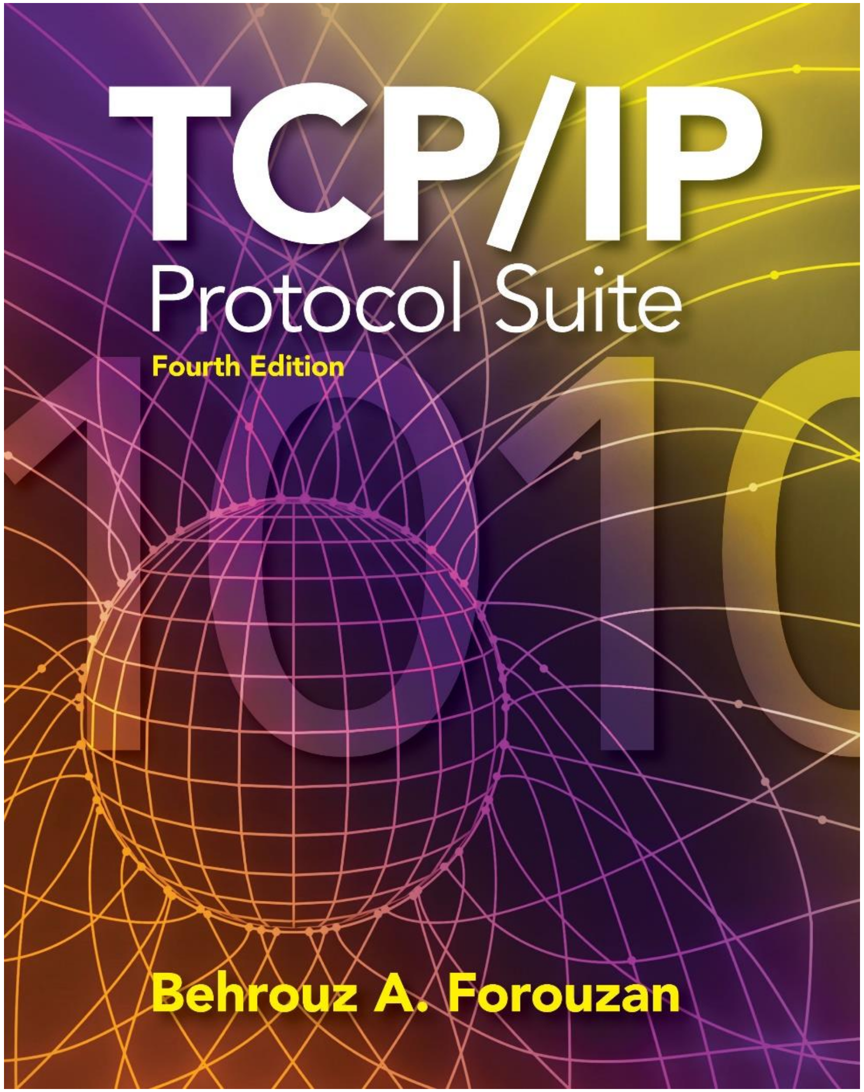
Chapter 25

Multimedia



OBJECTIVES:

- To show how audio/video files can be downloaded for future use or broadcast to clients over the Internet. The Internet can also be used for live audio/video interaction. Audio and video need to be digitized before being sent over the Internet.
- To discuss how audio and video files are compressed for transmission through the Internet.
- To discuss the phenomenon called Jitter that can be created on a packet-switched network when transmitting real-time data.
- To introduce the Real-Time Transport Protocol (RTP) and Real-Time Transport Control Protocol (RTCP) used in multimedia applications.
- To discuss voice over IP as a real-time interactive audio/video application.

OBJECTIVES (*continued*):

- To introduce the Session Initiation Protocol (SIP) as an application layer protocol that establishes, manages, and terminates multimedia sessions.
- To introduce quality of service (QoS) and how it can be improved using scheduling techniques and traffic shaping techniques.
- To discuss Integrated Services and Differential Services and how they can be implemented.
- To introduce Resource Reservation Protocol (RSVP) as a signaling protocol that helps IP create a flow and makes a resource reservation.

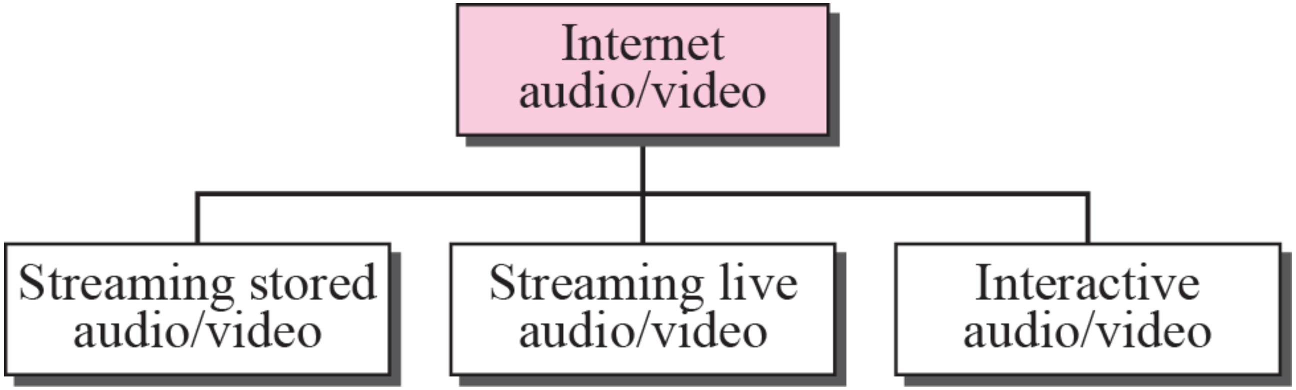
Chapter Outline

Chapter Outline (*continued*)

25-1 INTRODUCTION

We can divide audio and video services into three broad categories: streaming stored audio/video, streaming live audio/video, and interactive audio/video, as shown in Figure 25.1. Streaming means a user can listen (or watch) the file after the downloading has started.

Figure 25.1 *Internet audio/video*



Note

Streaming stored audio/video refers to on-demand requests for compressed audio/video files.

Note

Streaming live audio/video refers to the broadcasting of radio and TV programs through the Internet.

Note

Interactive audio/video refers to the use of the Internet for interactive audio/video applications.

25-2 DIGITIZING AUDIO AND VIDEO

Before audio or video signals can be sent on the Internet, they need to be digitized. We discuss audio and video separately.

Topics Discussed in the Section

- ✓ **Digitizing Audio**
- ✓ **Digitizing Video**

Note

Compression is needed to send video over the Internet.

25-3 AUDIO AND VIDEO COMPRESSION

To send audio or video over the Internet requires compression. In this section, we first discuss audio compression and then video compression.

Topics Discussed in the Section

- ✓ **Audio Compression**
- ✓ **Video Compression**

Figure 25.2 *JPEG gray scale*

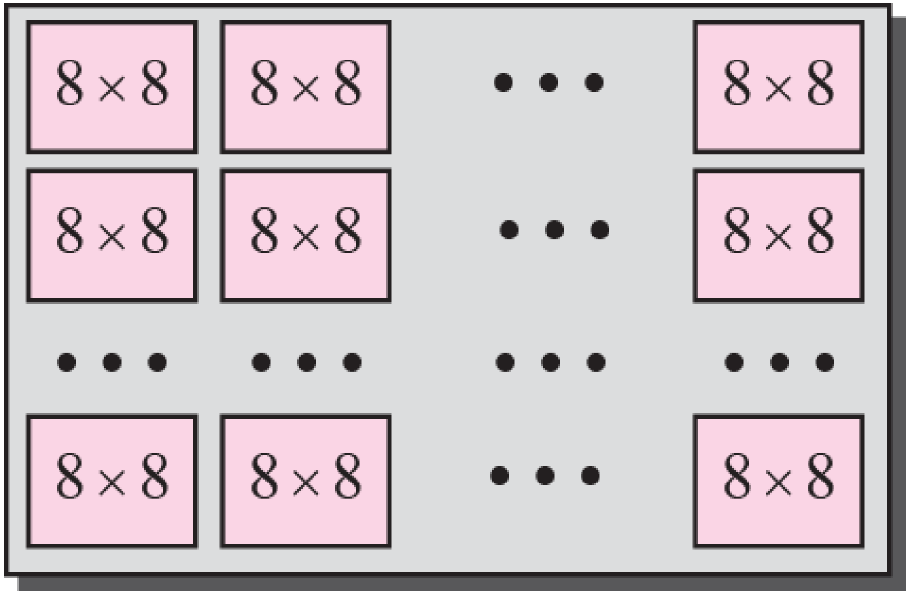


Figure 25.3 *JPEG process*

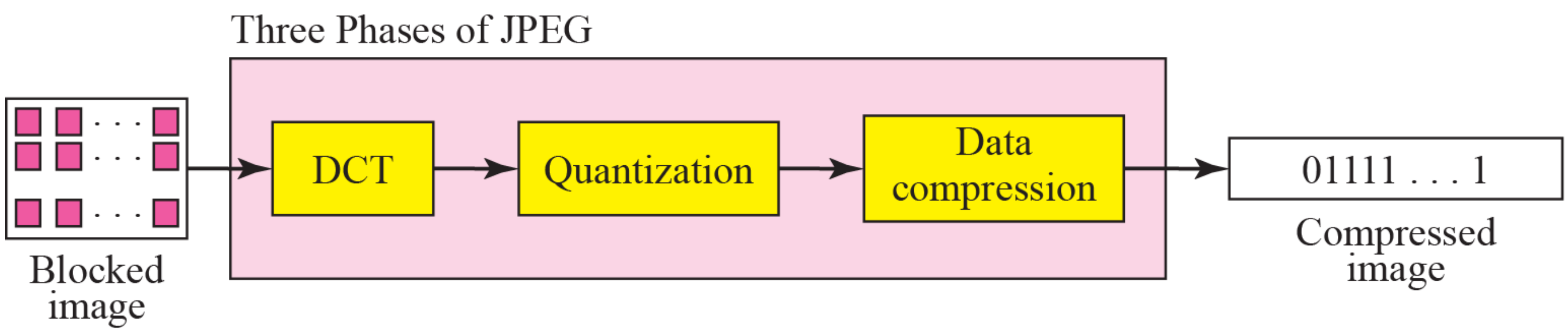


Figure 25.4 *Case 1: uniform gray scale*

Picture

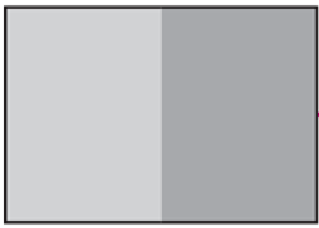
20	20	20	20	20	20	20	20
20	20	20	20	20	20	20	20
20	20	20	20	20	20	20	20
20	20	20	20	20	20	20	20
20	20	20	20	20	20	20	20
20	20	20	20	20	20	20	20
20	20	20	20	20	20	20	20
20	20	20	20	20	20	20	20

f(i,j)

160	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0

F(u,v)

Figure 25.5 *Case2: two sections*



Picture

20	20	20	20	50	50	50	50
20	20	20	20	50	50	50	50
20	20	20	20	50	50	50	50
20	20	20	20	50	50	50	50
20	20	20	20	50	50	50	50
20	20	20	20	50	50	50	50
20	20	20	20	50	50	50	50
20	20	20	20	50	50	50	50

f(i,j)

280	2109	0	39	0	225	0	22
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0

F(u,v)

Picture

20	30	40	50	60	70	80	90
20	30	40	50	60	70	80	90
20	30	40	50	60	70	80	90
20	30	40	50	60	70	80	90
20	30	40	50	60	70	80	90
20	30	40	50	60	70	80	90
20	30	40	50	60	70	80	90
20	30	40	50	60	70	80	90

f(i,j)

400	2146	0	231	21	3	21	28
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0

F(u,v)

