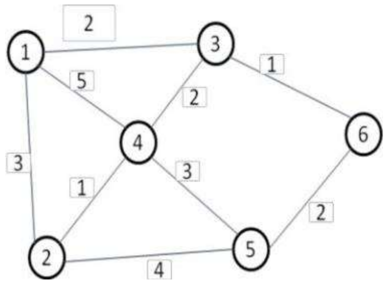
19. Consider the network given below use Dijktras algorithm to find shortest path from all nodes to destination node 2. **(10 M)**



UNIT II- Packet Switching Networks-2

Link-State Algorithm

- Basic idea: two stage procedure
- Each source node gets a map of all nodes and link metrics (link state) of the entire network
  -Learning who the neighbors are and what are delay to them
  - Construct a link state packet, and deliver it to other
- Find the shortest path on the map from the source node to all destination nodes;
  -Dijkstras algorithm Broadcast of link-state information
  -Every node i in the network broadcast to every other node in the network:

- It requires each link cost to be positive
- The main idea is to progressively identify the closest nodes from source node in order of increasing path cost
- The algorithm is iterative.
- The algorithm can be implemented by maintaining a set of N permanently labeled nodes, which consists of those nodes whose shortest paths have been determined.
- At each iteration the next closest node is added to the set N and the distance to the remaining nodes via the new node is evaluated.

Dijkstra Algorithm:
Finding shortest paths in order

- *N*: set of nodes for which shortest path already found
- Initialization: (*Start with source node s)*
  - *N = {s}, Ds* = 0, "*s* is distance zero from itself"
  - *Dj=Csj* for all $j \neq s$, distances of directly-connected neighbors
- Step A: (*Find next closest node i*)
  - Find $i \notin N$ such that
  - *Di* = min *Dj* for $j \notin N$
  - Add *i* to *N*
  - If *N* contains all the nodes, stop
- Step B: (*update minimum costs)*
  - For each node $j \notin N$
  - *Dj* = min (*Dj, Di+Cij*)

- Go to Step A

Execution of Dijkstra's algorithm

Iteration	**N**	D2	D3	D4	D5	D6
Initial	{1}	3	2	5	∝	∝
1	{1,3}	3	**2**	4	∝	3
2	{1,2,3}	**3**	**2**	4	7	3
3	{1,2,3,6}	**3**	**2**	4	5	**3**
4	{1,2,3,4,6}	**3**	**2**	**4**	5	**3**
5	{1,2,3,4,5,6}	**3**	**2**	**4**	**5**	**3**

Shortest Paths in Dijkstra's Algorithm

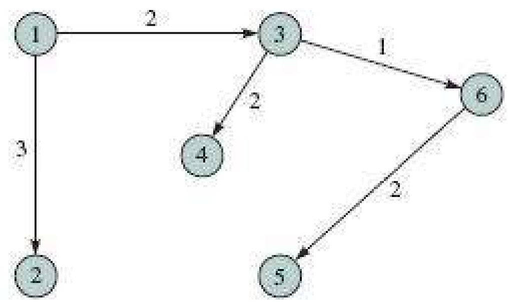


FIGURE 7.32 Shortest-path tree from node 1 to other nodes

Why is Link State Better?

- Fast, loopless convergence
- Support for precise metrics, and multiple metrics if necessary (throughput, delay, cost, reliability)
- Support for multiple paths to a destination
    - algorithm can be modified to find best two paths

Distance vector v/s Link state

Funtionality	***Distance vector***	***Link State***
Primary principle	Sends larger updates , about the complete network information only to neighboring routers	Sends smaller updates, about the link state of neighbors, to all routers
Learning about network	Learns about network only from neighbors	Learn about network only from all routers
Building the routing table	Based on inputs from only neighbors	Based on complete database collected from all routers
Advertisement of updates	Periodically (e.g. every 30 seconds)	Triggered updates, only when there is a change

Routing loops	More prone ; suffer from problems like count-to-infinity	Less prone to routing loops
Convergence (stabilisation)	Slow	Fast
Resources	Less CPU power and memory	More CPU power and memory required
Cost	Less cost	More than Distance vector
Scalability	Less scalable	More scalable than distance vector

Assignment Questions

1. Why is packet switching more suitable than message switching for interactive applications? Compare the delays in datagram packet switching and message switching. **(10 M)**
2. Differentiate between virtual circuits and datagram subnets. **(6M)**
3. Define routing. Explain Bellman-Ford routing algorithm with necessary illustration. What are the drawbacks of this algorithm? **(10 M)**
4. Explain Bellman-Ford routing algorithm with necessary illustration. What are the drawbacks of this algorithm?

   (10 M)

   Find the shortest path for below diagrams. Destination node is E for Fig1 and Destination node is 6 for Fig2

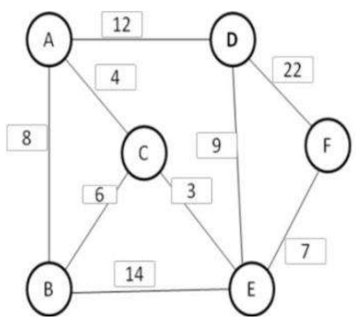


Fig 1

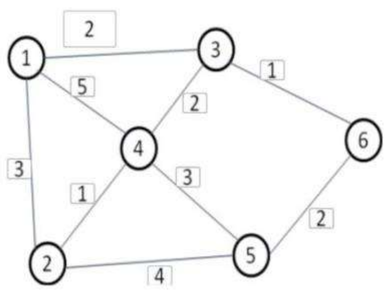
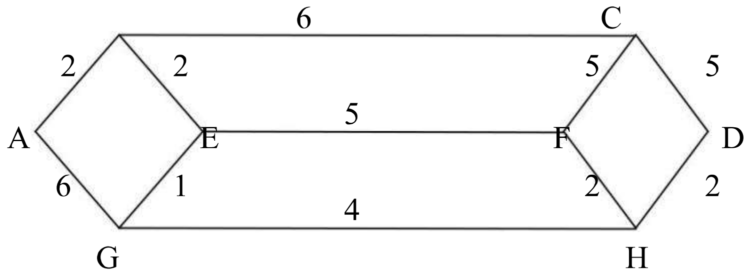
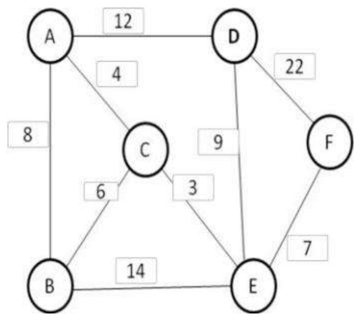


Fig 2

5 Using Dijkstra's algorithm find the shortest path between A and D



Find the shortest path between A and E for the below diagram



6. Compare the Bellman –Ford algorithm and Dijkstra's algorithm for finding the shortest paths from a source node to all other nodes in a network.

7. Good news spreads fast; bad news propagates slowly in bellman ford algorithm. Explain with an example.

8. Write short notes on i) Flooding ii) Deflection Routing iii) Hierarchical routing.

Traffic Management

- The main objectives of traffic management are efficient use of network resources & deliver QoS.
- Traffic Management is classified into three levels that are Packet level, Flow level and Flow aggregated level.

Traffic Management at Packet Level

- o Queueing & scheduling at switches, routers and multiplexers.

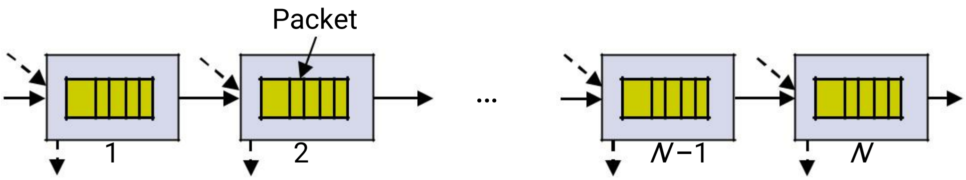


Figure: - End-to-End QoS of a packet along a path traversing N Queueing System

- The path traversed by packet through a network can be modeled as sequence of Queueing systems as shown in above figure.
- A packet traversing network encounters delay and possible loss at various multiplexing points.
- End-to-end performance is sum of the individual delays experienced at each system.
- Average end-to-end delay is the sum of the individual average delay.
- To meet the QoS requirements of multiple services, a queueing system must implement strategies for controlling the transmission bit rates.

The different strategies for Queue scheduling are:-

1. **FIFO QUEUEING**
2. **PRIORITY QUEUEING**
3. **FAIR QUEUEING**
4. **WEIGHTED FAIR QUEUEING**

1) FIFO QUEUEING

- Transmission Discipline: First-In, First-Out
- All packets are transmitted in order of their arrival.
- Buffering Discipline:- Discard arriving packets if buffer is full
- Cannot provide differential QoS to different packet flows
- Difficult to determine performance delivered
- Finite buffer determines a maximum possible delay
- Buffer size determines loss probability, but depends on arrival & packet length statistics.

FIFO Queueing with Discard Priority

FIFO queue management can be modified to provide different characteristics of packet-loss performance to different classes of traffic.

- The above Figure 7.42 (b) shows an example with two classes of traffic.
- When number of packets in a buffer reaches a certain threshold, arrivals of lower access priority (class 2) are not allowed into the system.
- Arrivals of higher access priority (class 1) are allowed as long as the buffer is not full.

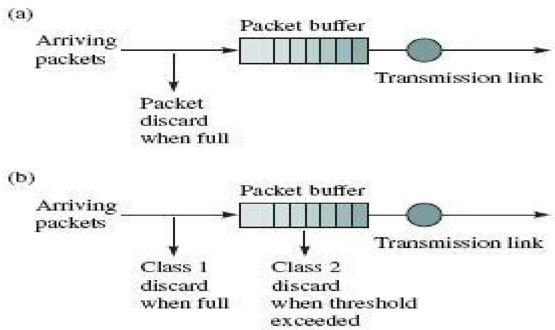


FIGURE 7.42 (a) FIFO queueing; (b) FIFO queueing with discard priority

2) Head of Line (HOL) Priority Queueing

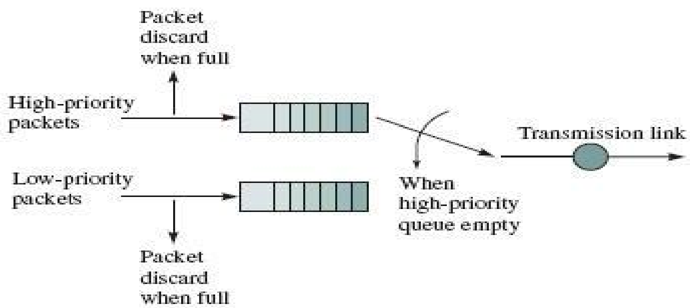


FIGURE 7.43 HOL priority queueing

- Second queue scheduling approach which defines number of priority classes.
- A separate buffer is maintained for each priority class.
- High priority queue serviced until empty and high priority queue has lower waiting time
- Buffers can be dimensioned for different loss probabilities
- Surge in high priority queue can cause low priority queue to starve for resources.
- It provides differential QoS.
- High-priority classes can hog all of the bandwidth & starve lower priority classes
- Need to provide some isolation between classes

Sorting packets according to priority tags/Earliest due Date Scheduling

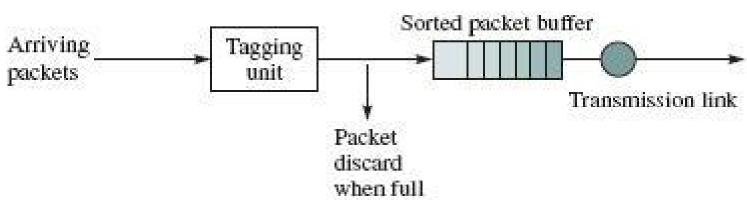


FIGURE 7.44 Sorting packets according to priority tag

- Third approach to queue scheduling
- Sorting packets according to priority tags which reflect the urgency of packet needs to be transmitted.
- Add Priority tag to packet, which consists of priority class followed by the arrival time of a packet.

- Sort the packet in queue according to tag and serve according to HOL priority system
- Queue in order of "due date".
- The packets which requires low delay get earlier due date and packets without delay get indefinite or very long due dates

3) Fair Queueing / Generalized Processor Sharing

- Fair queueing provides equal access to transmission bandwidth.
- Each user flow has its own logical queue which prevents hogging and allows differential loss probabilities
- C bits/sec is allocated equally among non-empty queues.
- The transmission rate = C / n bits/second, where n is the total number of flows in the system and C is the transmission bandwidth.
- Fairness: It protects behaving sources from misbehaving sources.
- Aggregation:
    - Per-flow buffers protect flows from misbehaving flows
    - Full aggregation provides no protection
    - Aggregation into classes provided intermediate protection
- Drop priorities:
    - Drop packets from buffer according to priorities
    - Maximizes network utilization & application QoS
    - Examples: layered video, policing at network edge.

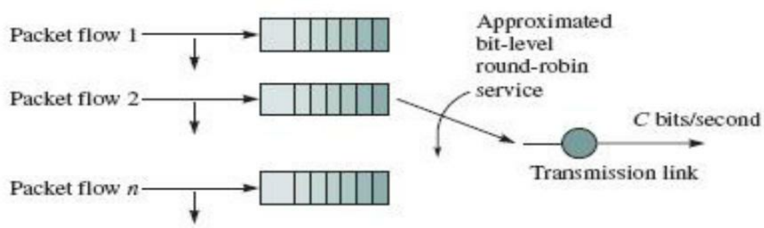


FIGURE 7.45 Fair queueing

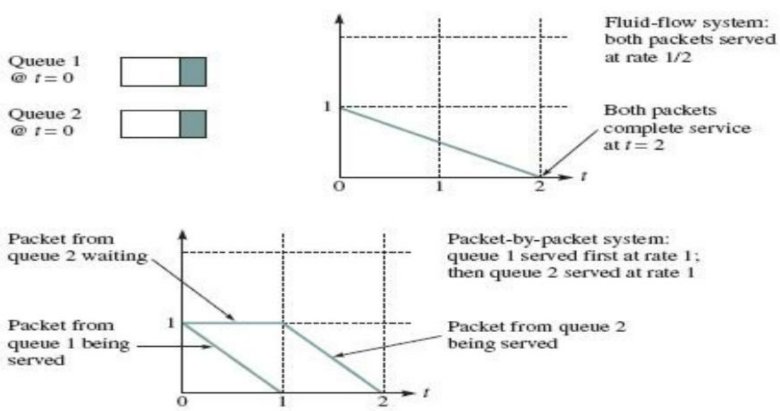


FIGURE 7.46 Fluid-flow and packet-by-packet fair queueing (two packets of equal length)

The above figure 7.46 illustrates the differences between ideal or fluid flow and packet-by-packet fair queueing for packets of equal length.

- Idealized system assumes fluid flow from queues, where the transmission bandwidth is divided equally among all non-empty buffers.
- The figure assumes buffer1 and buffer 2 has single L-bit packet to transmit at t=0 and no subsequent packet arrive.
- Assuming capacity of C=L bits/second=1 packet/second.
- Fluid-flow system transmits each packet at a rate of ½ and completes the transmission of both packets exactly at time=2 seconds.
- Packet-by-packet fair queueing system transmits the packet from buffer 1 first and then transmits from buffer 2, so the packet completion times are 1 and 2 seconds.

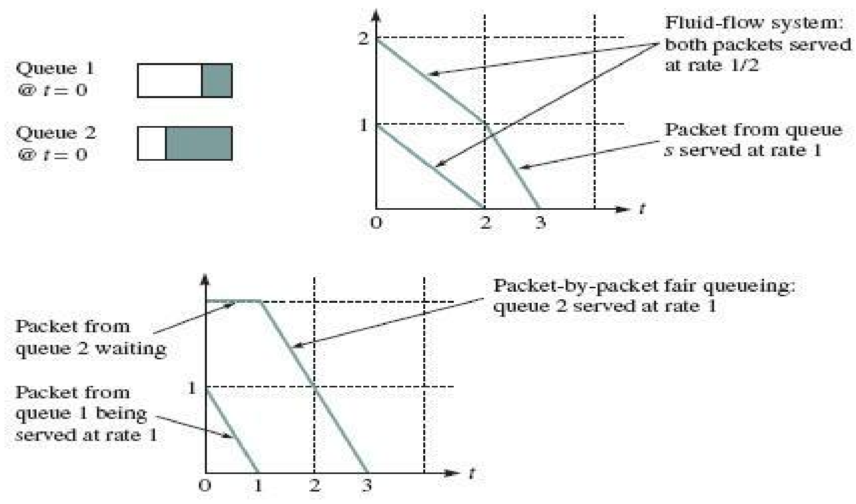


FIGURE 7.48 Fluid flow and packet-by-packet fair queueing (two packets of different lengths)

The above figure 7.48 illustrates the differences between ideal or fluid flow and packet-by-packet fair queueing for packets of variable length.

- The fluid flow fair queueing is not suitable, when packets have variable lengths.
- If the different user buffers are serviced one packet at a time in round-robin fashion, then we do not obtain fair allocation of transmission bandwidth.
- Finish tag is number used for the packet and the packet with smallest finish tag will be served first, and finish tag is computed as follows.
- Finish tag is used as priorities in packet-by-packet system.

Consider Bit-by-Bit Fair Queueing

- Assume n flows, n queues
- 1 round = 1 cycle serving all n queues
- If each queue gets 1 bit per cycle, then 1 round is the number of opportunities that each buffer has had to transmit a bit.
- Round number = number of cycles of service that have been completed

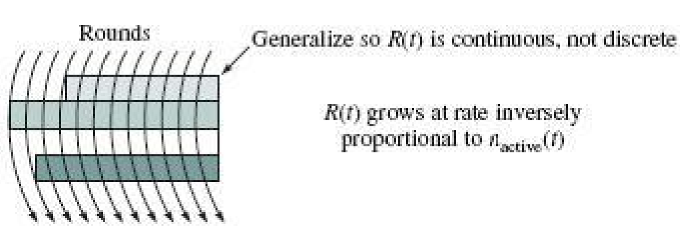


FIGURE 7.47 Computing the finishing time in packet-by-packet fair queueing and weighted fair queueing

- If packet arrives to idle queue:
  Finishing time = round number + packet size in bits
- If packet arrives to active queue:

Finishing time = finishing time of last packet in queue + packet size

Computing the Finishing Time

- *F(i,k,t)* = finish time of *k*th packet that arrives at time *t* to flow *i*
- *P(i,k,t)* = size of *k*th packet that arrives at time t to flow *i*
- *R(t)* = round number at time *t*

- Fair Queueing:

$$F(i,k,t) = \max\{F(i,k-1,t), R(t)\} + P(i,k,t)$$

4) Weighted Fair Queueing (WFQ)

- WFQ addresses the situation in which different users have different requirements.
- Each user flow has its own buffer and each user flow also has weight.
- Here weight determines its relative bandwidth share.
- If buffer 1 has weight 1 and buffer 2 has weight 3, then when both buffers are nonempty, buffer 1 will receive 1/(1+3)=1/4 of the bandwidth and buffer 2 will receive ¾ of the bandwidth.

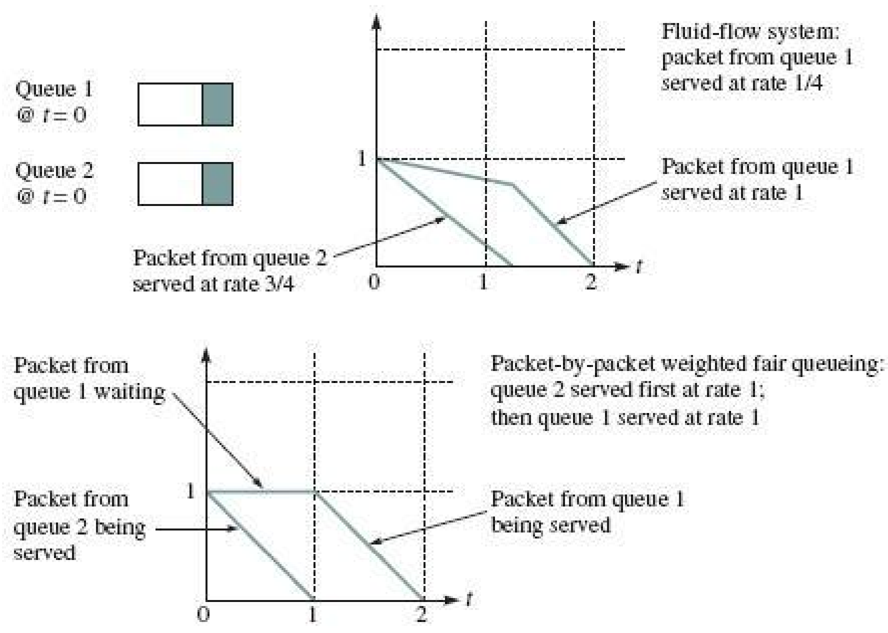


FIGURE 7.49 Fluid flow and packetized, weighted fair queueing

In the above figure,

- In Fluid-flow system, the transmission of each packet from buffer 2 is completed at time t=4/3, and the packet from buffer 1 is completed at t=2 seconds.
- In the above figure buffer1 would receive 1 bit/round and buffer 2 would receive 3 bits/second.
- Packet-by-packet weighted fair queueing calculates its finishing tag as follows

- The above figure also shows the completion times for Packet-by-packet weighted fair queueing.
- The finish tag for buffer1 is F(1,1)=R(0)+1/1 =1 and finish tag for buffer 2 is F(2,1) =R(0) + 1/3 =1/3.
- Therefore the packet from buffer 2 is served first and followed by packet from buffer 1.

