Data and Computer Communications

Chapter 24 – Internet Applications – Multimedia

Eighth Edition

by William Stallings

Lecture slides by Lawrie Brown

Internet Applications – Multimedia

Prior to the recent explosion of sophisticated research, scientists believed that birds required no special awareness or intelligence to perform their migrations and their navigational and homing feats. Accumulated research shows that in addition to performing the difficult tasks of correcting for displacement (by storms, winds, mountains, and other hindrances), birds integrate an astonishing variety of celestial, atmospheric, and geological information to travel between their winter and summer homes. In brief, avian navigation is characterized by the ability to gather a variety of informational cues and to interpret and coordinate them so as to move closer toward a goal.

—*The Human Nature of Birds*, Theodore Barber

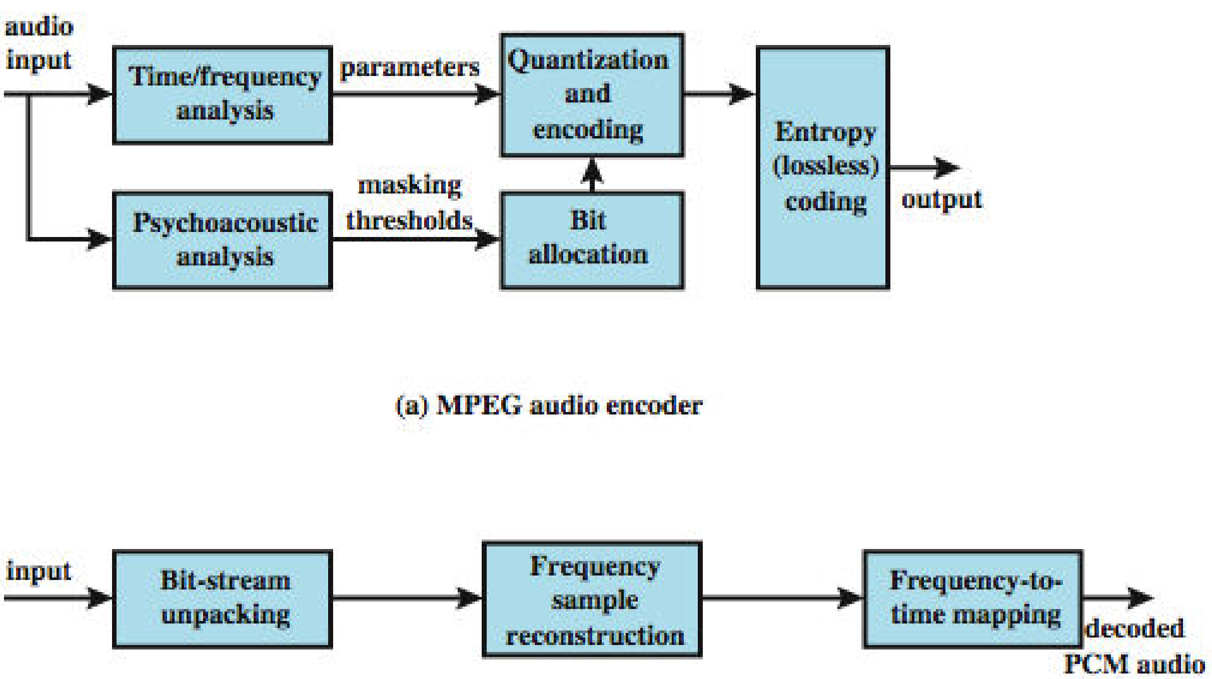
Audio and Video Compression

- multimedia applications need efficient use of transmission capacity
- hence audio/video compression algorithms
- techniques standardized by MPEG
- lossless compression loses no information
  - limited by redundancy in original data
- lossy compression provides acceptable approximation to original (typically use)

Simple Audio Compression

- must first digitize audio signal, eg. PCM
  - sample at twice highest frequency
  - then quantize using fixed number of bits
  - effectively compression algorithm
  - otherwise need unlimited number of bits
- compress further by reducing sampling frequency or number of bits
- or use more sophisticated approaches
  - as in MPEG Layer 3 (MP3) giving 10:1 compression

