Error Detection and Control Schemes

Corso di
Networked Multimedia Systems

Master Universitario di Primo Livello in
Progettazione e Gestione di Sistemi di Rete

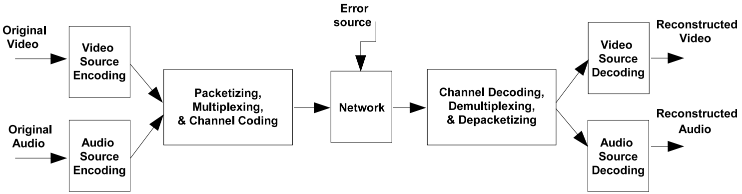
Università degli Studi di Verona

Facoltà di Scienze Matematiche,
Fisiche e Naturali

Error Detection and Control Schemes: OUTLINE

Multimedia transmission services

A typical transmission system for audiovisual data



Networked multimedia services and applications

- Audio and video downloading (E.g., video/audio on demand)
- Audio and video data streaming over IP
  (E.g., TV and radio broadcasting)
- Real-time interactive audio and video communication
  (E.g., Videoconferencing, Voice over IP)

Error sources

- Real networks are unreliable
- Internet is a packed-switched, best-effort delivery service: $\Rightarrow$ packet loss, delay, variable delay (packet jitter), packet resequencing
- Wireless packet networks are affected by noise, interference bursts, etc. $\Rightarrow$ packet errors

Today, an increasing number of network-based services place more stringent requirements on data networks. E.g., typical delay requirements for audio-visual applications are:

- Tight (up to 200 ms)
- Medium (200-500 ms)
- Loose (larger than 500 ms)

Application scenarios vs network requirements

Characteristics of continuous media streams (audio, voice, video, animation):

- Strict timing requirements:
  to be useful, data must be delivered before a certain instant
- Some tolerance of loss:
  depends on medium, coding techniques, and human perception
- Periodicity:
  frames per second for video, samples per second for audio

Application scenarios vs network requirements

Application Scenario	Interactive voice	Interactive video	Noninteractive video
ADU size	short fixed	long variable	long variable
ADU Priority	Single	Multiple	Multiple
ADU Availability Interval	Constant	Variable	Constant
Time Interval of Relevance	Constant	Variable	Variable
Data Rate (typical values)	Low (10-64 Kb/s)	Medium-high (1.5 Mb/s)	High (6 Mb/s)
Delay Budget (typical values)	Tight ($<$ 200 ms)	Medium (200-500 ms)	Loose ($>$ 500 ms)

source: (Li 2001)

Reliability for multipoint services

Transport services (protocols) may provide different degree of reliability; they can involve single as well as multiple sources and receivers:

- fully reliable multicast
- reliable real-time multicast transport
- single source (1:N or multicast)
- multiple sources (M:N multipeer service)

Error detection and control

Error detection and and control is based on the principle that a channel can be made arbitrarily reliable if some redundancy is added (Shannon)

General characteristics of error detection and retransmission schemes:

- Improved reliability
- Low throughput if channel error rate is high

Error-coding can be broadly classified in two categories:

- Source coding
- Channel coding

Error resilient source encoding

Redundancy is inserted in the source encoding process to recover from transmission errors.

The characteristics of the source can be exploited to some extent.

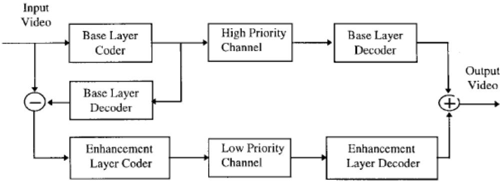
Reliability of transport can be improved at the price of increased bandwidth usage or increased delay.

Among available techniques:

- Layered coding
- Multiple description coding

Layered coding

Layered coding splits image/video into a base layer and one or several enhancement layers



source: (Wang & Zhu 1998)

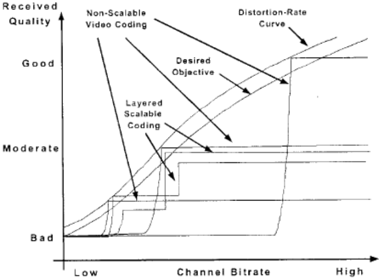
Layered coding

Possible LC schemes:

- Data partitioning
- Temporal scalability
- Spatial scalability
- SNR scalability
- Fine grain scalability (FGS)

