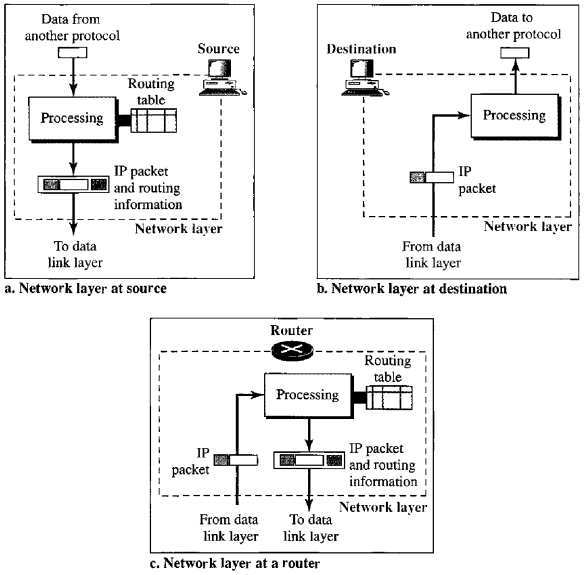
Figure 20.3 *Network layer at the source, router, and destination*



The network layer at the switch or router is responsible for routing the packet. When a packet arrives, the router or switch consults its routing table and finds the interface from which the packet must be sent. The packet, after some changes in the header, with the routing information is passed to the data link layer again.

The network layer at the destination is responsible for address verification; it makes sure that the destination address on the packet is the same as the address of the host. If the packet is a fragment, the network layer waits until all fragments have arrived, and then reassembles them and delivers the reassembled packet to the transport layer.

Internet as a Datagram Network

The Internet, at the network layer, is a packet-switched network. We discussed switching in Chapter 8. We said that, in general, switching can be divided into three broad categories: circuit switching, packet switching, and message switching. Packet switching uses either the virtual circuit approach or the datagram approach.

The Internet has chosen the datagram approach to switching in the network layer. It uses the universal addresses defined in the network layer to route packets from the source to the destination.

Switching at the network layer in the Internet uses the datagram approach to packet switching.

Internet as a Connectionless Network

Delivery of a packet can be accomplished by using either a connection-oriented or a connectionless network service. In a **connection-oriented service,** the source first makes a connection with the destination before sending a packet. When the connection is established, a sequence of packets from the same source to the same destination can be sent one after another. In this case, there is a relationship between packets. They are sent on the same path in sequential order. A packet is logically connected to the packet traveling before it and to the packet traveling after it. When all packets of a message have been delivered, the connection is terminated.

In a connection-oriented protocol, the decision about the route of a sequence of packets with the same source and destination addresses can be made only once, when the connection is established. Switches do not recalculate the route for each individual packet. This type of service is used in a virtual-circuit approach to packet switching such as in Frame Relay and ATM.

In **connectionless service,** the network layer protocol treats each packet independently, with each packet having no relationship to any other packet. The packets in a message may or may not travel the same path to their destination. This type of service is used in the datagram approach to packet switching. The Internet has chosen this type of service at the network layer.

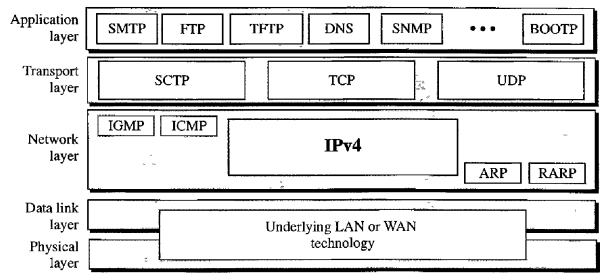
The reason for this decision is that the Internet is made of so many heterogeneous networks that it is almost impossible to create a connection from the source to the destination without knowing the nature of the networks in advance.

Communication at the network layer in the Internet is connectionless.

20.2 IPv4

The **Internet Protocol version 4 (IPv4)** is the delivery mechanism used by the TCP/IP protocols. Figure 20.4 shows the position of IPv4 in the suite.

Figure 20.4 *Position of IPv4 in TCP/IP protocol suite*



IPv4 is an unreliable and connectionless datagram protocol—a **best-effort delivery** service. The term *best-effort* means that IPv4 provides no error control or flow control (except for error detection on the header). IPv4 assumes the unreliability of the underlying layers and does its best to get a transmission through to its destination, but with no guarantees.

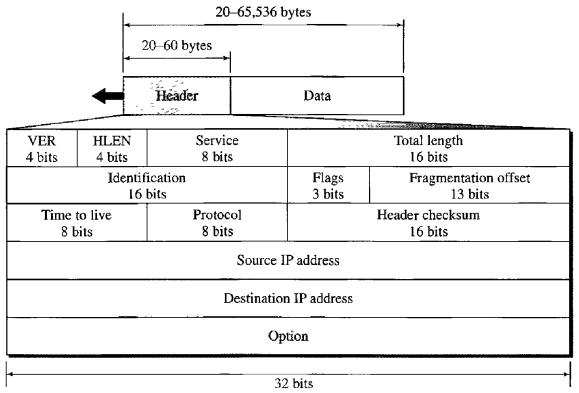
If reliability is important, IPv4 must be paired with a reliable protocol such as TCP. An example of a more commonly understood best-effort delivery service is the post office. The post office does its best to deliver the mail but does not always succeed. If an unregistered letter is lost, it is up to the sender or would-be recipient to discover the loss and rectify the problem. The post office itself does not keep track of every letter and cannot notify a sender of loss or damage.

IPv4 is also a connectionless protocol for a packet-switching network that uses the datagram approach (see Chapter 8). This means that each datagram is handled independently, and each datagram can follow a different route to the destination. This implies that datagrams sent by the same source to the same destination could arrive out of order. Also, some could be lost or corrupted during transmission. Again, IPv4 relies on a higher-level protocol to take care of all these problems.