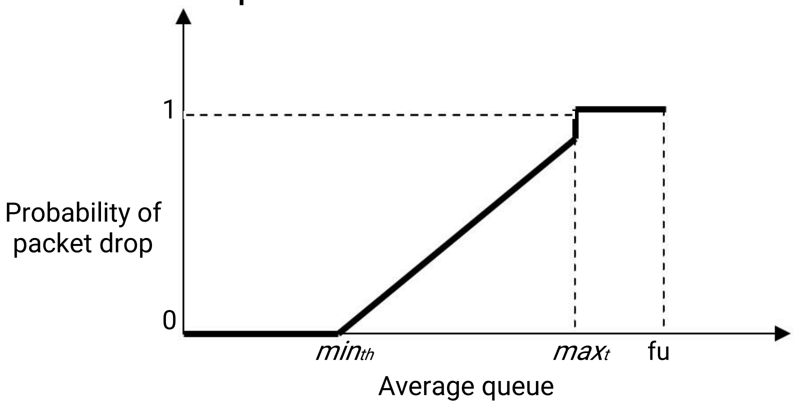
Buffer Management: - Random Early Detection (RED)

- An approach to preventing unfair buffer hogging by detecting congestion when a buffer begins to reach certain level and it notifies the source to reduce the rate at which they send packets.
- Packets produced by TCP will reduce input rate in response to network congestion
- RED is a buffer management technique that attempts to provide equal access to FIFO system by randomly dropping arriving packets before the buffer overflows.
- A dropped packet provides feedback information to the source and informs the source to reduce its transmission rate.
- Early drop: discard packets before buffers are full
- Random drop causes some sources to reduce rate before others, causing gradual reduction in aggregate input rate.
- Min_{th} and Max_{th} are the two thresholds defined
- RED algorithm uses average queue length, when average queue length is below Min_{th} , RED does not drop any arriving packets.
- When average queue length is between Min_{th} and Max_{th}, RED drops an arriving packet with an increasing probability as the average queue length increases.
- Packet drop probability increases linearly with queue length
- RED improves performance of cooperating TCP sources.
- RED increases loss probability of misbehaving sources

Algorithm:

- Maintain running average of queue length
- If Qavg < minthreshold, do nothing
- If Qavg > maxthreshold, drop packet
- If in between, drop packet according to probability
- Flows that send more packets are more likely to have packets dropped

Packet Drop Profile in RED



Traffic Management at the Flow Level

- Management of individual traffic flows & resource allocation to ensure delivery of QoS(e.g. Delay, jitter, loss)
- Traffic management at flow level operates on the order of milliseconds to seconds.
- It is concerned with managing the individual traffic flow to ensure the QoS (e.g. delay, jitter, loss) requested by user is satisfied.
- The purpose of Traffic Management at the Flow Level is to control the flows of traffic and maintain performance even in presence of traffic overload.
- The process of managing the traffic flow in order to control congestion is called congestion control.
- Congestion occurs when a surge of traffic overloads network resources

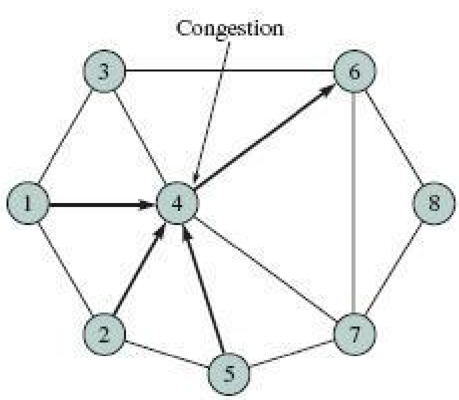


FIGURE 7.50 A congested switch

Approaches to Congestion Control:

- Preventive Approaches: Scheduling & Reservations
- Reactive Approaches: Detect & Throttle/Discard

Ideal effect of congestion control:
Resources used efficiently up to capacity available

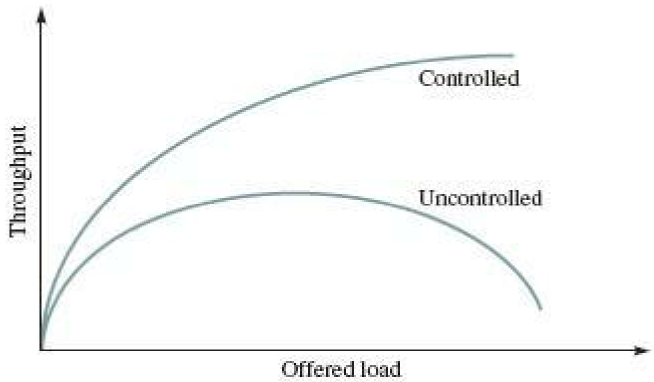


FIGURE 7.51 Throughput drops when congestion occurs

Open-loop control and closed-loop control are the two logical approaches of congestion control.

Open-Loop Control

- It prevents congestion from occurring.
- It does not depend on feedback information to react to congestion.
- Network performance is guaranteed to all traffic flows that have been admitted into the network
- It depends on three Key Mechanisms and they are:-
  - Admission Control
  - Policing
  - Traffic Shaping

Admission Control

- It is a network function that computes the resource (bandwidth and buffers) requirements of new flow and determines whether the resources along the path to be followed are available or not available.
- Before sending packet the source must obtain permission from admission control.
- Admission control decides whether to accept the flow or not.
- Flow is accepted, if the QoS of new flow does not violate QoS of existing flows
- QoS can be expressed in terms of maximum delay, loss probability, delay variance, or other performance measures.
- QoS requirements:
  - o Peak, Avg., Min Bit rate
  - o Maximum burst size
  - o Delay, Loss requirement
- Network computes resources needed
  - o "Effective" bandwidth
- If flow accepted, network allocates resources to ensure QoS delivered as long as source conforms to contract

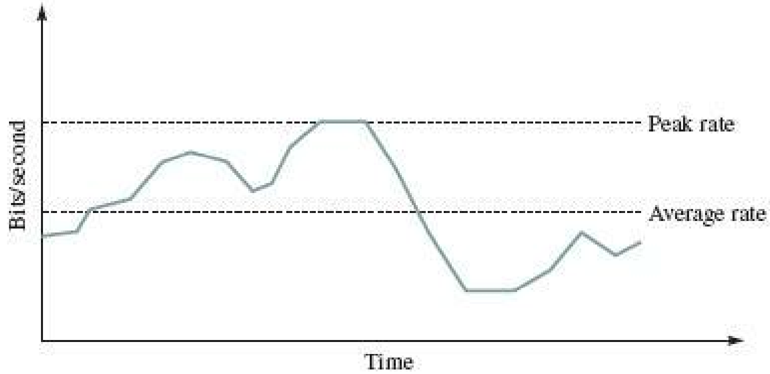


FIGURE 7.52 Example of a traffic flow

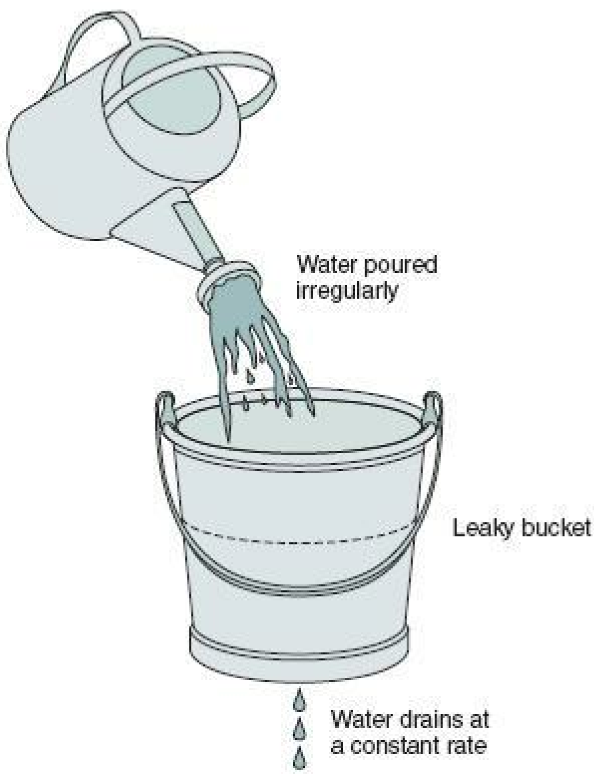
Policing

- Network monitors traffic flows continuously to ensure they meet their traffic contract.
- The process of monitoring and enforcing the traffic flow is called policing.
- When a packet violates the contract, network can discard or tag the packet giving it lower priority
- If congestion occurs, tagged packets are discarded first
- *Leaky Bucket Algorithm* is the most commonly used policing mechanism
  - o Bucket has specified leak rate for average contracted rate
  - o Bucket has specified depth to accommodate variations in arrival rate o

  Arriving packet is *conforming* if it does not result in overflow

Leaky Bucket algorithm can be used to police arrival rate of a packet stream

FIGURE 7.53 A leaky bucket



Let X = bucket content at last conforming packet arrival
Let ta be last conforming packet arrival time = depletion in bucket

Leaky Bucket Algorithm

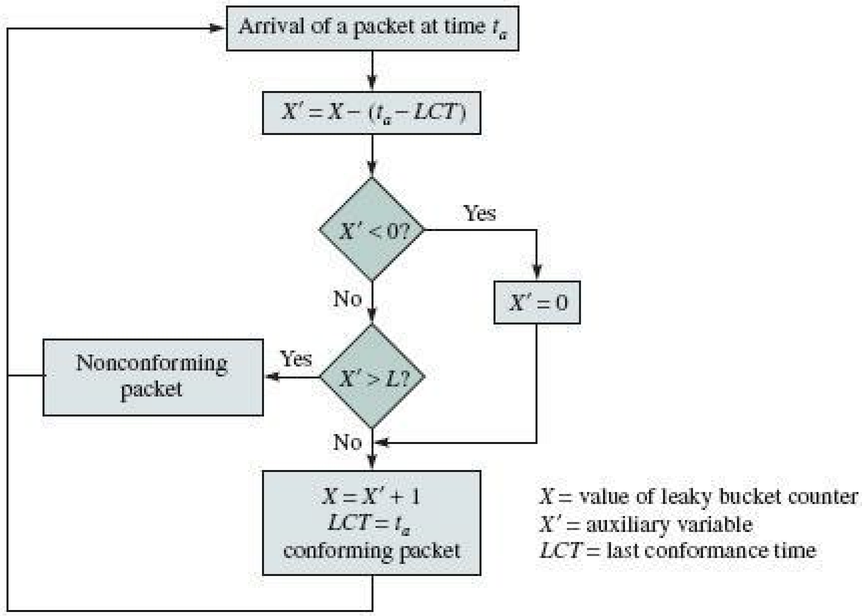


FIGURE 7.54 Leaky bucket algorithm used for policing

- The above figure shows the leaky bucket algorithm that can be used to police the traffic flow.
- At the arrival of the first packet, the content of the bucket is set to zero and the last conforming time (LCT) is set to the arrival time of the first packet.
- The depth of the bucket is L+I, where I depends on the traffic burstiness.
- At the arrival of the kth packet, the auxiliary variable X’ records the difference between the bucket content at the arrival of the last conforming packet and the interarrival time between the last conforming packet and the kth packet.
- If the auxiliary variable is greater than L, the packet is considered as nonconforming, otherwise the packet is conforming. The bucket content and the arrival time of the packet are then updated.

Leaky Bucket Example: - The operation of the leaky bucket algorithm is illustrated in the below figure.

- Here the value I is four packet times, and the value of L is 6 packet times.
- The arrival of the first packet increases the bucket content by four (packet times).
- At the second arrival the content has decreased to three, but four more are added to the bucket resulting in total of seven.
- The fifth packet is declared as nonconforming since it would increase the content to 11, which would exceed L+I (10).
- Packets 7, 8, 9 and 10 arrive back to back after the bucket becomes empty. Packets 7, 8 and 9 are conforming, and the last one is nonconforming.
- Non-conforming packets not allowed into bucket & hence not included in calculations.

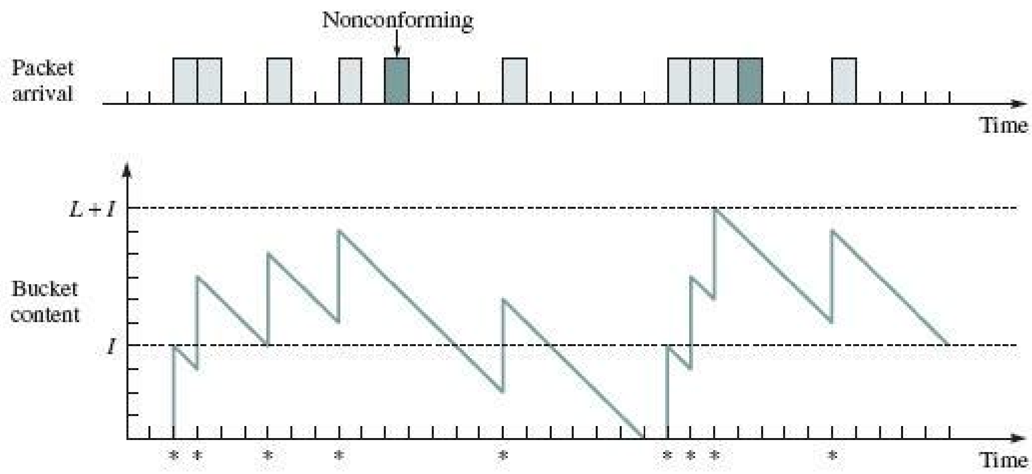


FIGURE 7.55 Behavior of leaky bucket

Dual Leaky Bucket

- Dual leaky bucket is use to police multiple traffic parameters like PCR, SCR, and MBS:
- Traffic is first checked for SCR at first leaky bucket.
- Nonconforming packets at first bucket are dropped or tagged.
- Conforming (untagged) packets from first bucket are then checked for PCR at second bucket.
- Nonconforming packets at second bucket are dropped or tagged.

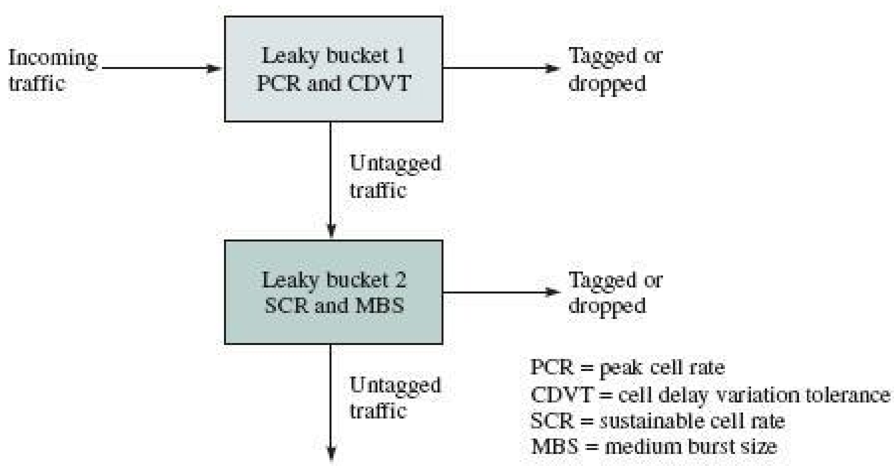
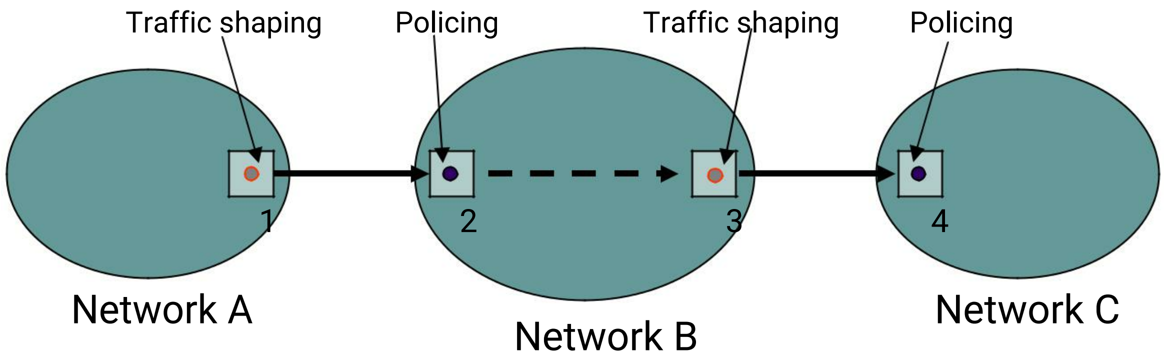


FIGURE 7.57 A dual leaky bucket configuration

Traffic Shaping



- Networks police the incoming traffic flow
- *Traffic shaping* is used to ensure that a packet stream conforms to specific parameters
- Networks can shape their traffic prior to passing it to another network
- In the above figure, the traffic shaping device is located at the node just before the traffic flow leaves a network, while the policing device is located at the node that receives the traffic flow from another network.

Leaky Bucket Traffic Shaper

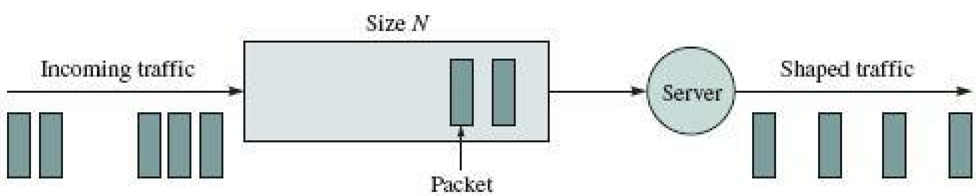


FIGURE 7.59 A leaky bucket traffic shaper

- Incoming packets are first stored in a buffer.
- Packets are served periodically so that the stream of packets at the output is smooth.
- Incoming packets are first stored in a buffer.
- Packets are served periodically so that the stream of packets at the output is smooth.
- A traffic shaping device needs to introduce certain delays for packets that arrive earlier than their scheduled departures and require a buffer to store these packets.
- Leaky bucket traffic shaper is too restrictive, since the output rate is constant when the buffer is not empty.

Token Bucket Traffic Shaper

- Token bucket is a simple extension of leaky bucket traffics shaper
- Tokens are generated periodically at constant rate and are stored in token bucket.
- Token rate regulates transfer of packets.
- If the token bucket is full, arriving tokens are discarded.
- A packet from the buffer can be taken out only if a token in the token bucket can be drawn

- If sufficient tokens available, packets enter network without delay
- If the token bucket is empty, arriving packets have to wait in the packet buffer.
- The size K determines how much burstiness allowed into the network

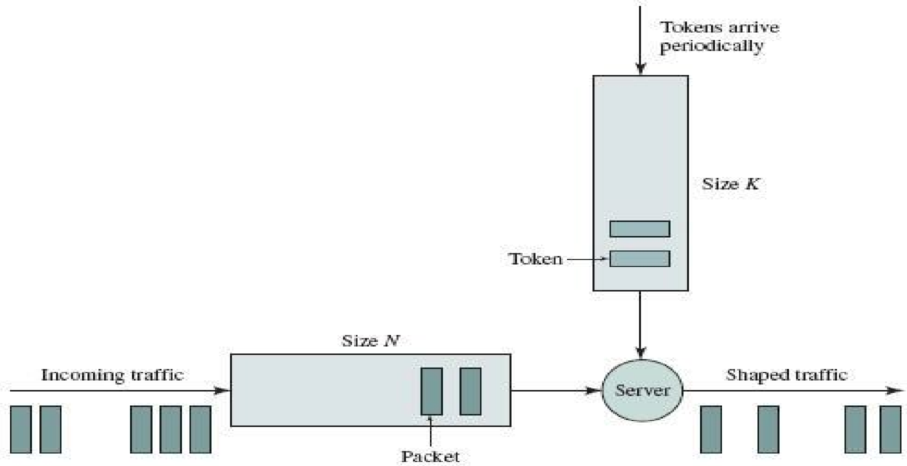


FIGURE 7.60 Token bucket traffic shaper

Closed-Loop Flow Control

- Congestion control
    - Feedback information is used to regulate the flow from sources into network based on buffer content, link utilization, etc.
    - Examples: TCP at transport layer; congestion control at ATM level
- Feedback information may be sent by End-to-end or Hop-by-hop.

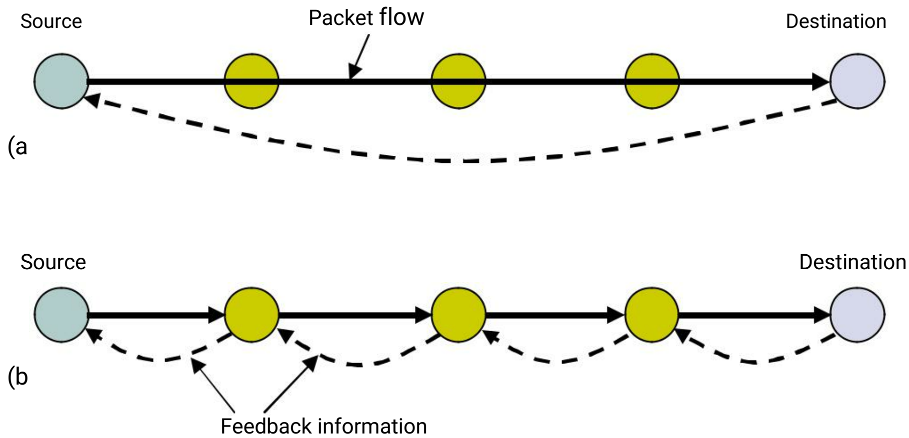
End-to-end closed loop control

- Feedback information about state of network is propagated back to source which regulate packet flow rate.
- Feedback information may be forwarded directly by a node that detects congestion, or it may be forwarded to destination first which then it relays information to source.
- The transmission of feedback information introduces propagation delay, so the information may not be accurate when the source receives the information.

