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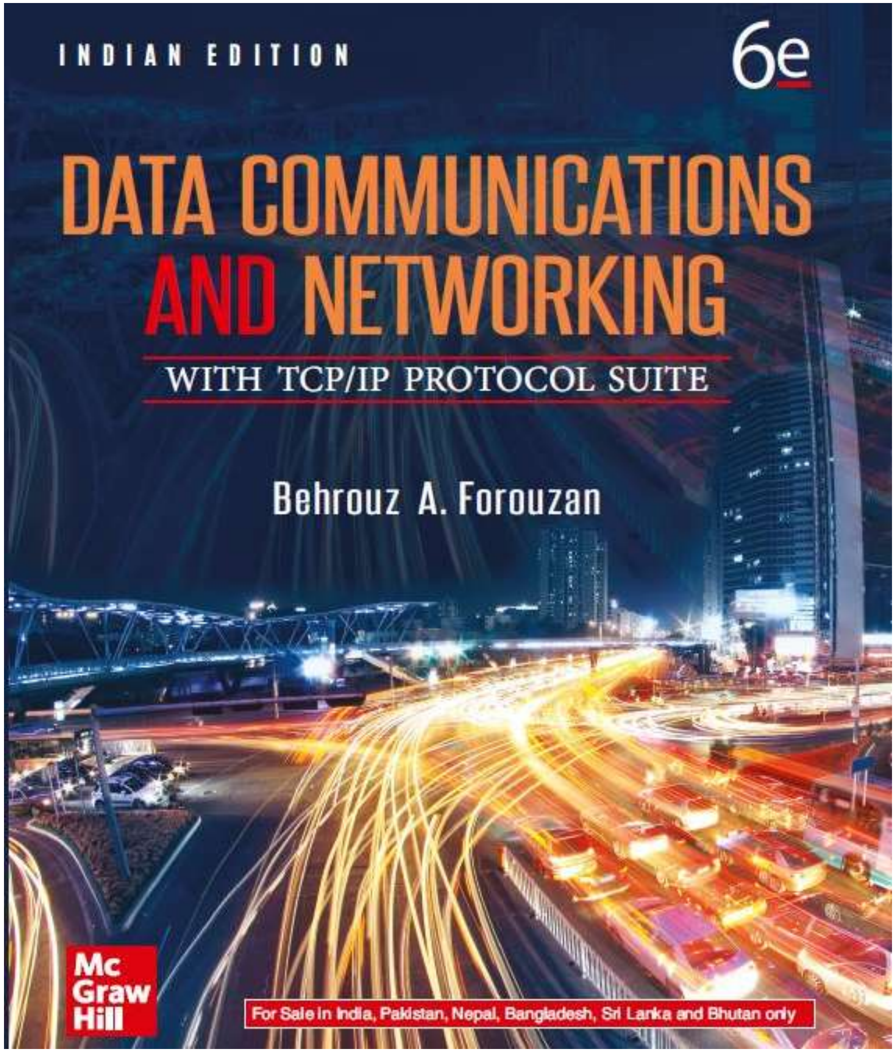
Chapter 07

Network Layer: Data Transfer

Data Communications and Networking, With TCP/IP protocol suite
Sixth Edition
Behrouz A. Forouzan



Chapter 7: Outline

7.1 Services

7.2 Packet Switching

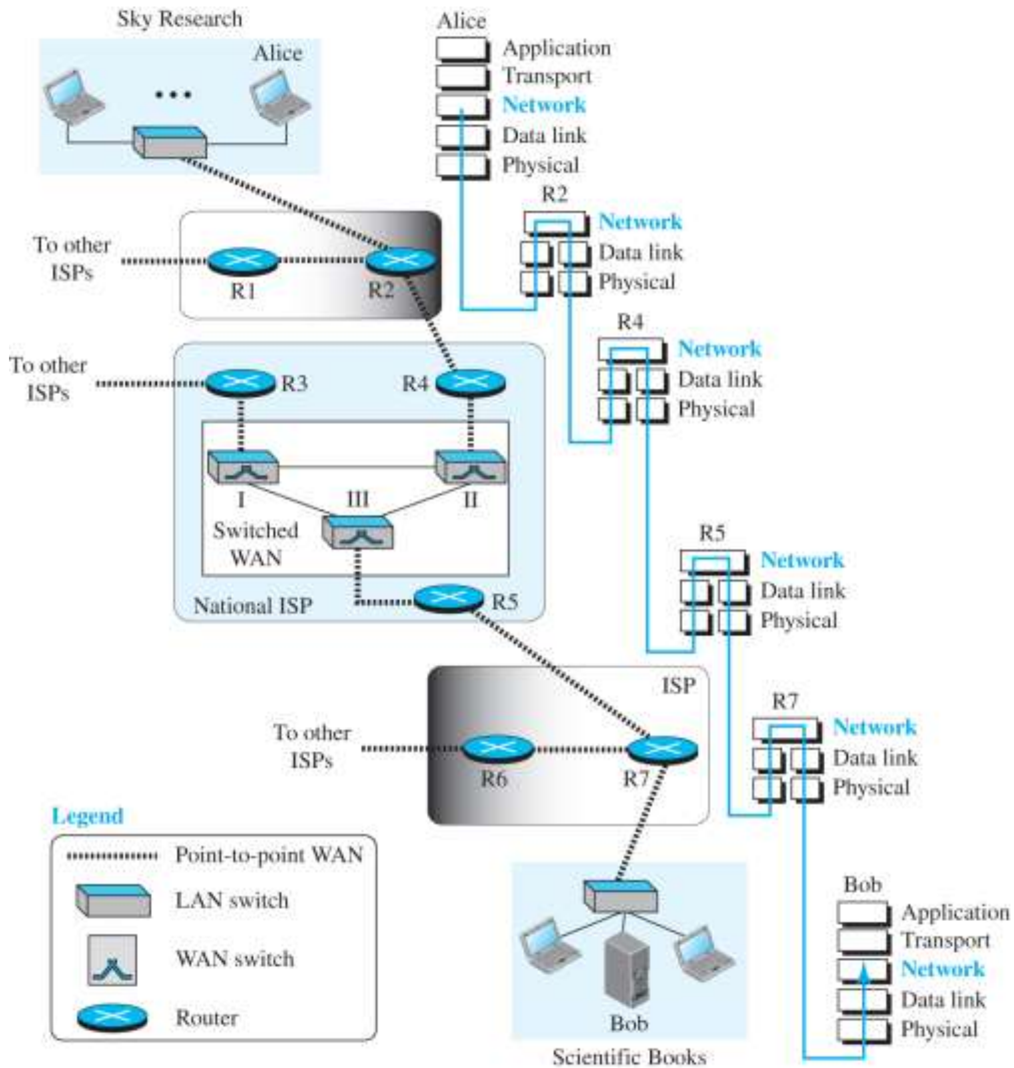
7.3 Performance

7.4 Internet Protocol V4

7.5 Internet Protocol V6

7.6 Transition from V4 To V6

Figure 7.1 Communication at the network layer



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7-1 SERVICES

We briefly discuss the services provided at the network layer.

7.1.1 Packetizing

The first duty of the network layer is definitely packetizing: encapsulating the payload in a network-layer packet at the source and decapsulating the payload from the network-layer packet at the destination. In other words, one duty of the network layer is to carry a payload from the source to the destination without changing it or using it. The network layer is doing the service of a carrier such as the postal office, which is responsible for delivery of packages from a sender to a receiver without changing or using the contents.

7.1.2 Routing

Other duties of the network layer, which are as important as the first, are routing and forwarding, which are directly related to each other.

7.1.3 Error Control

Although error control can be implemented in the network layer, the designers of the network layer in the Internet ignored this issue for the data being carried by the network layer. One reason for this decision is the fact that the packet in the network layer may be fragmented at each router, which makes error checking at this layer inefficient.

7.1.4 Flow Control

Flow control regulates the amount of data a source can send without overwhelming the receiver. If the upper layer at the source computer produces data faster than the upper layer at the destination computer can consume it, the receiver will be overwhelmed with data. To control the flow of data, the receiver needs to send some feedback to the sender to inform the latter that it is overwhelmed with data.

7.1.5 Congestion Control

Another issue in a network-layer protocol is congestion control. Congestion in the network layer is a situation in which too many datagrams are present in an area of the Internet. Congestion may occur if the number of datagrams sent by source computers is beyond the capacity of the network or routers.

7.1.6 Quality of Service

As the Internet has allowed new applications such as multimedia communication (in particular real-time communication of audio and video), the quality of service (QoS) of the communication has become more and more important. The Internet has thrived by providing better quality of service to support these applications. However, to keep the network layer untouched, these provisions are mostly implemented in the upper layer.

7.1.7 Security

Another issue related to communication at the network layer is security. Security was not a concern when the Internet was originally designed because it was used by a small number of users at universities for research activities; other people had no access to the Internet. The network layer was designed with no security provision. Today, however, security is a big concern. To provide security for a connectionless network layer, we need to have another virtual level that changes the connectionless service to a connection-oriented service.

7-2 PACKET SWITCHING

From the discussion of routing and forwarding in the previous section, we infer that a kind of switching occurs at the network layer. A router, in fact, is a switch that creates a connection between an input port and an output port (or a set of output ports), just as an electrical switch connects the input to the output to let electricity flow.

7.2.1 Datagram Approach

When the Internet started, to make it simple, the network layer was designed to provide a connectionless service in which the network-layer protocol treats each packet independently, with each packet having no relationship to any other packet. The idea was that the network layer is only responsible for delivery of packets from the source to the destination. In this approach, the packets in a message may or may not travel the same path to their destination.

7.2.2 Virtual-Circuit Approach

In a connection-oriented service (also called virtual-circuit approach), there is a relationship between all packets belonging to a message. Before all datagrams in a message can be sent, a virtual connection should be set up to define the path for the datagrams. After connection setup, the datagrams can all follow the same path. In this type of service, not only must the packet contain the source and destination addresses, it must also contain a flow label, a virtual circuit identifier that defines the virtual path the packet should follow.

7-3 PERFORMANCE

The upper-layer protocols that use the service of the network layer expect to receive an ideal service, but the network layer is not perfect. The performance of a network can be measured in terms of delay, throughput, and packet loss. Congestion control is an issue that can improve the performance.

7.3.1 Delay

All of us expect instantaneous response from a network, but a packet, from its source to its destination, encounters delays. The delays in a network can be divided into four types: transmission delay, propagation delay, processing delay, and queuing delay. Let us first discuss each of these delay types and then show how to calculate a packet delay from the source to the destination.

Transmission Delay 1

A source host or a router cannot send a packet instantaneously. A sender needs to put the bits in a packet on the line one by one. If the first bit of the packet is put on the line at time t1 and the last bit is put on the line at time t2, transmission delay of the packet is (t2 - t1). Definitely, the transmission delay is longer for a longer packet and shorter if the sender can transmit faster. In other words, the transmission delay is .

$$\text{Delay}_{\text{tr}} = \left(\text{Packet length}\right) / \left(\text{Transmission rate}\right)$$

Propagation Delay

Propagation delay is the time it takes for a bit to travel from point A to point B in the transmission media. The propagation delay for a packet-switched network depends on the propagation delay of each network (LAN or WAN). The propagation delay depends on the propagation speed of the media, which is 3 ´ 108 meters/second in a vacuum and normally much less in a wired medium; it also depends on the distance of the link. In other words, propagation delay is .

$$\text{Delay}_{\text{pg}} = (\text{Distance})/(\text{Propagation speed})$$

Processing Delay

The processing delay is the time required for a router or a destination host to receive a packet from its input port, remove the header, perform an error detection procedure, and deliver the packet to the output port (in the case of a router) or deliver the packet to the upper-layer protocol (in the case of the destination host). The processing delay may be different for each packet, but normally is calculated as an average.

$$\text{Delay}_{\text{pr}} = \text{Time required to process a packet}$$

Total Delay

Assuming equal delays for the sender, routers, and receiver, the total delay (source-to-destination delay) a packet encounters can be calculated if we know the number of routers, n, in the whole path.

$$\text{Delay}_{qu} = \text{The time a packet waits in queues}$$

Transmission Delay 2

A source host or a router cannot send a packet instantaneously. A sender needs to put the bits in a packet on the line one by one. If the first bit of the packet is put on the line at time t1 and the last bit is put on the line at time t2, transmission delay of the packet is (t2 - t1). Definitely, the transmission delay is longer for a longer packet and shorter if the sender can transmit faster. In other words, the transmission delay is:

$$\text{Delay}_{\text{tr}} = (\text{Packet length})/(\text{Transmission rate})$$

7.3.2 Throughput

Throughput at any point in a network is defined as the number of bits passing through the point in a second, which is actually the transmission rate of data at that point. In a path from source to destination, a packet may pass through several links (networks), each with a different transmission rate. How, then, can we determine the throughput of the whole path? To see the situation, assume that we have three links, each with a different transmission rate, as shown in Figure 7.2.

Figure 7.2 Throughput in a path with three links in a series

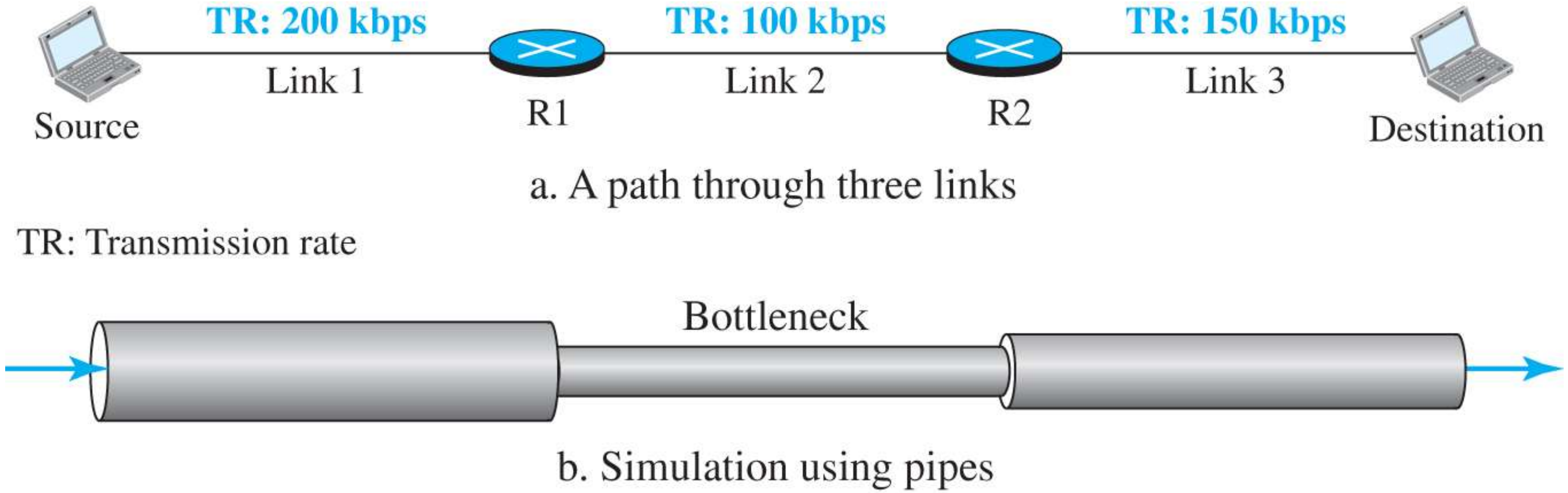

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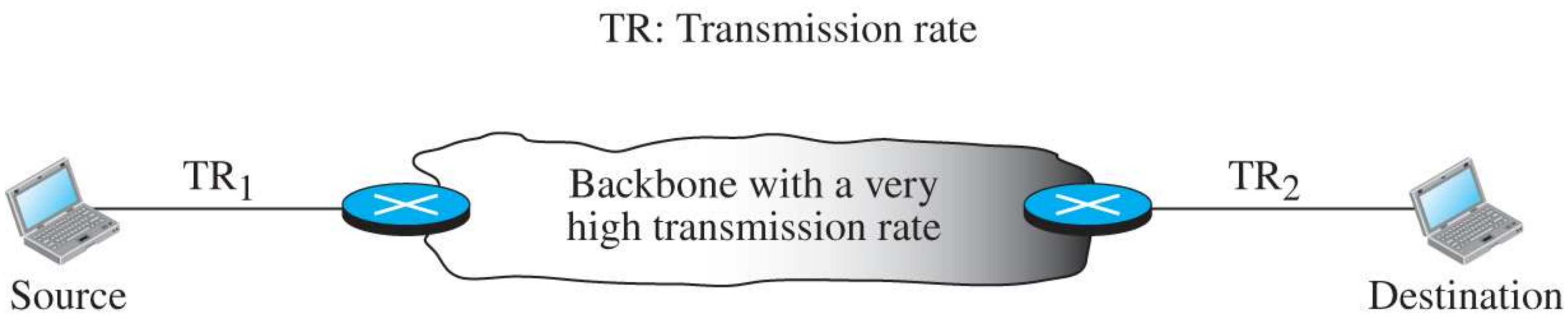
Figure 7.3 A path through the Internet backbone



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7.3.3 Packet Loss

Another issue that severely affects the performance of communication is the number of packets lost during transmission. When a router receives a packet while processing another packet, the received packet needs to be stored in the input buffer waiting for its turn. A router, however, has an input buffer with a limited size. A time may come when the buffer is full and the next packet needs to be dropped. The effect of packet loss on the Internet network layer is that the packet needs to be resent, which in turn may create overflow and cause more packet loss.

7.3.4 Congestion Control

Congestion control is a mechanism for improving performance. Although congestion at the network layer is not explicitly addressed in the Internet model, the study of congestion at this layer may help us to better understand the cause of congestion at the transport layer and find possible remedies to be used at the network layer. Congestion at the network layer is related to two issues, throughput and delay, which we discussed in the previous section.

7-4 INTERNET PROTOCOL VERSION 4

The network layer in the Internet has gone through several versions, but only two versions have survived: IP Version 4 (IPv4) and IP Version 6 (IPv6). Although IPv4 is almost depleted, we discuss it because there are still some areas that use this version and also because it is the foundation for IPv6.

7.4.1 IPv4 Addressing

The identifier used in the IP layer of the TCP/IP protocol suite to identify the connection of each device to the Internet is called the Internet address or IP address. An IPv4 address is a 32-bit address that uniquely and universally defines the connection of a host or a router to the Internet. The IP address is the address of the connection, not the host or the router, because if the device is moved to another network, the IP address may be changed.

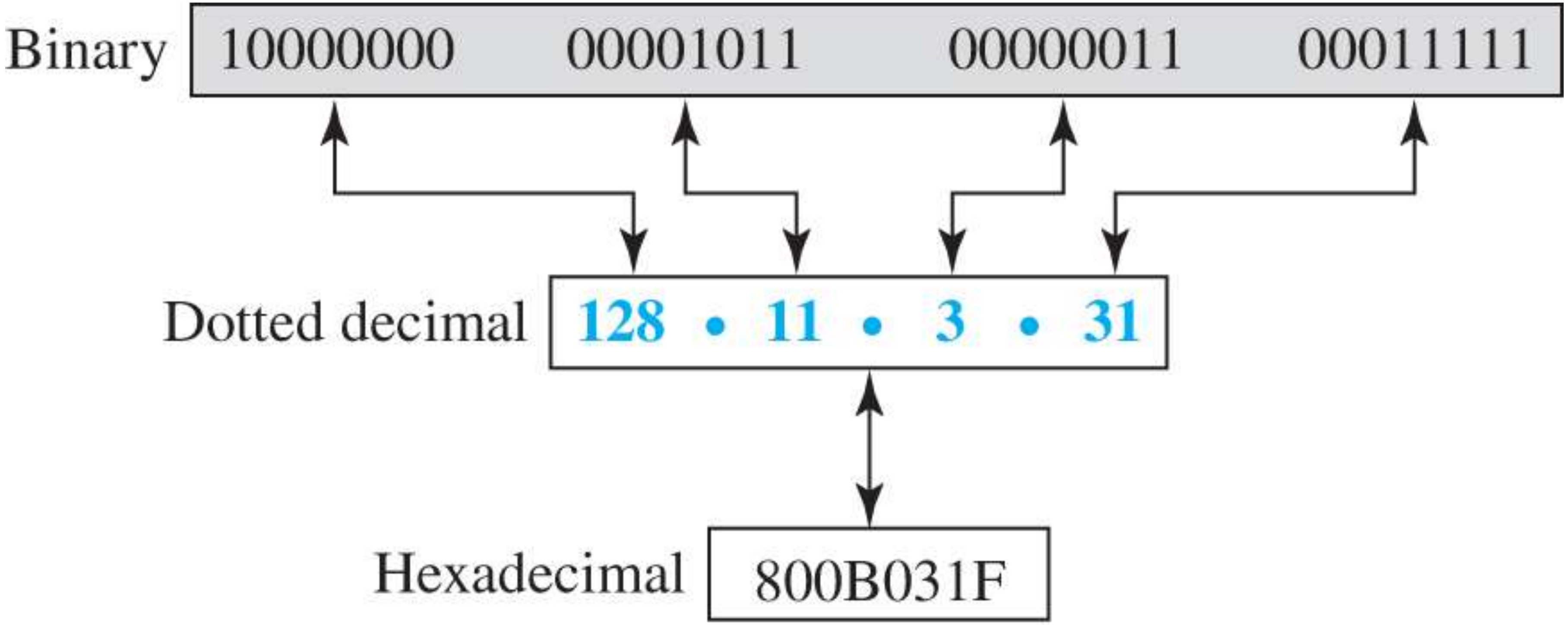
Address Space *1*

A protocol like IPv4 that defines addresses has an address space. An address space is the total number of addresses used by the protocol. If a protocol uses b bits to define an address, the address space is 2b because each bit can have two different values (0 or 1). IPv4 uses 32-bit addresses, which means that the address space is 232 or 4,294,967,296 (more than four billion). If there were no restrictions, more than 4 billion devices could be connected to the Internet.

Notation

There are three common notations to show an IPv4 address: binary notation (base 2), dotted-decimal notation (base 256), and hexadecimal notation (base 16).

