Lecture notes for SSY150: Multimedia and video communications

IP-Based Network for Video Communications

(for lecture 9)

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May 5, 2009

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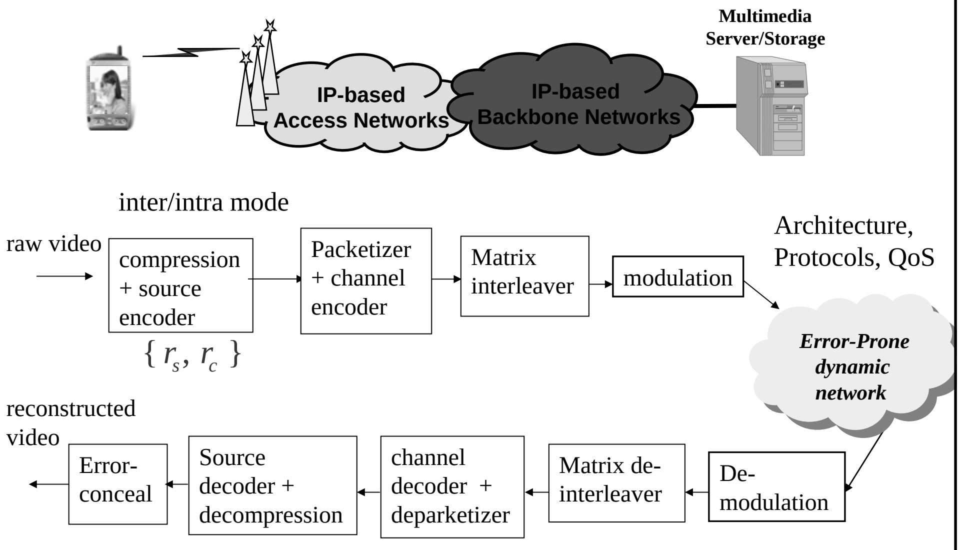
1. Introduction

Compressed Video over Communication Networks



2. Categories of Video Transportation Applications

- ***Conversational***

  (e.g. video telephony, video conferencing, distance learning)

  2-way, point-to-point (or, small multipoint) transmission.
  strict end-to-end delay (< a few hundred milliseconds)
  real time video codecs

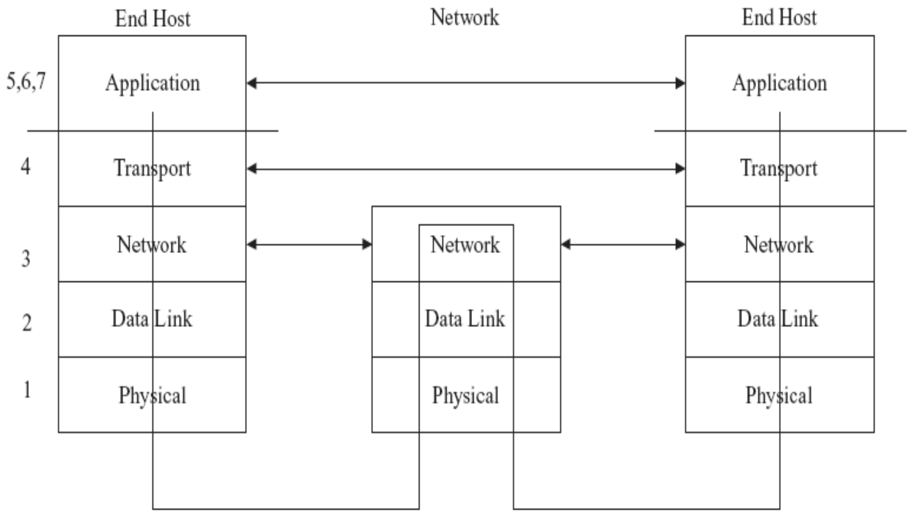
- ***Video download and storage***

  pre-encoded (offline) video stored on a server,
  video bit stream is the same as a regular data file,
  use reliable protocols (e.g., FTP, HTTP) to transmit

- ***Video streaming***

  few seconds of setup or buffering time
  once playback starts, it must be on real time
  transmit video stream: point→ point / multipoint / broadcast