Figure 25.7 *Reading the table*

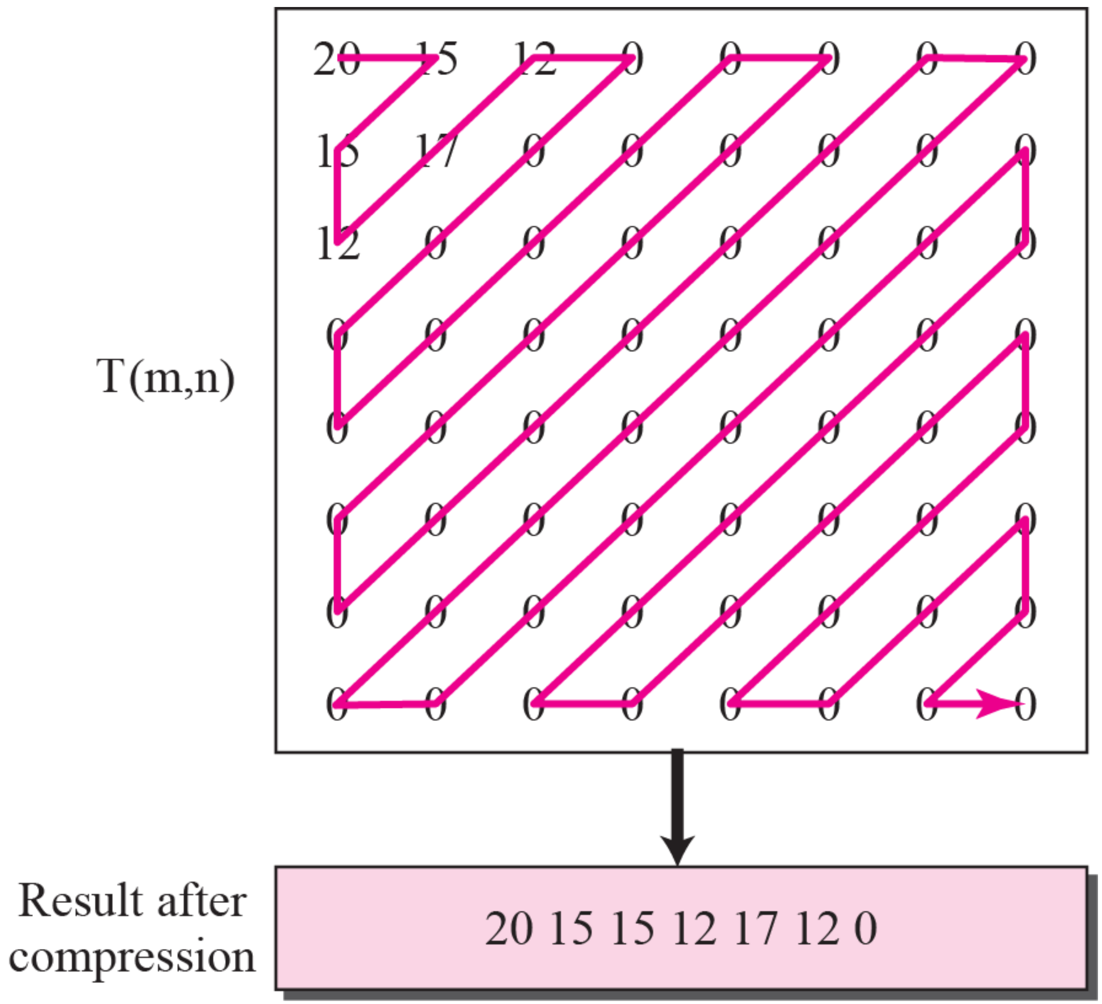


Figure 25.8 *MPEG frames*

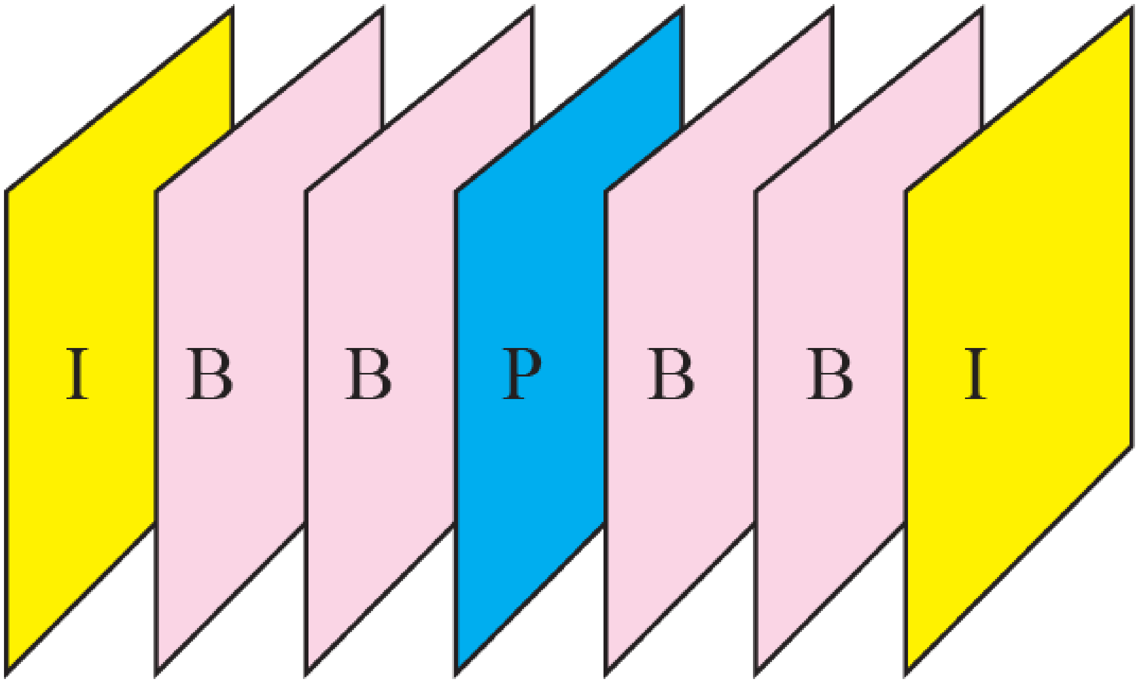
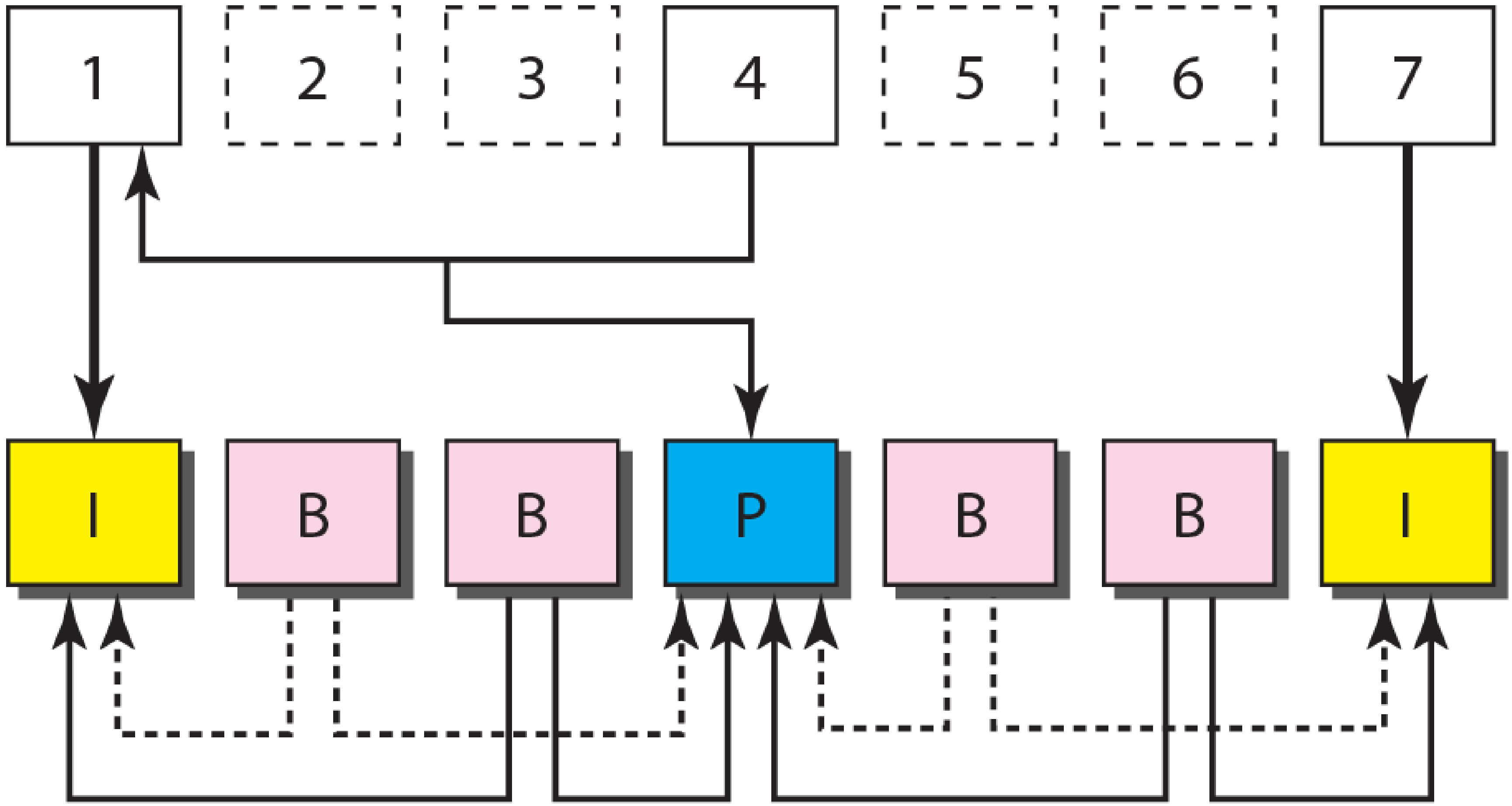


Figure 25.9 *MPEG frame construction*



25-4 STREAMING STORED AUDIO/VIDEO

Now that we have discussed digitizing and compressing audio/video, we turn our attention to specific applications. The first is streaming stored audio and video. Downloading these types of files from a Web server can be different from downloading other types of files. To understand the concept, let us discuss three approaches, each with a different complexity.

Topics Discussed in the Section

- ✓ **First Approach: Using a Web Server**
- ✓ **Second Approach: Using a Web Server with Metafile**
- ✓ **Third Approach: Using a Media Server**
- ✓ **Fourth Approach: Using a Media Server and RTSP**