Figure 7.4 Three different notations in IPv4 addressing



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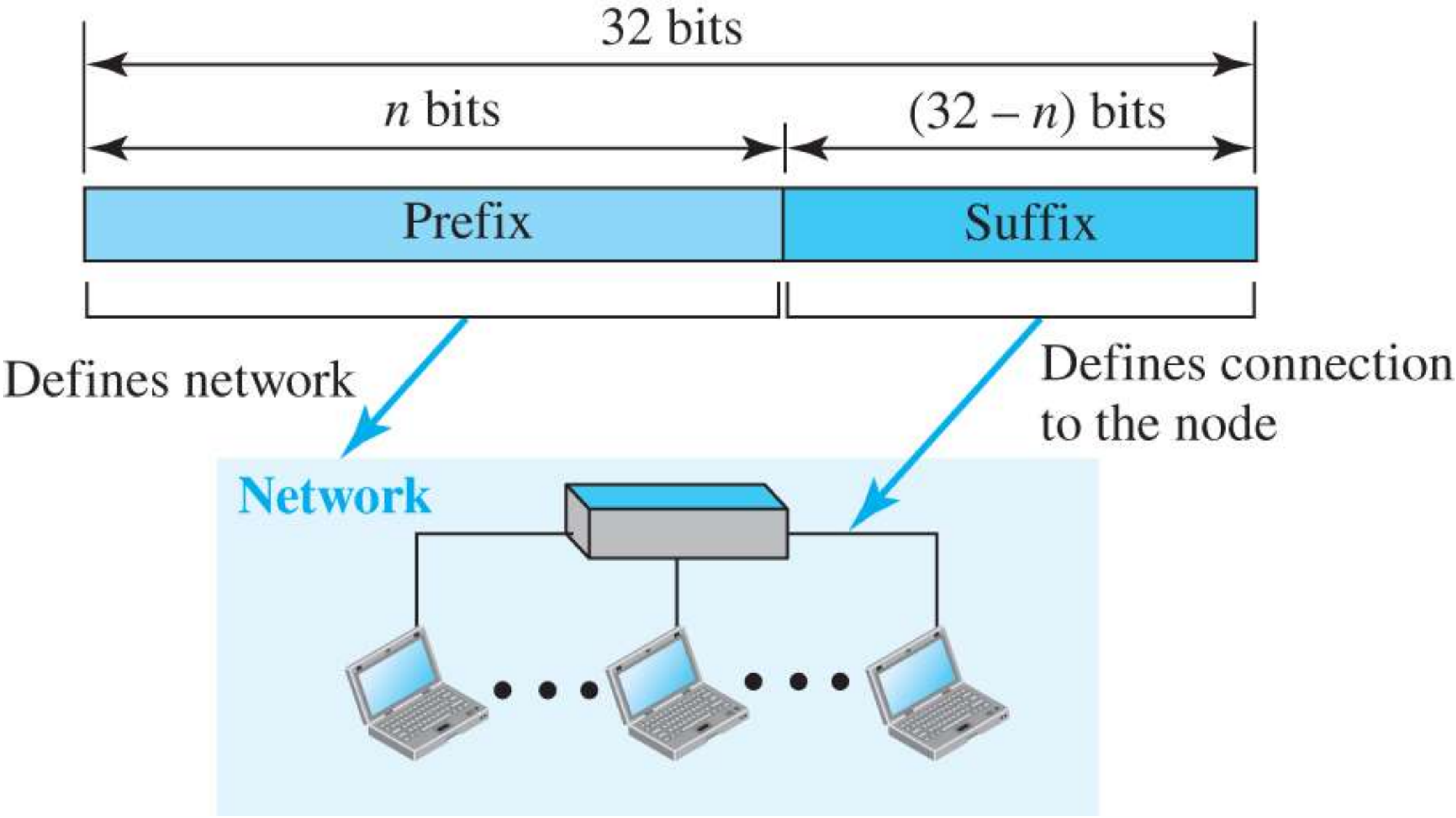
Hierarchy in Addressing

In any communication network that involves delivery, such as a telephone network or a postal network, the addressing system is hierarchical. A 32-bit IPv4 address is also hierarchical but divided only into two parts. The first part of the address, called the prefix, defines the network; the second part of the address, called the suffix, defines the node.

Figure 7.5 Hierarchy in addressing



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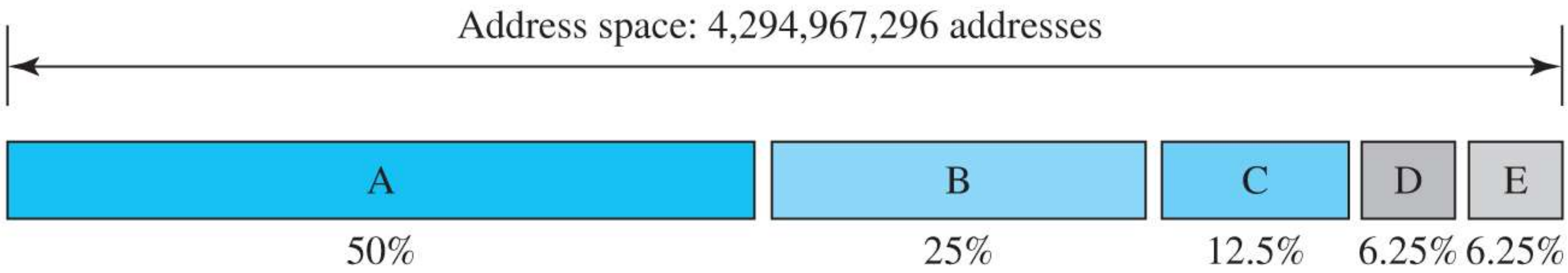
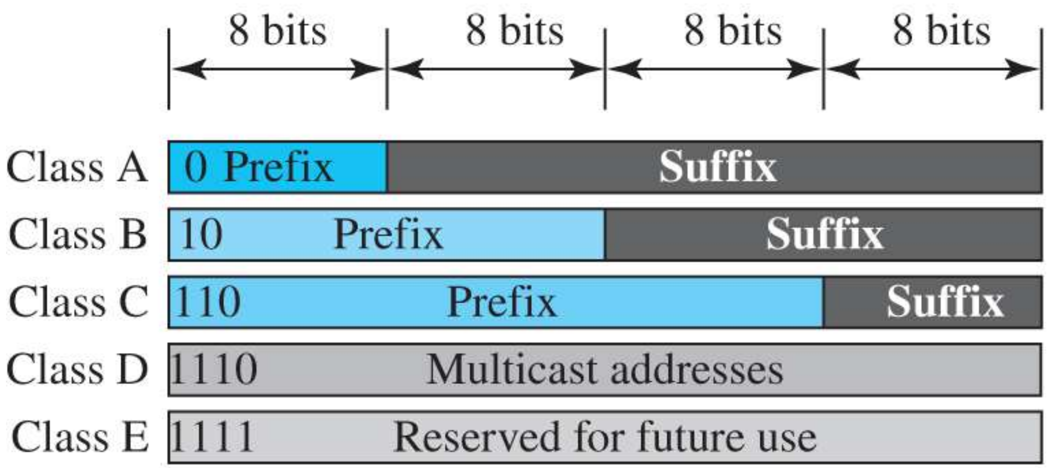
Classful Addressing

When the Internet started, an IPv4 address was designed with a fixed-length prefix, but to accommodate both small and large networks, three fixed-length prefixes were designed instead of one (n = 8, n = 16, and n = 24). The whole address space was divided into five classes (class A, B, C, D, and E), as shown in Figure 7.6. This scheme is referred to as classful addressing. Although classful addressing belongs to the past, it helps us to understand classless addressing, discussed later.

Figure 7.6 Occupation of the address space in classful addressing





Class	Prefixes	First byte
A	$n = 8$ bits	0 to 127
B	$n = 16$ bits	128 to 191
C	$n = 24$ bits	192 to 223
D	Not applicable	224 to 239
E	Not applicable	240 to 255

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Classless Addressing

With the growth of the Internet, it was clear that a larger address space was needed as a long-term solution. The larger address space, however, requires that the length of IP addresses also be increased, which means the format of the IP packets needs to be changed. Although the long-range solution has already been devised and is called IPv6, a short-term solution was also devised to use the same address space but to change the distribution of addresses to provide a fair share to each organization. The short-term solution still uses IPv4 addresses, but it is called classless addressing.

Figure 7.7 Variable-length blocks in classless addressing

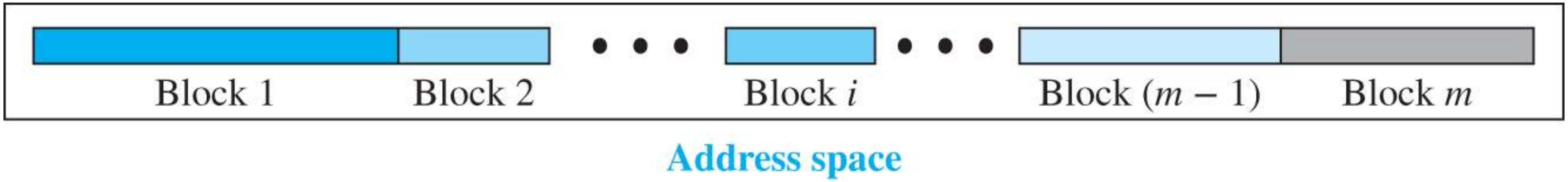

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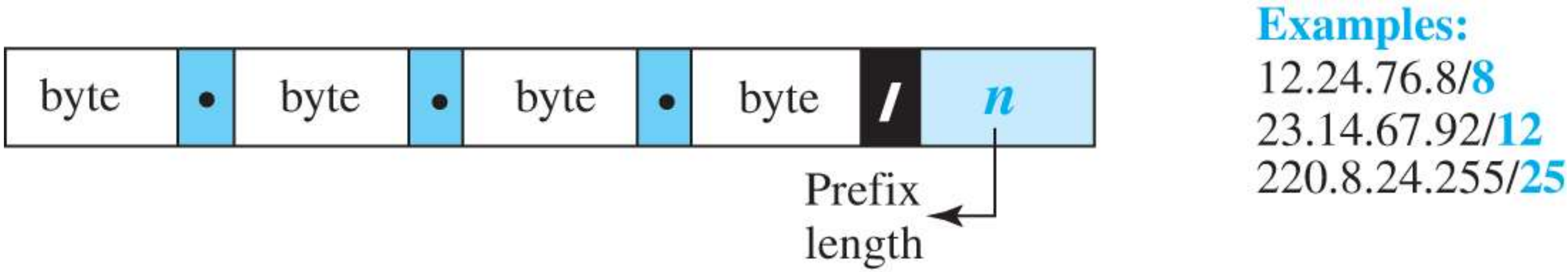
Figure 7.8 Slash notation (CIDR)



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Example 7.1

A classless address is given as 167.199.170.82/27. We can find the above three pieces of information as follows. The number of addresses in the network is $2^{32-n} = 25 = 32$ addresses. The first address can be found by keeping the first 27 bits and changing the rest of the bits to 0s.

Address: 167.199.170.82/**27**	10100111	11000111	10101010	01010010
First address: 167.199.170.64/**27**	10100111	11000111	10101010	01000000

The last address can be found by keeping the first 27 bits and changing the rest of the bits to 1s.

Address: 167.199.170.82/**27**	10100111	11000111	10101010	01011111
Last address: 167.199.170.95/**27**	10100111	11000111	10101010	01011111

Example 7.2

We repeat Example 7.1 using the mask. The mask in dotted-decimal notation is 256.256.256.224 The AND, OR, and NOT operations can be applied to individual bytes using calculators and applets at the book website.

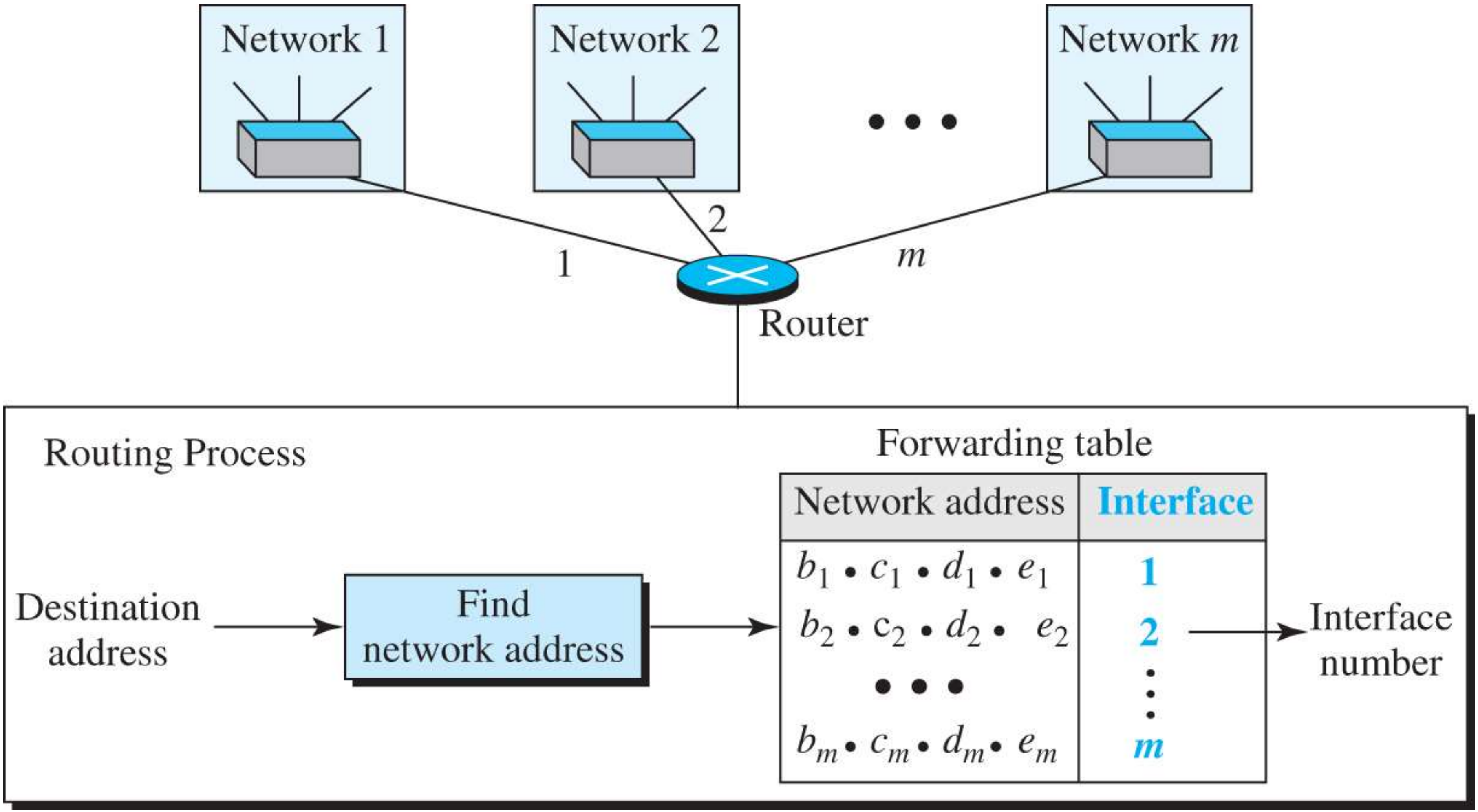
Number of addresses in the block:	N = **NOT** (mask) + 1 = 0.0.0.31 + 1 = 32 addresses
First address:	First = (address) **AND** (mask) = 167.199.170. 82
Last address:	Last = (address) **OR** (**NOT** mask) = 167.199.170. 255

Example 7.3

In classless addressing, an address cannot per se define the block the address belongs to. For example, the address 230.8.24.56 can belong to many blocks. Some of them are shown below with the value of the prefix associated with that block.

Prefix length:16	→	Block:	230.8.0.0	to	230.8.255.255
Prefix length:20	→	Block:	230.8.16.0	to	230.8.31.255
Prefix length:26	→	Block:	230.8.24.0	to	230.8.24.63
Prefix length:27	→	Block:	230.8.24.32	to	230.8.24.63
Prefix length:29	→	Block:	230.8.24.56	to	230.8.24.63
Prefix length:31	→	Block:	230.8.24.56	to	230.8.24.57

Figure 7.9 Network address



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Example 7.4

An ISP has requested a block of 1000 addresses. Since 1000 is not a power of 2, 1024 addresses are granted. The prefix length is calculated as $n = 32 - \log_2 1024 = 22.$ An available block, 18.14.12.0**/22**, is granted to the ISP. It can be seen that the first address in decimal is 302,910,464, which is divisible by 1024.

Example 7.5 (1)

An organization is granted a block of addresses with the beginning address 14.24.74.0**/24**. The organization needs to have 3 subblocks of addresses to use in its three subnets: one subblock of 10 addresses, one subblock of 60 addresses, and one subblock of 120 addresses. Design the subblocks.

Solution

There are $2^{32-24} = 256$ addresses in this block. The first address is 14.24.74.0**/24**; the last address is 14.24.74.255**/24**. To satisfy the third requirement, we assign addresses to subblocks, starting with the largest and ending with the smallest one.

Example 7.5 ***(2)***

a. The number of addresses in the largest subblock, which requires 120 addresses, is not a power of 2. We allocate 128 addresses. The subnet mask for this subnet can be found as $n_1 = 32 - \log_2 128 = 25$. The first address in this block is 14.24.74.0**/25**; the last address is 14.24.74.127**/25**.

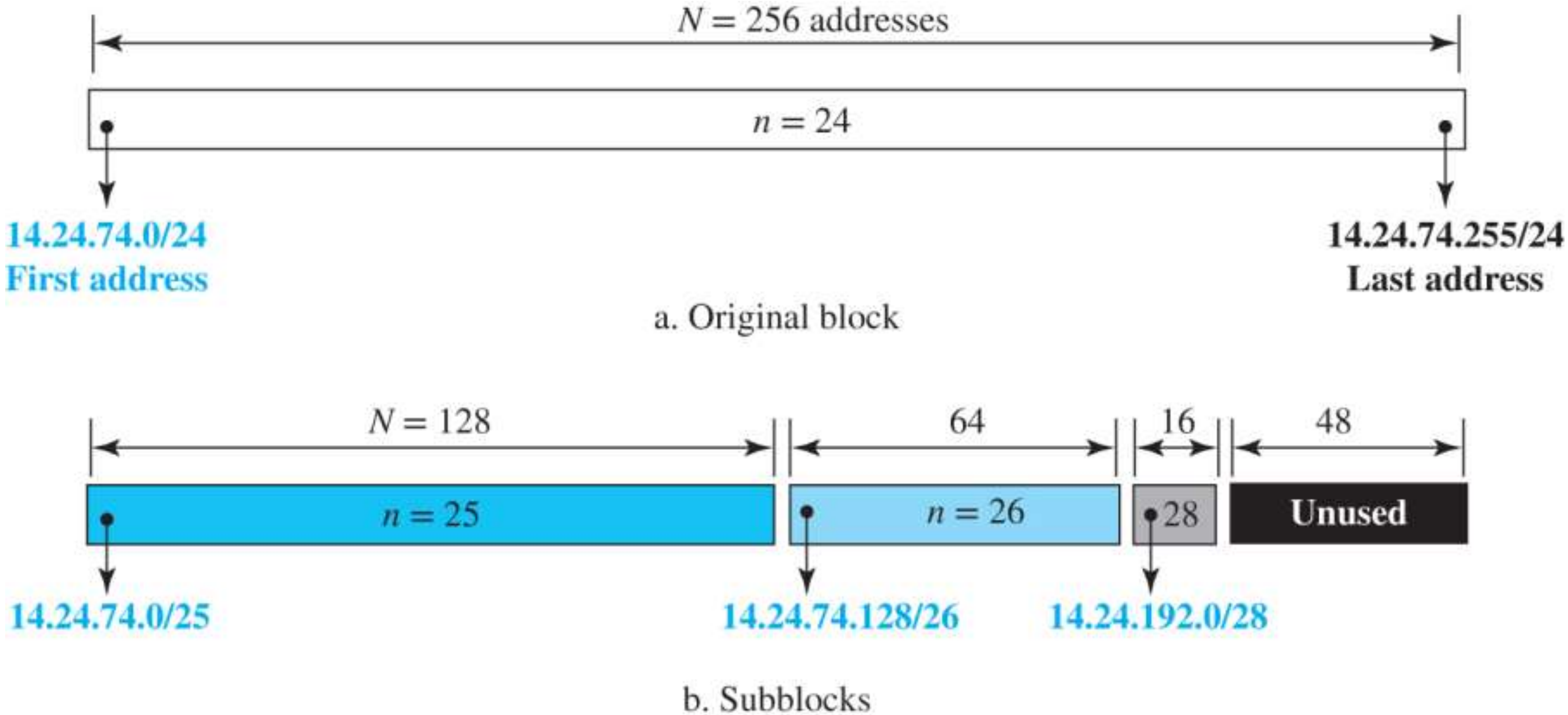
b. The number of addresses in the second largest subblock, which requires 60 addresses, is not a power of 2 either. We allocate 64 addresses. The subnet mask for this subnet can be found as $n_2 = 32 - \log_2 64 = 26.$ The first address in this block is 14.24.74.128**/26**; the last address is 14.24.74.191**/26**.

Example 7.5 *(3)*

c. The number of addresses in the largest subblock, which requires 120 addresses, is not a power of 2. We allocate 128 addresses. The subnet mask for this subnet can be found as $n_1 = 32 - \log_2 128 = 25$. The first address in this block is 14.24.74.0**/25**; the last address is 14.24.74.127**/25**.

If we add all addresses in the previous subblocks, the result is 208 addresses, which means 48 addresses are left in reserve. The first address in this range is 14.24.74.208. The last address is 14.24.74.255. We don’t know about the prefix length yet. Figure 4.36 shows the configuration of blocks. We have shown the first address in each block.

Figure 7.10 Solution to Example 4.5



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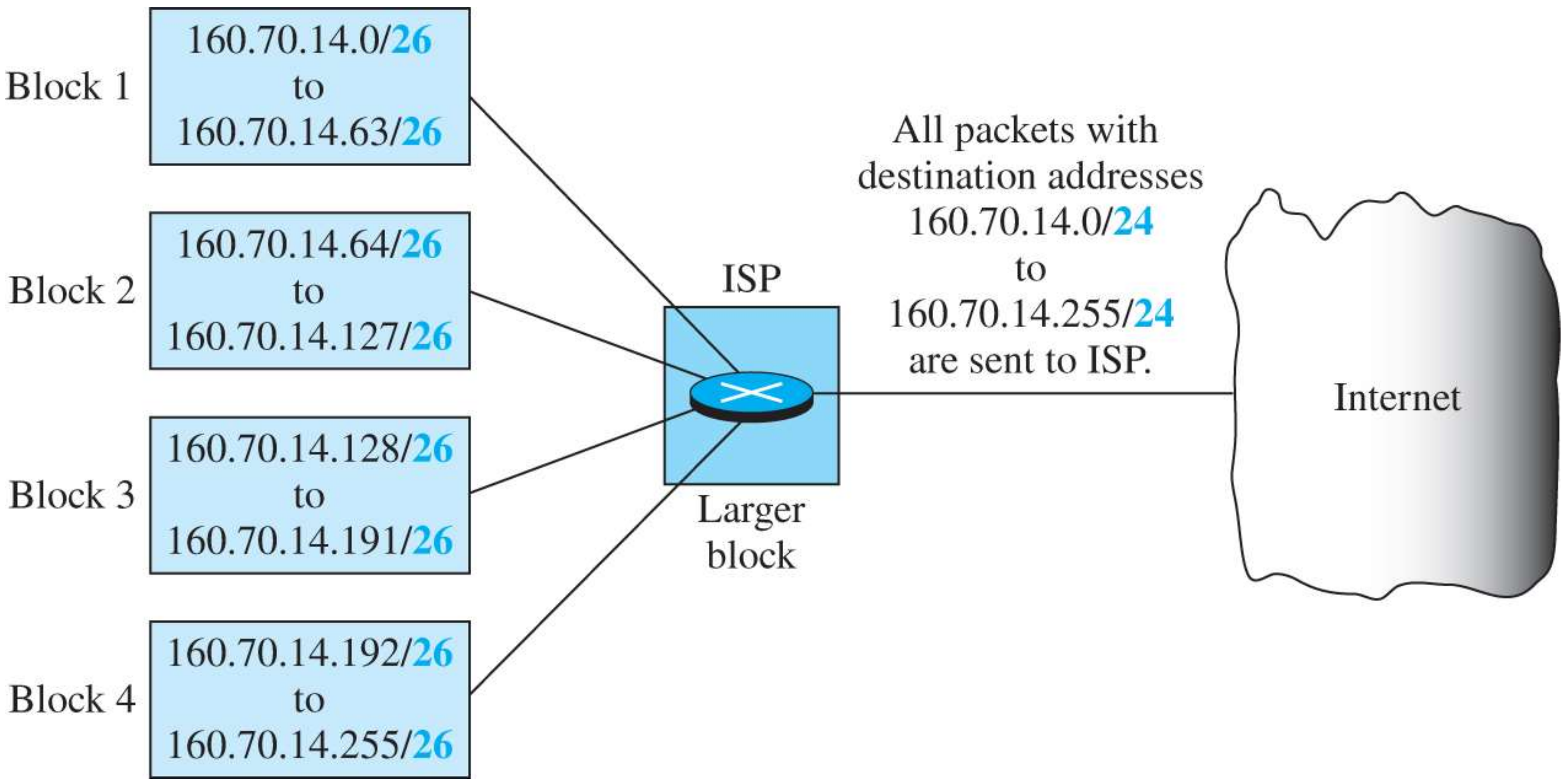
Example 7.6

Figure 7.11 shows how four small blocks of addresses are assigned to four organizations by an ISP. The ISP combines these four blocks into one single block and advertises the larger block to the rest of the world. Any packet destined for this larger block should be sent to this ISP. It is the responsibility of the ISP to forward the packet to the appropriate organization. This is similar to routing we can find in a postal network. All packages coming from outside a country are sent first to the capital and then distributed to the corresponding destination.

