Networking Applications

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Multimedia Applications

Outline

- Audio and Video Services
- Digitizing Audio and Video
- Audio and Video Compression
- Streaming Stored Audio/Video
- Streaming Live Audio/Video
- Real-Time Interactive Audio/Video
- Voice over IP (VoIP)
    - SIP and H.323

Audio and Video Services

- Streaming stored audio/video
  - Streaming implies a user can listen or watch the file after downloading has started
  - On-demand
- Streaming live audio/video
- Interactive audio/video
  - Teleconferencing
  - Interactive communication with a group

Digitizing Audio

- Analog signal needs to be digitized to produce a digital signal
  - ➢Sampling process → signal sampled at some fixed rate (example 8,000 samples per second). Each sample is an arbitrary real number
  - ➢Quantization → Each of the samples rounded to one of finite number of values
  - ➢Each of the quantization values represented by a fixed number of bits
- Voice sampled at 8000 samples per second with 8 bits per sample (digital signal of 64 Kbps)
- Music sampled at 44100 samples per second with 16 bits per sample (digital signal of 705.6 Kbps for mono and 1.41 Mbps for stereo)

Digitizing Video 1/2

- A video is a sequence of frames (if displayed fast enough, get impression of motion)
  - In North America, TV sends 25 frames per second
  - To avoid flickering, a frame needs to be refreshed
  - TV industry repaints each frame twice
    - 50 frames to be sent, or
    - If memory at receiver, 25 frames with each frame repainted from memory
- Frame divided into pixels (picture elements)
  - Black and White, each 8-bit pixel → one of 256 different gray levels
  - Color, pixel is 24-bits, 8 bits for each primary color (RGB)

Digitizing Video 2/2

•Number of bits per second for lowest resolution 1024 x 768 pixels

$$2 \times 25 \times 1024 \times 768 \times 24 = 944 \text{ Mbps}$$

•Need for a technology supporting a very high data rate

•To use lower-rate technologies → video compression

Audio Compression

- Predictive Encoding
  - ➢Differences between samples are encoded instead of encoding all sample values
  - ➢GSM (13 Kbps), G.729 (8 Kbps), and G.723.3 (6.4 or 5.3 Kbps)
- Perceptual Encoding (CD-quality audio)
  - ➢MP3 (MPEG audio layer 3)
  - ➢Uses study of how people perceive sound (we have flaws in our auditory system! → Some sounds mask other sounds)
  - ➢Frequency masking (can not hear speaker with loud sound in background) versus temporal masking (can not hear for a short time even after loud sound has stopped)

