Q - Can DNS use TCP? In which cases DNS uses TCP?
A - DNS uses TCP for Zone Transfer over Port: 53 TCP is used if the size of the packet goes over 512 bytes. It is necessary to maintain a consistent DNS database between DNS Servers. This is achieved by the TCP protocol. This communication happens between DNS Servers only. The connection is established between the DNS Server to transfer the zone data and Source and Destination DNS Servers will make sure that data is consistent by using TCP ACK bit.

DNS uses UDP for DNS Queries over Port: 53
A client computer will always send a DNS Query using UDP Protocol over Port 53. The query and response work can be done in one RTT. So this approach is fast as compared to TCP as in case of TCP, connection has to established first. If a client computer does not get response from a DNS Server, it must re-transmit the DNS Query using the TCP after 3-5 seconds of interval.

Q - How does BGP work?
A - BGP is an Exterior Gateway Routing Protocol and is used to do routing between ASs or ISPs. Has 4 message types: Open(Open a BGP session with the peer), Keeplalive(duh!), Update(Routing updates) and Notification(Bad!). Main reason to use BGP is influence inbound or outbound(mostly) routes. This means giving preference to one path over other depending on various factors and policies. Has a long metric list. N-W-L-Li-AS path-O-M-N-I. Has two flavors. eBGP and iBGP.

TSHOOT - Before Neighbors:
Loopback reachability, Neighbor command verification for right AS, update source

TSHOOT - After Neighbors (Routes not showing up in the RT):
Next hop self, BGP Synchronization

Q - How does a Juniper Router decides its path (BGP)?
A - Discuss BGP metric, ties and tie-breakers

Q - Types of Routing? Types of Dynamic Routing Protocols?
A - Static and Dynamic. LSRP and DVRP

Q - What's the difference between a router and switch?
A - **Router**: L3 device, High forwarding rates, Highly configurable, connects two LANs, WANs, PDU-Packet, each port has own broadcast domain, does forwarding on software logic (Routing Tables)
Switch: L2 device, less configurable, connects devices in a LAN, PDU-Frame, each port has own collision domain and whole switch is one broadcast domain, does forwarding on hardware logic (CAM and ASICs)

Q - Define bandwidth
A - bandwidth is technically the bit-rate of available data capacity of a network. The width of the band limits the data rate that can be carried on the medium Bandwidth is measured in kilobits per second or “Kbps.”

Q - Define throughput
A - bit rate (bits/time unit) at which bits are transferred between sender/receiver

Q - Two switching techniques

A - **Store-and-Forward:** Store-and-Forward switching will wait until the entire frame has arrived prior to forwarding it. This method stores the entire frame in memory. So if there are 3 links between the source and the destination, the transmission delay Dtr will be: 3L/R (rate of transmission for 3 links = 3* * 1/R) Once the frame is in memory, the switch checks the destination address, source address, and the CRC. If no errors are present, the frame is forwarded to the appropriate port. This process ensures that the destination network is not affected by corrupted or truncated frames.

Cut-Through: Cut-Through switching will begin forwarding the frame as soon as the destination address is identified. The difference between this and Store-and-Forward is that Store-and-Forward receives the whole frame before forwarding.Since frame errors cannot be detected by reading only the destination address, Cut-Through may impact network performance by forwarding corrupted or truncated frames. These bad frames can create broadcast storms wherein several devices on the network respond to the corrupted frames simultaneously.

Q - DISADV of Packet Switching

A - Queuing of packets can lead to excessive congestion. Which leads to Packet Delay and Loss. We need congestion control mechanisms.