Figure 7.11 Example of address aggregation



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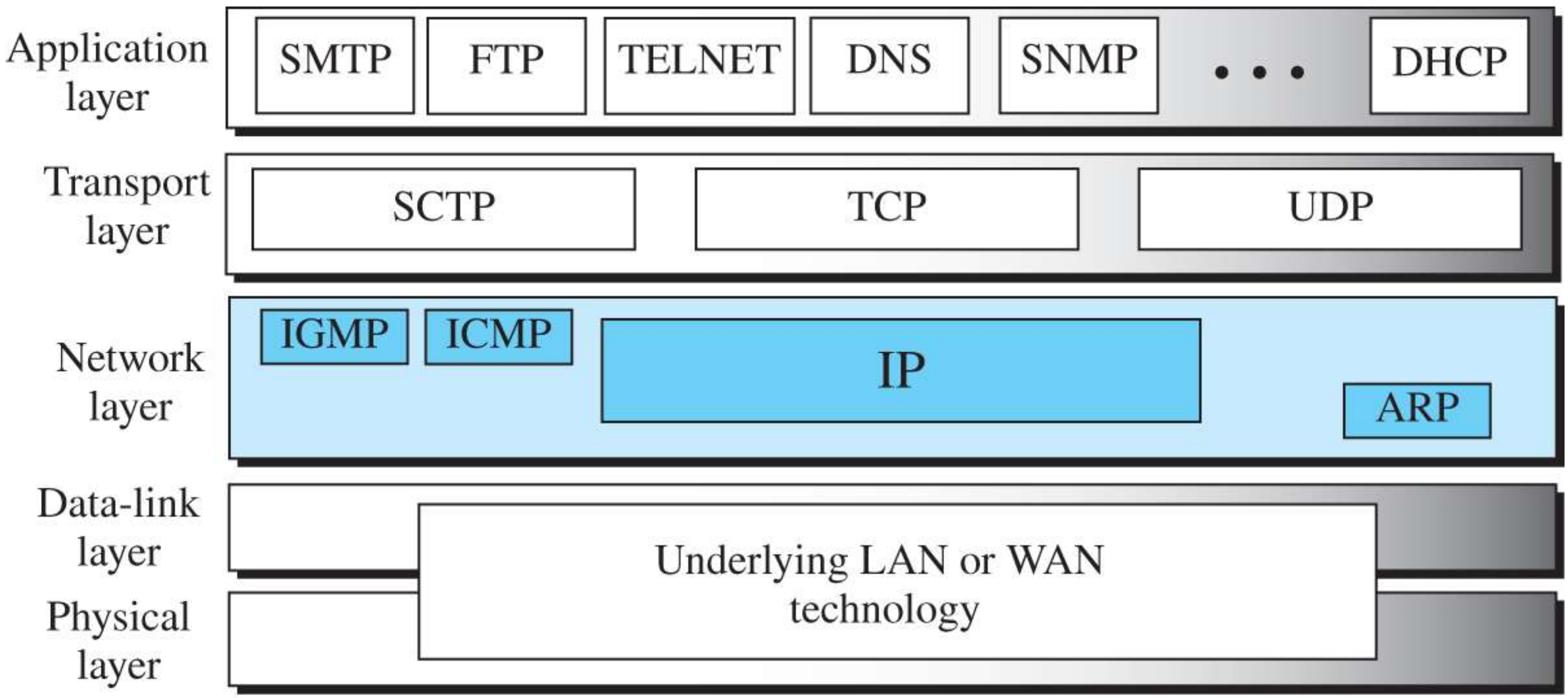
7.4.2 Four Related Protocols

The network layer in version 4 can be thought of as one main protocol and three auxiliary ones. The main protocol, IPv4, is responsible for packetizing, forwarding, and delivery of a packet. The ICMPv4 helps IPv4 to handle some errors that may occur in delivery. The IGMP is used to help IPv4 in multicasting. ARP is used in address mapping.

Figure 7.12 Position of IP and other network-layer protocols in TCP/IP protocol suite



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Datagram Format

Packets used by the IP are called datagrams. Figure 7.13 shows the IPv4 datagram format. A datagram is a variable-length packet consisting of two parts: header and payload (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.

Figure 7.13 IP datagram

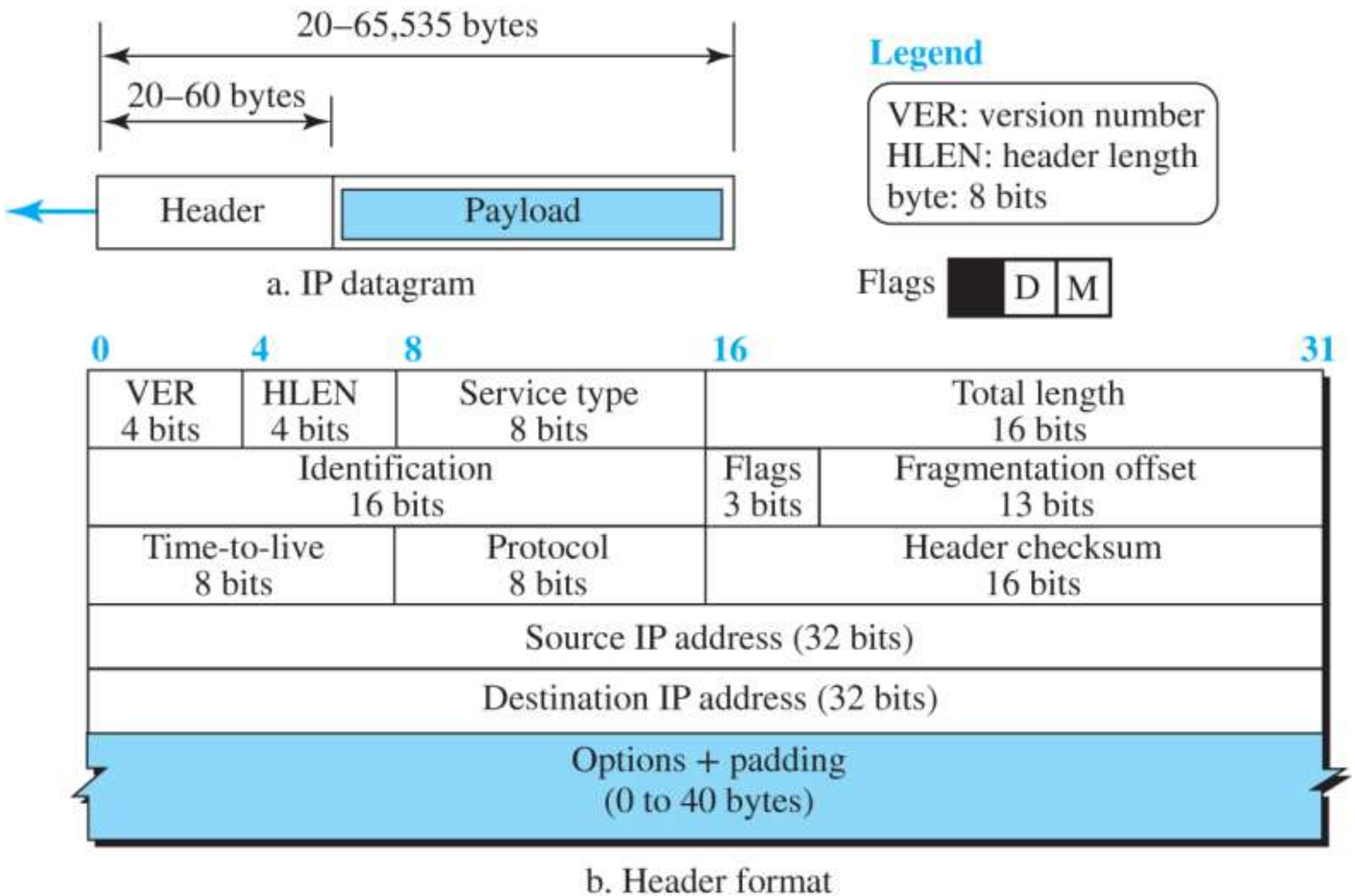

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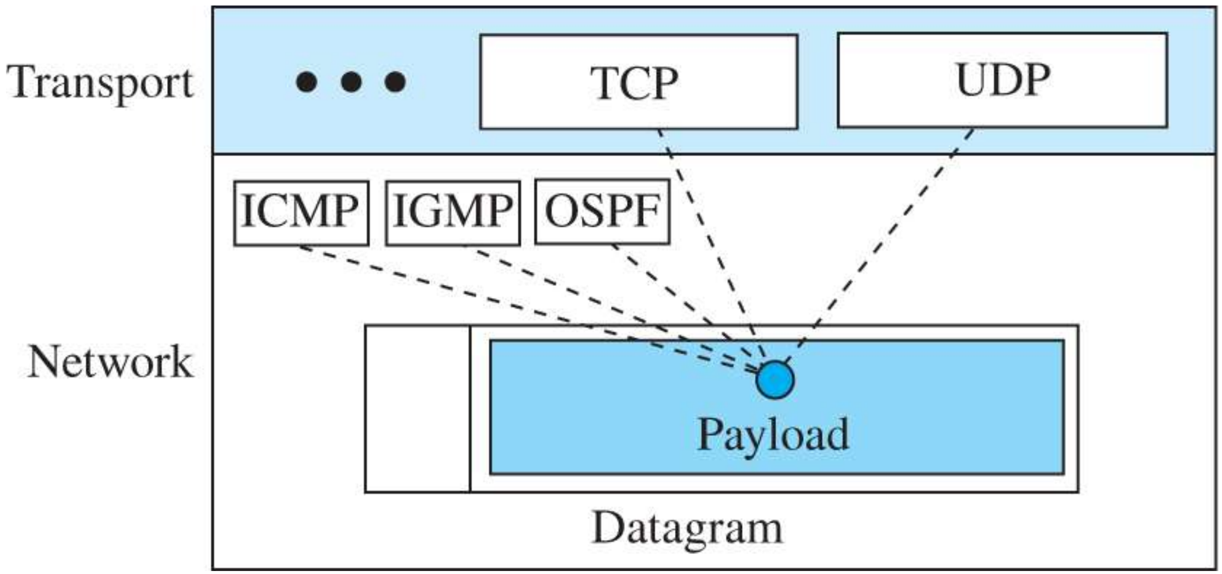
Figure 7.14 Multiplexing and demultiplexing using the value of the protocol field



Some protocol values

ICMP	01
IGMP	02
TCP	06
UDP	17
OSPF	89

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Example 7.7

An IPv4 packet has arrived with the first 8 bits as $(01000010)_2$. The receiver discards the packet. Why?

Solution

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

Example 7.8

In an IPv4 packet, the value of HLEN is $(1000)_2$. 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 7.9

In an IPv4 packet, the value of HLEN is 5, and the value of the total length field is $(0028)_{16}$. 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 $(0028)_{16}$ or 40 bytes, which means the packet is carrying 20 bytes of data (40 - 20).

Example 7.10

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

$$\left(\boldsymbol{4500002800010000010 2}\ldots\right)_{\boldsymbol{16}}$$

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

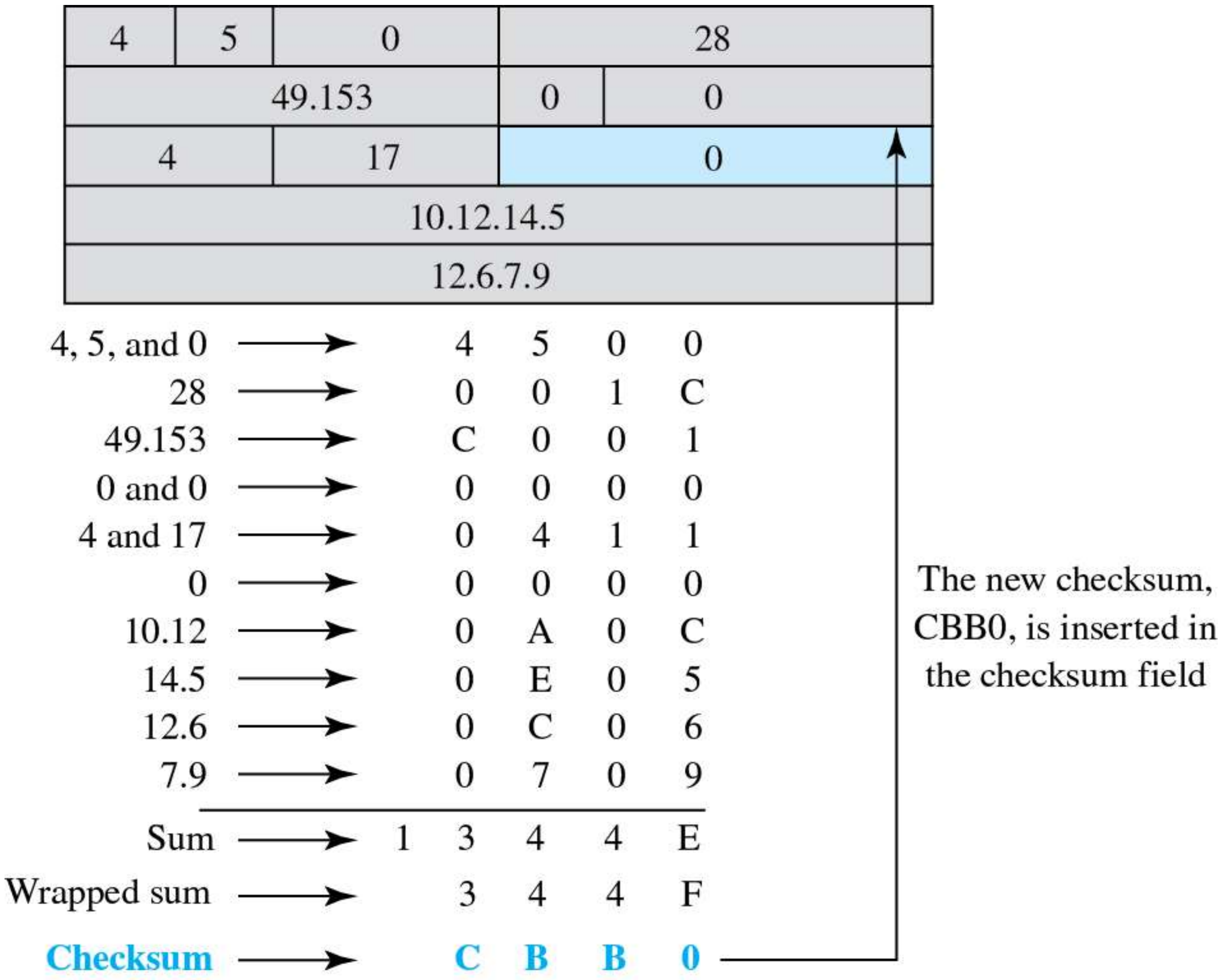
Solution

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 $(01)_{16}$. This means the packet can travel only one hop. The protocol field is the next byte $(02)_{16}$, which means that the upper-layer protocol is IGMP.

Example 7.11

Figure 7.15 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 after wrapping the leftmost digit. The result is inserted in the checksum field.

Figure 7.15 Example of checksum calculation



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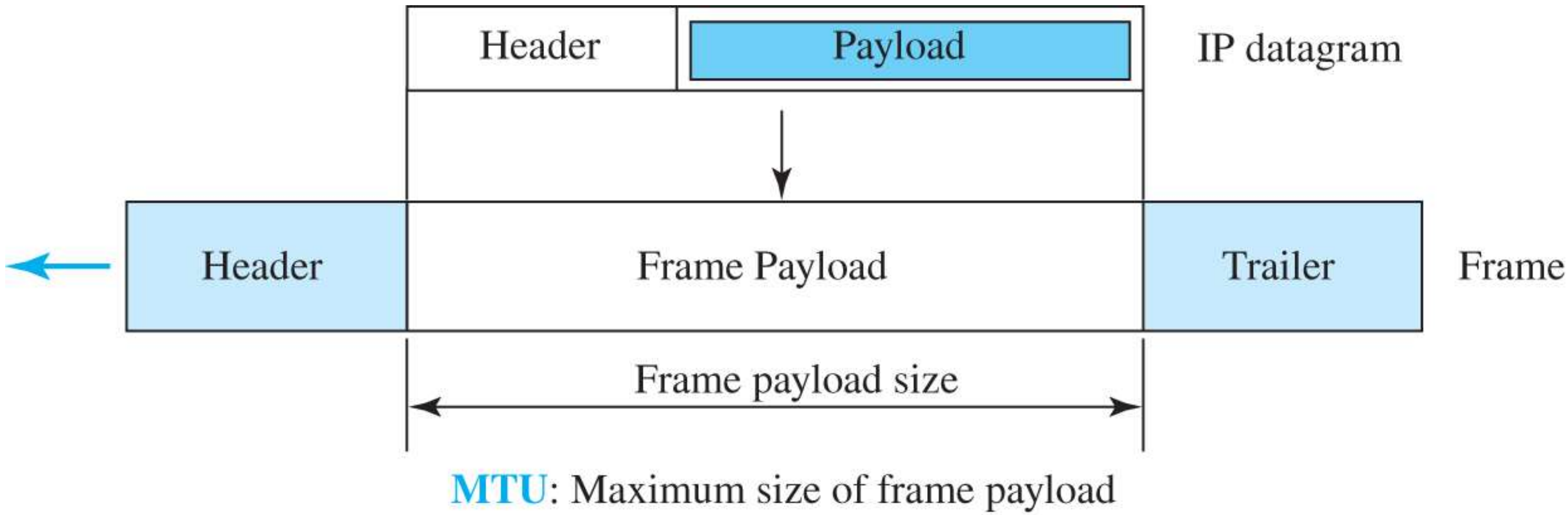
Fragmentation

A datagram can travel through different networks. Each router decapsulates the IP 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

Each link-layer protocol has its own frame format. One of the features of each format is the maximum size of the payload that can be encapsulated in a frame, total size of the datagram must be less than the maximum size (see Figure 7.16).

Figure 7.16 Maximum transfer unit (MTU)



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Fields Related to Fragmentation

We mentioned before that three fields in an IP datagram are related to fragmentation: identification, flags, and fragmentation offset.

The 16-bit identification field identifies a datagram originating from the source host.

The 3-bit flags field defines three flags.

The 13-bit fragmentation offset field shows the relative position of this fragment with respect to the whole datagram.

Figure 7.17 Fragmentation example

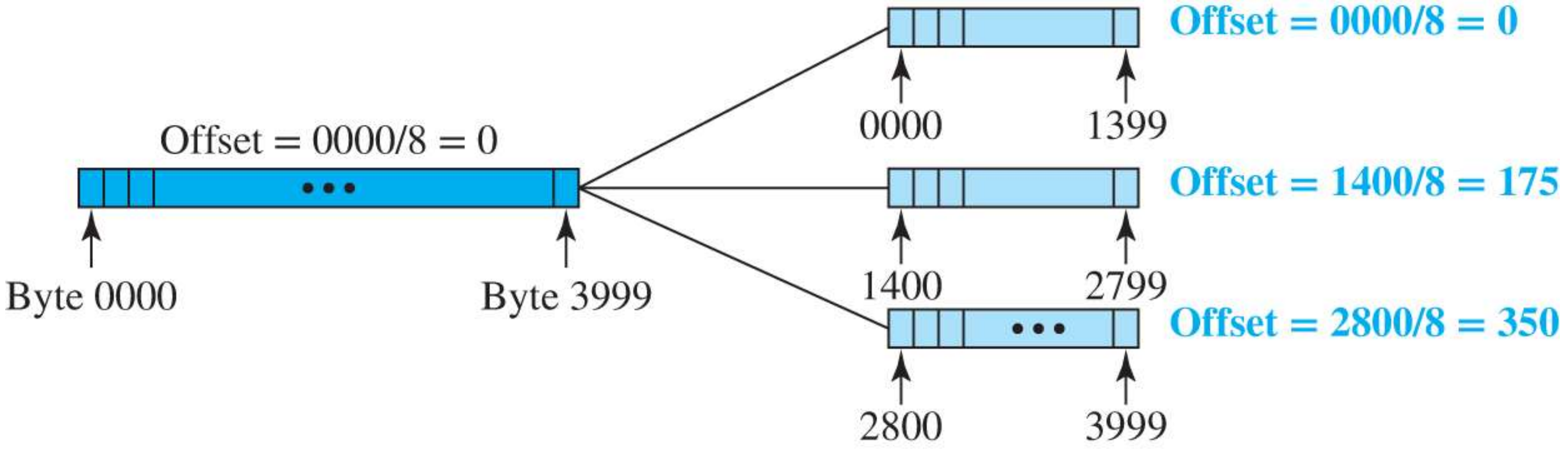

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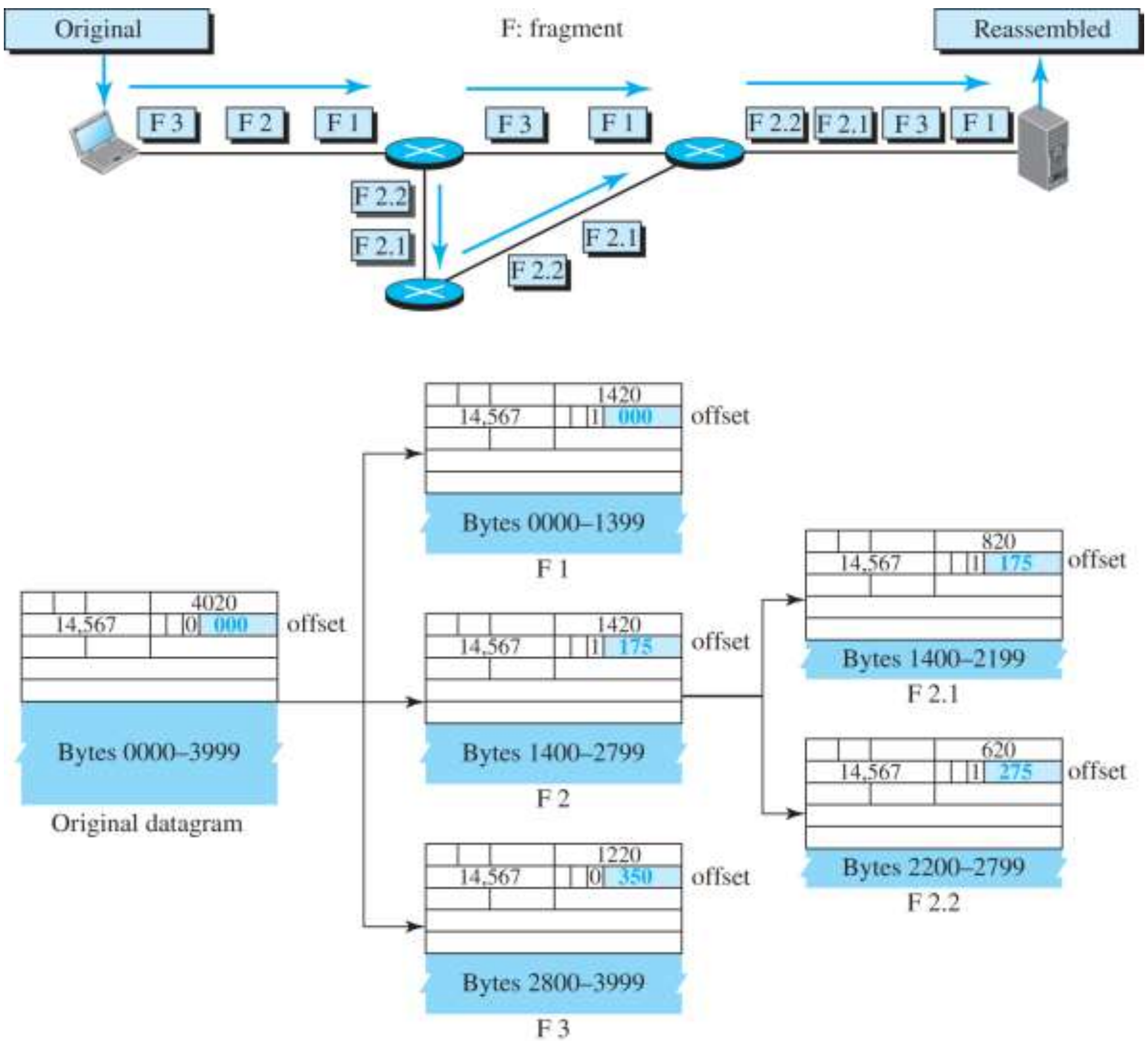
Figure 7.18 Detailed fragmentation example



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Example 7.12

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 non-fragmented packet is considered the last fragment.