Figure 25.10 *Using a Web server*

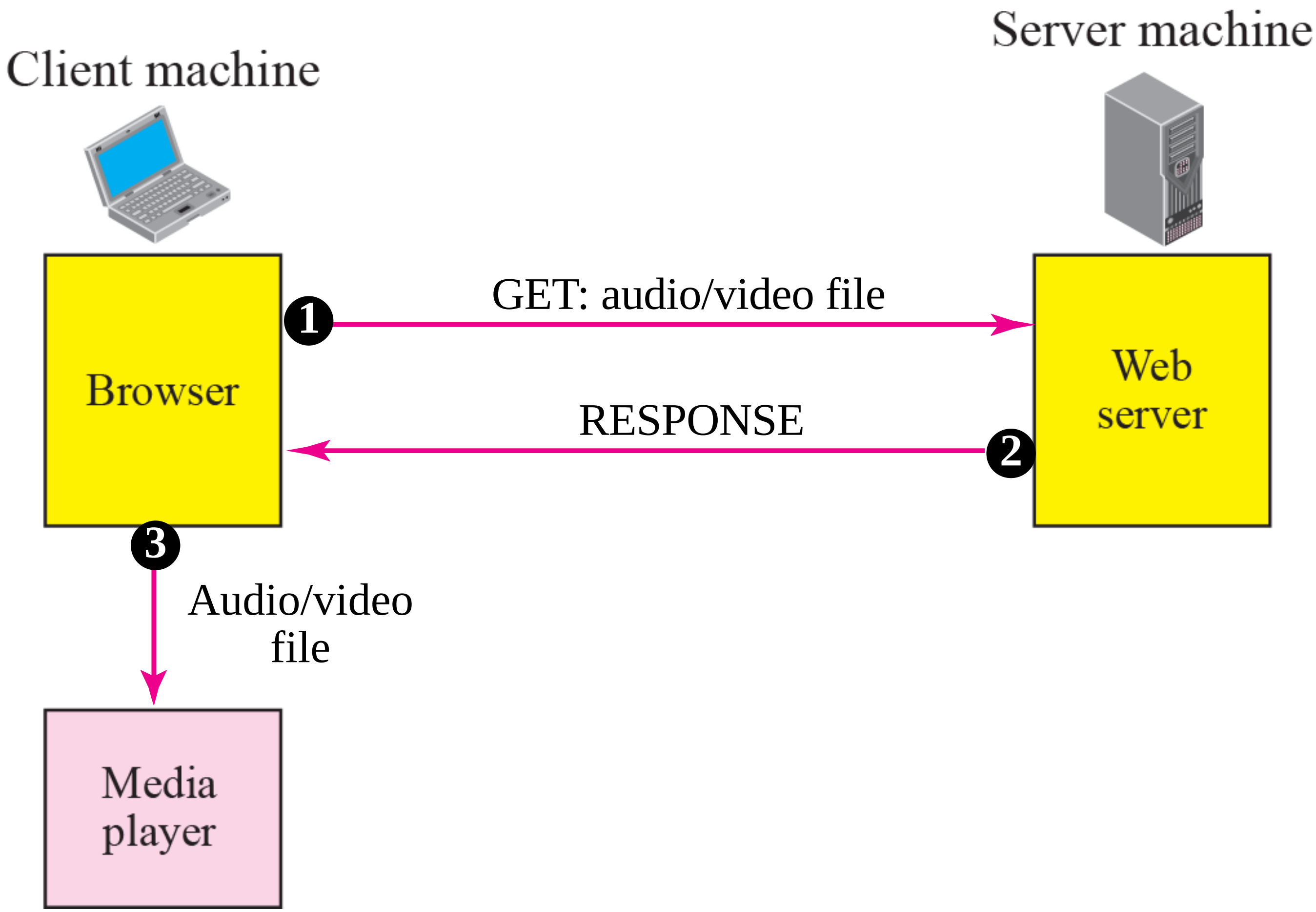


Figure 25.11 *Using a Web server with a metafile*

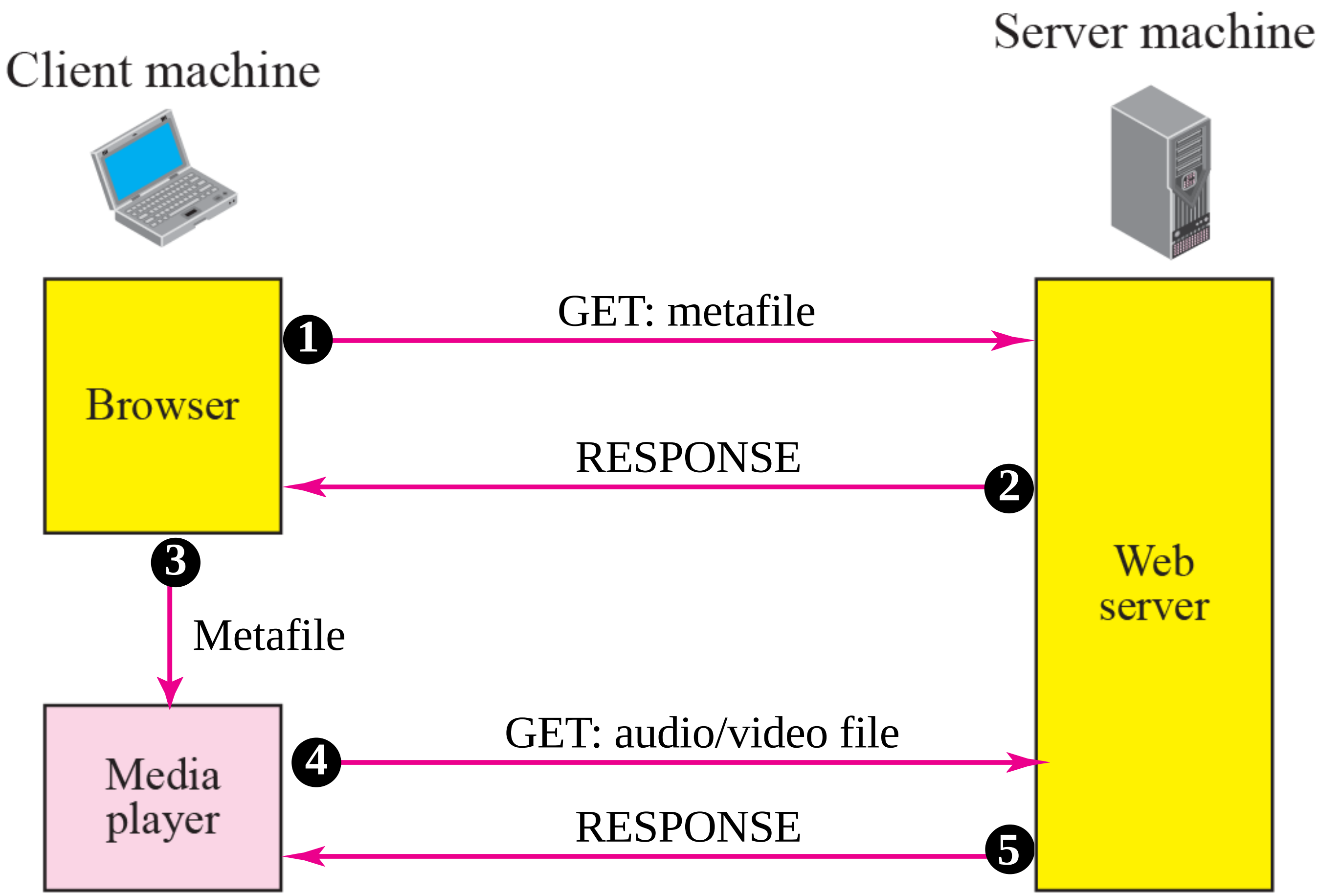


Figure 25.12 *Using a media server*

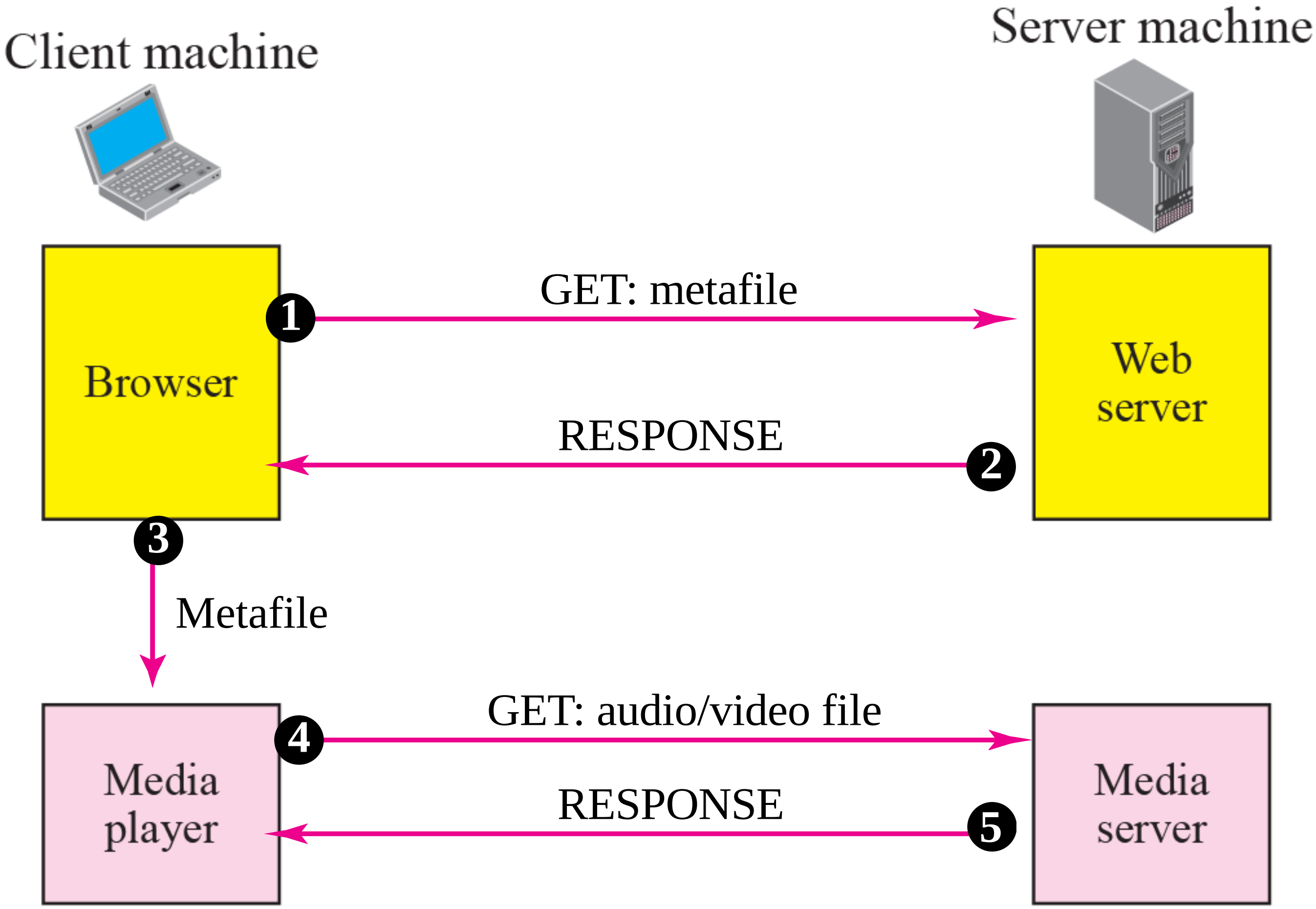
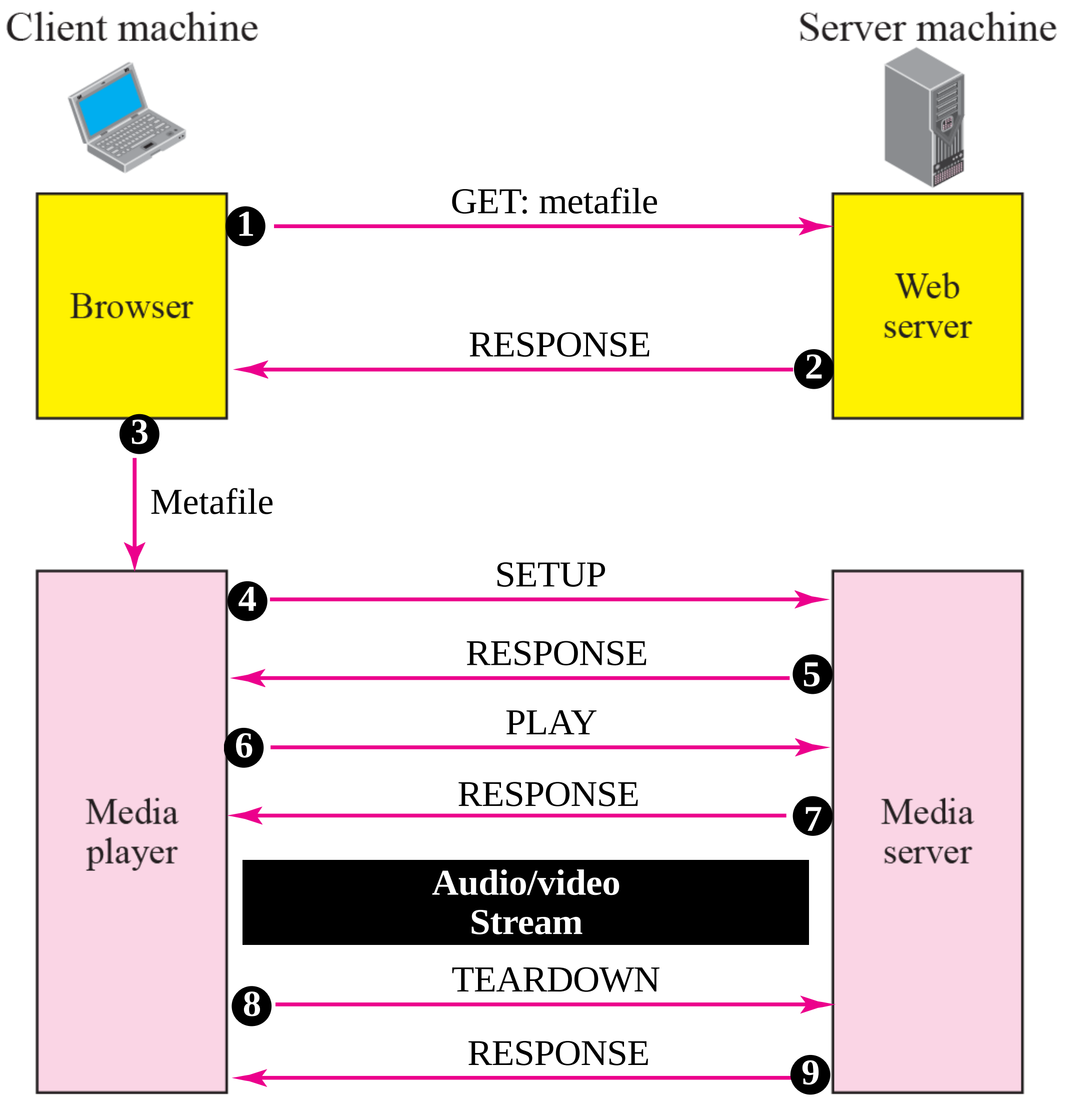


Figure 25.13 *Using a media server and RSTP*



25-5 STREAMING LIVE AUDIO/VIDEO

Streaming live audio/video is similar to the broadcasting of audio and video by radio and TV stations. Instead of broadcasting to the air, the stations broadcast through the Internet. There are several similarities between streaming stored audio/video and streaming live audio/video. They are both sensitive to delay; neither can accept retransmission. However, there is a difference. In the first application, the communication is unicast and on-demand. In the second, the communication is multicast and live.

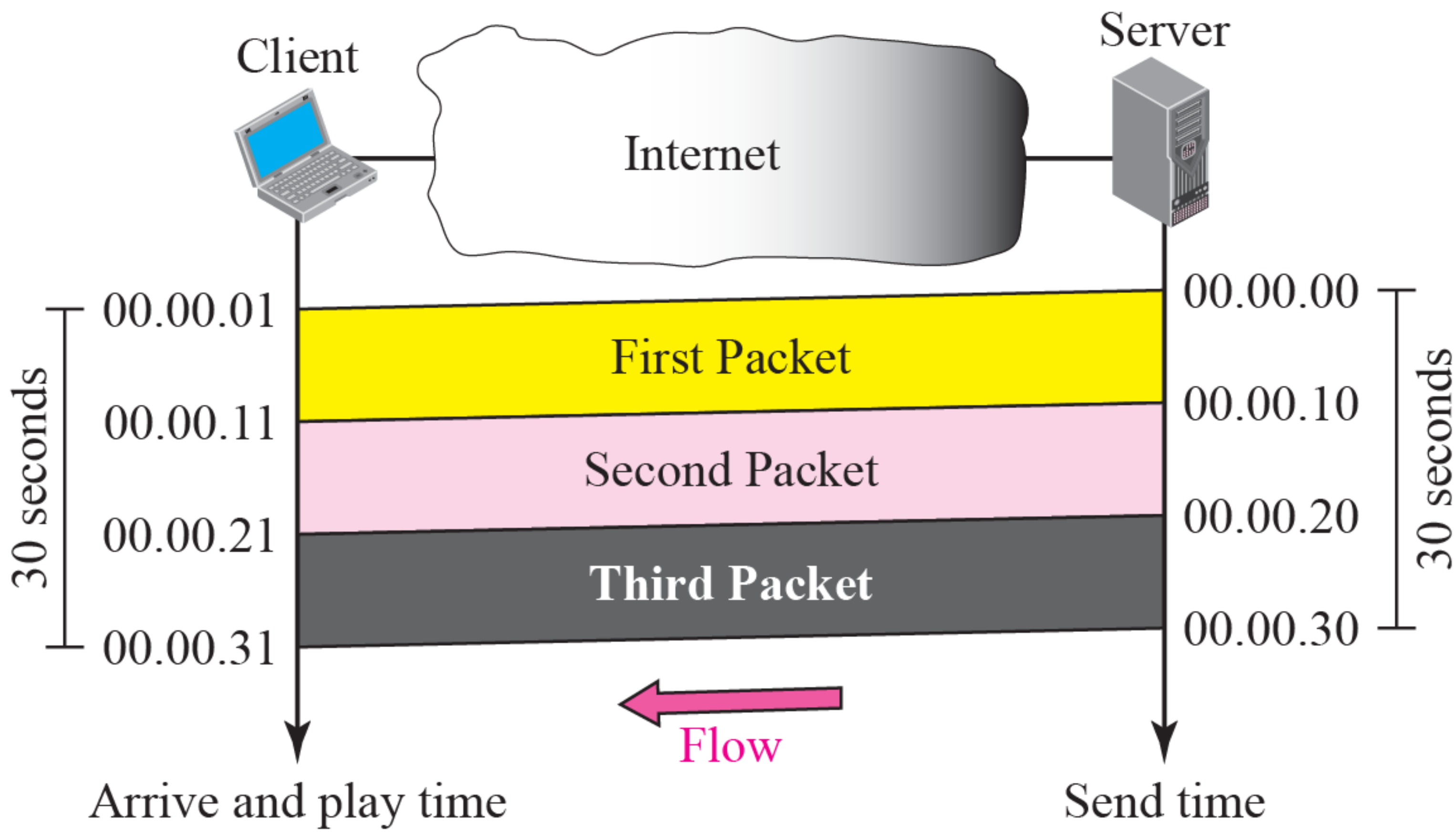
25-6 REAL-TIME INTERACTIVE AUDIO/VIDEO

In real-time interactive audio/video, people communicate with one another in real time. The Internet phone or voice over IP is an example of this type of application. Video conferencing is another example that allows people to communicate visually and orally.

Topics Discussed in the Section

- ✓ **Characteristics**

Figure 25.14 *Time relationship*



Note

Jitter is introduced in real-time data by the delay between packets.