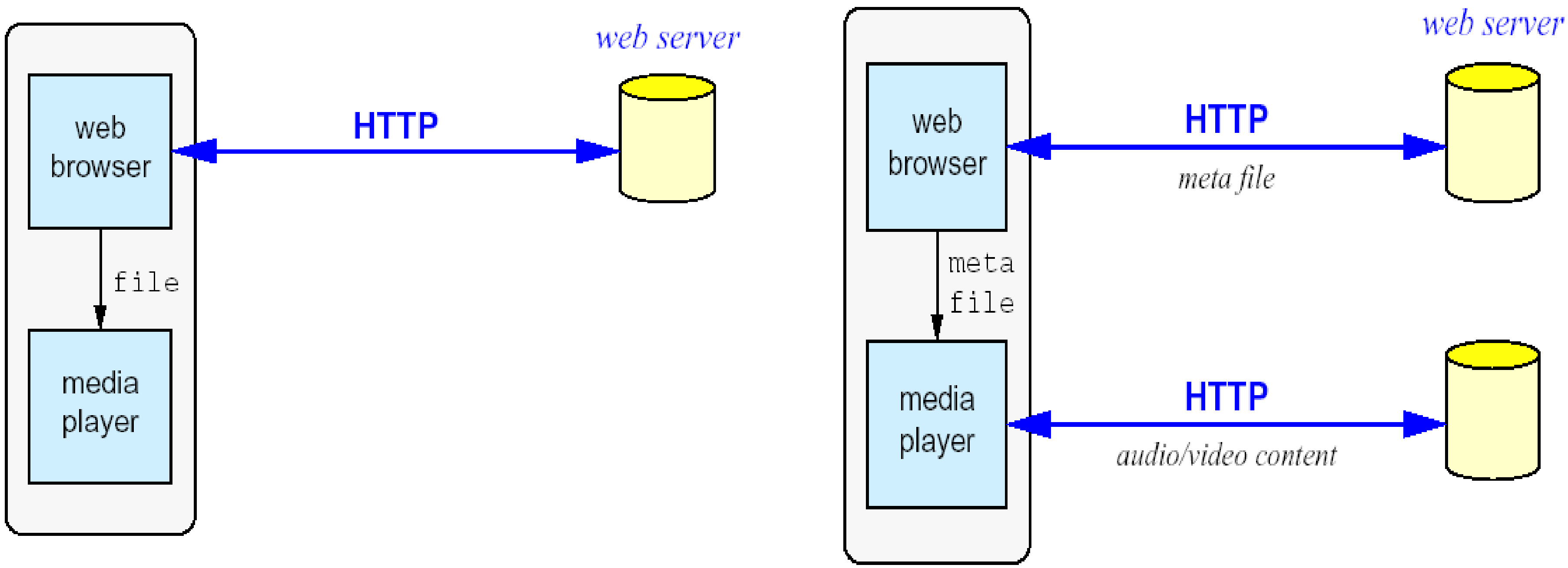
Video Compression

- Use MPEG (Motion Picture Experts Group) as an example
- Spatially compressing each frame (spatial combination of pixels) and temporally compressing a set of frames
- Spatial compression: each frame is simply a picture that can be independently compresses (e.g., divide into blocks, apply a transformation to reveal redundancies and remove them → lossy vs lossless compression)
- Temporal compression: redundant frames are removed
  - Divide frames into I-frames (Intra-coded), P-frames (Predicted, related to preceding I or P frame), and B-frames (Bidirectional)
  - Example frame sequence: I-B-B-P-B-B-I
  - MPEG1 (CD-ROM 1.5 Mbps), MPEG2 (high quality DVD 3-6 Mbps)

Streaming Stored Audio/Video 1/6

- Using a Web Server
  - ➢Download file as a simple text file using Get and Response mechanisms
  - ➢Browser uses a helper application (media player) to play file
  - ➢Does not involve streaming (file downloaded completely before played)
- Using a Web Server with a Metafile
  - ➢Media player directly connected to Web server
  - ➢Server stores two files: actual audio/video file and a metafile holding information about audio/video file
  - ➢Media player uses URL in metafile to access audio/video file, using HTTP services (over TCP → problem with retransmissions)