Q - Types of delay

A - D_Node = D_proc(check bit errors, determine output link) + D_queue(router buffer delay) + D_trans(L/R) + D_prop(length of physical link/medium propagation speed)

Q - why we need checksum in both layer-2 and layer-3 of OSI layer

A - Simply put, different layers of the OSI model have checksums so you can assign blame appropriately. Suppose there is a web server running on some system (assume TCP port 80, i.e. OSI Layer 4) Suppose there is a software error in the Operating System of that webserver that corrupts certain IP payloads. (IPv4 OSI Layer 3)

If we only rely on Ethernet (i.e. OSI Layer 2) checksums, then that error would go un-noticed until something crashes or throws an error, because the Ethernet NIC would simply transmit the (already corrupted) data that it received from the Operating System IP stack. However, if TCP, IP and Ethernet all have checksums, we can isolate the layer where the error occurred, and notify the appropriate Operating System or application component of the error.

Q - What is IP Fragmentation? What are its disadvantages? How to avoid fragmentation?

A - The process of breaking the datagrams into smaller pieces so that they can be transmitted on a link with a smaller **MTU** (MTU of a layer is the size of the largest PDU that the layer/link can pass/transfer) than the original datagram size.

MSS is the largest amount of data that a communication device can receive in a single TCP segment. This does not count the TCP header and the IP header (20 bytes each).

Small MSS would mean reduction or elimination of the need of IP fragmentation. This is because a TCP segment will be packed inside an IP datagram. IP datagrams have their own size issues (MTU). If the segment size is large, the IP datagram will be large too which would increase the need of IP fragmentation.

Disadvantage of IP fragmentation:

1. There is a small increase in CPU and memory overhead to fragment an IP datagram. This holds true for the sender as well as for a router in the path between a sender and a receiver. Creating fragments simply involves creating fragment headers and copying the original datagram into the fragments. IP fragmentation can cause excessive retransmissions when fragments encounter packet loss and reliable protocols such as TCP must retransmit all of the fragments in order to recover from the loss of a single fragment. Fragmentation causes more overhead for the receiver when reassembling the fragments because the receiver must allocate memory for the arriving fragments and coalesce them back into one datagram after all of the fragments are received.

Reassembly on a host is not considered a problem because the host has the time and memory resources to devote to this task. But, reassembly is very inefficient on a router whose primary job is to forward packets as quickly as possible. A router is not designed to hold on to packets for any length of time. Also a router that does reassembly chooses the largest buffer available (18K) with which to work because it has no way to know the size of the original IP packet until the last fragment is received.

2. Another fragmentation issue involves how dropped fragments are handled. If one fragment of an IP datagram is dropped, then the entire original IP datagram must be resent,
and it will also be fragmented.

Senders typically use two approaches to decide the size of IP datagrams to send over the network.
a) The first is for the sending host to send an IP datagram of size equal to the MTU of the first hop of the source destination pair.
b) The second is to run the path **MTU discovery (PMTUD) algorithm:**
TCP MSS as described earlier takes care of fragmentation at the two endpoints of a TCP connection, but it does not handle the case where there is a smaller MTU link in the middle between these two endpoints. PMTUD was developed in order to avoid fragmentation in the path between the endpoints. It is used to dynamically determine the lowest MTU along the path from a packet's source to its destination.
For IPv4 packets, Path MTU Discovery works by setting the Don't Fragment (DF) flag bit in the IP headers of outgoing packets. Then, any device along the path whose MTU is smaller than the packet will drop it, and send back an Internet Control Message Protocol (ICMP) Fragmentation Needed (Type 3, Code 4) message containing its MTU, allowing the source host to reduce its Path MTU appropriately. The process is repeated until the MTU is small enough to traverse the entire path without fragmentation.