Example 7.13

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).

Example 7.14

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 7.15

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 7.16

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 * 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.

7.4.3 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 (in multiples of 4 bytes) to preserve the boundary of the header.

Single-Byte Options

There are two single-byte options.

No Operation

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

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.

Security of IPv4 Datagrams

We give a brief idea about the security issues in IP protocol.

Packet Sniffing

An intruder may intercept an IP packet and make a copy of it.

Packet Modification

The attacker intercepts the packet, changes its contents, and sends the new packet to the receiver..

IP Spoofing

An attacker can masquerade as somebody else and create an IP packet that carries the source address of another computer.

IPSec

The IP packets today can be protected from the previously mentioned attacks using a protocol called IPSec (IP Security). This protocol, which is used in conjunction with the IP protocol, creates a connection-oriented service between two entities in which they can exchange IP packets without worrying about the three attacks discussed above. We will discuss IPSec in detail in Chapter 13.

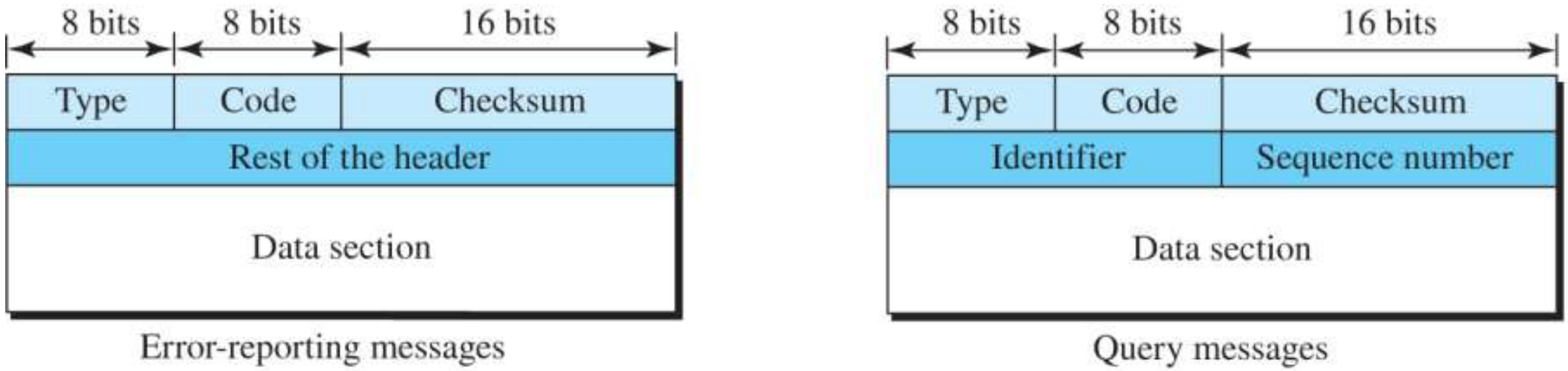
7.4.4 ICMPv4

The IPv4 has no error-reporting or error-correcting mechanism. The IP protocol also lacks a mechanism for host and management queries. The Internet Control Message Protocol version 4 (ICMPv4) has been designed to compensate for the above two deficiencies.

Messages

ICMP messages are divided into two broad categories: error-reporting messages and query messages. The error-reporting messages report problems that a router or a host (destination) may encounter when it processes an IP packet. The query messages, which occur in pairs, help a host or a network manager get specific information from a router or another host. For example, nodes can discover their neighbors. Also, hosts can discover and learn about routers on their network and routers can help a node redirect its messages.

Figure 7.19 General format of ICMP messages



Type and code values

Error-reporting messages

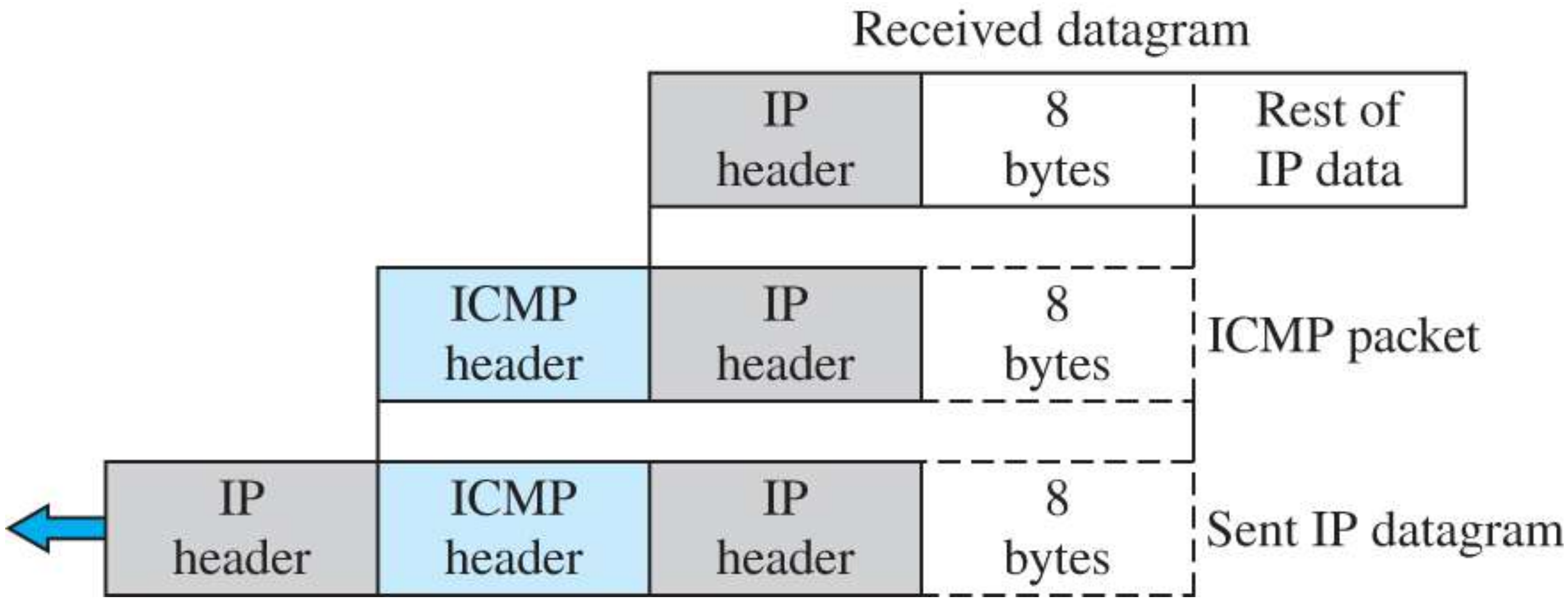
03: Destination unreachable (codes 0 to 15)
04: Source quench (only code 0)
05: Redirection (codes 0 to 3)
11: Time exceeded (codes 0 and 1)
12: Parameter problem (codes 0 and 1)

Query messages

08 and 00: Echo request and reply (only code 0)
13 and 14: Timestamp request and reply (only code 0)

Access the text alternative for slide images.

Figure 7.20 Contents of data field for error messages



Access the text alternative for slide images.

Deprecated Messages

Three pairs of messages are declared obsolete by IETF:

1. *Information request and replay messages*
2. *Address mask request and reply messages*
3. *Router solicitation and advertisement messages*

Debugging Tools

There are several tools that can be used in the Internet for debugging. We can determine the viability of a host or router. We can trace the route of a packet. We introduce two tools that use ICMP for debugging: ping and traceroute.

Ping

We can use the ping program to find if a host is alive and is responding.

Example 7.17

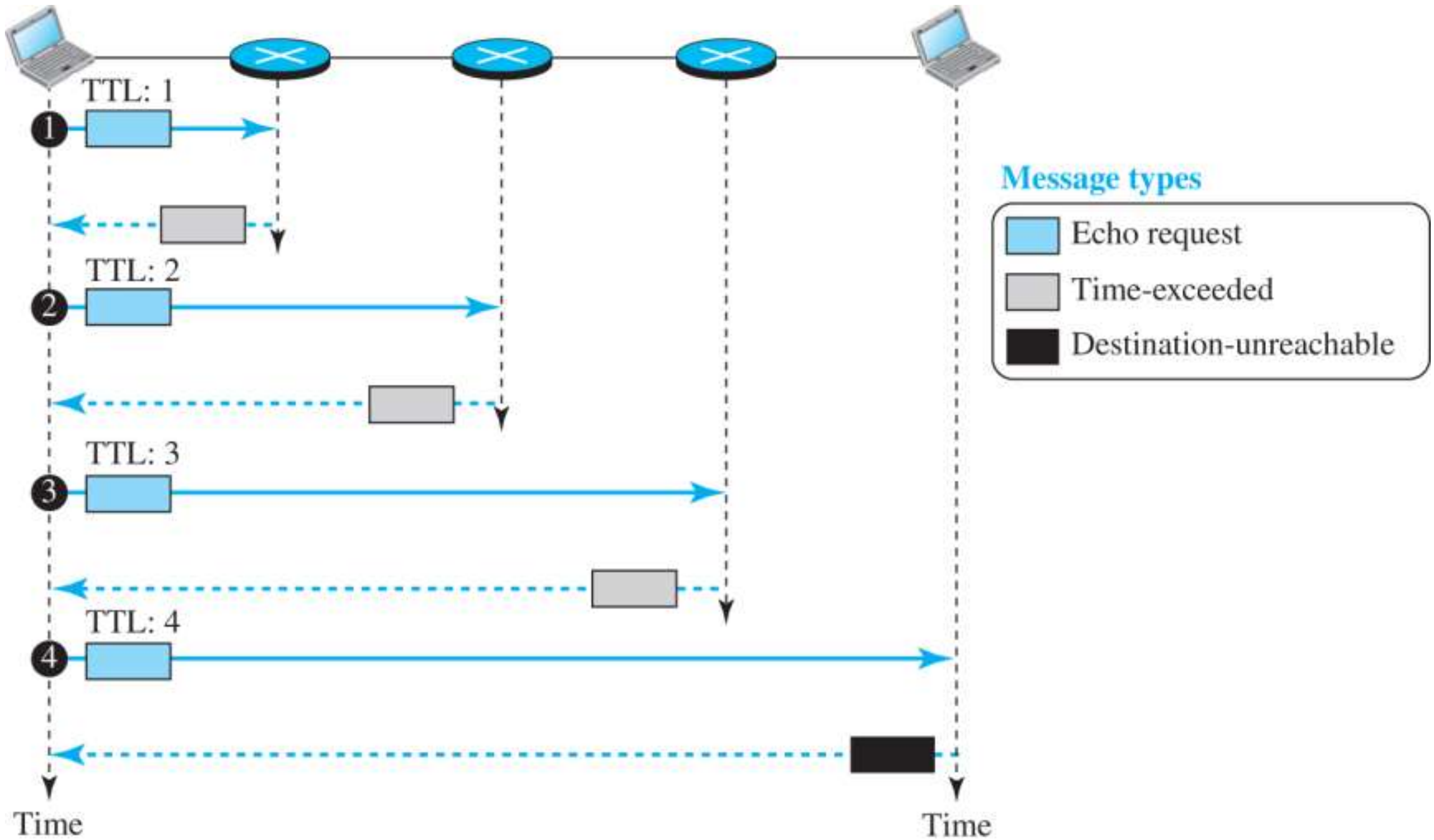
The following shows how we send a ping message to the auniversity.edu site.

```
$ ping auniversity.edu
PING auniversity.edu (152.181.8.3) 56 (84)   bytes of data.     ttl=62     time=1.91 ms
64 bytes from auniversity.edu (152.181.8.3): icmp_seq=0         ttl=62     time=2.04 ms
64 bytes from auniversity.edu (152.181.8.3): icmp_seq=1         ttl=62     time=1.90 ms
64 bytes from auniversity.edu (152.181.8.3): icmp_seq=2         ttl=62     time=1.90 ms
64 bytes from auniversity.edu (152.181.8.3): icmp_seq=3         ttl=62     time=1.97 ms
64 bytes from auniversity.edu (152.181.8.3): icmp_seq=4         ttl=62     time=1.93 ms
64 bytes from auniversity.edu (152.181.8.3): icmp_seq=5         ttl=62     time=2.00 ms
--- auniversity.edu statistics ---
6 packets transmitted, 6 received, 0% packet loss
rtt min/avg/max = 1.90/1.95/2.04 ms
```

Traceroute or Tracert

The traceroute program in UNIX or tracert in Windows can be used to trace the path of a packet from a source to the destination. It can find the IP addresses of all the routers that are visited along the path. The program is usually set to check for the maximum of 30 hops (routers) to be visited. The number of hops in the Internet is normally less than this. Since these two programs behave different in Unix and Windows, we explain them separately.

Figure 7.21 Example of traceroute program



Access the text alternative for slide images.

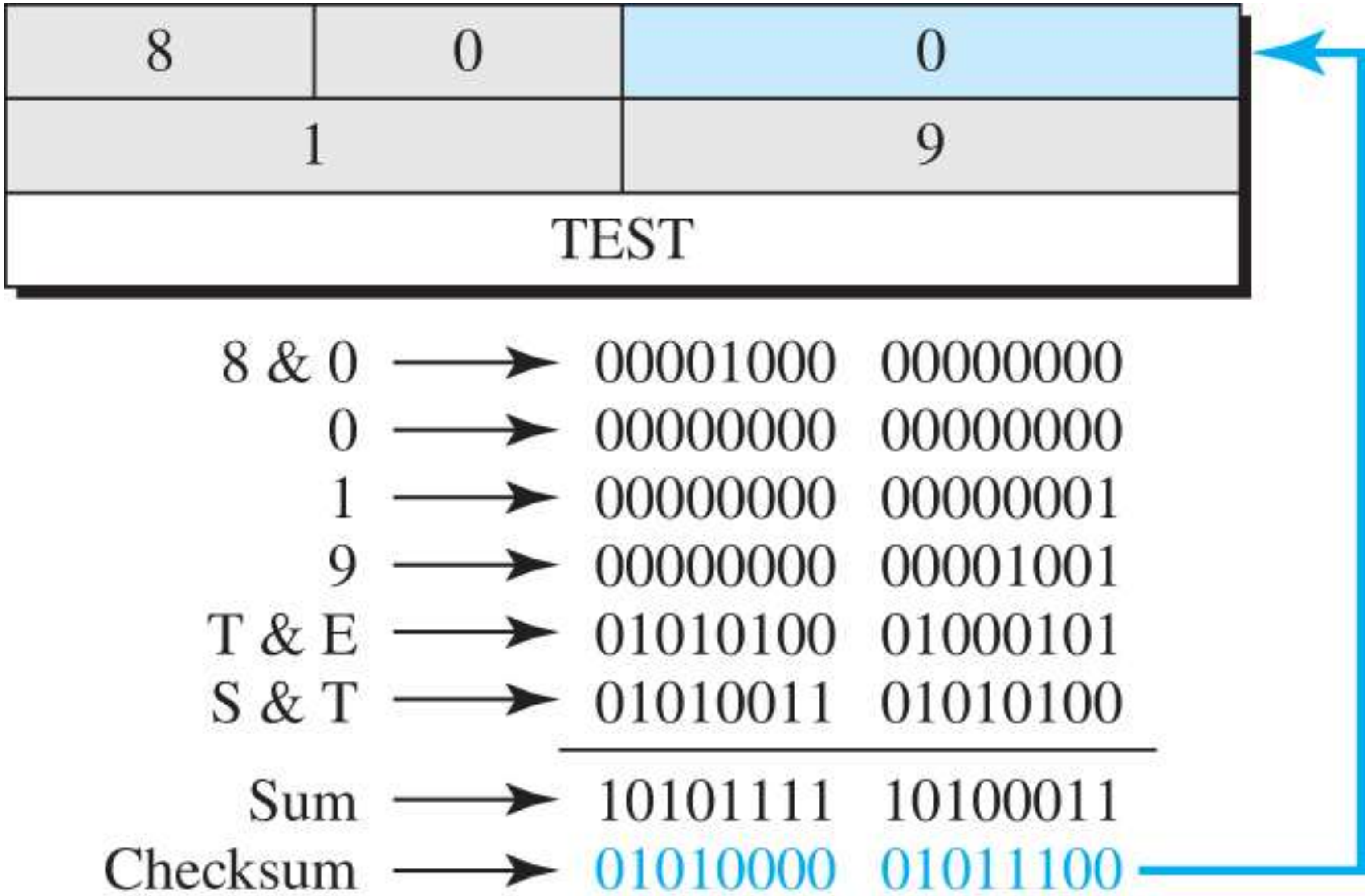
ICMP Checksum

In ICMP, the checksum is calculated over the entire message (header and data).

Example 7.18

Figure 7.22 shows an example of checksum calculation for a simple echo-request message. We randomly chose the identifier to be 1 and the sequence number to be 9. The message is divided into 16-bit (2-byte) words. The words are added and the sum is complemented. Now the sender can put this value in the checksum field.

Figure 7.22 Example of checksum calculation



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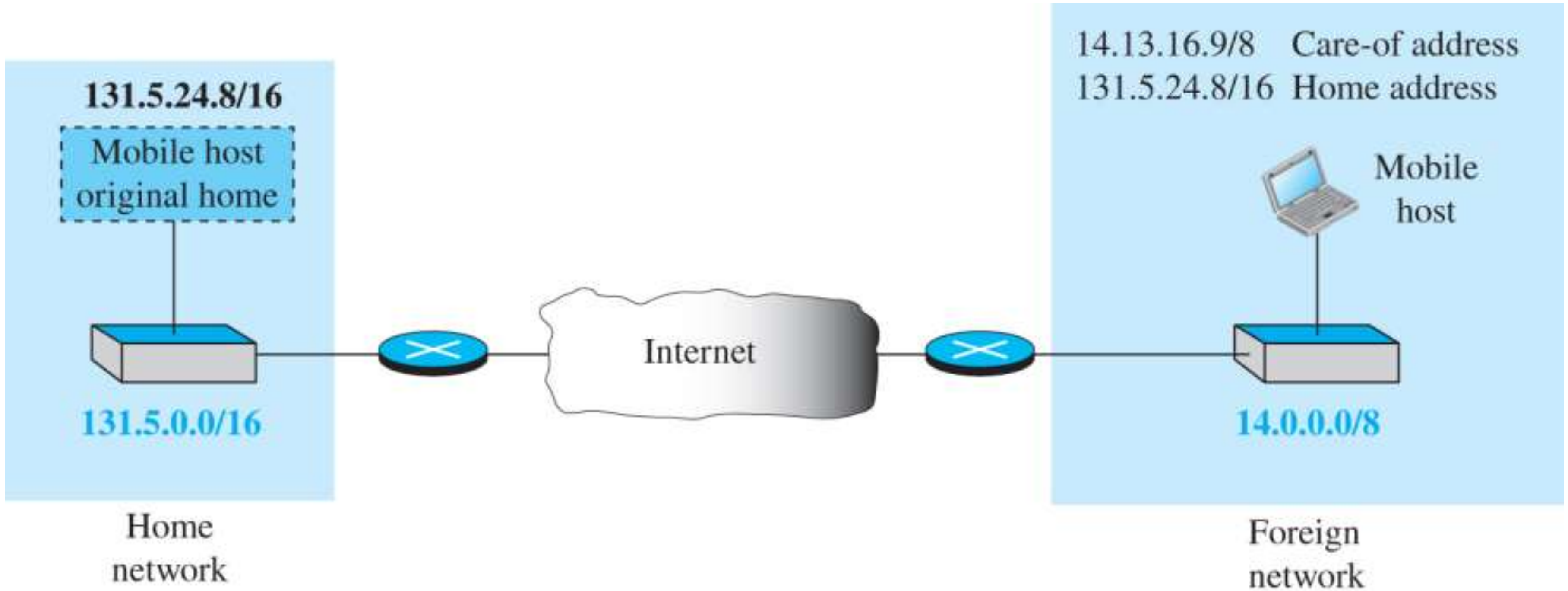
7.4.5 Mobile IP

In the last section of this chapter, we discuss mobile IP. As mobile and personal computers such as notebooks become increasingly popular, we need to think about mobile IP, the extension of IP protocol that allows mobile computers to be connected to the Internet at any location where the connection is possible. In this section, we discuss this issue.

Addressing

The main problem that must be solved in providing mobile communication using the IP protocol is addressing.

Figure 7.23 Home address and care-of address



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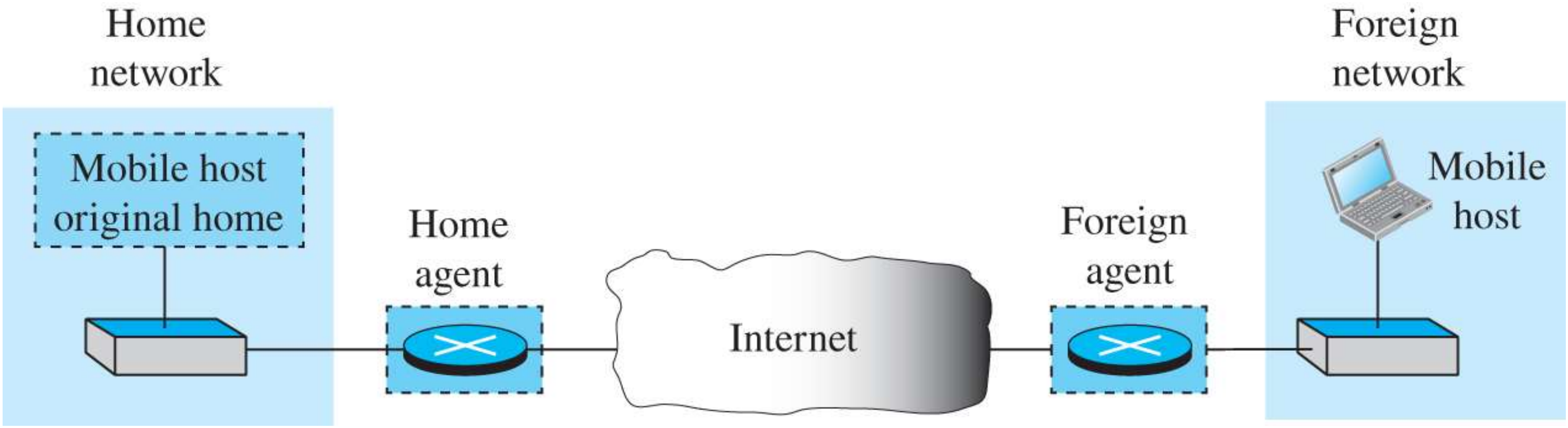
Agents

To make the change of address transparent to the rest of the Internet requires a home agent and a foreign agent. Figure 7.24 shows the position of a home agent relative to the home network and a foreign agent relative to the foreign network.

