Multimedia over the Internet

François Fluckiger
CERN, Geneva

francois.fluckiger@cern.ch
Who am I?

François Fluckiger

CERN

Geneva, Switzerland

Birth place of the World-Wide Web

Head, Internet Infrastructure

Part-time Professor of Computer Science at the

University of Geneva
For further reading ...

- Reference book:

  "Understanding Networked Multimedia"


  by F.Fluckiger

Where to find more in the Book?

See pointer at the bottom

See UNM book, p. xyw
Part 1
Requirements of Multimedia over the Internet
• **Key requirements**

• **Bit rate requirements**
  - Audio requirements
  - Video requirements

• **Delay requirements**
  - Jitter
  - Inter-media synchronization

• **On compression ...**
Multimedia characteristics

- **Information types**
  - Text
  - Images (pixel matrix)
  - Graphics (logical objects)
  - Animation (moving graphics)
  - Video (moving images)
  - Audio

- **Continuous media** (time is part of their semantics)
Audio/video network requirements

- **Key requirements**
  - Bit rates
  - Transit delay variation
  - Multicasting capabilities (for distribution)

- **Other requirements**
  - Transit delay
  - Error rate
• Key requirements

• **Bit rate requirements**
  • Audio requirements
  • Video requirements

• **Delay requirements**
  • Jitter
  • Inter-media synchronization

• On compression ...
Types of applications

- **Traditional real-time applications** e.g. PABXs
  
  *constant bit rate* (CBR)

- **Traditional bulk data applications** e.g. file transfer, email
  
  *available bit rate* (ABR)

- **Modern real-time applications** e.g. compressed audio, video
  
  *variable bit rate* (VBR)
Quality of Service and bit rate

- **CBR** applications

- **ABR** applications

- **VBR** applications

From S. Shenker,
Fundamental Design Issues for the Future Internet, 1995
Principle (or platitude)

The grass is always greener on the other side of the hill ...
• Key requirements

• Bit rate requirements
  • Audio requirements
  • Video requirements

• Delay requirements
  • Jitter
  • Inter-media synchronization

• On compression ...
Nyquist theorem

- To faithfully represent an analog signal
  - if maximum frequency $f$
  - sampling rate at least $2f$

- Application to audio
  - if sampling rate is 8 kHz
  - bandwidth is 3.4 kHz

Sampling at $f$: impossible to reconstruct
Sampling at $2f$: easier to reconstruct
Sound, Audio, Speech, ...

- **Sound**: vibration of matter
- **Audio**: audible sound (by humans)
  - human audible spectrum: 20 Hz - 20kHz
- **Speech**: a particular type of sound
  - we hear better than we talk
  - speech spectrum: 50-10 kHz
- **Music**: a particular case of non-speech sound
## Audio bit rate requirements

<table>
<thead>
<tr>
<th>Quality</th>
<th>Technique or standard</th>
<th>Kbps</th>
<th>Compr.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Telephone quality</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Standard</td>
<td>G.711 PCM</td>
<td>64</td>
<td></td>
</tr>
<tr>
<td>Standard</td>
<td>G.721 ADPCM</td>
<td>32</td>
<td>Y</td>
</tr>
<tr>
<td>Lower</td>
<td>G.728 LD-CELP</td>
<td>16</td>
<td>Y</td>
</tr>
<tr>
<td>Lower</td>
<td>GSM</td>
<td>13</td>
<td>Y</td>
</tr>
<tr>
<td>Standard-</td>
<td><strong>G.729 LD-CELP</strong></td>
<td>8</td>
<td>Y</td>
</tr>
<tr>
<td>Lower+</td>
<td>CELP</td>
<td>5-7</td>
<td>Y</td>
</tr>
<tr>
<td><strong>CD Quality</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Consumer CD-audio</td>
<td>CD-DA</td>
<td>1441 (stereo)</td>
<td></td>
</tr>
<tr>
<td>Consumer CD-audio</td>
<td>MPEG with FFT</td>
<td>192-256</td>
<td>Y</td>
</tr>
<tr>
<td>Sound studio quality</td>
<td>MPEG with FFT</td>
<td>384</td>
<td>Y</td>
</tr>
<tr>
<td>Consumer CD-audio</td>
<td>MPEG2.5 Layer III</td>
<td>128 (stereo)</td>
<td>Y</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Which bit rate is actually needed?