Hop-by-hop control

- It reacts faster than end-to-end counterpart due to shorter propagation delay.
- State of the network is propagated to the upstream node as shown in below figure.
- When a node detects congestion it tells to its upstream neighbor to slow down its transmission rate.
- The Back Pressure created from one down stream node to another upstream node may continue all the way to the source.

End-to-End vs. Hop-by-Hop Congestion Control



Implicit vs. Explicit Feedback: - The information can be implicit or explicit. **Explicit Feedback**

- The node detecting congestion initiates an explicit message to notify the source about the congestion in the network.
- The explicit message can be sent as separate packet often called as choke packets or piggybacked on a data packet.
- The explicit message may be bit information or it may contain rich amount of information.

Implicit Feedback

- In implicit Feedback, no such explicit messages are sent between the nodes.
- Here congestion is controlled by using time out based on missing acknowledgements from destination to decide whether congestion has been encountered in the network.
- TCP congestion control is one example that regulates the transmission rate by using the implicit feedback information derived from missing acknowledgement.

Traffic Management at the flow aggregated level / Traffic Engineering

- Routing of aggregate traffic flows across the network for efficient utilization of resources and meeting of service levels
- Traffic Management at the Flow-Aggregate Level is called “Traffic Engineering”.
- Management exerted at flow aggregate level
- Distribution of flows in network to achieve efficient utilization of resources (bandwidth)
- Shortest path algorithm to route a given flow not enough
    - Does not take into account requirements of a flow, e.g. bandwidth requirement
    - Does not take account interplay between different flows
- Must take into account aggregate demand from all flows.
- Refer figure 7.63 and page number 560-561 for more information.

Why Internetworking?

- To build a "network of networks" or internet
    - operating over multiple, coexisting, different network technologies
    - providing ubiquitous(universal) connectivity through IP packet transfer
    - achieving huge economies of scale
- To provide *universal communication services*
    - independent of underlying network technologies
    - providing common interface to user applications
- To provide *distributed applications*
    - Rapid deployment of new applications
        - Email, WWW, Peer-to-peer
    - Application independent of network technologies
        - New networks can be introduced

TCP/IP Architecture

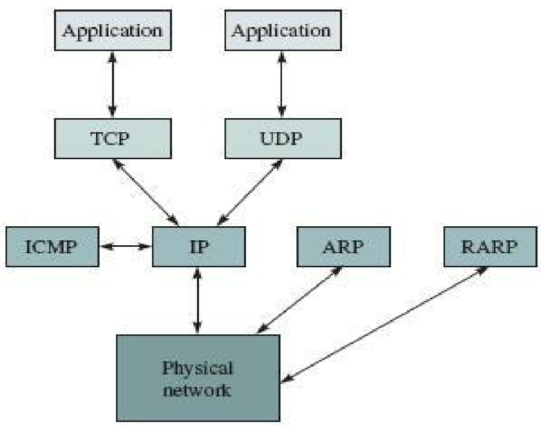


FIGURE 8.1 TCP/IP protocol suite

- The TCP/IP protocol suite usually refers not only to the two most well-known protocols called TCP and IP but also to other related protocols such as UDP, ICMP, HTTP, TELNET and FTP.
- Basic structure of TCP/IP protocol suite is shown in above figure.
- Protocol data unit (PDU) exchanged between peer TCP protocols is called segments.
- Protocol data unit (PDU) exchanged between peer UDP protocols is called datagrams.
- Protocol data unit (PDU) exchanged between peer IP protocols is called packets.

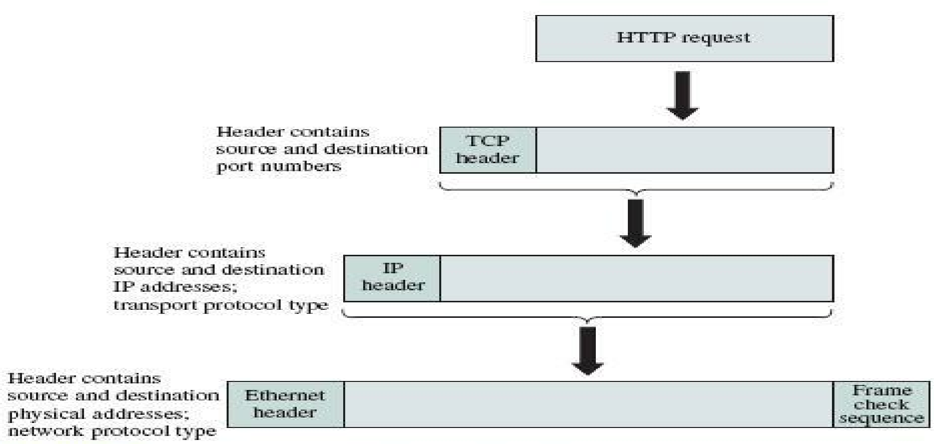


FIGURE 8.2 Encapsulation of PDUs in TCP/IP and addressing information in the headers

- In the above figure an HTTP GET command is passed to the TCP layer, which encapsulates the message into a TCP segment.
- The segment header contains an ephemeral port number for the client process and well known port 80 for HTTP server process.
- The TCP segment is passed to IP layer where it is encapsulated in an IP packet.
- The IP packet contains source and destination network address.
- IP packet is then passed through network interface and encapsulated into PDU of underlying network.
- In the network interface, the IP packet is encapsulated into an Ethernet frame, which contains physical addresses that identify the physical endpoints for the Ethernet sender and receiver.

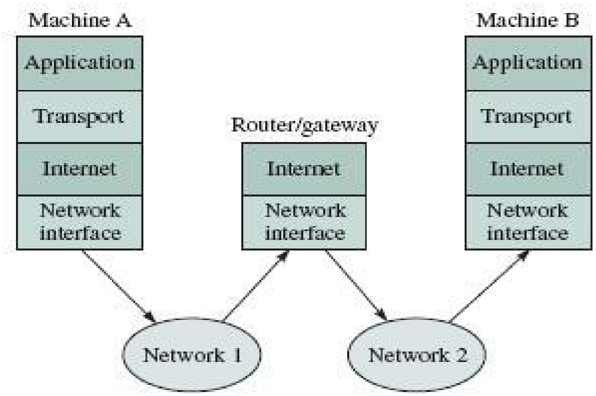


FIGURE 8.3 The Internet and network interface layers

- IP packets transfer information across Internet
- **Host A IP → router→ router…→ router→ Host B IP**
- IP layer in each router determines next hop (router)
- Network interfaces transfer IP packets across networks

- **Internet Names**
  - Each host has a unique name
    - Independent of physical location
    - Domain Name will facilitates memorization by humans
  - Host Name
    - Name given to host computer
  - User Name
    - Name assigned to user

Internet Addresses

- Each host has globally unique logical 32 bit IP address
- Separate address for each physical connection to a network
- Routing decision is done based on destination IP address
- IP address has two parts:
  - netid and hostid
  - netid unique
  - netid facilitates routing
- Dotted Decimal Notation is used for representation: Ex: - int1.int2.int3.int4
  128.100.10.13
  DNS(Domain Name Service) resolves IP name to IP address

Physical Addresses

- LANs (and other networks) assign physical addresses to the physical attachment to the network
- The network uses its own address to transfer packets or frames to the appropriate destination
- IP address needs to be resolved to physical address at each IP network interface
- Example: Ethernet uses 48-bit addresses
    - o Each Ethernet network interface card (NIC) has globally unique Medium Access Control (MAC) or physical address
    - o First 24 bits identify NIC manufacturer; second 24 bits are serial number o 00:90:27:96:68:07 12 hex numbers

Internet Protocol

- It provides best effort, connectionless packet delivery, packets may be lost, out of order, or even duplicated, so it is the responsibility of higher layer protocols to deal with these, if necessary.
- The header is of fixed-length component of 20 bytes plus variable-length consisting of options that can be up to 40 bytes.

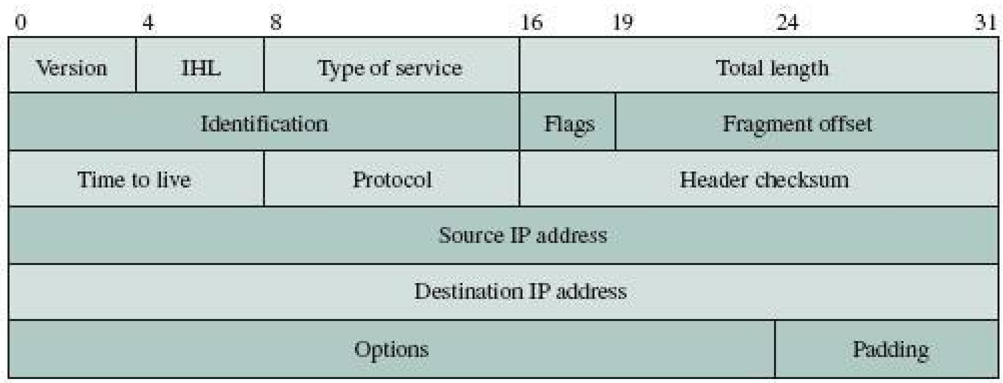


FIGURE 8.4 IP version 4 header

Version: This field identifies the current IP version and it is 4.
Internet header length (IHL): It specifies the length of the header in 32-bit words. If no options are used, IHL will have value of 5.
Type of service (TOS): This field specifies the priority of packet based on delay, throughput, reliability and cost. Three bits are used to assign priority levels and four bits are used for specific requirement (i.e. delay, throughput, reliability and cost).
Total length: The total length specifies the number of bytes of the IP packet including header and data, maximum length is 65535 bytes.
Identification, Flags, and Fragment Offset: These fields are used for fragmentation and reassembly.
Time to live (TTL): It specifies the number of hops; the packet is allowed to traverse in the network. Each router along the path to the destination decrements this value by one. If the value reaches zero before the packet reaches the destination, the router discards the packet and sends an error message back to the source.
Protocol: specifies upper-layer protocol that is to receive IP data at the destination.
Examples include TCP (protocol = 6), UDP (protocol = 17), and ICMP (protocol = 1).

Header checksum: verifies the integrity of the IP header of the IP packet.

- IP header uses check bits to detect errors in the ***header***
- A checksum is calculated for header contents
- Checksum recalculated at every router, so algorithm selected for ease of implementation in software

Source IP address and **destination IP address:** contain the addresses of the source and destination hosts.
Options: Variable length field allows packet to request special features such as security level, route to be taken by the packet, and timestamp at each router. Detailed descriptions of these options can be found in [RFC 791].
Padding: This field is used to make the header a multiple of 32-bit words. **IP Header**

Processing

1. Compute header checksum for correctness and check that fields in header (e.g. version and total length) contain valid values
2. Consult routing table to determine next hop
3. Change fields that require updating (TTL, header checksum)

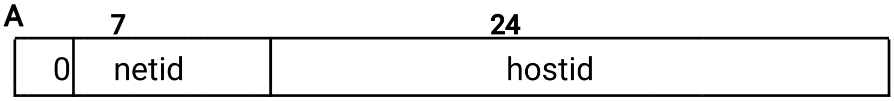
IP Addressing

- RFC 1166
- Each host on Internet has unique 32 bit IP address
- Each address has two parts: Netid and Hostid
- Netid is unique & administered by
    - o American Registry for Internet Numbers (ARIN) o Reseaux IP Europeens (RIPE)
    - o Asia Pacific Network Information Centre (APNIC)
- The Net ID identifies the network the host is connected to.
- The host ID identifies each individual system connected to network.
- Dotted Decimal Notation is used for representation:
- The IP address of 10000000 10000111 01000100 00000101 is 128.135.68.5 in dotted-decimal notation

Classful IP Addresses

- The IP address structure is divided into five address classes: Class A, Class B, Class C, Class D and Class E
- The class is identified by the Most Significant Bit (MSB) of the address as shown below.
- Class A has 7 bits for network IDs and 24 bits for host IDs, allowing up to 126 networks and about 16 million hosts per network.
- Class B has 14 bits for network IDs and 16 bits for host IDs, allowing about 16,000 networks and about 64,000 hosts per network.
- Class C has 21 bits for network IDs and 8 bits for host IDs, allowing about 2 million networks and 254 hosts per network.
- Class D addresses is used for multicast services that allow host to send information to a group of hosts simultaneously.
- Class E addresses are reserved for experiments.

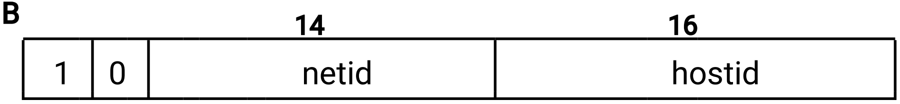
Class A



- 126 networks with up to 16 million hosts

1.0.0.0 to 127.255.255.255

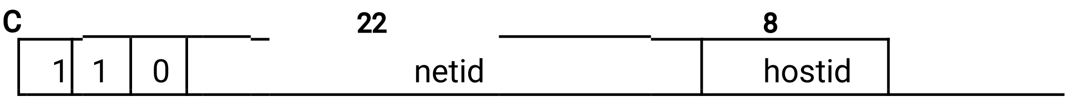
Class B



- 16,382 networks with up to 64,000 hosts

128.0.0.0 to 191.255.255.255

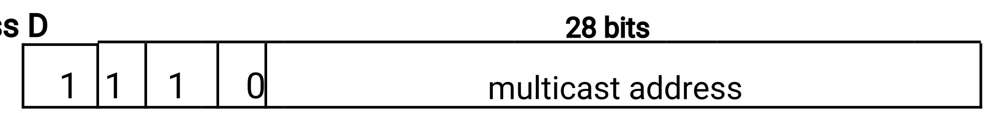
Class C



- 2 million networks with up to 254 hosts

192.0.0.0 to 223.255.255.255

Class D



224.0.0.0 to 239.255.255.255

Class E

240.0.0.0 to 254.255.255.255

Subnet Addressing

- Subnet addressing was introduced in the mid 1980s when most large organizations are moving their computing platforms from mainframes to networks of workstations.
- Subnetting adds another level of hierarchical level called **"Subnet".**
- Inside the organization the network administrator can choose any combination of lengths for subnet and host ID fields.
- Example: - consider an organization that has been assigned a class B IP address with a network ID of 150.100. Suppose the organization has many LANS, each consisting of not more than 100 hosts. Then seven bits are sufficient to uniquely identify each host in a subnetwork. The other nine bits can be used to identify the subnetworks within organization
- To find the subnet number, the router needs to store an additional quantity called subnet mask, which consists of binary 1s for every bit position of the address except the host ID field where binary 0s are used.
- For the IP address 150.100.12.176, the subnet mask is
  11111111 11111111 11111111 10000000, which corresponds to 255.255.255.128.

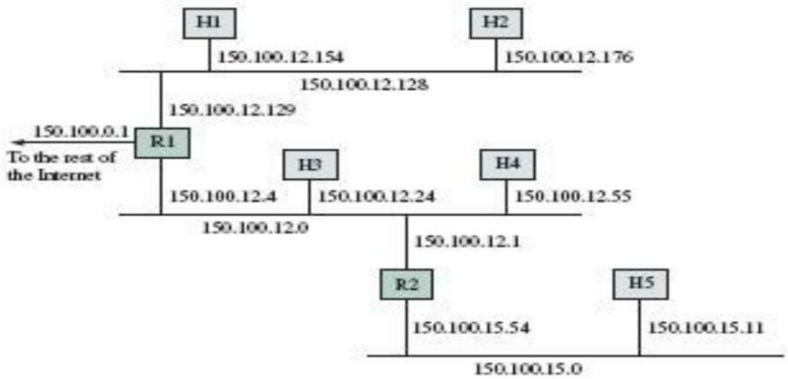
- The router can determine the subnet number by performing a binary AND between subnet mask and the IP address.

The IP address is 10010110 01100100 00001100 10110000 i.e. 150.100.12.176 AND with subnet mask 11111111 11111111 11111111 10000000 i.e. 255.255.255.128 to get subnet number 10010110 01100100 00001100 10000000 i.e.150.100.12.128 and which is also called as **First Address** and is used to identify the subnetwork inside the organization.

- The IP address 150.100.12.255 is used to broadcast packets inside the subnetwork. Thus the host connected to subnetwork must have IP address in the range 150.100.12.129 to 150.100.12.254.

IP Routing

- IP layer in end-system hosts and in the router work together to route packets from source to destination.
- IP layer in each host and router maintains a routing table, which is used to route the packets based on IP address.
- If a destination host is directly connected to the originating host by a link or by a LAN, then the packet is sent directly to destination host using appropriate network interface, otherwise, the routing table specifies that the packet is to send to default gateway.
- When a router receives an IP packet from one of the network interfaces, then router examines its routing table to see whether the packet is destined to itself or not, if so, delivers to router's own address, then the router determines the next–hop router and associated network interface, and then forwards the packet.
- Each row in routing table must provide information like: destination IP address, IP address of next -hop router, several flag fields, outgoing network interface, and other information such as subnet mask, physical address.
- H flag indicates whether the route in the given row is to a host (H=1) or to a network.
- G flag indicates whether the route in the given row is to a router (gateway, G=1) or to a directly connected destination (G=0).
- Each time a packet is to be routed, the routing table is searched in the following order.
- First, the destination column is searched to see whether table contains an entry for complete destination IP address.
- If so, then IP packet is forwarded according to next-hop entry and G flag.
- Second, if the table does not contain complete destination IP address, then routing table is searched for the destination network ID.
- If an entry found, the IP packet is forwarded according to next-hop entry and G flag.
- Third, if table does not contain destination network ID, the table is searched for default router entry, and if one is available, the packet is forwarded there.
- Finally if none of the above searches are successful, the packet is declared undeliverable and an ICMP "host unreachable error" packet is sent back to originating host.



Example—Routing with Subnetworks

Suppose that host H5 wishes to send an IP packet to host H2 in Figure 8.7. H2 has IP address 150.100.12.176 (1001 0110. 0110 0100. 0000 1100. 1011 0110). Let us trace the operations in carrying out this task.

The routing table in H5 may look something like this:

Destination	Next-Hop	Flags	Network Interface
127.0.0.1	127.0.0.1	H	lo0
default	150.100.15.54	G	emd0
150.100.15.0	150.100.15.11		emd0

The first entry is the loopback interface, the H indicates a host address, and lo0 by convention is always the loopback interface. The second entry is the default entry, with next-hop router R2 (150.100.15.54), which is a router, so G = 1, and with Ethernet interface emd0. The third entry does not have H set, so it is a network address; G is also not set, so a direct route is indicated and the next-hop entry is the IP address of the outgoing network interface.

CIDR

- **CIDR** stands for Classless Inter-Domain Routing.
- CIDR was developed in the 1990s as a standard scheme for routing network traffic across the Internet.
- Before CIDR technology was developed, Internet routers managed network traffic based on the class of IP addresses. In this system, the value of an IP address determines its subnetwork for the purposes of routing.
- CIDR is an alternative to traditional IP subnetting that organizes IP addresses into subnetworks independent of the value of the addresses themselves. CIDR is also known as supernetting as it effectively allows multiple subnets to be grouped together for network routing.

CIDR Notation: - CIDR specifies an IP address range using a combination of an IP address and its associated network mask. CIDR notation uses the following format -

xxx.xxx.xxx.xxx/n

where n is the number of (leftmost) '1' bits in the mask. For example, 192.168.12.0/23 applies the network mask 255.255.254.0 to the 192.168 network, starting at 192.168.12.0. This notation represents the address range 192.168.12.0 - 192.168.13.255. Compared to traditional class-based networking, 192.168.12.0/23 represents an *aggregation* of the two Class C subnets 192.168.12.0 and 192.168.13.0 each having a subnet mask of 255.255.255.0. In other words,

192.168.12.0/23 = 192.168.12.0/24 + 192.168.13.0/24

Additionally, CIDR supports Internet address allocation and message routing independent of the traditional class of a given IP address range. For example,

10.4.12.0/22 represents the address range 10.4.12.0 - 10.4.15.255 (network mask 255.255.252.0). This allocates the equivalent of four Class C networks within the much larger Class A space.

You will sometimes see CIDR notation used even for non- CIDR networks. In non -CIDR IP subnetting, however, the value of *n* is restricted to either 8 (Class A), 16 (Class B) or 24 (Class C). Examples:

- 10.0.0.0/8
- 172.16.0.0/16
- 192.168.3.0/24

CIDR aggregation requires the network segments involved to be contiguous (numerically adjacent) in the address space. CIDR cannot, for example, aggregate 192.168.12.0 and 192.168.15.0 into a single route unless the intermediate .13 and .14 address ranges are included (*i.e.*, the 192.168.12/22 network).

ARP (Address Resolution Protocol)

- ❖ The address resolution protocol (**ARP**) is a protocol used by the Internet Protocol (IP) specifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol.
- ❖ The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet.
- ❖ It is also used for IP over other LAN technologies, such as Token Ring, FDDI, or IEEE 802.11, and for IP over ATM.
- ❖ ARP is a Link Layer protocol because it only operates on the local area network or point-to-point link that a host is connected to.
- ❖ The hardware address is also known as the Medium Access Control (MAC) address, in reference to the standards which define Ethernet.
- ❖ The Ethernet address is a link layer address and is dependent on the interface card which is used.
- ❖ IP operates at the network layer and is not concerned with the link addresses of individual nodes which are to be used. The ARP is therefore used to translate IP addresses into MAC address.