Streaming Stored Audio/Video 2/6

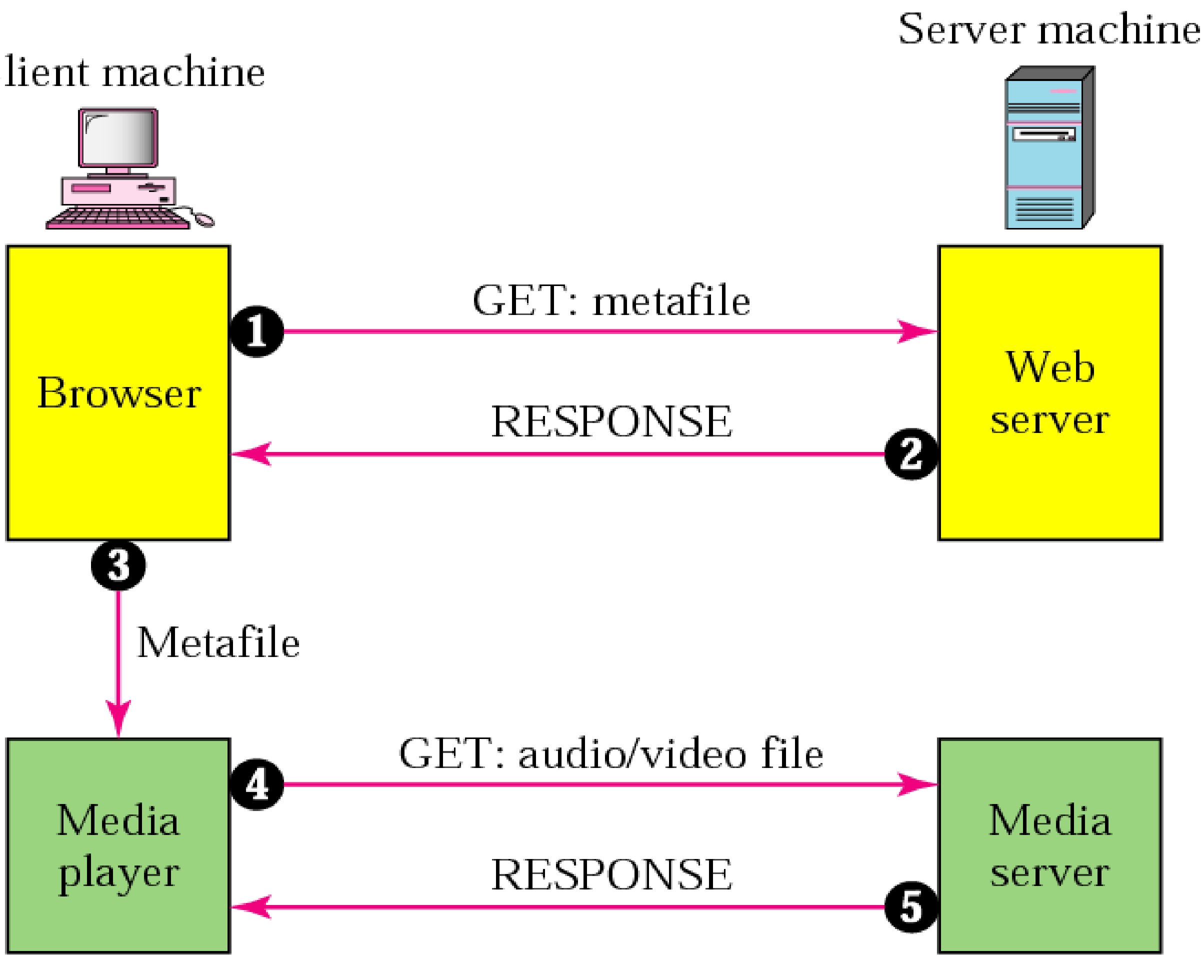


Streaming Stored Audio/Video 3/6

- Using a Media Server
  - Get metafile from Web server using HTTP mechanisms
  - Metafile passed to media player
  - Media player uses URL in metafile to access media server and download file

