Data and Computer Communications

Tenth Edition

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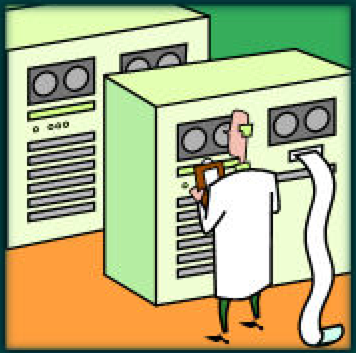
CHAPTER 25

Internet Multimedia Support

"One can now picture a future investigator in his laboratory. His hands are free, and he is not anchored. As he moves about and observes, he photographs and comments. Time is automatically recorded to tie the two records together. If he goes into the field, he may be connected by radio to his recorder. As he ponders over his notes in the evening, he again talks his comments into the record. His typed record, as well as his photographs, may both be in miniature, so that he projects them for examination."



—*As We May Think,*VannevarBush, *The Atlantic*, July 1945

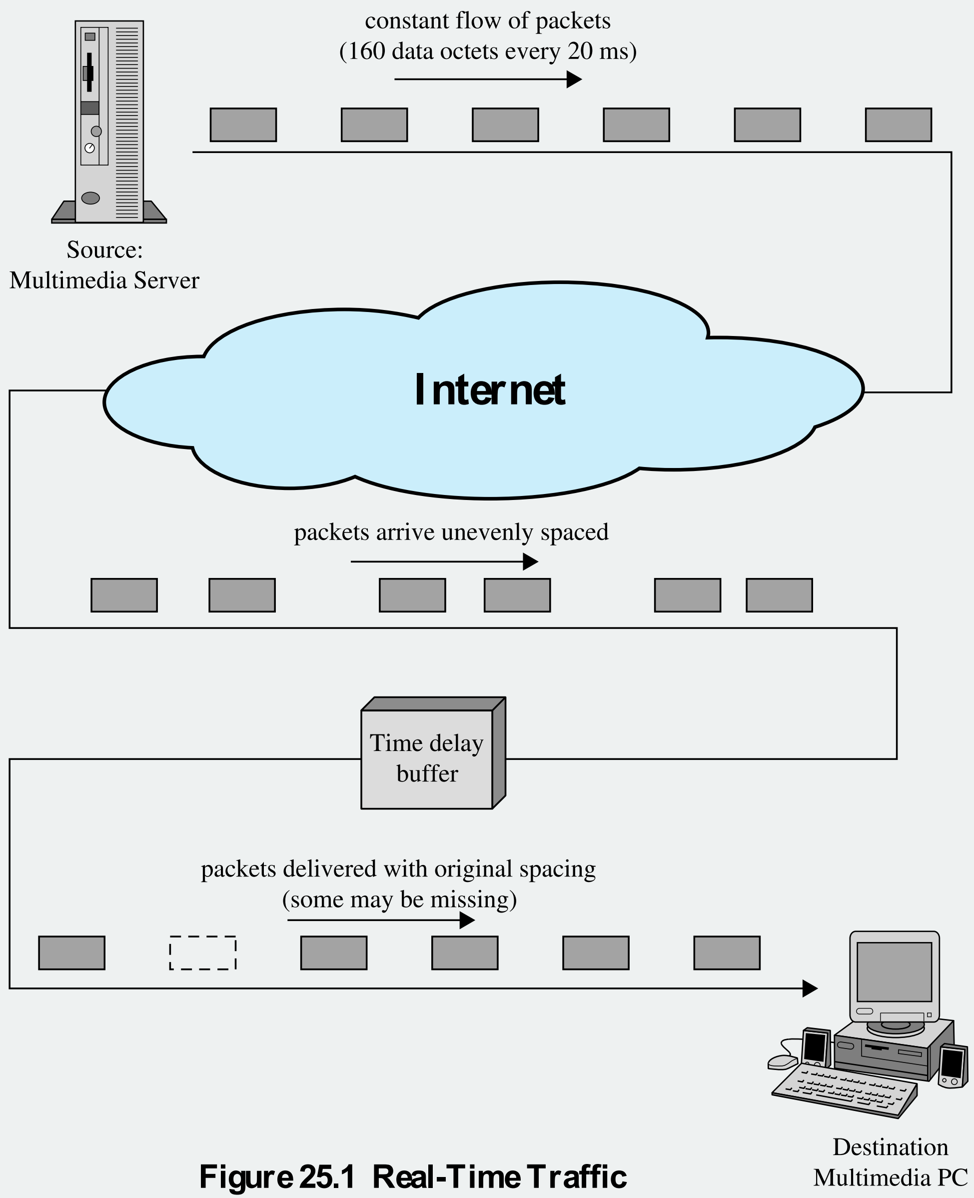
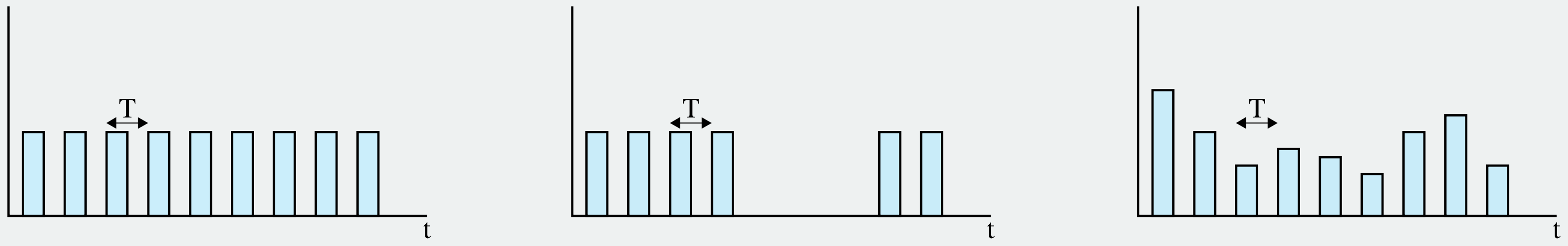