- In the below figure suppose host H1 wants to send an IP packet to H3, but does not know the MAC address of H3. H1 first broadcast an ARP request packet asking the destination host, which is identified by H3's IP address, to reply. All hosts in the network receive the packet, but only the intended host, which is H3, responds to H1.
- The ARP response packet contains H3's MAC address and IP addresses.
- H1 caches H3's MAC address in its ARP table so that H1 can simply look up H3's MAC address in the table for future use.

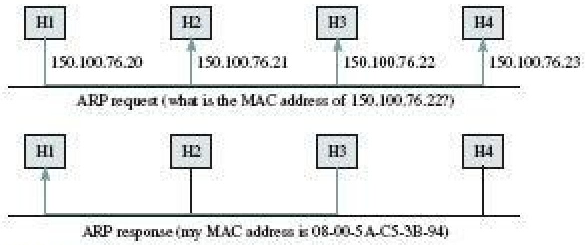


FIGURE 8.8 Address Resolution Protocol

- ❖ The ARP client and server processes operate on all computers using IP over Ethernet. The processes are normally implemented as part of the software driver that drives the network interface card.

RARP (Reverse Address Resolution Protocol)

- ❖ **RARP** is a link layer networking protocol, used to resolve an IP address from a given hardware address (such as an Ethernet address).
- ❖ RARP requires one or more server hosts to maintain a database of mappings from Link Layer address to protocol address.
- ❖ To obtain its IP address, the host broadcasts an RARP request packet containing its MAC address on the network.
- ❖ All hosts in the network receive the packet, but only the server replies to the host by sending an RARP response containing the host's MAC and IP address.

IP fragmentation and Reassembly

- ❖ The Internet Protocol allows **IP fragmentation** so that datagrams can be fragmented into pieces small enough to pass over a link with a smaller MTU than the original datagram size.
- ❖ The Identification field, and Fragment offset field along with Don't Fragment and More Fragment Flags are used for Fragmentation and Reassembly of IP datagrams.
- ❖ In a case where a router in the network receives a PDU larger than the next hop's MTU, it has two options. Drop the PDU and send an ICMP message which says "Packet too Big", or to Fragment the IP packet and send over the link with a smaller MTU.
- ❖ If a receiving host receives an IP packet which is fragmented, it has to reassemble the IP packet and hand it over to the higher layer.
- ❖ Reassembly is intended to happen in the receiving host but in practice it may be done by an intermediate router, for example network address translation requires re-calculating checksums across entire packets, and so routers supporting this will often recombine packets as part of the process.

- The details of the fragmentation mechanism, as well as the overall architectural approach to fragmentation, are different in IPv4 and IPv6.
- In IPv4, routers do the fragmentation, whereas in IPv6, routers do not fragment, but drop the packets that are larger than the MTU size. Though the header formats are different for IPv4 and IPv6, similar fields are used for fragmentation, so the algorithm can be reused for fragmentation and reassembly.
- 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.
- Thus senders typically use two approaches to decide the size of IP datagrams to send over the network.
- 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.
- The second is to run the "Path MTU discovery" algorithm, to determine the path MTU between two IP hosts, so that IP fragmentation can be avoided.
- The flag field has three bits, one unused bit, one “don’t fragment”(DF) bit, and one “more fragment”(MF) bit.
- If DF bit is set to 1, it forces the router not to fragment the packet. If the packet length is greater than MTU, the router will discard the packet and send an error message to the source host.
- The MF bit tells the destination host whether or not more fragments follow. If there are more, the MF bit is set to 1; otherwise, it is set to 0.
- Fragment offset field identifies the location of a fragment in a packet.

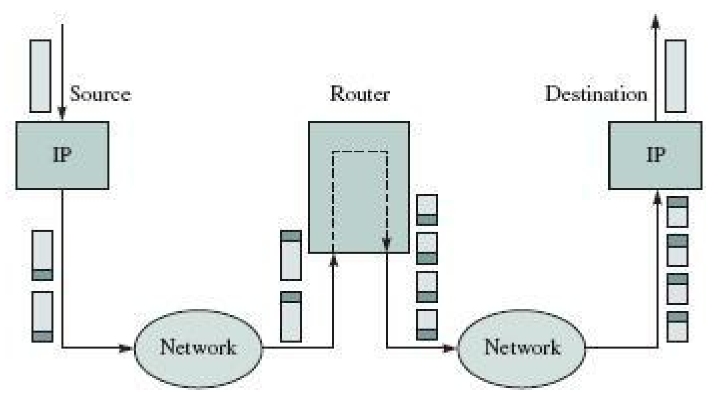


Figure: Packet fragmentation

Example—Fragmenting a Packet

Suppose a packet arrives at a router and is to be forwarded to an X.25 network having an MTU of 576 bytes. The packet has an IP header of 20 bytes and a data part of 1484 bytes. Perform fragmentation and include the pertinent values of the IP header of the original packet and of each fragment.

The maximum possible data length per fragment $= 576 - 20 = 556$ bytes. However, 556 is not a multiple of 8. Thus we need to set the maximum data length to 552 bytes. We can break 1484 into $552 + 552 + 380$ (other combinations are also possible).

Table 8.1 shows the pertinent values for the IP header where x denotes a unique identification value. Other values, except the header checksum, are the same as in the original packet.

	Total length	ID	MF	Fragment offset
Original packet	1504	x	0	0
Fragment 1	572	x	1	0
Fragment 2	572	x	1	69
Fragment 3	400	x	0	138

TABLE 8.1 Values of the IP header in a fragmented packet

Deficiencies of IP

- Lack of error control, flow control and congestion control
- Lack of assistance mechanisms

What happens if something goes wrong?

- If a router must discard a datagram because it can not find a router to the final destination
- The time-to-leave field has a zero value
- If the final destination host must discard all fragments of a datagram because it has not received all fragments within a pre-determined time limit

IP has no built in mechanisms to notify the original hosts, in erroneous situations **IP also lacks a** mechanism for host and management queries

- A host wants to know whether a router or another host is active
- Sometimes network manager needs information from another host or router

Internet Control Message Protocol [ICMP] is companion to IP, designed to compensate these deficiencies

- ICMP is a network layer protocol
- Its messages are encapsulated inside IP datagrams before going to lower layer
- Ping and Traceroute uses ICMP messages,

ICMP Messages

1) Error Reporting Messages
2) Query Messages

1) Error Reporting
    - Destination unreachable
    - Source quench

- Time exceeded
- Parameters problems
- Redirection

ICMP messages [Error reporting]

1. Destination unreachable
When the subnet or a router can not locate the destination Or
When a packet with DF bit, can not be delivered because a 'small-packet' network stands in the way

2. Time exceeded
When a packet is dropped because its counter has reached zero. This event is a symptom that packets are looping enormous congestion or the time values are being set too low.

3. Parameter problem
Indicates that an illegal value has been detected in the header field
Indicates a bug in the sending host's IP software Or Possibly in the software of a router transited.

4. Source quench
To throttle hosts that send too many packets, When a host receives this message, it slows down sending packets

5. Redirect
Is used when a router notices that a packet seems to be routed wrong It is used by the router to tell the sending host about the probable error.

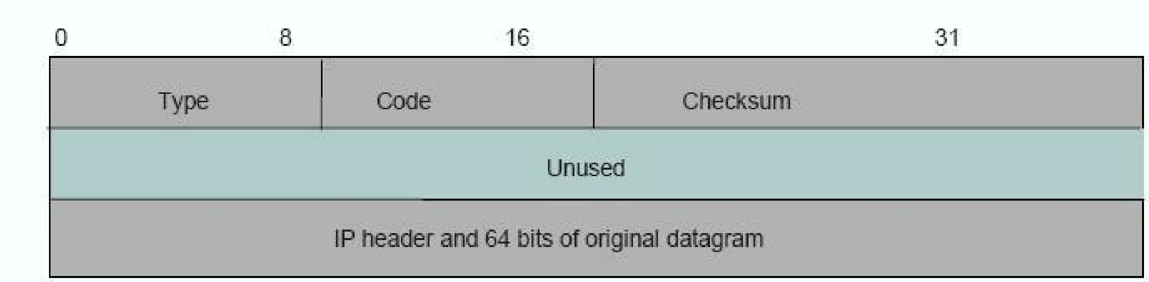
2) Query Messages
   - Echo request and reply
   - Time-stamp request and reply
   - Address mask request and reply

1. ECHO & ECHO Reply
To see if a given destination is reachable and alive, upon receipt of ECHO message, the destination is expected to send an ECHO REPLY message back.

2. Time stamp & Time stamp reply
Similar to ECHO queries, except that the arrival time of the message and departure time of the reply are recorded in the reply.
This facility is used to measure network performance.

Message type	Description
Destination unreachable	Packet could not be delivered
Time exceeded	Time to live field hit 0
Parameter problem	Invalid header field
Source quench	Choke packet
Redirect	Teach a router about geography
Echo request	Ask a machine if it is alive
Echo reply	Yes, I am alive
Timestamp request	Same as Echo request, but with timestamp
Timestamp reply	Same as Echo reply, but with timestamp

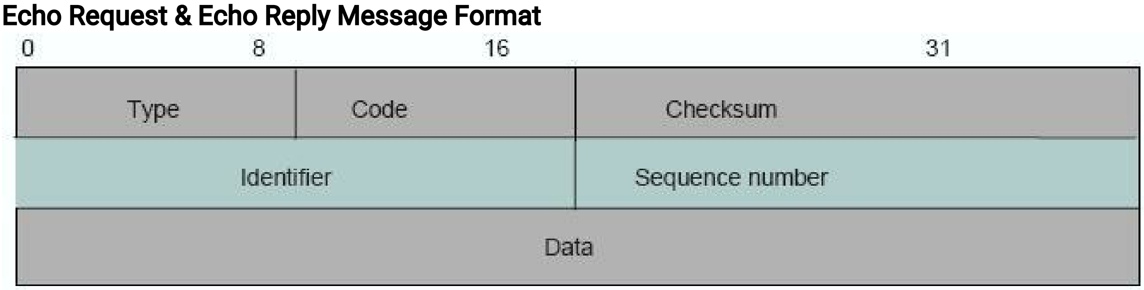
ICMP Basic Error Message Format



Type of message: some examples

0 Network Unreachable; 3 Port Unreachable
1 Host Unreachable 4 Fragmentation needed
2 Protocol Unreachable 5 Source route failed
11 Time-exceeded, code=0
if TTL exceeded
• Code: purpose of message
• IP header & 64 bits of original datagram
– To match ICMP message with original data in IP packet

Echo Request & Echo Reply Message Format



Echo request: type=8; Echo reply: type=0
– Destination replies with echo reply by copying data in request onto reply message
• Sequence number to match reply to request
• ID to distinguish between different sessions using echo services
• Used in PING

ICMP functions

1) **Announce network errors:** Such as host or Entire portion of the network being unreachable, due to some type of failure. A TCP or UDP packet directed at a port number with no receiver attached is also reported via ICMP.
2) **Announce network congestion:** When a router begins buffering too many packets, due to an inability to transmit them as fast as they are being received, it will generate ICMP Source Quench messages. Directed at the sender, these messages should cause the rate of packet transmission to be slowed.
3) **Assist Troubleshooting:** ICMP supports an Echo function, which just sends a packet on a round--trip between two hosts. Ping, a common network management tool, is based on this feature. Ping will transmit a series of packets, measuring average round—trip times and computing loss percentages.
4) **Announce Timeouts:** If an IP packet's TTL field drops to zero, the router discarding the packet will often generate an ICMP packet announcing this fact.

UNIT 2 Question Bank

1. Consider a packet-by-packet fair queuing system with three logical buffers and with a service rate of one unit / second. Show the sequence of transmissions for this system for the following packet arrival pattern:
   (i) Buffer 1: arrival at time t=0, length =2 ;arrival at t=4, length= 1
   (ii) Buffer 2: arrival at time t=1, length =3 ;arrival at t=2, length= 1 (iii) Buffer 3: arrival at time t=3, length =5 ;
   (10M)
2. With a neat diagram explain the internal network operation of the network.
3. What is congestion? Discuss the general principles of congestion control?
   (10M)
4. Explain the random early detection. ***(6M)***
5. Explain leaky bucket algorithm ***(8M)***
6. Explain the Token bucket policy for the traffic shaping. ***(July 07, 5M)***
7. A computer on a 6Mbps network is regulated by a token bucket. The token bucket is filled at a rate of 1Mbps. It is initially filled to capacity with 8 megabits. How long the computer transmit at the full 6Mbps ***(5M)***
8. Explain FIFO and Priority Queues for the traffic management at the packet level
9. Explain Fair Queuing for the traffic management at the packet level
10. Explain Weighted-Fair Queuing for the traffic management at the packet level
11. Write short notes on admission control and policing.
12. Write short notes on traffic shaping
13. Distinguish between end-to-end and hop-by-hop closed loop control
14. Distinguish between implicit and explicit feedback
15. Give the differences between leaky bucket and token bucket algorithm
16. Write a note on Traffic management at the flow-aggregate level.
17. Explain with diagram the TCP/IP architecture
18. Explain IPV4 header.***(6M)***
19. Explain the IP addressing scheme. ***(6M)***
20. Distinguish between address resolution protocol and reverse address resolution protocol. ***(5M)***

21. Illustrate with a diagram the five address formats used in internet***(6M)***
22. Briefly explain Address Resolution Protocol. ***(5M)***
23. What is ICMP? Explain the functions of ICMP. ***(5M)***
24. A university has 150 LANs with 100 hosts in each LAN.
    a. Suppose the university has one Class B address. Design an appropriate subnet addressing scheme.
    b. Design an appropriate CIDR addressing scheme.

    (6M)
25. A network on the internet has a subnet mask 255.255.240.0. What is the maximum no. of hosts it can handle? ***(6M)***
26. A large number of consecutive IP address are available at 198.16.0.0. Suppose that four organizations A, B, C and D request for 4000, 2000, 4000 and 8000 addresses respectively. For each of these, give the first IP address assigned, the last IP address assigned, and the mask in dotted decimal notation.

    (4M)
27. A large number of consecutive IP address are available starting at 200.40.160.0. Suppose that 3 organizations A, B, and C request for 4000, 2000 and 1000 addresses respectively. For each of these, give the first IP address assigned, the last IP address assigned, and the mask in dotted decimal notation.***(6M)***
28. Write a note on CIDR
29. What is fragmentation? How packets are fragmented and reassembled by the IP?
30. Identify the address class of the following IP addresses: 200.58.20.165; 128.167.23.20; 16.196.128.50; 50.156.10.10; 250.10.24.96.
31. Convert the IP addresses in Problem above to their binary representation.
32. Identify the range of IPv4 addresses spanned by Class A, Class B, and Class C.
33. What are all the possible subnet masks for the Class C address space? List all the subnet masks in dotted-decimal notation, and determine the number of hosts per subnet supported for each subnet mask.
34. A host in an organization has an IP address 150.32.64.34 and a subnet mask 255.255.240.0. What is the address of this subnet? What is the range of IP addresses that a host can have on this subnet?
35. A small organization has a Class C address for seven networks each with 24 hosts. What is an appropriate subnet mask?
36. A packet with IP address 150.100.12.55 arrives at router R1 in Figure 8.8. Explain how the packet is delivered to the appropriate host.
37. Perform CIDR aggregation on the following /24 IP addresses: 128.56.24.0/24; 128.56.25.0./24; 128.56.26.0/24; 128.56.27.0/24.
38. Perform CIDR aggregation on the following /24 IP addresses: 200.96.86.0/24; 200.96.87.0/24; 200.96.88.0/24; 200.96.89.0/24.
39. Suppose a router receives an IP packet containing 600 data bytes and has to forward the packet to a network with maximum transmission unit of 200 bytes. Assume that the IP header is 20 bytes long. Show the fragments that the router creates and specify the relevant values in each fragment header (i.e., total length, fragment offset, and more bit).

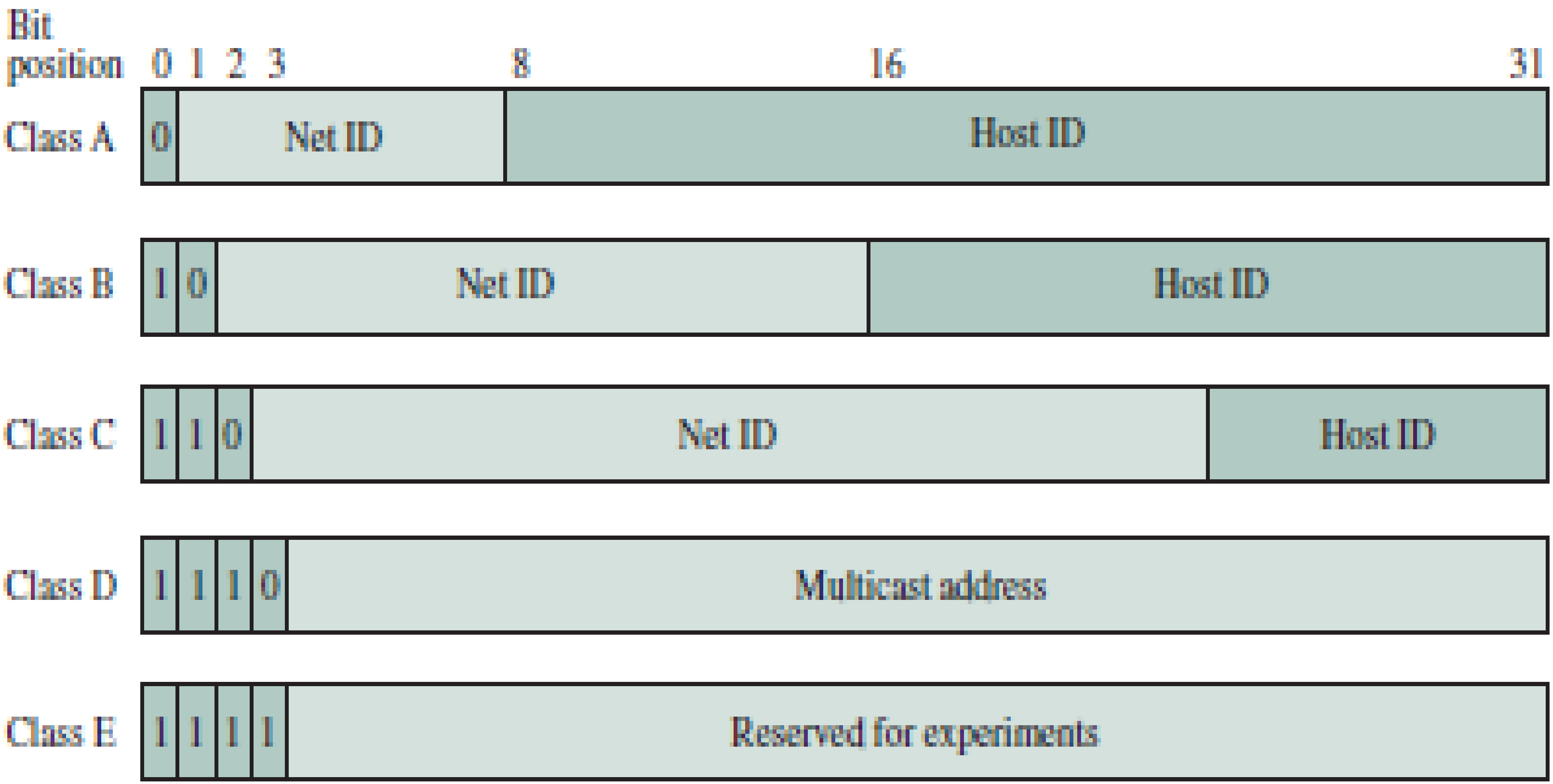
Internet Protocol

IP Addressing

- numeric identifier assigned to each machine on an IP network
- consists of network ID(NID) and host ID(HID) .
- Network ID identifies the network the host is connected to
- Host ID identifies the network connection to the host – assigned by the network administrator at the local site

IP address Structure

- Composed of 5 address classes:
- Class A, B, C, D and E
- Class A – 7 bits for NID, 24 bits for host ID
- Class B – 14 bits for NID, 16 bits for host ID
- Class C – 21 bits for NID, 8 bits for host ID
- Class D – used for multicast services
- Class E – reserved for experiments



IP addresses are usually written in dotted-decimal notation. The address is broken into four bytes.