Streaming Stored Audio/Video 4/6

- Using a Media Server

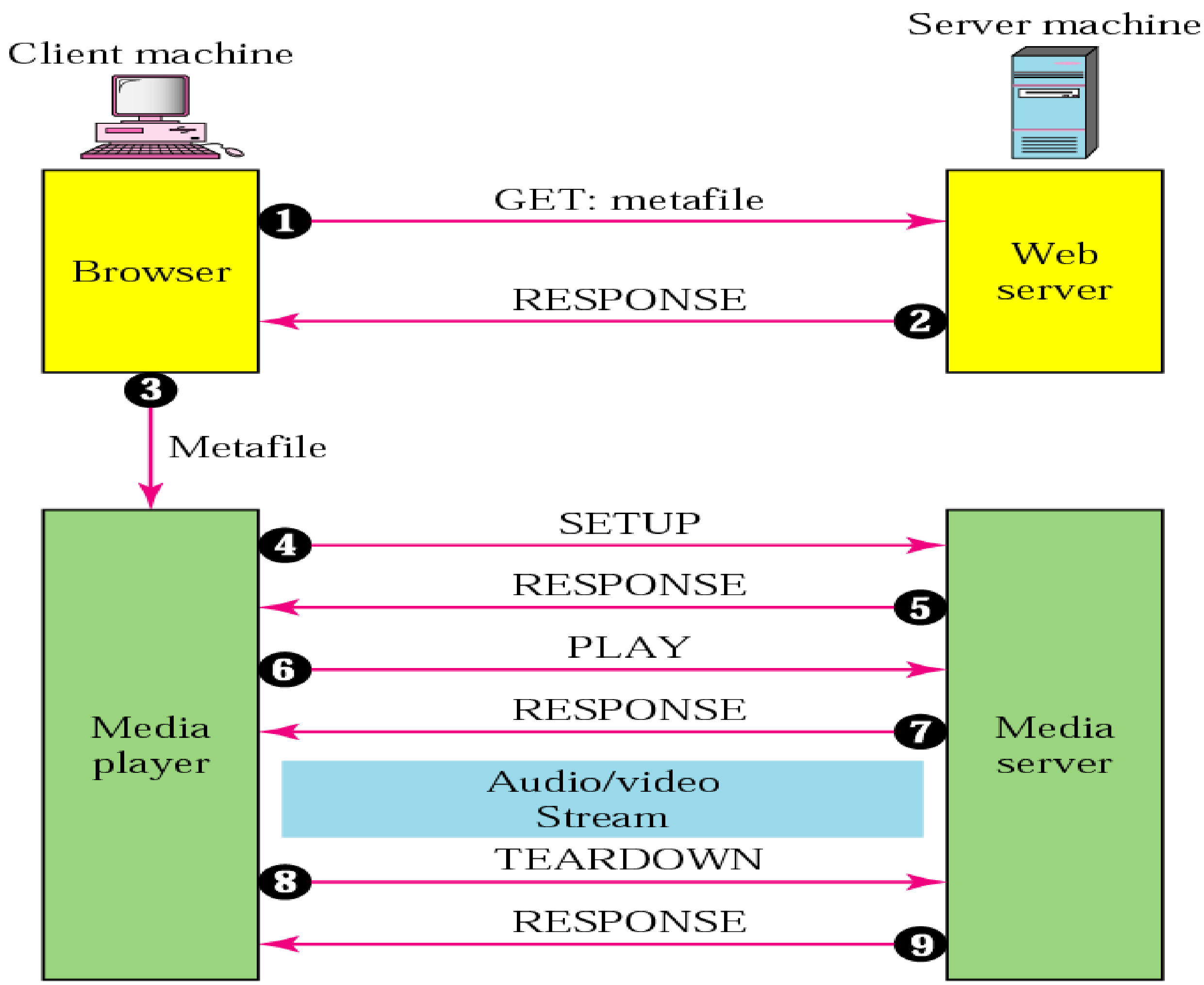


Streaming Stored Audio/Video 5/6

- Using a Media Server and RTSP (Real-Time Streaming Protocol)
  - http://www.cs.columbia.edu/~hgs/rtsp/
  - RTSP is a control protocol to add more functionalities to the streaming process (provide user interactivity)
  - After getting metafile, media player sends a SETUP message to create a connection with the media server (Media server responds)
  - Media player sends a PLAY message to start playing (downloading)