Figure 25.15 *Jitter*

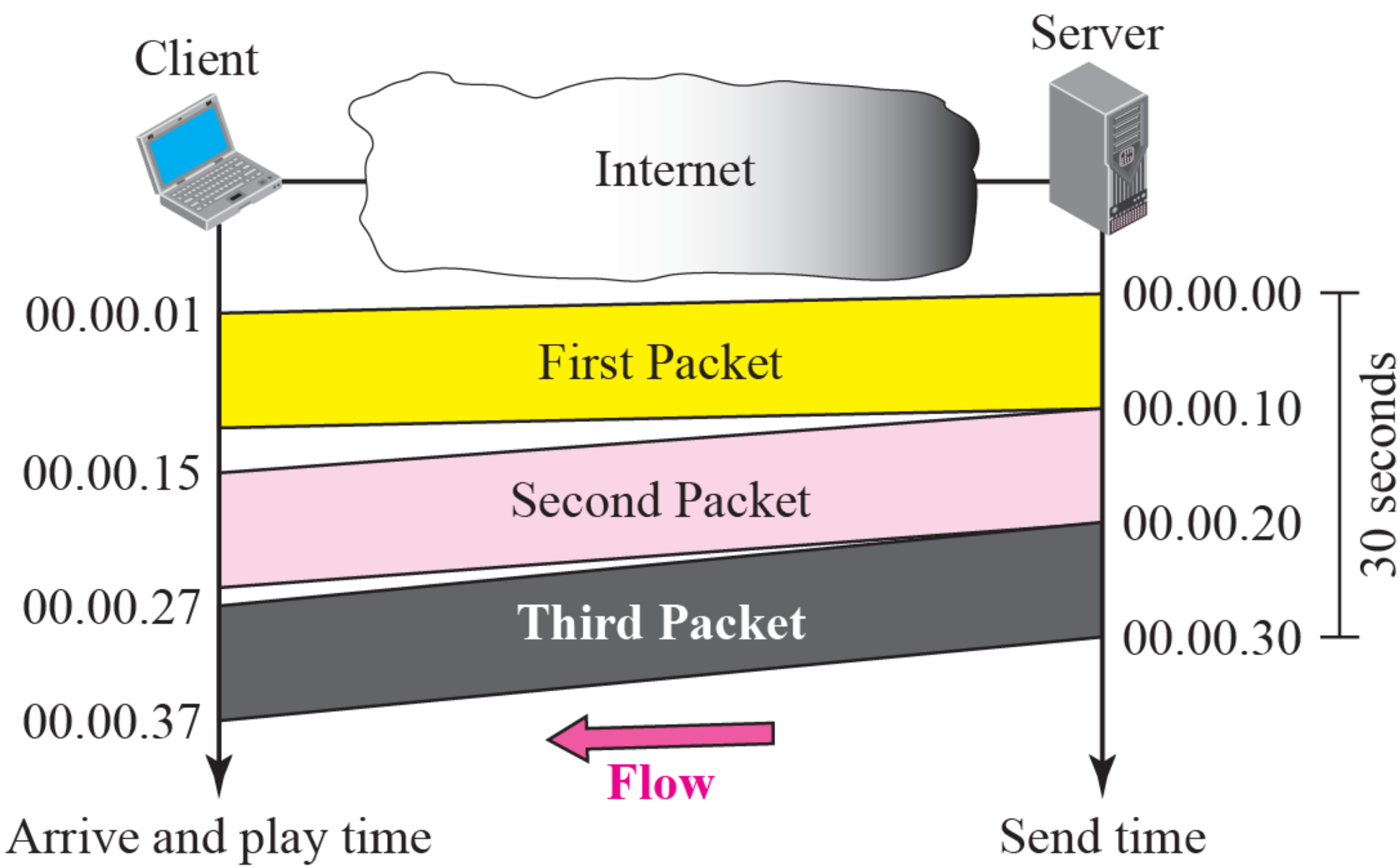
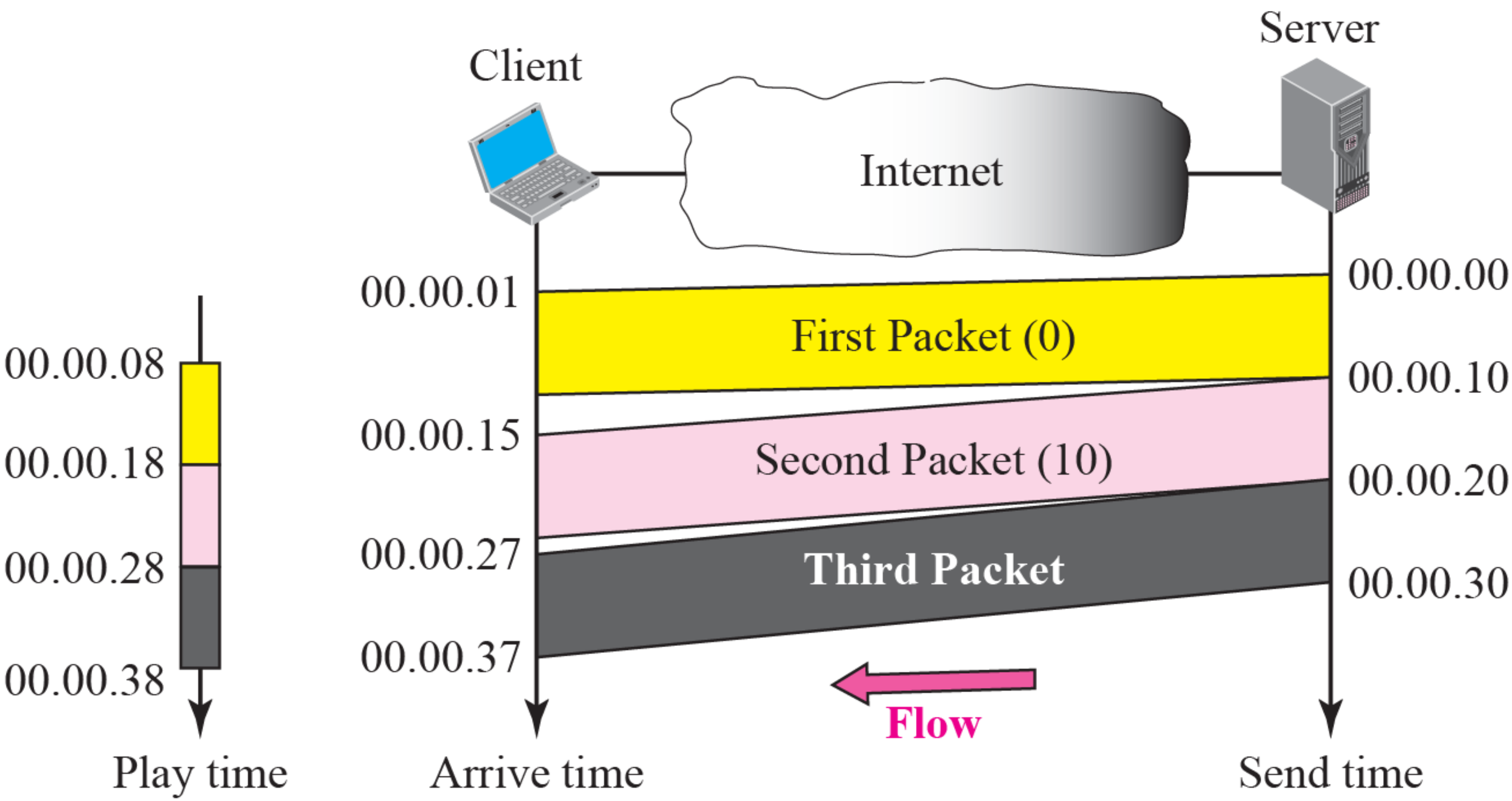


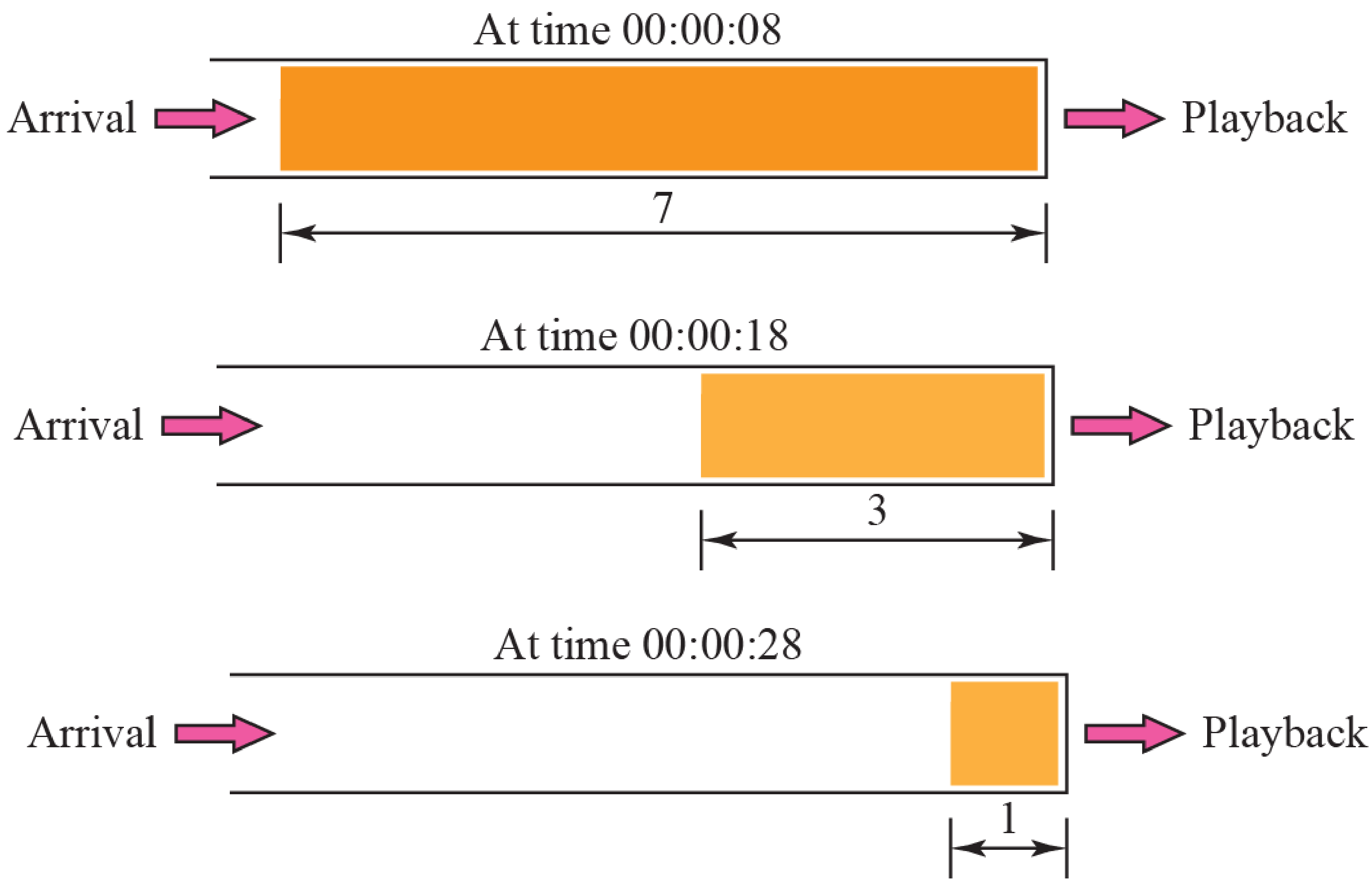
Figure 25.16 *Timestamp*



Note

To prevent jitter, we can timestamp the packets and separate the arrival time from the playback time.

Figure 25.17 *Playback buffer*



Note

A playback buffer is required for real-time traffic.

Note

A sequence number on each packet is required for real-time traffic.

Note

Real-time traffic needs the support of multicasting.

Note

Translation means changing the encoding of a payload to a lower quality to match the bandwidth of the receiving network.

Note

Mixing means combining several streams of traffic into one stream.

Note

TCP, with all its sophistication, is not suitable for interactive multimedia traffic because we cannot allow retransmission of packets.

Note

UDP is more suitable than TCP for interactive traffic. However, we need the services of RTP, another transport layer protocol, to make up for the deficiencies of UDP.

25-7 RTP

Real-time Transport Protocol (RTP) is the protocol designed to handle real-time traffic on the Internet. RTP does not have a delivery mechanism (multicasting, port numbers, and so on); it must be used with UDP. RTP stands between UDP and the application program. The main contributions of RTP are timestamping, sequencing, and mixing facilities.