Figure 7.24 Home agent and foreign agent



Access the text alternative for slide images.

Three Phases

To communicate with a remote host, a mobile host goes through three phases: agent discovery, registration, and data transfer, as shown in Figure 7.25.

Figure 7.25 Remote host and mobile host communication

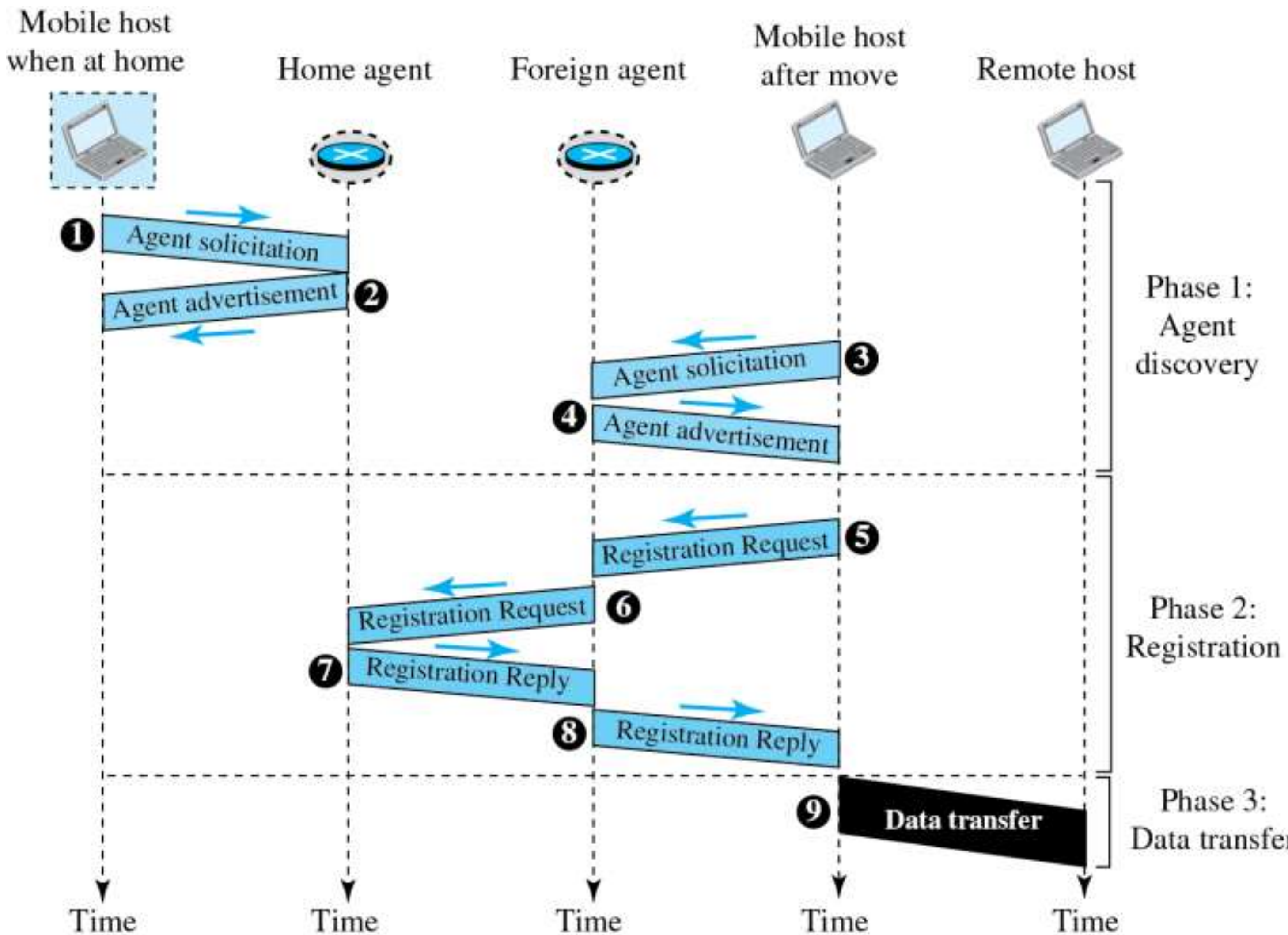


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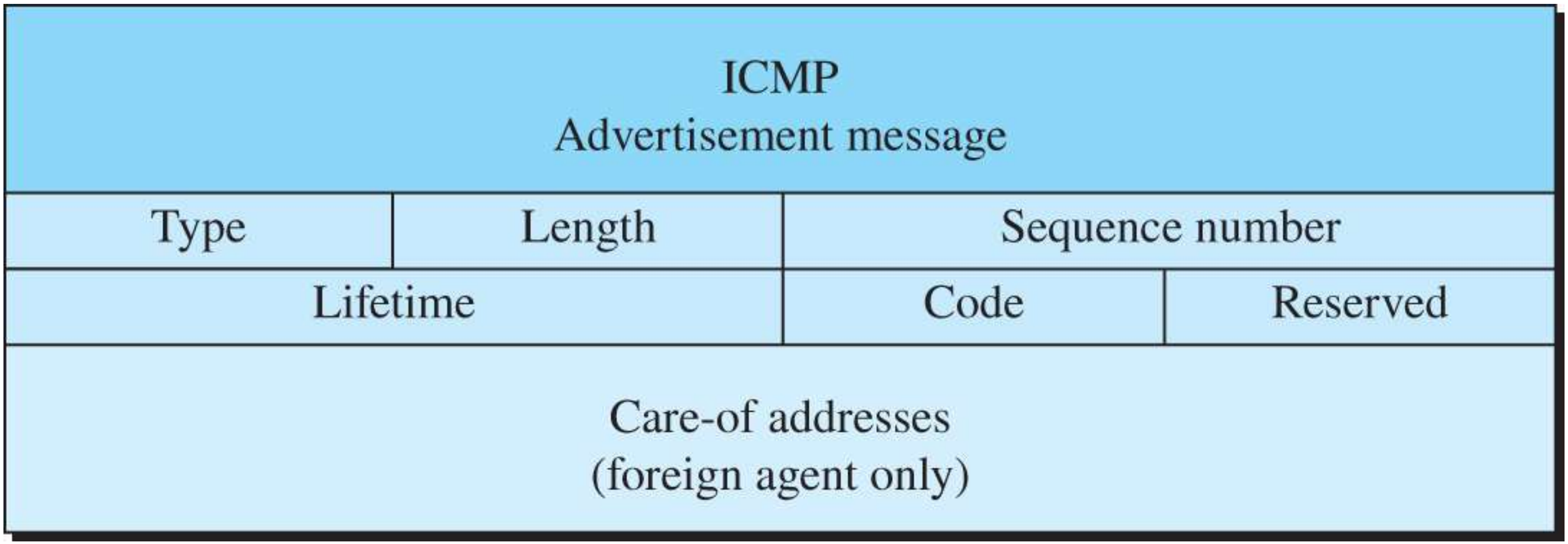
Figure 7.26 Agent advertisement



Access the text alternative for slide images.

Table 7.1 Code Bits

Bit	*Meaning*
0	Registration required. No collocated care-of address.
1	Agent is busy and does not accept registration at this moment.
2	Agent acts as a home agent.
3	Agent acts as a foreign agent.
4	Agent uses minimal encapsulation.
5	Agent uses generic routing encapsulation (GRE).
6	Agent uses generic routing encapsulation (GRE).
7	Unused (0).

Registration

The second phase in mobile communication is registration. After a mobile host has moved to a foreign network and discovered the foreign agent, it must register. There are four aspects of registration:

1. *The mobile host must register itself with the foreign agent.*
2. *The mobile host must register itself with its home agent.*
3. *The mobile host must renew registration if it has expired.*
4. *The mobile host must cancel its registration when it returns.*

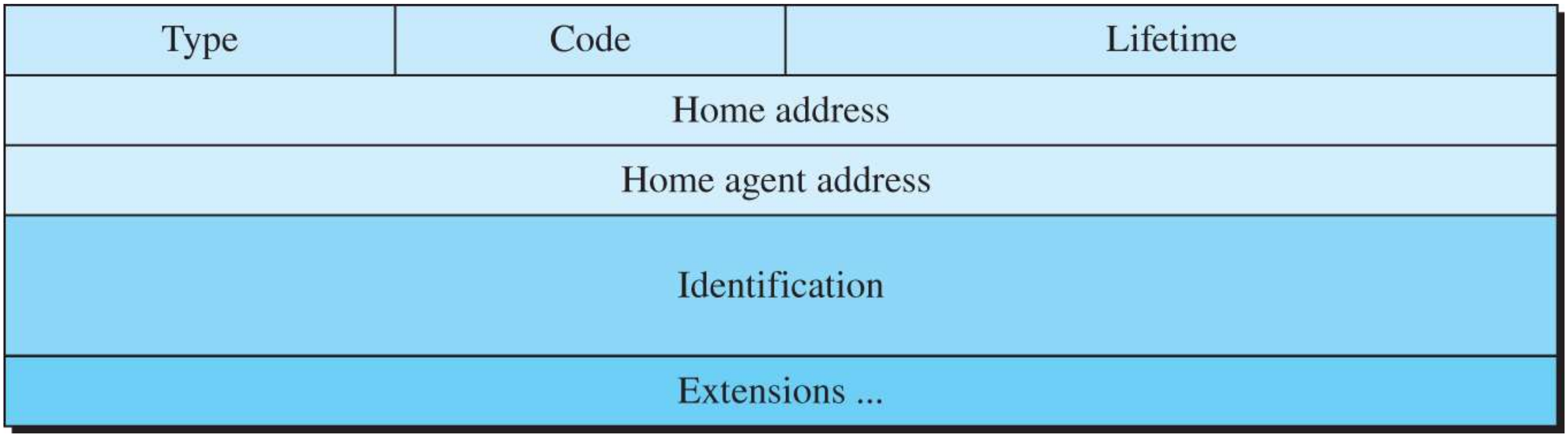
Figure 7.27 Registration request format

Access the text alternative for slide images.

Table 7.2 Registration request flag field bits

Bit	*Meaning*
0	Mobile host requests that home agent retain its prior care-of address.
1	Mobile host requests that home agent tunnel any broadcast message.
2	Mobile host is using collocated care-of address.
3	Mobile host requests that home agent use minimal encapsulation.
4	Mobile host requests generic routing encapsulation (GRE).
5	Mobile host requests header compression.
6 –7	Reserved bits.

Figure 7.28 Registration reply format



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Data Transfer

After agent discovery and registration, a mobile host can communicate with a remote host. Figure 7.29 shows the idea.

Figure 7.29 Data transfer

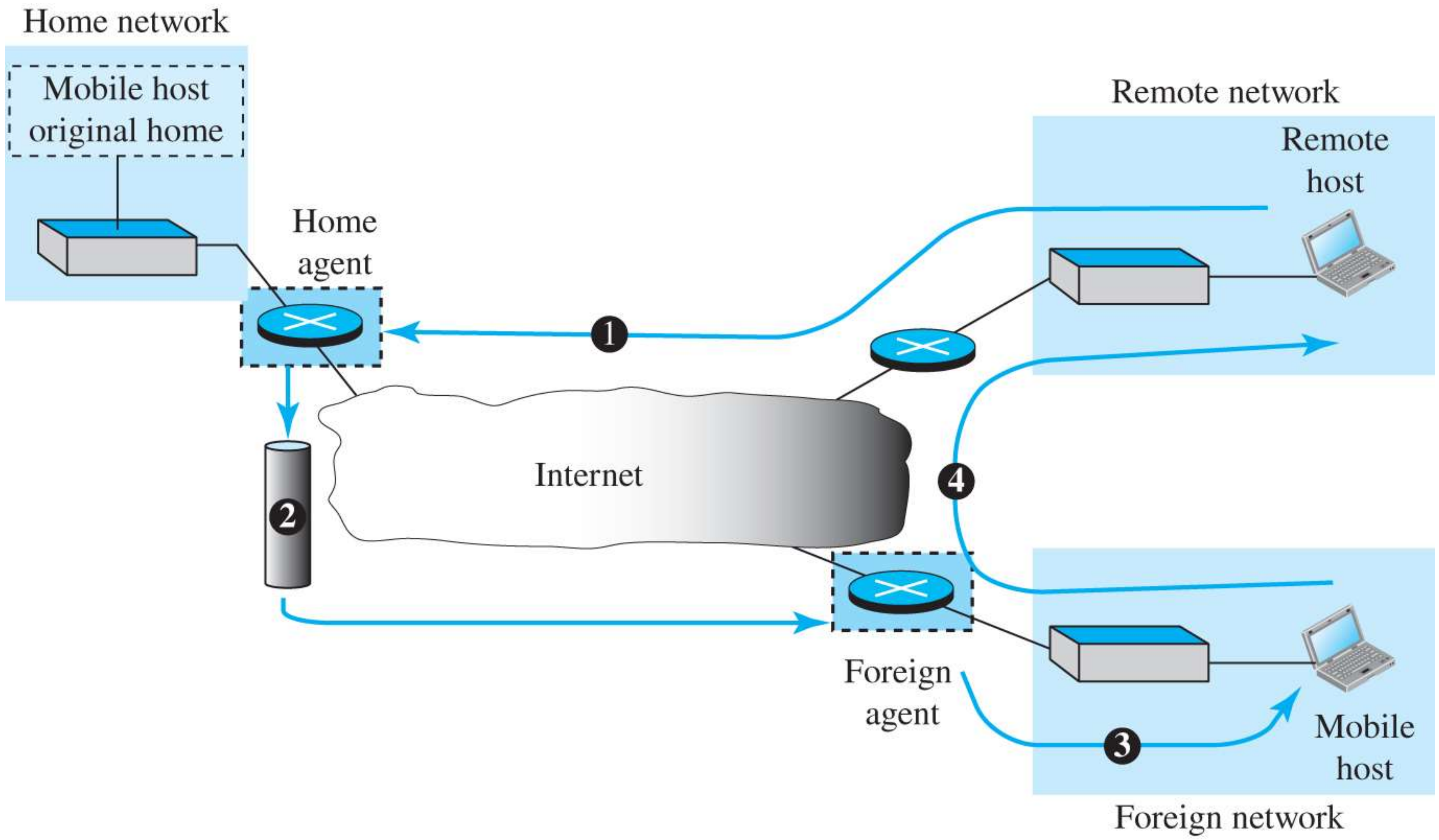

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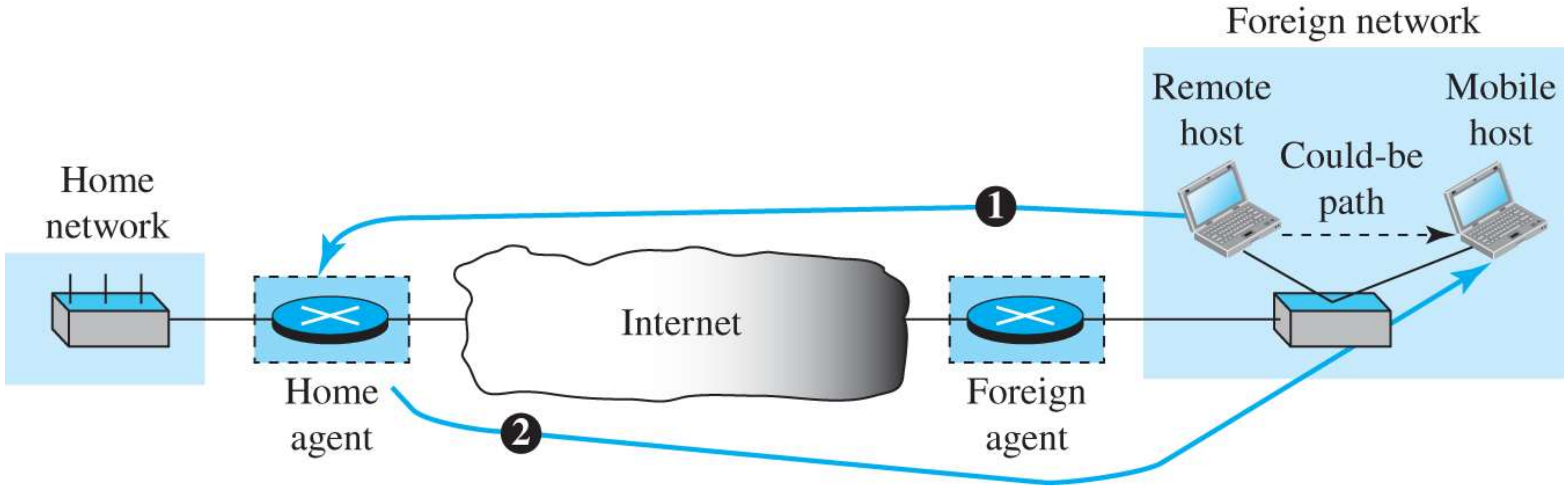
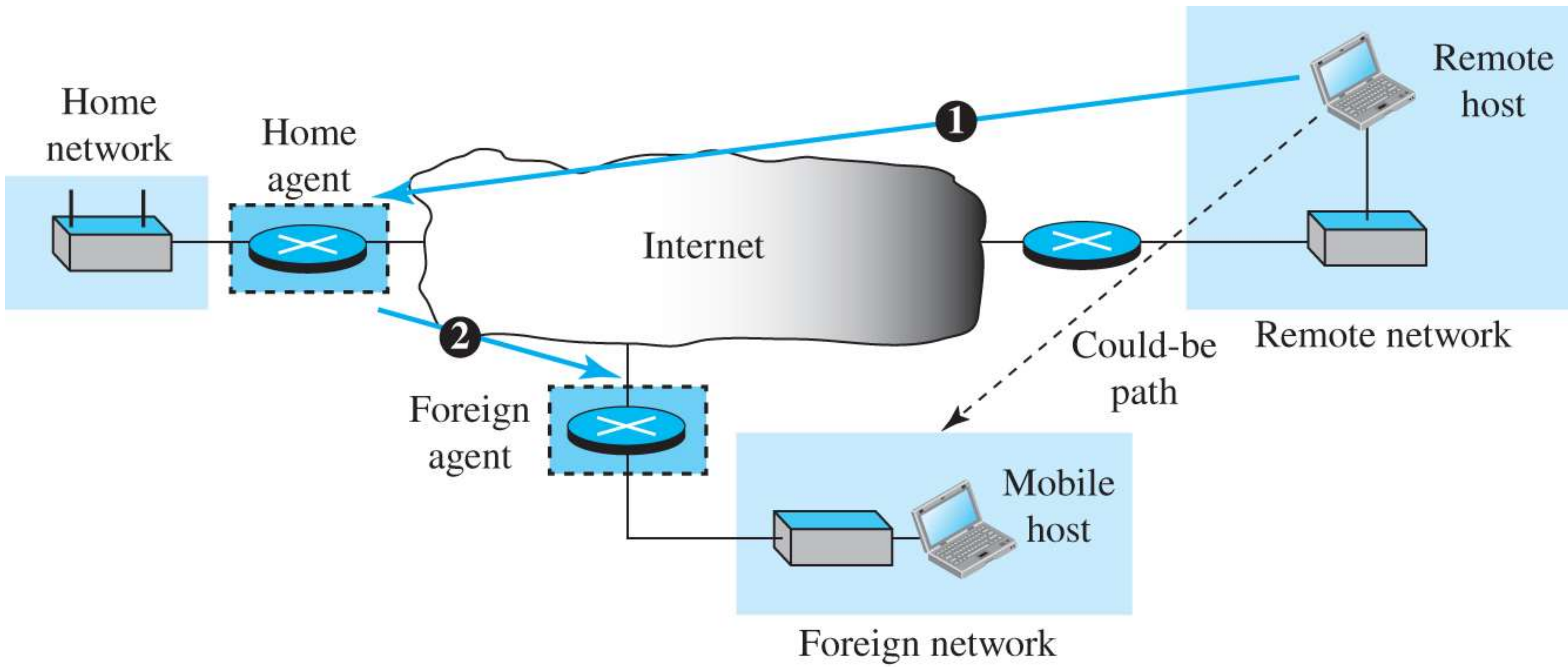
Figure 7.30 Double crossing



[Access the text alternative for slide images.]

Figure 7.31 Triangle routing



Access the text alternative for slide images.

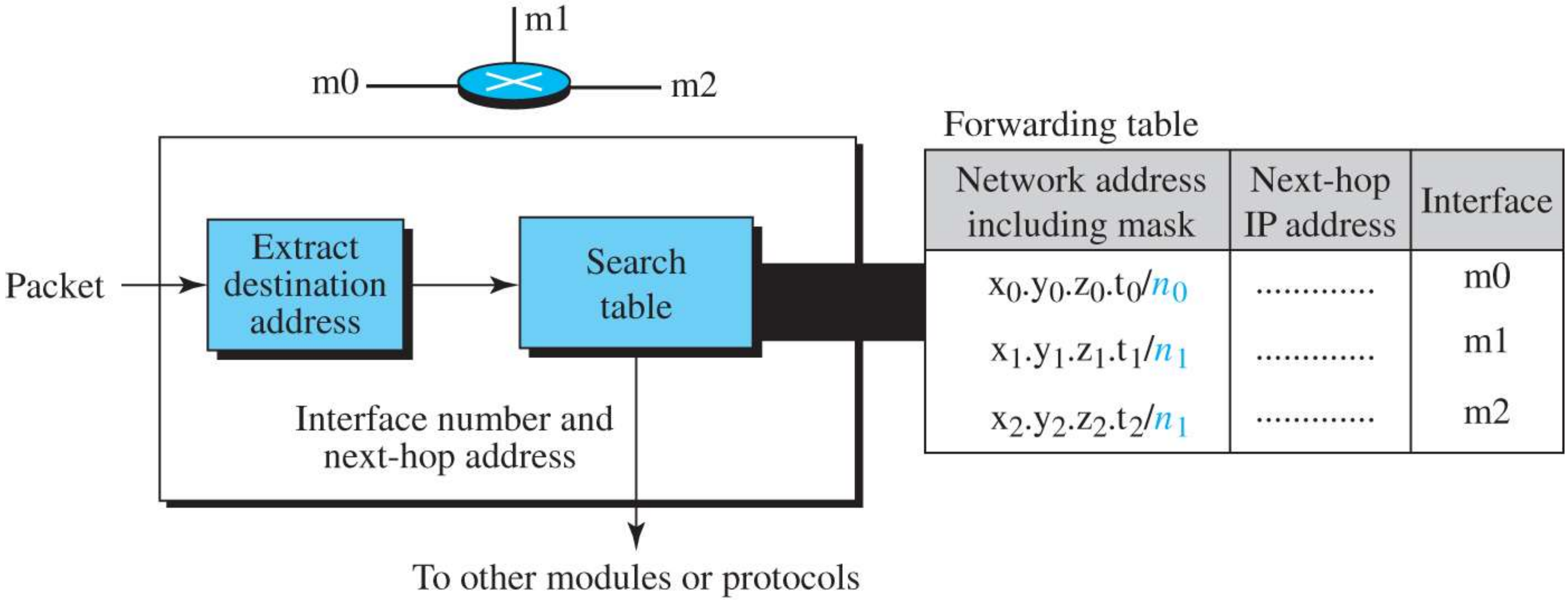
7.4.6 Forwarding of IP Packets

We discussed the concept of forwarding at the network layer earlier in this chapter. In this section, we extend the concept to include the role of IP addresses in forwarding. As we discussed before, forwarding means to place the packet in its route to its destination. Since the Internet today is made of a combination of links (networks), forwarding means to deliver the packet to the next hop (which can be the final destination or the intermediate connecting device). Although the IP protocol was originally designed as a connectionless protocol, today the tendency is to change it to connection-oriented protocol. We discuss both cases.