Figure 25.1 Real-Time Traffic



(a) Continuous data source

(b) Voice source with silent intervals

(c) Compressed video source

Figure 25.2 Real-Time Packet Transmission (based on [ARAS94])

Requirements for Real-Time Communication

- Low jitter
- Low latency
- Ability to easily integrate non-real-time and real-time services
- Adaptable to dynamically changing network and traffic conditions
- High effective capacity utilization
- Good performance for large networks and large numbers of connections
- Modest buffer requirements within the network
- Low overhead in header bits per packet
- Low processing overhead per packet within the network and at the end system

Hard Versus Soft Real-Time Applications

Soft

- Can tolerate the loss of some portion of the communicated data
- Impose fewer requirements on the network, therefore permissible to focus on maximizing network utilization, even at the cost of some lost or misordered packets

Hard

- Have zero loss tolerance
- A deterministic upper bound on jitter and high reliability takes precedence over network utilization considerations

Voice Over IP (VoIP)

- The transmission of speech across IP-based network
- Works by encoding voice information into a digital format, which can be carried across IP networks in discrete packets
- Has two main advantages over traditional telephony:
  - Is usually cheaper to operate than an equivalent telephone system with a PBX and conventional telephone network service
  - Readily integrates with other services, such as combining Web access with telephone features through a single PC or terminal

VoIP Signaling

- Before voice can be transferred using VoIP a call must be placed
- The calling user supplies the phone number of a URI which then triggers a set of protocol interactions resulting in the placement of the call
- The heart of the call placement process is the Session Initiation Protocol (SIP)

