B211 Internet Computing

Multimedia Delivery over the Internet

Lecture Objectives

- Understand the special requirements for transporting multimedia content.
- Understand the basic methods used to deliver multimedia content over the Internet.

Lecture Outline

- Multimedia Content
- Delivering Multimedia Content over the Internet and the Web.
- Multimedia Transport Protocols.
  - RTSP and RTP
  - RSVP
  - MMS

Multimedia Content

- Images, audio and video
  - As oppose to plain text.
- Multimedia content can be served through HTTP by using conventional web servers, and having special media plugins or helper applications on the client-side.

Steps in Preparing Multimedia Content

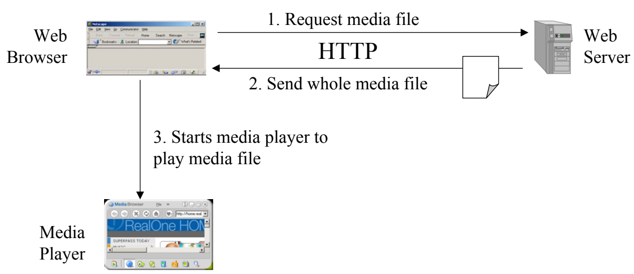
- Capture
  - get original data (eg from microphone or video camera) into digital form
- Encode
  - convert to format that clients can play
- Store
  - put on server
- Deliver
  - from server to client
- Decode
  - play on the client side

Example Content Encodings

- Video
  - RealVideo
  - AVI (Audio Video Interleave)
  - Quicktime movies
  - MPEG (Moving Pictures Expert Group)
- Audio
  - RealAudio
  - AU
  - WAV
  - MP3 (MPEG Audio Layer 3)

Download and Play

- The conventional way using a standard HTTP to deliver multimedia content:



Streaming Content

- Hierarchy of multimedia content:

Pixel (picture element)
Frame (grid of pixels) Sound samples
Stream (Sequence of frames and sounds over time)
Session (Synchronised set of streams)
Presentation (Set of multimedia sessions)

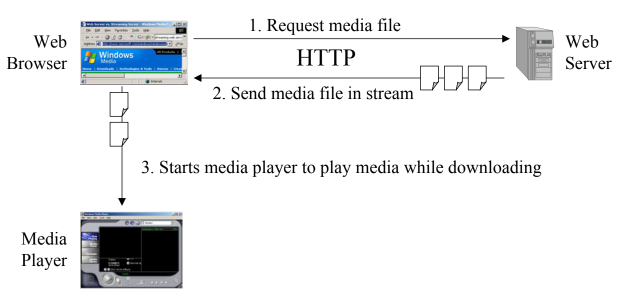
Streaming

- In *streaming,* unlike simple download-and-play, content is displayed by the client media player while the download is happening.
- Streaming clients:
  - download an initial portions of the audio/video and start displaying without waiting for everything.
  - have a continuous buffer of upcoming sounds/images, so that the display won't stop if there is a slight delay in traffic.

Streaming Applications

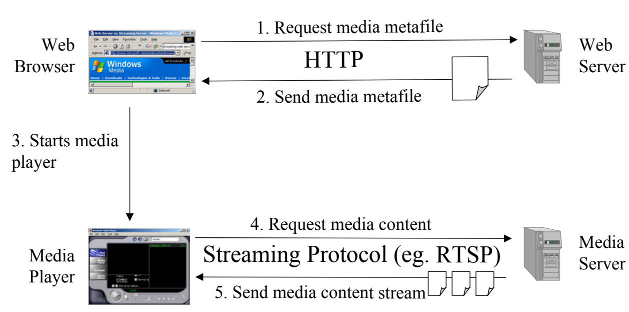
- Some examples of using streaming:
  - Audio or video on demand through web sites
  - Internet telephony
  - Audio or video conferencing
  - Radio or TV broadcasts
  - Online multi-player games
  - Virtual reality simulations

Streaming Through a Web Server



- This can only be done if the media player and the media file format supports progressive HTTP downloads and playback like this. An example is Microsoft's ASF files.