Topics Discussed in the Section

- ✓ **RTP Packet Format**
- ✓ **UDP Port**

Figure 25.18 *RTP packet header format*

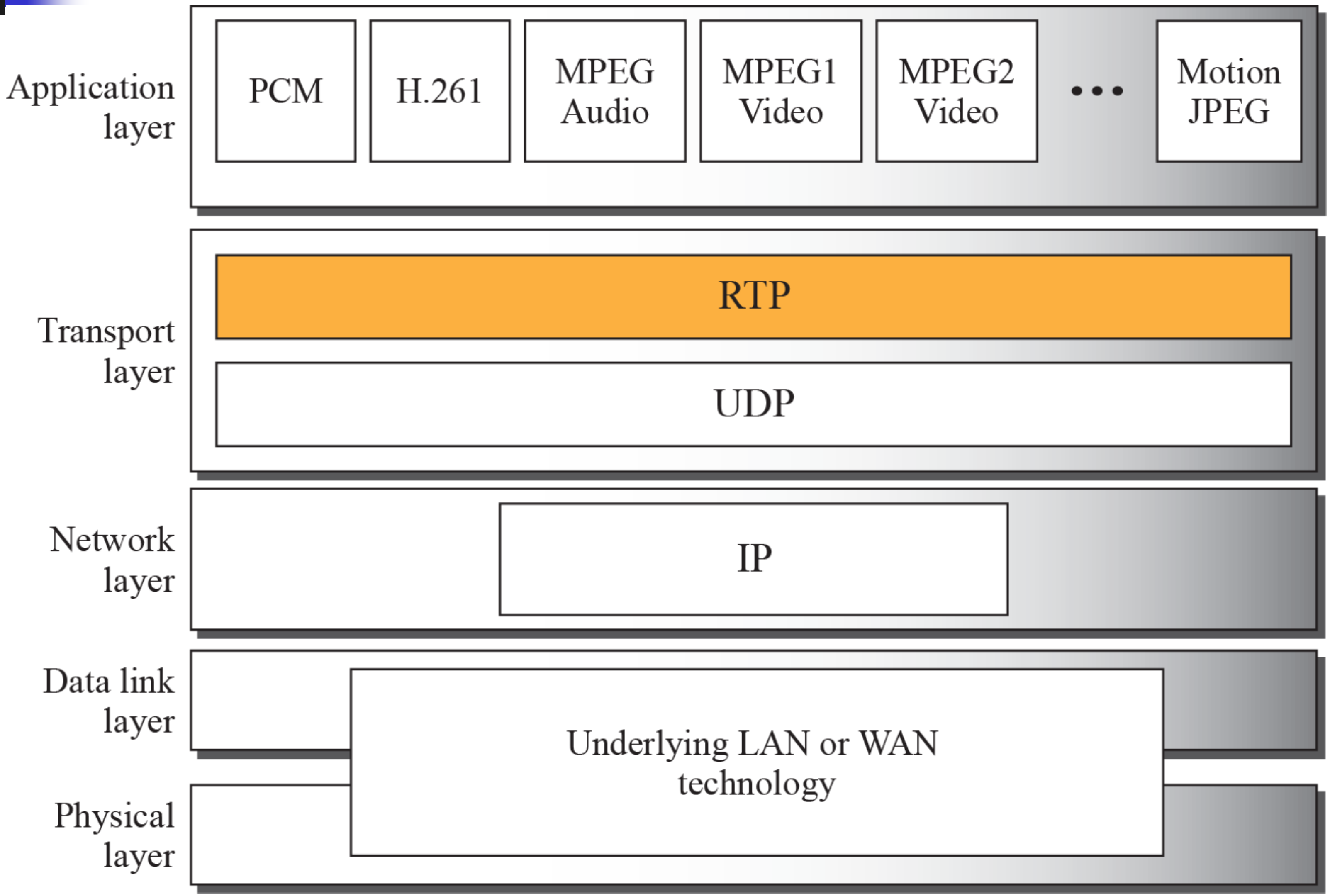


Figure 25.19 *RTP packet header format*

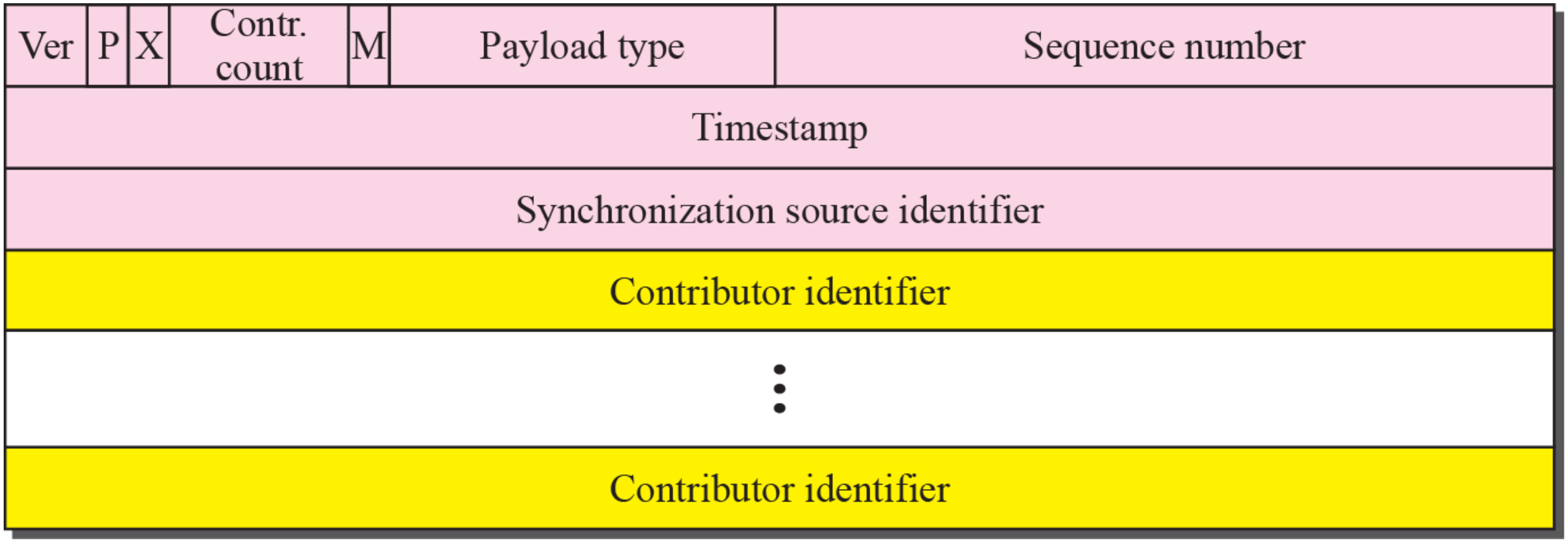


Table 25.1 *Payload Types*

Type	*Application*	*Type*	*Application*	*Type*	*Application*
0	PCMμ Audio	7	LPC audio	15	G728 audio
1	1016	8	PCMA audio	26	Motion JPEG
2	G721 audio	9	G722 audio	31	H.261
3	GSM audio	10–11	L16 audio	32	MPEG1 video
5–6	DV14 audio	14	MPEG audio	33	MPEG2 video

Note

RTP uses a temporary even-numbered UDP port.

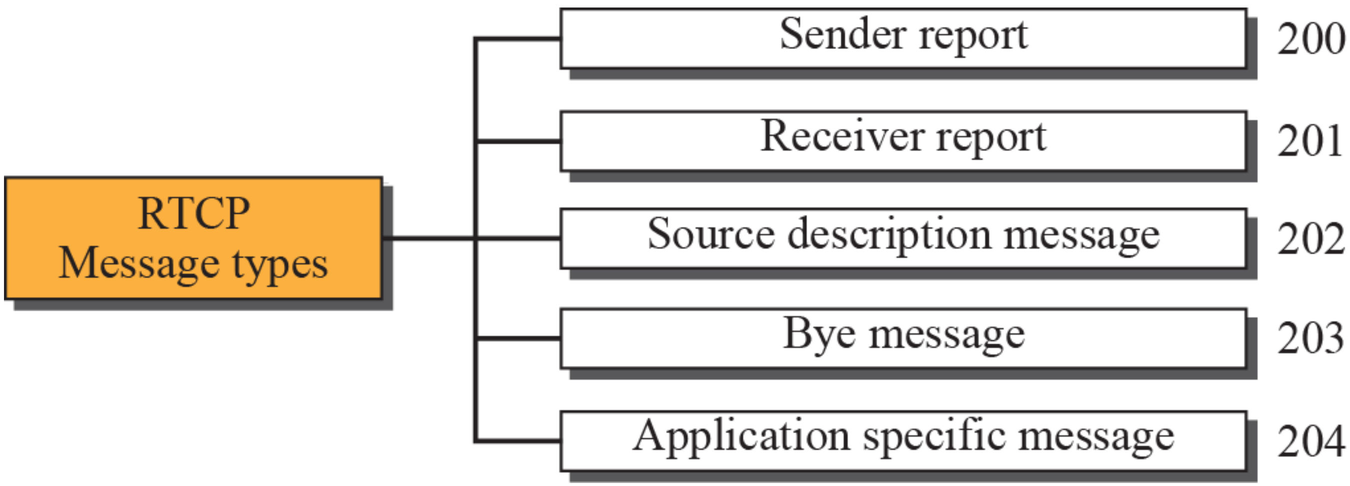
25-8 RTCP

RTP allows only one type of message, one that carries data from the source to the destination. In many cases, there is a need for other messages in a session. These messages control the flow and quality of data and allow the recipient to send feedback to the source or sources. Real-Time Transport Control Protocol (RTCP) is a protocol designed for this purpose.

Topics Discussed in the Section

- ✓ Sender Report
- ✓ Receiver Report
- ✓ Source Description Message
- ✓ Bye Message
- ✓ Application-Specific Message
- ✓ UDP Port

Figure 25.20 *RTCP message types*



Note

RTCP uses an odd-numbered UDP port number that follows the port number selected for RTP.

25-9 VOICE OVER IP

Let us concentrate on one real-time interactive audio/video application: voice over IP, or Internet telephony. The idea is to use the Internet as a telephone network with some additional capabilities. Instead of communicating over a circuit-switched network, this application allows communication between two parties over the packet-switched Internet. Two protocols have been designed to handle this type of communication: SIP and H.323. We briefly discuss both.

