CS 218 F 2003

Nov 3 lecture:

- Streaming video/audio
- Adaptive encoding (eg, layered encoding)
- TCP friendliness

References:

- J. Padhye, V.Firoiu, D. Towsley, J. Kurose" Modeling TCP Throughput: a Simple Model and its Empirical Validation" Sigcomm 98
- S. Floyd, J. Padhye, J.Widmer "Equation Based Congestion Control for Unicast Applications", Sigcomm 2000
- Rejaie et al " RAP: end to end control for real time streams" Infocom 99
- Tang et al " RCS: Rate Control Scheme..for high bit error rates"

Streaming

- Important and growing application due to reduction of storage costs, increase in high speed net access from homes, enhancements to caching and introduction of QoS in IP networks

- Audio/Video file is segmented and sent over either TCP or UDP, public segmentation protocol: **Real-Time Protocol (RTP)**

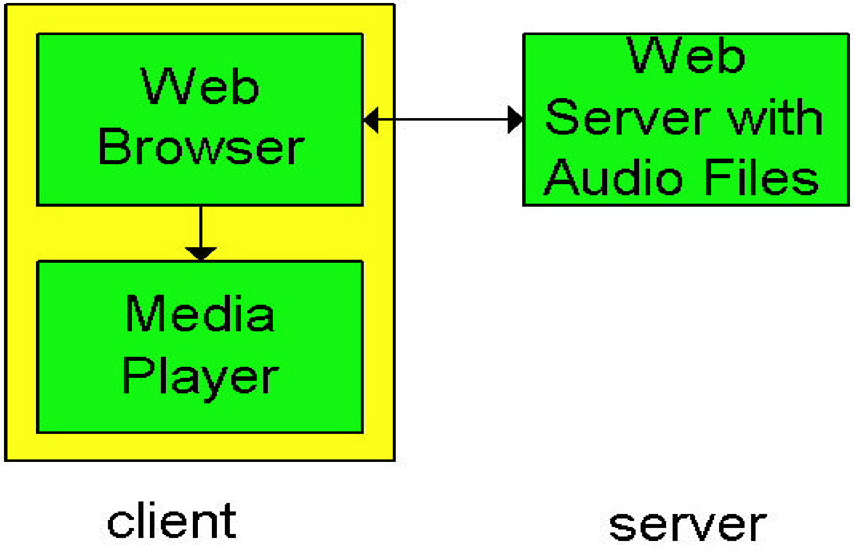
Streaming

- User interactive control is provided, e.g. the public protocol **Real Time Streaming Protocol (RTSP)**
- **Helper Application:** displays content, which is typically requested via a Web browser; e.g. RealPlayer; typical functions:
  - Decompression
  - Jitter removal
  - Error correction: use redundant packets to be used for reconstruction of original stream
  - GUI for user control

Streaming From Web Servers

- Audio: in files sent as HTTP objects
- Video (interleaved audio and images in one file, or two separate files and client synchronizes the display) sent as HTTP object(s)

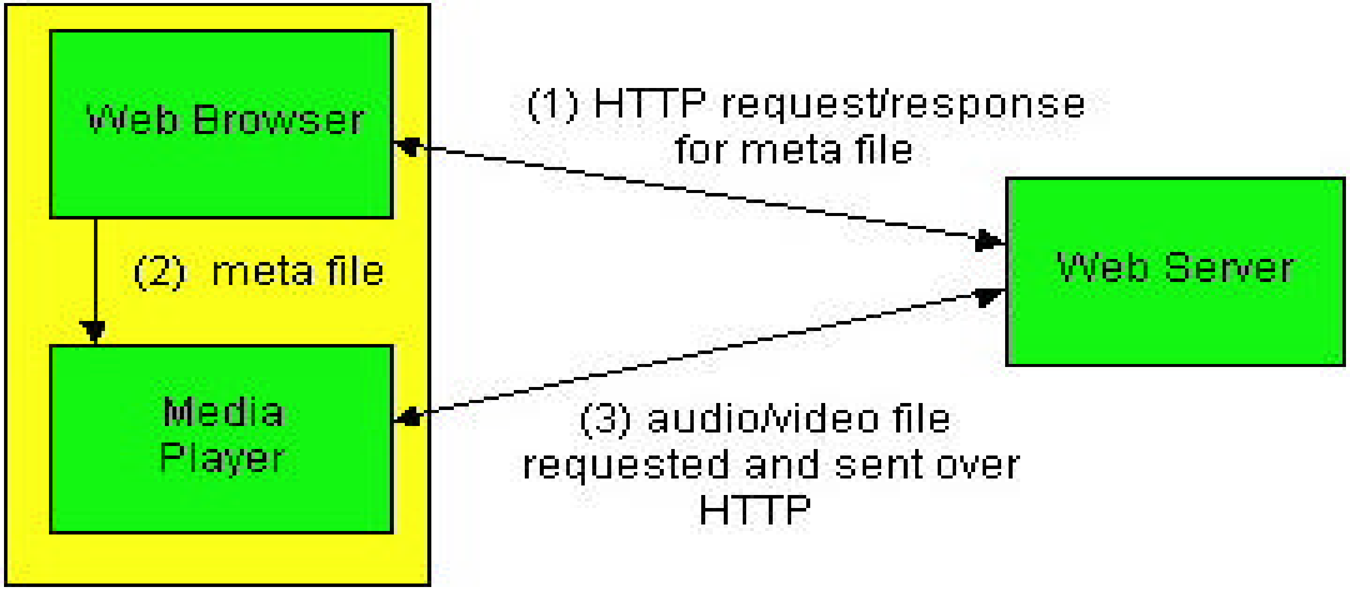
- A simple architecture is to have the Browser requests the object(s) and after their reception pass them to the player for display
  - No pipelining