3. Components and Adjustable Parameters in the End-to-End System: A Big Picture



IP 5 layer models

Parameters/components related to error control and rate control

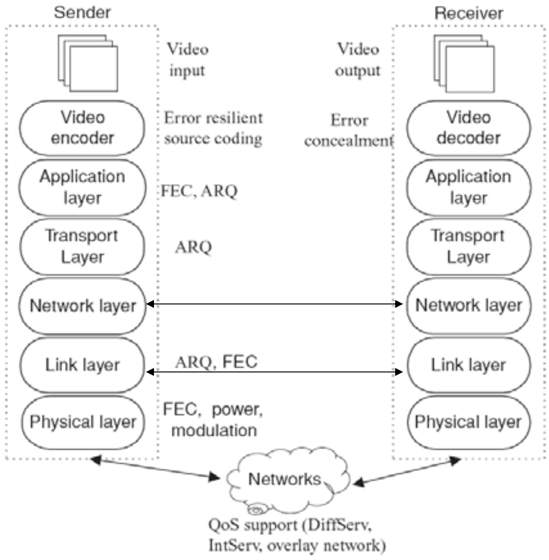


Figure: components that error and rate control can be applied for improving end-end performance

(From Dr. Fan zhai)

Components for error control & rate control

<u>**At the sender side**</u>:

Source encoder:

- error resilient (compression methods, VLC, scalable, multiple description coding)

- *rate control*: inter/inter mode selection for bit allocations, motion compensation schemes, intra refreshing rate

<u>At the sender side</u> (cont'd):

FEC for inter and intra packets (application/link/physical layer)

to combat network/channel errors and control of congestion

- channel coding *across* packets (application/transport layer)
- channel coding *<u>within</u>* packets (link layer)

4

ARQ (automatic repeat request): re-transmit lost packets in the physical layer (application/transport/link layer)

Control/adjust transmission power in wireless for prioritized packet transmission (physical layer)

<u>At the receiver side</u>:

- error concealment

In the network:

QoS classes based on congestion control models (e.g. TCP-friendly …)

QoS support for IntServ and DiffServ (integrated services, differentiated services)

Intelligent network management: e.g. Routing

Congestion control: aims at sharing network resources fairly with other users in the transport layer.

Challenges

- Quality of Service (QoS): in terms of: bandwidth, packet loss probability, delay distribution, throughput.

There is a mismatch between the video applications and communication networks!

- Channel noise, especially in wireless channel due to fading, multi-path, shadowing effect, resulting in higher bit error rates and low throughput for wireless video communications. Current standards discard (at the link layer) a packet with unrecoverable bit errors. Packet losses due to erasure channels.

- Error control: Limit the effect of channel/network errors

- Congestion control: share network resources fairly with other users at the transport layer, routing;

- Throughput: limited by the network resources

More challenges for wireless networks

- **Special issues needs to consider:**
  - \+ Delay;
  - \+ Portability of terminals (i.e. Minimum computation power);
  - \+ Robustness to mobile channel impairment

- **Special features required for compression algorithms**
  - \+ Scalable compression rate, as the channel link rate changes depending on channel quality
  - \+ Robustness to transmission errors + packet loss
  - \+ Minimum end-to-end delay, as extra delay is likely to result from wireless transmission.
  - \+ Intelligently conceal of errors, as a subscriber may move, or channel quality may deteriorate.

4. Options for improving end-to-end performance

Error resilient source coding

- Select coding/compression methods: DCT, DWT, …

- Select inter/intra mode:
  control the coding rate and prevent error propagation

  for intra mode:
  quantization step size (→ different errors),
  variable length codes (VLC), intra refreshing rate

  for inter mode:
  MC schemes, coding of MC errors

We may employ:
Joint source and channel coding

Shannon's Separation Principle breaks down:

- Finite block length in coding
- Lack of complete knowledge of channel statistics

Shannon theory for communication holds

a) Infinite block length → infinite delay

b) Stationary source and channel

Joint source and channel compression/coding that optimizes the end-to-end packet video performance

Packetization schemes and packet protection

- **UEP** (unequal error protection) **of packets**:
  different rates for packets of MVs, or DCT coefficients

- **Packetization schemes:**
  - <u>source</u> packetizer:
    ensure packets being encoded independently,
    size of packets: tradeoff between efficiency in header/payload ratio and packet loss
  - <u>re-packetizer</u> in the *intermediate layer*:
    e.g. interleaving, FEC
  - packetizer in the transport/network layer:
    avoid fragment/split (max. size ~1500 bytes)

Channel coding methods

- Channel coding can be applied to

  Inter-packets: usually for packet loss errors

  Intra-packet: usually for bit errors (especially for wireless channels)

- Channel coding methods for multimedia data packets
  - RS (Reed-Solomon) codes
  - RCPC codes (rate-compatible punctured convolutional codes)
  - Turbo codes

  ....

Reducing Dominant Types of Errors

Different networks have different dominant types of errors. One should strike for reducing the dominant types of errors in the network of concern.