Network overheads incl.:
- RTP header (12 bytes)
- Transport Protocol header (usually UDP, 8 bytes)
- IP header (20 bytes)
- Example:
  raw G.711 64 Kbps requires from 68 to 80 Kbps

However, speech contains silence
Silences in speech

- **Monologue**
  - Typically 15% silence

- **Bi-party telephone conversation**
  - 20% silence for overall conversation
  - 60% silence for each party

- If silence suppressed, required bit rate is in effect <40% of nominal raw bit rate
Audio does not eat bandwidth

Voice packets will swim in an ocean of data packets
• Key requirements

• Bit rate requirements
  • Audio requirements
  • **Video requirements**

• Delay requirements
  • Jitter
  • Inter-media synchronization

• **On compression ...**
# Video bit rate requirements

<table>
<thead>
<tr>
<th>Quality</th>
<th>Technique or standard</th>
<th>Mbps</th>
<th>Compr.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video conf. quality</td>
<td>H.261</td>
<td>0.1</td>
<td>Y</td>
</tr>
<tr>
<td>VCR quality</td>
<td>MPEG-1</td>
<td>1.2</td>
<td>Y</td>
</tr>
<tr>
<td>Broadcast quality</td>
<td>MPEG-2</td>
<td>2-4 (1)</td>
<td>Y</td>
</tr>
<tr>
<td>Studio-quality digital TV</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Uncompressed</td>
<td>ITU-R 601</td>
<td>166</td>
<td></td>
</tr>
<tr>
<td>Compressed</td>
<td>MPEG-2</td>
<td>3 to 6 (2)</td>
<td>Y</td>
</tr>
<tr>
<td>HDTV</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Uncompressed</td>
<td>CD-DA</td>
<td>2000</td>
<td></td>
</tr>
<tr>
<td>Compressed</td>
<td>MPEG-2</td>
<td>25 to 34</td>
<td>Y</td>
</tr>
</tbody>
</table>

(1): future; current implementations: 4 to 7
(2): future; current implementations: 6 to 10

Reproduced from "Understanding Networked Multimedia" by François Fluckiger, Prentice Hall 1995
- **Key requirements**
- **Bit rate requirements**
  - Audio requirements
  - Video requirements
- **Delay requirements**
  - Jitter
  - Inter-media synchronization
- **On compression ...**
**Network Transit Delay**

- **Telephone conversation:**
  - Round-trip delay $< 400$ ms
  - for natural conversation

- **Virtual reality**
  - Round-trip delay $< 100$ ms
  - for impression of immersion
Transit delay variation (Jitter)

- Receiver to wait a delay offset before playout

- Called delay equalization

- Increases overall end-to-end latency
• Key requirements
• Bit rate requirements
  • Audio requirements
  • Video requirements
• Delay requirements
  • Jitter
  • Inter-media synchronization
• On compression ...
Inter-media synchronization

- Called *orchestration*

- Particular case: *lip synchronization*

  - A skew of 80-100 ms is generally tolerated

  $\text{Net} < 100 \text{ ms}$
Audio/video relative priorities

- The ear behaves as a differentiator
- The eye behaves as an integrator
- Tolerance of transmission errors affecting sound much lower than for video
- When audio and video streams part of the same application competing for network resources, audio stream should have priority

See UNM book, p. 347-348
• Key requirements
• Bit rate requirements
  • Audio requirements
  • Video requirements
• Delay requirements
  • Jitter
  • Inter-media synchronization
• On compression ...
Audio-compression techniques

Encoding techniques
- DPCM, Delta
- ADPCM

Source compression techniques
Based on psycho-acoustic model
- Transform encoding (all sounds)
  - Fast Fourier Transform (FFT)
  - Wavelet transform
- Source modeling/synthesis coding (for speech)
Psycho-acoustic and Masking

- Response of ear to frequency:
  - ear most sensitive between 2 - 5 kHz

- Masking:
  - ear does not register energy in some frequencies band, when there is more energy in a nearby band
Physiology and masking

- Ear membrane vibrates as a function of frequency
  - High frequencies:
    - at one end
  - Low frequencies:
    - at opposite end
- Vibration of a area forces close areas to vibrate at the same frequency, and not at their own

Ear Membrane

low freq.  

high freq.

High amplitude vibrating area

Masked areas
Voice modeling techniques

- Human Vocal system model relies
  - on a set of cylinders of differing diameters
    (e.g. 10 in LPC-10)
  - excited by a signal at a certain frequency

- Operates over 20 ms, on standard PCM samples
Principle (or platitude)

This is what we perceive that count, not what the physical reality is!

or

The Reality is what we perceive
Information rate, bit rate, entropy

- Information rate is different from bit rate

- Information content or entropy of a sample:
  - a function of how different it is from the predicted value

- Shannon’s theory:
  - any signal which is totally predictable: carries no information
    - (e.g. a sine wave)
  - noise is completely unpredictible: high entropy
Real, contentless signals, noise

Real signals

lie somewhere between the two extremes

Real signal

total predictability (no information)

total unpredictability (noise)
Real signals only occupy a portion of the Signal level - Frequency plane.