Topics Discussed in the Section

- ✓ **SIP**
- ✓ **H.323**

Figure 25.21 *SIP messages*

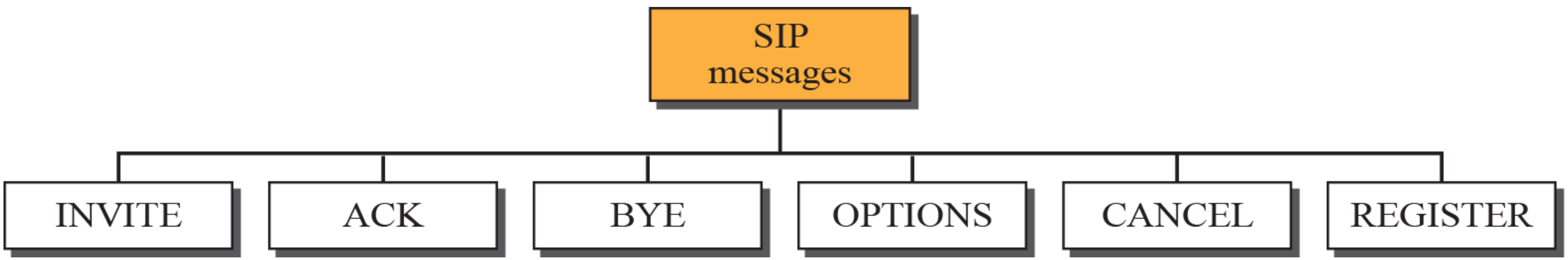


Figure 25.22 *SIP formats*

Figure 25.23 *SIP simple session*

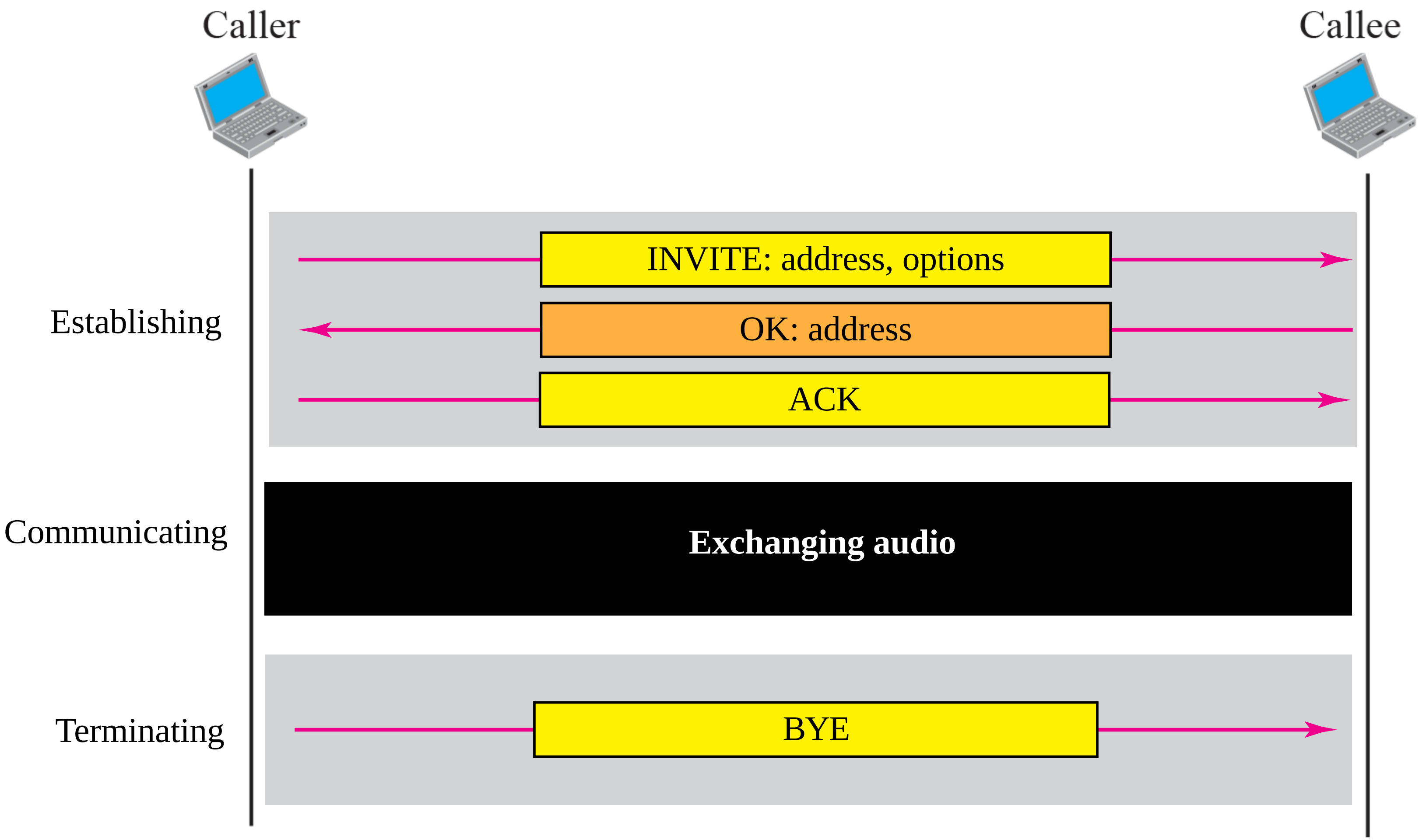


Figure 25.24 *Tracking the callee*

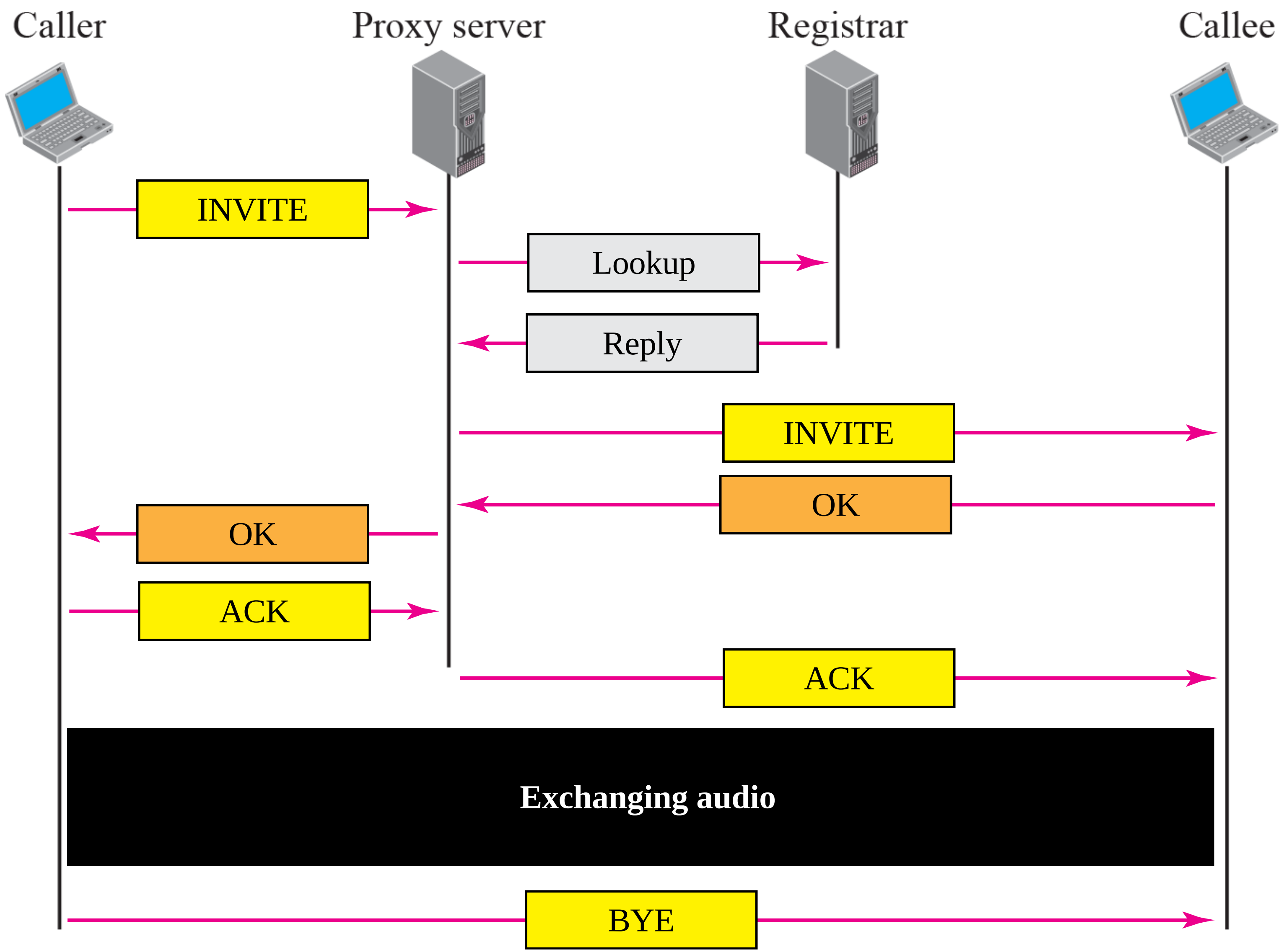


Figure 25.25 *H.323 architecture*

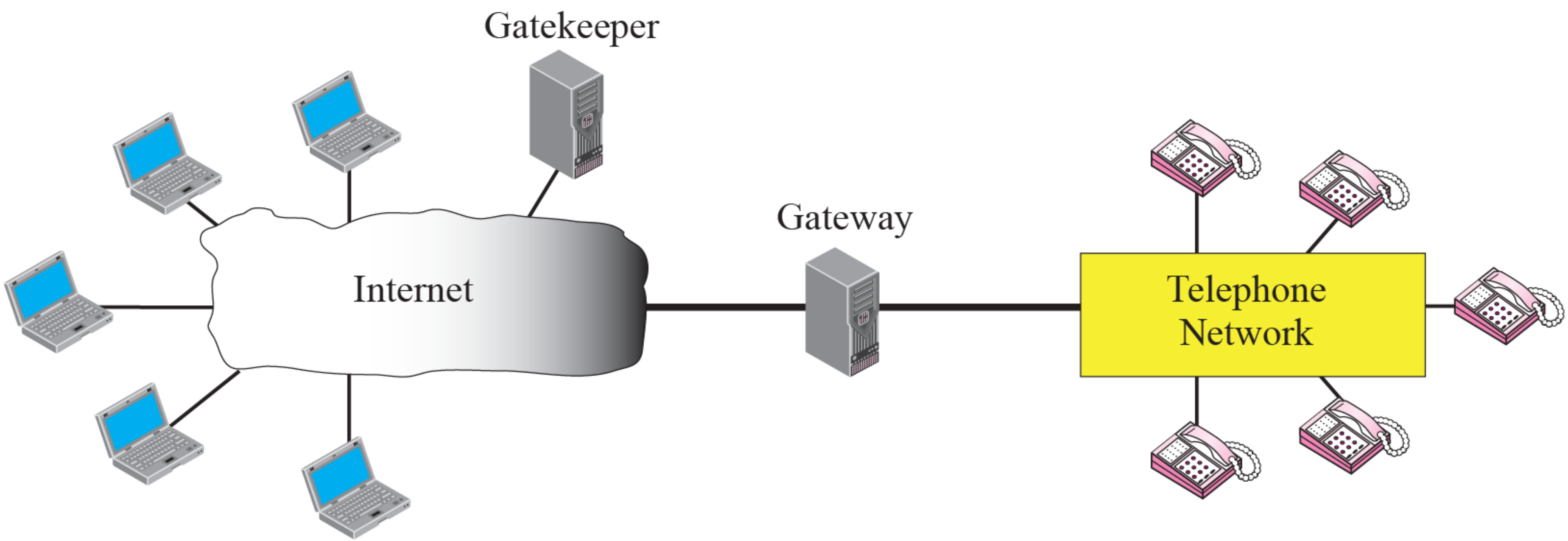


Figure 25.26 *H.323 protocols*

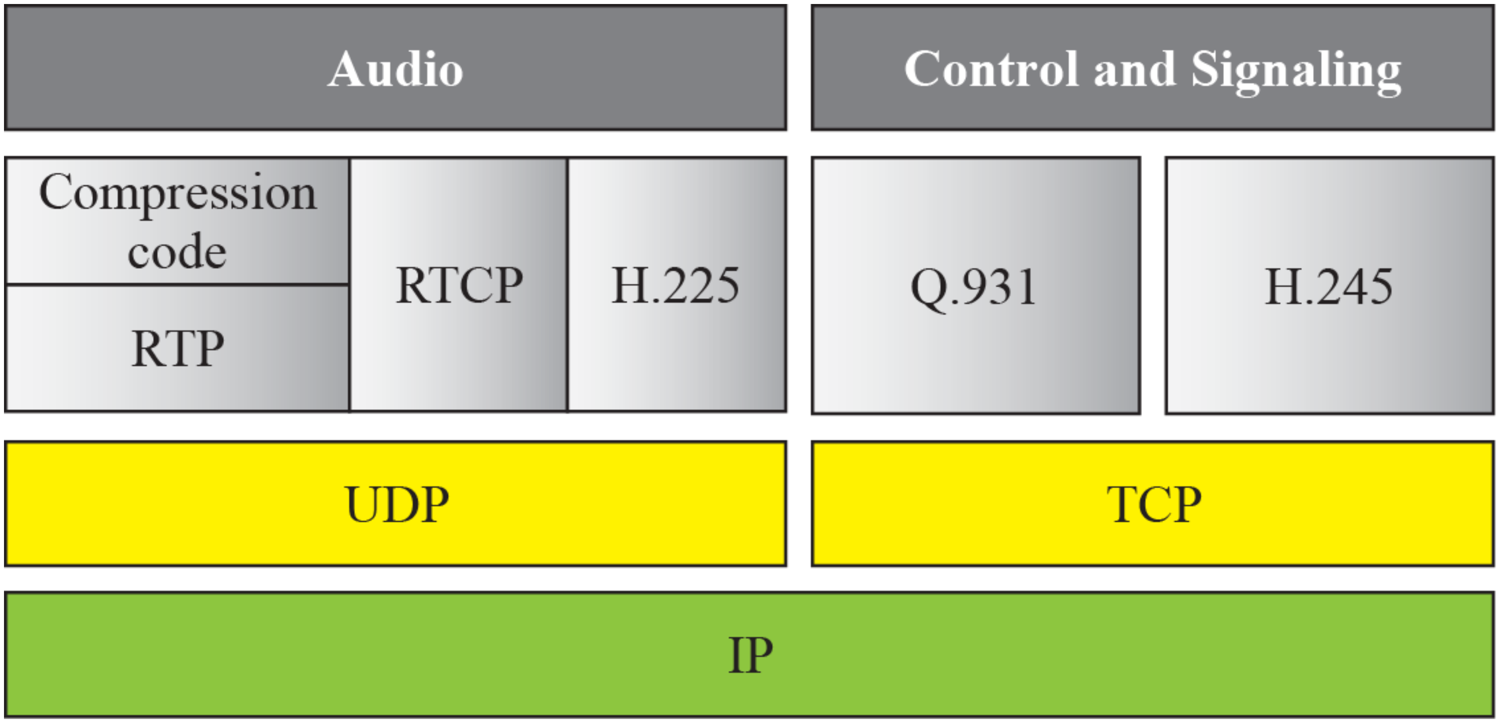
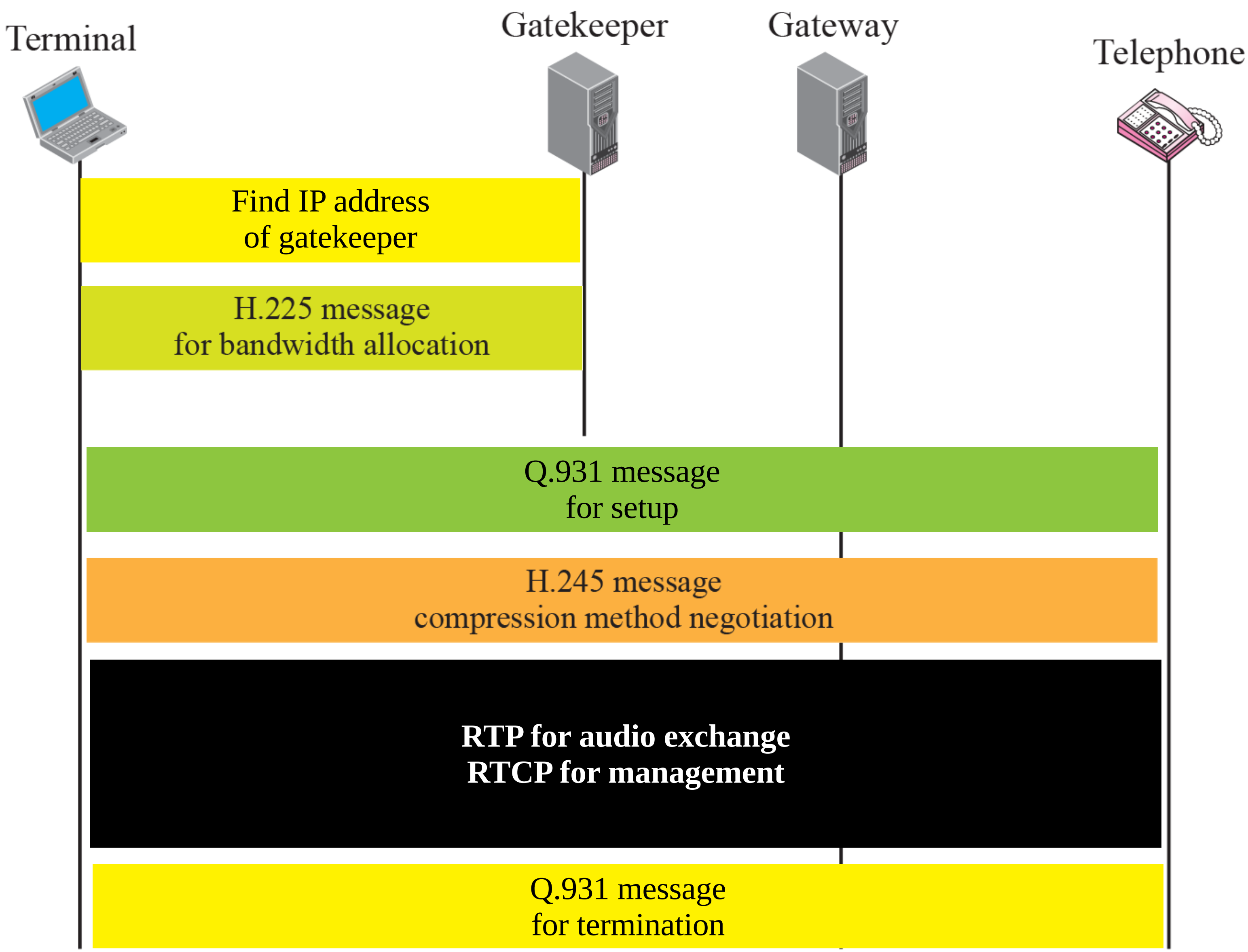


Figure 25.27 *H.323 example*



25-10 QUALITY OF SERVICE

Quality of service (QoS) is an internetworking issue that has been discussed more than defined. We can informally define quality of service as something a flow of data seeks to attain. Although QoS can be applied to both textual data and multimedia, it is more an issue when we are dealing with multimedia.

