Application Layer - 4

RTP & VoIP

29-6 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; it must be used with UDP. RTP stands between UDP and the application program. The main contributions of RTP are time-stamping, sequencing, and mixing facilities.

RTP Packet Format
UDP Port

Figure 29.18 *RTP*

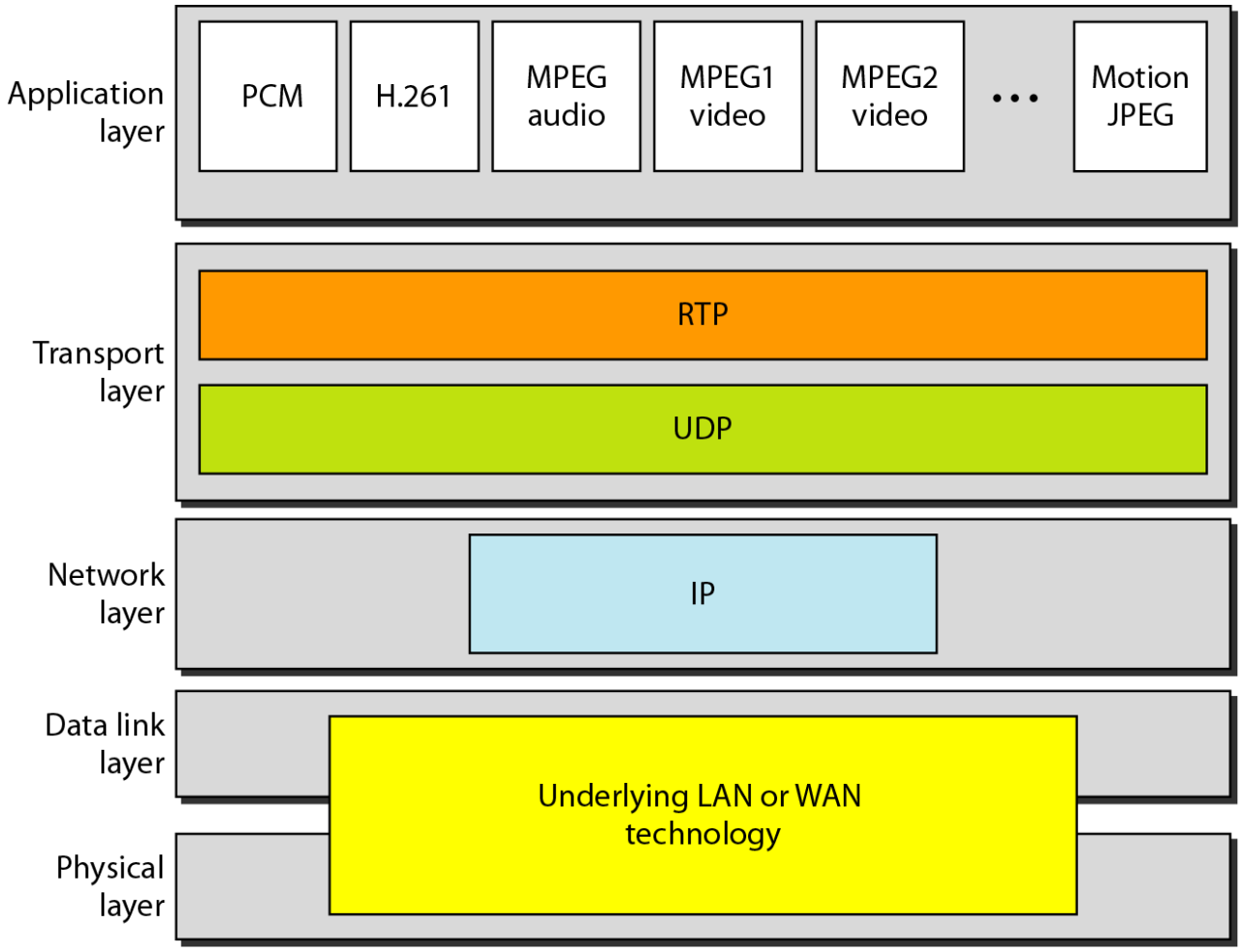


Figure 29.19 *RTP packet header format*

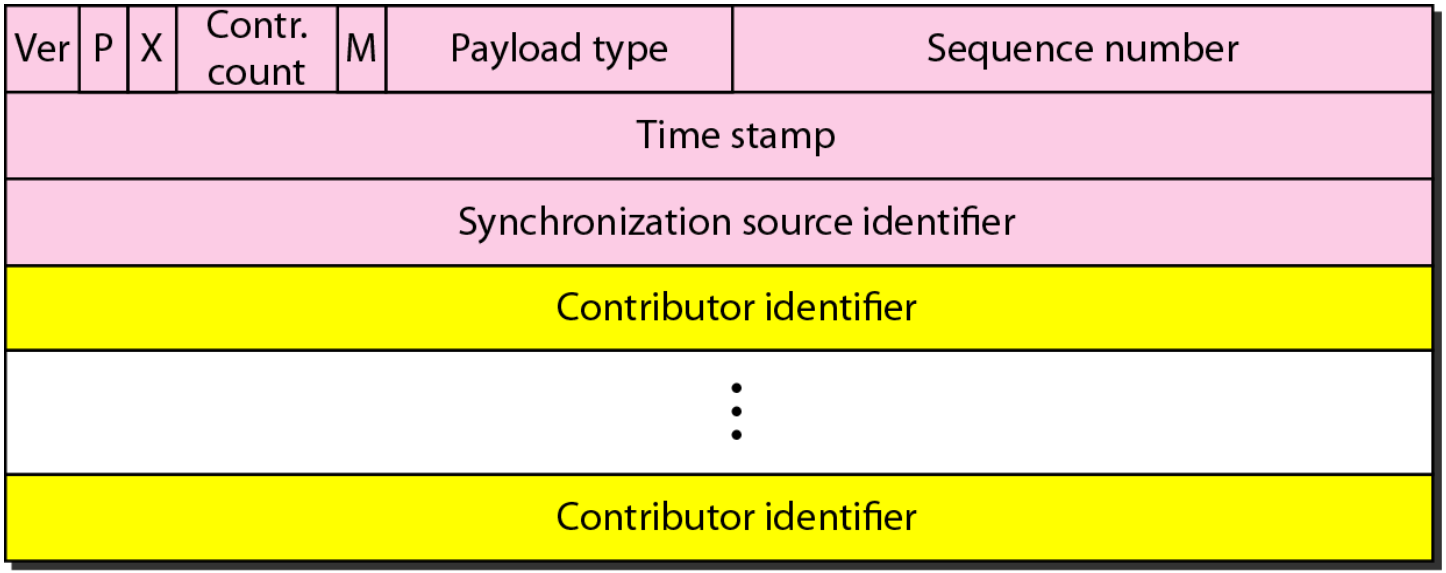
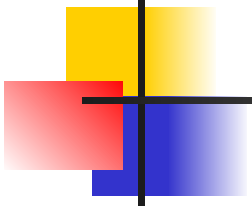