Streaming Through a Media Server



Example Media Meta Files

- Windows ASX (Advanced Streaming Redirector) File

```
<ASX version = "3.0">
  ...
  <Entry>
    <ref href="mms://mmsmediaserver.com.au/show.asf" />
    ...
  </Entry>

</ASX>
```


- Streaming multimedia Integration Language (SMIL) file with RTSP

```
<smil>
  ...

  <body>
    <video src="rtsp://rtspmediserver.com.au/show.mov"/>
  ...
  </body>
</smil>
```


- Embedded Quicktime files in HTML:

```
<html>
  ...

  <body>
    <EMBED src="show.mov"
      qtsrc="rtsp://qtmediaserver.com.au/show2.mov">

  ...
  </body>
</html>
```


Live vs On-Demand Streaming

- In on-demand streaming, content is prepared as described on page 5.
  - Users requests prepared content, and are delivered what they requested.
- In live streaming, the steps of capturing, encoding, storing and delivering are done in real-time, as events are happening.

Properties of Streaming Applications

- Start-up delay and buffering
  - To counter network latencies and traffic congestion.
  - Prevents start-and-stop presentation by having a buffer.
- Content generation
  - On-demand content can be compressed, and can provide advanced VCR (video cassette recorder) features. Live content cannot.
- Unicast vs Multicast
  - Unicast: streamed to a single recipient.
  - Multicast: streamed to multiple recipients.
- Session establishment
  - Through client-pull (eg user clicking on a hyperlink)
  - Through client-to-client signaling (eg. in conferencing)
  - Through server broadcast

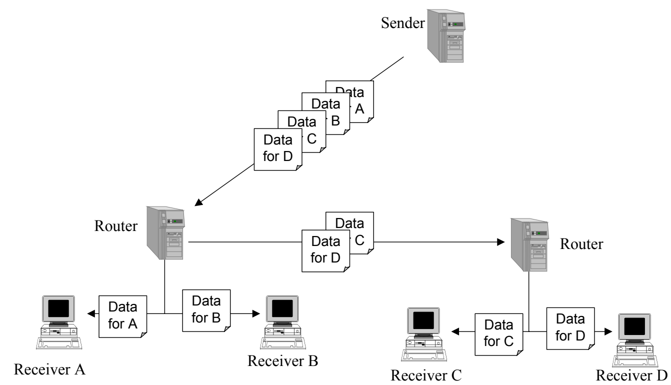
Properties of Multimedia Delivery

- Content cannot be delayed.
  - Unlike normal files, where delays in delivery are not as critical.
- Content can suffer slight loss.
  - It just means the image or sound quality is a bit degraded.
  - Unlike normal files, where the whole file has to arrive intact.
- Typically very large size, so required large bandwidth.

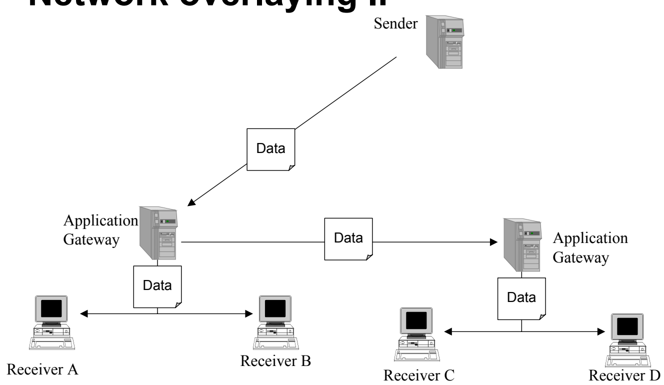
IP Networks for Multimedia Delivery

- IP is not the ideal protocol for delivering multimedia content
  - It was designed in days when text was the prevalent format.
- It suffers from:
  - No guarantee on Quality-of-Service (QoS): streams can't be assured they will have certain priorities for resources.
  - Congestion control problems
  - being inherently a unicast network, not natural for multicasting.
- In spite of this, IP is the default protocol used in most multimedia delivery.
  - the cost of replacing IP with another network is too high in cost to consider.