Layered coding for video

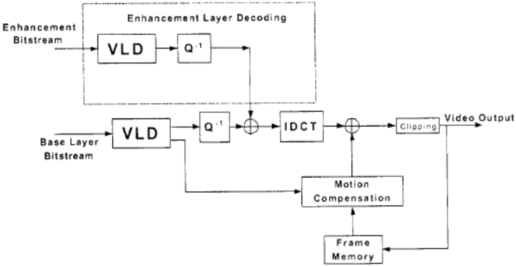
- Streaming video over the Internet requires high scalability
- Layered coding for video provides optimal quality at a given bit rate
- MPEG-2 and MPEG-4 standards includes several scalability techniques: SNR, temporal, spatial scalability, and FGS



source: (Li 2001)

SNR scalability for video

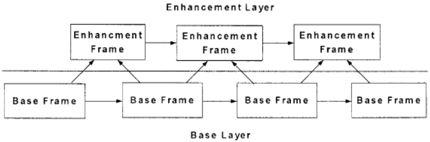
- A video sequence is encoded into two layers at same frame rate, same spatial resolution, and different quantization accuracy
- At decoder, enhancement layer is used to improve DCT coefficient reconstruction



source: (Li 2001)

Temporal scalability for video

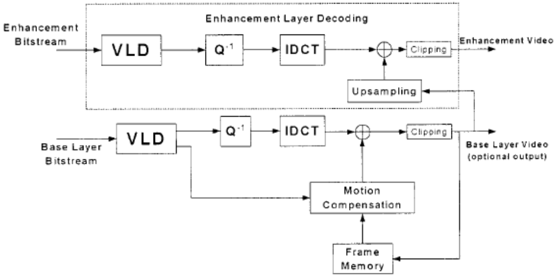
- A video sequence is encoded into two layers at same spatial resolution but different frame rate
- The base layer is encoded at a lower frame rate, and enhanced layer provides missing frames



source: (Li 2001)

Spatial scalability for video

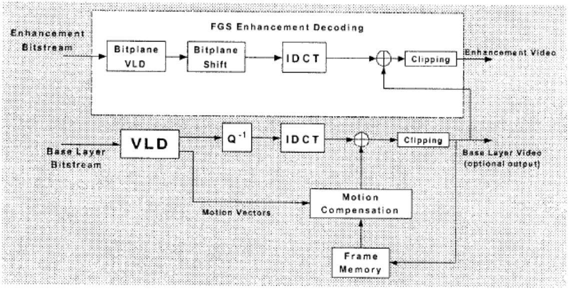
- A video sequence is encoded into two layers at same frame rate, but different spatial resolution
- The base layer is encoded at a lower spatial resolution
- At decoder, the base-layer is upsampled and combined with enhanced layer picture



source: (Li 2001)

FGS scalability for video

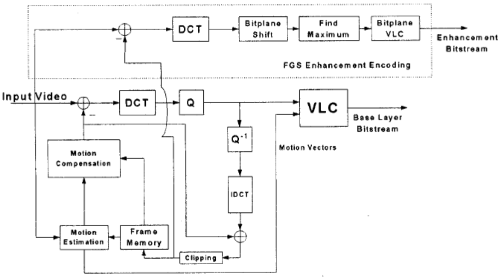
- As spatial scalability, but uses bit-plane coding of DCT coefficients
- The base layer uses nonscalable coding to reach lower bound
- The enhancementlayer codes the difference between original and reconstructed picture using bit-plane coding



source: (Li 2001)

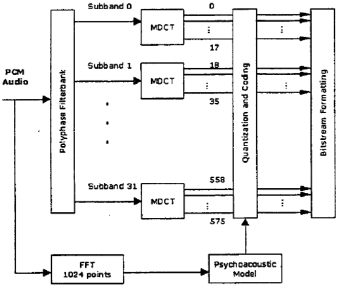
FGS scalability for video

- FGS Encoder



source: (Li 2001)

Layered encoding for audio



source: (Jin & Scordilis 2001)

- 1 Sub-band coding uses psychoacoustic masking theory to allocate bits to evenly spaced frequency sub-bands.
- 2 Layered designs of sub-band sample quantization and bit allocation is employed to produce sub-streams, which can be combined to produce layers.