Effective Audio Compression



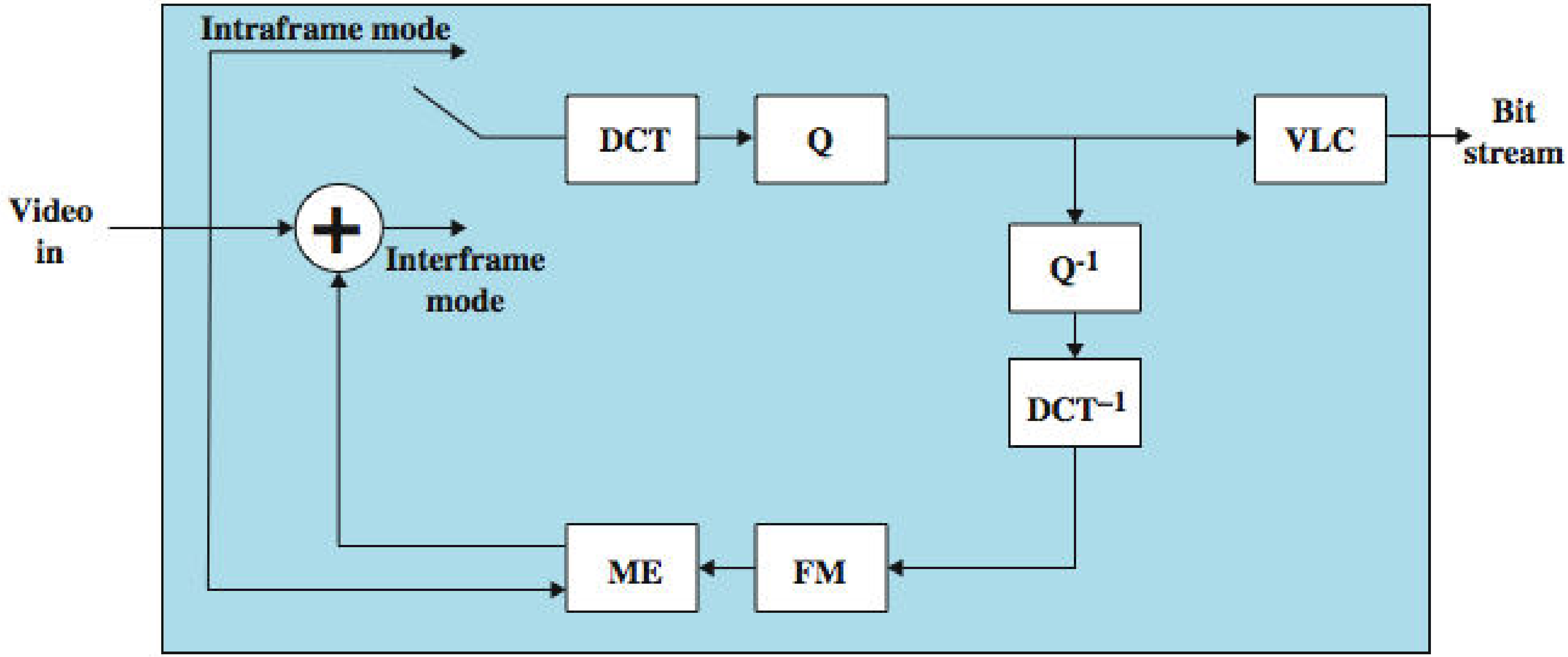
(a) MPEG audio encoder

(b) MPEG audio decoder

Video Compression

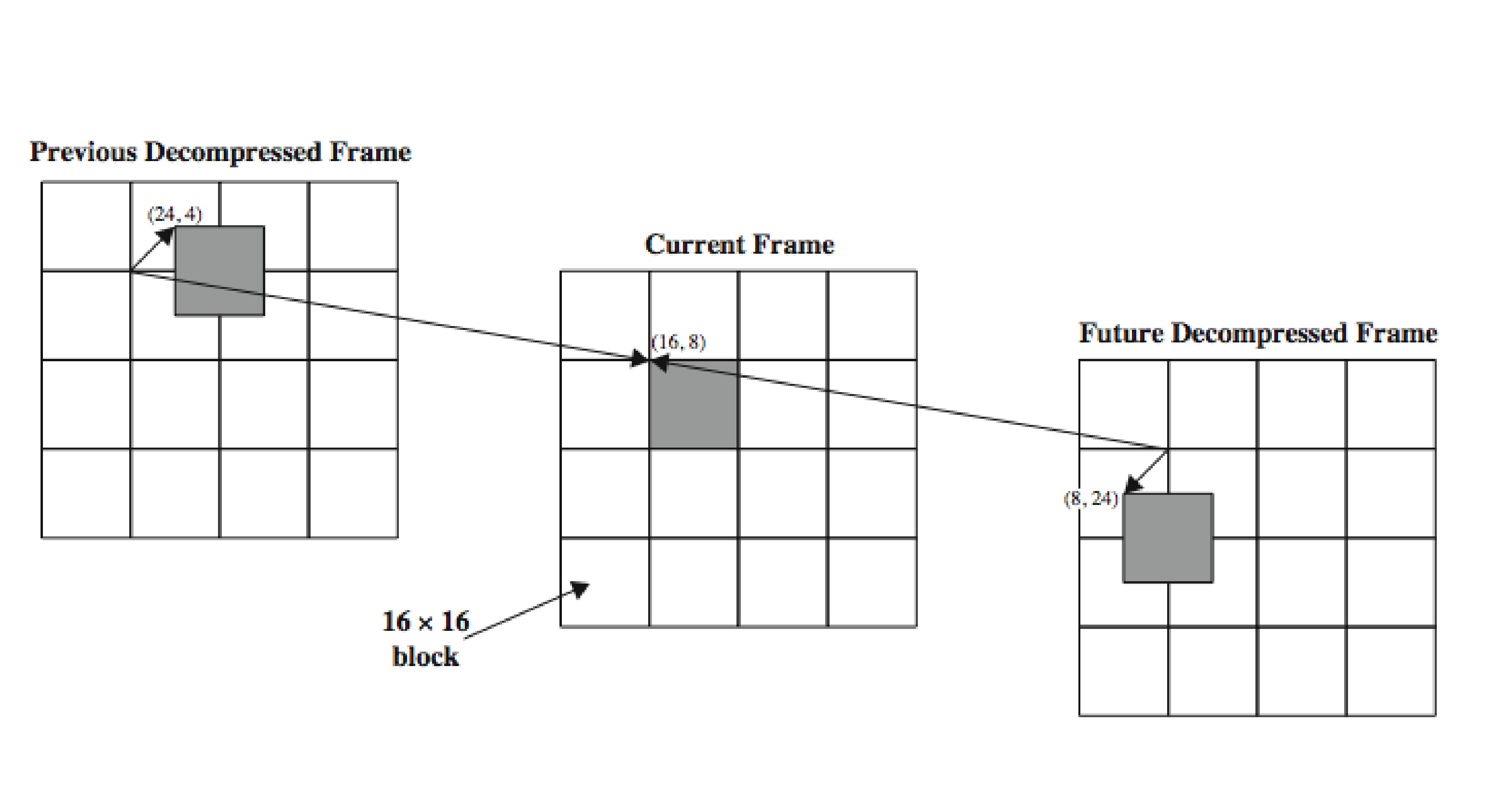
- moving picture a sequence of still images
- hence can compress each individually
- but get greater efficiency by using similarities between adjacent images
- encode just differences between them
- approach used in MPEG

MPEG Video Compression



DCT: Discrete cosine transform
Q: quantizer
VLC: variable length coder
FM: frame memory
ME: motion estimator

MPEG Video Compression



MPEG Video Compression

- important features in video compression
  - random access - needs access frames
  - fast forward / reverse - scan stream using access frames
- MPEG foundation is motion compensation
  - prediction
  - interpolation

Prediction

- MPEG uses 16x16 pixel macroblocks for motion compensation
- each block encoded separately
- with reference to preceeding anchor frame most closely matching it
  - matching block not on 16-pixel boundary
  - compare against decompressed frame
- MPEG then records motion vector and prediction error for current frame