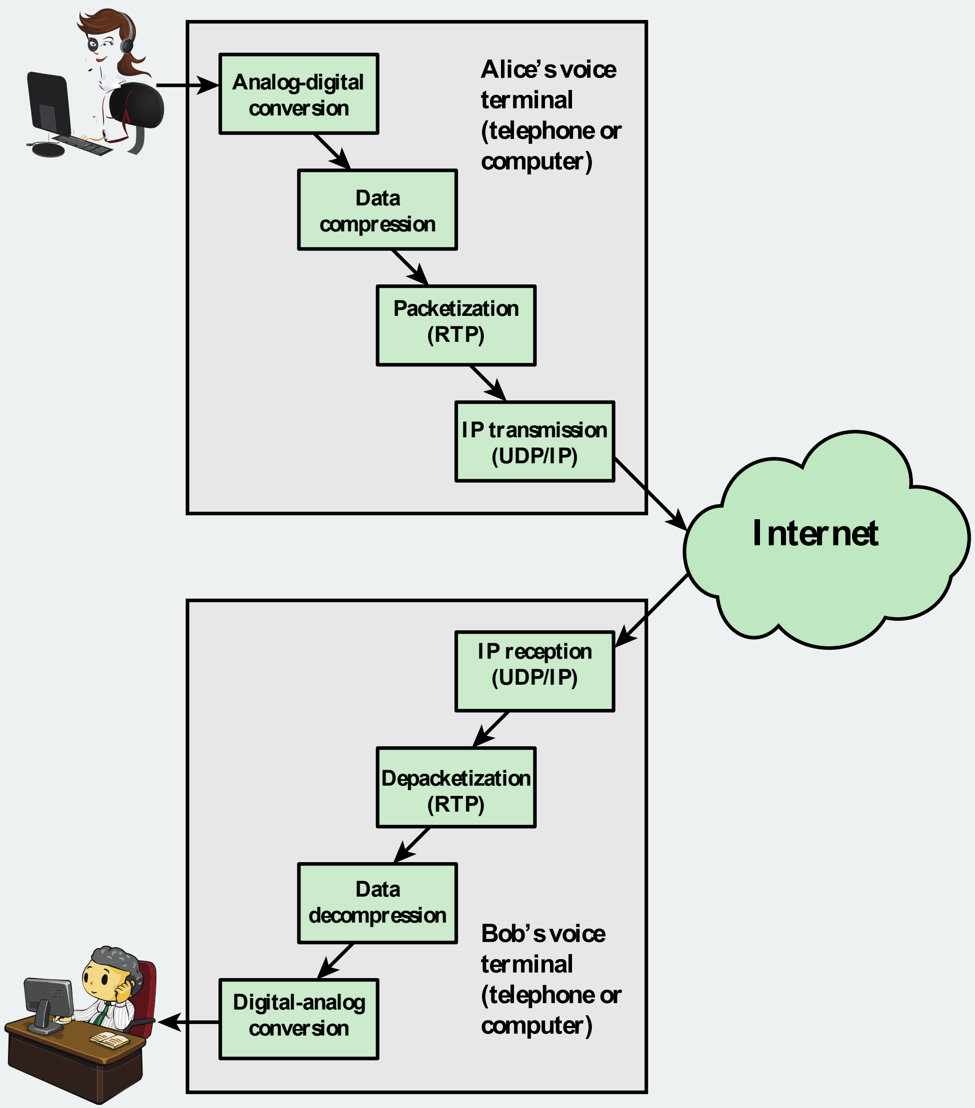


Figure 25.3 VoIP Processing

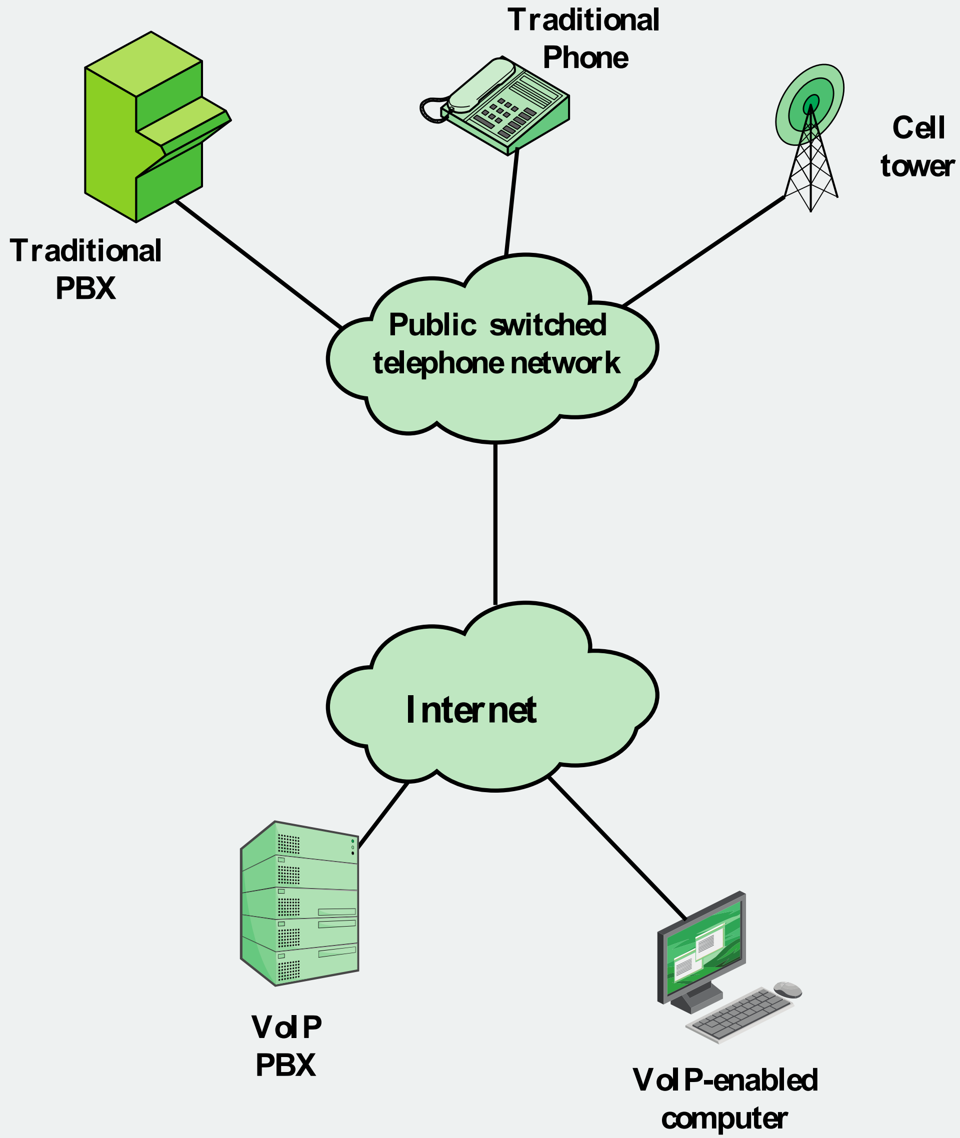
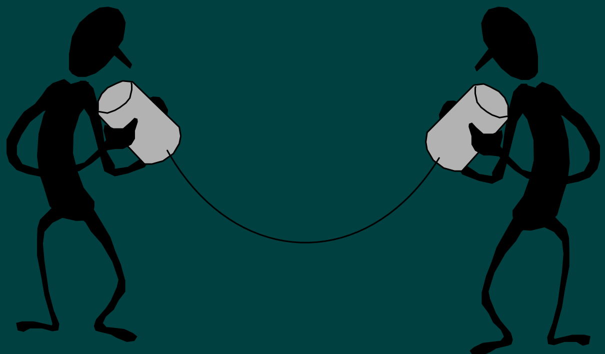


Figure 25.4 VoIP Context

VoIP Context

- The deployment of the VoIP infrastructure has been accompanied by a variety of end-user products including:
  - Traditional telephone handset
  - Conferencing units
  - Mobile units
  - Softphone
- Infrastructure equipment developed to support VoIP:
  - IP PBX
  - Media gateway