Datagram

Packets in the IPv4 layer are called **datagrams.** Figure 20.5 shows the IPv4 datagram format.

Figure 20.5 *IPv4 datagram format*

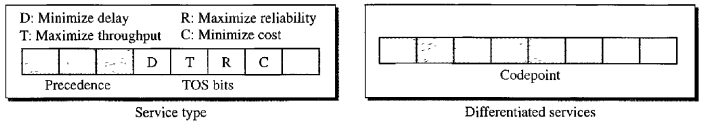


A datagram is a variable-length packet consisting of two parts: header and data. The header is 20 to 60 bytes in length and contains information essential to routing and

delivery. It is customary in TCP/IP to show the header in 4-byte sections. A brief description of each field is in order.

- **Version (VER).** This 4-bit field defines the version of the IPv4 protocol. Currently the version is 4. However, version 6 (or IPng) may totally replace version 4 in the future. This field tells the IPv4 software running in the processing machine that the datagram has the format of version 4. All fields must be interpreted as specified in the fourth version of the protocol. If the machine is using some other version of IPv4, the datagram is discarded rather than interpreted incorrectly.
- **Header length (HLEN).** This 4-bit field defines the total length of the datagram header in 4-byte words. This field is needed because the length of the header is variable (between 20 and 60 bytes). When there are no options, the header length is 20 bytes, and the value of this field is 5 ($5 \times 4 = 20$). When the option field is at its maximum size, the value of this field is 15 ($15 \times 4 = 60$).
- **Services.** IETF has changed the interpretation and name of this 8-bit field. This field, previously called **service type,** is now called **differentiated services.** We show both interpretations in Figure 20.6.

Figure 20.6 *Service type or differentiated services*



1. **Service Type**
   In this interpretation, the first 3 bits are called precedence bits. The next 4 bits are called **type of service (TOS)** bits, and the last bit is not used.
   a. **Precedence** is a 3-bit subfield ranging from 0 (000 in binary) to 7 (111 in binary). The precedence defines the priority of the datagram in issues such as congestion. If a router is congested and needs to discard some datagrams, those datagrams with lowest precedence are discarded first. Some datagrams in the Internet are more important than others. For example, a datagram used for network management is much more urgent and important than a datagram containing optional information for a group.

The precedence subfield was part of version 4, but never used.

   b. **TOS bits** is a 4-bit subfield with each bit having a special meaning. Although a bit can be either 0 or 1, one and only one of the bits can have the value of 1 in each datagram. The bit patterns and their interpretations are given in Table 20.1. With only 1 bit set at a time, we can have five different types of services.

Table 20.1 *Types of service*

TOS Bits	*Description*
0000	Normal (default)
0001	Minimize cost
0010	Maximize reliability
0100	Maximize throughput
1000	Minimize delay

Application programs can request a specific type of service. The defaults for some applications are shown in Table 20.2.

Table 20.2 *Default types of service*

Protocol	*TOS Bits*	*Description*
ICMP	0000	Normal
BOOTP	0000	Normal
NNTP	0001	Minimize cost
IGP	0010	Maximize reliability
SNMP	0010	Maximize reliability
TELNET	1000	Minimize delay
FTP (data)	0100	Maximize throughput
FTP (control)	1000	Minimize delay
TFTP	1000	Minimize delay
SMTP (command)	1000	Minimize delay
SMTP (data)	0100	Maximize throughput
DNS (UDP query)	1000	Minimize delay
DNS (TCP query)	0000	Normal
DNS (zone)	0100	Maximize throughput

It is clear from Table 20.2 that interactive activities, activities requiring immediate attention, and activities requiring immediate response need minimum delay. Those activities that send bulk data require maximum throughput. Management activities need maximum reliability. Background activities need minimum cost.

2. **Differentiated Services**
In this interpretation, the first 6 bits make up the **codepoint** subfield, and the last 2 bits are not used. The codepoint subfield can be used in two different ways.
   a. When the 3 rightmost bits are 0s, the 3 leftmost bits are interpreted the same as the precedence bits in the service type interpretation. In other words, it is compatible with the old interpretation.

   b. When the 3 rightmost bits are not all 0s, the 6 bits define 64 services based on the priority assignment by the Internet or local authorities according to Table 20.3. The first category contains 32 service types; the second and the third each contain 16. The first category (numbers 0, 2, 4, . . . , 62) is assigned by the Internet authorities (IETF). The second category (3, 7, 11, 15, . . . , 63) can be used by local authorities (organizations). The third category (1, 5, 9, . . . , 61) is temporary and can be used for experimental purposes. Note that the numbers are not contiguous. If they were, the first category would range from 0 to 31, the second from 32 to 47, and the third from 48 to 63. This would be incompatible with the TOS interpretation because XXX000 (which includes 0, 8, 16, 24, 32, 40, 48, and 56) would fall into all three categories. Instead, in this assignment method all these services belong to category 1. Note that these assignments have not yet been finalized.

Table 20.3 *Values for codepoints*

Category	*Codepoint*	*Assigning Authority*
1	XXXXX0	Internet
2	XXXX11	Local
3	XXXX01	Temporary or experimental

❑ **Total length.** This is a 16-bit field that defines the total length (header plus data) of the IPv4 datagram in bytes. To find the length of the data coming from the upper layer, subtract the header length from the total length. The header length can be found by multiplying the value in the HLEN field by 4.

$$\text{Length of data} = \text{total length} - \text{header length}$$

Since the field length is 16 bits, the total length of the IPv4 datagram is limited to 65,535 ($2^{16} - 1$) bytes, of which 20 to 60 bytes are the header and the rest is data from the upper layer.

The total length field defines the total length of the datagram including the header.

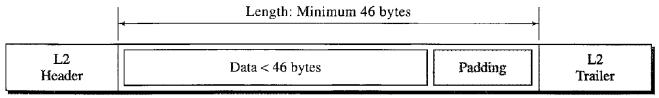
Though a size of 65,535 bytes might seem large, the size of the IPv4 datagram may increase in the near future as the underlying technologies allow even more throughput (greater bandwidth).