For example, an IP address of

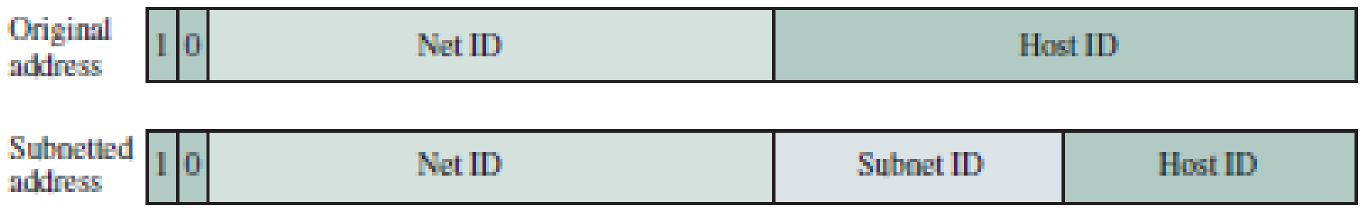
10000000 10000111 01000100 00000101

is written as

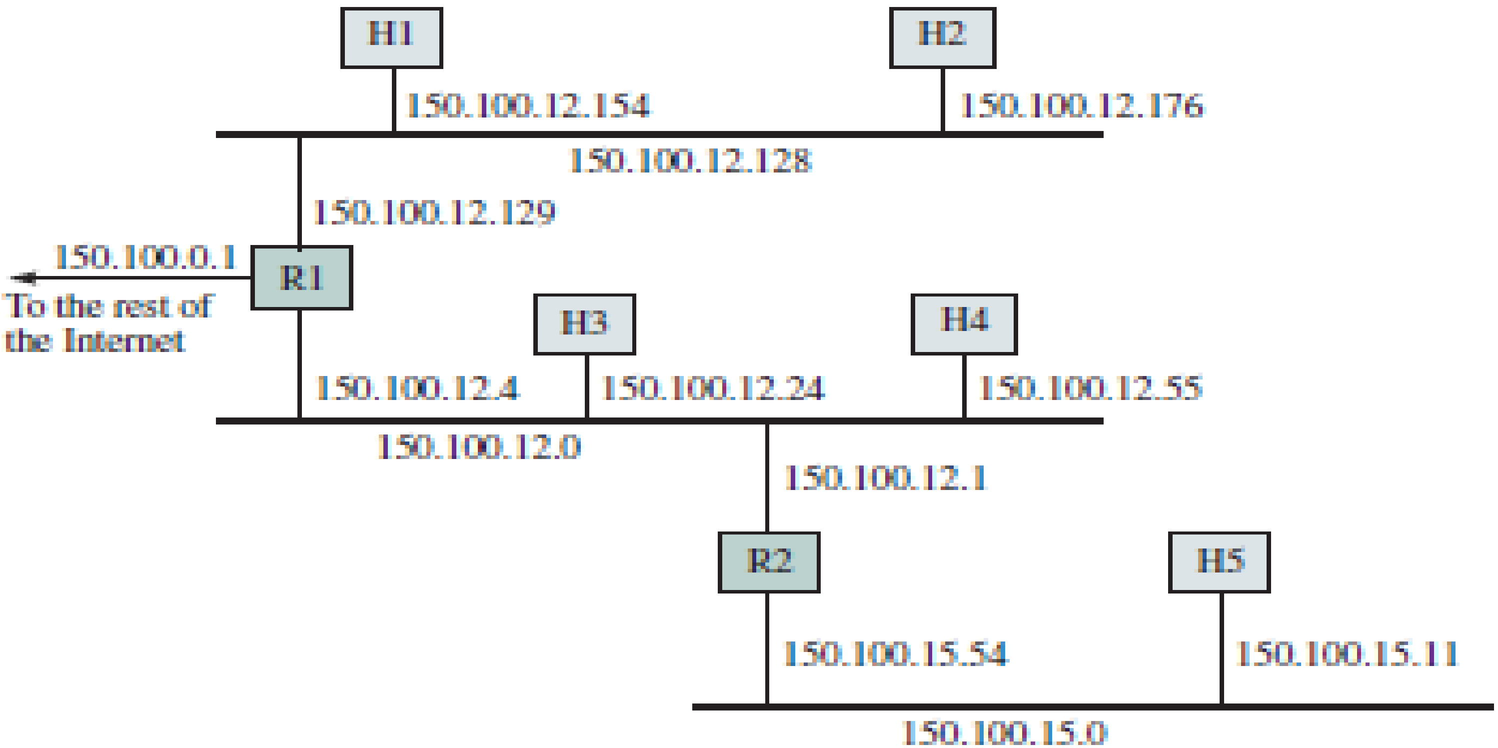
128.135.68.5

Subnet Addressing

- To allow a single network address to span multiple physical networks is called *subnet addressing*
- For the subnet address scheme to work, every machine on the network must know which part of the host address will be used as the subnet address - subnet mask
- The 1's in the subnet mask represent the positions that refer to the network or subnet addresses
- The 0's represent the positions that refer to the host part of the address.



- If a packet with a destination IP address of 150.100.12.176 arrives at site from the outside network, which subnet should a router forward this packet to?
- Assume subnet mask is 255.255.255.128



- The router can determine the subnet number by performing a binary AND between the subnet mask and the IP address.

- IP address: 10010110 01100100 00001100 10110000(150.100.12.176)
- Subnet mask: 11111111 11111111 11111111 10000000(255.255.255.128)
- Subnet number: 10010110 01100100 00001100 10000000(150.100.12.128)
- This number(150.100.12.128) is used to forward the packet to the correct subnet work inside the organization.

IP Routing

- IP layer in the end-system hosts and in the routers work together to route packets from sources to destinations.
- IP layer in each host and router maintains a routing table – to determine how to handle each packet
- Job of the host
- Job of the router

CLASSLESS INTER DOMAIN ROUTING(CIDR)

- To prevent the wastage of IP addresses by allocating a subset of class A, B or C
- An arbitrary prefix length is used to indicate the network number known as CIDR
- Use of network masks
- CIDR routing table is composed of 32 bit IP address and 32 bit mask
- Supernetting technique p- single routing entry to cover a block of classful addresses.

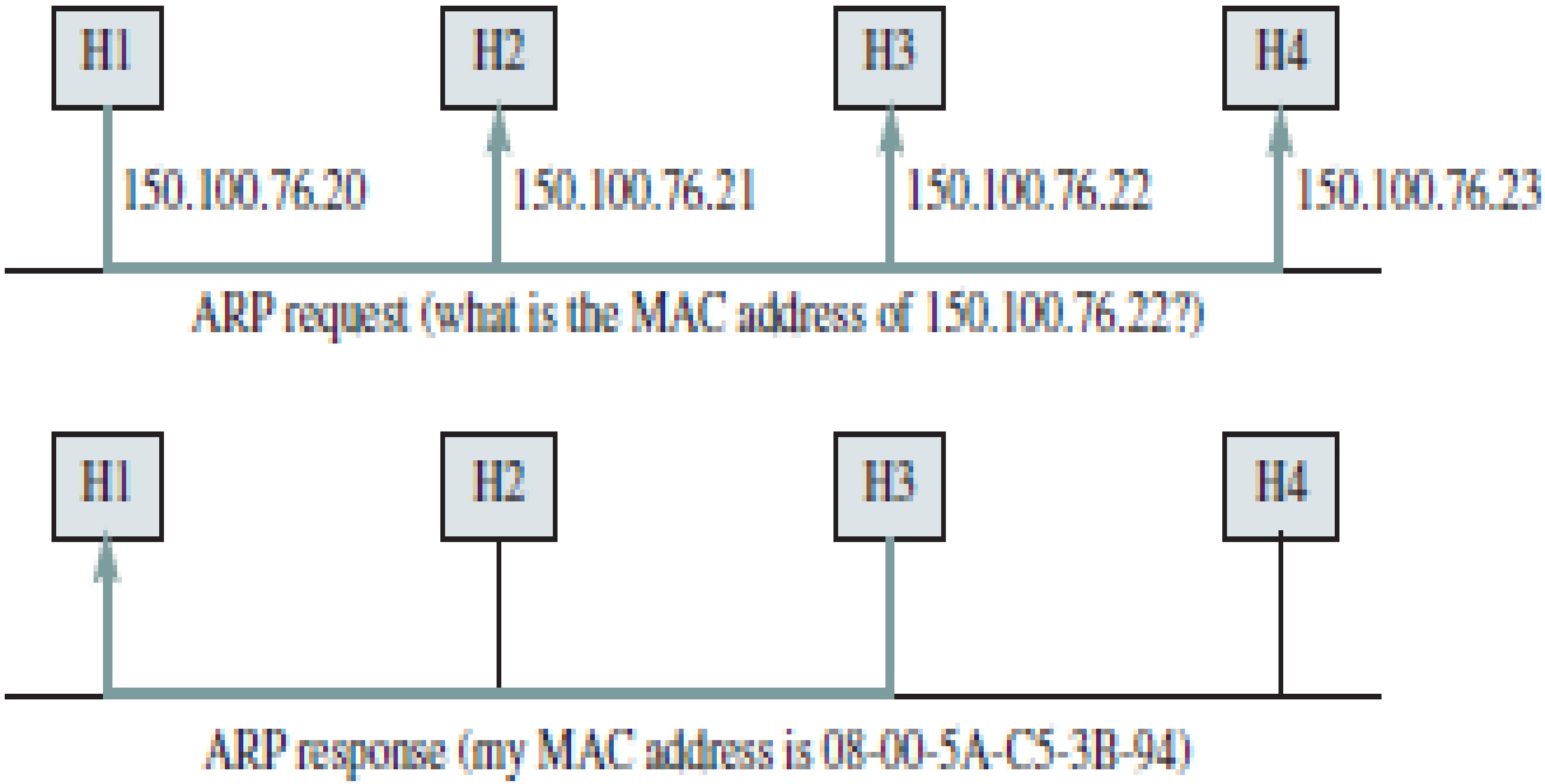
- Using a CIDR notation, a prefix 205.100.0.0 of length 22 is written as 205.100.0.0/22. The /22 notation indicates that the network mask is 22 bits or 255.255.252.0
- For example, instead of having four entries for a contiguous set of Class C addresses(e.g. 205.100.0.0 - 205.100.3.0), CIDR allows a single routing entry 205.100.0.0/22, which includes all IP addresses from 205.100.0.0 205.100.3.255

Mask 255.255.252.0 = 11111111 11111111 11111100 00000000
Supernet address 205.100.0.0 = 11001101 01100100 00000000 00000000

- The use of variable-length prefixes requires that the routing tables be searched to find the longest prefix match.
- For example, a routing table may contain entries for the above supernet 205.100.0.0/22 as well as for the even larger supernet 205.100.0.0/20.
- A packet with destination address 205.100.1.1 will match both of these entries, so the algorithm must select the match with the longest prefix.

Address Resolution Protocol

- IP packets must be delivered to the underlying network technology – different address format – ARP
- Suppose H1 wants to send an IP packet to H3 but does not know the MAC address of H3
- H1 first broadcasts an ARP request packet asking the destination host(which is identified by H3's IP address) to reply.
- All host in the network receive the packet, but only the intended host(which is H3) responds to H1.
- The ARP response packet contains H3's MAC and IP addresses.
- For future use, H1 caches H3's IP and MAC addresses in its ARP table

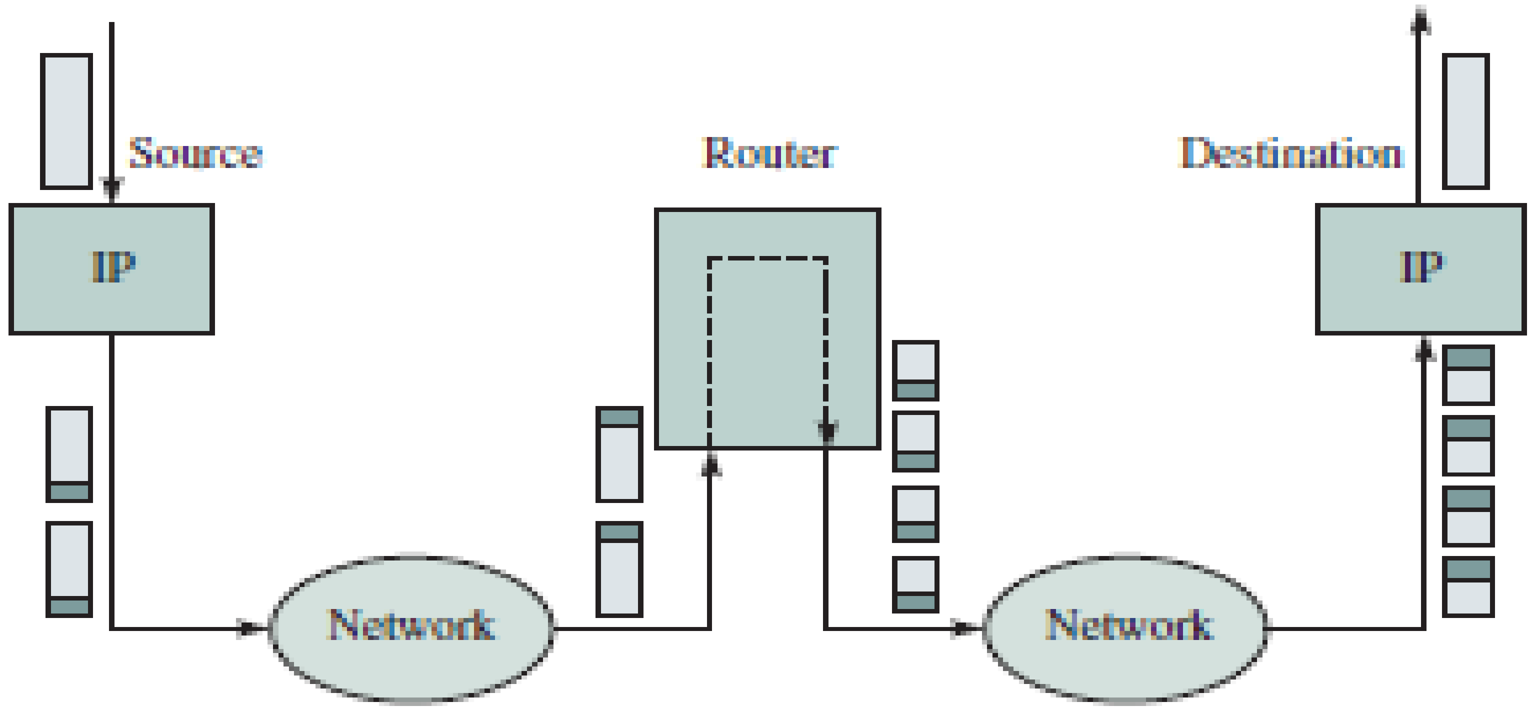

H1
H2
H3
H4
150.100.76.20
150.100.76.21
150.100.76.22
150.100.76.23
ARP request (what is the MAC address of 150.100.76.22?)
H1
H2
H3
H4
ARP response (my MAC address is 08-00-5A-C5-3B-94)

Reverse Address Resolution Protocol

- In some situations, a host may know its MAC address but not its IP address
- For example, when a diskless computer is being bootstrapped, it can read the MAC address from its Ethernet card. However, its IP address is usually kept separately in a disk at the server.
- To obtain its IP address, the host first broadcasts an RARP request packet containing its MAC address on the
- network.
- All hosts on the network receive the packet, but only the server replies to the host by sending an RARP response packet containing the host's MAC and IP addresses.
- Limitation: The server must be located on the same physical network as the host.

Fragmentation and Reassembly

- Fragmentation means the division of a packet into smaller units to accommodate a protocols MTU.
- Each network imposes a certain packet-size limitation on the packets that can be carried, called the maximum transmission unit(MTU).
- When IP wants send a packet that is larger than MTU of physical-network, IP breaks packet into smaller fragments.
- Each fragment is sent independently to the destination.



- Destination IP is responsible for reassembling the fragments into the original packet.
- To reassemble the fragments, the destination will wait until it has received all the fragments belonging to the same packet

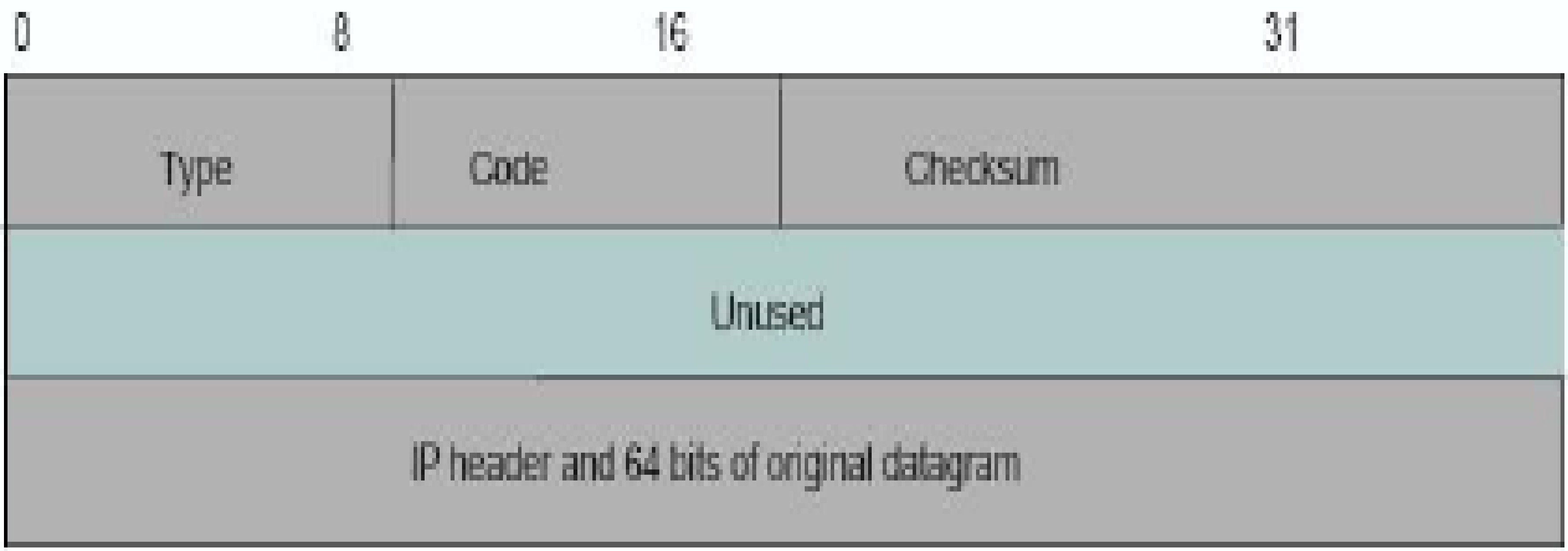
- Drawback:
  - Total overhead increases because each fragment must have a header
  - Performance penalty: If one of the fragments is lost packet cannot be reassembled at the destination
  - rest of the fragments have to be discarded. This process wastes transmission bandwidth.

Identification, Flags and Fragment-offset

- Identification, flags and fragment offset field in the IP header
- *Identification:*
  - This is used to identify to which packet a particular fragment belongs to
- *Flag*
  - *One unused bit*
  - *One "don't fragment" bit*
  - *One "more fragment" bit*
- *Fragment offset*
  - identifies location of a fragment in a packet

ICMP: Error and Control Messages

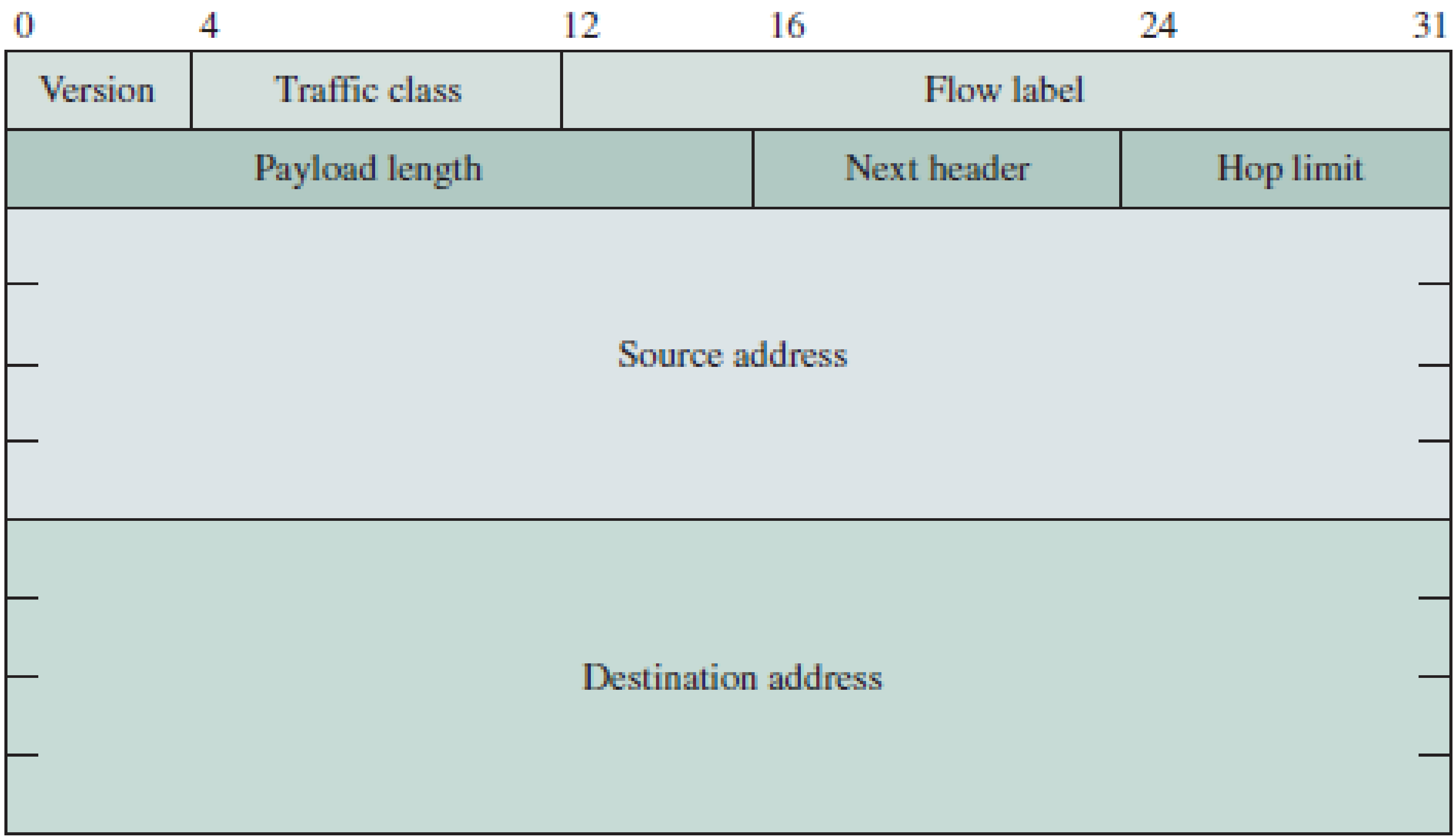
- Protocol to handle error and control messages
- Various fields in the header are:
  - Type
  - Code
  - Checksum
  - IP header plus original datagram


0
8
16
31
Type
Code
Checksum
Unused
IP header and 64 bits of original datagram

Echo Request and Echo Reply

- When destination receives an echo request message from a source, the destination simply replies with a corresponding echo reply message back to the source
- The echo request and echo reply messages are used in the PING program and are often used to determine whether a remote host is alive

IPv6


0
4
12
16
24
31
Version
Traffic class
Flow label
Payload length
Next header
Hop limit
Source address
Destination address

- Version:
  - This specifies version number of protocol. For IPv6,version=6.
- Traffic class:
  - This specifies priority of packet. This is used to support differential service
- Flow label:
  - This is used to identify QoS requested by packet.
- Payload length:
  - This indicates length of data. Maximum length=65535 bytes

- Next header
  - This identifies type of extension header that follows the basic header
- Hop limit
  - This specifies number of hops the packet can travel before being dropped by a router
- Source address & destination address
  - These identify source host and destination host respectively

Changes From IPv4 TO IPv6

- *Longer address fields:*
  - Length of address field is extended from 32 bits to 128 bits. The address space can support up to 3.4*1038 hosts.
- *Simplified header format:*
  - Some of the header fields in IPv4 such as identification, flags and fragment offset do not appear in the IPv6 header.
- *Flexible support for options:*
  - The options in IPv6 appear in optional extension headers that are encoded in a more efficient and flexible fashion than they are in IPv4.