Session Initiation Protocol (SIP)

- Defined in RFC 3261
- An application level control protocol for setting up, modifying, and terminating real-time sessions between participants over an IP data network
- Key driving force is to enable Internet telephony
- Can support any type of single media or multimedia session, including teleconferencing

SIP Components and Protocols

Can be viewed of consisting of components defined on two dimensions:

Client/server

- A client is any network element that sends SIP requests and receives SIP responses
- A server is a network element that receives requests in order to service them and sends back responses to those requests

Individual network elements

- User agent
- Redirect server
- Proxy server
- Registrar
- Location service
- Presence server

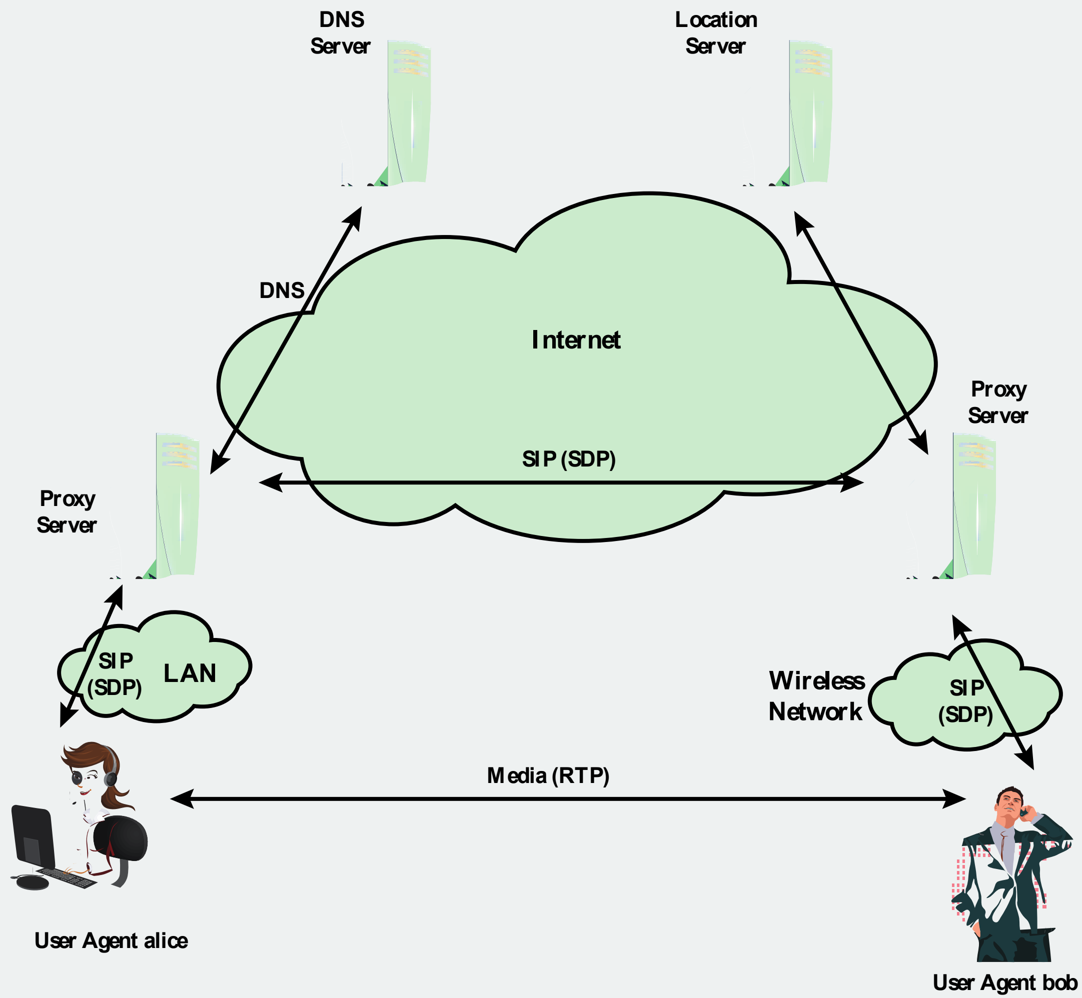
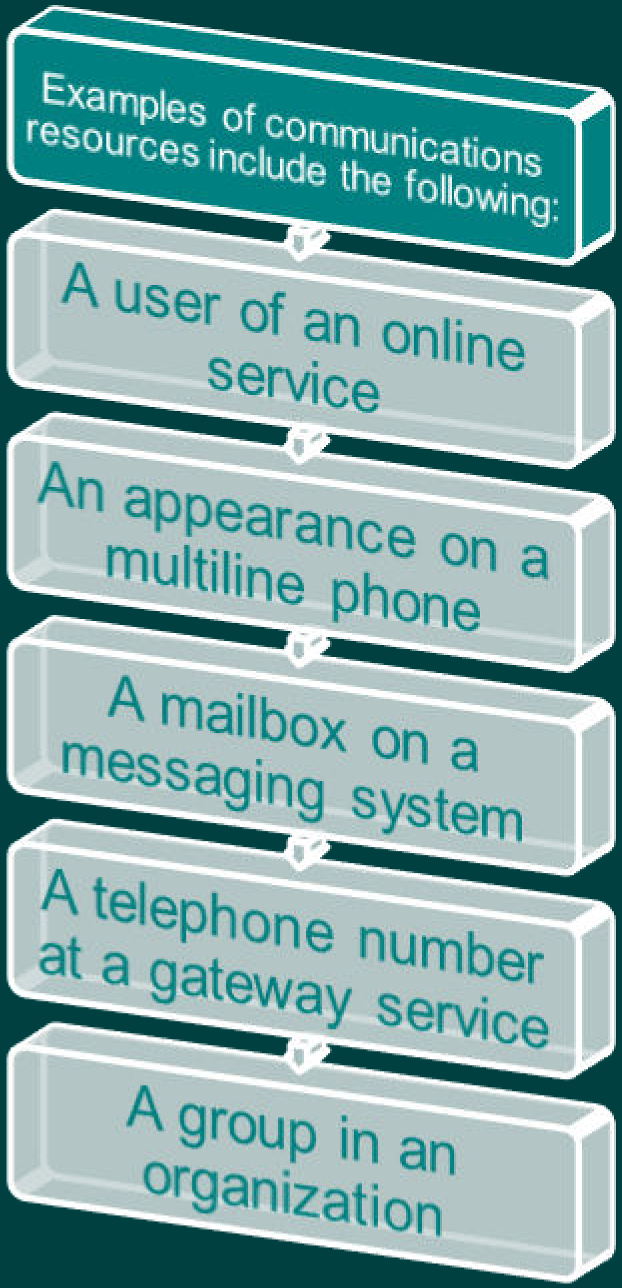