IPv6 routers do not support fragmentation or the Don't Fragment option. For IPv6, Path MTU Discovery works by initially assuming the path MTU is the same as the MTU on the link layer

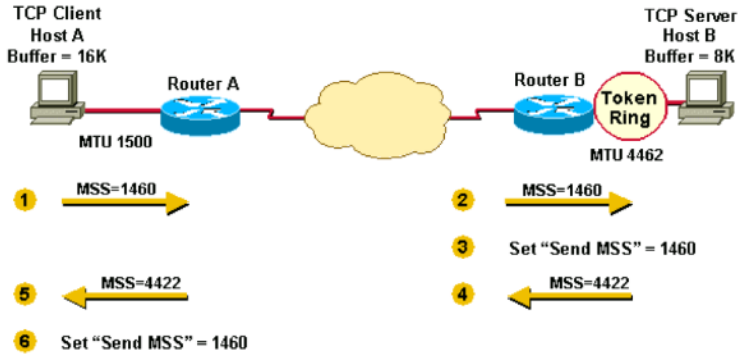
interface where the traffic originates. Then, similar to IPv4, any device along the path whose MTU is smaller than the packet will drop the packet and send back an ICMPv6 Packet Too Big (Type 2) message containing its MTU, allowing the source host to reduce its Path MTU appropriately. The process is repeated until the MTU is small enough to traverse the entire path without fragmentation.

In order to assist in avoiding IP fragmentation at the endpoints of the TCP connection, the selection of the MSS value was changed to the minimum buffer size and the MTU of the outgoing interface (- 40). MSS numbers are 40 bytes smaller than MTU numbers because MSS is just the TCP data size, which does not include the 20 byte IP header and the 20 byte TCP header. MSS is based on default header sizes; the sender stack must subtract the appropriate values for the IP header and the TCP header dependent on what TCP or IP options are used.

The way MSS now works is that each host will first compare its outgoing interface MTU with its own buffer and choose the lowest value as the MSS to send. The hosts will then compare the MSS size received against their own interface MTU and again choose the lower of the two values.

Scenario 2 illustrates this additional step taken by the sender in order to avoid fragmentation on the local and remote wires. Notice how the MTU of the outgoing interface is taken into account by each host (before the hosts send each other their MSS values) and how this helps to avoid fragmentation.

Scenario 2



1. Host A compares its MSS buffer (16K) and its MTU (1500 - 40 = 1460) and uses the lower value as the MSS (1460) to send to Host B.
2. Host B receives Host A's send MSS (1460) and compares it to the value of its outbound interface MTU - 40 (4422).
3. Host B sets the lower value (1460) as the MSS for sending IP datagrams to Host A.
4. Host B compares its MSS buffer (8K) and its MTU (4462-40 = 4422) and uses 4422 as the MSS to send to Host A.
5. Host A receives Host B's send MSS (4422) and compares it to the value of its outbound interface MTU -40 (1460).
6. Host A sets the lower value (1460) as the MSS for sending IP datagrams to Host B.

1460 is the value chosen by both hosts as the send MSS for each other. Often the send MSS value will be the same on each end of a TCP connection.

In Scenario 2, fragmentation does not occur at the endpoints of a TCP connection because both outgoing interface MTUs are taken into account by the hosts. Packets can still become fragmented in the network between Router A and Router B if they encounter a link with a lower MTU than that of either hosts' outbound interface.

Fig: How end to end fragmentation is avoided

Large Segment Offload or (TSO - TCP segment offload)

When a system needs to send large chunks of data out over a computer network, the chunks first need breaking down into smaller segments that can pass through all the network elements like routers and switches between the source and destination computers. This process is referred to as segmentation. Often the TCP protocol in the host computer performs this segmentation. Offloading this work to the NIC is called TCP segmentation offload (TSO).

For example, a unit of 64kB (65,536 bytes) of data is usually segmented to 46 segments of 1448 bytes each before it is sent through the NIC and over the network. With some intelligence in the NIC, the host CPU can hand over the 64 KB of data to the NIC in a single transmit-request, the NIC can break that data down into smaller segments of 1448 bytes, add the TCP, IP, and data link layer protocol headers - according to a template provided by the host's TCP/IP stack - to each segment, and send the resulting frames over the network. This significantly reduces the work done by the CPU. As of 2014 many new NICs on the market support TSO.