When we discuss fragmentation in the next section, we will see that some physical networks are not able to encapsulate a datagram of 65,535 bytes in their frames. The datagram must be fragmented to be able to pass through those networks.

One may ask why we need this field anyway. When a machine (router or host) receives a frame, it drops the header and the trailer, leaving the datagram. Why include an extra field that is not needed? The answer is that in many cases we really do not need the value in this field. However, there are occasions in which the

datagram is not the only thing encapsulated in a frame; it may be that padding has been added. For example, the Ethernet protocol has a minimum and maximum restriction on the size of data that can be encapsulated in a frame (46 to 1500 bytes). If the size of an IPv4 datagram is less than 46 bytes, some padding will be added to meet this requirement. In this case, when a machine decapsulates the datagram, it needs to check the total length field to determine how much is really data and how much is padding (see Figure 20.7).

Figure 20.7 *Encapsulation of a small datagram in an Ethernet frame*

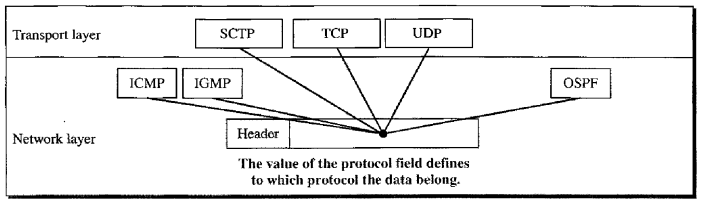


- **Identification.** This field is used in fragmentation (discussed in the next section).
- **Flags.** This field is used in fragmentation (discussed in the next section).
- **Fragmentation offset.** This field is used in fragmentation (discussed in the next section).
- **Time to live.** A datagram has a limited lifetime in its travel through an internet. This field was originally designed to hold a timestamp, which was decremented by each visited router. The datagram was discarded when the value became zero. However, for this scheme, all the machines must have synchronized clocks and must know how long it takes for a datagram to go from one machine to another. Today, this field is used mostly to control the maximum number of hops (routers) visited by the datagram. When a source host sends the datagram, it stores a number in this field. This value is approximately 2 times the maximum number of routes between any two hosts. Each router that processes the datagram decrements this number by 1. If this value, after being decremented, is zero, the router discards the datagram.

  This field is needed because routing tables in the Internet can become corrupted. A datagram may travel between two or more routers for a long time without ever getting delivered to the destination host. This field limits the lifetime of a datagram.

  Another use of this field is to intentionally limit the journey of the packet. For example, if the source wants to confine the packet to the local network, it can store 1 in this field. When the packet arrives at the first router, this value is decremented to 0, and the datagram is discarded.
- **Protocol.** This 8-bit field defines the higher-level protocol that uses the services of the IPv4 layer. An IPv4 datagram can encapsulate data from several higher-level protocols such as TCP, UDP, ICMP, and IGMP. This field specifies the final destination protocol to which the IPv4 datagram is delivered. In other words, since the IPv4 protocol carries data from different other protocols, the value of this field helps the receiving network layer know to which protocol the data belong (see Figure 20.8).

Figure 20.8 *Protocol field and encapsulated data*



The value of this field for each higher-level protocol is shown in Table 20.4.

Table 20.4 *Protocol values*

Value	*Protocol*
1	ICMP
2	IGMP
6	TCP
17	UDP
89	OSPF

- **Checksum.** The checksum concept and its calculation are discussed later in this chapter.
- **Source address.** This 32-bit field defines the IPv4 address of the source. This field must remain unchanged during the time the IPv4 datagram travels from the source host to the destination host.
- **Destination address.** This 32-bit field defines the IPv4 address of the destination. This field must remain unchanged during the time the IPv4 datagram travels from the source host to the destination host.

Example 20.1

An IPv4 packet has arrived with the first 8 bits as shown:

01000010

The receiver discards the packet. Why?

Solution

There is an error in this packet. The 4 leftmost bits (0100) show the version, which is correct. The next 4 bits (0010) show an invalid header length ($2 \times 4 = 8$). The minimum number of bytes in the header must be 20. The packet has been corrupted in transmission.

Example 20.2

In an IPv4 packet, the value of HLEN is 1000 in binary. How many bytes of options are being carried by this packet?

Solution

The HLEN value is 8, which means the total number of bytes in the header is 8×4, or 32 bytes. The first 20 bytes are the base header, the next 12 bytes are the options.

Example 20.3

In an IPv4 packet, the value of HLEN is 5, and the value of the total length field is 0x0028. How many bytes of data are being carried by this packet?

Solution

The HLEN value is 5, which means the total number of bytes in the header is 5×4, or 20 bytes (no options). The total length is 40 bytes, which means the packet is carrying 20 bytes of data (40 – 20).

Example 20.4

An IPv4 packet has arrived with the first few hexadecimal digits as shown.

0x45000028000100000102 . . .

How many hops can this packet travel before being dropped? The data belong to what upper-layer protocol?

Solution

To find the time-to-live field, we skip 8 bytes (16 hexadecimal digits). The time-to-live field is the ninth byte, which is 01. This means the packet can travel only one hop. The protocol field is the next byte (02), which means that the upper-layer protocol is IGMP (see Table 20.4).

Fragmentation

A datagram can travel through different networks. Each router decapsulates the IPv4 datagram from the frame it receives, processes it, and then encapsulates it in another frame. The format and size of the received frame depend on the protocol used by the physical network through which the frame has just traveled. The format and size of the sent frame depend on the protocol used by the physical network through which the frame is going to travel. For example, if a router connects a LAN to a WAN, it receives a frame in the LAN format and sends a frame in the WAN format.

Maximum Transfer Unit (MTU)

Each data link layer protocol has its own frame format in most protocols. One of the fields defined in the format is the maximum size of the data field. In other words, when a datagram is encapsulated in a frame, the total size of the datagram must be less than this maximum size, which is defined by the restrictions imposed by the hardware and software used in the network (see Figure 20.9).

