Multimedia communication
Delay and loss control

- Lecture material:
  - E. de Souza e Silva, et al., “Performance Issues of Multimedia Applications,” Performance 2002, tutorial
- Reading for next lecture:
  - N. Laoutaris, I. Stavrakakis, “Adaptive playout strategies for packet video receivers with finite buffer capacity,” IEEE ICC 2001.
  - X. Yu, J.W. Modestino, X. Tian, “The accuracy of Gilbert models in predicting packet-loss statistics for a single multiplexer network model,” IEEE Infocom, 2005

Multimedia communication

- Multimedia communication classes and requirements
- Delay control
  - Playout buffer control
- Loss control
  - Forward Error Correction (FEC)

Multimedia transmission

- Different cases of media distribution according to
  - availability of content
  - delay requirements
- File download
  - content is available at servers or at users for P2P applications
  - download then playback
  - no delay limitations
- Video on Demand – offline streaming
  - content is available at the servers or at the users
  - playback while downloading (offline streaming)
  - some delay limitation
- Live streaming – online streaming
  - content is generated during playback
  - playback delay have to be minimized

Streaming

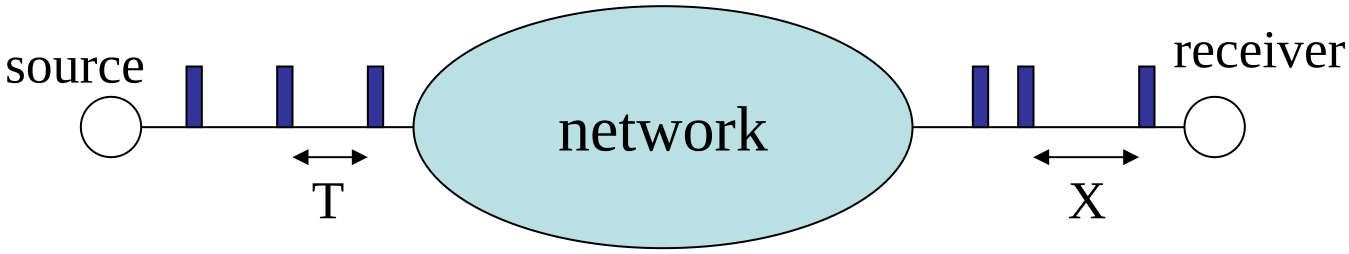
- Delay-jitter control
  - to compensate delay variations (jitter) due to varying congestion at the network nodes
  - method: playout buffer at the receiver, playout delay control
- Loss control
  - to deal with packet losses, when retransmission is too slow
  - method: increase redundancy at the source, forward error correction
- All control methods have to consider the end-to-end delay limits

- In the case of live streaming the traffic characteristics is not known a-priori, this makes jitter and loss control challenging

- In general it is hard to find appropriate end-to-end delay and loss models.