Entropy area: where the information lies.

Redundancy area adds no information.

Compression: removes (part of) the redundancy.
The Effect of compression

Compression removes redundancy ... but
Redundancy is essential for resistance to errors.
The Effect of compression

Compression removes redundancy

- Redundancy essential for resistance to errors
- Compressed data more sensitive to errors
End of

Part 1

Requirements of Multimedia over the Internet
Part 2

Transporting Multimedia over the Internet
Who has a **good or fair understanding** of the difference between …

- FTP and HTTP
- HTTP and TCP
- TCP and UDP
- UDP and RTP
- RTP and RTCP
TCP behaviour

- **Slow start**

- **Sender aware of packets dropped**

- **Sender decreases bit rate when packet dropped**
**UDP behaviour**

- **UDP sends blindly to a receiver**
- **No feedback from the receiver**
- **Sender unaware whether packets are dropped/lost**

**Process:**
- Full speed start
- Less data to send
- More data, packets dropped
- UDP maintain packet rate
  - No feedback from the receiver
  - Sender unaware whether packets are dropped/lost
Protocols for real-time audio and video

- Audio/video applications cannot operate over TCP

- They use UDP
  - which has no timestamp, feedback, ...

- All applications use RTP (Real-Time Transport Protocol)
  - time-stamp
  - packet loss detection
Real-Time Protocol

- **RTP**: an Internet IETF standard

- **Supports**
  - timing reconstruction: timestamp (4 bytes)
  - loss detection: sequence number (2 bytes)

- **Lighter than TCP**
  - no retransmission, no flow control
  - TCP header: 20 bytes; RTP header: 12 bytes
Real-Time Protocol services

- Two parts in RTP
  - \textit{RTP} per se: for carrying data
  - \textit{RTCP}: to identify participants, monitor the quality of the service

- Session control (\textit{RTCP})
  - Receivers send periodically “reports”
  - “Reports” indicate how good the reception is
- **RTP**
  - Needs and Principles
  - **Header overhead**
- **End-systems improvements**
  - Redundancy coding
  - Error concealment
- **Quality of Service**
- **Unfair competition ...**
On header overhead

- IP+UDP+RTP headers = 40 bytes

- At 64 Kbps PCM
  - 20 ms = 160 Bytes
  - overall rate = 80 Kbps

- At 8 Kbps (e.g. with G.729)
  (e.g. over modem lines)
  - 20 ms = 20 Bytes
IP, UDP, RTP compression

- IP/UDP/RTP compression specified by
  - Robust Header Compression (ROHC) IETF draft

- Can reduce to 1 byte (best case)

- Operates on a link-by-link basis
Basic principles

- **Fixed fields removal**
  - parts of the headers remain unchanged between pkts

- **Differential encoding**
  - some fields vary in a predictive, monotonic way

- **Re-coding combinations of fields**
  - some fields may be combined and hash coded
• RTP
  • Needs and Principles
  • Header overhead

• End-systems improvements
  • Redundancy coding
  • Error concealment

• Quality of Service

• Unfair competition ...
Low-bit rate redundancy

Compression aims at removing redundancies ... but redundancies improve resistance to data errors

- re-code each packet at lower resolution
- insert re-coded packet into one subsequent pkt(s)
Example of redundancy: RAT (UCL)

- Primary coding = 64 Kbps
- Redundancy coding = 4.8 Kbps
- Experiment shows reasonable repair with high loss rate (40%)
• RTP
  • Needs and Principles
  • Header overhead
• End-systems improvements
  • Redundancy coding
    • Error concealment
• Quality of Service
• Unfair competition ...
Error concealment (audio example)

Replace missing packet with

- silence
  - “OK” if pkt<16ms, loss rate<1%; beyond, clipping effect (1)

- white noise
  - (better than silence)

(1) “OK” means tolerable; does not mean unnoticed
Phonemic Restoration

- brain uses **phonemic restoration**: 

  “the ability of the brain to subconsciously repair a missing segment of speech with the correct sound”

- phonemic restoration
  - occurs better when missing segment replaced by *noise* instead of silence
• RTP
  • Needs and Principles
  • Header overhead