The value of the MTU depends on the physical network protocol. Table 20.5 shows the values for some protocols.

Figure 20.9 *Maximum transfer unit (MTU)*

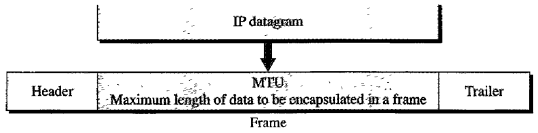


Table 20.5 *MTUs for some networks*

Protocol	*MTU*
Hyperchannel	65,535
Token Ring (16 Mbps)	17,914
Token Ring (4 Mbps)	4,464
FDDI	4,352
Ethernet	1,500
X.25	576
PPP	296

To make the IPv4 protocol independent of the physical network, the designers decided to make the maximum length of the IPv4 datagram equal to 65,535 bytes. This makes transmission more efficient if we use a protocol with an MTU of this size. However, for other physical networks, we must divide the datagram to make it possible to pass through these networks. This is called **fragmentation.**

The source usually does not fragment the IPv4 packet. The transport layer will instead segment the data into a size that can be accommodated by IPv4 and the data link layer in use.

When a datagram is fragmented, each fragment has its own header with most of the fields repeated, but with some changed. A fragmented datagram may itself be fragmented if it encounters a network with an even smaller MTU. In other words, a datagram can be fragmented several times before it reaches the final destination.

In IPv4, a datagram can be fragmented by the source host or any router in the path although there is a tendency to limit fragmentation only at the source. The reassembly of the datagram, however, is done only by the destination host because each fragment becomes an independent datagram. Whereas the fragmented datagram can travel through different routes, and we can never control or guarantee which route a fragmented datagram may take, all the fragments belonging to the same datagram should finally arrive at the destination host. So it is logical to do the reassembly at the final destination. An even stronger objection to reassembling packets during the transmission is the loss of efficiency it incurs.

When a datagram is fragmented, required parts of the header must be copied by all fragments. The option field may or may not be copied, as we will see in the next section. The host or router that fragments a datagram must change the values of three fields:

flags, fragmentation offset, and total length. The rest of the fields must be copied. Of course, the value of the checksum must be recalculated regardless of fragmentation.

Fields Related to Fragmentation

The fields that are related to fragmentation and reassembly of an IPv4 datagram are the identification, flags, and fragmentation offset fields.

- **Identification.** This 16-bit field identifies a datagram originating from the source host. The combination of the identification and source IPv4 address must uniquely define a datagram as it leaves the source host. To guarantee uniqueness, the IPv4 protocol uses a counter to label the datagrams. The counter is initialized to a positive number. When the IPv4 protocol sends a datagram, it copies the current value of the counter to the identification field and increments the counter by 1. As long as the counter is kept in the main memory, uniqueness is guaranteed. When a datagram is fragmented, the value in the identification field is copied to all fragments. In other words, all fragments have the same identification number, the same as the original datagram. The identification number helps the destination in reassembling the datagram. It knows that all fragments having the same identification value must be assembled into one datagram.
- **Flags.** This is a 3-bit field. The first bit is reserved. The second bit is called the *do not fragment* bit. If its value is 1, the machine must not fragment the datagram. If it cannot pass the datagram through any available physical network, it discards the datagram and sends an ICMP error message to the source host (see Chapter 21). If its value is 0, the datagram can be fragmented if necessary. The third bit is called the *more fragment* bit. If its value is 1, it means the datagram is not the last fragment; there are more fragments after this one. If its value is 0, it means this is the last or only fragment (see Figure 20.10).

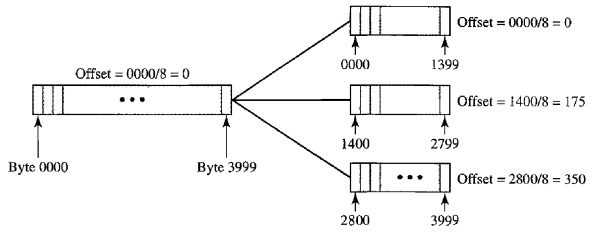
Figure 20.10 *Flags used in fragmentation*

- **Fragmentation offset.** This 13-bit field shows the relative position of this fragment with respect to the whole datagram. It is the offset of the data in the original datagram measured in units of 8 bytes. Figure 20.11 shows a datagram with a data size of 4000 bytes fragmented into three fragments.

  The bytes in the original datagram are numbered 0 to 3999. The first fragment carries bytes 0 to 1399. The offset for this datagram is 0/8 = 0. The second fragment carries bytes 1400 to 2799; the offset value for this fragment is 1400/8 = 175. Finally, the third fragment carries bytes 2800 to 3999. The offset value for this fragment is 2800/8 = 350.

  Remember that the value of the offset is measured in units of 8 bytes. This is done because the length of the offset field is only 13 bits and cannot represent a

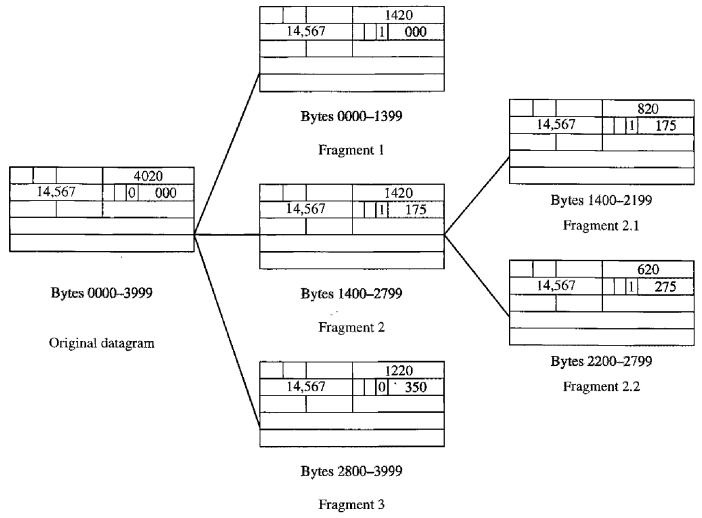
Figure 20.11 *Fragmentation example*



sequence of bytes greater than 8191. This forces hosts or routers that fragment datagrams to choose a fragment size so that the first byte number is divisible by 8.