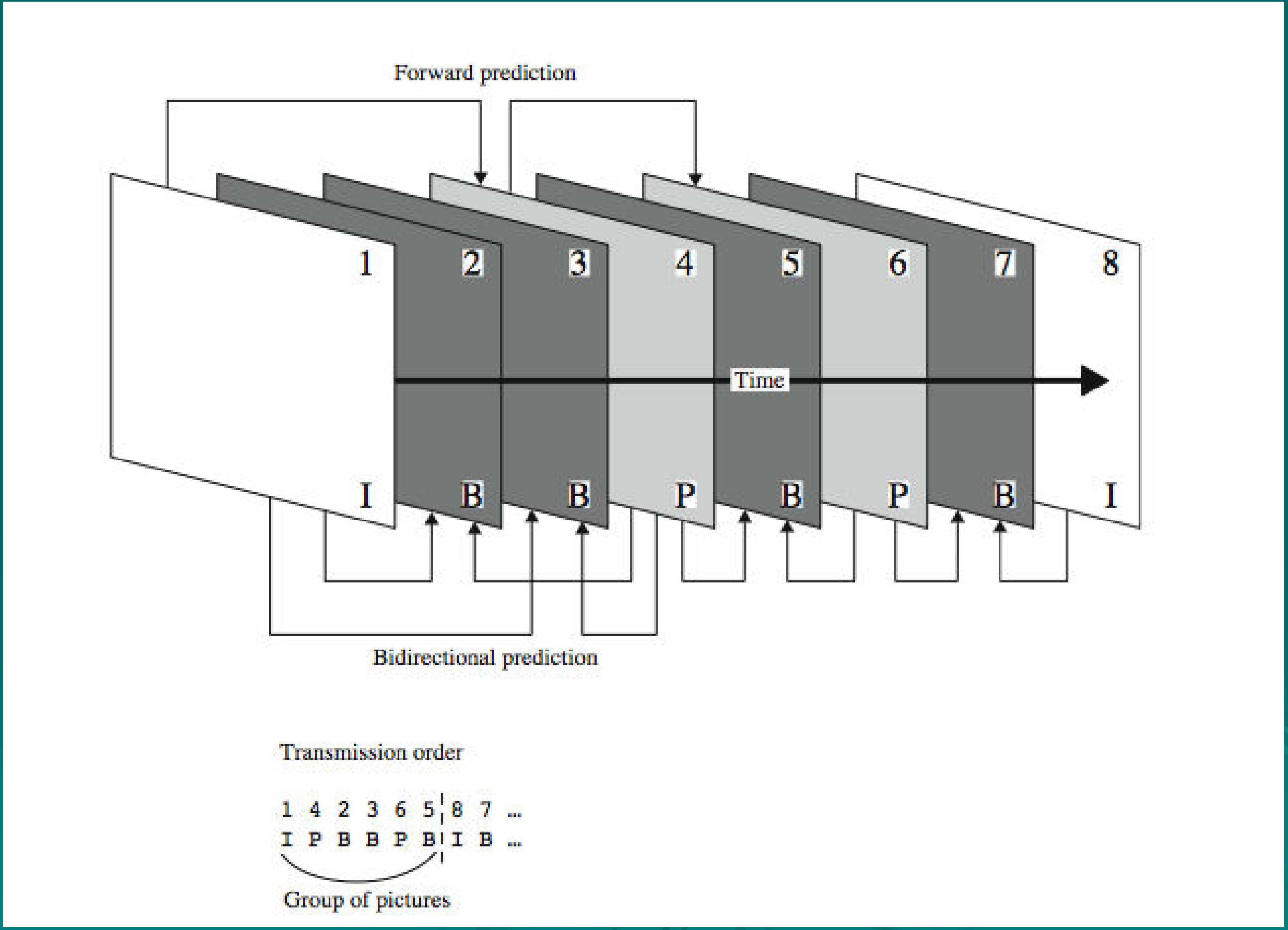
Interpolation

- have further compression improvement by using two reference frames
  - bidirectional interpolation
- process current against frames before and after
- encode using:
  - block from before (forward prediction)
  - block from after (backward prediction)
  - average of blocks before and after (averaging)
- interpolation encodes more info than prediction

MPEG Frame Ordering

- MPEG uses three types of frames:
  - intraframe (I)
  - predicted (P)
  - bidirectional interpolated (B)
- relative frequency is configurable
  - balance need for random access and FF/Rev with computational complexity and size
  - noting B frames rely only on I and P frames

MPEG Frame Ordering



Real-Time Traffic

- increasing deployment of high-speed nets sees increasing real-time traffic use
- has different requirements to traditional non real-time traffic
  - traditionally throughput, delay, reliability
  - real-time more concerned with timing issues
  - with deadline for delivery of data block

Real-Time Traffic Example

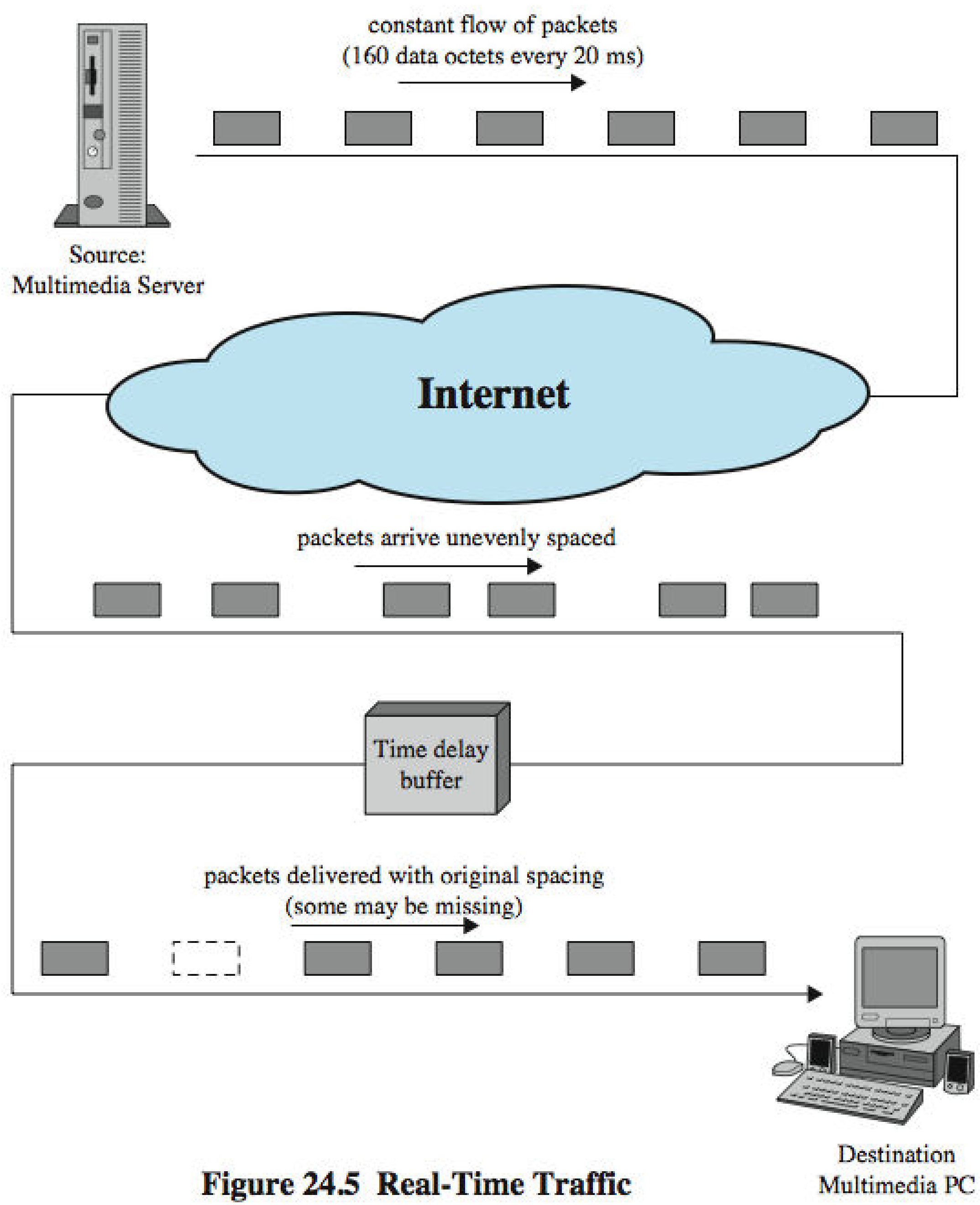
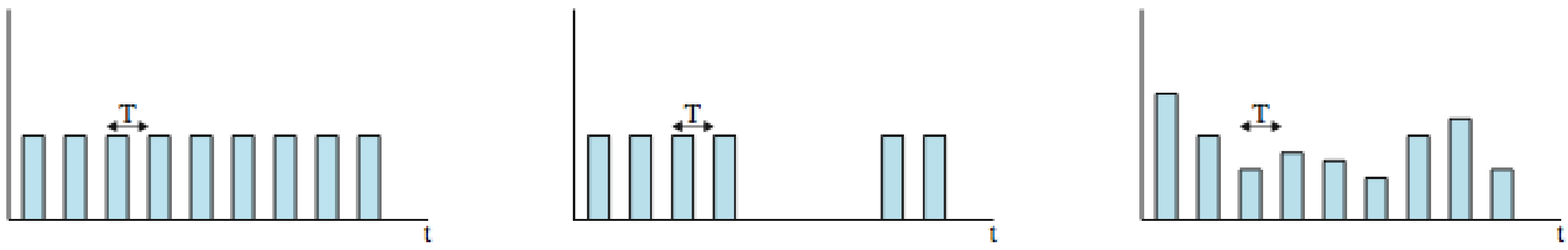


Figure 24.5 Real-Time Traffic

Real-Time Traffic Profiles



(a) Continuous data source

(b) Voice source
with silent intervals

(c) Compressed video source

Real-Time Traffic Requirements

- low jitter
- low latency
- integrate non-real-time and real-time services
- adapts to changing network / traffic conditions
- good performance for large nets / connections
- modest buffer requirements within the network
- high effective capacity utilization
- low overhead in header bits per packet
- low processing overhead

Hard vs Soft Real-Time Apps

- soft real-time applications
  - tolerate loss of some data
  - hence impose fewer requirements on network
  - can focus on maximizing network utilization
- hard real-time applications
  - zero loss tolerance
  - hence deterministic upper bound on jitter and high reliability take precedence over utilization