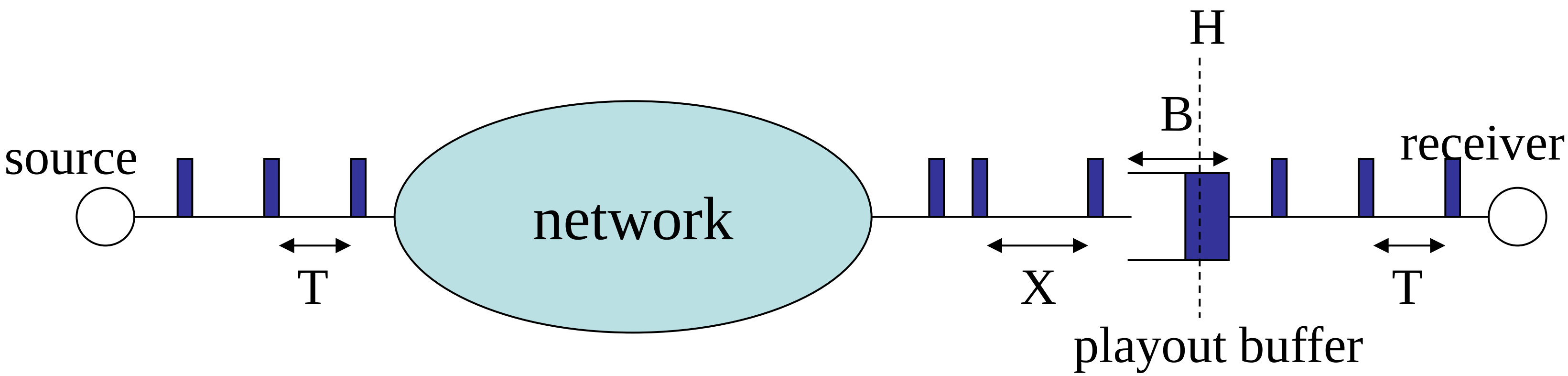
Delay-jitter control

- What is jitter:
  - T packet generation interval (constant or random v.)
  - X interarrival time at the receiver (random v.)
  - Jitter: J=X-T (random variable)