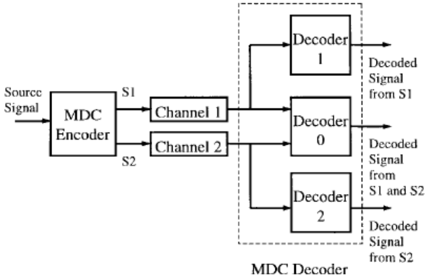
Layered coding: conclusions

Pros and Cons of Layered coding:

- Advantages:
  1. bit-rate can be instantly adjusted by discarding a certain number of enhancement layers
  2. compatible with heterogeneous terminals (from palmtop to TV)
  3. can be combined with UEP
- Disadvantages:
  1. reliable transmission must be guaranteed for base layer
  2. losses in the base layer can lead to disastrous decoding effects
  3.Limited number of bit-rate levels due to limited number of layers (solved by FGS);

Multiple description coding (MDC)

Multiple description coding assumes there are several parallel channels between source and destination



source: (Wang & Zhu 1998)

Multiple description coding (MDC)

Characteristics of MDC:

- MDC generates multiple descriptions and sends them through several independent channels
- Descriptions are correlated and have similar importance
- Any received description provides a basic level of quality
- More descriptions together provide the improved quality
- A different decoder will be invoked, depending on which descriptions are received correctly

Multiple description coding (MDC)

Some MDC schemes:

- Splitting adjacent samples among several channels
- Multiple-description scalar quantization (MDSQ)
- MDC based on correlation-inducing LT

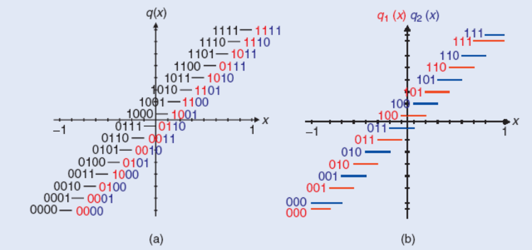
Multiple description coding (MDC)

Characteristics of Multiple-description scalar quantization (MDSQ):

- Two indexes for each quantization level produce two substreams
- If both indexes are received: equivalent to a fine quantizer
- If one index out of two: equivalent to a coarse quantizer
- MDSQ has been applied to transform-based image and video coding

Multiple description coding (MDC)

Scalar quantization for MDSQ



source: (Goyal 2001)

- a) a four-bit ubiform scalar quantizer
- b) two three-bit quantizers that complement each other (to about 4-bit res.)

Note that in b) the total rate over both channel is 6 bits per sample

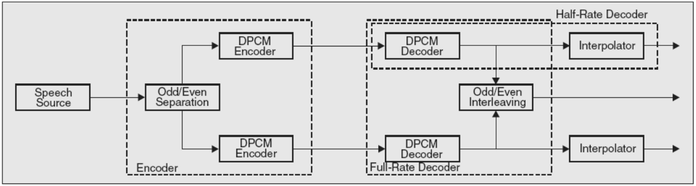
Multiple description coding (MDC)

Characteristics of MDC based on correlation-inducing linear transform :

- Use of linear transforms that do not completely decorrelate coefficients
- If all coefficient group are received maximum quality is achieved
- If some coefficient groups are lost they can be estimated from the received groups

Multiple description coding for speech

Multiple description for speech coding can be based on separation of odd and even samples:



source: (Goyal 2001)

- 1. Usually speech is bandlimited to 3.2 kHz at 8 kHz
- 2. In MDC scheme initial sampling is 12 kHz
- 3. Speech decoded from both channels has 12 kHz quality
- 4. Speech decoded from one channel has 6 kHz quality with aliasing

Multiple description coding (MDC)

Advantages of MDC:

- 1. reliability requirements on channels are less stringent
- 2. acceptable description with any one bit stream
- 3. incremental improvement with more bit streams

Disadvantages of MDC:

- 1. Each description must carry sufficient information about the original signal
- 2. Overlap in the information contained in different descriptions

Channel encoding

Channel encoding strategies add redundancy to data prior to transmission to reduce the effects of errors

Principal channel coding error-control schemes:

- Error correction (FEC):
  Redundancy is used by the decoder to reduce the error at the receiver (return path not necessary)
- Retransmission (ARQ):
  Redundancy is used to detect errors and, upon detection, to request repeat transmission (a return path is necessary)

Forward-Error Correction (FEC)

FEC is a special case of channel coding

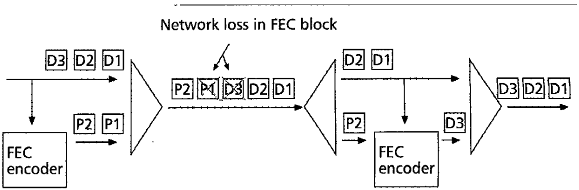
- 1 Linear block codes (Reed-Solomon, Parity check, BCH, ...):
  1. Encoding step adds extra "redundant" bits to a block of digital data
  2. Decoding is based on algebraic properties. A certain amount of errors and erasures can be detected and recovered
- 2 Convolutional codes:
  1. Encoding is based on operations on buffers
  2. Decoding is based on Viterbi algorithm