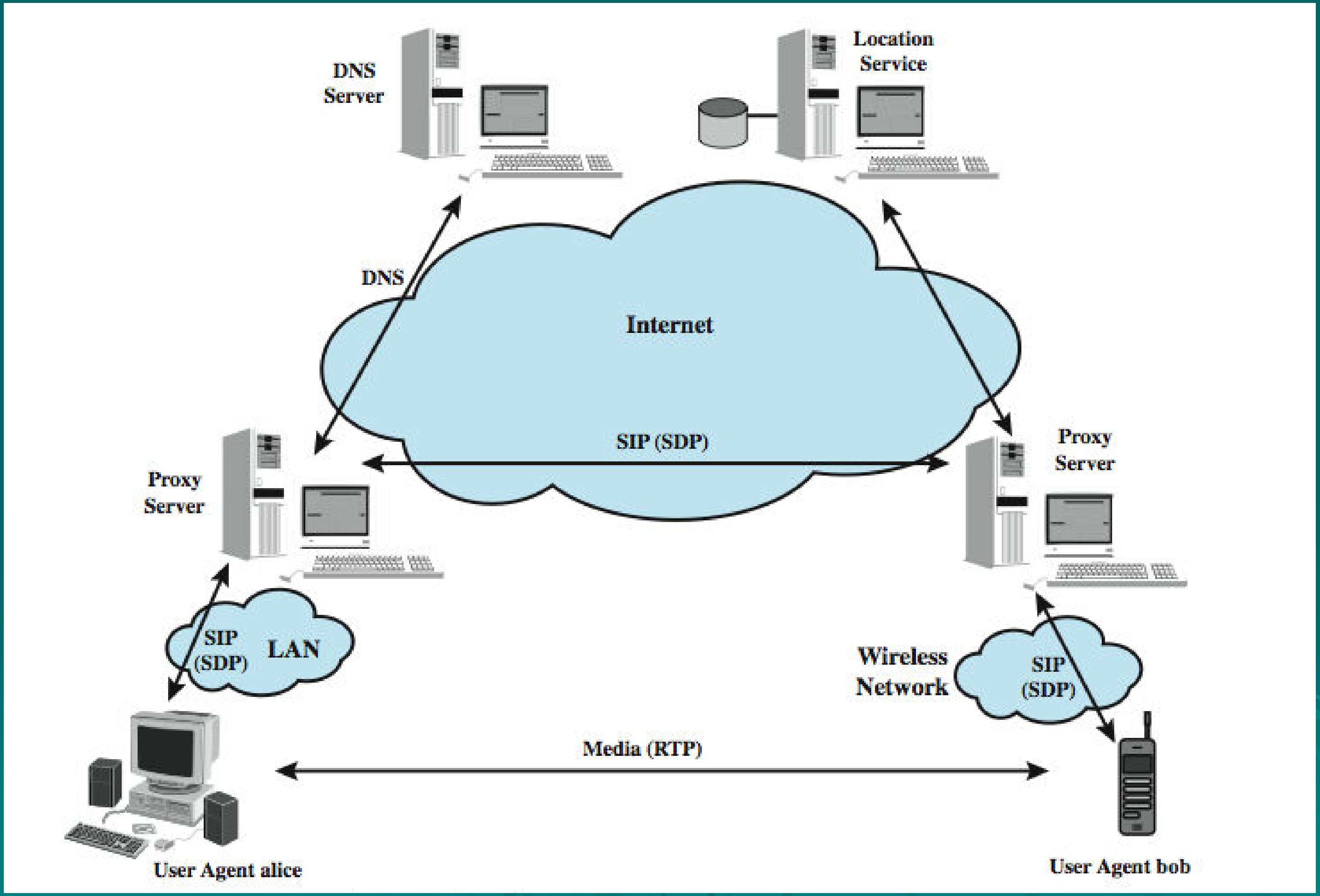
Session Initiation Protocol (SIP)

- control protocol for setting up, modifying, and terminating real-time sessions
- defined in RFC 3261
- five multimedia communications facets:
  - user location
  - user availablility
  - user capabilities
  - session setup
  - session management

SIP Design Elements

- based on earlier protocols
- HTTP request/response transaction model
  - client invokes server method/function
  - receives at least one response
  - using most header fields, encoding rules, and status codes of HTTP
- DNS like recursive and iterative searches
- incorporates the use of a Session Description Protocol (SDP)

SIP Components



SIP Servers and Protocols

- servers are logical devices
  - may be distinct servers or combined in one
- user agent uses SIP to setup session
- initiation dialogue uses SIP involving one or more proxies to relay to remote agent
- proxies act as redirect servers if needed
  - consulting location service DB
  - protocol used here outside SIP
  - DNS also important
- SIP uses UDP for performance reasons
- can use TLS for security if desired

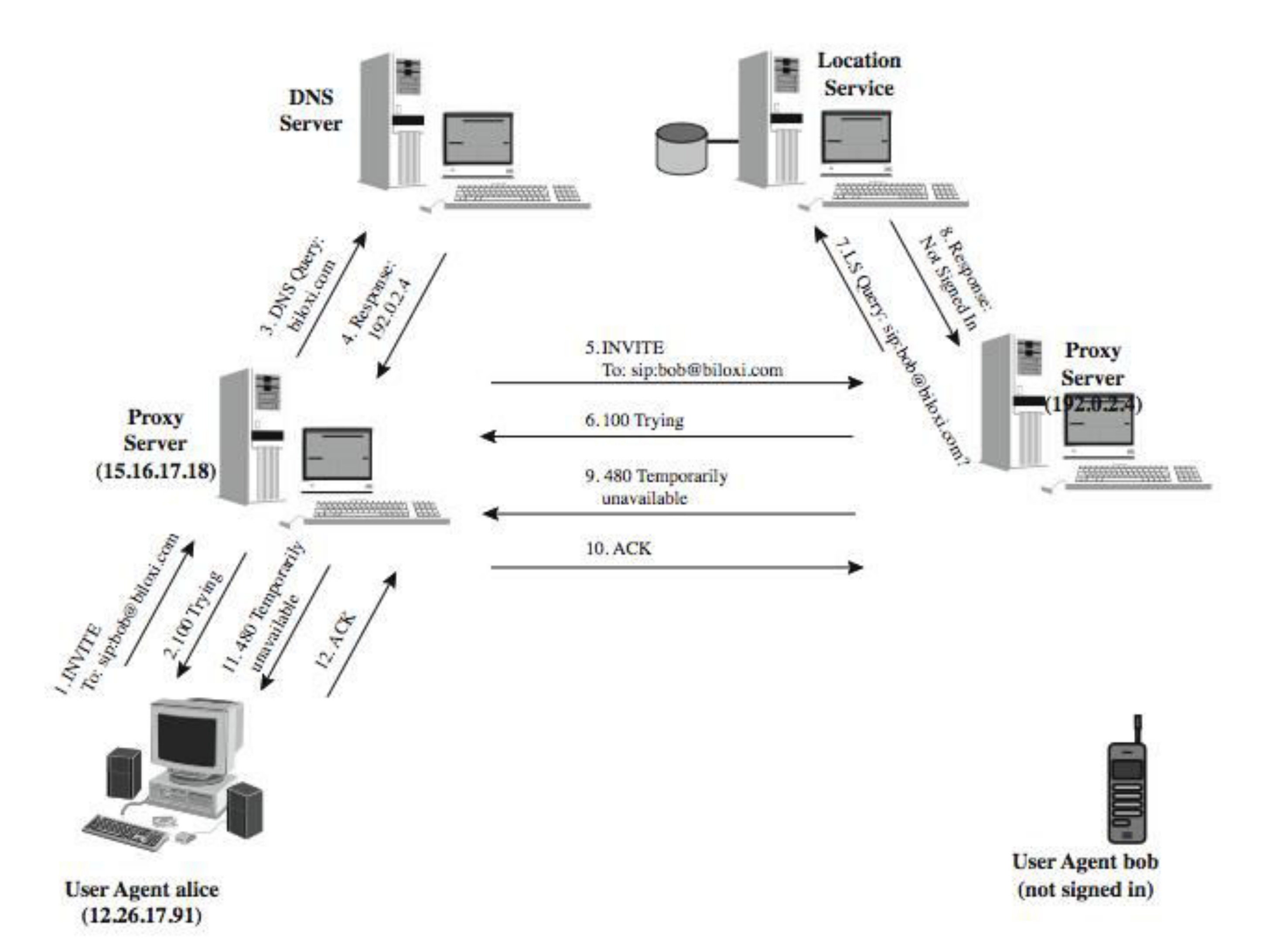
Session Description Protocol (SDP)

