any particular signalling protocol, Resource Reservation Protocol (RSVP) ~~described~~ is often regarded as the signalling protocol in ISA.

The major ~~advantage of~~ char. of ISA is that it leaves the existing best-effort service class mostly unchanged (except for the further subdivision of the class), so it doesn't involve any change to existing application. ISA also leaves the forwarding mechanism in the n/w unchanged.

Although ISA is a straight forward QoS model, end-to-end service guarantees can't be supported unless all nodes along the route support ISA.

* Differentiated Service Architecture

It minimise the signalling & concentrates an aggregated flows & per hop behaviour (PHB) applied to a n/w-wide set of the traffic classes. Arriving flows are classified according to pre-det-ermined rules, which aggregate many application flows into a lim-ited & manageable set (perhaps 2 to 8) of class flows.

Traffic entering the n/w domain at the edge router is first classified for the consistent treatment at each transit router inside the n/w. Treatment will usually be applied by separat-ing traffic into different queues according to the class of traffic, so that high priority pkts can be assigned the appro-priate priority level at an o/p port.

Differentiated service Architecture approach separates the cla-ssification & queuing functions. Pkts carry self-evident pri-ority marking in the Type-of-Service byte inside pkt. headers. (Type-of-Service is part of the legacy IP archite-cture).

DSA outlines an initial architectural philosophy that serves as a framework for inter-provider agreement & makes it poss-ible to extended QoS beyond a single n/w domain. The DSA is more scalable than ISA because it handles flow aggregates & minimizes signalling, thus avoiding the complexity of per-flow soft states at each node. DSA will likely be applied more commonly in ~~inter~~ enterprise backbones & in service provider networks.

To provide QoS support, a n/w must somehow allow for the controlled unfairness in the use of its resources. Controlling to a granularity as fine as a flow of data requires advanced signalling protocols.

In DSA, scalability & flexibility are achieved by following hierarchical model for n/w resource management;

Interdomain Resource Management: Unidirectional service levels, & hence traffic contracts, are agreed at each boundary point betn. a customer & a provider for the traffic entering the provider n/w.

Intradomain Resource Management: The service provider is solely responsible for the configuration & provisioning of resources within it's domain (i.e, the n/w). Furthermore, service policies are also left to the provider.

The DSA is an elegant way to provide much needed service discrimination within a commercial n/w. Customers willing to pay more will see their applications receive better service than those paying less. This scheme exhibits an "auto-funding" property: "popular" traffic classes generate more revenues, which can be used to increase their provisioning.

MPLS

... about what policies should be supported in

Multi-protocol Label Switching (MPLS)

MPLS is a data carrying mechanism which emulates some properties of a circuit switched networks over a packet switched network. MPLS operates at an OSI model layer that is generally considered to lie between traditional definitions of layer 2 (data link layer) and layer 3 (network layer), and thus is often referred to as "Layer 2.5" protocol. It was designed to provide a unified data-carrying service for both circuit-based clients and packet-switching clients which provide a datagram service model. It can be used to carry different kinds of traffic, including IP packets, as well as native ATM, SONET, and Ethernet frames.

MPLS is replacing the previous technologies such as frame relay and ATM which were deployed with essential identical goals. This is because MPLS is better aligned with current and future technology needs.

MPLS distribute with cell-switching and signaling-protocol baggage of ATM. MPLS recognizes that small ATM cells are not needed in the core of modern networks, since modern optical networks are so fast (at 10 Gbit/s and well beyond) that even full length 1500 byte do not incur significant real-time queuing delays (the need to reduce such delays, to support voice traffic, having been the motivation for the cell nature of ATM). At the same time it attempts to preserve the traffic engineering and out-of-band control that frame relay and ATM attractive for deploying large scale networks.

MPLS is considered in:

- High speed IP backbones
- Legacy ATM
- MPLS capable ATM
- Optical fibre networks

How MPLS works

MPLS works by preappending packets with an MPLS header, containing one or more 'labels'. This is called a label stack.