Table 20.1 ***Payload types***

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

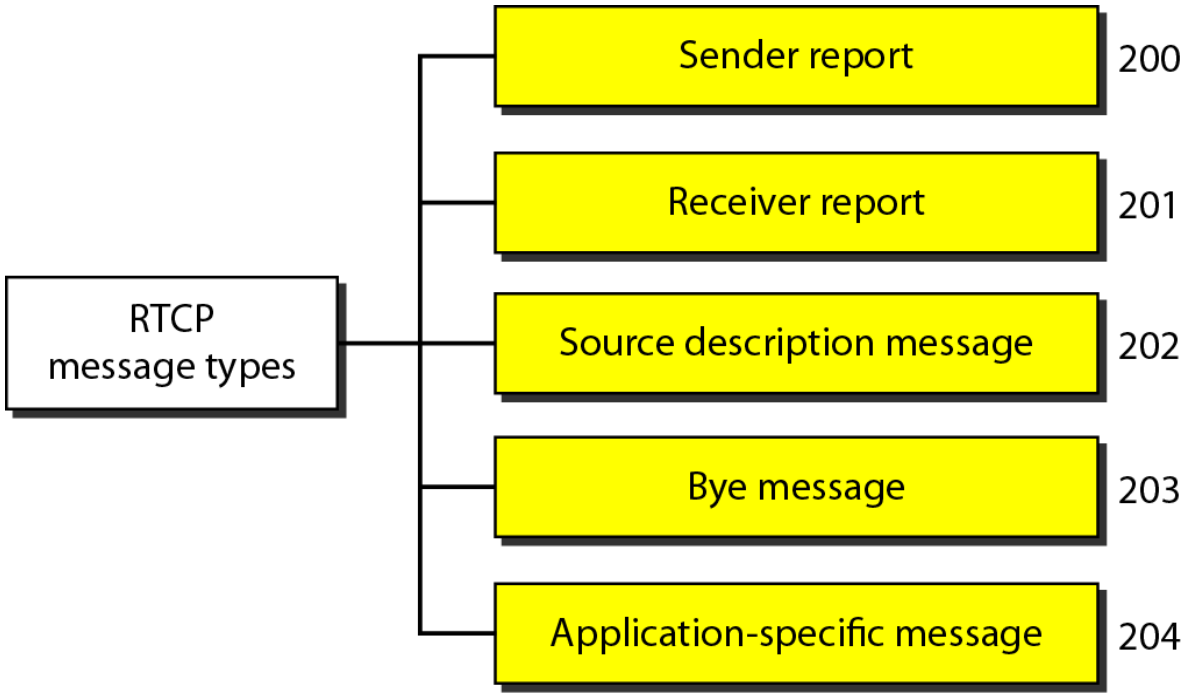


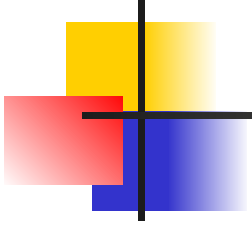
RTP uses a temporary even-numbered UDP port.

29-7 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.

Figure 29.20 *RTCP message types*





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

29-8 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. Two protocols have been designed to handle this type of communication: SIP and H.323.