• End-systems improvements
  • Redundancy coding
  • Error concealment

• Quality of Service

• Unfair competition ...
Improving QoS

Combination of solutions

- **Integrated Services**: RSVP Protocol
- **Differentiated Services**
IntServ principles

- **Resource reservation** is necessary

- Reservations on a **per-flow** basis

- Routers have to maintain **flow-specific states**
RSVP protocol (simplified)

- “path” control message sent periodically by source
- “path” establish an RSVP route in intermediary routers
- Sink replies with a “resv” message, according to its capabilities
- “resv” reserve resources in node on the route back
- If “path” not repeated after time-out, resources released
- “path” and “resv” are carried by ordinary best-effort datagrams

See UNM book, p. 441
Concern: Scalability

- Problem
  - how many soft-states can network handle at a time?
  - problem of granularity of the flows
RSVP scalability

- Millions per-flow RSVP reservations in high speed backbone?

  - **Problem 1:**
    - **Signaling overhead**
      - CPU: PATH/RESV processing
      - Memory: states

  - **Problem 2:**
    - **Data pkt overhead:**
      - CPU: Multi-field classification

many per-application-flow reservations
excessive router overhead
Solution to problem 1 (Signaling Overhead)

- **RSVP aggregation** in core

- Only one reservation between ingress/egress pairs or routers
  - If N boundary routers,
  - N^2-N RSVP reservations

- Size of the aggregate does not need change for every new flow request
Problem 2: overheard of classification

- Recognizing pkts which belong to reserved pipes:
  - heavy process!

- Requires
  - examining multiple fields
    (source, dest addresses, ports)
  - match them against “filters”
Differentiated Services: principles

- **Rationale**
  - Knowing pkt priority needs heavy classification process
  - Classification may be faster if packets are "marked" (1) with a priority

(1) “Marking” pkts also called “coloring”
**Differentiated Services Packet marking**

- Packets carry explicitly their priority

---

**RSVP flows**

**Best effort flows**

- High Priority
- Low Priority
Principle (or platitude)

- Systems with no reservation (e.g. connectionless networks) **scale well**, but are **poor at QoS guarantees**
  
  Too bad for IP, Ethernet

- Systems with reservations (e.g. connection-oriented networks) are **good at QoS guarantees and poor at scaling**
  
  Too bad for RSVP, ATM
• RTP
  • Needs and Principles
  • Header overhead
• End-systems improvements
  • Redundancy coding
  • Error concealment
• Quality of Service
• Unfair competition …
Unresponsive flows

- Unresponsive flows do not react to congestion indication (pkt loss)

- Can create
  - bandwidth starvation inflicted to well-behaved responsive traffic
  - Congestion collapse (network busy transmitting pkts that will never reach dest.)
Unfair competition

Case 1:
- 2 LANs (10 Mbps)
- interconnected with T1 and a pair of routers

Competition between
- 3 TCP connections and
- 1 UDP connection

From S. Floyd et al, February 97
Unfair competition

Delivered bandwidth ("goodput")

TCPs

UDP

From S.Floyd et al, February 97
Case 2:
- 2 LANs (10 Mbps)
- interconnected with T1 and a pair of routers
- UDP receiver connected via ISDN (dual)

Competition between
- 3 TCP connections and
- 1 UDP connection

From S. Floyd et al, February 97
Congestion Collapse

Delivered bandwidth ("goodput")

TCPs

UDP

UDP sending rate

From S. Floyd et al, February 97
Future or non-TCP applications

- In the future, UDP traffic may be penalized in public Internets
- Real-time Applications will have to behave "more responsively" to congestion indications
  - adaptive compression levels
  - hierarchical encoding/compression
  - frame, block rate reduction for video
  - ...

In the future, UDP traffic may be penalized in public Internets.

Real-time Applications will have to behave "more responsively" to congestion indications:
- adaptive compression levels
- hierarchical encoding/compression
- frame, block rate reduction for video
- ...

Future or non-TCP applications
“640KB should be enough for anybody”

Bill Gates, 1981
“We’ll have infinite bandwidth in a decade’s time.”

Bill Gates, 1994
Predictions is a difficult art, in particular when ... 

... predicting the future

W. Allen
When you have no memory of the past, you cannot predict the future.
Where the future is?

Dr Raphael Nunez
- Prof. Cognitive Psychology and Ethno-Mathematics,
  Uni of Friburg, Switzerland

*Studied how a people in South America see* 
mentally “the future”

- Spatial Localization of the future becomes a 
phenomenon of “naturalization”
Where the future is?
Thank you