  - Audio/video file downloaded using a protocol that runs over UDP
  - Connection broken using TEARDOWN message
  - Media server responds
  - Media player can use a PAUSE message, and later resume with a PLAY message

Streaming Stored Audio/Video 6/6

- Using a Media Server and RTSP



Streaming Live Audio/Video

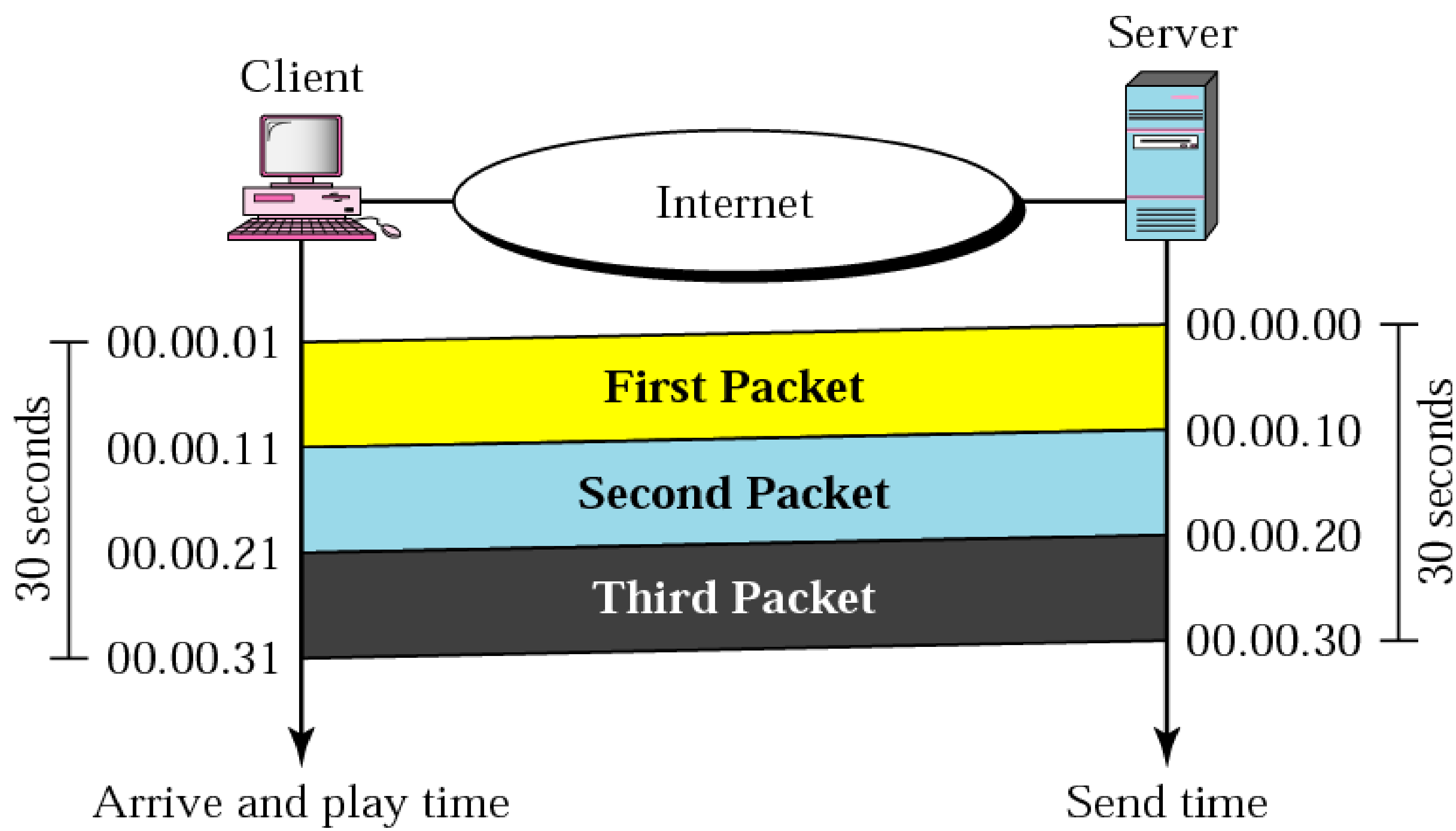
- Similar to broadcasting audio and video by radio and TV stations
- Sensitive to delay
- Does not accept retransmissions
- Stored live audio/video: unicast and on-demand (pull)
- Live audio/video: multicast and live (push)