- *Flow label capability:*
  - IPv6 adds a "flow label" to identify a certain packet "flow" that requires a certain QoS
- *Security:*
  - IPv6 supports build-in authentication and confidentiality.
- *Large packets:*
  - IPv6 supports payloads that are longer than 65 Kbytes called jumbo payloads.
- *Fragmentation at source only:*
  - Routers do not perform packet fragmentation. If a packet needs to be fragmented, the source should check the minimum MTU along the path and perform the necessary fragmentation.

- *No checksum fields:*
  - The checksum field has been removed to reduce packet processing time in a router. Packets carried by the physical network(such as Ethernet, token shop) are typically already checked.

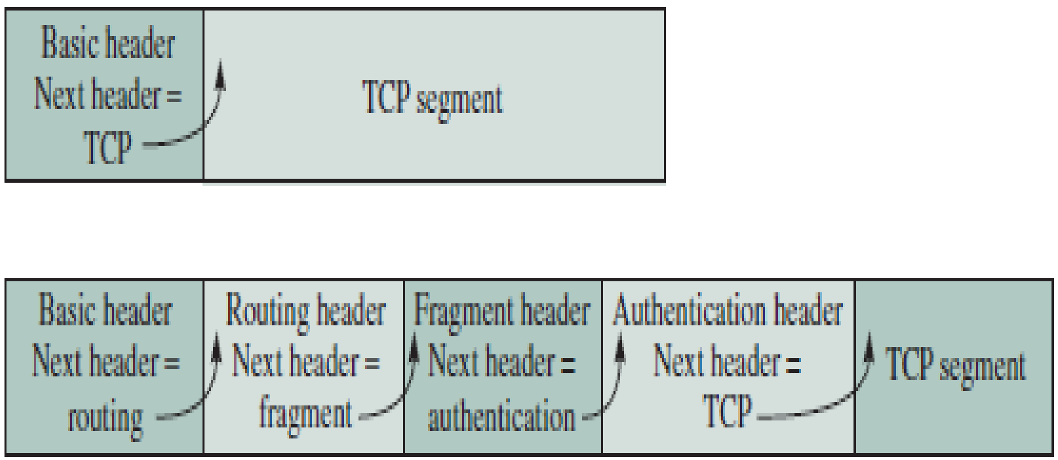
Network Addressing

- IPv6 addresses are divided into three categories:
  - Unicast addresses
  - Multicast addresses
  - Anycast addresses
- Special purpose addresses
  - 0::0 – used as the source address
  - ::1
- IPv4 compatible addresses are needed during the transition period where an IPv6 packet needs to be "tunneled" across an IPv4 network.
- IP mapped addresses are used to indicate IPv4 hosts and routers that do not support IPv6

	n bits	m bits	o bits	p bits	(125-m-n-o-p) bits
010	Registry ID	Provider ID	Subscriber ID	Subnet ID	Interface ID

- Provides based unicast addresses are identified by the prefix 010. These addresses will be mainly used by the
- Internet service providers to assign addresses to their subscribers.

Extension Headers

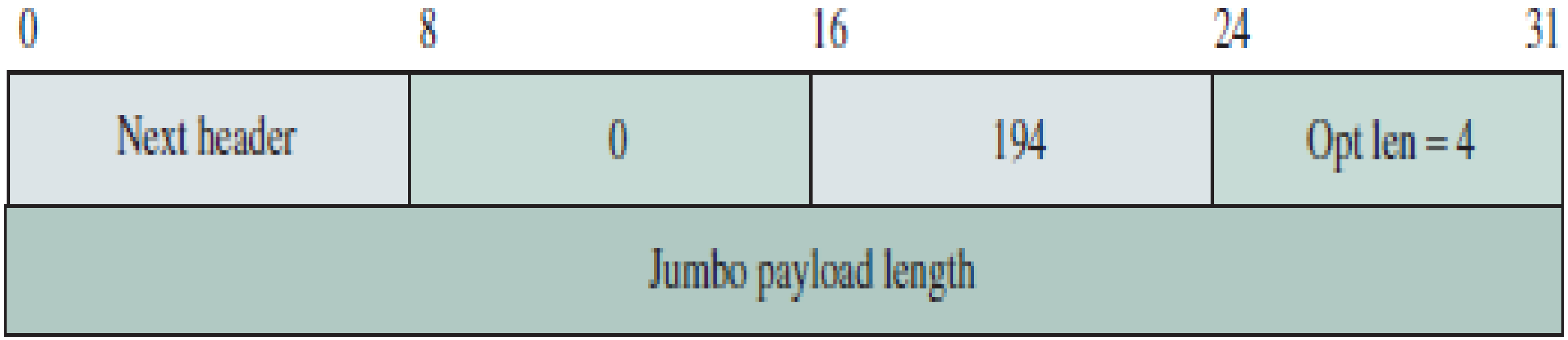


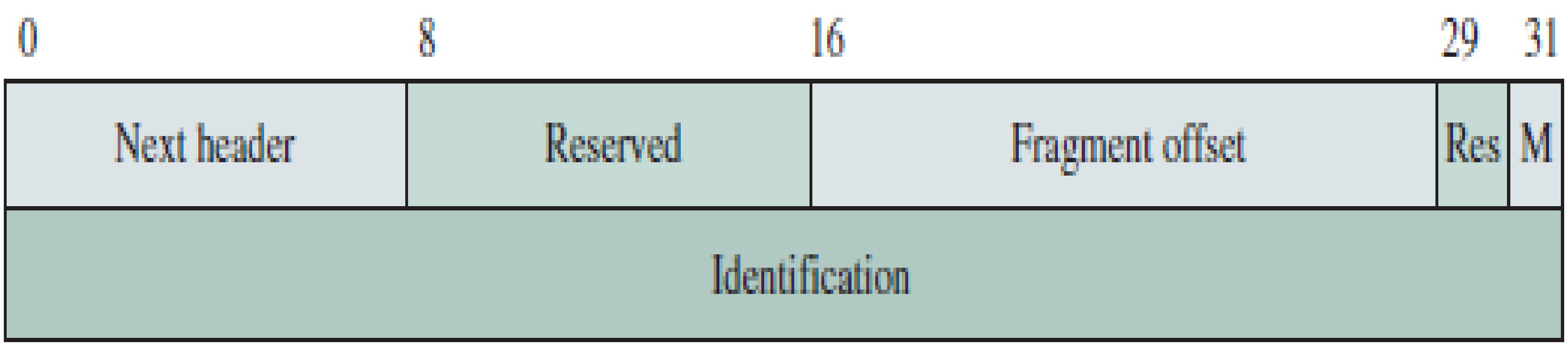
Header code	Header type
0	Hop-by-hop options header
43	Routing header
44	Fragment header
51	Authentication header
52	Encapsulating security payload header
60	Destination options header

- To support extra functionalities, IPv6 allows an arbitrary number of extension headers to be placed between the basic header and the payload
- Encoded more efficiently and flexibly
- Header codes and Header type

Large Packet

- Next header: This identifies type of header immediately following this header
- The value 194 defines a jumbo payload option
- Payload length: This must be set to 0.
- Option length: This specifies size of jumbo payload length field
- Jumbo payload length: This specifies payload size. Maximum payload size=232 bytes


0
8
16
24
31
Next header
0
194
Opt len = 4
Jumbo payload length


0
8
16
29
31
Next header
Reserved
Fragment offset
Res
M
Identification

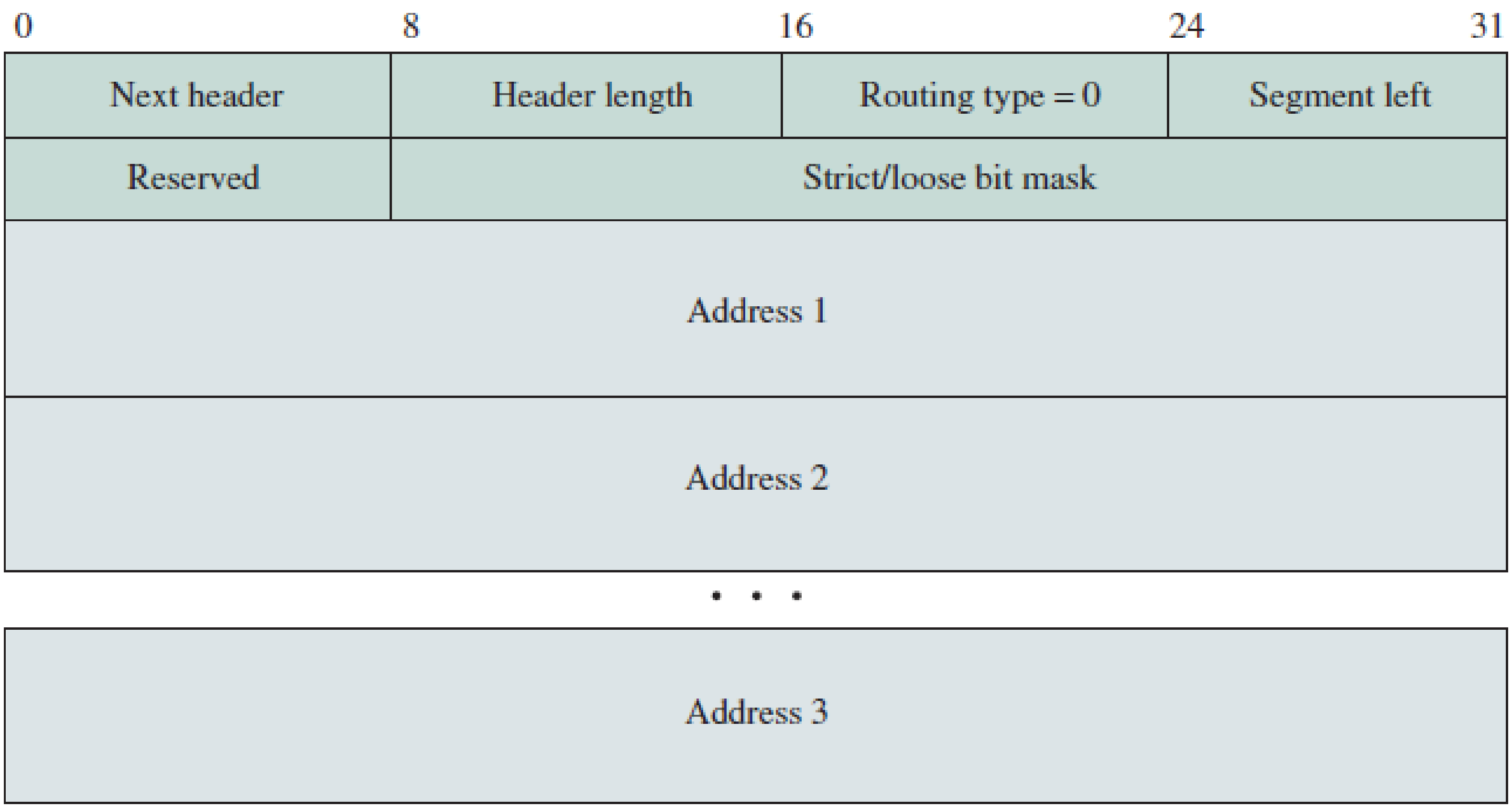
Fragmentation

- A source can find the minimum MTU along the path from the source to the destination by performing a "path MTU discovery" procedure
- Advantage of doing fragmentation at the source only is that routers can process packets faster
- Disadvantage: Path between a source and a destination must remain reasonably static

Source Routing

- IPv6 allows for source routing to specify the sequence of routers to be visited by the packet to reach the destination.
- Header length: This specifies the length of the routing extension header (in units of 64 bits), excluding first 64 bits.
- Currently, only type=0 is specified
- Segment left: This identifies the number of route segments remaining before the destination is reached. Maximum value=23.

- Each bit in the strict/loose bit mask indicates whether the next destination address must be followed strictly

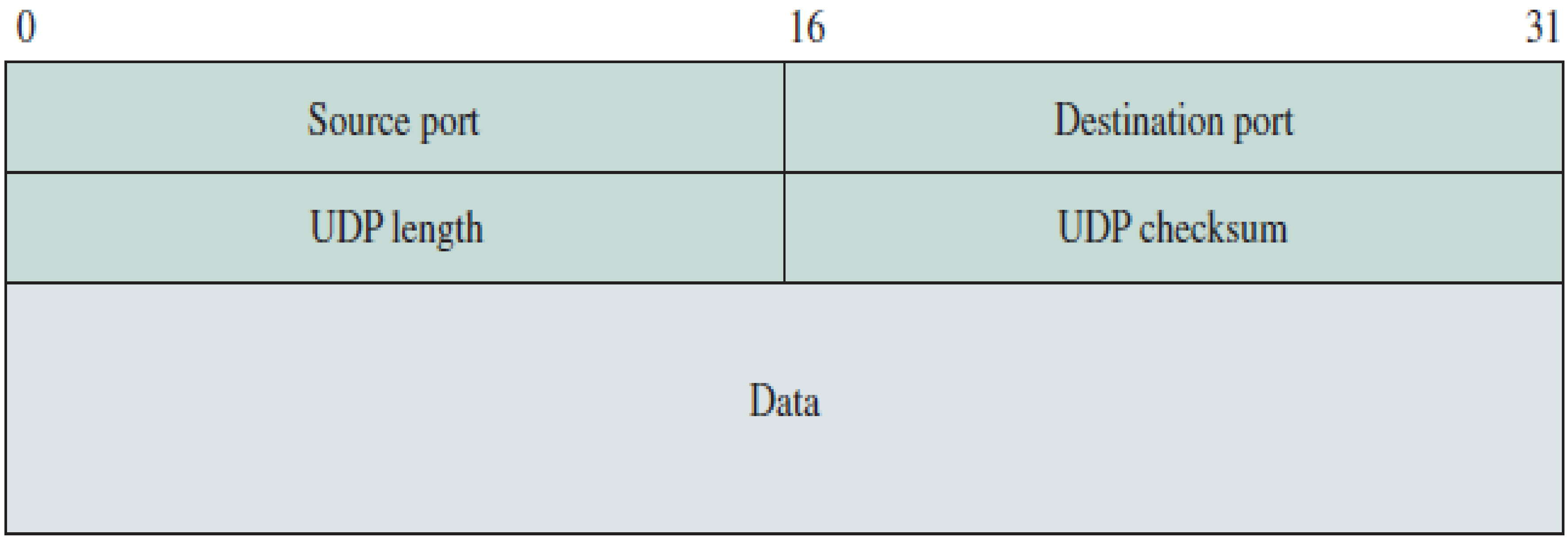


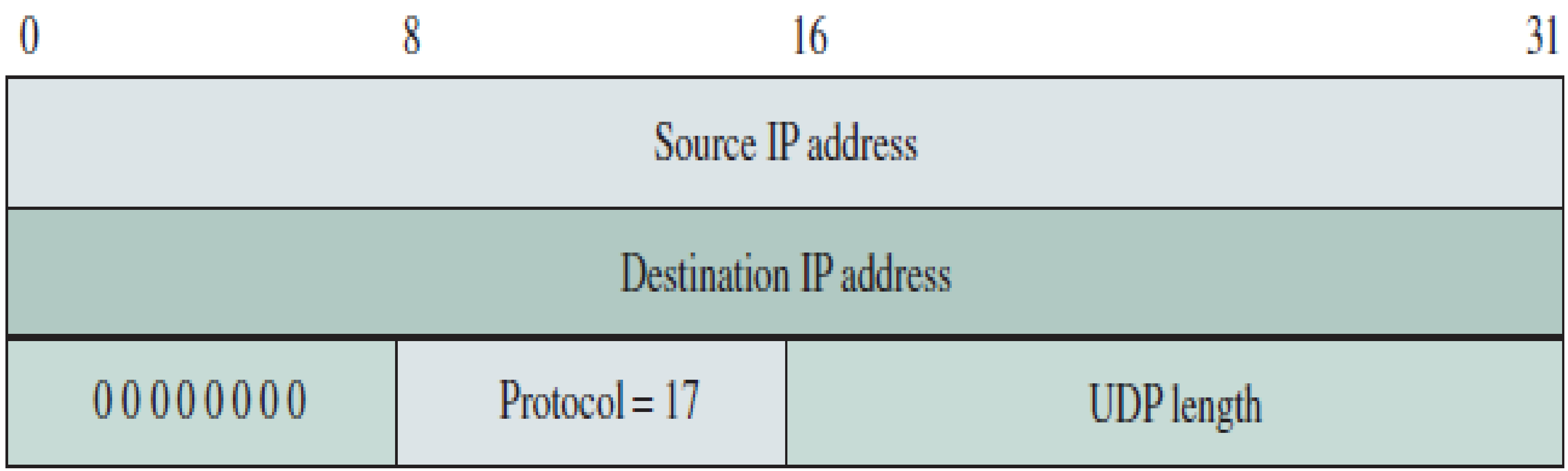
Migration Issues From IPv4 TO IPv6

- A tunnel is a path created between two nodes so that the tunnel appears as a single link to the user
- An IPv4 tunnel allows IPv6 packets to be forwarded across an IPv4 network without the IPv6 user having to worry about how packets are actually forwarded in the IPv4 network.
- A tunnel is typically realized by encapsulating each user packet in another packet that can be forwarded along the tunnel.

User Datagram Protocol (UDP)

- This is an unreliable, connectionless transport layer protocol.
- This provides only two additional services beyond IP: demultiplexing and error checking on data.
- This can optionally check the integrity of the entire datagram.
- Applications that do not require zero packet loss such as in packet voice systems are well suited to UDP.
- Applications that use UDP include DNS, SNMP, RTP & TFTP.


0
16
31
Source port
Destination port
UDP length
UDP checksum
Data


0
8
16
31
Source IP address
Destination IP address
0 0 0 0 0 0 0 0
Protocol = 17
UDP length

- The destination port allows the UDP module to demultiplex datagrams to the correct application in a given host.
- The source port identifies the particular application in the source host to receive replies.
- The length field indicates the number of bytes in datagram(including header and eat).
- Checksum field detects errors in the datagram and its use is optional.

- Checksum computation procedure is similar to that in computing IP checksum except for 2 new twists.
  - First, if the length of the datagram is not a multiple of 16 bits, the datagram will be padded out with 0s to make it a multiple of 16bits.
  - Second, UDP adds a pseudoheader to the beginning of the datagram when performing the checksum computation.
- The pseudoheader is also created by the source and destination hosts only during the checksum computation and is not transmitted.