For Internet/wired networks:

bursty packet loss (or, erasure channels)

For wireless networks:

high bit errors, in addition to packet loss (narrow bandwidths, multi-path fading and shadow effects)

Prevent packet losses

Packet losses:

a) occur when the network buffer overflows (network congestion)
b) discard when a packet does not arrive in time (excessive delay)
c) due to packet truncation
d) due to unrecoverable bit errors in a packet

Ways to combat the problem

- UEP in source encoders (intra/inter)
- FEC in the application/transport layer (e.g. RS codes)
- Interleaving
- Cross-layer/ joint source-channel coding (end-to-end optimization)
- Re-transmission of packets if applications allow (ARQ)

5. Main QoS parameters

Packet delay, round trip time delay, latency, delay jitter, throughput

Packet delay is caused by:

- Physical distance in transport information between 2 comm. nodes
- Encoder /decoder processing time, encoder/decoder buffer delay
- Queuing delay in the network caused by network congestion.

The round trip time delay (RTT): the time interval between sending a packet and receiving the acknowledgement of the packet.

End-to-end delay (latency): is the average delay time between the input (in the sender) and the output in the receiver.

Delay jitter (variation): is mainly caused by network congestion, multiple paths

Throughput: is the *effective* output rate over the network (excluding re-transmitted packets and the lost packets)

Main QoS parameters

- Throughput
- Packet loss / truncation
- Round trip delay time (RTT)
- end-to-end delay (latency), and delay jitter

Important notice:
QoS parameters in the lower network layers may not always reflect the QoS requirement in the application layer !

6. Network / channel modeling for computing packet loss probability

Estimate: packet loss rate

Packet loss can be estimated:

- **Empirical**, by computing at the receiver side:

$$\frac{\text{the number of lost packets}}{\text{the number of expected packets}}$$

- **Theoretical**, by using mathematical models

  (e.g. packet erasure channels with random delays)

Channel/Network Models

- Erasure channel (in the network layer)
- AWGN channel (in the physical layer, bit errors)
- Raleigh fading channel (in the physical layer, bit errors)
- Racian fading channel (in the physical layer, bit errors)

Channel / network modeling

Models can be built in different layers

Internet: packet error is modeled in the **<u>network (IP) layer</u>**
Error packets are discarded in the link layer (not forward to the network layer).

Wireless:
bit errors are modeled for the channel in the **<u>physical layer</u>**.
This requires: mapping QoS parameters from the lower layer to the higher layer!
(e.g. To be able to effectively adapt QoS parameters in the video application layer).

Modeling network packet losses

Network errors: Packet loss + Packet truncation
Delay: Queuing delay in the network

Model: independent time-invariant packet erasure channel with random delay

Overall packet losses
= packet losses in the network +excessively delayed packets

$$\boxed{\rho_k = \varepsilon_k + (1-\varepsilon_k)\, \nu_k} \qquad (1)$$

where: ε_k probability of packet k is lost in the network

ν_k probability of packet k is lost due to excessive delay

$$\nu_k = \int_{\tau>\tau_0} p(\tau \mid \text{packet } k \text{ received})\, d\tau \qquad \text{(see (2))}$$

ε_k: from a 2-state Markov chain model (see (3)), or, Bernoulli process

Modeling packet delays in the network

- Network delay τ varies randomly ~ a self-similar law.
- τ is heavy tail distributed, rather than Poisson distributed.
- A simple model*: shifted Gamma distribution

$$\boxed{p(\tau \mid \text{packet received}) = \frac{\alpha}{\Gamma(n)}\left(\alpha\left(\tau-\gamma\right)\right)^{(n-1)} e^{-\alpha(\tau-\gamma)}, \quad \tau \geq \gamma} \qquad (2)$$

n: number of routers, each being a M/M/1 queue with service rate α,
γ: total end-to-end processing time

(Chou & Niao '06, IEEE multimedia)

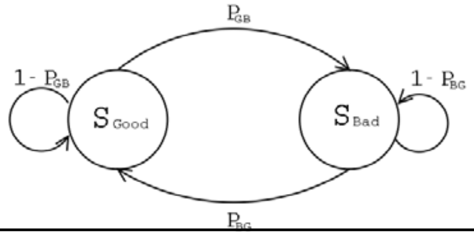
Modeling packet losses in the network layer (erasure channel modeling)

Gilbert-Elliot / 2-state Markov chain model

Channel states: 'good', 'bad' ~ success/failure of packets

Transition matrix $A = \begin{bmatrix} 1-P_{GB} & P_{GB} \\ P_{BG} & 1-P_{BG} \end{bmatrix}$

State probabilities: $P_G = \dfrac{P_{BG}}{P_{BG}+P_{GB}}, \quad P_B = \dfrac{P_{GB}}{P_{BG}+P_{GB}}$