Multicasting using an IP Network



More Efficient Multicasting Network overlaying IP



IPv6

- One of the design criteria for the new IP version 6 was to support transport of multimedia content.
- Unlike IPv4, IPv6 packets can indicate:
  - high priorities
  - the fact that multiple packets belong in a single traffic "flow"
- Routers can use the information to route the packets appropriately.
  - Eg. process the higher priority packets first.
- This helps with QoS and congestion problems.

Multimedia Delivery Using HTTP

- HTTP (over IP) can be used to deliver multimedia content
  - Refer to diagrams on page 7 and 11 again.
- HTTP is also not the ideal for this purpose:
  - Requires large portions (or whole) of content to be downloaded before playing.
  - Content has to be copied from browser to media player.
  - Must interleave audio and video streams into one HTTP response message.
  - Guarantees reliable transmission when reliability (at the cost of timeliness) is not desired.
  - Hard to incorporate VCR functions (like pause, rewind, etc) with only single HTTP request/responses exchanges.

Protocols for Multimedia Delivery

- We can replace the use of HTTP with a few specialised protocols for efficient multimedia delivery.
  - This only involves having specialised clients (the media players) and servers (the streaming servers).
    - » Compare this with replacing IP as the default protocol at a lower level, which will involve replacing router software - too difficult!
- Example multimedia protocols at different levels:

Media Content	MPEG, WMV, MOV, RAM, ASF
Presentation Description	SMIL, ASX, HTML
Session Description	SDP
Session Establishment	RTSP, SIP/SAP
Data Transport	RTP/RTCP, RSVP, MMS

• Refer to Unit reader 6 part 12.3 for acronyms

Key Streaming Technologies

- The current streaming market is dominated by 3 technologies:
  - Microsoft:
    - » Server: Microsoft Windows Media Services (part of .NET environment)
    - » Client: Windows Media Player
    - » Transport: MMS (Microsoft Media Streaming), HTTP
  - Real Networks
    - » Server: Helix Universal Server
    - » Client: RealOne, RealJukeBox
    - » Transport: RTSP over RTP

- Quicktime
  - » Servers: Quicktime Streaming Server and Darwin Sreaming Server
  - » Client: Quicktime player
  - » Transport: RTSP over RTP

- Look up the references at the end of this lecture for the content formats
  - each streaming servers can stream, and
  - each client media players can play.

Real Time Streaming Protocol (RTSP)

- Allows clients to access multimedia content by sending requests to URLs
  - Syntax of RSTP request response messages are based on HTTP's format.
  - Works much like HTTP, except with different available request methods, response status codes, and headers.
  - Selects an underlying transport protocol (eg. RTP) to deliver the content.

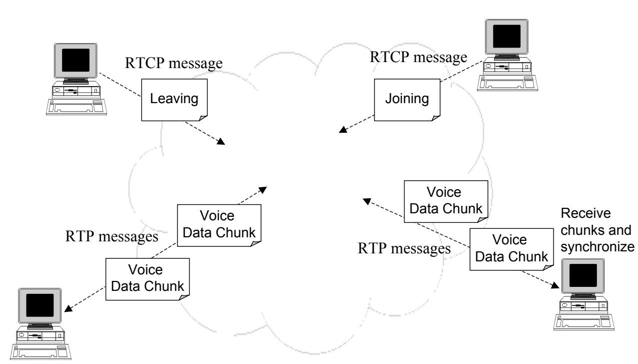
Real Time Transport Protocol (RTP)

- Designed to support real-time traffic.
  - Eg. audio and video conferencing, live event broadcasting.
- Can also be used for any on-demand delivery.

Real Time Control Protocol (RTCP)

- RTP also defined RTCP.
- RTCP involves the control information sent by the various senders to coordinate the multicasting session.
  - All participant periodically send RTCP messages to the others to convey information (eg. changes in congestions, a new receiver in the multicast, a receiver leaving the session, etc).