Figure 25.5 SIP Components and Protocols

SIP URI

- A resource within a SIP network is identified by a Uniform Resource Identifier
- SIP URI's have a format based on e-mail address formats
- If secure transmission is required "sips" is used
  - Transported over TLS

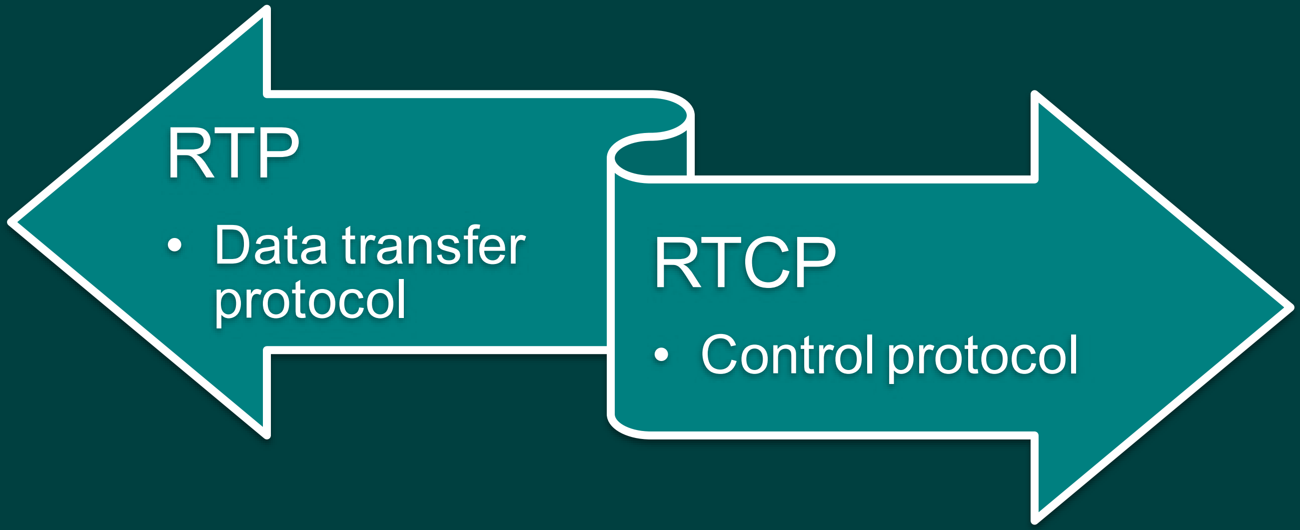


Session Description Protocol (SDP)

- Describes the content of sessions, including telephony, Internet radio, and multimedia applications
- Includes information about:
  - Media streams
  - Addresses
  - Ports
  - Payload types
  - Start and stop times
  - Originator

Real-Time Transport Protocol (RTP)

- Defined in RFC 3550
- Best suited to soft real-time communication
- Lacks the necessary mechanisms to support hard real-time traffic
- Two protocols that make up RTP are:



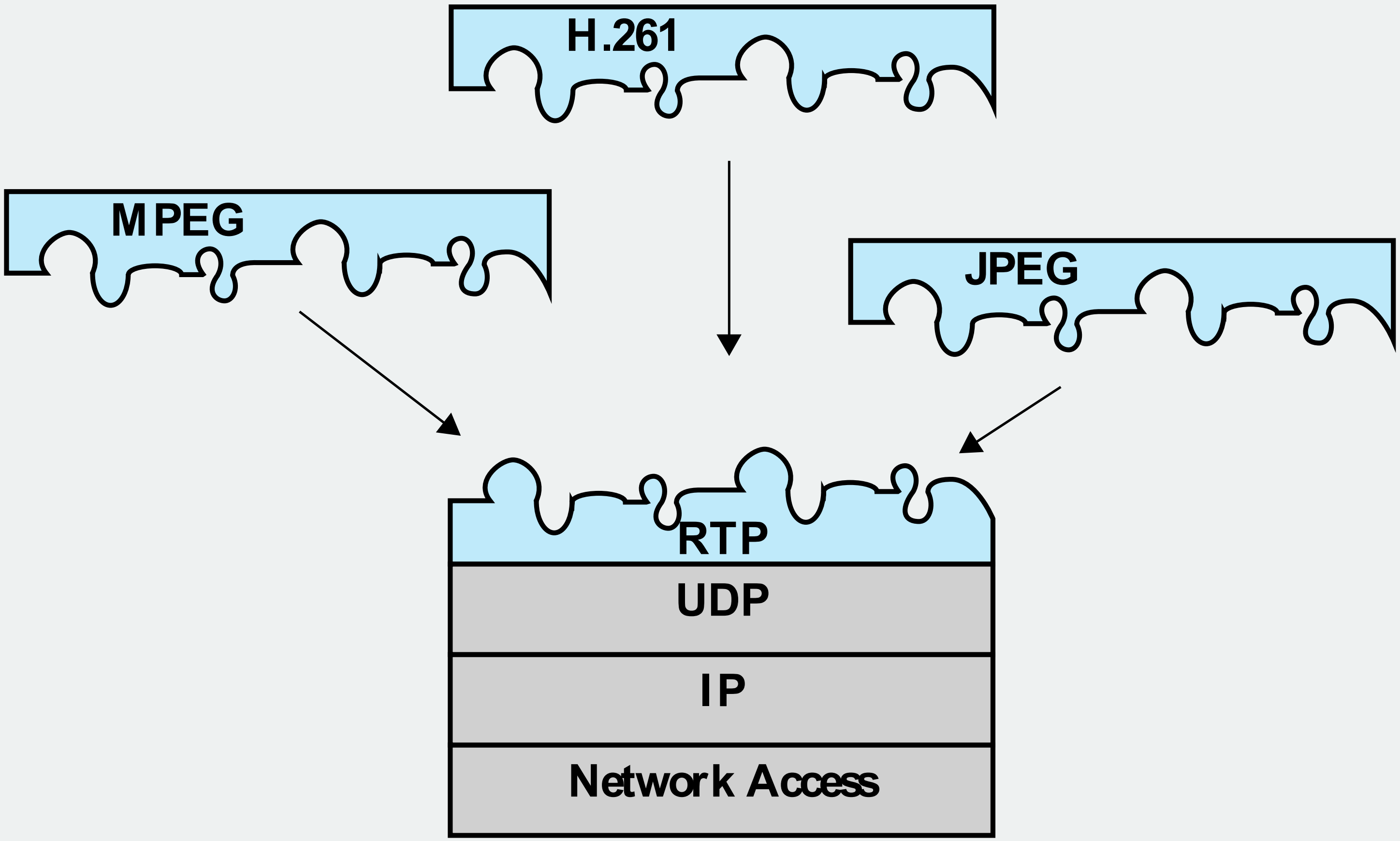


Figure 25.6 RTP Protocol Architecture [THOM96]

RTP Concepts

- RTP supports the transfer of real-time data among a number of participants in a session
  - A session is a logical association among two or more RTP entities that is maintained for the duration of the data transfer
    - Defined by:
      - RTP port number
      - RTCP port number
      - Participant IP addresses