What is the ROMMON Mode?
A - The ROM Monitor is a bootstrap program that initializes the hardware and boots the Cisco IOS XR software when you power on or reload a router. A version of the ROM Monitor software exists on each card. If the Cisco IOS XR software cannot boot on a card, the card startup ends in ROM Monitor mode

SDN Architecture

A- **SDN** - The separation of the control plane and the data plane of a network. The **Controller** in Control Plane is the brain of the Networking Device. Rather than each device having its own brain, the Controller is a single brain controlling hundreds of devices.

Benefits - Easier management(policies). The Controller can push centralized polices and configuration down to the data plane. An application programmer can create an application that commands the Controller through the REST or JAVA API(North Bound Communication using the North Bound API). The controller then uses OpenFlow(South Bound API) to update the flow tables in the switches giving them instruction of the flow.(South Bound Communication). OpenFlow only updates the flow tables. Does not install routing protocols.

OpenFlow: is an open source communication interface that allows access and manipulation of the data plane of a device (both physical and hypervisor) via a controller.

3 Layers of SDN Architecture:

1. Application Layer - Where the Application resides. This application talks to the Controller using the REST or JAVA API
2. Control Layer - The Control Plane where the Controller resides. Receives communication from the application(North) and sends instructions to the
Infrastructure layer via the OpenFlow
3. Infrastructure Layer - The Data Plane. Performs forwarding.

East West Communication - Controller X talks to Controller Y in the Control Layer.

CONGESTION AVOIDANCE

Q - Fair Queuing and Weighted Fair Queuing (WFQ) and QoS and Congestion Topics

Both FQ and WFQ are a way **(algorithms)** packet scheduling takes place in networking devices like routers and switches.

Fair Queuing: The concept implies a separate data packet queue (or job queue) for each traffic flow (or for each program process) as opposed to the traditional approach with one FIFO queue for all packet flows (or for all process jobs).

WFQ: a natural generalization of fair queuing (FQ): whereas FQ shares the link's capacity in equal subparts, WFQ allows to specify, for each flow, which fraction of the capacity will be given.

WFQ can be utilized by controlling the QoS.

QoS: quality of service refers to traffic prioritization and resource reservation control mechanisms. It is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow.

Protocols that are used to achieve QoS:

1. The type of service (ToS) field in the IP(v4) header (now superseded by DiffServ):
The modern redefinition of the ToS field is a six-bit Differentiated Services Code Point (DSCP) field and a two-bit Explicit Congestion Notification (ECN) field. While Differentiated Services is somewhat backwards compatible with ToS, ECN is not.

ECN: Conventionally, TCP/IP networks signal congestion by dropping packets. **But** when ECN is successfully negotiated, an ECN-aware router may set a mark in the IP header instead of dropping a packet in order to signal impending congestion. The receiver of the packet echoes the congestion indication to the sender, which reduces its transmission rate as if it detected a dropped packet.
Use of ECN on a TCP connection is optional; for ECN to be used, it must be negotiated at connection establishment by including suitable options in the SYN and SYN-ACK segments.
(the underlying network infrastructure should also support it.)

2. Differentiated services (DiffServ) (DSCP)
Differentiated services or DiffServ is a computer networking architecture that specifies a simple, scalable and **coarse-grained mechanism** for classifying and managing network traffic and providing quality of service (QoS) on modern IP networks.

DiffServ operates on the principle of **traffic classification**, where each data packet is placed into a limited number of traffic classes, rather than differentiating network traffic based on the requirements of an individual flow. **Each router on the network is configured to differentiate traffic based on its class.** Each traffic class can be managed differently, ensuring preferential treatment for higher-priority traffic on the network.

3. Integrated services (IntServ)

IntServ specifies a fine-grained QoS system, which is often contrasted with DiffServ's coarse-grained control system. There are two parts to a flow spec:

1. **What does the traffic look like?** Done in the Traffic SPECification part, also known as TSPEC.

TSPECs include **token bucket algorithm** parameters. The idea is that there is a token bucket which slowly fills up with tokens, arriving at a constant rate. Every packet which is sent requires a token, and if there are no tokens, then it cannot be sent. Thus, the rate at which tokens arrive dictates the average rate of traffic flow, while the depth of the bucket dictates how 'bursty' the traffic is allowed to be.