Topics Discussed in the Section

- ✓ Flow Characteristics
- ✓ Flow Classes
- ✓ Techniques to Improve QoS
- ✓ Resource Reservation
- ✓ Admission Control

Figure 25.28 *Flow characteristics*

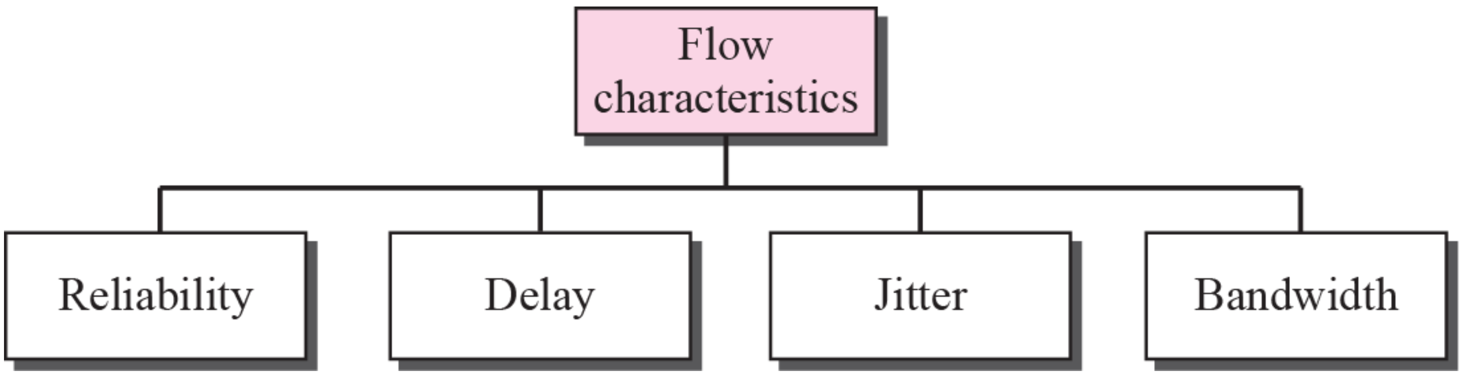


Figure 25.29 *FIFO queues*

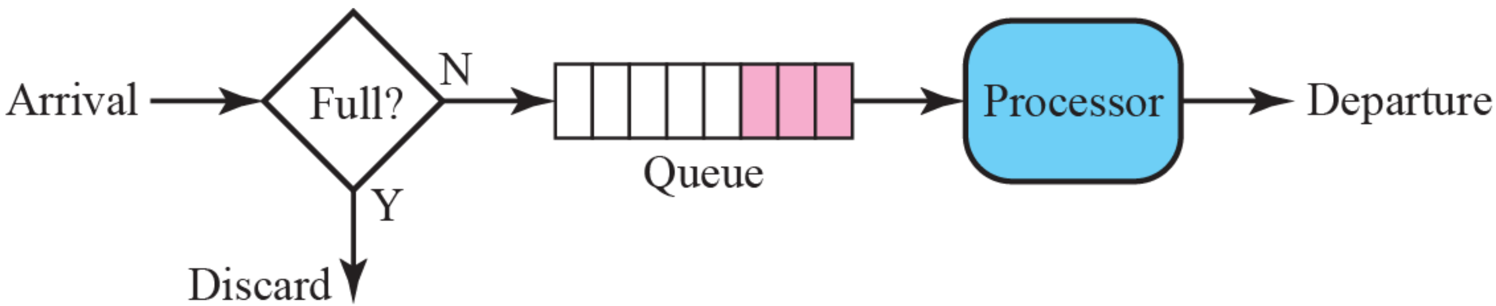


Figure 25.30 *Priority queues*

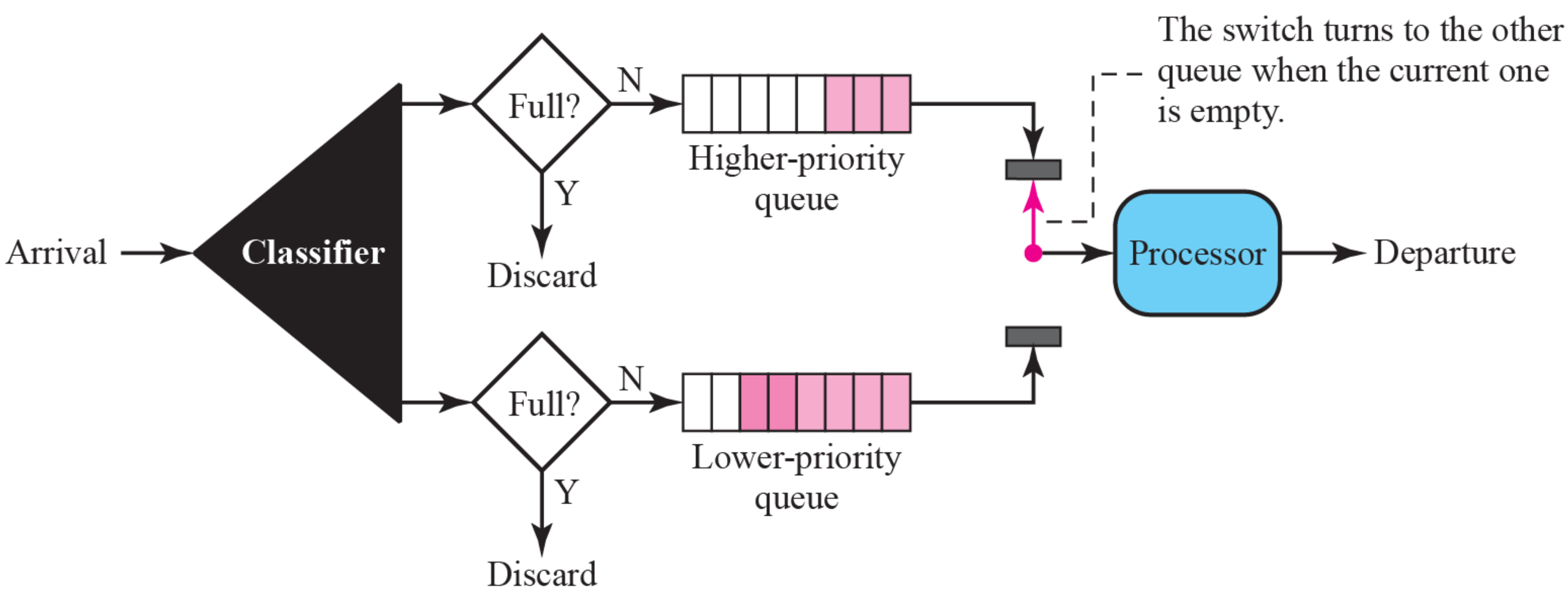


Figure 25.31 *Weighted fair queuing*

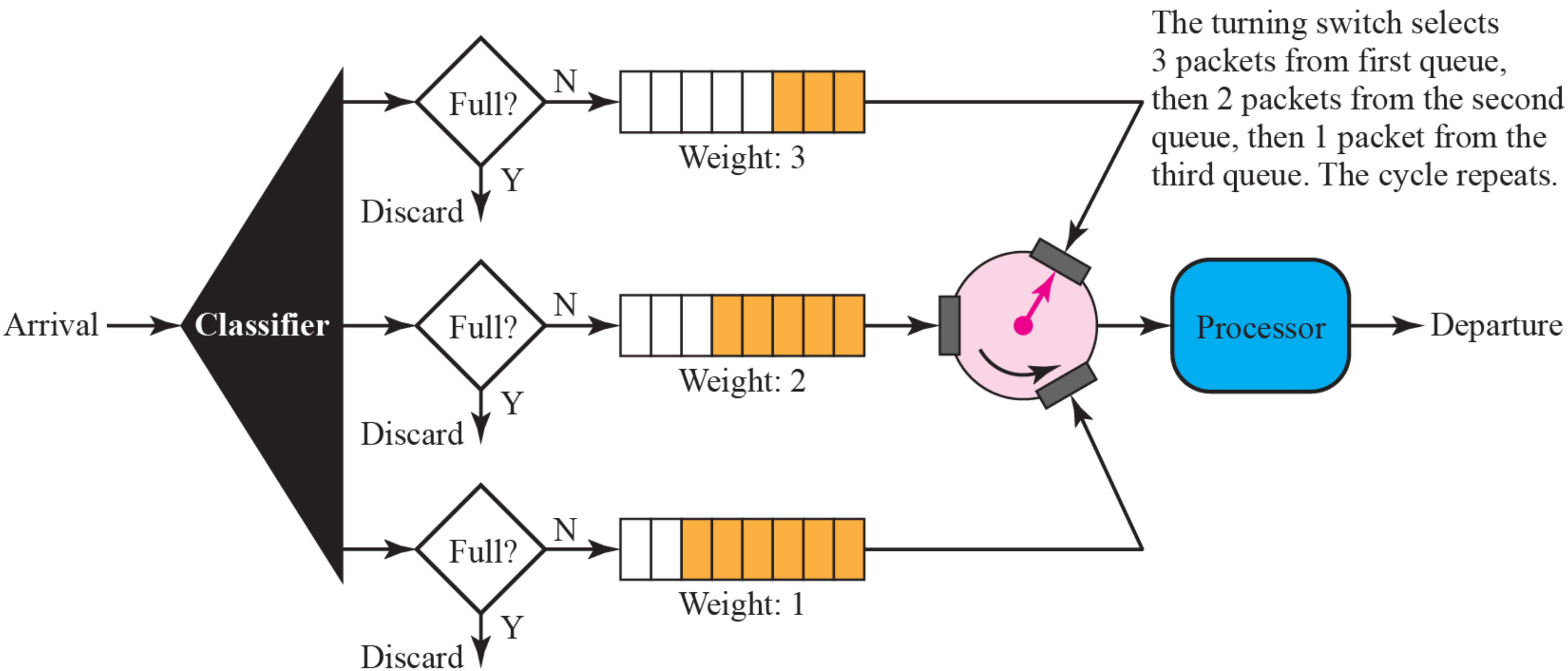


Figure 25.32 *Leaky bucket*

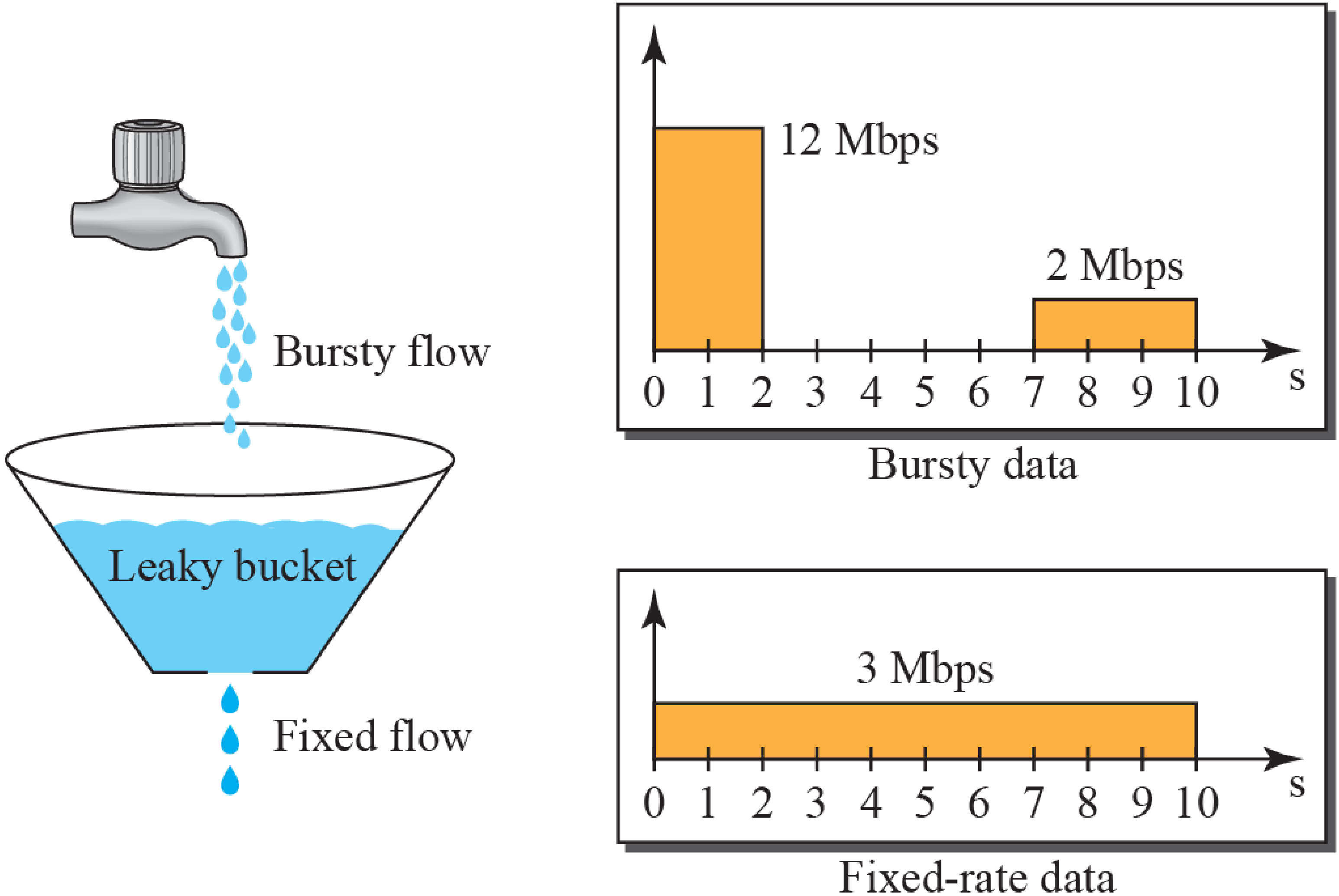
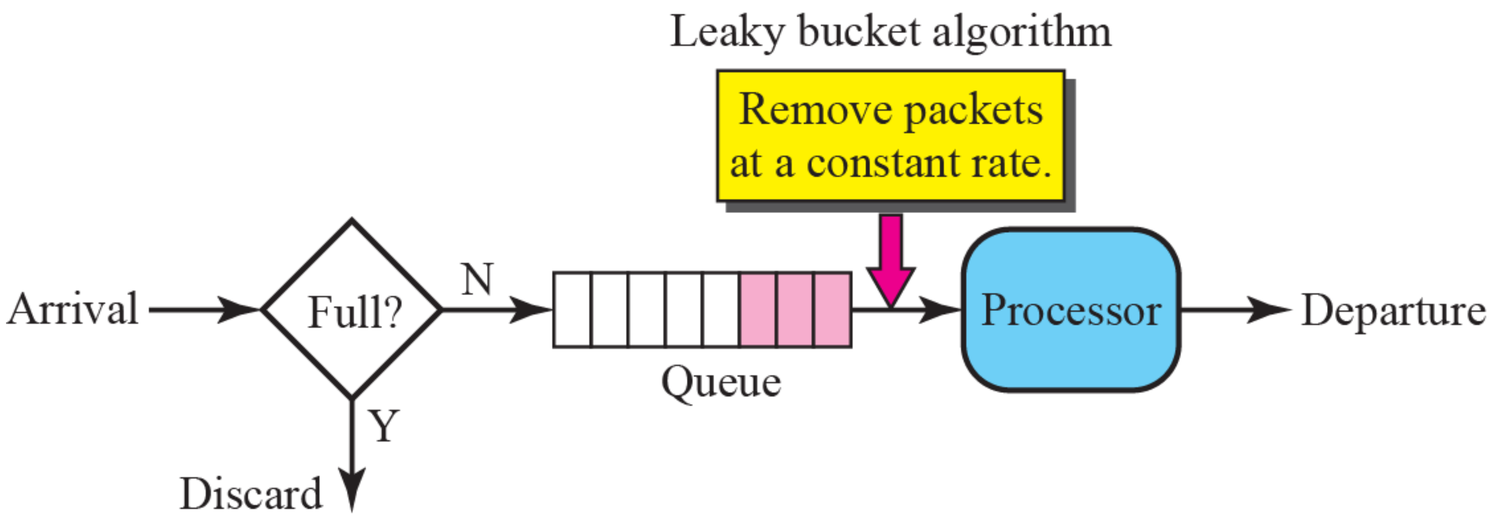