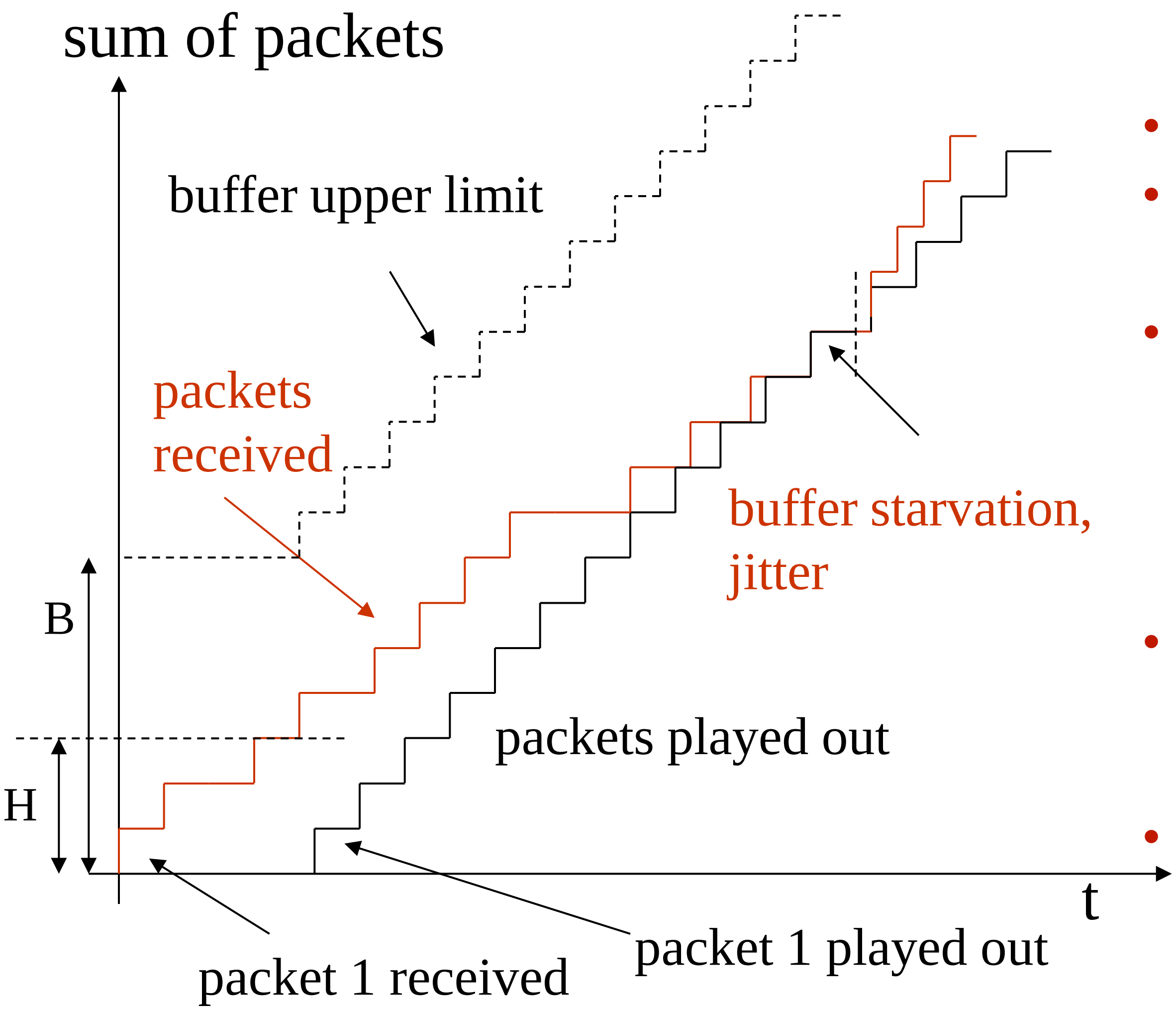


Playout buffer

- Playout buffer to compensate for jitter
  - at the receiver side
  - to store received, but not decoded packets
  - decoding rate determined by the coding scheme
  - buffer size B
  - threshold H, when decoding starts