RTP Relays

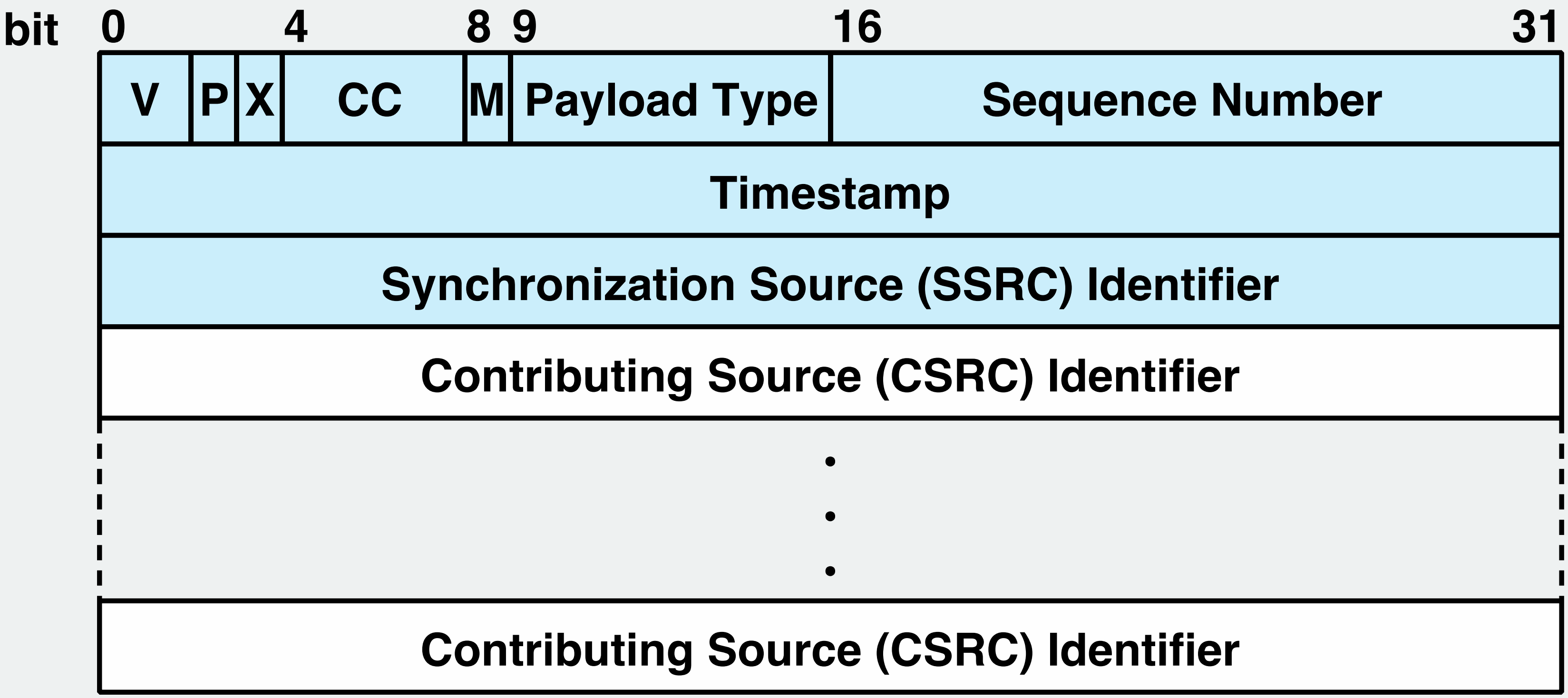
- A relay operating at a given protocol layer is an intermediate system that acts as both a destination and a source in a data transfer
- Two kinds:
  - Mixer
  - Translator

Mixer

- RTP relay that receives streams of RTP packets from one or more sources, combines these streams, and forwards a new RTP packet stream to one or more destinations
- May change the data format or simply perform the mixing function
- Provides the timing information in the combined packet stream and identifies itself as the source of synchronization

Translator

- A simple device that produces one or more outgoing RTP packets for each incoming RTP packet
- May change the format of the data in the packet or use a different lower-level protocol suite to transfer from one domain to another
- Examples of translator use include:
  - Convert a video to a lower quality format
  - If an application-level firewall prevents the forwarding of RTP packets
  - Replicate an incoming multicast RTP packet and send it to a number of unicast destinations



V = Version
P = Padding
X = Extension
CC = CSRC count
M = Marker

Figure 25.7 RTP Header

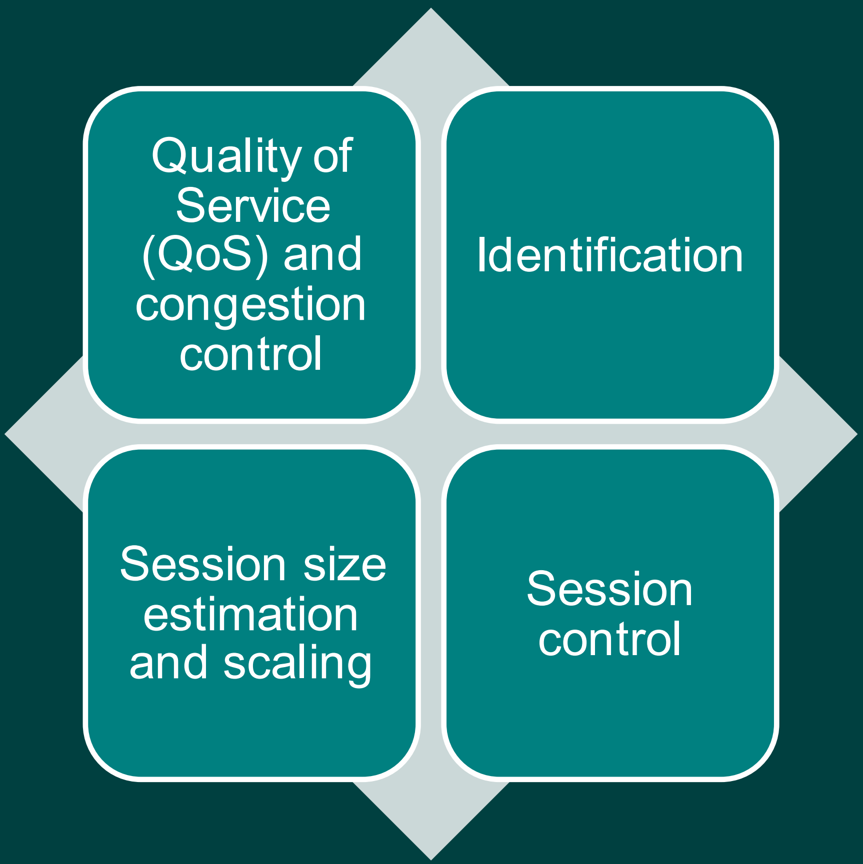
Table 25.1

Payload Types for Standard Audio and Video Encodings (RFC 3551)