Figure 20.12 shows an expanded view of the fragments in Figure 20.11. Notice the value of the identification field is the same in all fragments. Notice the value of the flags field with the *more* bit set for all fragments except the last. Also, the value of the offset field for each fragment is shown.

Figure 20.12 *Detailed fragmentation example*



The figure also shows what happens if a fragment itself is fragmented. In this case the value of the offset field is always relative to the original datagram. For example, in the figure, the second fragment is itself fragmented later to two fragments of 800 bytes and 600 bytes, but the offset shows the relative position of the fragments to the original data.

It is obvious that even if each fragment follows a different path and arrives out of order, the final destination host can reassemble the original datagram from the fragments received (if none of them is lost) by using the following strategy:

1. The first fragment has an offset field value of zero.
2. Divide the length of the first fragment by 8. The second fragment has an offset value equal to that result.
3. Divide the total length of the first and second fragments by 8. The third fragment has an offset value equal to that result.
4. Continue the process. The last fragment has a *more* bit value of 0.

Example 20.5

A packet has arrived with an *M* bit value of 0. Is this the first fragment, the last fragment, or a middle fragment? Do we know if the packet was fragmented?

Solution

If the *M* bit is 0, it means that there are no more fragments; the fragment is the last one. However, we cannot say if the original packet was fragmented or not. A nonfragmented packet is considered the last fragment.

Example 20.6

A packet has arrived with an *M* bit value of 1. Is this the first fragment, the last fragment, or a middle fragment? Do we know if the packet was fragmented?

Solution

If the *M* bit is 1, it means that there is at least one more fragment. This fragment can be the first one or a middle one, but not the last one. We don't know if it is the first one or a middle one; we need more information (the value of the fragmentation offset). See Example 20.7.

Example 20.7

A packet has arrived with an *M* bit value of 1 and a fragmentation offset value of 0. Is this the first fragment, the last fragment, or a middle fragment?

Solution

Because the *M* bit is 1, it is either the first fragment or a middle one. Because the offset value is 0, it is the first fragment.

Example 20.8

A packet has arrived in which the offset value is 100. What is the number of the first byte? Do we know the number of the last byte?

Solution

To find the number of the first byte, we multiply the offset value by 8. This means that the first byte number is 800. We cannot determine the number of the last byte unless we know the length of the data.

Example 20.9

A packet has arrived in which the offset value is 100, the value of HLEN is 5, and the value of the total length field is 100. What are the numbers of the first byte and the last byte?

Solution

The first byte number is $100 \times 8 = 800$. The total length is 100 bytes, and the header length is 20 bytes (5×4), which means that there are 80 bytes in this datagram. If the first byte number is 800, the last byte number must be 879.

Checksum

We discussed the general idea behind the checksum and how it is calculated in Chapter 10. The implementation of the checksum in the IPv4 packet follows the same principles. First, the value of the checksum field is set to 0. Then the entire header is divided into 16-bit sections and added together. The result (sum) is complemented and inserted into the checksum field.

The checksum in the IPv4 packet covers only the header, not the data. There are two good reasons for this. First, all higher-level protocols that encapsulate data in the IPv4 datagram have a checksum field that covers the whole packet. Therefore, the checksum for the IPv4 datagram does not have to check the encapsulated data. Second, the header of the IPv4 packet changes with each visited router, but the data do not. So the checksum includes only the part that has changed. If the data were included, each router must recalculate the checksum for the whole packet, which means an increase in processing time.

Example 20.10

Figure 20.13 shows an example of a checksum calculation for an IPv4 header without options. The header is divided into 16-bit sections. All the sections are added and the sum is complemented. The result is inserted in the checksum field.

Options

The header of the IPv4 datagram is made of two parts: a fixed part and a variable part. The fixed part is 20 bytes long and was discussed in the previous section. The variable part comprises the options that can be a maximum of 40 bytes.

Options, as the name implies, are not required for a datagram. They can be used for network testing and debugging. Although options are not a required part of the IPv4 header, option processing is required of the IPv4 software. This means that all implementations must be able to handle options if they are present in the header.

The detailed discussion of each option is beyond the scope of this book. We give the taxonomy of options in Figure 20.14 and briefly explain the purpose of each.

No Operation

A **no-operation option** is a 1-byte option used as a filler between options.

Figure 20.13 *Example of checksum calculation in IPv4*

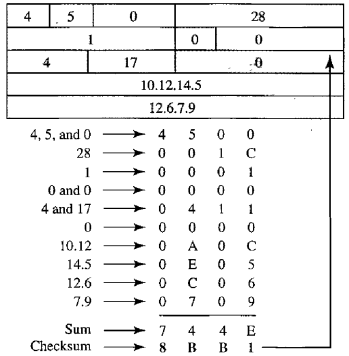
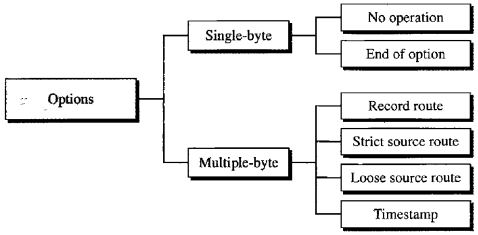


Figure 20.14 *Taxonomy of options in IPv4*



End of Option

An **end-of-option option** is a 1-byte option used for padding at the end of the option field. It, however, can only be used as the last option.

Record Route

A **record route option** is used to record the Internet routers that handle the datagram. It can list up to nine router addresses. It can be used for debugging and management purposes.