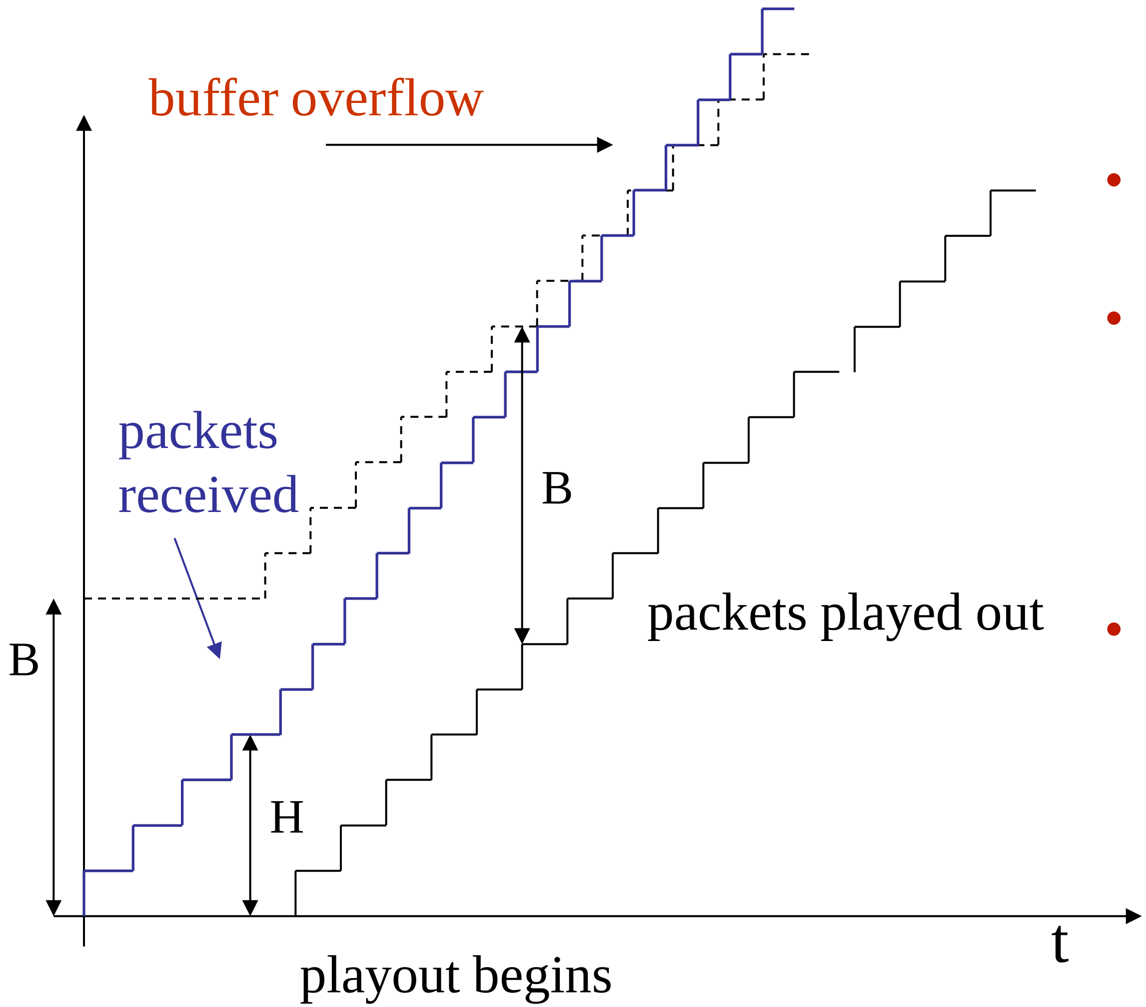


Playout buffer - starvation



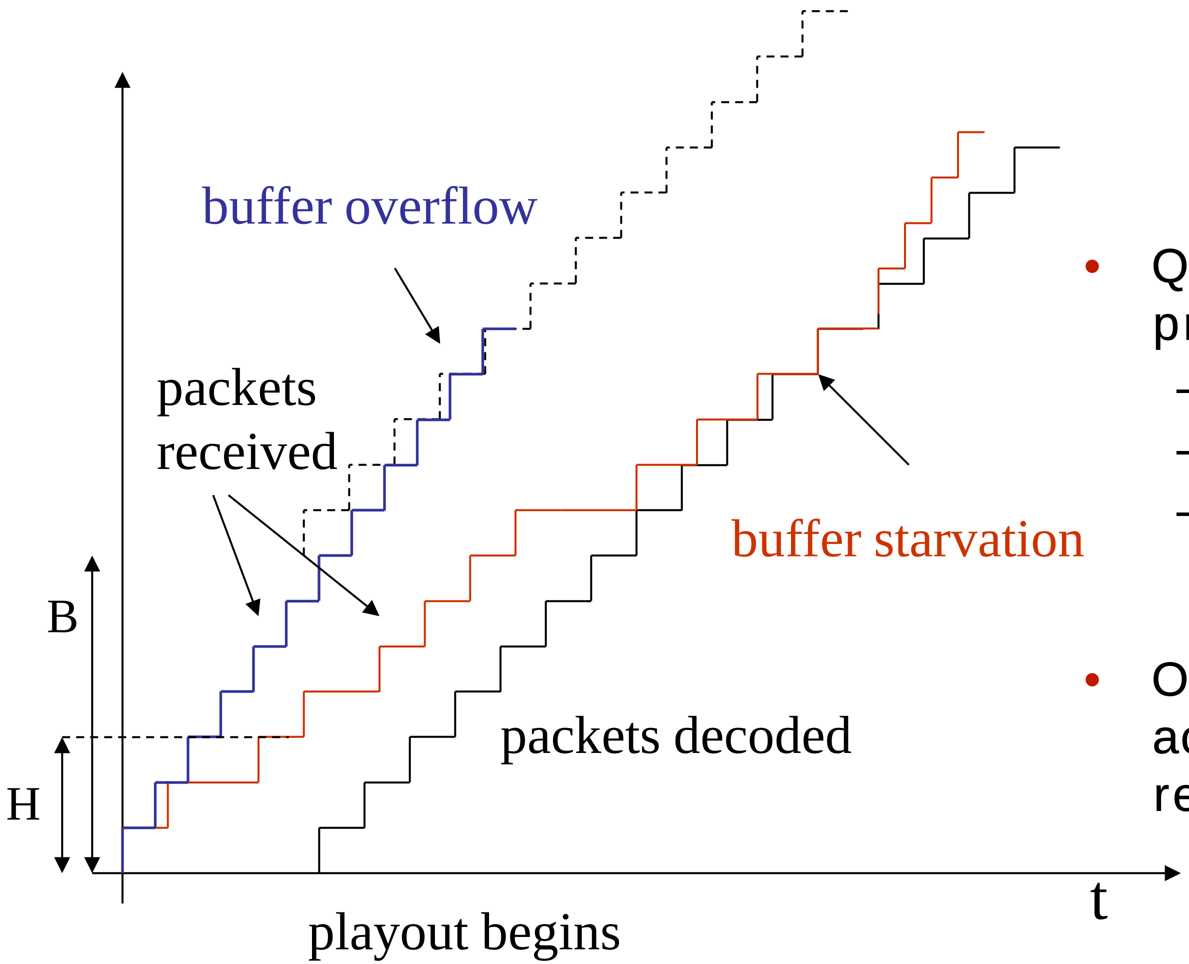
- B: playout buffer size
- H: playout threshold
- Buffer starvation: the buffer is empty, there is no available content to play out
- Jitter, increased delay between two samples
- H should be increased

Playout buffer



- For larger H:
- Buffer overflow: buffer can not store more packets, packet loss
- Large H also increases the end-to-end delay (network delay of the first packet + H)

Playout buffer



- Question: select H to provide acceptable:
  - end-to-end delay,
  - probability of starvation
  - probability of packet loss due to buffer overflow
- Other control solutions to achieve smooth reception?