Real-Time Interactive Audio/Video 1/6

- Internet phone or Voice over IP
- Time Relationship (Example: 3 video packets, each worth 10s of video information)
  - Case 1: equal delay for each packet to reach destination (e.g., 1 s)
  - Case 2: different delays? (1s followed by 5s followed by 7s)
  - Might cause gaps between packet playback
  - Causes jitter due to variable delay between packets
- To deal with jitter use Timestamps (shows time it was produced relative to first (or previous) packet
  - Receiver can add this time to the time at which it starts the playback (receiver knows when packet is to be played)
  - This way we *separate arrival time from playback time*

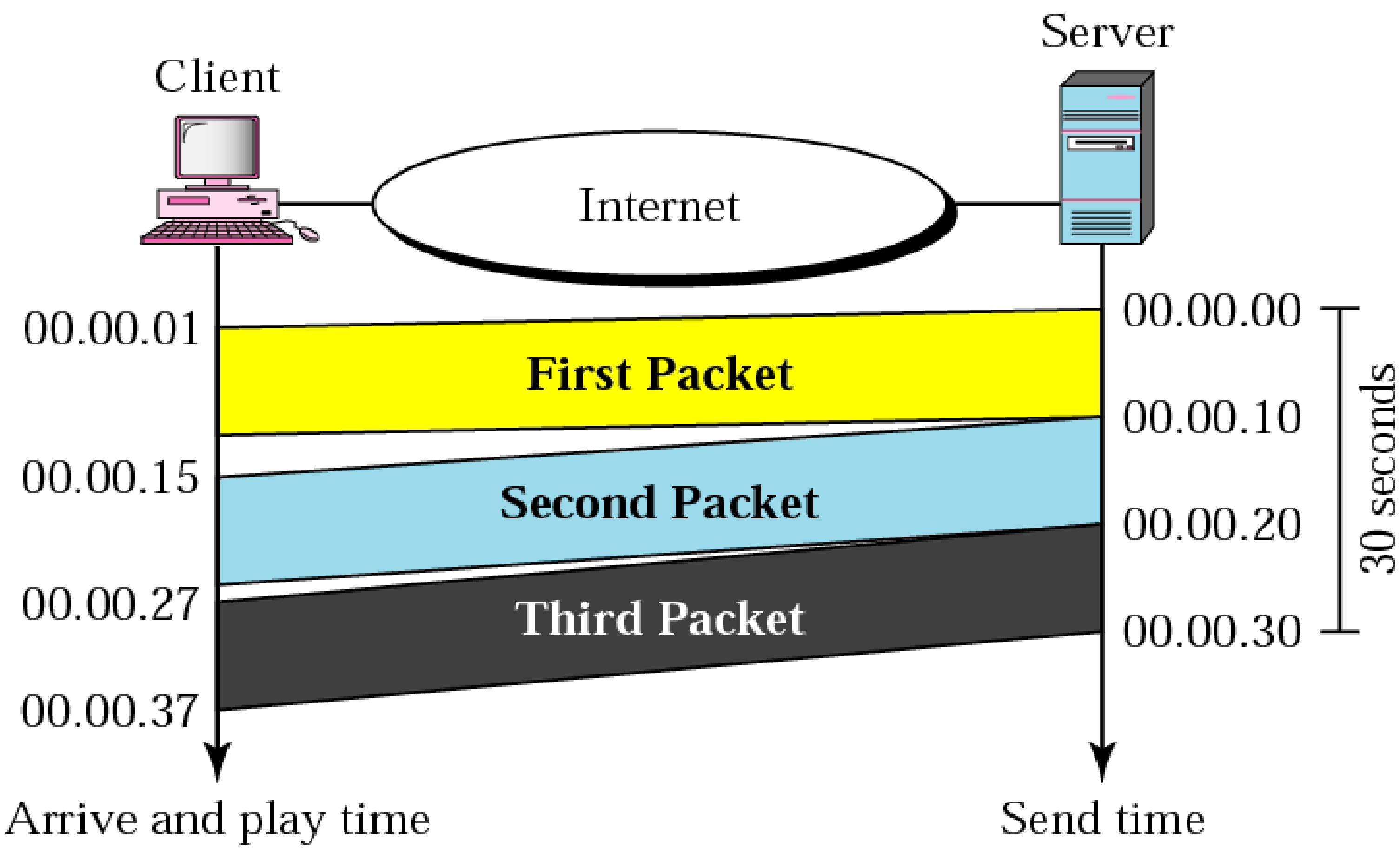
Real-Time Interactive Audio/Video 2/6

Case 1: equal delay for each packet to reach destination (e.g., 1 s)