Strict Source Route

A **strict source route option** is used by the source to predetermine a route for the datagram as it travels through the Internet. Dictation of a route by the source can be useful

for several purposes. The sender can choose a route with a specific type of service, such as minimum delay or maximum throughput. Alternatively, it may choose a route that is safer or more reliable for the sender's purpose. For example, a sender can choose a route so that its datagram does not travel through a competitor's network.

If a datagram specifies a strict source route, all the routers defined in the option must be visited by the datagram. A router must not be visited if its IPv4 address is not listed in the datagram. If the datagram visits a router that is not on the list, the datagram is discarded and an error message is issued. If the datagram arrives at the destination and some of the entries were not visited, it will also be discarded and an error message issued.

Loose Source Route

A **loose source route option** is similar to the strict source route, but it is less rigid. Each router in the list must be visited, but the datagram can visit other routers as well.

Timestamp

A **timestamp option** is used to record the time of datagram processing by a router. The time is expressed in milliseconds from midnight, Universal time or Greenwich mean time. Knowing the time a datagram is processed can help users and managers track the behavior of the routers in the Internet. We can estimate the time it takes for a datagram to go from one router to another. We say *estimate* because, although all routers may use Universal time, their local clocks may not be synchronized.

20.3 IPv6

The network layer protocol in the TCP/IP protocol suite is currently IPv4 (Internetworking Protocol, version 4). IPv4 provides the host-to-host communication between systems in the Internet. Although IPv4 is well designed, data communication has evolved since the inception of IPv4 in the 1970s. IPv4 has some deficiencies (listed below) that make it unsuitable for the fast-growing Internet.

- Despite all short-term solutions, such as subnetting, classless addressing, and NAT, address depletion is still a long-term problem in the Internet.
- The Internet must accommodate real-time audio and video transmission. This type of transmission requires minimum delay strategies and reservation of resources not provided in the IPv4 design.
- The Internet must accommodate encryption and authentication of data for some applications. No encryption or authentication is provided by IPv4.

To overcome these deficiencies, **IPv6 (Internetworking Protocol, version 6),** also known as **IPng (Internetworking Protocol, next generation),** was proposed and is now a standard. In IPv6, the Internet protocol was extensively modified to accommodate the unforeseen growth of the Internet. The format and the length of the IP address were changed along with the packet format. Related protocols, such as ICMP, were also modified. Other protocols in the network layer, such as ARP, RARP, and IGMP, were

either deleted or included in the ICMPv6 protocol (see Chapter 21). Routing protocols, such as RIP and OSPF (see Chapter 22), were also slightly modified to accommodate these changes. Communications experts predict that IPv6 and its related protocols will soon replace the current IP version. In this section first we discuss IPv6. Then we explore the strategies used for the transition from version 4 to version 6.

The adoption of IPv6 has been slow. The reason is that the original motivation for its development, depletion of IPv4 addresses, has been remedied by short-term strategies such as classless addressing and NAT. However, the fast-spreading use of the Internet, and new services such as mobile IP, IP telephony, and IP-capable mobile telephony, may eventually require the total replacement of IPv4 with IPv6.

Advantages

The next-generation IP, or IPv6, has some advantages over IPv4 that can be summarized as follows:

- **Larger address space.** An IPv6 address is 128 bits long, as we discussed in Chapter 19. Compared with the 32-bit address of IPv4, this is a huge (2^{96}) increase in the address space.
- **Better header format.** IPv6 uses a new header format in which options are separated from the base header and inserted, when needed, between the base header and the upper-layer data. This simplifies and speeds up the routing process because most of the options do not need to be checked by routers.
- **New options.** IPv6 has new options to allow for additional functionalities.
- **Allowance for extension.** IPv6 is designed to allow the extension of the protocol if required by new technologies or applications.
- **Support for resource allocation.** In IPv6, the type-of-service field has been removed, but a mechanism (called *flow label*) has been added to enable the source to request special handling of the packet. This mechanism can be used to support traffic such as real-time audio and video.
- **Support for more security.** The encryption and authentication options in IPv6 provide confidentiality and integrity of the packet.

Packet Format

The IPv6 packet is shown in Figure 20.15. Each packet is composed of a mandatory base header followed by the payload. The payload consists of two parts: optional extension headers and data from an upper layer. The base header occupies 40 bytes, whereas the extension headers and data from the upper layer contain up to 65,535 bytes of information.

Base Header

Figure 20.16 shows the **base header** with its eight fields.

These fields are as follows:

- **Version.** This 4-bit field defines the version number of the IP. For IPv6, the value is 6.
- **Priority.** The 4-bit priority field defines the priority of the packet with respect to traffic congestion. We will discuss this field later.

Figure 20.15 *IPv6 datagram header and payload*

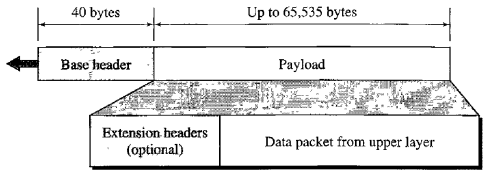
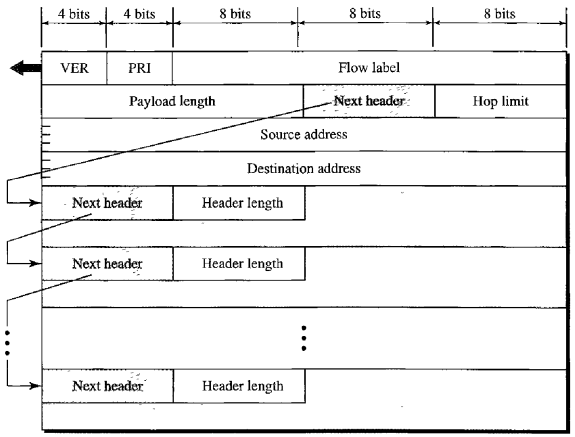


Figure 20.16 *Format of an IPv6 datagram*



- **Flow label.** The **flow label** is a 3-byte (24-bit) field that is designed to provide special handling for a particular flow of data. We will discuss this field later.
- **Payload length.** The 2-byte payload length field defines the length of the IP datagram excluding the base header.
- **Next header.** The **next header** is an 8-bit field defining the header that follows the base header in the datagram. The next header is either one of the optional extension headers used by IP or the header of an encapsulated packet such as UDP or TCP. Each extension header also contains this field. Table 20.6 shows the values of next headers. Note that this field in version 4 is called the *protocol*.
- **Hop limit.** This 8-bit **hop limit** field serves the same purpose as the TTL field in IPv4.
- **Source address.** The source address field is a 16-byte (128-bit) Internet address that identifies the original source of the datagram.