Internet Routing Protocols

- Autonomous System – set of routers and networks that are administered by a single organization
- Three categories are:
  - Stub AS/Single homed AS
  - Multihomed AS
  - Transit AS
- Routing Protocols:
  - Interior gateway protocol
  - Exterior gateway protocol

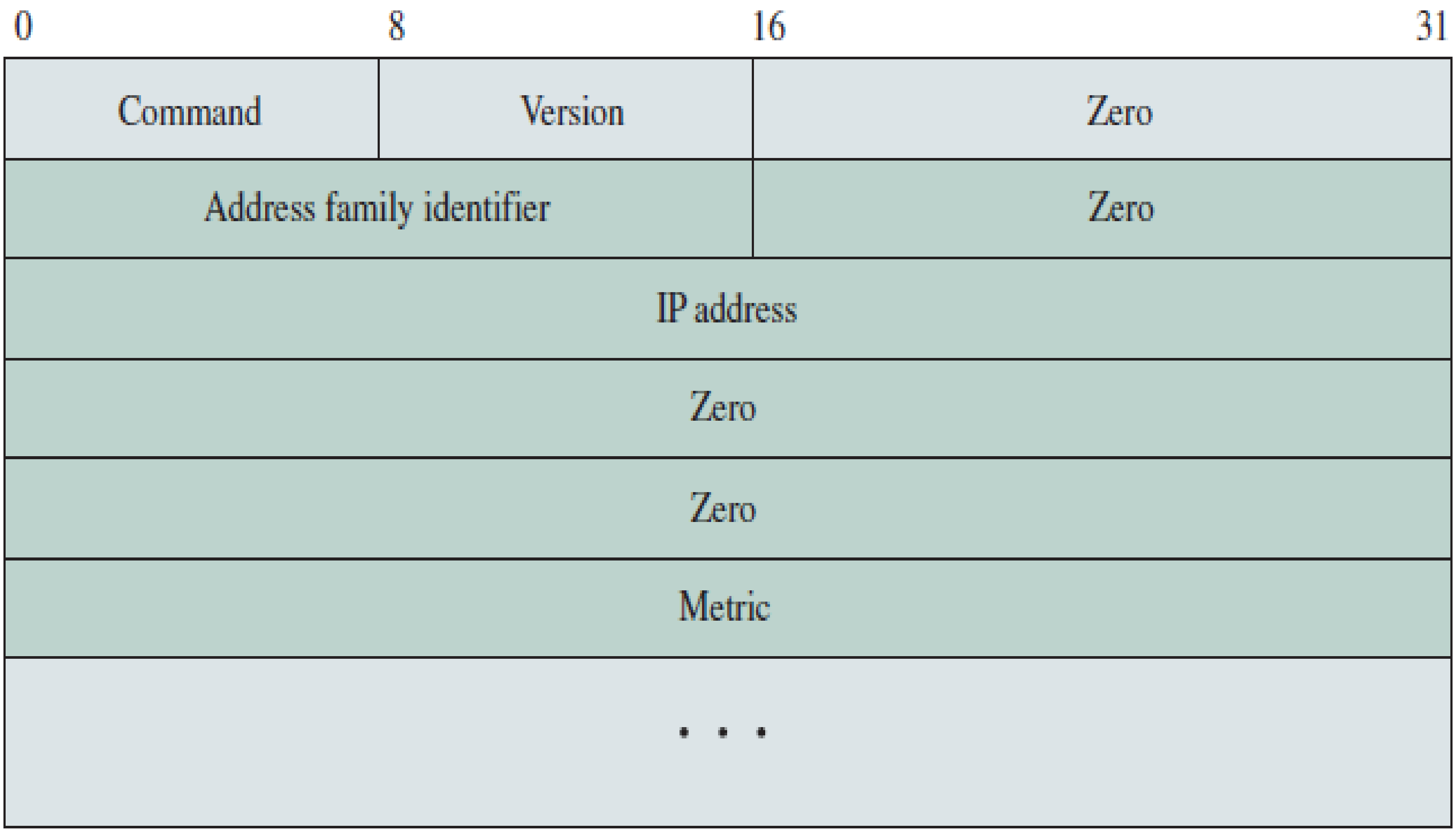
Routing Information Protocol

- Distance – vector algorithm
- Runs on UDP with the standard port number as 520
- Metric used – number of hops
- Max limited to 15
  - suitable for small networks (local area environments)
  - value of 16 is reserved to represent infinity
  - small number limits the *count-to-infinity* problem

Working of RIP

- Router sends update message to neighbors every 30 sec
- A router expects to receive an update message from each of its neighbors within 180 seconds in the worst case
- If router does not receive update message from neighbor X within this limit, it assumes the link to X has failed and sets the corresponding minimum cost to 16 (infinity)
- If the router later receives a valid minimum cost to X from another neighbor router will replace infinity with the new minimum cost.

- Routers run RIP in active mode (advertise distance vector tables)
- Hosts can run RIP in passive mode (update distance vector tables, but do not advertise)
- RIP datagrams broadcast over LANs & specifically addressed on pt-pt or multi-access non-broadcast nets
- Two RIP packet types:
  - *request* to ask neighbor for distance vector table
  - *response* to advertise distance vector table


0
8
16
31
Command
Version
Zero
Address family identifier
Zero
IP address
Zero
Zero
Metric
• • •

RIP Message Format

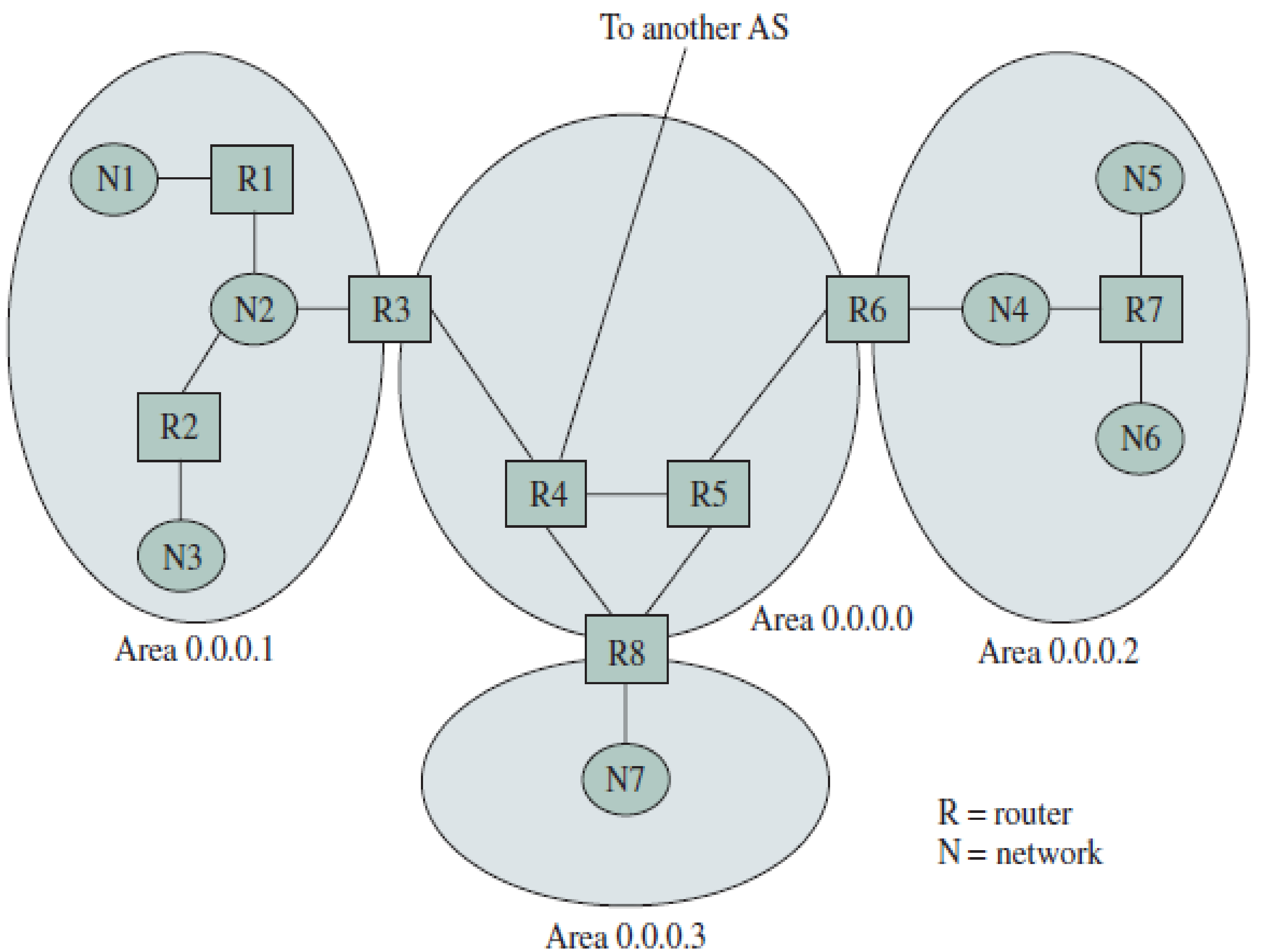
- Command: request or response
- Version: v1 or v2
- One or more of:
  - Address Family: 2 for IP
- IP Address: network or host destination
- Metric: number of hops to destination
- Does not have access to subnet mask information
- Cannot work with variable-length subnet masks

Open Shortest Path First

- Each router is enabled to learn complete network topology
- Each router monitors the *link state* to each neighbor and floods the link-state information to other routers
- Each router builds an identical *link-state database*
- Allows router to build shortest path tree with router as root
- OSPF typically converges faster than RIP when there is a failure in the network

OSPF Features

- *Multiple routes* to a given destination, one per type of service
- Support for *variable-length subnetting* by including the subnet mask in the routing message
- More *flexible link cost* which can range from 1 to 65,535
- Distribution of traffic over *multiple paths* of equal cost
- *Authentication* to ensure routers exchange information with trusted neighbors
- Uses *notion of area* to partition sites into subsets
- Support *host-specific routes* as well as net-specific routes
- *Designated router* to minimize table maintenance overhead


To another AS
N1
R1
N2
R3
R2
N3
R4
R5
R6
N4
R7
N5
N6
R8
N7
Area 0.0.0.1
Area 0.0.0.0
Area 0.0.0.2
Area 0.0.0.3
R = router
N = network

- To improve scalability, AS may be partitioned into *areas*
  - Area is identified by 32-bit Area ID
  - Router in area only knows complete topology inside area & limits the flooding of link-state information to area
  - *Area border routers* summarize info from other areas
- Each area must be connected to *backbone area* (0.0.0.0)
  - Distributes routing info between areas
- *Internal router* has all links to nets within the same area
- *Area border router* has links to more than one area
- *backbone router* has links connected to the backbone
- *Autonomous system boundary (ASB) router* has links to another autonomous system.

Neighbor, Adjacent & Designated Routers

- *Neighbor routers*: two routers that have interfaces to a common network
  - Neighbors are discovered dynamically by *Hello protocol*
- Each neighbor of a router described by a state
  - down, attempt, init, 2-way, Exchange, Loading, Full
- *Adjacent router*: neighbor routers become adjacent when they synchronize topology databases by exchange of link state information
  - Neighbors on point-to-point links become adjacent
  - Routers on multiaccess nets become adjacent only to *designated & backup designated routers*
  - Reduces size of topological database & routing traffic

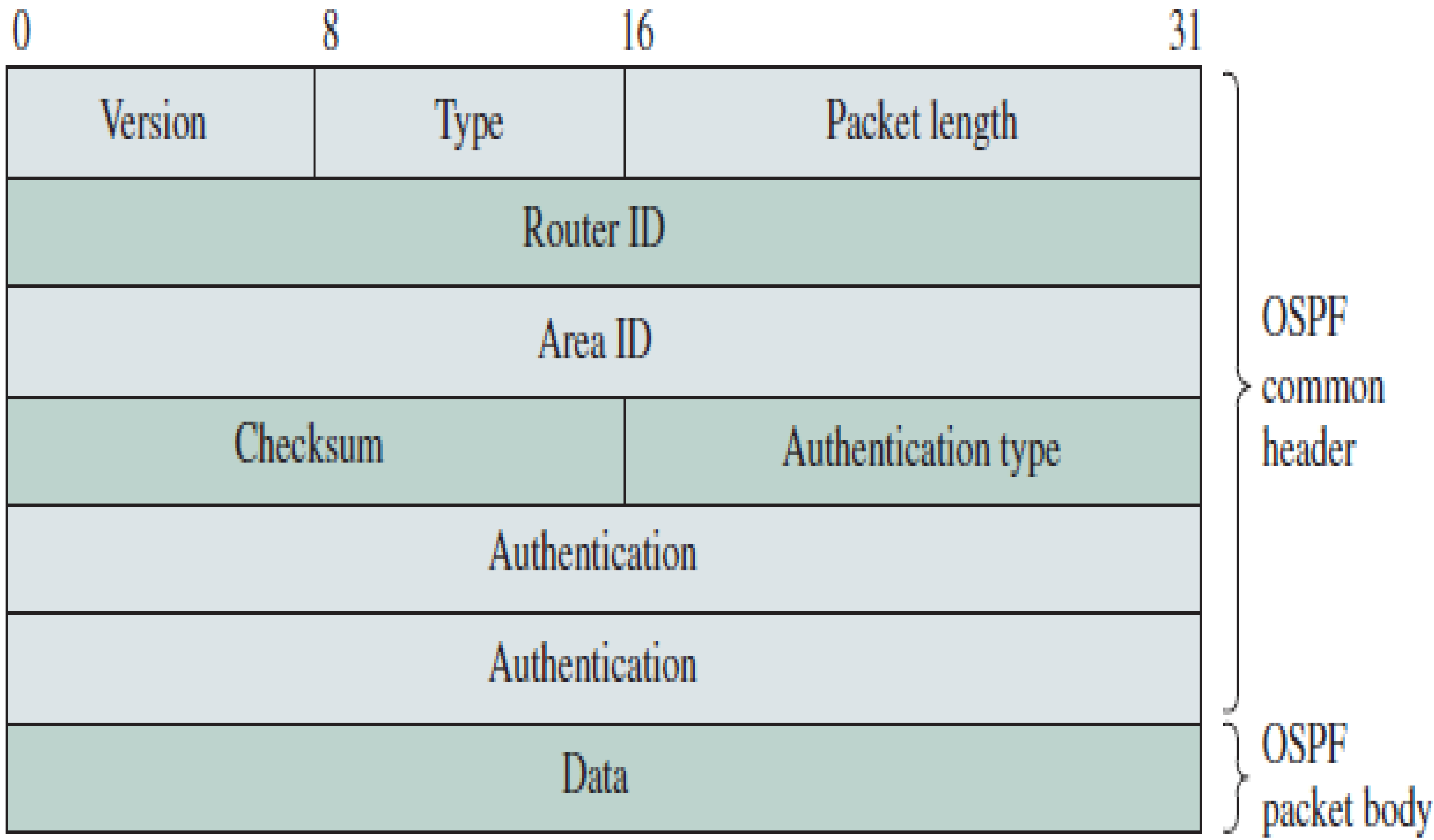
Designated Routers

- Reduces number of adjacencies
- Elected by each multiaccess network after neighbor discovery by hello protocol
- Election based on priority & id fields
- Generates link advertisements that list routers attached to a multi-access network
- Forms adjacencies with routers on multi-access network
- Backup prepared to take over if designated router fails

OSPF Operation

- Runs directly over IP, using IP protocol 89
- Composed of OSPF header followed by body of a particular packet type
- Stages of OSPF operation
  - Neighbors are discovered through sending Hello messages and designated routers are elected in multiple access
  - Adjacencies are established and link state databases are synchronized
  - LSAs are exchanged by adjacent routers

OSPF Header

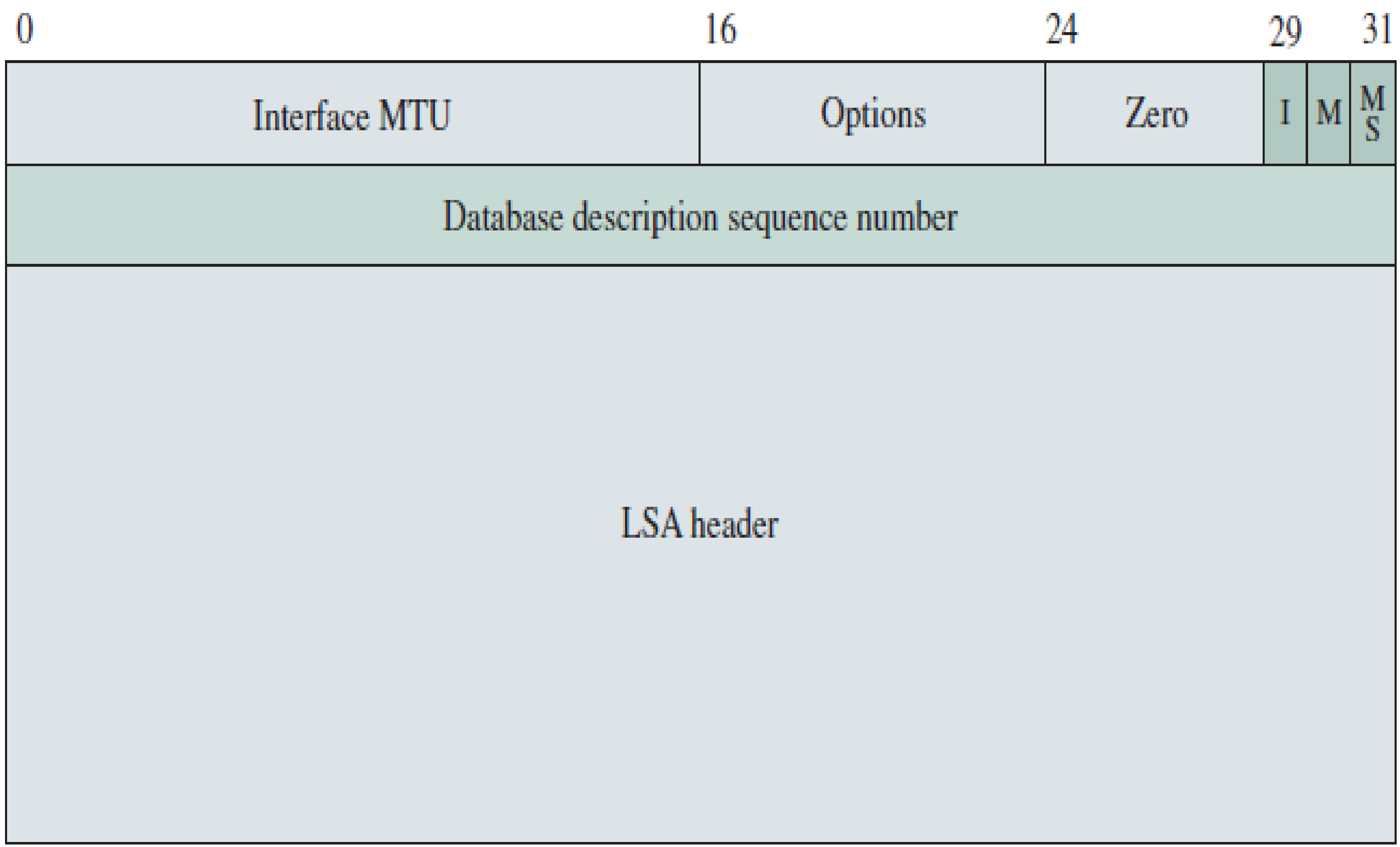

0
8
16
31
Version
Type
Packet length
Router ID
Area ID
Checksum
Authentication type
Authentication
Authentication
Data
OSPF common header
OSPF packet body

Stage 1

0	16	24 31
Network mask		
Hello interval	Options	Priority
Dead interval		
Designated router		
Backup designated router		
Neighbor 1		
. . .		
Neighbor n		

- Send Hello packets to establish & maintain neighbor relationship
- Hello interval: number of seconds between Hello packets
- Priority: used to elect designated router & backup

Stage 2



OSFP database description packet

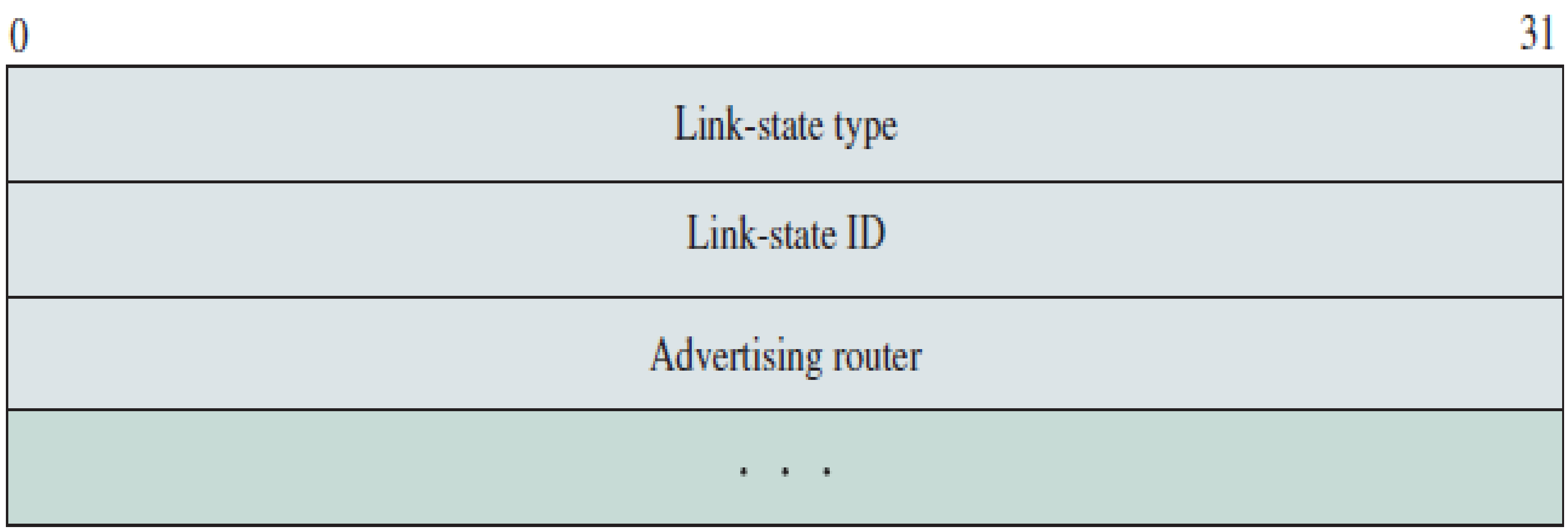
- Once neighbor routers become adjacent, they exchange database description packets to synchronize their link-state databases
- Init bit 1 if pkt is first in sequence of database description packets
- More bit 1 if there are more database description packets to follow
- Master/Slave bit indicates that the router is the master.
- Link state ad (LSA) header describes state of router or network; contains info to uniquely identify entry in LSA (type, ID, and advertising router).
- Can have multiple LSA headers

<table>
<tr><th>0</th><th>16</th><th>24   31</th></tr>
<tr><td>Link-state age</td><td>Options</td><td>Link-state type</td></tr>
<tr><td colspan="3">Link-stage ID</td></tr>
<tr><td colspan="3">Advertising router</td></tr>
<tr><td colspan="3">Link-state sequence number</td></tr>
<tr><td>Link-state checksum</td><td colspan="2">Length</td></tr>
</table>