Streaming From Web Server (more)

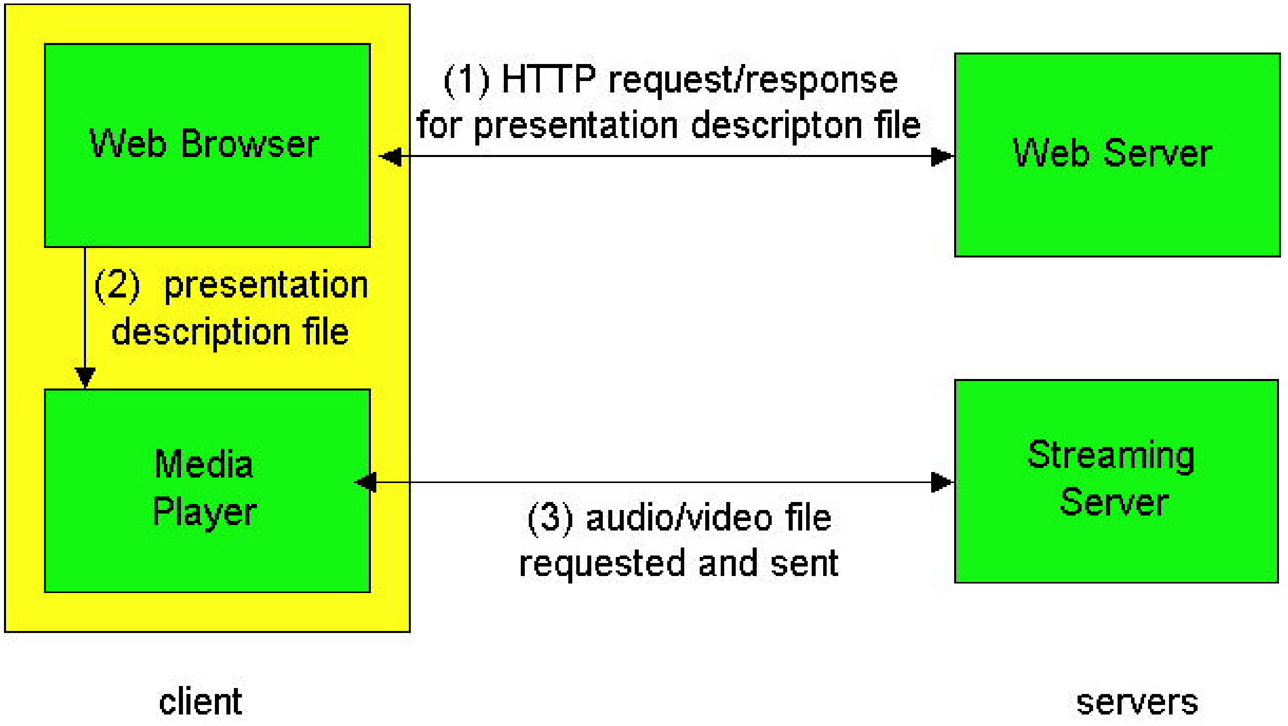
- Alternative: set up connection between server and player, then download
- Web browser requests and receives a **Meta File** (a file describing the object) instead of receiving the file itself;
- Browser launches the appropriate Player and passes it the Meta File;
- Player sets up a TCP connection with Web Server and downloads the file