- defined in RFC 2327
- have SDP encoded body in SIP message
- specifies information on media encodings parties can and will use
- after exchange parties know IP addresses, transmission capacity, media types
- may then exchange data using a suitable transport protocol, eg. RTP
- change session parameters with SIP messages

SIP Uniform Resource Identifier (URI)

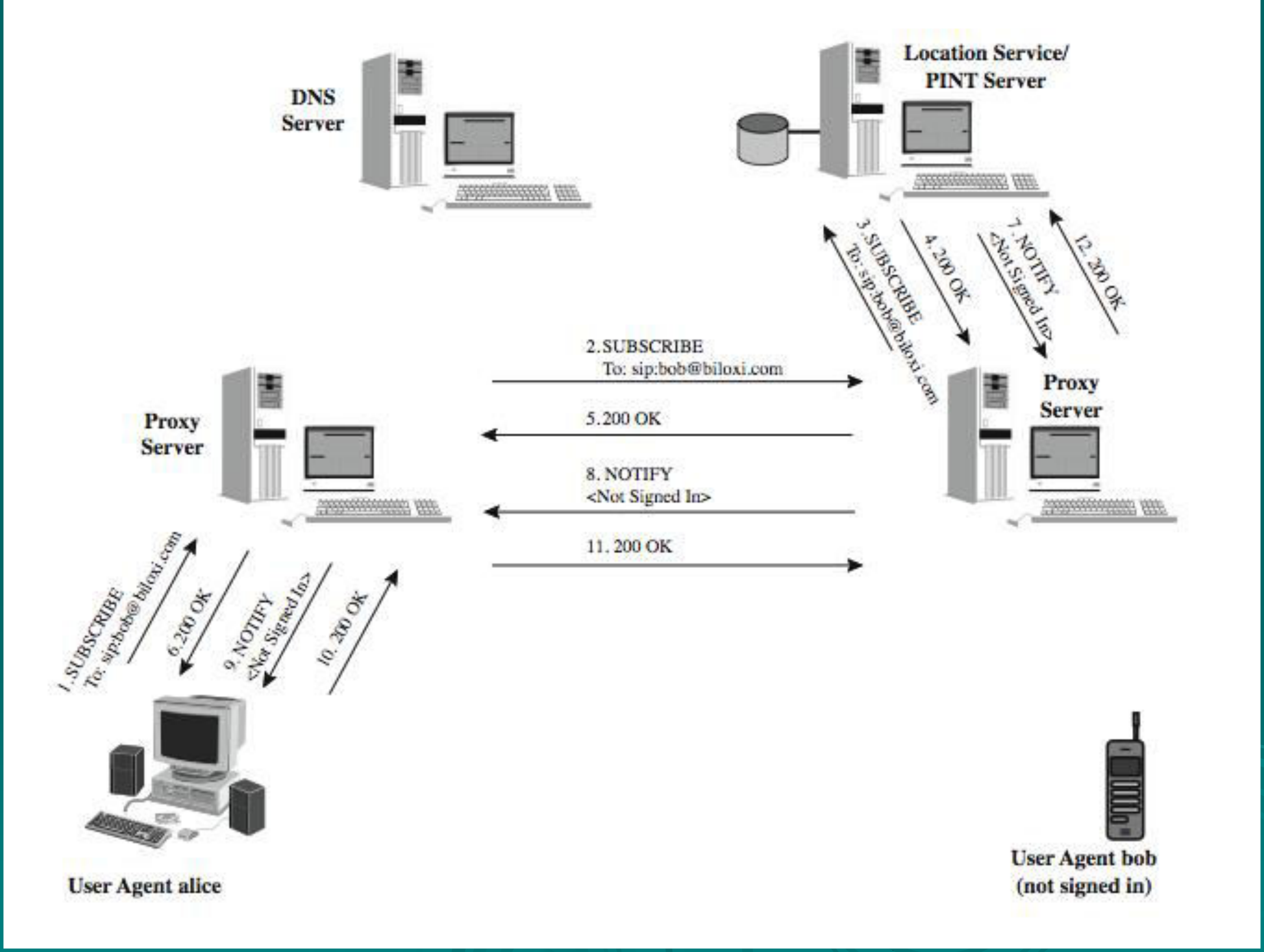
- identifies a resource within a SIP network
  - eg. user, mailbox, phone number, group
- format based on email address
  - eg. sip:bob@biloxi.com
- may also include password, port number and other parameters
- “sips” for secure transmission over TLS