Each label stack entry contains four fields: 32 bit

- A 20-bit label value
- A 3-bit field for QoS priority (experimental)

- A 1-bit bottom of stack flag. If this is set, it signifies the current label is the last in the stack.
- An 8-bit TTL (time to live) field

MPLS versus ATM:

- Both MPLS and ATM provide a connection-oriented service for transporting data across computer networks.
- MPLS is able to work with variable length packets while ATM transports fixed-length (53-byte) cells.
- In ATM networks packets must be segmented, transported, and re-assembled, which adds significant complexity and overhead to the data stream. MPLS on the otherhand, simply adds a label to the head of each packet and transmits it on the network.
- Other difference is the nature of connection. An MPLS connection (LSP) is unidirectional whereas ATM point-to-point connections (Virtual Circuits) on the other hand are bi-directional; PVC ATM connections are unidirectional.
- Both ATM and MPLS support tunneling of connections inside connections. MPLS uses label stacking to accomplish this while ATM uses virtual paths.
- One biggest advantage of MPLS is that it is complementary to IP, whereas ATM's are incompatible with IP which make ATM's largely unsuitable in today's predominantly IP networks.

Resource Reservation Protocol

The *Resource Reservation Protocol (RSVP)* is a network-control protocol that enables Internet applications to obtain differing qualities of service (QoS) for their data flows. Such a capability recognizes that *different applications have different network performance requirements*. Some applications, including the more traditional interactive and batch applications, require reliable delivery of data but do not impose any stringent requirements for the timeliness of delivery. Newer application types, including videoconferencing, IP telephony, and other forms of multimedia communications require almost the exact opposite: Data delivery must be timely but not necessarily reliable. Thus, ***RSVP was intended to provide IP networks with the capability to support the divergent performance requirements of differing application types***. It is important to note that *RSVP is not a routing protocol*. RSVP works in conjunction with routing protocols and installs the equivalent of dynamic access lists.

receiver to request a special and do

Real Time Transport Protocol.

The Real Time Transport Protocol (or RTP) defines a standardized packet format for delivering audio and video over the internet. RTP was first developed in 1996 as RFC 1889 and was made obsolete in 2003 by RFC 355. RTP can also be used in conjunction with RSVP protocol which enhances the field of multimedia applications.

RTP dose not have a standard TCP or UDP port on which it communicates. The only standard that obeys is that UDP communications are done via an even port and the next higher odd port is used for RTP control protocol (RTCP) communications.

RTP can carry any data with real time characteristics such as interactive audio and video. Call setup and tear-down is usually performed by the SIP protocol.

RTP was originally designed as a multicast protocol, but has since been applied in many unicast applications. It is frequently used in streaming media systems as well as videoconferencing and push to talk systems, making it the technical foundation of VoIP industry. It goes along with RTCP and is built on top of the UDP.

The services provided by RTP include:

- Payload-type identification- Indication of what kind of content is being carried.
- Sequence numbering- PDU sequence number
- Time stamping- allow synchronization and jitter calculations
- Delivery monitoring

RTP however do not provide mechanisms to ensure timely delivery and does not guarantees QoS. These things have to be provided by some other mechanisms.

Using RTP can lead to out-of-order delivery, also RTP does not directly support flow control and congestion control. However, the protocols do deliver the necessary data to the application to make sure it can put the received packets in the correct order. RTCP provides information about reception quality which the application can use to make local adjustments. Eg. If congestion is forming, the application could decide to lower the data rate.

Session Initiation Protocol

SIP (Session Initiation Protocol) is an application-layer control protocol that can establish, modify, and terminate multimedia sessions such as Internet telephony calls (VOIP). SIP can also invite participants to already existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility - users can maintain a single externally visible identifier regardless of their network location.

SIP supports five facets of establishing and terminating multimedia communications:

- ***User location:*** determination of the end system to be used for communication;
- ***User availability:*** determination of the willingness of the called party to engage in communications;
- ***User capabilities:*** determination of the media and media parameters to be used;
- ***Session setup:*** "ringing", establishment of session parameters at both called and calling party;
- ***Session management:*** including transfer and termination of sessions, modifying session parameters, and invoking services.