2. **What guarantees does it need?** Done in the service Request SPECification part, also known as RSPEC.

RSPECs specify what requirements there are for the flow: it can be normal internet 'best effort', in which case no reservation is needed. This setting is likely to be used for webpages, FTP, and similar applications.

4. Resource Reservation Protocol (RSVP)

A protocol designed **to reserve resources across a network** for an integrated services Internet. RSVP operates over an IPv4 or IPv6 Internet Layer and **provides receiver-initiated setup of resource reservations** for multicast or unicast data flows with scaling and robustness. It does not transport application data but is similar to a control protocol, like Internet Control Message Protocol (ICMP)

5. RSVP-TE

Resource Reservation Protocol - Traffic Engineering **is an extension** of the resource reservation protocol (RSVP) for traffic engineering. It supports the reservation of resources across an IP network. Applications running on IP end systems can use RSVP to **indicate to other nodes the nature (bandwidth, jitter, maximum burst, and so forth) of the packet streams they want to receive.**

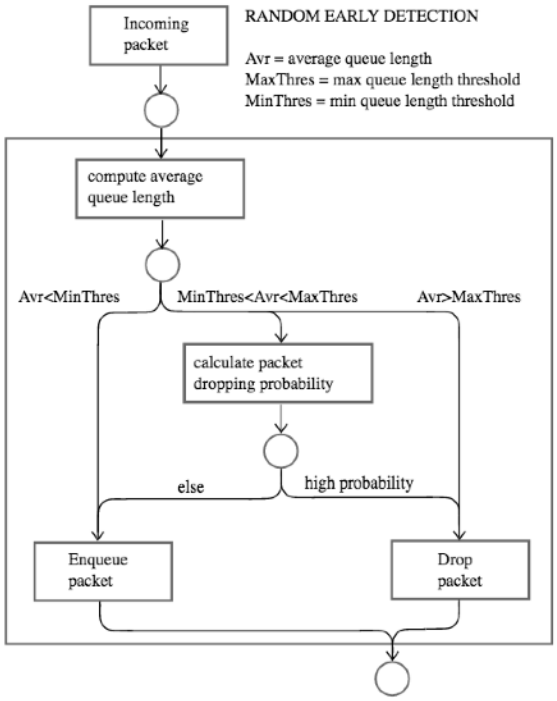
RSVP-TE generally **allows the establishment of MPLS label switched paths (LSPs)**, taking into consideration network constraint parameters such as available bandwidth and explicit hops.

6. Multiprotocol Label Switching (MPLS) provides eight QoS classes

DiffServ and IntServ Difference:

DiffServ is a coarse-grained, class-based mechanism for traffic management. In contrast, IntServ is a fine-grained, flow-based mechanism.

Congestion Avoidance Mechanisms in routers using queueing disciplines(Tail Drop, RED, WRED, Flow based WRED):



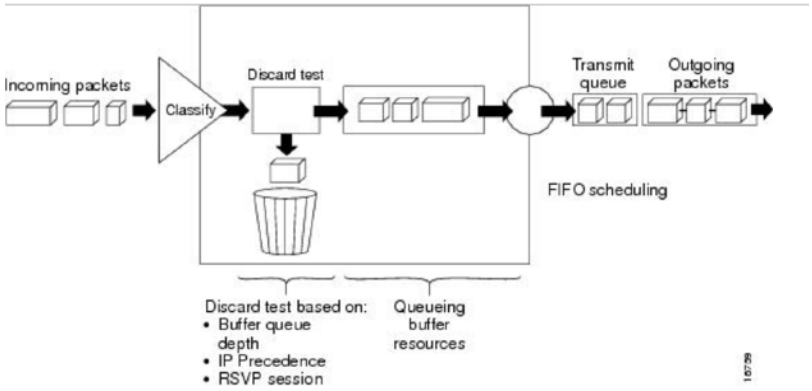
Queuing principles: are queue management algorithms used by Internet routers, e.g. in the network schedulers, and network switches to decide when to drop packets