Table 20.6 *Next header codes for IPv6*

Code	*Next Header*
0	Hop-by-hop option
2	ICMP
6	TCP
17	UDP
43	Source routing
44	Fragmentation
50	Encrypted security payload
51	Authentication
59	Null (no next header)
60	Destination option

- **Destination address.** The destination address field is a 16-byte (128-bit) Internet address that usually identifies the final destination of the datagram. However, if source routing is used, this field contains the address of the next router.

Priority

The priority field of the IPv6 packet defines the priority of each packet with respect to other packets from the same source. For example, if one of two consecutive datagrams must be discarded due to congestion, the datagram with the lower **packet priority** will be discarded. IPv6 divides traffic into two broad categories: congestion-controlled and noncongestion-controlled.

Congestion-Controlled Traffic If a source adapts itself to traffic slowdown when there is congestion, the traffic is referred to as **congestion-controlled traffic.** For example, TCP, which uses the sliding window protocol, can easily respond to traffic. In congestion-controlled traffic, it is understood that packets may arrive delayed, lost, or out of order. Congestion-controlled data are assigned priorities from 0 to 7, as listed in Table 20.7. A priority of 0 is the lowest; a priority of 7 is the highest.

Table 20.7 *Priorities for congestion-controlled traffic*

Priority	*Meaning*
0	No specific traffic
1	Background data
2	Unattended data traffic
3	Reserved
4	Attended bulk data traffic
5	Reserved
6	Interactive traffic
7	Control traffic

The priority descriptions are as follows:

- **No specific traffic.** A priority of 0 is assigned to a packet when the process does not define a priority.
- **Background data.** This group (priority 1) defines data that are usually delivered in the background. Delivery of the news is a good example.
- **Unattended data traffic.** If the user is not waiting (attending) for the data to be received, the packet will be given a priority of 2. E-mail belongs to this group. The recipient of an e-mail does not know when a message has arrived. In addition, an e-mail is usually stored before it is forwarded. A little bit of delay is of little consequence.
- **Attended bulk data traffic.** A protocol that transfers data while the user is waiting (attending) to receive the data (possibly with delay) is given a priority of 4. FTP and HTTP belong to this group.
- **Interactive traffic.** Protocols such as TELNET that need user interaction are assigned the second-highest priority (6) in this group.
- **Control traffic.** Control traffic is given the highest priority (7). Routing protocols such as OSPF and RIP and management protocols such as SNMP have this priority.

Noncongestion-Controlled Traffic This refers to a type of traffic that expects minimum delay. Discarding of packets is not desirable. Retransmission in most cases is impossible. In other words, the source does not adapt itself to congestion. Real-time audio and video are examples of this type of traffic.

Priority numbers from 8 to 15 are assigned to **noncongestion-controlled traffic.** Although there are not yet any particular standard assignments for this type of data, the priorities are usually based on how much the quality of received data is affected by the discarding of packets. Data containing less redundancy (such as low-fidelity audio or video) can be given a higher priority (15). Data containing more redundancy (such as high-fidelity audio or video) are given a lower priority (8). See Table 20.8.

Table 20.8 *Priorities for noncongestion-controlled traffic*

Priority	*Meaning*
8	Data with greatest redundancy
. . .	. . .
15	Data with least redundancy

Flow Label

A sequence of packets, sent from a particular source to a particular destination, that needs special handling by routers is called a *flow* of packets. The combination of the source address and the value of the *flow label* uniquely defines a flow of packets.

To a router, a flow is a sequence of packets that share the same characteristics, such as traveling the same path, using the same resources, having the same kind of security, and so on. A router that supports the handling of flow labels has a flow label table. The table has an entry for each active flow label; each entry defines the services required by

the corresponding flow label. When the router receives a packet, it consults its flow label table to find the corresponding entry for the flow label value defined in the packet. It then provides the packet with the services mentioned in the entry. However, note that the flow label itself does not provide the information for the entries of the flow label table; the information is provided by other means such as the hop-by-hop options or other protocols.

In its simplest form, a flow label can be used to speed up the processing of a packet by a router. When a router receives a packet, instead of consulting the routing table and going through a routing algorithm to define the address of the next hop, it can easily look in a flow label table for the next hop.

In its more sophisticated form, a flow label can be used to support the transmission of real-time audio and video. Real-time audio or video, particularly in digital form, requires resources such as high bandwidth, large buffers, long processing time, and so on. A process can make a reservation for these resources beforehand to guarantee that real-time data will not be delayed due to a lack of resources. The use of real-time data and the reservation of these resources require other protocols such as Real-Time Protocol (RTP) and Resource Reservation Protocol (RSVP) in addition to IPv6.

To allow the effective use of flow labels, three rules have been defined:

1. The flow label is assigned to a packet by the source host. The label is a random number between 1 and $2^{24} - 1$. A source must not reuse a flow label for a new flow while the existing flow is still active.
2. If a host does not support the flow label, it sets this field to zero. If a router does not support the flow label, it simply ignores it.
3. All packets belonging to the same flow have the same source, same destination, same priority, and same options.

Comparison Between IPv4 and IPv6 Headers

Table 20.9 compares IPv4 and IPv6 headers.