Forwarding Based on Destination Address

We first discuss forwarding based on the destination address. This is a traditional approach, which is prevalent today. In this case, forwarding requires a host or a router to have a forwarding table. When a host has a packet to send or when a router has received a packet to be forwarded, it looks at this table to find the next hop to deliver the packet to.

Figure 7.32 Simplified forwarding module in classless address



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Example 7.19

Make a forwarding table for router R1 using the configuration in Figure 7.33.

Figure 7.33 Configuration for Example 7.19

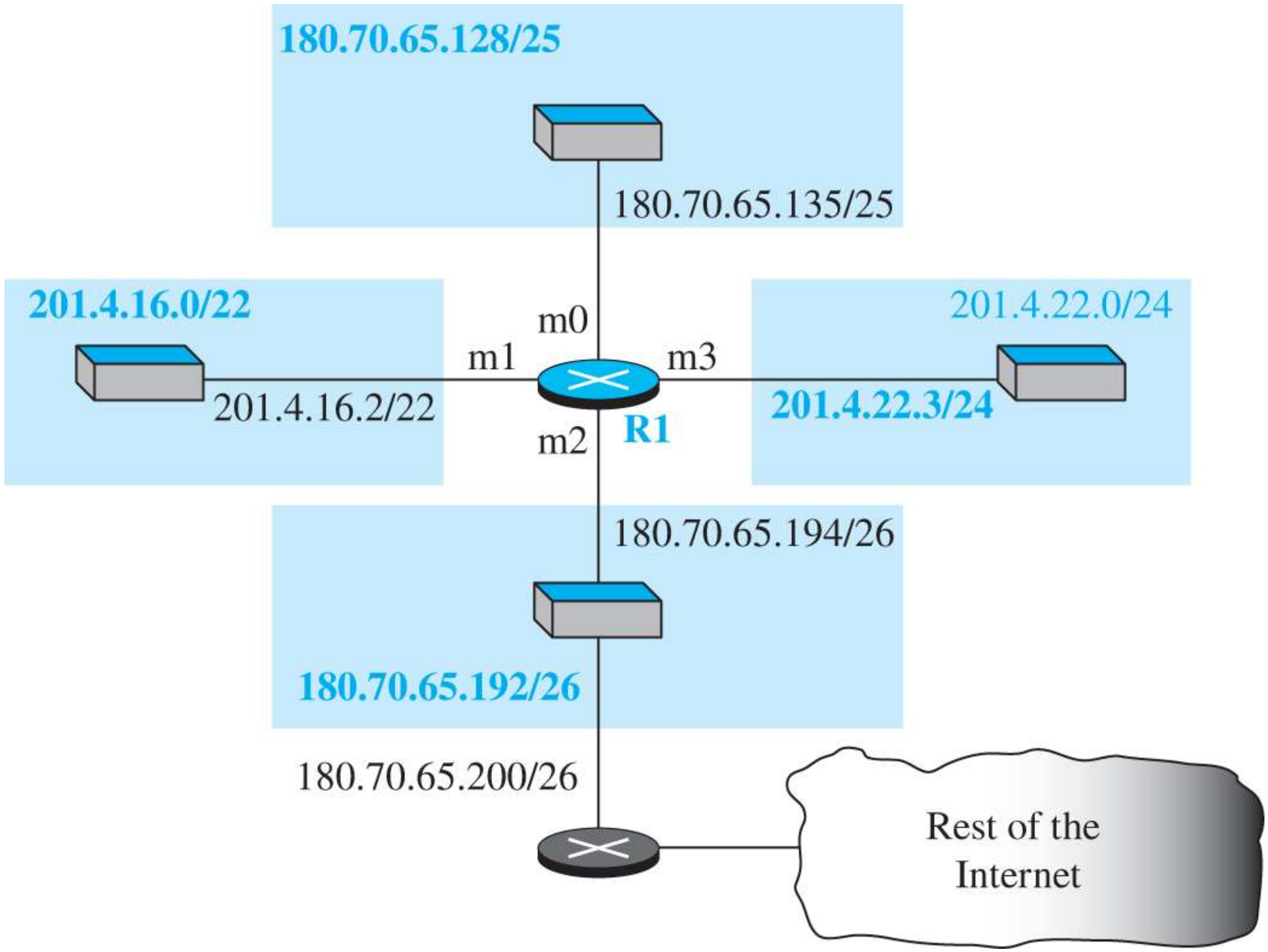

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Table 7.3 Forwarding table for router R1

Network address/mask	Next hop	Interface
180.70.65.192/**26**	—	m2
180.70.65.128/**25**	—	m0
201.4.22.0/**24**	—	m3
201.4.16.0/**22**	—	m1
Default	180.70.65.200	m2

Example 7.20

Instead of Table 7.3, we can use Table 7.4, in which the network address/mask is given in bits.

Table 7.4 Forwarding table for router R1 using prefix bit

Leftmost bits in the destination address	*Next hop*	*Interface*
10110100 01000110 01000001 11	—	m2
10110100 01000110 01000001 1	—	m0
11001001 00000100 00011100	—	m3
11001001 00000100 000100	—	m1
Default	180.70.65.200	m2

Example 7.21

The router performs the following steps

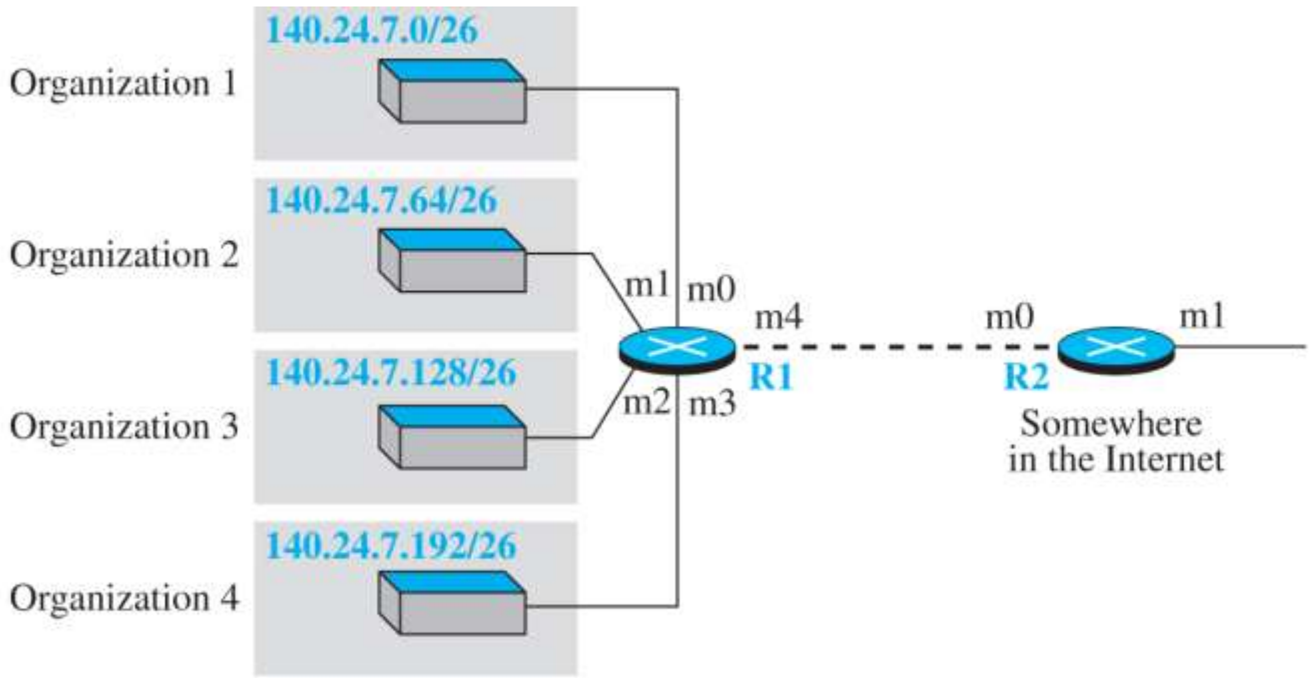
1. The first mask (/26) is applied to the destination address. The result is 180.70.65.128, which does not match the corresponding network address.

2. The second mask (/25) is applied to the destination address. The result is 180.70.65.128, which matches the corresponding network address. The next-hop address and the interface number m0 are extracted for forwarding the packet.

Address Aggregation

When we use classful addressing, there is only one entry in the forwarding table for each site outside the organization. The entry defines the site even if that site is subnetted. When a packet arrives at the router, the router checks the corresponding entry and forwards the packet accordingly. When we use classless addressing, it is likely that the number of forwarding table entries will increase. To alleviate the problem, the idea of address aggregation was designed. In Figure 7.34 we have two routers.

Figure 7.34 Address aggregation



Forwarding table for R1

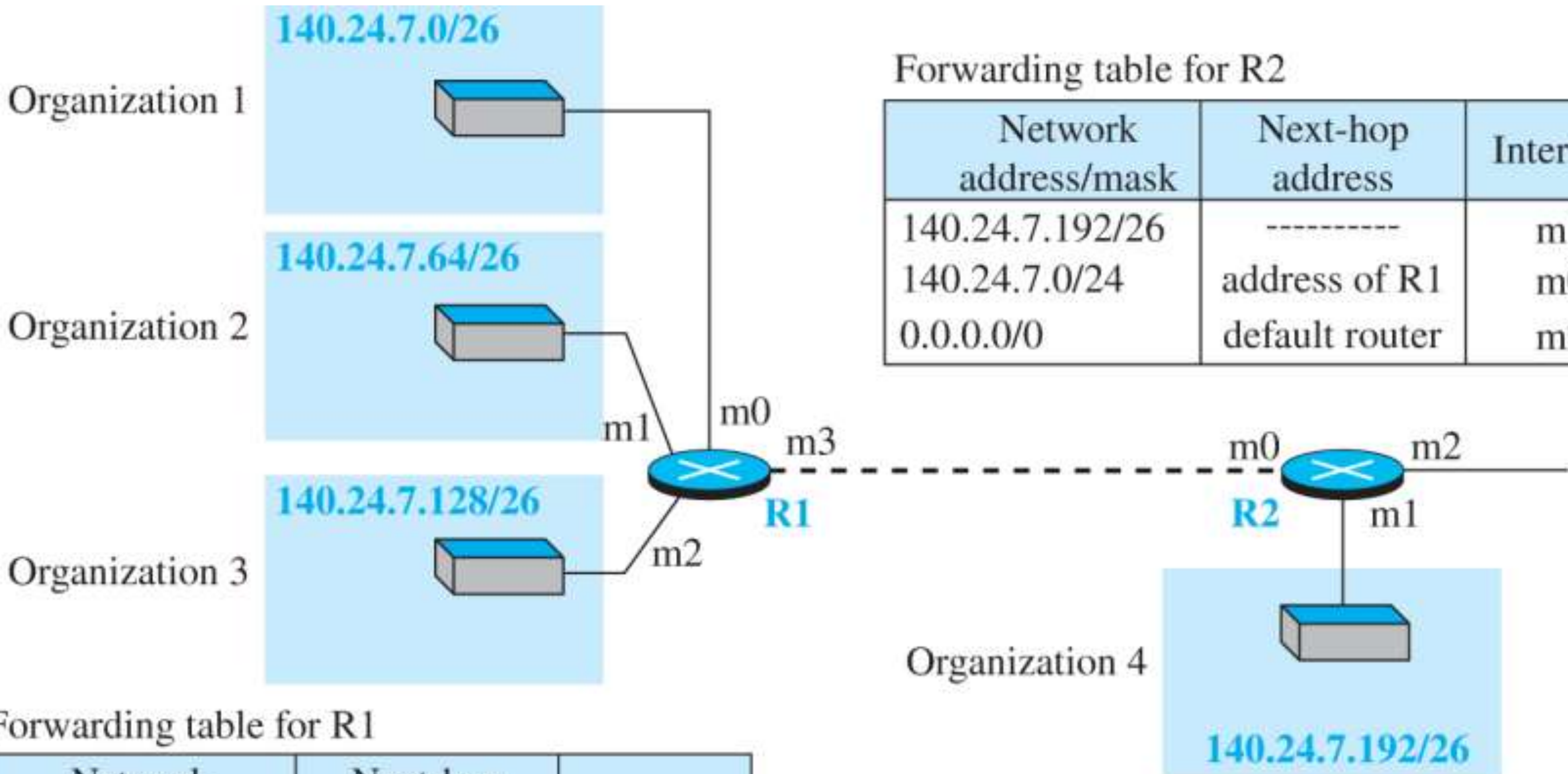
Network address/mask	Next-hop address	Interface
140.24.7.0/26	----------	m0
140.24.7.64/26	----------	m1
140.24.7.128/26	----------	m2
140.24.7.192/26	----------	m3
0.0.0.0/0	address of R2	m4

Forwarding table for R2

Network address/mask	Next-hop address	Interface
140.24.7.0/24	----------	m0
0.0.0.0/0	default router	m1

Access the text alternative for slide images.

Figure 7.35 Longest mask addressing



Forwarding table for R2

Network address/mask	Next-hop address	Interface
140.24.7.192/26	----------	m1
140.24.7.0/24	address of R1	m0
0.0.0.0/0	default router	m2

Forwarding table for R1

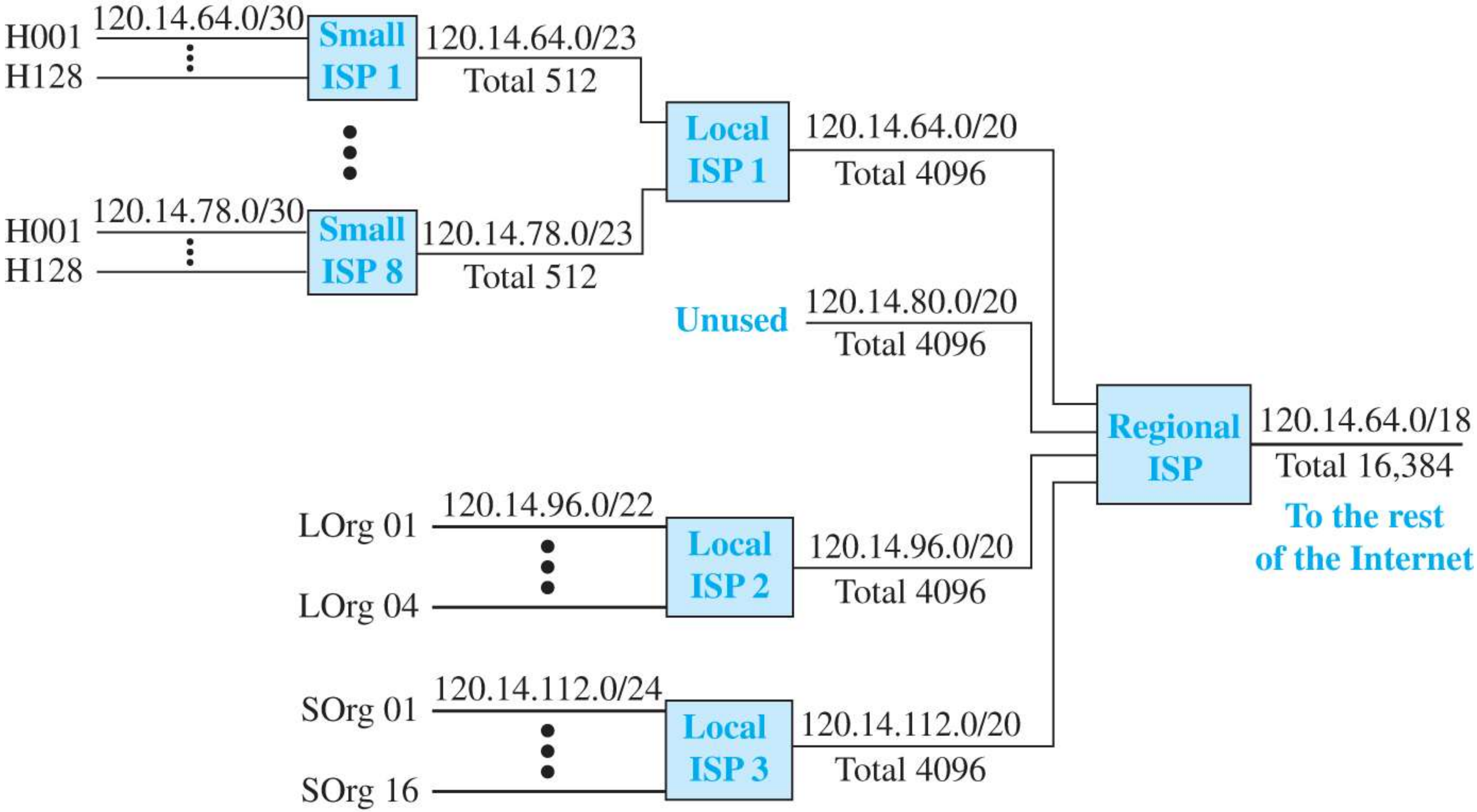
Network address/mask	Next-hop address	Interface
140.24.7.0/26	----------	m0
140.24.7.64/26	----------	m1
140.24.7.128/26	----------	m2
0.0.0.0/0	default router	m3

Access the text alternative for slide images.

Example 7.22

As an example of hierarchical routing, let us consider . A regional ISP is granted 16,384 addresses starting from 120.14.64.0. The regional ISP has decided to divide this block into 4 subblocks, each with 4096 addresses. Three of these subblocks are assigned to three local ISPs, the second subblock is reserved for future use. Note that the mask for each block is /20 because the original block with mask /18 is divided into 4 blocks.

Figure 7.36 Hierarchical routing with ISPs



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Forwarding Table Search Algorithm

In classless addressing, there is no network information in the destination address. The simplest, but not the most efficient, search method is called the longest prefix match (as we discussed before). The forwarding table can be divided into buckets, one for each prefix. The router first tries the longest prefix. If the destination address is found in this bucket, the search is complete. If the address is not found, the next prefix is searched, and so on. It is obvious that this type of search takes a long time.

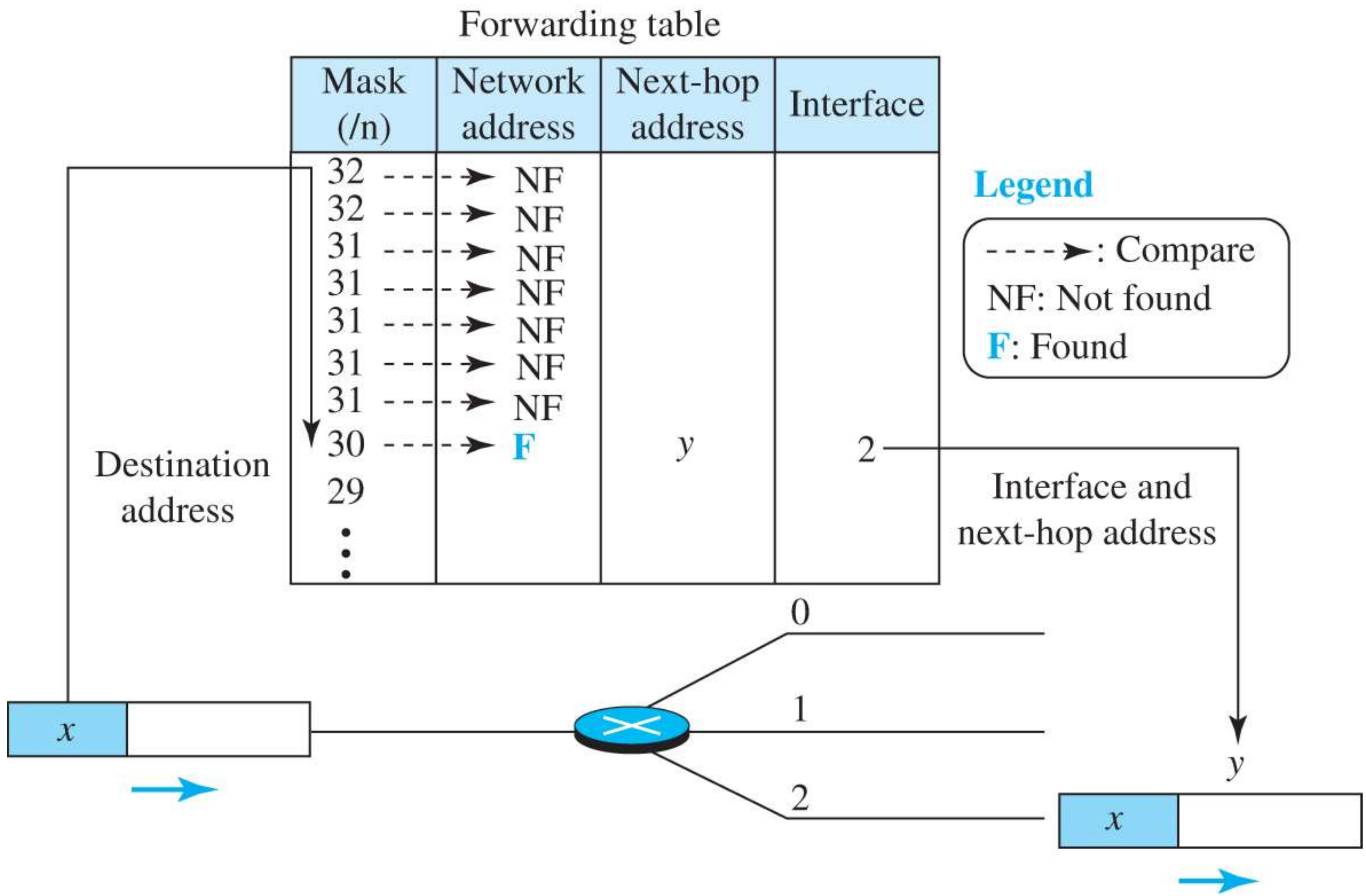
Forwarding Based on Label

In the 1980s, an effort started to somehow change IP to behave like a connection-oriented protocol in which the routing is replaced by switching. As we discussed earlier in the chapter, in a connectionless network (datagram approach), a router forwards a packet based on the destination address in the header of the packet. On the other hand, in a connection-oriented network (virtual-circuit approach), a switch forwards a packet based on the label attached to the packet. Routing is normally based on searching the contents of a table; switching can be done by accessing a table using an index. In other words, routing involves searching; switching involves accessing.

Example 7.23

Figure 7.37 shows a simple example of searching in a forwarding table using the longest mask algorithm. Although there are some more efficient algorithms today, the principle is the same.