Tail Drop:
Tail Drop, or Drop Tail, is a very simple. In contrast to the more complex algorithms like RED and WRED, **in Tail Drop the traffic is not differentiated. Each packet is treated identically.** With tail drop, when the queue is filled to its maximum capacity, the newly arriving packets are dropped until the queue has enough room to accept incoming traffic.
PROBLEMS: Tail drop leads to the problem of **TCP Global Synchronization** - as all TCP connections "hold back" simultaneously, and then step forward simultaneously. Because each sender will reduce the transmission rate at the same time when packet loss occurs.

Random Early Detection (RED):
Random early detection (RED), also known as *random early discard* or *random early drop* addresses this problem. RED monitors the average queue size and drops (or marks when used in conjunction with ECN) packets based on statistical probabilities. If the buffer is almost empty, all incoming packets are accepted. **As the queue grows, the probability for dropping an incoming packet grows too**. When the buffer is full, the probability has reached 1 and all incoming packets are dropped.
RED is more fair than tail drop, in the sense that it does not possess a bias against bursty traffic that uses only a small portion of the bandwidth. The more a host transmits, the more likely it is that its packets are dropped as the probability of a host's packet being dropped is proportional to the amount of data it has in a queue. **Early detection helps avoid TCP global synchronization.**

PROBLEMS: RED does not accommodate quality of service (QoS) differentiation. Its variants, WRED and DWRED do.

Weighted random early detection (WRED):
It is an extension to random early detection (RED) where a single queue may have several **different queue thresholds**. Each queue threshold is associated to a **particular traffic class**. WRED combines the capabilities of the RED algorithm with the IP Precedence feature to provide for preferential traffic handling of higher priority packets. WRED **can selectively discard lower priority traffic** when the interface begins to get congested and provide differentiated performance characteristics for different classes of service.
WRED **is also RSVP-aware**, and it can provide the controlled-load QoS service of integrated service.



Flow Based WRED:
Flow-based WRED is a feature that forces WRED to afford **greater fairness** to all flows on an interface **in regard to how packets are dropped.** Flow-based WRED relies on the following two main approaches to remedy the problem of unfair packet drop:
•It **classifies incoming traffic into flows based on parameters** such as destination and source addresses and ports.
•It maintains state about active flows, which are flows that have packets in the output queues.

CONGESTION AVOIDANCE IN UDP

UDP itself does not avoid congestion. Congestion control measures must be implemented at the application level.

RTP runs over UDP and RTCP(Real-time Transport Control Protocol) working with RTP provides measures for QoS(Quality of Service) like packet loss, delay, jitter, etc to report back to the sender so it knows when to slow down or change codecs.

DCCP - Datagram Congestion Control Protocol

DCCP provides a way to gain access to congestion control mechanisms without having to implement them at the application layer. It provides a congestion-controlled flow of unreliable datagrams. It allows for flow-based semantics like in Transmission Control Protocol (TCP), but does not provide reliable in-order delivery (this would mean overhead)
Challenge - TCP's congestion control is so tightly coupled to its reliable semantics that few TCP mechanisms are directly applicable without substantial change.

SCTP - Stream Control Transmission Protocol
Stream Control Transmission Protocol (SCTP) is a transport-layer protocol, serving in a similar role to the popular protocols TCP and UDP. SCTP provides some of the same service features of both: it is message-oriented like UDP and ensures reliable, in-sequence transport of messages with congestion control like TCP; it differs from these in providing multi-homing and redundant paths to increase resilience and reliability.
Multihoming support in which one or both endpoints of a connection can consist of more than one IP address, enabling transparent fail-over between redundant network paths.

//TODO
Throughput, Latency and Bandwidth
Low throughput remedies - Increasing TCP congestion window size, TCP acceleration, forward error correction.