Can be used to correct bit errors and to recover from packet loss (Packet-level FEC)

Forward-Error Correction (FEC)

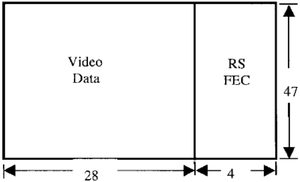
FEC error correction scheme:

- 1 An error-correcting code (e.g. block or convolutional) is used
- 2 Sender adds parity-check bits to allow for error detection at the receiving end
- 3 If an error is detected, receiver attempts to locate and correct



Forward-Error Correction (FEC)

Example of FEC error correction for video data transmission:



source: (Wang & Zhu 1998)

- 1 RS codes is combined with block interleaving to recover ATM cells
- 2 RS is applied to every block of 28 bytes to form a block of 32 bytes

Packet-level FEC

FEC is usually deployed at the physical-link layer in conjunction with advanced line-coding schemes.

Even when a network's physical link is free from bit errors, packets may still get dropped (e.g., queue overflows)

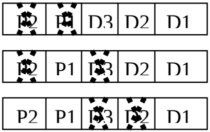
Packet-level FEC works by:

- Adding another error-recovery packet for every N packets that are sent
- This FEC packet contains information that can be used to reconstruct any single packet within the group of N
- If one of these N packets happens to be lost during transfer, the FEC packet is used to reconstitute the lost packet

Packet-level FEC

Recovery of lost packets with FEC:

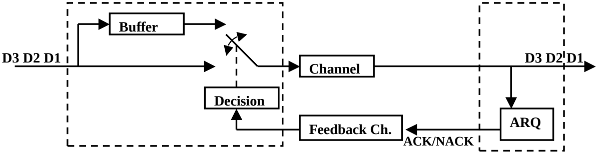
- Transmitter sends N data packets (transmission group, or TG), and adds additional H parity packets
- Receiver can can reconstruct the N data packets if up to H packets are lost
- Example:
  - $N = 3,\ H = 2$



- Reconstruction delay at the receiver increases for larger N
- Loss repair using parity is particularly efficient for multicast transmission.

Automatic-Repeat-Request (ARQ)

In ARQ schemes, sender adds parity-check bits to allow for error detection at the receiving end. If an error is detected, receiver requests a retransmission.



ARQ-based schemes

ARQ-based schemes consist of three parts:

- Lost data detection

  Loss can be detected by the receiver or by the sender
  Receiver: gap-based or timeout-based detection
  Sender: timeout-based.

- Acknowledgement strategy

  Receiver uses ACKs or NACKs to signal received or missing data.
  Requires a feedback channel.

- Retransmission strategy

  Best known: Stop-and-Wait, Go-back-N, Selective repeat

ARQ retransmission strategies

Traditional ARQ retransmission strategies:

- Stop-and-Wait

  Sender transmits PDU of information and then waits for a response (ACK/NACK) or timeout

- Go-back-N

  Sender transmits N numbered PDUs and retransmit all PDUs following a lost packet upon receiver request (efficient in case of correlated packet losses)

- Selective repeat

  Sender retransmits only lost packets upon receiver request

ARQ characteristics

Pros and Cons of ARQ:

- Pros:
  1. Only what is needed is retransmitted
  2. Easy to implement
  3. Error performance: ARQ outperforms FEC
  4. Very robust if 3-4 retransmissions allowed
- Cons:
  1. Requires a feedback channel, and protection of ACK/NACKs
  2. Decreased throughput
  3. A round-trip delay is introduced for every retransmission (not ideal for VoIP and Videoconference applications)
  4. Not adequate when delay is critical (e.g., interactive audio/video)

ARQ Example: Multicast Video

Example of ARQ scheme for non-interactive multicast video transmission

Layered Video Multicast with Retransmission (LVMR)

- MPEG video is separated in three layers
- Different layers are transmitted to different multicast groups
- Layer base: I-frames
- Layers 2-3: P- and B- frames
- If frame is lost, Rx sends NACK only if there is time to receive it before playout time
- Rx will send unicast NACK to a designated node

Soft ARQ (1)

Motivations

- In many applications based on multimedia data transfer, a certain level of end-to-end delay can be afforded
- Buffering and retransmission can be used to face delay jitter, out-of-order, and packet losses.
- Traditional ARQ not always provides the optimal retransmission choice

Alternatives for optimal retransmission

- The sender only retransmits packets that are worth transmitting
- **Soft ARQ**: a delay-constrained version of ARQ