Example Audio Conference



Mixed Audio and Video in RTP

- In sessions involving both audio and video, the two media are transmitted as separate RTP sessions.
  - RTCP packets are transmitted for each medium using different ports and different addresses.
  - This is to allow some participants to receive only one medium if they choose

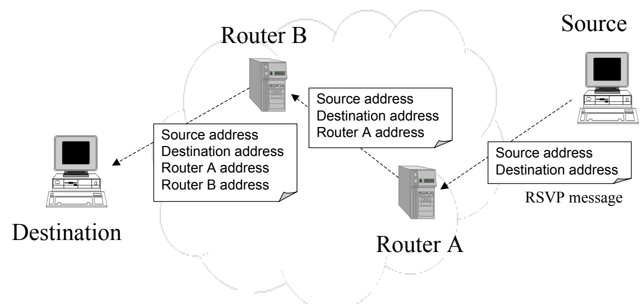
Limitations of RTP

- RTP does not handle:
  - timely delivery - how long it takes to get to the receivers.
  - Guarantees that packets will reach receivers.
- It depends on other protocols to handle them, eg RSVP.

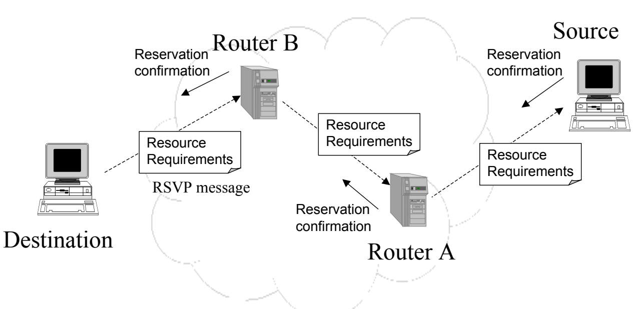
Resource Reservation Protocol (RSVP)

- Defines a procedures for packet recipients to tell sender and routers what it's capacities and limitations are.
  - Reserves resources, eg. bandwidth.
- Works on the assumption that a receiver knows what its requirements are.

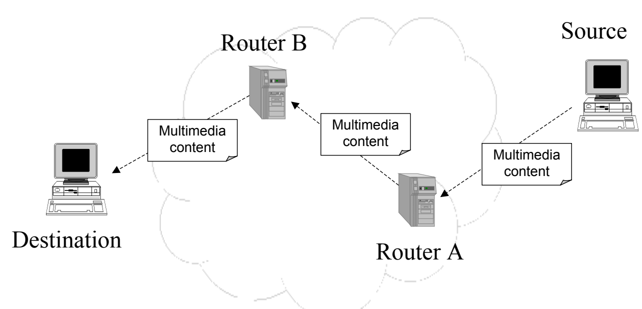
RSVP Path Determination



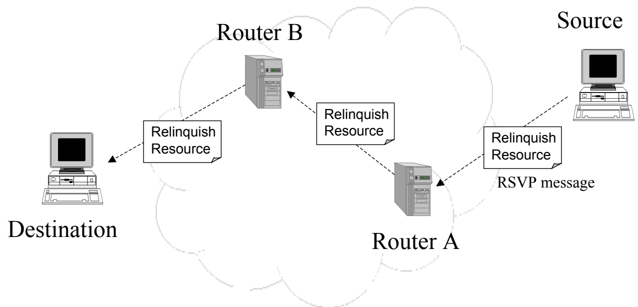
RSVP Resource Reservation



Content Transport through path fixed by RSVP



RSVP Path Teardown



Microsoft Media Server (MMS) Streaming

- Microsoft's proprietary protocol for streaming media.
- Rollover mechanism: Client media player will
  - First try to receive stream in UDP (most cost efficient). If fails then
  - Try using TCP. If fails, then
  - Try using HTTP

Further Reading

- RTSP, RTP and RSVP
  - http://www.rtsp.org/
  - http://www.cs.columbia.edu/~hgs/rtp/
  - http://www.isi.edu/div7/rsvp/rsvp.html
- Streaming Media Technologies
  - http://www.real.com/
  - http://www.apple.com/quicktime/
  - http://www.microsoft.com/windows/windowsmedia/default.asp

Reference

- Unlike other lectures, this one follows your Unit Reader section 6 quite closely. Read that section for further elaborations.