SIP Example



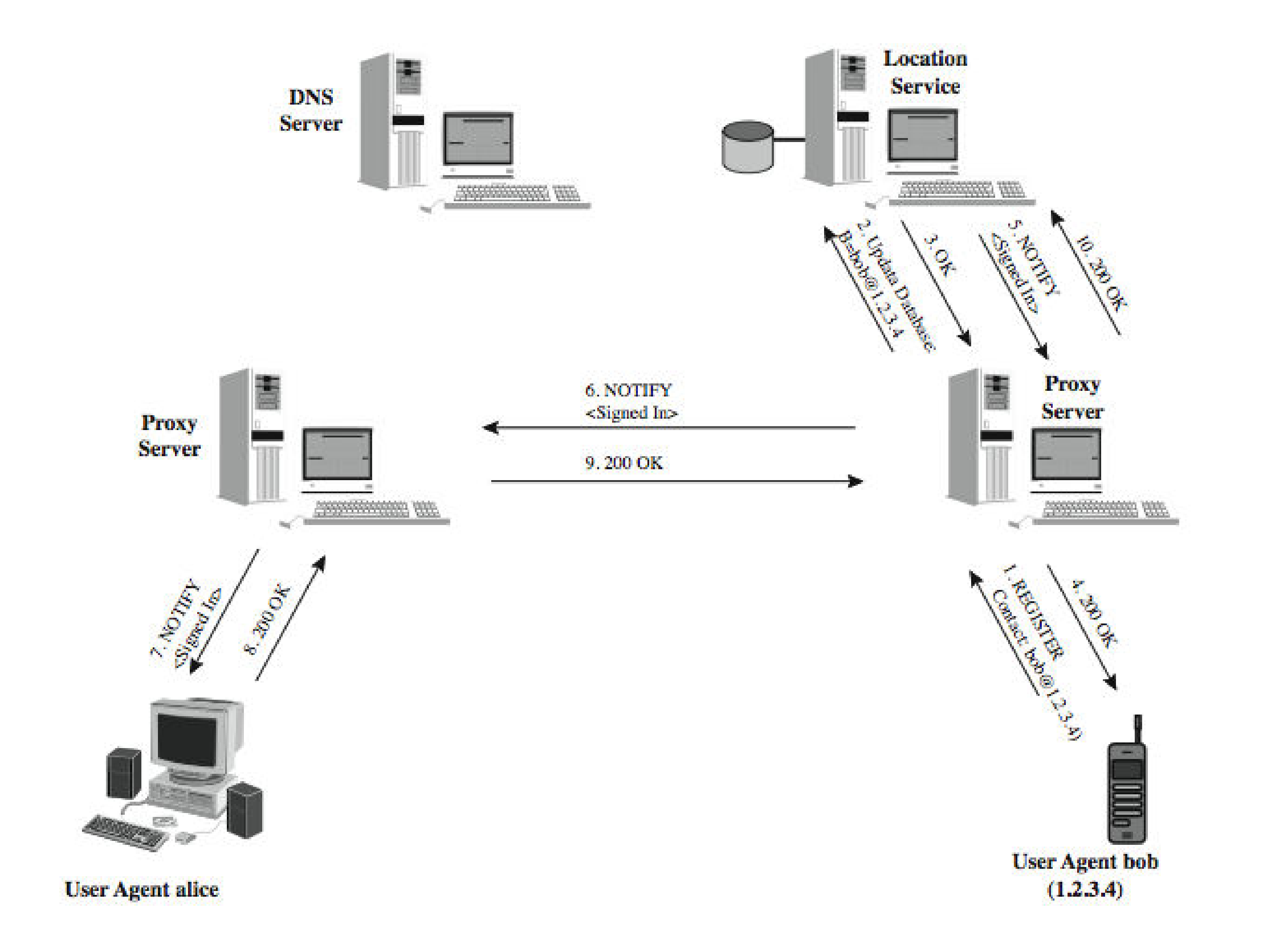
DNS Server
Location Service
3. DNS Query: biloxi.com
4. Response: 192.0.2.4
7. LS Query: sip:bob@biloxi.com?
8. Response: Not Signed In
Proxy Server (192.0.2.4)
5. INVITE
To: sip:bob@biloxi.com
6. 100 Trying
9. 480 Temporarily unavailable
10. ACK
Proxy Server (15.16.17.18)
1. INVITE To: sip:bob@biloxi.com
2. 100 Trying
11. 480 Temporarily unavailable
12. ACK
User Agent alice (12.26.17.91)
User Agent bob (not signed in)

SIP Example

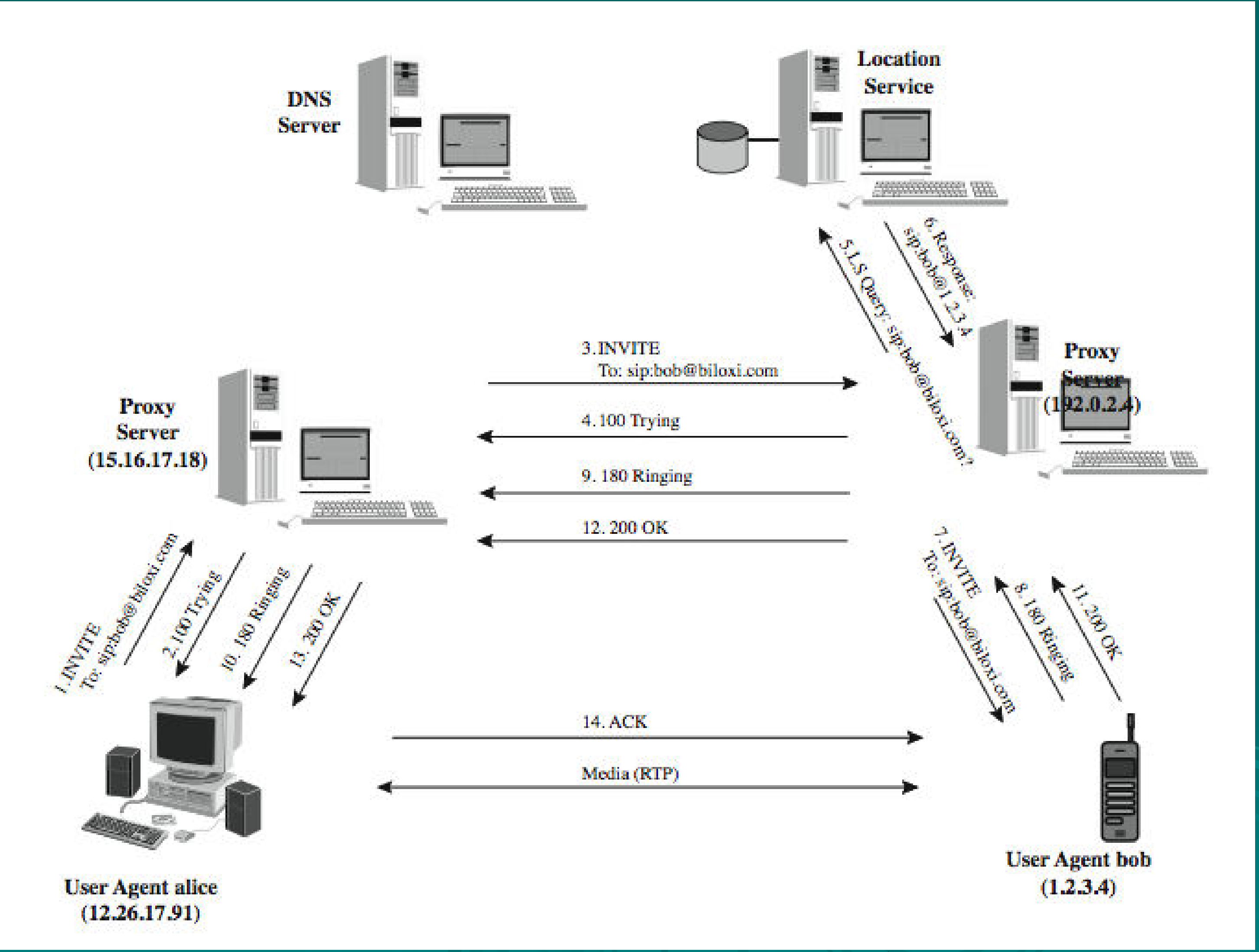


DNS Server
Location Service/ PINT Server
3. SUBSCRIBE To: sip:bob@biloxi.com
4. 200 OK
7. NOTIFY <Not Signed In>
12. 200 OK
2. SUBSCRIBE To: sip:bob@biloxi.com
5. 200 OK
8. NOTIFY <Not Signed In>
11. 200 OK
Proxy Server
Proxy Server
1. SUBSCRIBE To: sip:bob@biloxi.com
6. 200 OK
9. NOTIFY <Not Signed In>
10. 200 OK
User Agent alice
User Agent bob (not signed in)

SIP Example


DNS Server
Location Service
Proxy Server
Proxy Server
2. Update Database B:bob@1.2.3.4
3. OK
5. NOTIFY <Signed In>
10. 200 OK
6. NOTIFY <Signed In>
9. 200 OK
7. NOTIFY <Signed In>
8. 200 OK
1. REGISTER Contact: bob@1.2.3.4
4. 200 OK
User Agent alice
User Agent bob (1.2.3.4)

SIP Example



DNS Server
Location Service
6. Response: sip:bob@1.2.3.4
5. LS Query: sip:bob@biloxi.com?
Proxy Server (192.0.2.4)
3. INVITE To: sip:bob@biloxi.com
4. 100 Trying
9. 180 Ringing
12. 200 OK
Proxy Server (15.16.17.18)
1. INVITE To: sip:bob@biloxi.com
2. 100 Trying
10. 180 Ringing
13. 200 OK
7. INVITE To: sip:bob@biloxi.com
8. 180 Ringing
11. 200 OK
14. ACK
Media (RTP)
User Agent alice (12.26.17.91)
User Agent bob (1.2.3.4)