Figure 29.21 *SIP messages*

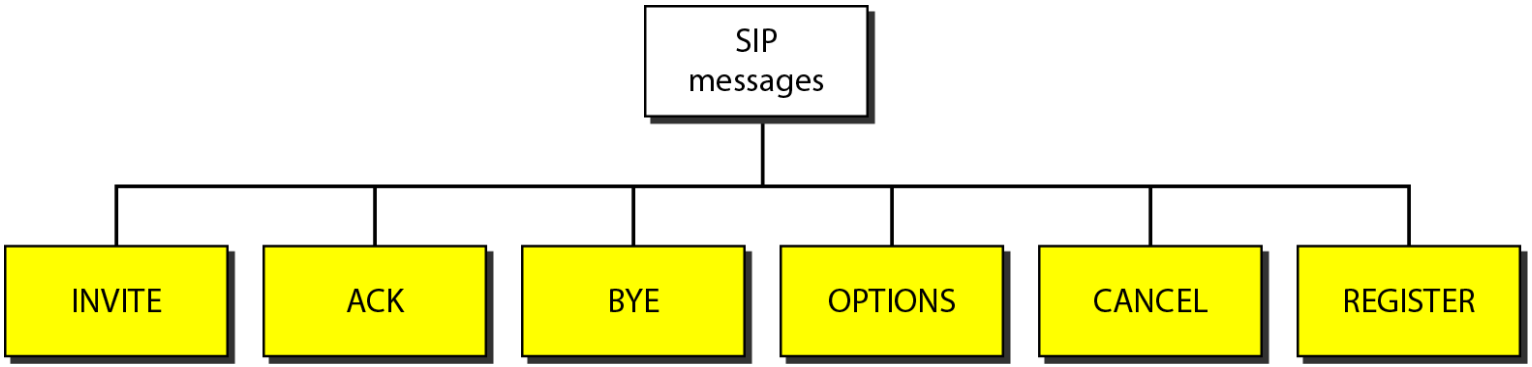


Figure 29.22 *SIP formats*