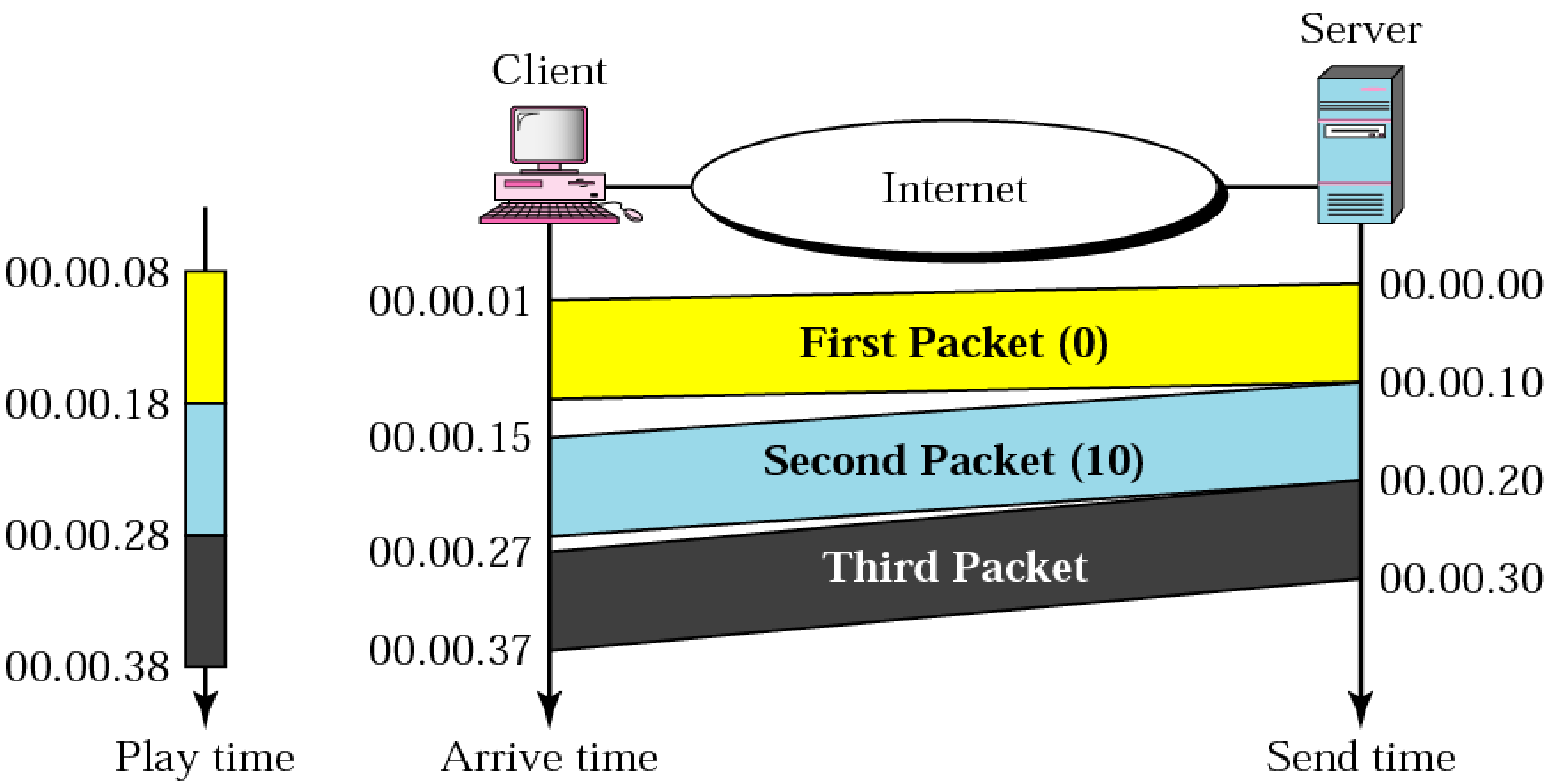
Real-Time Interactive Audio/Video 3/6

Case 2: different delays? (1s followed by 5s followed by 7s)



Real-Time Interactive Audio/Video 4/6

- Use timestamps: separate arrival time from play time
- Assume playback started at 00.00.08s



Real-Time Interactive Audio/Video 5/6

- To separate arrival time from playback time, need a buffer to store data until it is played back (playback buffer)
    - ➢Receiver delays playing data until a threshold is reached
    - ➢Data stored in buffer at a variable rate, but extracted and played back at a fixed rate
    - ➢Data arriving after playback deadline is ignored
- Need for multicasting support
- *Ordering*: need a sequence number for ordering purposes
- *Transcoding*: change the format of high-bandwidth video to a low-quality video (e.g., from MPEG to H263)
- *Mixing*: combine streams from multiple sources into one stream

Real-Time Interactive Audio/Video 6/6

- Support from transport layer protocol
  - Previous procedures can be implemented in application layer
  - Common in real-time applications → implement in transport layer
  - TCP not suitable because of error control mechanisms
  - UDP suitable for interactive multimedia traffic (multimedia, no retransmission, however does not support timestamping, sequencing, or mixing
  - Use Real-Time Transport Protocol (RTP)

Real-Time Transport Protocol 1/2

- See http://www.cs.columbia.edu/~hgs/rtp/
- Handle real-time traffic on the Internet
- Does not have a delivery mechanism → must be used with UDP and application program
- Offers timestamping, sequencing, and mixing services
- RTP packet encapsulated in a UDP datagram
  - No well-known port assigned to RTP
  - Uses an ephemeral even port, with next port used by companion protocol (RTCP)
- RTCP offers flow control and data quality, and allows recipient to send feedback to source

Real-Time Transport Protocol 2/2

- RTCP Messages
  - Sender Report
    - Sent periodically by active senders to report transmission and reception statistics (includes an absolute timestamp for synchronization)
  - Receiver Report
    - For passive participants that do not send RTP packets to inform senders and other receivers about the quality of service
  - Source Description Message
    - Periodically sent by sender (name, email address, Tel. No)
  - Bye Message → Shutdown a stream
  - Application-Specific Message

Voice over IP

Extra References

•Voice over IP and IP Telephony: References

http://www.cse.wustl.edu/~jain/refs/hot_refs.htm

Voice Over IP Reference Page from *protocols.com*

http://www.protocols.com/pbook/VoIP.htm

•IEEE Spectrum Feature Article in March 2005

http://ieeexplore.ieee.org/xpl/articleDetails.jsp?arnumber=1402719