Gilbert-Elliot 2-state Markov model

- **Probability of packet losses:**

$$\varepsilon_k = P_B = \frac{P_{GB}}{P_{GB}+P_{BG}} \qquad (3)$$

- **Average burst length:**

$$L_B = \frac{1}{P_{BG}}$$

The above model is used in the network layer, sometimes also used to describe the success/failure of the link layer packets for estimating the UDP throughput R_T

Modeling IP-based wireless channels

Wireless channel model is in the physical layer:
fading channels and AWGN channels mainly causing bit errors

need to convert to the model to the IP level! → packet erasure channel

A packet *k* is treated as lost, if a symbol in a packet cannot be recovered

Packet loss probability: $$\boxed{\rho_k = 1-(1-p_b)^{B_k}}$$

p_b : probability of BER after channel coding

B_k : Source packet size in bits

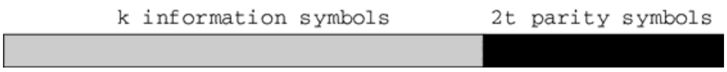
Probability of packet loss is a function of:
+ transmission power used for sending each packet
+ packet length
+ channel coding rate

7. FEC for error correction/detection, and basics of error hiding (concealment)

(a) FEC: RS codes + block interleaving

Reed-Solomon codes

• Is a Block-based code suitable for packet video (systematic encoding)



• Good correction capabilities against bursty errors.

Reed-Solomon codes

BCH codes in a Galois field: GF(2^m)

RS(n,k): codeword length = 2^m -1 (m: bits per symbol)

message length = k symbols

parity check = $n - k$ symbols

maximum corrected error = $\lfloor$(n-k)/2$\rfloor$ symbols

Block interleaving

Problems:

Bursts of symbol errors, or loss of an entire packet

Solution:

Use interleaver to spread errors.

input: codewords in rows

output: columns as to the channel

Interleaver

0,1,2,3,4,5,6,7,8,9,10,11,12,13,14,15 →

3	2	1	0
7	6	5	4
11	10	9	8
15	14	13	12

↓

12,8,4,0,13,9,5,1,14,10,6,2,15,11,7,3

De-interleaver

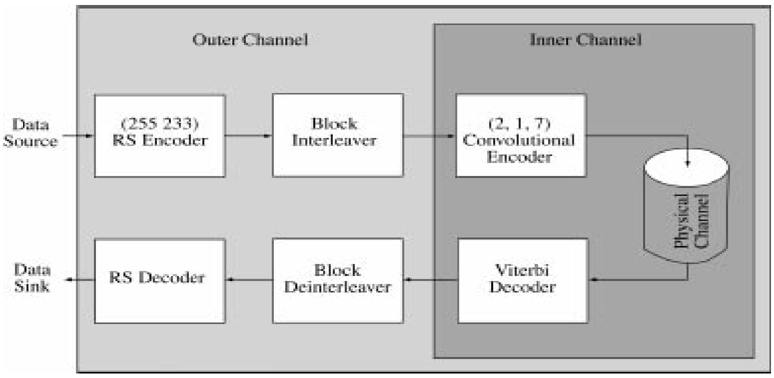
12,8,4,0,13,9,5,1,14,10,6,2,15,11,7,3 →

3	7	11	15
2	6	10	14
1	5	9	13
0	4	8	12

↓

0,1,2,3,4,5,6,7,8,9,10,11,12,13,14,15

Example: apply 2 channel codecs for video systems



- H.261 uses a (511,493) BCH code with dmin=6,which can correct a burst length of no longer than 6.
- For MPEG2, RS codes have been used.

• In European digital terrestrial TV broadcasting system, a RS(204,188) code is obtained by shortening RS(255,233) with dmin=17 as an outer code and a (2,1,7) convolutional code (sometimes rate punctured) as the inner code.

Comparison of Major Coding Techniques

Coding Technique	Coding gain(dB) at r=10^-5	Coding gain(dB) at 10^-8	Data rate Capability
Concatenated(RS and Viterbi)	6.5-7.5	8.5-9.5	Moderate
Concatenated(RS and biorthogonal)	5.0-7.0	7.0-9.0	Moderate
Block codes(soft decision)	5.0-6.0	6.5-7.5	Moderate
Concatenated(RS and short blocks)	4.5-5.5	6.5-7.5	Very High
Viterbi decoding	4.0-5.5	5.0-6.5	High
Block codes (hard decision)	3-0-4.0	4.5-5.5	High
Convolutional Codes-Threshold decoding	1.5-3.0	2.5-4.0	Very High

(b) Error concealment (basics)

Assumes image frames are highly correlated in spatial, temporal and spatial-temporal domain.