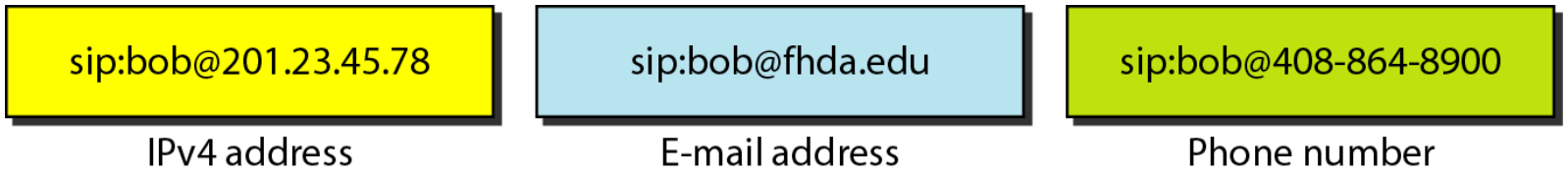


Figure 29.23 *SIP simple session*

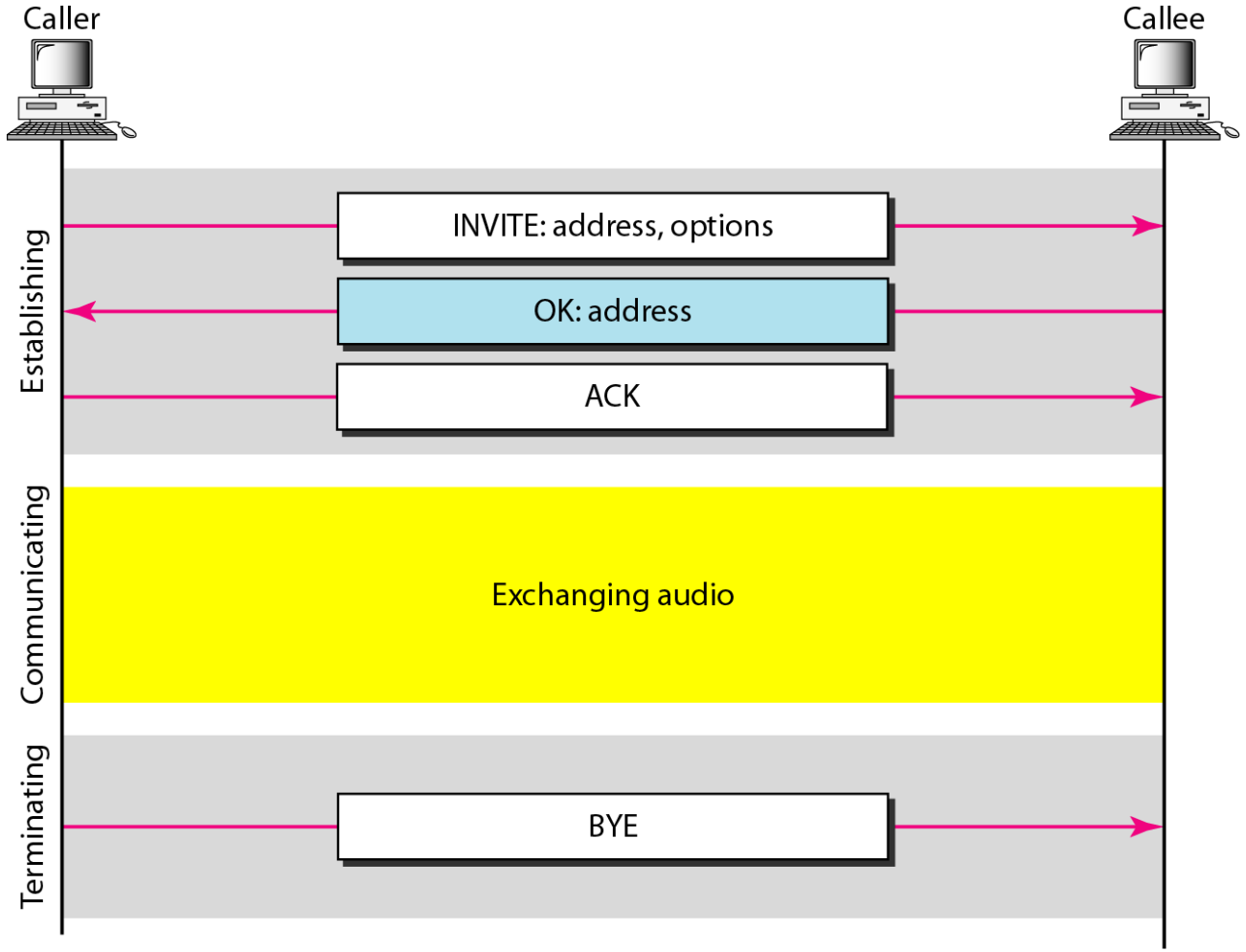


Figure 29.24 *Tracking the callee*