SIP is a component that can be used with other IETF protocols to build a complete multimedia architecture, such as the Real-time Transport Protocol (RTP) for transporting real-time data and providing QoS feedback, the Real-Time streaming protocol (RTSP) for controlling delivery of streaming media, the Media Gateway Control Protocol (MEGACO) for controlling gateways to the Public Switched Telephone Network (PSTN), and the Session Description Protocol (SDP) for describing multimedia sessions. Therefore, SIP should be used in conjunction with other protocols in order to provide complete services to the users. However, the basic functionality and operation of SIP does not depend on any of these protocols.

SIP provides a suite of security services, which include denial-of-service prevention, authentication (both user to user and proxy to user), integrity protection, and encryption and privacy services.

SIP works with both IPv4 and IPv6. For Internet telephony sessions, SIP works as follows:
Callers and callees are identified by SIP addresses. When making a SIP call, a caller first locates the appropriate server and then sends a SIP request. The most common SIP operation is the invitation. Instead of directly reaching the intended callee, a SIP request may be redirected or may trigger a chain of new SIP requests by proxies. Users can register their location(s) with SIP servers. SIP addresses (URL) can be embedded in Web pages and therefore can be integrated as part of powerful implementations such as Click to talk.

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Protocol Structure - SIP (Session Initiation Protocol)

SIP messages can be transmitted either over TCP or UDP SIP messages are text based and use the ISO 10646 character set in UTF-8 encoding. Lines must be terminated with CRLF. Much of the message syntax and header field are similar to HTTP. Messages can be request messages or response messages.

Request message has the following format:

Method	Request URI	SIP version

- **Method** - The method to be performed on the resource. Possible methods are Invite, Ack, Options, Bye, Cancel, Register.
- **Request-URI** - A SIP URL or a general Uniform Resource Identifier, this is the user or service to which this request is being addressed.
- **SIP version** - The SIP version being used.

The format of the Response message header is shown in the following illustration:

SIP version	Status code	Reason phrase

- **SIP version** - The SIP version being used.
- **Status-code** - A 3-digit integer result code of the attempt to understand and satisfy the request.
- **Reason-phrase** - A textual description of the status code.

Megaco Signaling Protocol

Megaco H.248, the Media Gateway Control Protocol, is for control of elements in a physically decomposed multimedia gateway, which enables separation of call control from media conversion. The Media Gateway Control Protocol (Megaco) is a result of joint efforts of the IETF and the ITU-T Study Group 16. Therefore, the IETF defined Megaco is the same as ITU-T Recommendation H.248.

Megaco/H.248 addresses the relationship between the Media Gateway (MG), which converts circuit-switched voice to packet-based traffic, and the Media Gateway Controller (MGC, sometimes called a call agent or softswitch, which dictates the service logic of that traffic). Megaco/H.248 instructs an MG to connect streams coming from outside a packet or cell data network onto a packet or cell stream such as the Real-Time Transport Protocol (RTP). Megaco/H.248 is essentially quite similar to MGCP from an architectural standpoint and the controller-to-gateway relationship, but Megaco/H.248 supports a broader range of networks, such as ATM.

There are **two basic components** in Megaco/H.248: **terminations and contexts.** Terminations represent streams entering or leaving the MG (for example, analog telephone lines, RTP streams, or MP3 streams). Terminations have properties, such as the maximum size of a jitter buffer, which can be inspected and modified by the MGC.

Terminations may be placed into contexts, which are defined as when two or more termination streams are mixed and connected together. The normal, "active" context might have a physical termination (say, one DS0 in a DS3) and one ephemeral one (the RTP stream connecting the gateway to the network). Contexts are created and released by the MG under command of the MGC. A context is created by adding the first termination, and it is released by removing (subtracting) the last termination.

A termination may have more than one stream, and therefore a context may be a multistream context. Audio, video, and data streams may exist in a context among several terminations.

Protocol Structure - Megaco/H.248 (Media Gateway Control Protocol)