Meta file requests



Using a Streaming Server

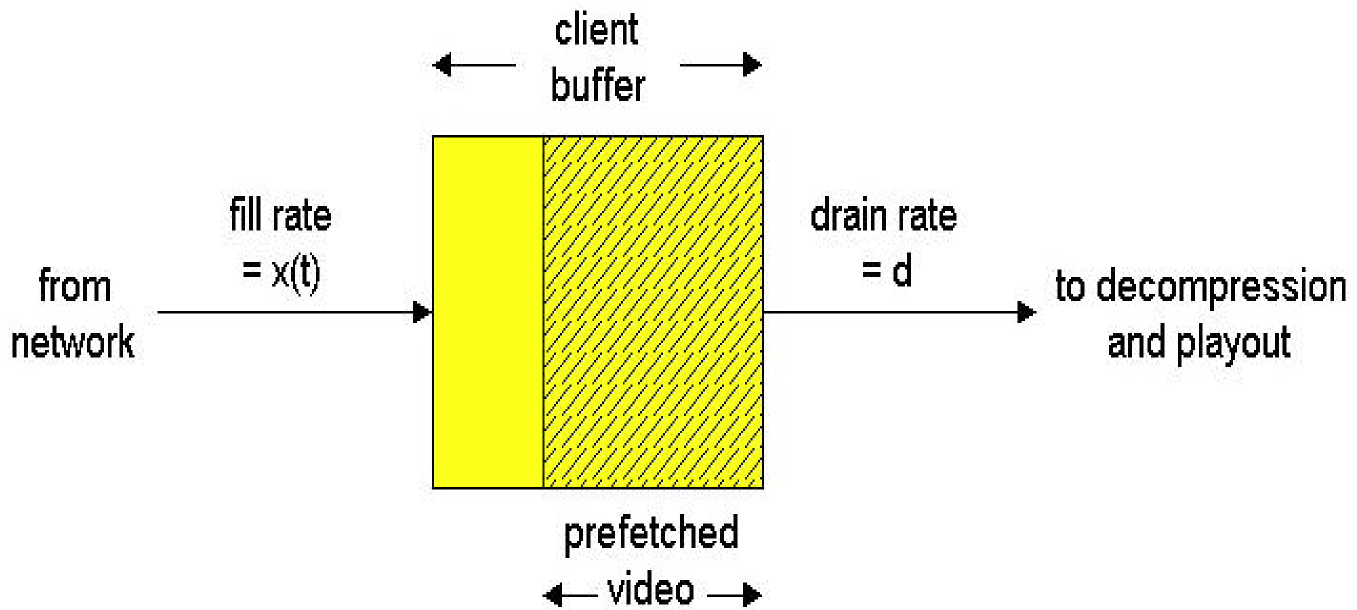
- This gets us around HTTP, allows a choice of UDP vs. TCP and the application layer protocol can be better tailored to Streaming; many enhancements options are possible (see next slide)



Options When Using a Streaming Server

- Use UDP, and Server sends at a rate (Compression and Transmission) appropriate for client; to reduce jitter, Player buffers initially for 2-5 seconds, then starts display

- Use TCP, and sender sends at maximum possible rate under TCP; retransmit when error is encountered; Player uses a much large buffer to smooth delivery rate of TCP

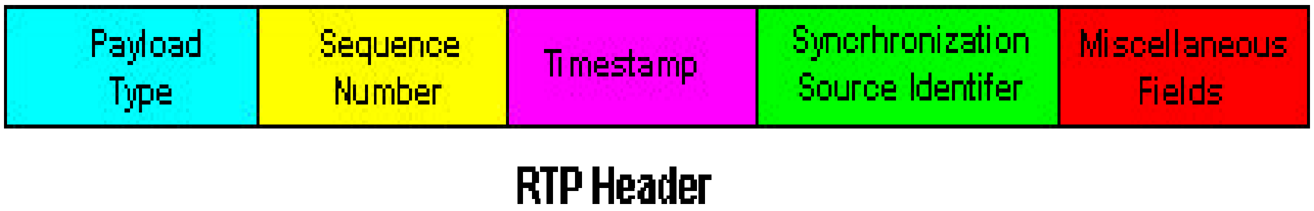


Real Time Streaming Protocol (RTSP)

- For user to control display: rewind, fast forward, pause, resume, etc...
- Out-of-band protocol (uses two connections, one for control messages (Port 554) and for media stream)
- RFC 2326 permits use of either TCP or UDP for the control messages connection, sometimes called the RTSP Channel
- As before, meta file is communicated to web browser which then launches the Player; Player sets up an RTSP connection for control messages in addition to the connection for the streaming media

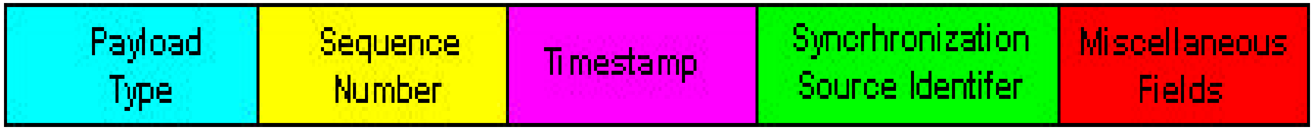
Real-Time Protocol (RTP)

- Provides standard packet format for real-time application
- Typically runs over UDP
- Specifies header fields below
- **Payload Type**: 7 bits, providing 128 possible different types of encoding; eg PCM, MPEG2 video, etc.
- **Sequence Number**: 16 bits; used to detect packet loss



Real-Time Protocol (RTP)

- **Timestamp**: 32 bytes; gives the sampling instant of the first audio/video byte in the packet; used to remove jitter introduced by the network
- **Synchronization Source identifier (SSRC)**: 32 bits; an id for the source of a stream; assigned randomly by the source



RTP Header

RTP Control Protocol (RTCP)

- Protocol specifies report of packets exchanged between sources and destinations of multimedia information
- Three reports are defined: Receiver reception, Sender, and Source description
- Reports contain statistics such as the number of packets sent, number of packets lost, inter-arrival jitter
- Used to modify sender transmission rates and for diagnostics purposes