The errors from a lost packet can be concealed by using:

- Spatial correlation
- Temporal correlation
- Joint spatio-temporal

Example: concealment of packet losses using Gaussian mixture models (GMMs)

Simulated packet losses (25%) using the Gilbert model

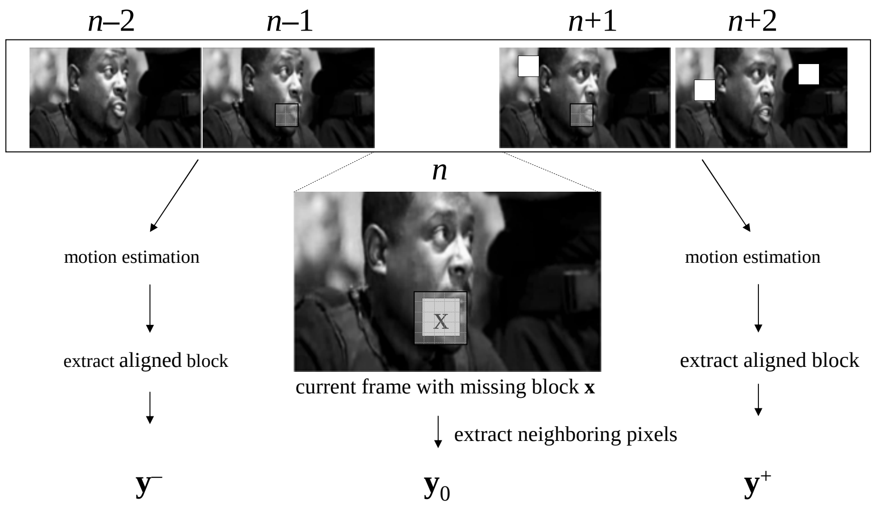
Methods used for concealment

- Assume packet loss leads to rectangular loss in time and space
- Use Gaussian mixture models, train models on a large video database
- Use adaptive estimation to recover loss

Results: (PSNR= 19.7 dB for the reconstructed parts)

(from P.Hedelin)

Example: use spatial and temporal correlation



(from P.Hedelin)

GMMs with conditional mean estimation

Compute:

Conditional mean: $\hat{\mathbf{x}} = E[\mathbf{x}\,|\,\mathbf{y}]$

Gaussian pdf : $f_{\mathbf{x}|\mathbf{y}}(\mathbf{x},\mathbf{y}) = \dfrac{f_{\mathbf{x},\mathbf{y}}(\mathbf{x},\mathbf{y})}{f_{\mathbf{y}}(\mathbf{y})}$

Gaussian mixtures: $f_{\mathbf{x},\mathbf{y}}(\mathbf{x},\mathbf{y}) = \sum_{i=1}^{M} \rho_i\, f_G(\mathbf{x},\mathbf{y};\mathbf{m}_i,\mathbf{C}_i)$

One image frame extracted from a video

before the concealment

after the concealment

(from P.Hedelin)

8. Performance evaluation: the expected end-to-end distortions

A. Pixel-based computation:

<u>expected</u> end-to-end distortion for packet *k*

$$E(D_k) = \frac{1}{N_k}\sum_{j=1}^{N_k} E\left[d\left(f_j, \tilde{f}_j\right)\right]$$

where: N_k total number of pixels in *k*-th packet

f_j j-th pixel value within the original image packet

$\tilde{f}_j$ j-th pixel value from the reconstructed packet at the receiver

MSE as the distortion: $$E(D_k) = \frac{1}{N_k}\sum_{j=1}^{N_k}(f_j - \tilde{f}_j)^2$$

PSNR as the distortion: $$E(D_k) = 10\log_{10}\left(\frac{255^2}{MSE_k}\right) \quad (dB)$$

B. Packet-based computation:

expected end-to-end distortion for packet *k*:

$$E(D_k) = (1-\rho_k)E[D_{R,k}] + \rho_k E[D_{L,k}]$$

probability of a packet *k* lost: ρ_k , R/L: packet received/lost

<u>e.g., in an error concealment scheme,</u>
assume if packet *k* is lost, then we replace it by packet *(k-1)*

$$E(D_k) = (1-\rho_k)E[D_{R,k}] + \rho_k(1-\rho_{k-1})E[D_{C_R,k}] + \rho_k\rho_{k-1}E[D_{C_L,k}]$$

$D_{C_R,k}$: distortion after concealment when previous packet is received

$D_{C_L,k}$: distortion after concealment when previous packet is also lost