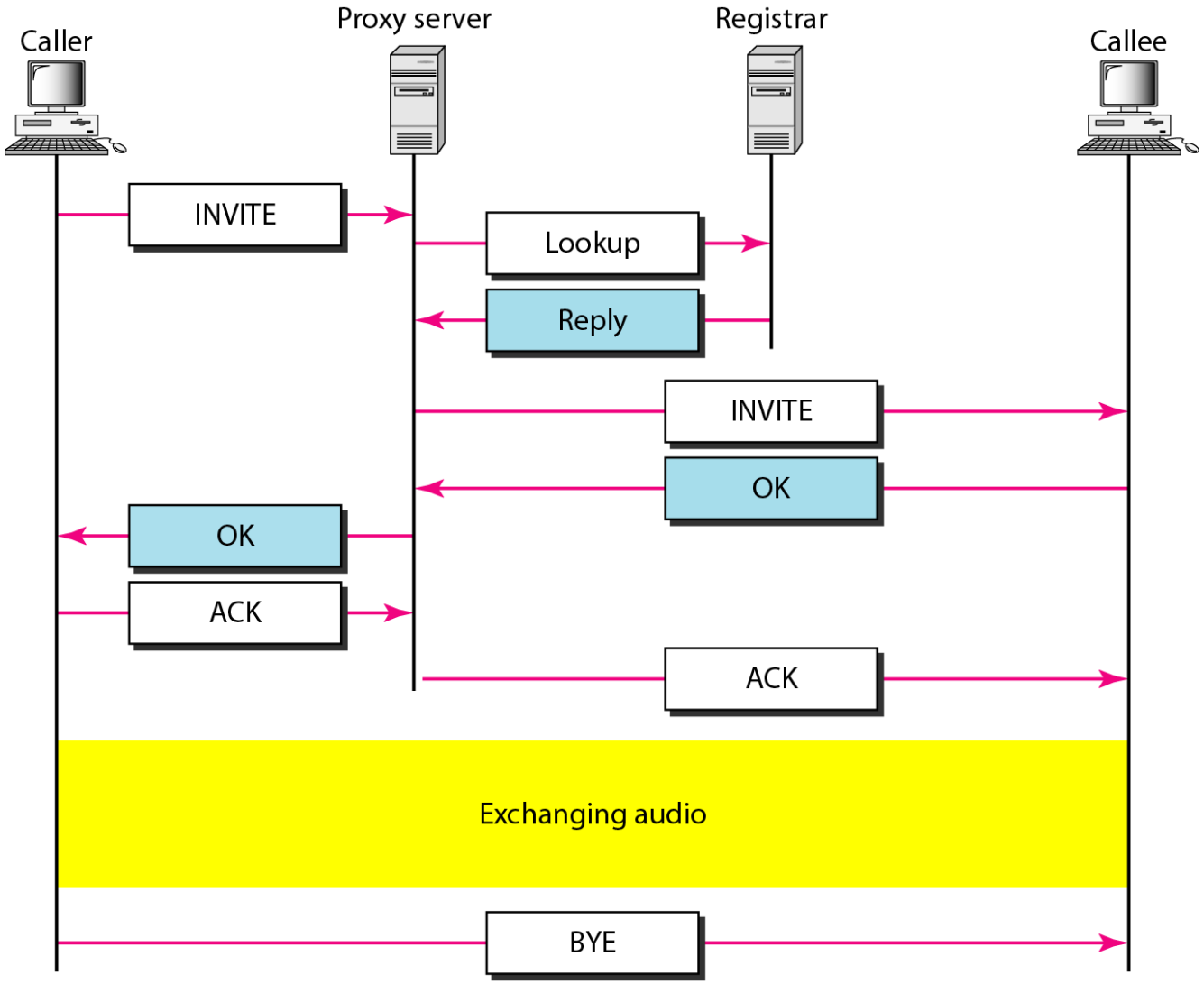


Figure 29.25 *H.323 architecture*

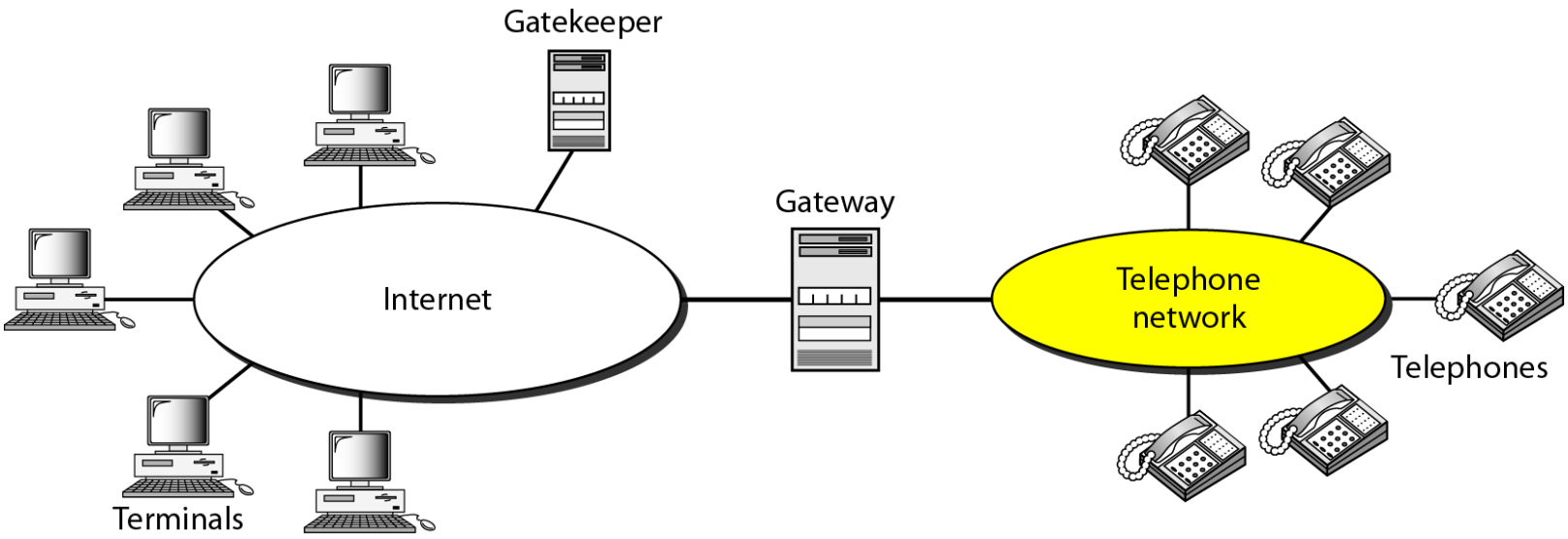