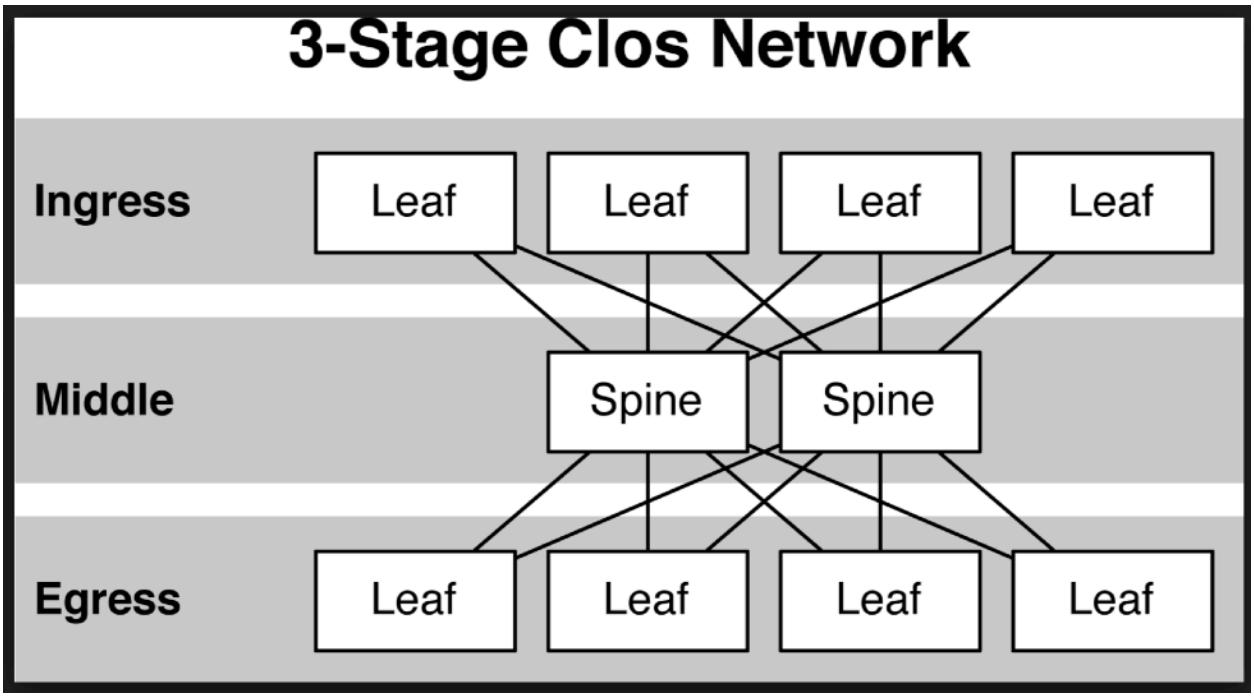
Network Topology Types

CLOS Network:
It is a type of multi-stage circuit switching network named after the inventor Charles Close. Clos networks are defined by three integers n, m, and r.

- n represents the number of sources which feed into each of r ingress stage crossbar switches.
- Each ingress stage crossbar switch has m outlets, and there are m middle stage crossbar switches. There is exactly one connection between each ingress stage switch and each middle stage switch.
- There are r egress stage switches, each with m inputs and n outputs. Each middle stage switch is connected exactly once to each egress stage switch.

Thus, the ingress stage has r switches, each of which has n inputs and m outputs. The middle stage has m switches, each of which has r inputs and r outputs. The egress stage has r switches, each of which has m inputs and n outputs.

Examples: TRILL, FabricPath, Brocade's VCS
Benefits: The key advantage of Clos networks is that the number of crosspoint(s) (which make up each crossbar switch) required can be far fewer than would be the case if the entire switching system were implemented with one large crossbar switch.


3-Stage Clos Network
Ingress
Middle
Egress
Leaf
Leaf
Leaf
Leaf
Spine
Spine
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Leaf
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