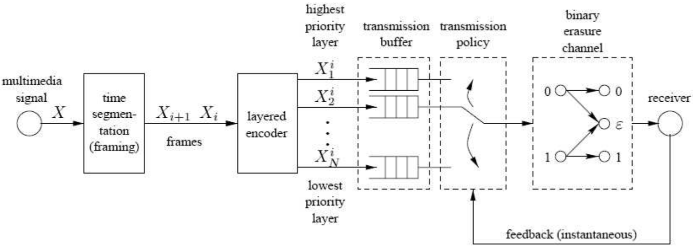
Soft ARQ (2)

Soft ARQ outline

- Retransmission decisions are based on playout delay and on importance of packets
- Can use layered coding to obtain a multi-resolution representation of media data
- Layers correspond to different levels of importance for packets

Soft ARQ (3)

System diagram of a SoftARQ based layered transmission:



source: (Podolsky et al., 2001)

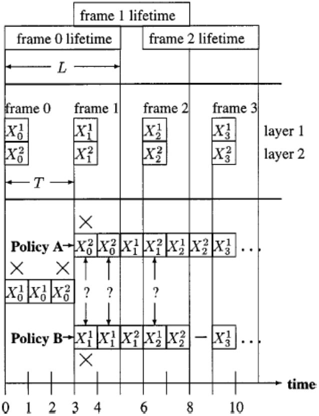
- Binary erasure channel (BEC) model: packages are dropped with probability ϵ
- Transmission policy: rules indicating which message the sender should transmit next

Soft ARQ (4)

Example of transmission sequence:

Configuration

- 2 layers
- data lifetime $L = 5$
- inter-frame period $T = 3$
- Policy A: older packets first (earlier playback)
- Policy B: higher priority packets first (greater distortion reduction)



source: (Podolsky et al., 2001)

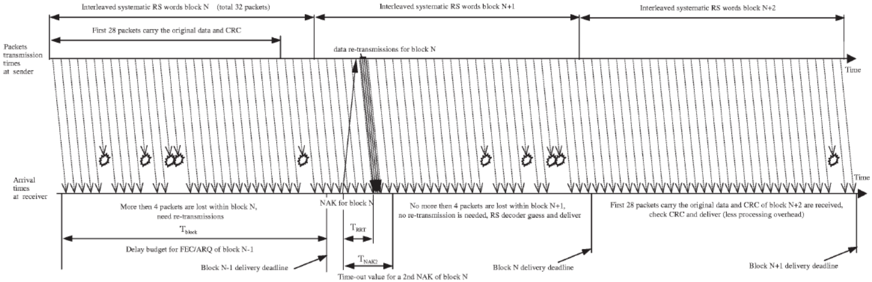
Hybrid ARQ

ARQ and FEC can be effectively combined to limit the redundancy introduced by FEC.

- Hybrid ARQ type I
  Sender adds a certain amount of redundancy using FEC; ARQ is used only when loss rate is too high
- Hybrid ARQ type II
  Sender adds FEC redundancy (parity data) only when retransmission is required.
  Better for reliable multicast (bandwidth saving)

Hybrid ARQ

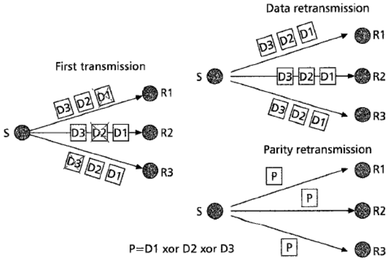
HARQ type I: example



source: (Chen at Al. 2004)

Hybrid ARQ

HARQ type II for reliable multicasting



source: (Carle & Biersack 1997)

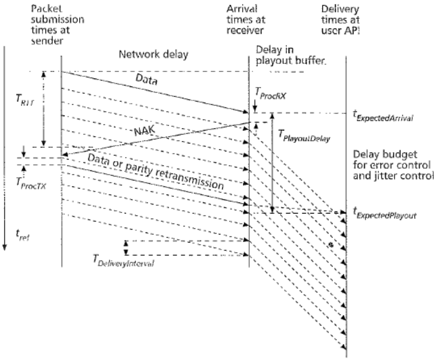
Hybrid ARQ

End-to-End delay of continuous media data transmission

- retransmission for real-time applications asks for time constraints
- jitter control and error control asks for a delay budget
- a playout buffer can be adopted
- playout buffer delay is adjusted to meet expected playout time even with retransmission
- delay budget dimensioned to take into account all transmission and processing times

Hybrid ARQ

End-to-End delay of continuous media data transmission



source: (Carle & Biersack 1997)

References

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