Figure 7.37 Example 7.23

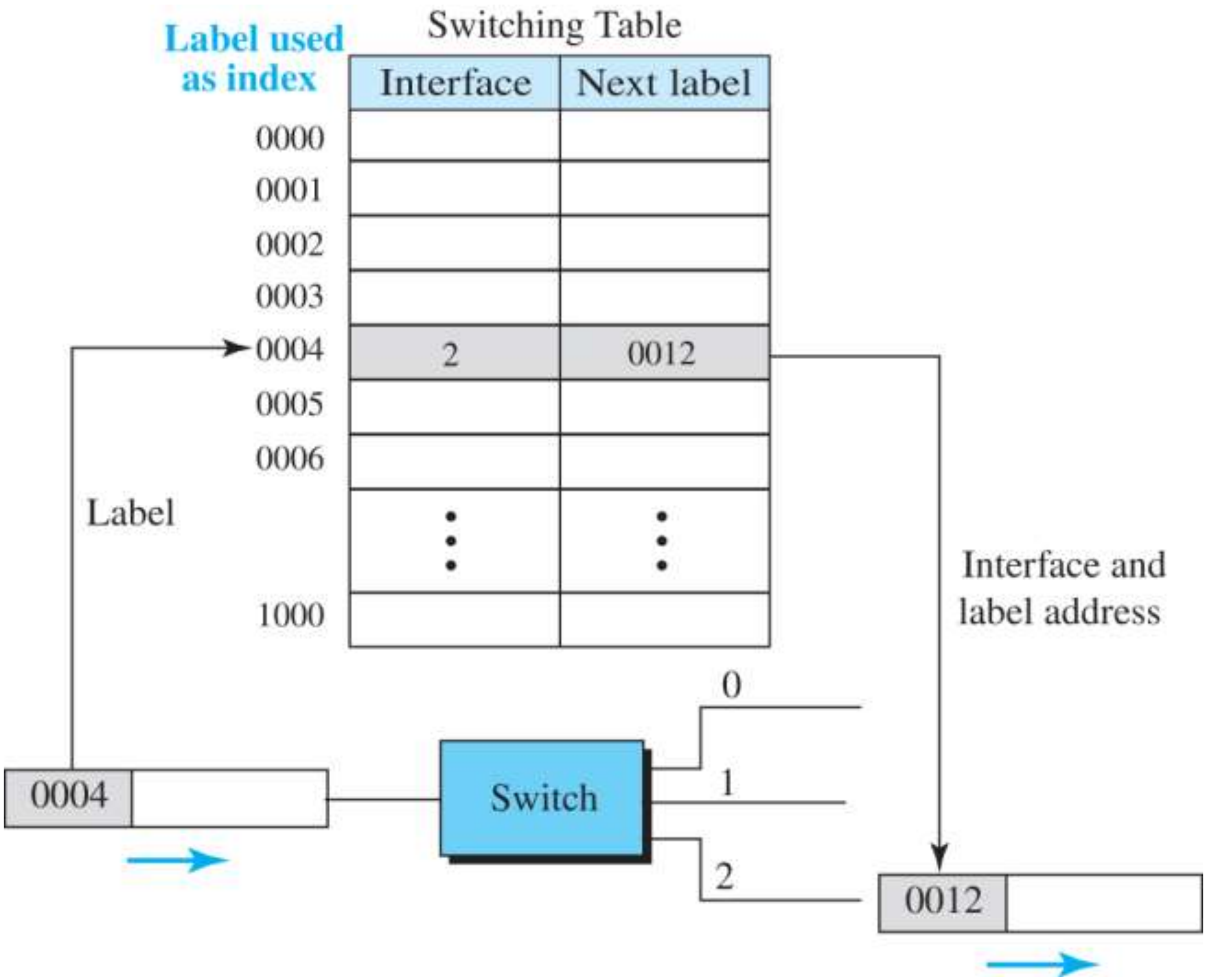


Access the text alternative for slide images.

Example 7.24

Figure 7.38 shows a simple example of using a label to access a switching table. Since the labels are used as the index to the table, finding the information in the table is immediate.

Figure 7.38 Example 7.24



Access the text alternative for slide images.

A New Header

- *To simulate connection-oriented switching using a protocol like IP, the first thing that is needed is to add a field to the packet that carries the label discussed later. The IPv4 packet format does not allow this extension The solution is to encapsulate the IPv4 packet in an MPLS packet. Figure 7.39 shows the encapsulation.*
- *The MPLS header is actually a stack of subheaders that is used for multilevel hierarchical switching as we will discuss shortly. Figure 7.40 shows the format of an MPLS header in which each subheader is 32 bits (4 bytes) long.*

Figure 7.39 MPLS header added to an IP packet

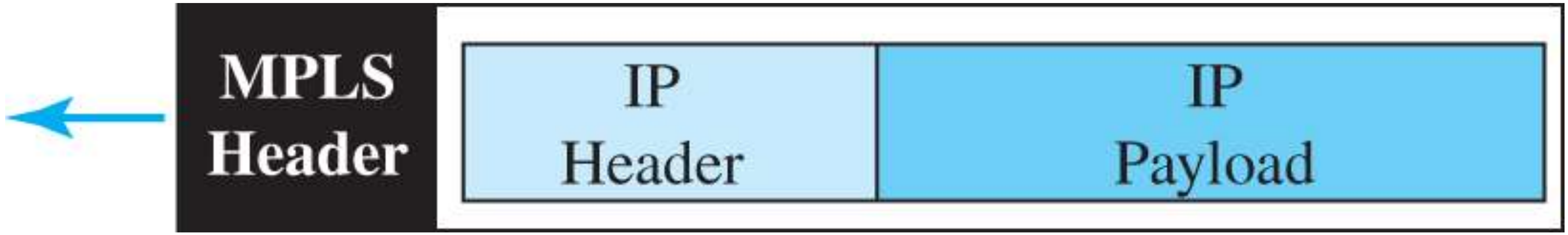


Figure 7.40 MPLS header made of a stack of labels

Access the text alternative for slide images.

Hierarchical Switching

A stack of labels in MPLS allows hierarchical switching. This is similar to conventional hierarchical routing. For example, a packet with two labels can use the top label to forward the packet through switches outside an organization; the bottom label can be used to route the packet inside the organization to reach the destination subnet.

7.4.7 Routers as Packet Switches

A stack of labels in MPLS allows hierarchical switching. This is similar to conventional hierarchical routing. For example, a packet with two labels can use the top label to forward the packet through switches outside an organization; the bottom label can be used to route the packet inside the organization to reach the destination subnet.

7.5 NEXT GENERATION IP (IPv6)

The address depletion of IPv4 and other shortcoming of this protocol prompted a new version of IP protocol in the early 1990s, which is called Internet Protocol version 6 (IPv6) or IP new generation (Ipng).

7.5.1 IPv6 Addressing

The main reason for migration from IPv4 to IPv6 was the small size of the address space of IPv4. An IPv6 address is 128 bytes or 16 bytes, four times the address length in IPv4.

Representation

An IPv6 address is 128 bits or 16 bytes long; four times the address length of IPv4.

Binary (128 bits)	**1111111011110110 1011 … 1111111100000000**
Colon hexadecimal	***FEF6:BA98:7654:3210:ADEF:BBFF:2922:FF00***

Address Space 2

The address space of IPv6 contains 2^{128} addresses. This address space is 2^{96} times the IPv4 address—definitely no address depletion—as shown, the size of the space is

340, 282, 366, 920, 938, 463, 374, 607, 431, 768, 211, 456

Address Space Allocation

Like the address space of IPv4, the address space of IPv6 is divided into several blocks of varying size and each block is allocated for a special purpose. Most of the blocks are still unassigned and have been set aside for future use. Table 7.5 shows only the assigned blocks. In this table, the last column shows the fraction each block occupies in the whole address space.

Table 7.5 Prefixes for assigned IPv6 addresses

Block prefix	*CIDR*	*Block assignment*	*Fraction*
0000 0000	0000::/8	Special addresses	1/256
001	**2000::/3**	**Global unicast**	**1/8**
1111 110	FC00::/7	Unique local unicast	1/128
1111 1110 10	FE80::/10	Link local addresses	1/1024
1111 1111	FF00::/8	Multicast addresses	1/256

Figure 7.41 Global unicast address

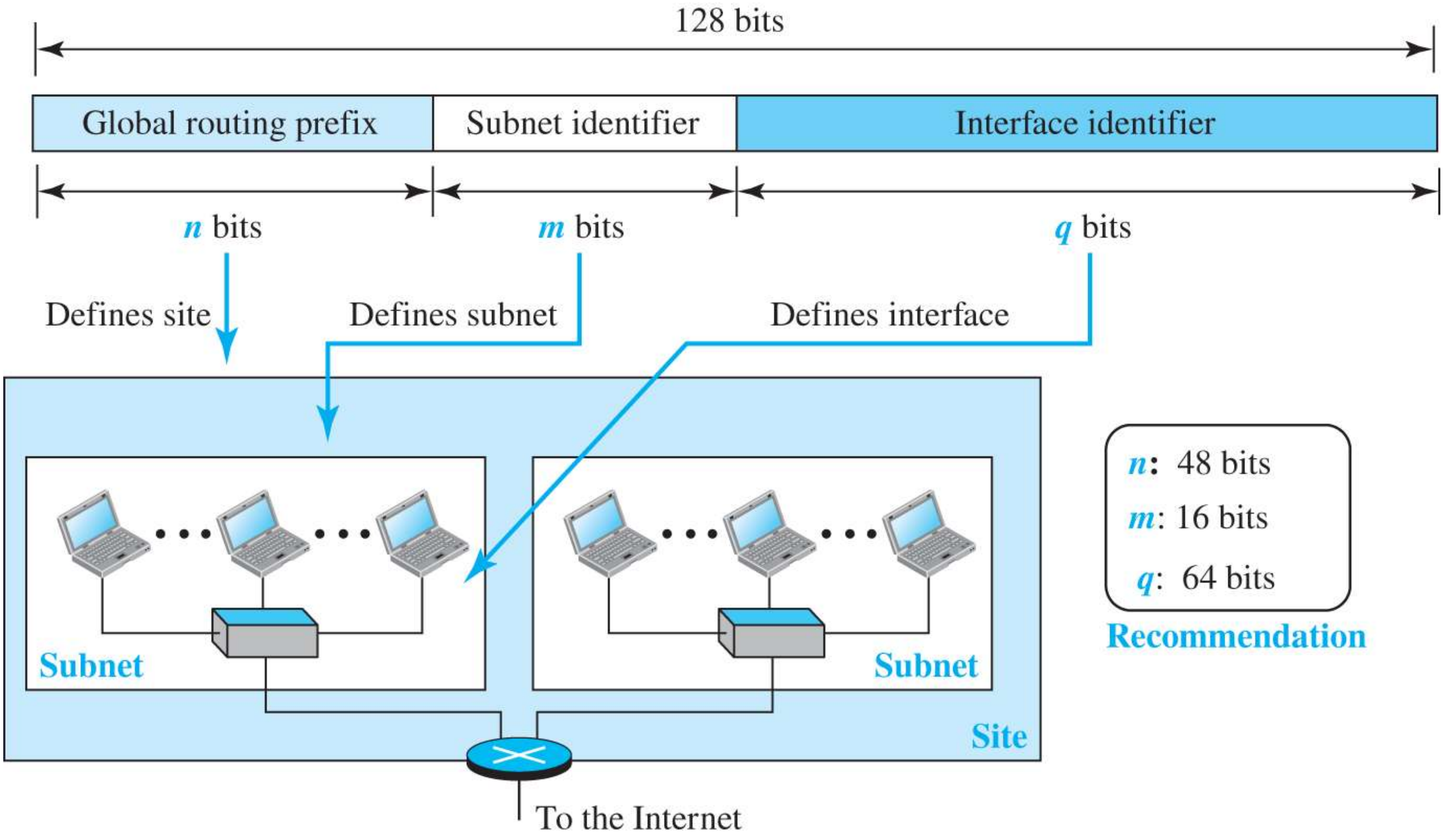


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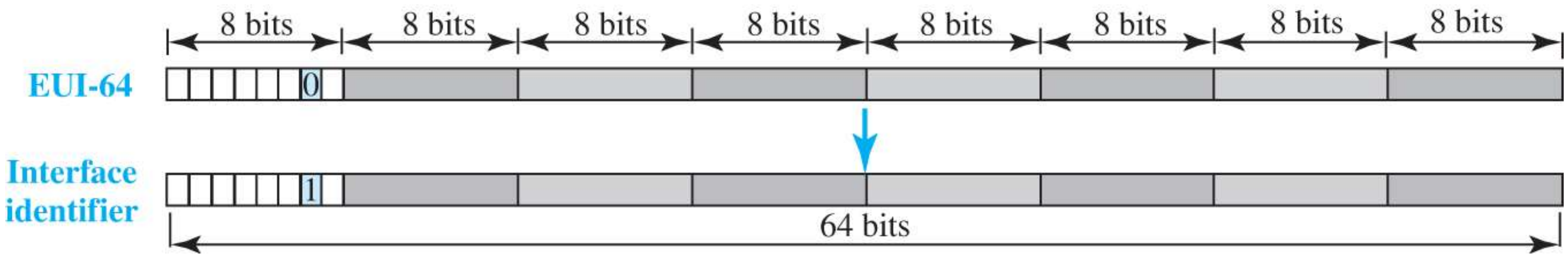
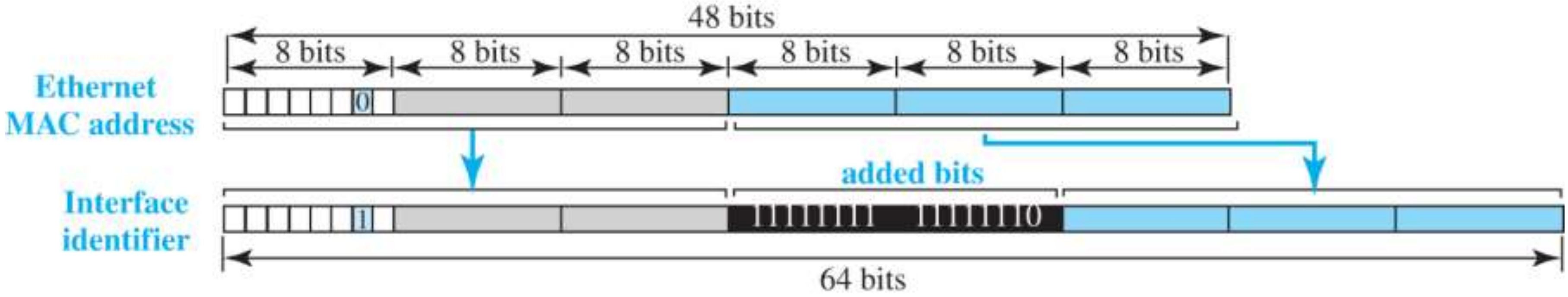
Figure 7.42 Mapping EUI-64



Access the text alternative for slide images.

Figure 7.43 Mapping for Ethernet MAC



Access the text alternative for slide images.

Example 7.25

An organization is assigned the block 2000:1456:2474/48. What is the CIDR notation for the blocks in the first and second subnets in this organization.

Solution

Theoretically, the first and second subnets should use the block with subnet identifier 0001_{16} and 0002_{16}. This means that the blocks are

2000:1456:2474:0000/64

and

2000:1456:2474:0001/64.

Example 7.26

Find the interface identifier if the physical address in the EUI is (**F5-A9-23-EF-07-14-7A-D2**)$_{16}$ using the format we defined for Ethernet addresses.

Solution

We only need to change the seventh bit of the first octet from 0 to 1 and change the format to colon hex notation. The result is **F7A9:23EF:0714:7AD2.**

Example 7.27

Find the interface identifier if the Ethernet physical address is (**F5-A9-23-14-7A-D2**)$_{16}$ using the format we defined for Ethernet addresses.

Solution

We only need to change the seventh bit of the first octet from 0 to 1, insert two octet $FFFE_{16}$ and change the format to colon hex notation. The result is **F7A9:23FF:FE14:7AD2** in colon hex.

Example 7.28

An organization is assigned the block 2000:1456:2474/48. What is the IPv6 address of an interface in the third subnet if the IEEE physical address of the computer is

$$\mathbf{(F5\text{-}A9\text{-}23\text{-}14\text{-}7A\text{-}D2)}_{16}\,.$$

Solution

The interface identifier for this interface is

F7A9:23FF:FE14:7AD2.

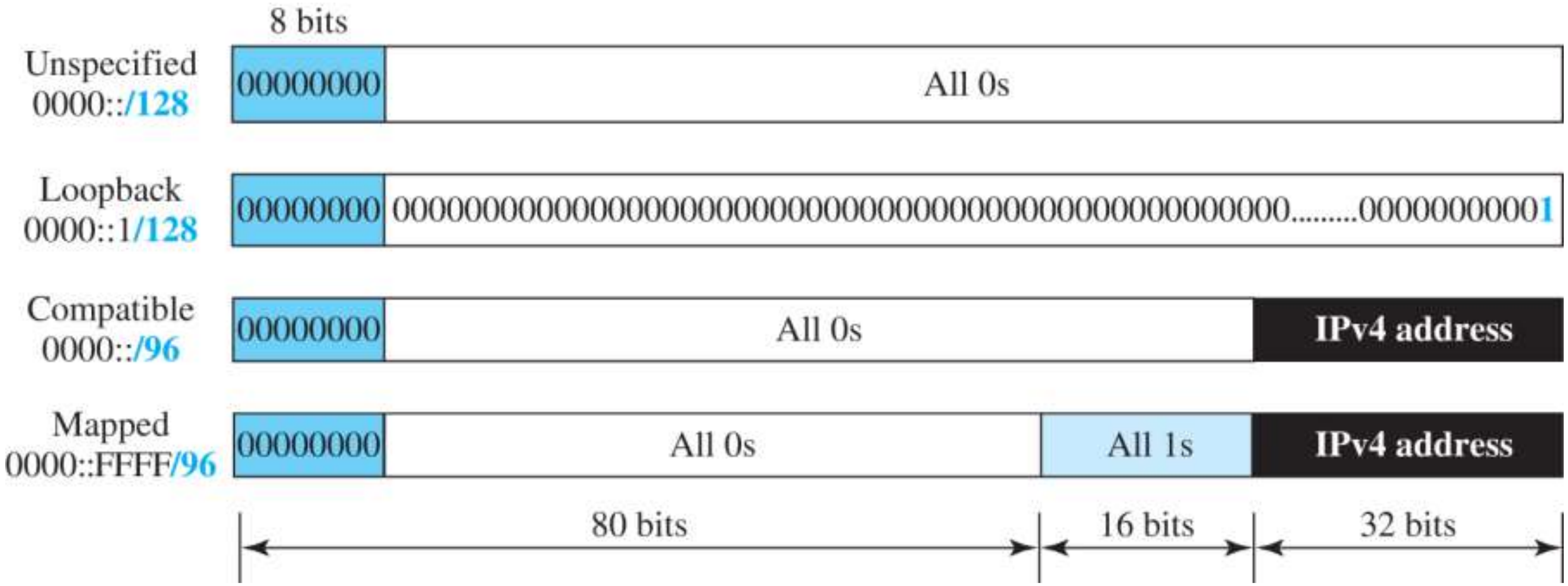
If we assign this identifier to the global prefix and the subnet identifier, we get

2000:1456:2474:0003:F7A9:23FF:FE14:7AD2/128

Special Addresses

After discussing the global unicast block, let us discuss the characteristics and purposes of assigned and reserved blocks in the first row of Table 7.5. Addresses that use the prefix (0000::/8) are reserved, but part of this block is used to define some special addresses. Figure 7.44 shows the assigned addresses in this block.

Figure 7.44 Special addresses

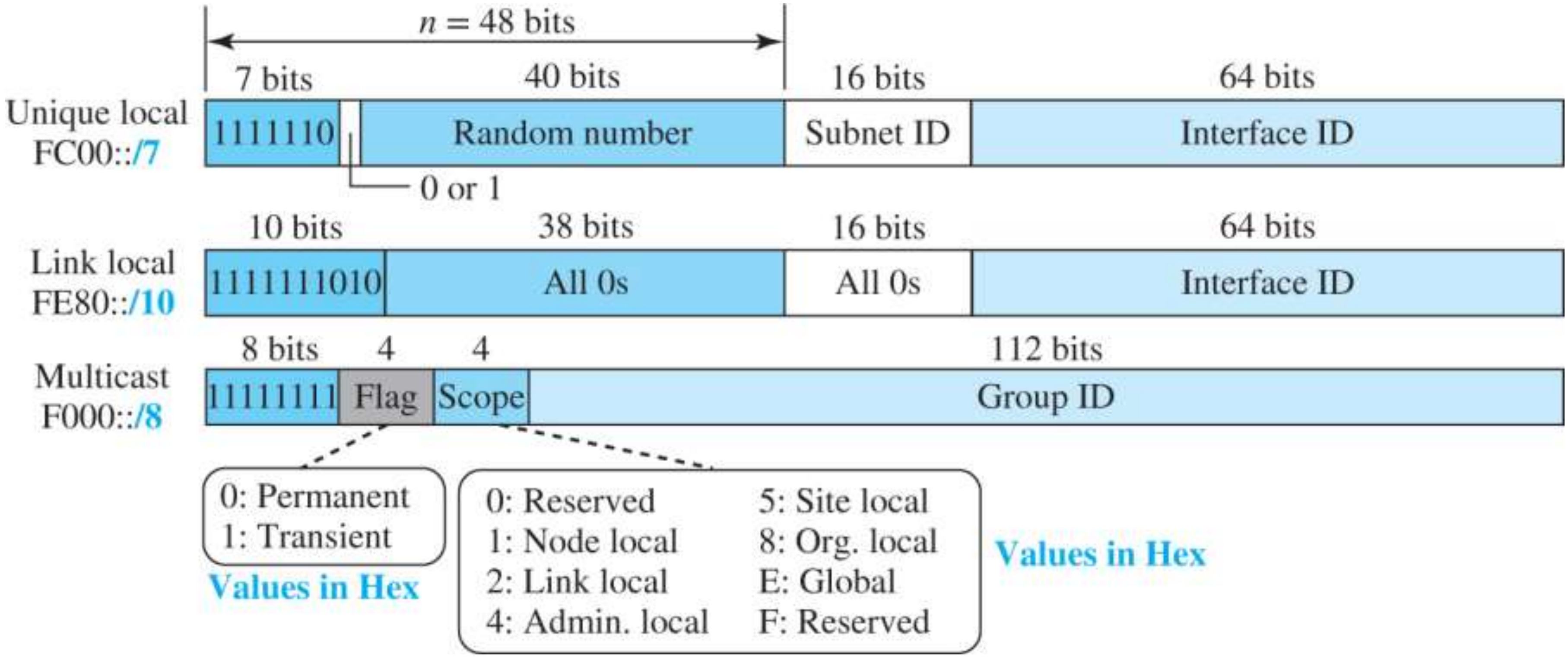


[Access the text alternative for slide images.]

Other Assigned Blocks

IPv6 uses two large blocks for private addressing and one large block for multicasting, as shown in Figure 7.45.

Figure 7.45 Unique local unicast block



Access the text alternative for slide images.

Autoconfiguration

One of the interesting features of IPv6 addressing is the auto-configuration of hosts. As we discussed in IPv4, the host and routers are originally configured manually by the network manager. However, the Dynamic Host Configuration Protocol, DHCP, can be used to allocate an IPv4 address to a host that joins the network. In IPv6, DHCP protocol can still be used to allocate an IPv6 address to a host, but a host can also configure itself.

Example 7.29 ***(1)***

Assume a host with Ethernet address (**F5-A9-23-11-9B-E2**)$_{16}$ has joined the network. What would be its global unicast address if the global unicast prefix of the organization is 3A21:1216:2165 and the subnet identifier is A245:1232.