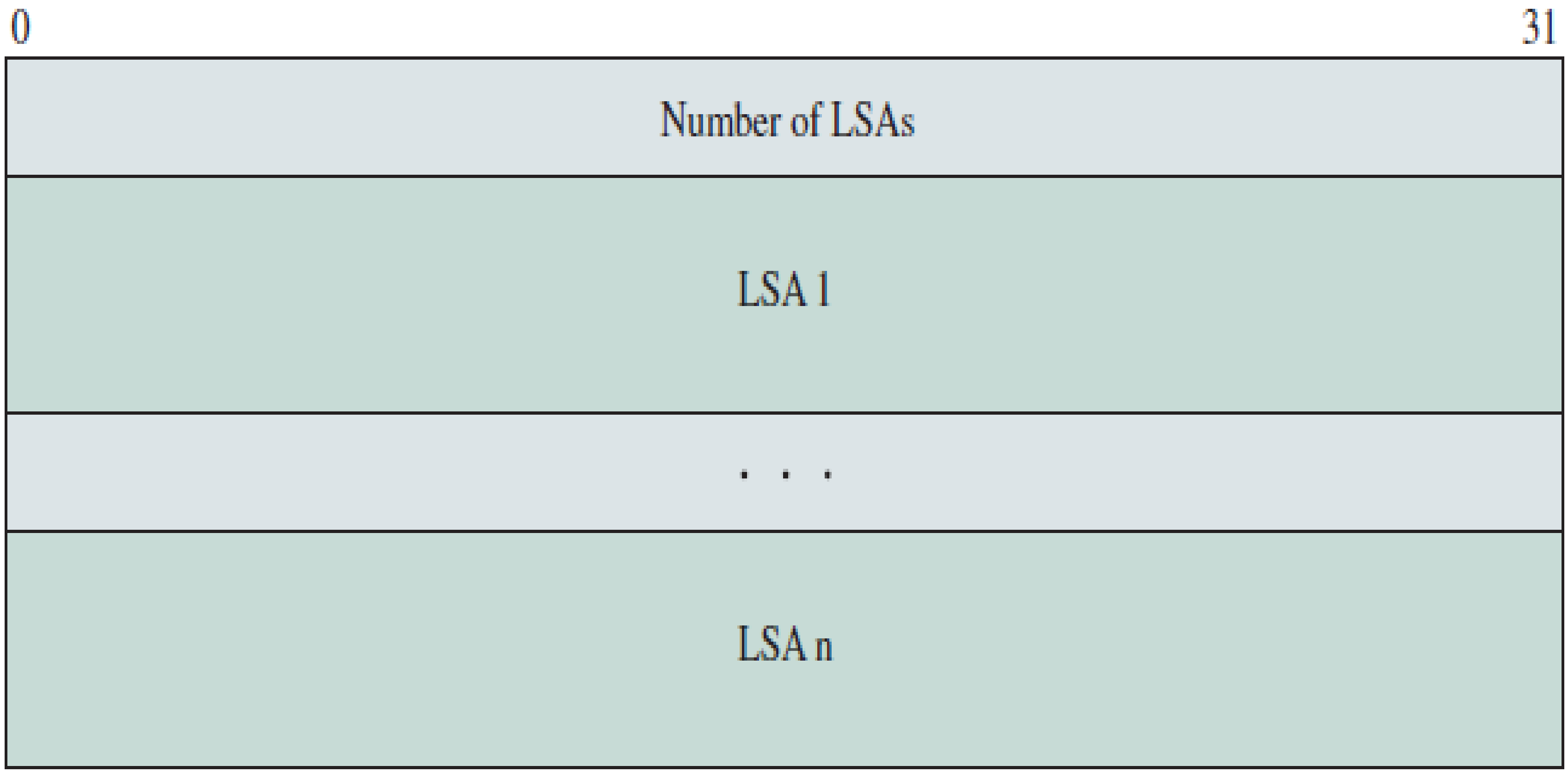
LSA Header

Stage 3

- Router sends a LS request packet to neighbor to update part of its link-state database
- Each LSA request is specified by the link state type, link state ID, and the advertising router.



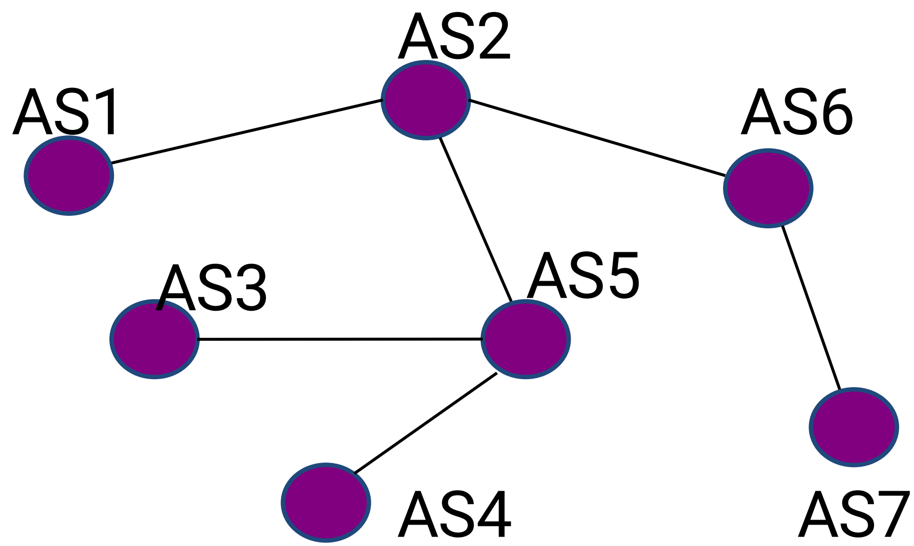
OSPF link-state request packet



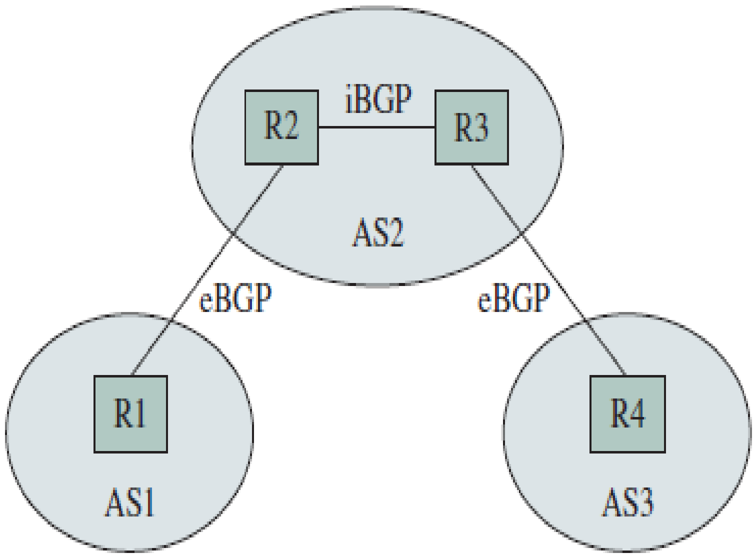
OSPF link-state update packet

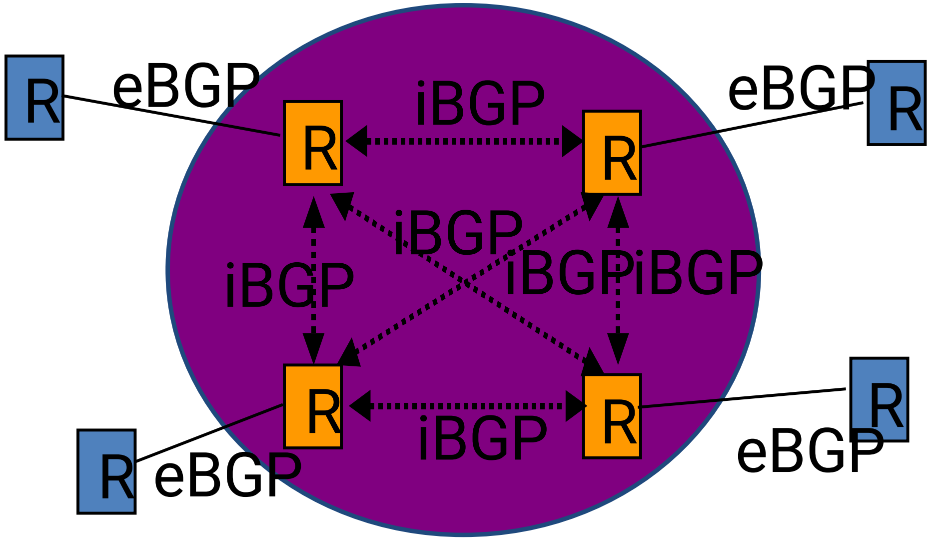
Border Gateway Protocol

- Exterior gateway protocol
- Exchange of network reachability information among BGP routers.
- TCP connection
- BGP messages are exchanged between the BGP routers/speakers
- Network reachability information – Sequence of AS
- Path vector protocol
- Redistribution of routing information



- *BGP speaker*: a router running BGP
- *Peers or neighbors*: two speakers exchanging information on a connection
- BGP peers use TCP (port 179) to exchange messages
- Initially, BGP peers exchange entire BGP routing table
  - Incremental updates sent subsequently
  - Reduces bandwidth usage and processing overhead
  - Keepalive messages sent periodically (30 seconds)
- *Internal BGP* (iBPG) between BGP routers in same AS
- *External BGP* (eBGP) connections across AS borders





- eBGP to exchange reachability information in different AS's
  - eBGP peers directly connected
- iBGP to ensure net reachability info is consistent among the BGP speakers in the same AS
  - usually not directly connected
  - iBGP speakers exchange info learned from other iBGP speakers, and thus fully meshed

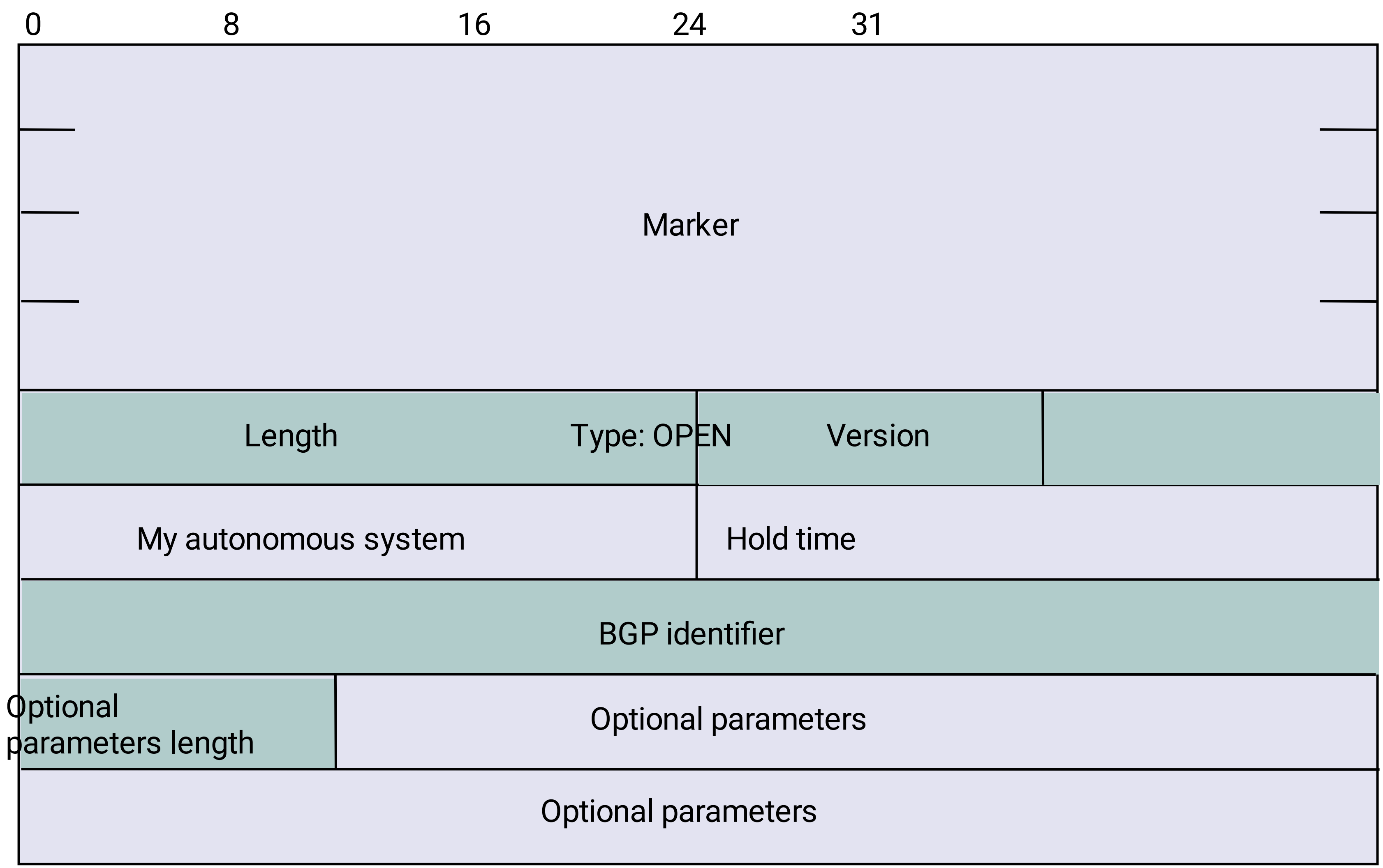
Path Selection

- Each BGP speaker
  - Evaluates paths to a destination from an AS border router
  - Selects the best that complies with policies
  - Advertises that route to all BGP neighbors
- BGP assigns a preference order to each path & selects path with highest value; BGP does not keep a cost metric to any path
- When multiple paths to a destination exist, BGP maintains all of the paths, but only advertises the one with highest preference value

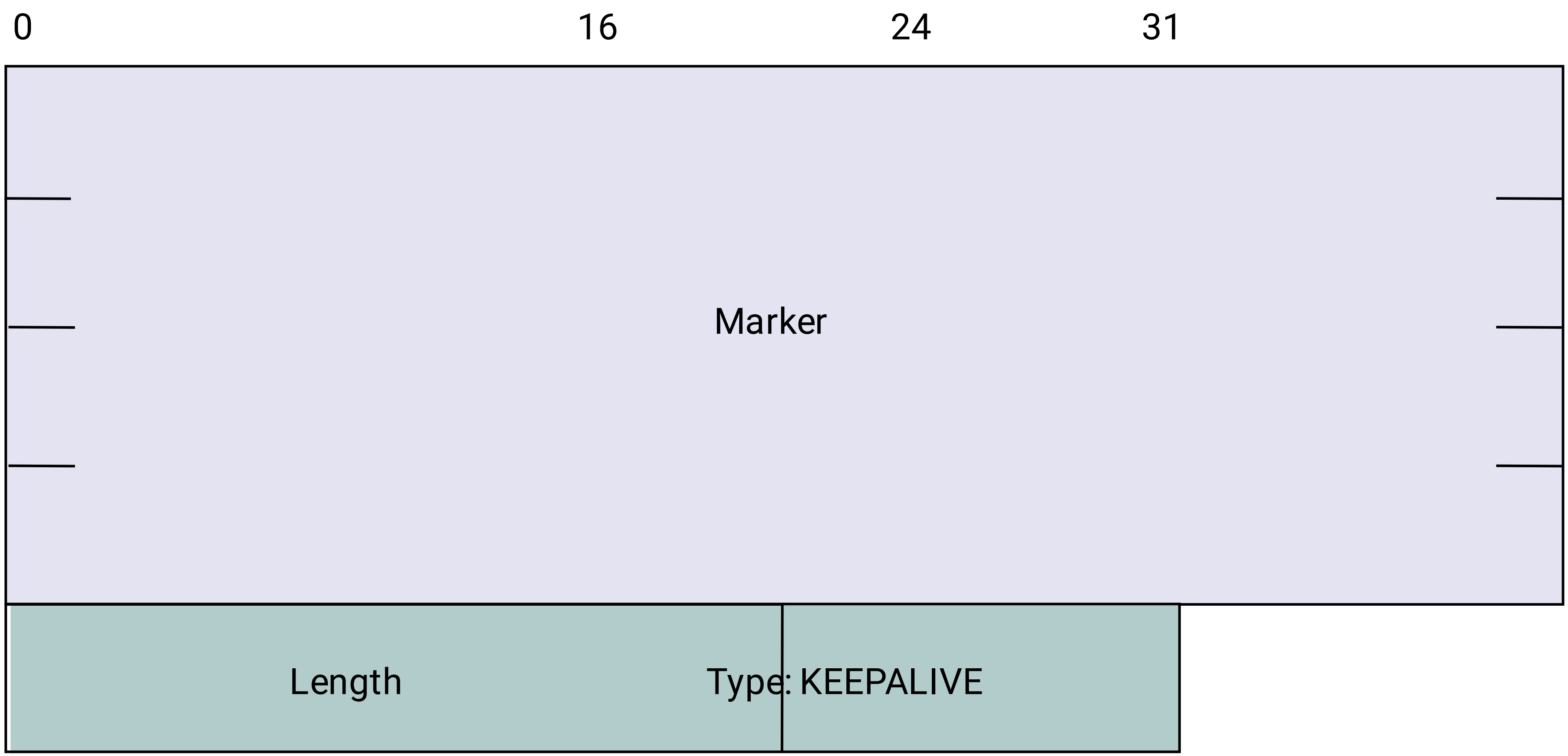
BGP Protocol

- Opening & confirming of a BGP connection with a neighbor router
- Maintaining the BGP connection
- Sending reachability information
- Notification of error conditions

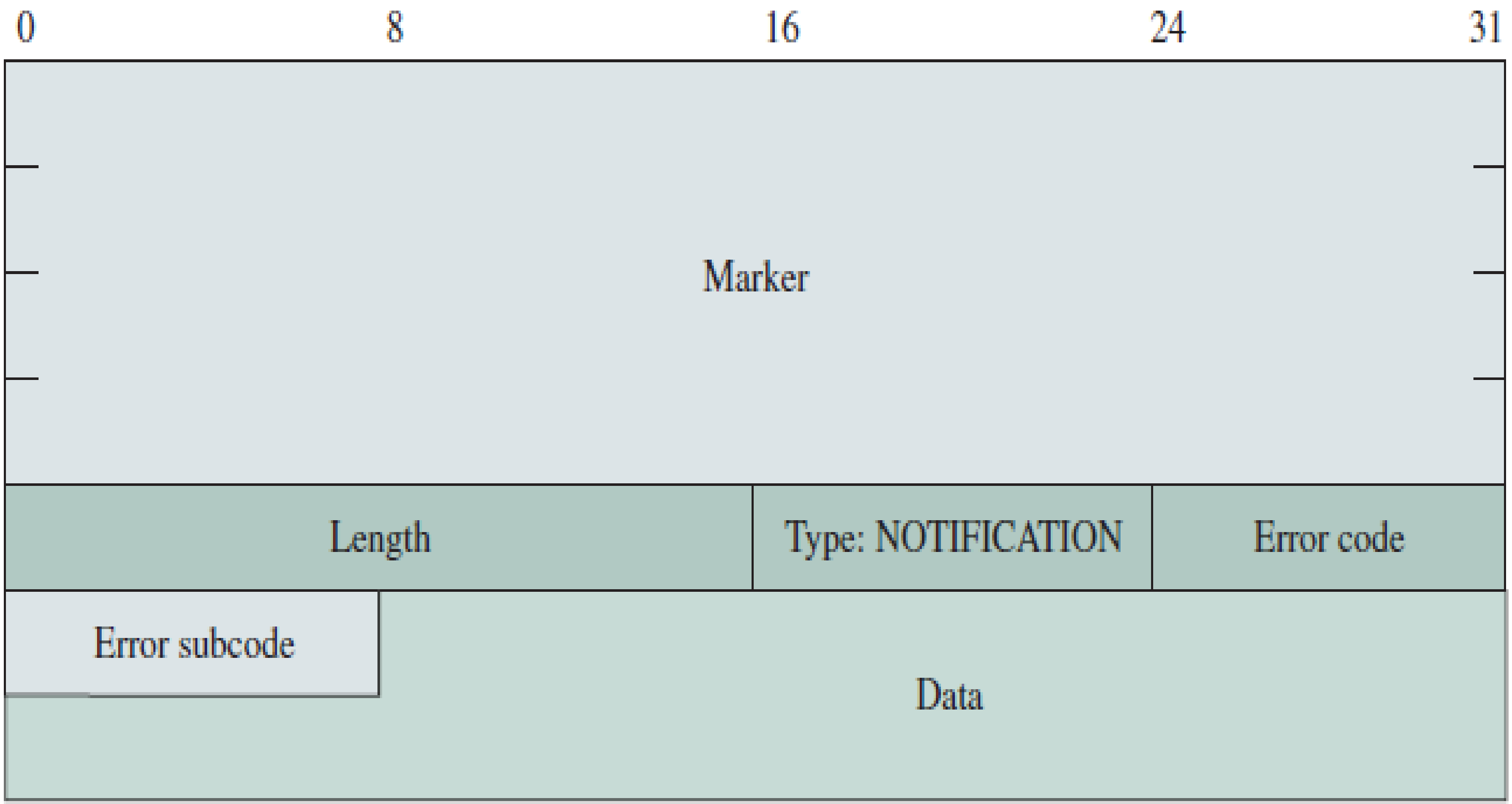
BGP Open Message



BGP Keep Alive Message



BGP Notification Message



BGP Update Message

Unfeasible routes length (two octets)
Withdrawn routes (variable)
Total path attribute length (two octets)
Path attributes (variable)
Network layer reachability information (variable)