SIP Messages

- SIP a text based protocol, cf. HTTP
- have request messages
  - first line a method name and request-URI
- have response messages
  - first line a response code

SIP Requests

- defined by RFC 3261
- REGISTER
- INVITE
- ACK
- CANCEL
- BYE
- OPTIONS

SIP Request Example

INVITE sip:bob@biloxi.com SIP/2.0

Via: SIP/2.0/UDP 12.26.17.91:5060

Max-Forwards: 70

To: Bob <sip:bob@biloxi.com>

From: Alice <sip:alice@atlanta.com>;tag=1928301774

Call-ID: a84b4c76e66710@12.26.17.91

CSeq: 314159 INVITE

Contact: <sip:alice@atlanta.com>

Content-Type: application/sdp

Content-Length: 142

SIP Response

- Provisional (1xx)
- Success (2xx)
- Redirection (3xx)
- Client Error (4xx)
- Server Error (5xx)
- Global Failure (6xx)

SIP Response Example

SIP/2.0 200 OK

Via: SIP/2.0/UDP server10.biloxi.com

Via: SIP/2.0/UDP bigbox3.site3.atlanta.com

Via: SIP/2.0/UDP 12.26.17.91:5060

To: Bob <sip:bob@biloxi.com>;tag=a6c85cf

From: Alice <sip:alice@atlanta.com>;tag=1928301774

Call-ID: a84b4c76e66710@12.26.17.91

CSeq: 314159 INVITE

Contact: <sip:bob@biloxi.com>

Content-Type: application/sdp

Content-Length: 131

Session Description Protocol (SDP)

- describes content of sessions
- includes information on:
  - media streams
  - addresses
  - ports
  - payload types
  - start and stop times
  - originator

Real-Time Transport Protocol (RTP)

- TCP has disadvantages for real-time use
  - is point-to-point, not suitable for multicast
  - includes retransmission mechanisms
  - has no timing mechanisms
- UDP can address some needs but not all
- have Real-Time Transport Protocol (RTP)
  - defined in RFC 1889
  - best suited to soft real-time applications
  - data transfer (RTP) & control (RTCP) protocols

RTP Protocol Architecture

- have close coupling between RTP and application-layer functionality
  - view RTP as framework used by applications
- imposes structure and defines common functions
- key concepts:
  - application-level framing
  - integrated layer processing

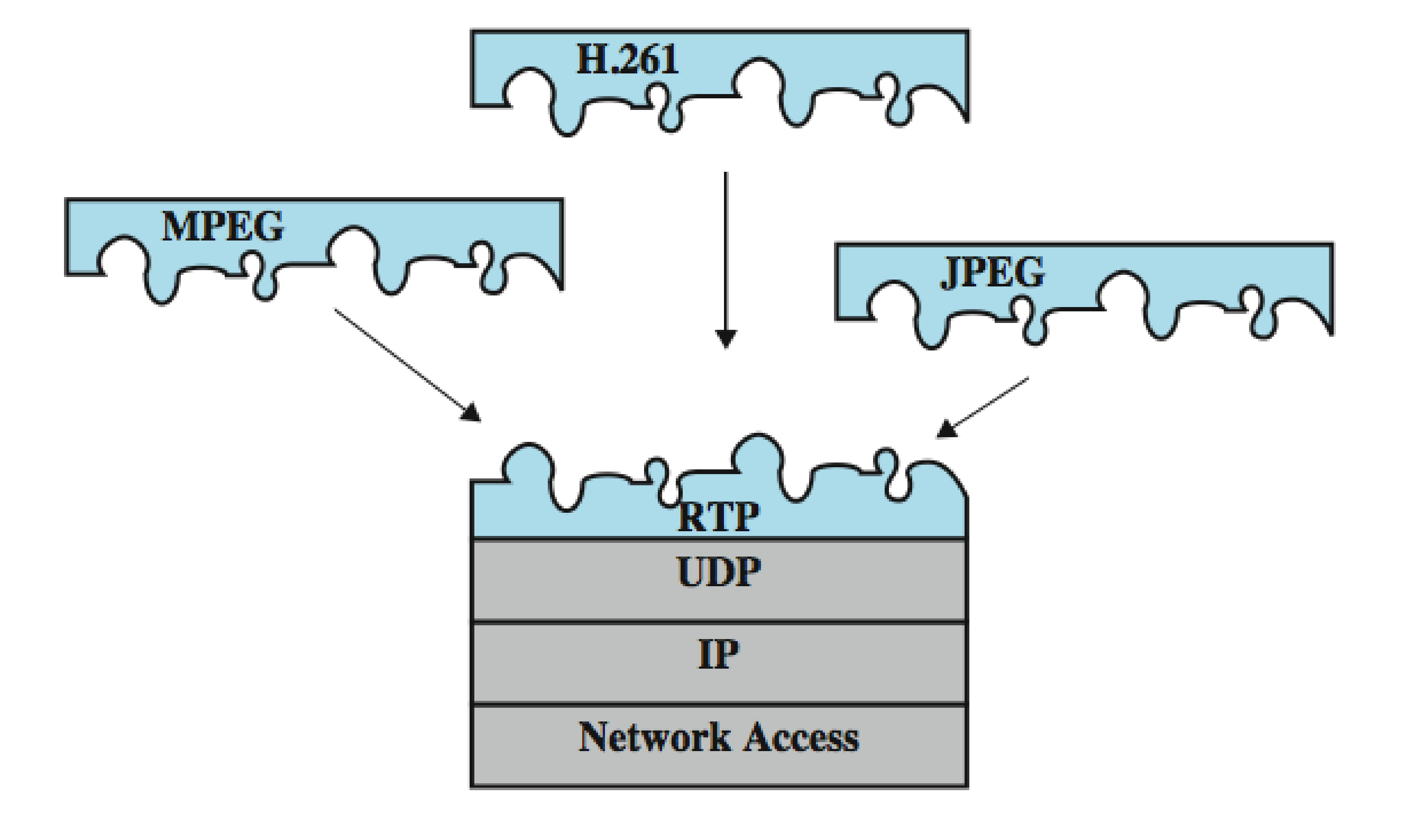
Application-Level Framing

- TCP transparently performs data recovery
- have scenarios where more appropriately done by application layer
  - when less than perfect delivery acceptable
  - when application can better provide data
- have application-level data units (ADUs)
  - preserved by lower layer processing
  - form unit of error recovery
  - if lose part of ADU discard and retransmit entire ADU

Integrated Layer Processing

- layered protocols have sequential processing of functions in each layer
  - limits parallel or re-ordered functions
- instead integrated layer processing allows tight coupling between adjacent layers for greater efficiency
- concept that strict layering is inefficient is not new, cf. RPC implementation