Figure 29.26 *H.323 protocols*

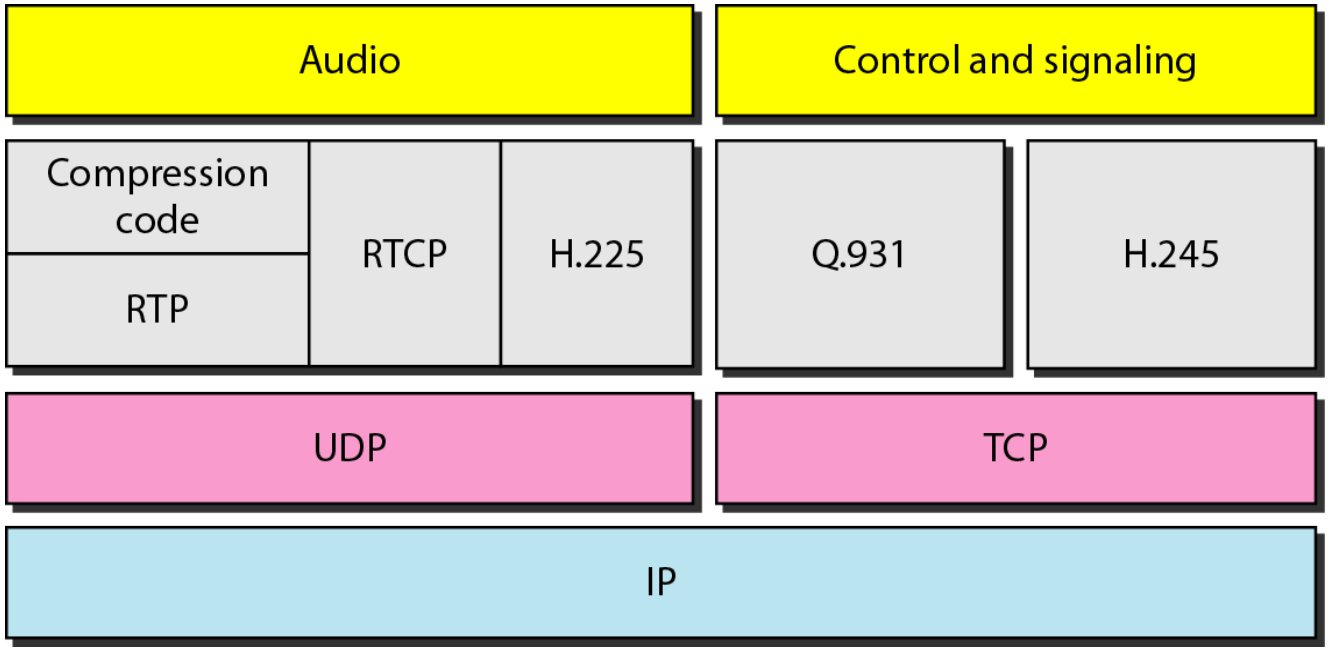


Figure 29.27 *H.323 example*