Solution

The host first creates its interface identifier as **F7A9:23FF:FE11:9BE2** using the Ethernet address read from its card. The host then creates its link-local address as

FE80::F7A9:23FF:FE11:9BE2

Example 7.29 (2)

Assuming that this address is unique, the host sends a router solicitation message and receives the router advertisement message that announces the combination of global unicast prefix and the subnet identifier as

3A21:1216:2165:A245:1232.

The host then appends its interface identifier to this prefix to find and store its global unicast address as:

3A21:1216:2165:A245:1232:F7A9:23FF:FE11:9BE2

Renumbering

To allow sites to change the service provider, renumbering of the address prefix (n) was built into IPv6 addressing. As we discussed before, each site is given a prefix by the service provider to which it is connected. If the site changes the provider, the address prefix needs to be changed. A router to which the site is connected can advertise a new prefix and let the site use the old prefix for a short time before disabling it. In other words, during the transition period, a site has two prefixes.

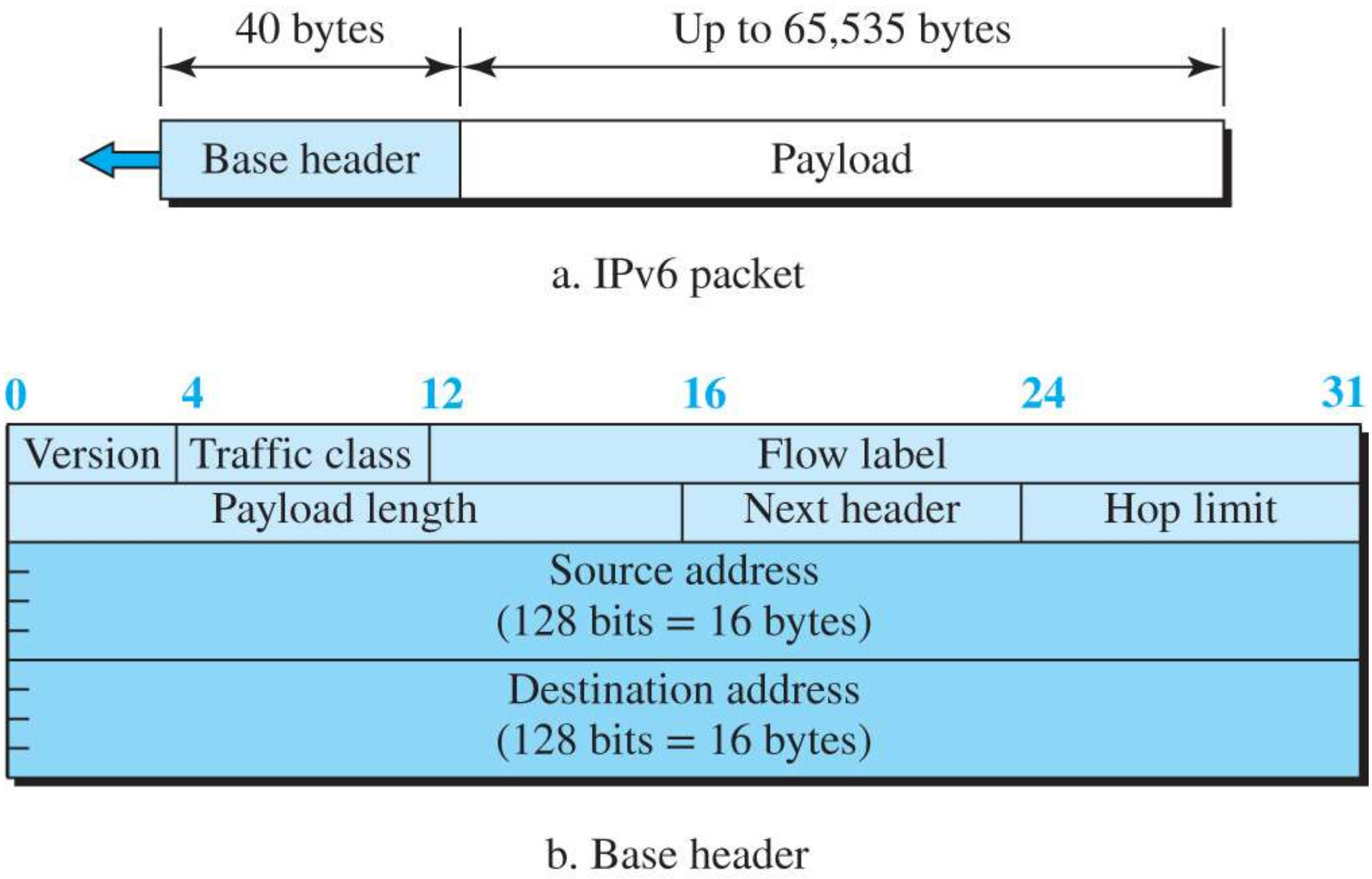
7.5.2 The IPv6 Protocol

The change of the IPv6 address size requires the change in the IPv4 packet format. The designer of IPv6 decided to implement remedies for other shortcomings now that a change is inevitable. The following shows other changes implemented in the protocol in addition to changing address size and format.

Packet Format

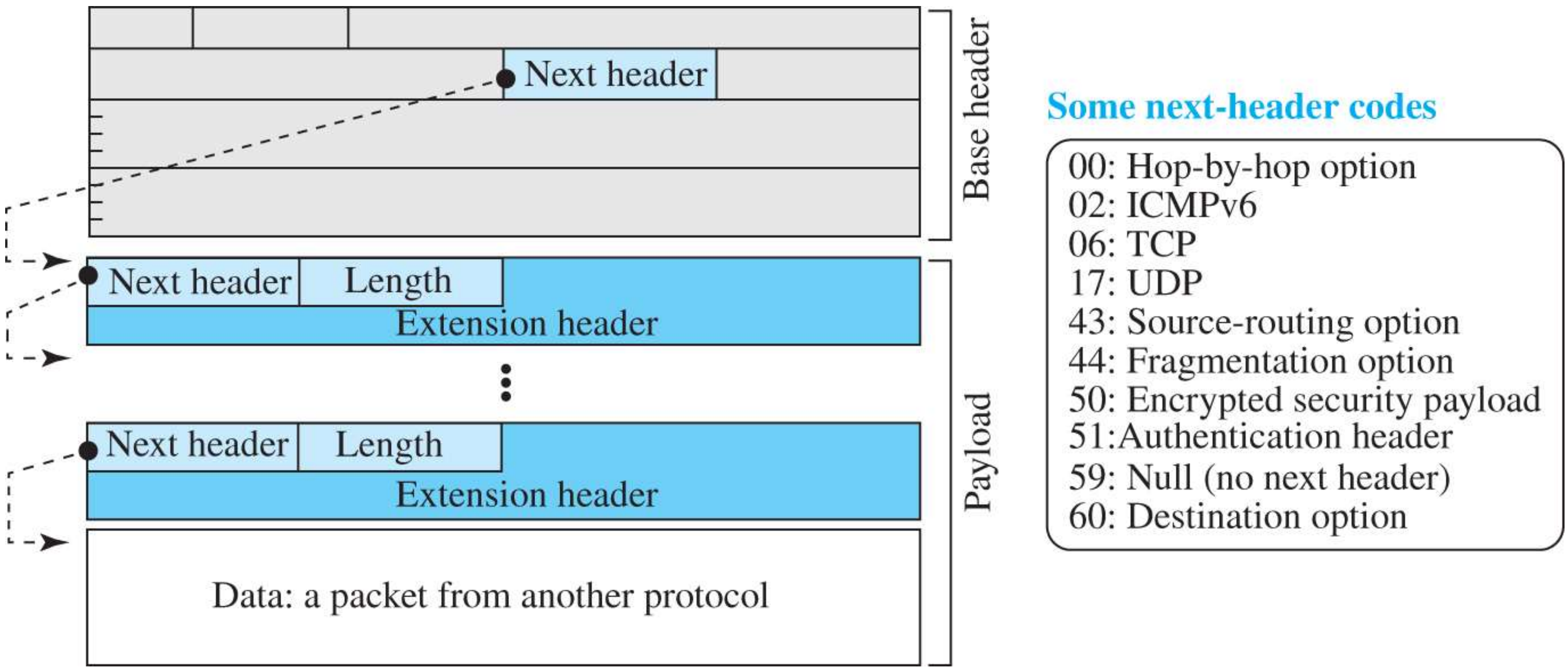
The IPv6 packet is shown in Figure 7.46. Each packet is composed of a base header followed by the payload. The base header occupies 40 bytes, whereas payload can be up to 65,535 bytes of information. The description of fields follows.

Figure 7.46 IPv6 datagram



Access the text alternative for slide images.

Figure 7.47 Payload in an IPv6 datagram



Access the text alternative for slide images.

Concept of Flow and Priority in IPv6

The IP protocol was originally designed as a connectionless protocol. However, the tendency is to use the IP protocol as a connection-oriented protocol. The MPLS technology described earlier allows us to encapsulate an IPv4 packet in an MPLS header using a label field. In version 6, the flow label has been directly added to the format of the IPv6 datagram to allow us to use IPv6 as a connection-oriented protocol.

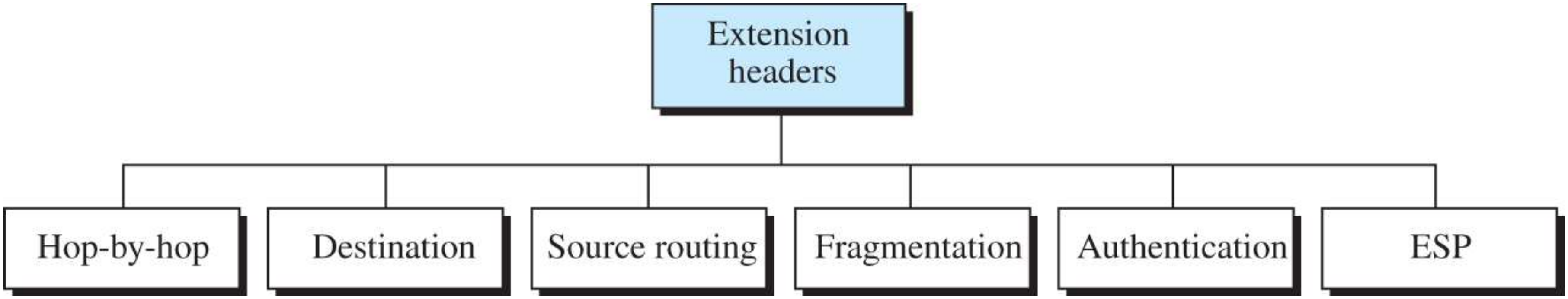
Fragmentation and Reassembly

There is still fragmentation and reassembly of datagrams in the IPv6 protocol, but there is a major difference in this respect. IPv6 datagrams can be fragmented only by the source, not by the routers; the reassembly takes place at the destination.

Extension Header

An IPv6 packet is made of a base header and some extension headers. The length of the base header is fixed at 40 bytes. However, to give more 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. These are hop-by-hop option, source routing, fragmentation, authentication, encrypted security payload, and destination option (see Figure 7.48).

Figure 7.48 Extension headers



Comparison of Options (IPv4 and IPv6) *1*

The following shows a quick comparison between the options used in IPv4 and the options used in IPv6.

- *The no-operation and end-of-option options are replaced by Pad1 and PadN.*
- *The record route option is not implemented in IPv6 because it was not used.*
- *The timestamp option is not implemented because it was not used.*
- *The source route option is called the source route extension header in IPv6.*

Comparison of Options (IPv4 and IPv6) 2

The following shows a quick comparison between the options used in IPv4 and the options used in IPv6.

- *The fragmentation fields in the base header section of IPv4 have moved to the fragmentation extension header in IPv6.*

- *The authentication extension header is new in IPv6.*

- *The encrypted security payload extension header is new in IPv6.*

7.5.3 The ICMPv6 Protocol

Another protocol that has been modified in version 6 of the TCP/IP protocol suite is ICMP. This new version, ICMPv6, follows the same strategy and purposes of version 4. ICMPv6, however, is more complicated than ICMPv4: some protocols that were independent in version 4 are now part of ICMPv6 and some new messages have been added to make it more useful.

Figure 7.49 Comparison of network layer in version 4 and version 6

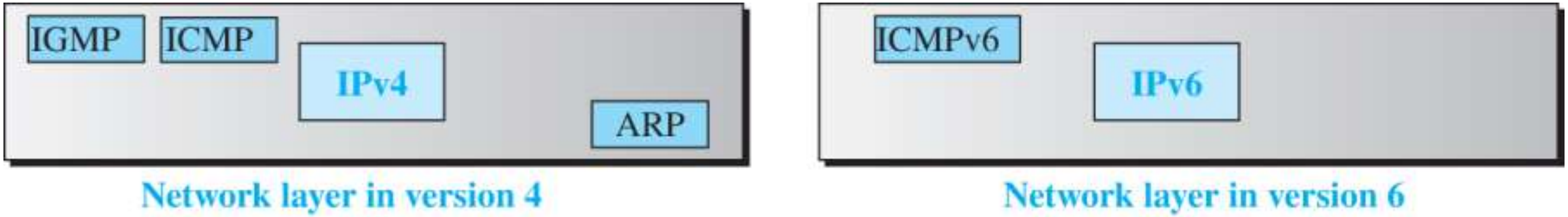


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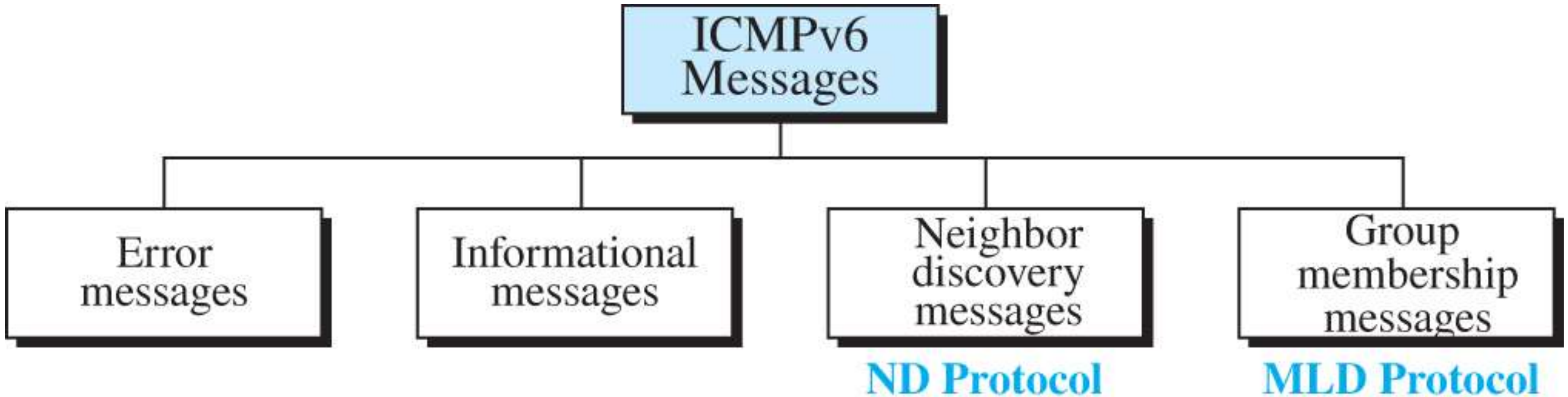
Figure 7.50 Categories of ICMPv6 messages



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Error-Reporting Messages

As we saw in our discussion of version 4, one of the main responsibilities of ICMPv6 is to report errors. Four types of errors are handled: destination unreachable, packet too big, time exceeded, and parameter problems. Note that the source-quenched message, which is used to control congestion in version 4, is eliminated in this version because the priority and flow label fields in IPv6 are supposed to take care of congestion.

Neighbor-Discovery Messages

Several messages in ICMPv4 have been redefined in ICMPv6 to handle the issue of neighbor discovery. Some new messages have also been added to provide extension. The most important issue is the definition of two new protocols that clearly define the functionality of these group messages: the Neighbor-Discovery (ND) protocol and the Inverse-Neighbor-Discovery (IND) protocol.

Group Membership Messages

The management of multicast delivery handling in IPv4 is given to the IGMPv3 protocol. In IPv6, this responsibility is given to the Multicast Listener Delivery protocol. MLDv1 is the counterpart to IGMPv2; MLDv2 is the counterpart to IGMPv3. The

material discussed in this section is taken from RFC 3810. The idea is the same as we discussed in IGMPv3, but the sizes and formats of the messages have been changed to fit the larger multicast address size in IPv6. Like IGMPv3, MLDv2 has two types of messages: membership-query message and membership-report message.

7-6 TRANSITION FROM IPv4 TO IPv6

Although we have a new version of the IP protocol, how can we make the transition to stop using IPv4 and start using IPv6? in the Internet can move The transition must be smooth to prevent any problems between IPv4 and IPv6 systems.

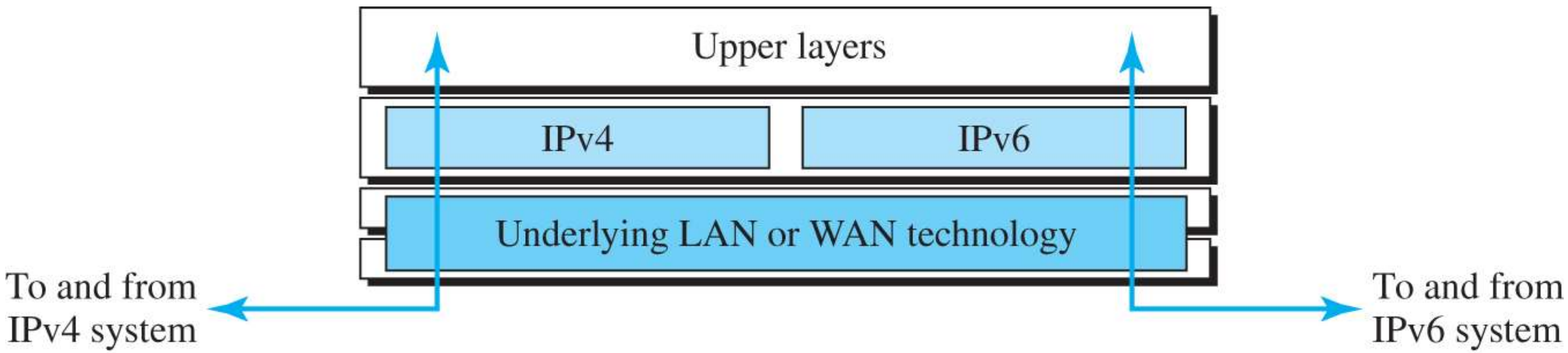
7.6.1 Strategies

Three strategies have been devised for transition: dual stack, tunneling, and header translation. One or all of these three strategies can be implemented during the transition period.

Dual Stack

It is recommended that all hosts, before migrating completely to version 6, have a dual stack of protocols during the transition. In other words, a station must run IPv4 and IPv6 simultaneously until all the Internet uses IPv6. See Figure 7.51 for the layout of a dual-stack configuration

Figure 7.51 Dual stack

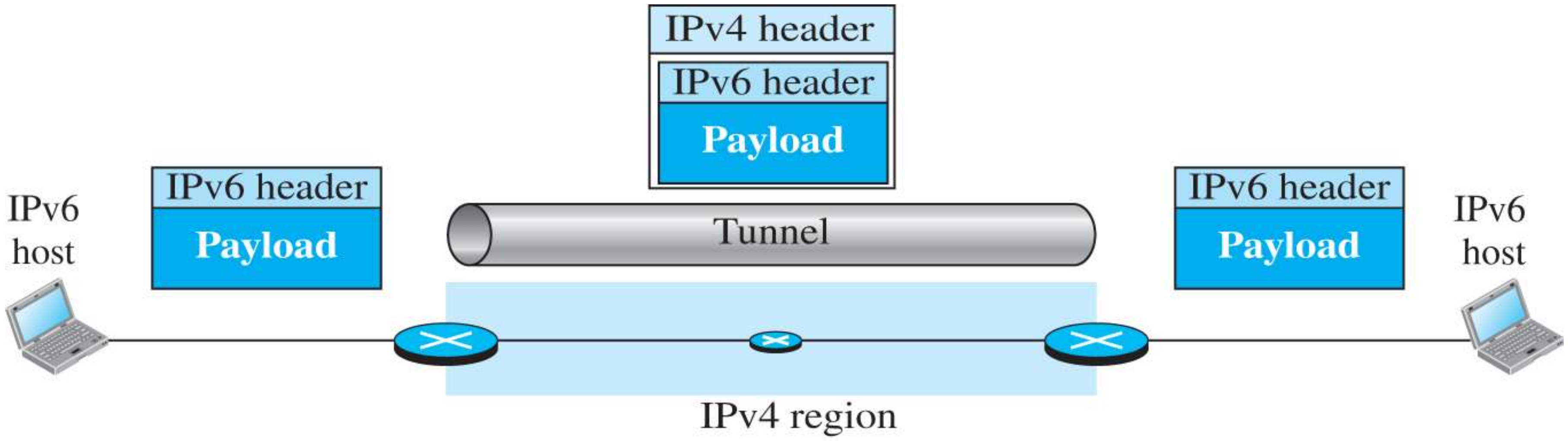


[Access the text alternative for slide images.](#)

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 7.52.

Figure 7.52 Tunneling strategy

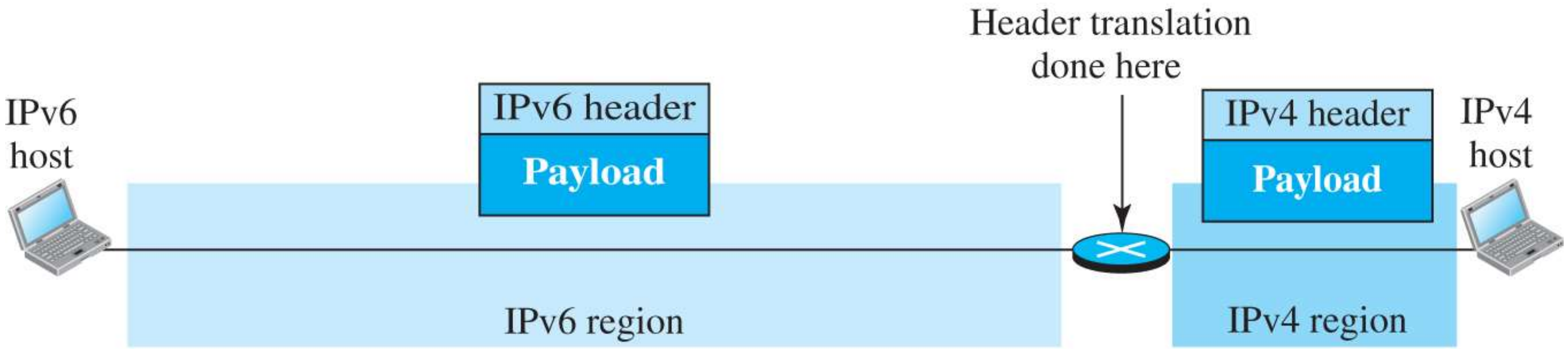


Access the text alternative for slide images.

Header Translation

Header translation is necessary when the majority of the Internet has moved to IPv6 but some systems still use IPv4. The sender wants to use IPv6, but the receiver does not understand IPv6. Tunneling does not work in this situation because the packet must be in the IPv4 format to be understood by the receiver. In this case, the header format must be totally changed through header translation. The header of the IPv6 packet is converted to an IPv4 header (see Figure 7.53).

Figure 7.53 Header translation strategy



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