Figure 25.33 *Leaky bucket implementation*

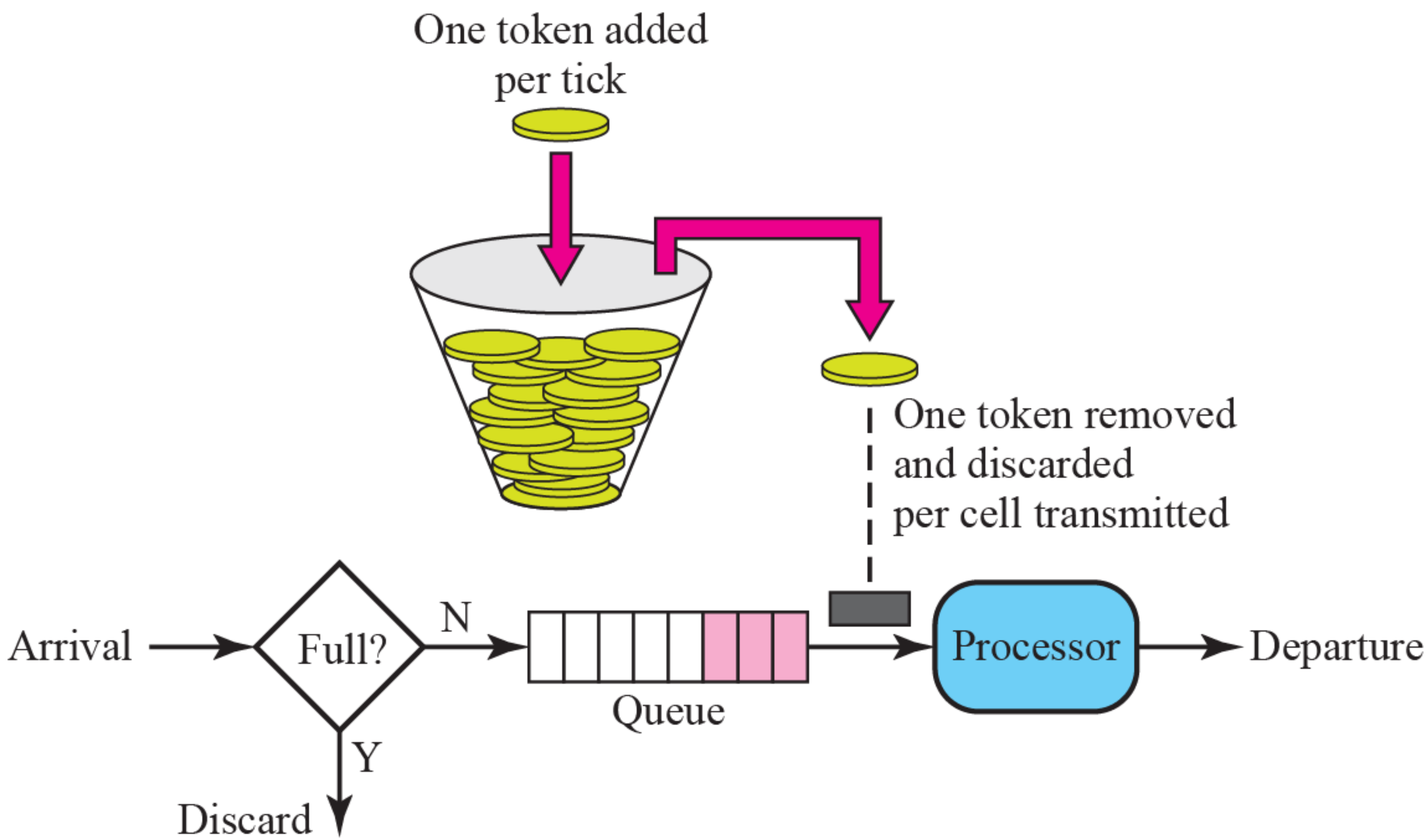


Note

A leaky bucket algorithm shapes bursty traffic into fixed-rate traffic by averaging the data rate.

It may drop the packets if the bucket is full.

Figure 25.34 *Token bucket*



Note

The token bucket allows bursty traffic at a regulated maximum rate.

25-11 INTEGRATED SERVICES

IP was originally designed for best-effort delivery. This means that every user receives the same level of services. This type of delivery does not guarantee the minimum of a service, such as bandwidth, to applications such as real-time audio and video. Integrated Services, sometimes called IntServ, is a flow-based QoS model, which means that a user needs to create a flow, a kind of virtual circuit, from the source to the destination and inform all routers of the resource requirement.

Topics Discussed in the Section

- ✓ **Signaling**
- ✓ **Flow Specification**
- ✓ **Admission**
- ✓ **Service Classes**
- ✓ **RSVP**
- ✓ **Problems with Integrated Services**

Note

Integrated Services is a flow-based QoS model designed for IP.

Figure 25.35 *Path messages*

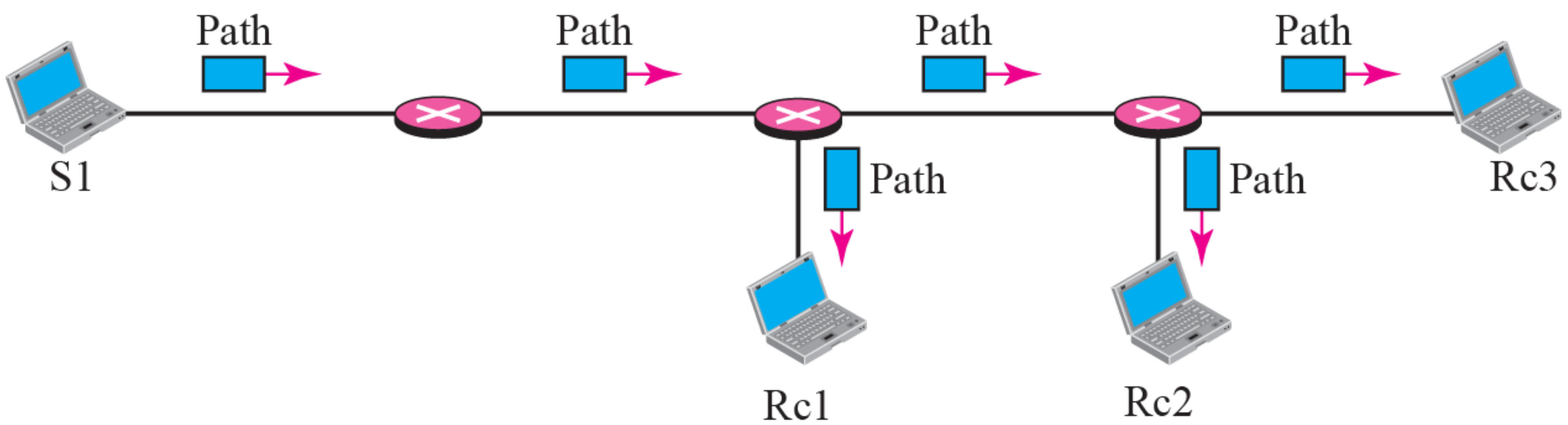


Figure 25.36 *Resv messages*

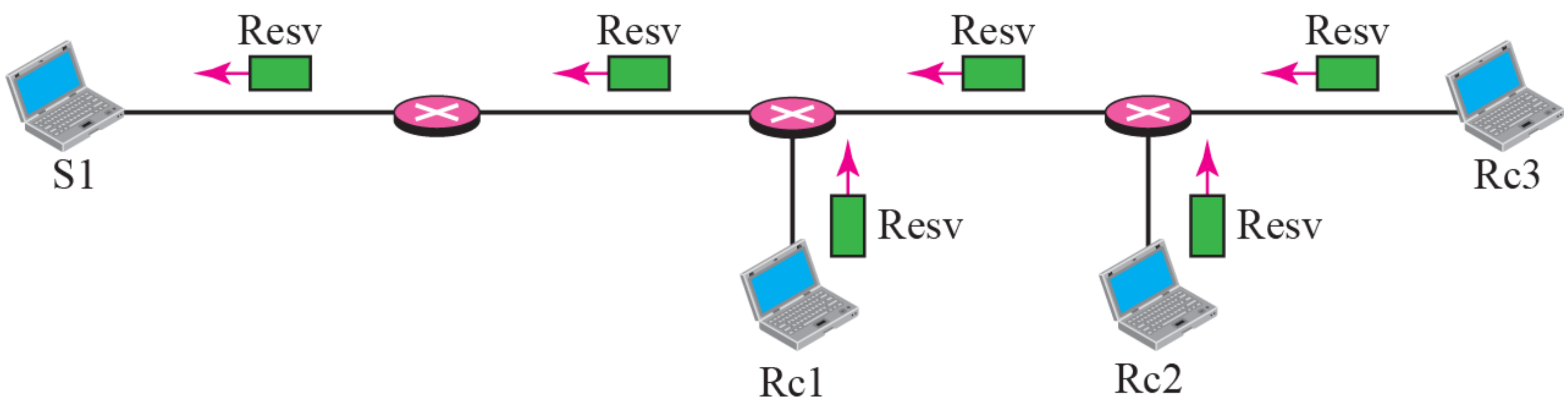


Figure 25.37 *Reservation merging*

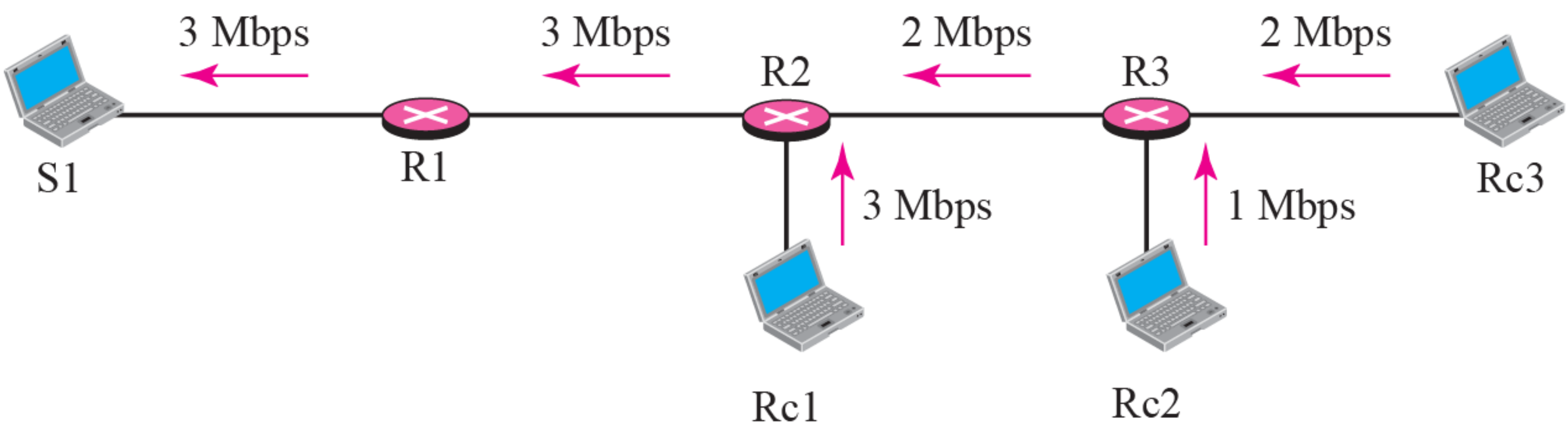
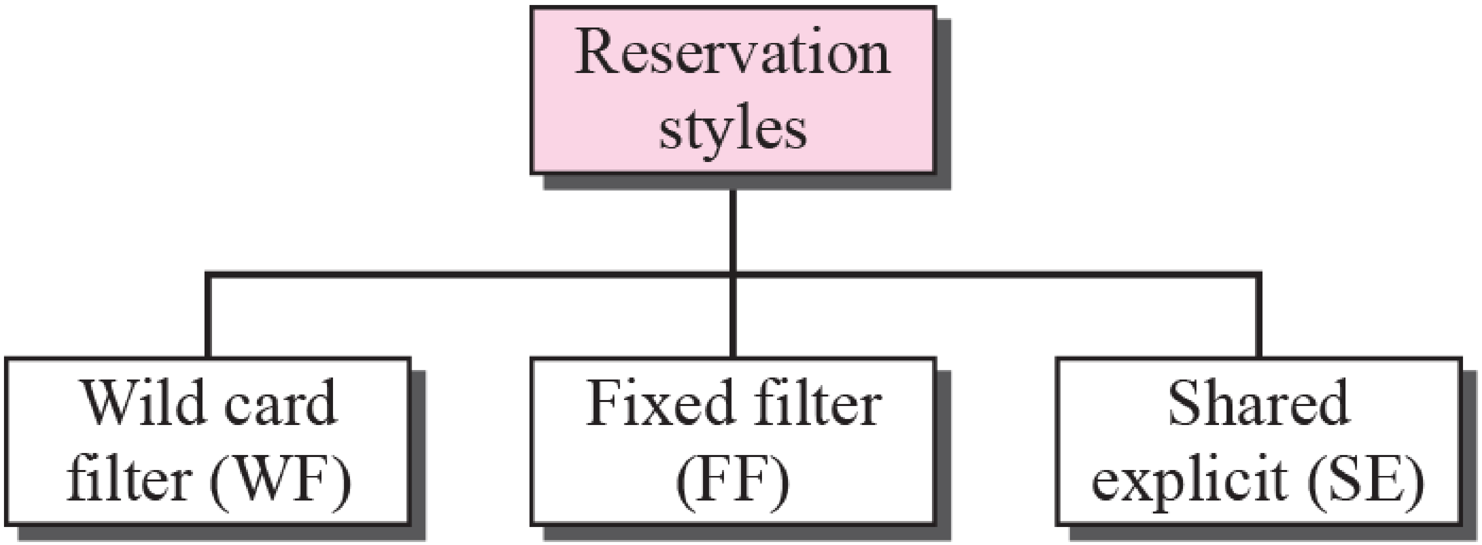


Figure 25.38 *Reservation styles*



25-12 DIFFERENTIATED SERVICES

Differentiated Services (DS or Diffserv) was introduced by the IETF (Internet Engineering Task Force) to handle the shortcomings of Integrated Services.

Topics Discussed in the Section

- ✓ **DS Field**

Note

Differentiated Services is a class-based QoS model designed for IP.

Figure 25.39 *DS field*

Figure 25.40 *Traffic conditioner*