Integrated Layer Processing



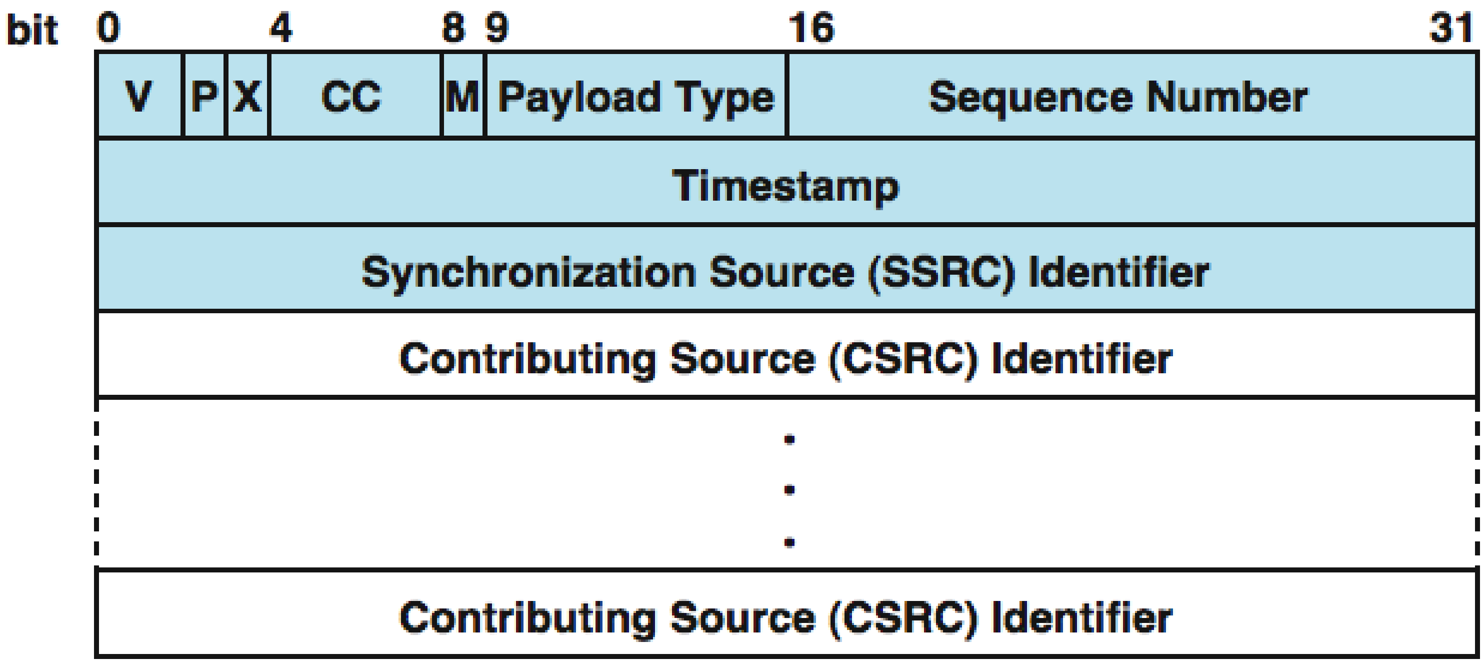
RTP Data Transfer Protocol

- supports transfer of real-time data
- amongst participants in a session
- define session by
  - RTP port (UDP dest port)
  - RTCP port (dest port for RTCP transfers)
  - participant IP addresses (multicast or unicast)
- strength is multicast transmission
  - includes identity of source, timestamp, payload format

RTP Relays

- relay on intermediary system
  - acts as both destination and source
  - to relay data between systems
- mixer
  - combines streams from multiple sources
  - forwards new stream to one or more dests
  - may change data format if needed
- translator
  - simpler, sends 1+ RTP packets for each 1 in

RTP Data Transfer Header



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

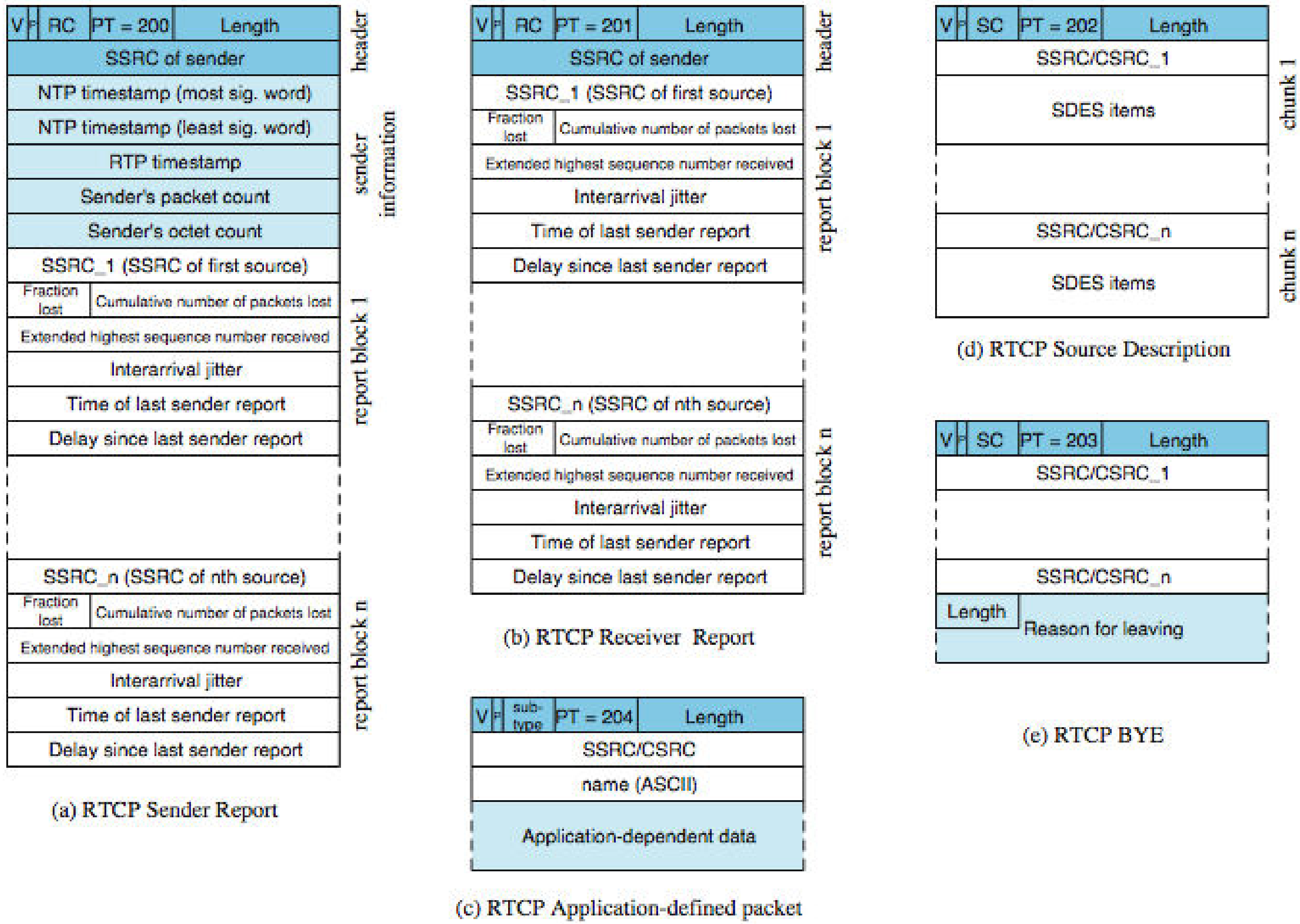
RTP Control Protocol (RTCP)

- separate control protocol
  - same transport (eg. UDP) but different port
  - packets sent periodically to all members
- RTCP functions:
  - Quality of Service (QoS), congestion control
  - identification
  - session size estimation and scaling
  - session control

RTCP Packet Types

- have multiple RTCP packets in datagram
- Sender Report (SR)
- Receiver Report (RR)
- Source Description (SDES)
- Goodbye (BYE)
- Application Specific

RCTP Packets



Summary

- audio and video compression
- real-time traffic
- session initiation protocol (SIP)
- real-time transport protocol (RTP)