0	PCMU audio
1	1016 audio
2	G721 audio
3	GSM audio
4	unassigned audio
5	DV14 audio (8 kHz)
6	DV14 audio (16 kHz)
7	LPC audio
8	PCMA audio
9	G722 audio
10	L16 audio (stereo)
11	L16 audio (mono)
12	QCELP wireless
13	Comfort noise
14	MPA audio
15	G728 audio
16-23	unassigned audio
24	unassigned video
25	CelB video
26	JPEG video
27	unassigned
28	nv video
29-30	unassigned video
31	H261 video
32	MPV video
33	MP2T video
34-71	unassigned
72-76	reserved
77-95	unassigned
96-127	dynamic

RTP Control Protocol (RTCP)

- RFC 3550 outlines four functions performed by RTCP:



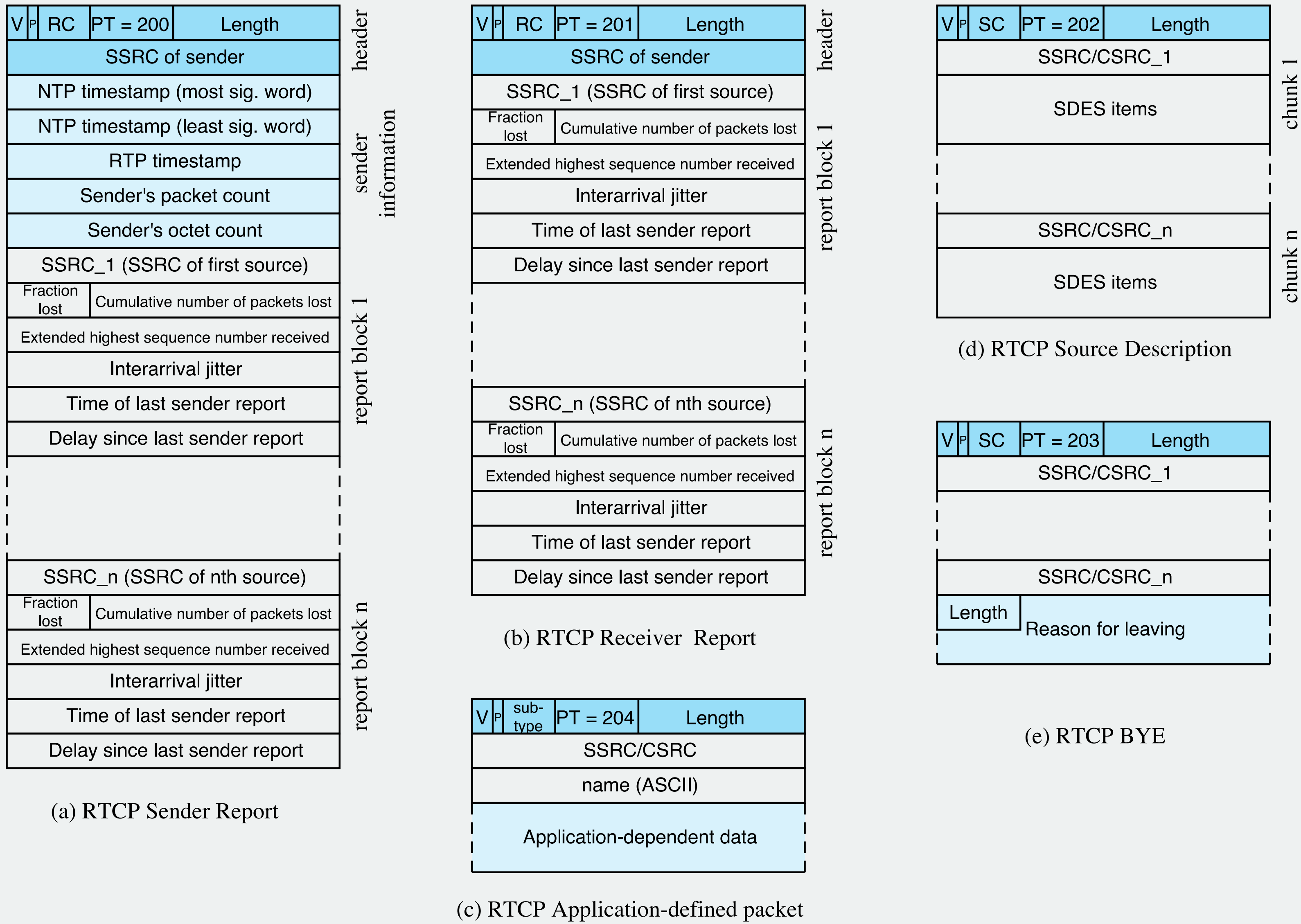


Figure 25.8 RTCP Formats

Table 25.2
SDES Types (RFC 3550)

Value	Name	Description
0	END	End of SDES list
1	CNAME	Canonical name: unique among all participants within one RTP session
2	NAME	Real user name of the source
3	EMAIL	E-mail address
4	PHONE	Telephone number
5	LOC	Geographic location
6	TOOL	Name of application generating the stream
7	NOTE	Transient message describing the current state of the source
8	PRIV	Private experimental or application-specific extensions

Summary

- Real-time traffic
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  - SIP messages
  - Session description protocol
- RTP
  - RTP protocol architecture
  - RTP data transfer protocol
  - RTCP