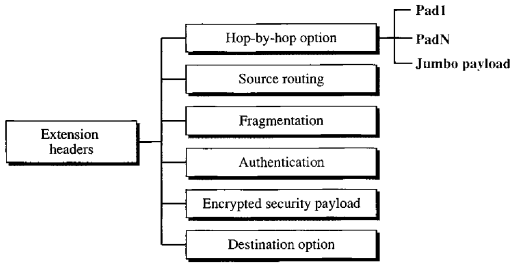
Table 20.9 *Comparison between IPv4 and IPv6 packet headers*

Comparison
1. The header length field is eliminated in IPv6 because the length of the header is fixed in this version.
2. The service type field is eliminated in IPv6. The priority and flow label fields together take over the function of the service type field.
3. The total length field is eliminated in IPv6 and replaced by the payload length field.
4. The identification, flag, and offset fields are eliminated from the base header in IPv6. They are included in the fragmentation extension header.
5. The TTL field is called hop limit in IPv6.
6. The protocol field is replaced by the next header field.
7. The header checksum is eliminated because the checksum is provided by upper-layer protocols; it is therefore not needed at this level.
8. The option fields in IPv4 are implemented as extension headers in IPv6.

Extension Headers

The length of the base header is fixed at 40 bytes. However, to give greater functionality to the IP datagram, the base header can be followed by up to six **extension headers.** Many of these headers are options in IPv4. Six types of extension headers have been defined, as shown in Figure 20.17.

Figure 20.17 *Extension header types*



Hop-by-Hop Option

The **hop-by-hop option** is used when the source needs to pass information to all routers visited by the datagram. So far, only three options have been defined: **Pad1, PadN,** and **jumbo payload.** The Pad1 option is 1 byte long and is designed for alignment purposes. PadN is similar in concept to Pad1. The difference is that PadN is used when 2 or more bytes is needed for alignment. The jumbo payload option is used to define a payload longer than 65,535 bytes.

Source Routing The source routing extension header combines the concepts of the strict source route and the loose source route options of IPv4.

Fragmentation

The concept of **fragmentation** is the same as that in IPv4. However, the place where fragmentation occurs differs. In IPv4, the source or a router is required to fragment if the size of the datagram is larger than the MTU of the network over which the datagram travels. In IPv6, only the original source can fragment. A source must use a **path MTU discovery technique** to find the smallest MTU supported by any network on the path. The source then fragments using this knowledge.

Authentication

The **authentication** extension header has a dual purpose: it validates the message sender and ensures the integrity of data. We discuss this extension header when we discuss network security in Chapter 31.

Encrypted Security Payload

The **encrypted security payload (ESP)** is an extension that provides confidentiality and guards against eavesdropping. We discuss this extension header in Chapter 31.

Destination Option The **destination option** is used when the source needs to pass information to the destination only. Intermediate routers are not permitted access to this information.

Comparison Between IPv4 Options and IPv6 Extension Headers

Table 20.10 compares the options in IPv4 with the extension headers in IPv6.

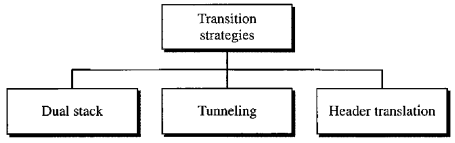
Table 20.10 *Comparison between IPv4 options and IPv6 extension headers*

Comparison
1. The no-operation and end-of-option options in IPv4 are replaced by Pad1 and PadN options in IPv6.
2. The record route option is not implemented in IPv6 because it was not used.
3. The timestamp option is not implemented because it was not used.
4. The source route option is called the source route extension header in IPv6.
5. The fragmentation fields in the base header section of IPv4 have moved to the fragmentation extension header in IPv6.
6. The authentication extension header is new in IPv6.
7. The encrypted security payload extension header is new in IPv6.

20.4 TRANSITION FROM IPv4 TO IPv6

Because of the huge number of systems on the Internet, the transition from IPv4 to IPv6 cannot happen suddenly. It takes a considerable amount of time before every system in the Internet can move from IPv4 to IPv6. The transition must be smooth to prevent any problems between IPv4 and IPv6 systems. Three strategies have been devised by the IETF to help the transition (see Figure 20.18).

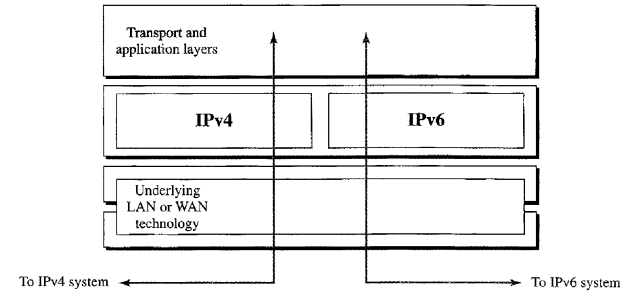
Figure 20.18 *Three transition strategies*



Dual Stack

It is recommended that all hosts, before migrating completely to version 6, have a **dual stack** of protocols. In other words, a station must run IPv4 and IPv6 simultaneously until all the Internet uses IPv6. See Figure 20.19 for the layout of a dual-stack configuration.

Figure 20.19 *Dual stack*



To determine which version to use when sending a packet to a destination, the source host queries the DNS. If the DNS returns an IPv4 address, the source host sends an IPv4 packet. If the DNS returns an IPv6 address, the source host sends an IPv6 packet.

Tunneling

Tunneling is a strategy used when two computers using IPv6 want to communicate with each other and the packet must pass through a region that uses IPv4. To pass through this region, the packet must have an IPv4 address. So the IPv6 packet is encapsulated in an IPv4 packet when it enters the region, and it leaves its capsule when it exits the region. It seems as if the IPv6 packet goes through a tunnel at one end and emerges at the other end. To make it clear that the IPv4 packet is carrying an IPv6 packet as data, the protocol value is set to 41. Tunneling is shown in Figure 20.20.

Figure 20.20 *Tunneling strategy*