Loss control for multimedia

- Loss control for elastic flows is often based on retransmission
  - This may not work for streaming applications due the the playout delay limit
  - At the same time some loss is tolerated by the application
- Solutions with lower delay – based on information redundancy
  - Application layer, utilizing the characteristics of the media content
  - Network layer – multimedia specific or not…

Loss control on application layer

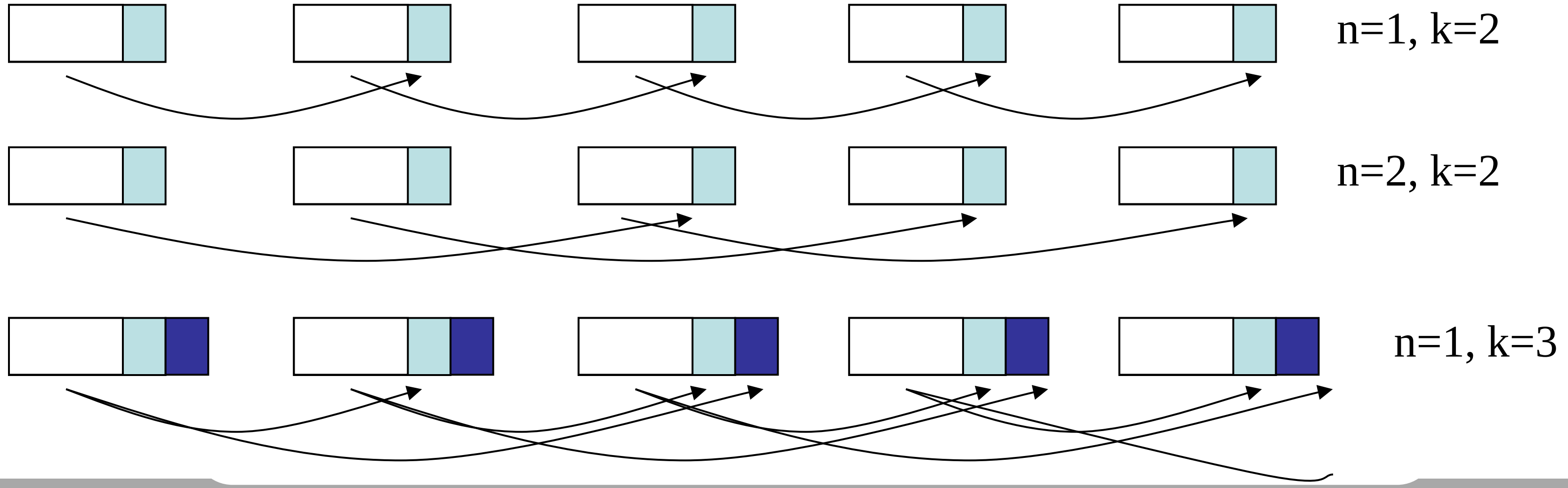
- Error concealment at the receiver
  - reconstruct the signal from available information
  - reuse last sample
  - interpolate from neighboring samples
- Error resilience with source coding
  - error propagation due to source coding (e.g., interframe coding in video coders)
  - limit error propagation (e.g., I frames in MPEG)
  - increases bandwidth
- Layered source coding
  - divide information into important and less important parts
  - transmit important information with high priority (requires network support!) or with higher redundancy
- Multiple description coding (this is source coding as well)
  - code the information in two redundant streams
  - receiving both streams gives good quality but receiving one stream only gives acceptable quality as well

Loss control for multimedia

- Loss control in the network layer (IP)
  - Interleaving of packets
    - loss can happen in bursts (buffer overflow)
    - single packet losses are easier to compensate for
    - scrambles the packets at the source
    - introduces latency, but does not consume extra bandwidth
  - Forward Error Correction (FEC)
    - adds redundancy to the stream of packets
    - uses this redundancy to reconstruct lost packets
    - introduces latency and uses extra bandwidth

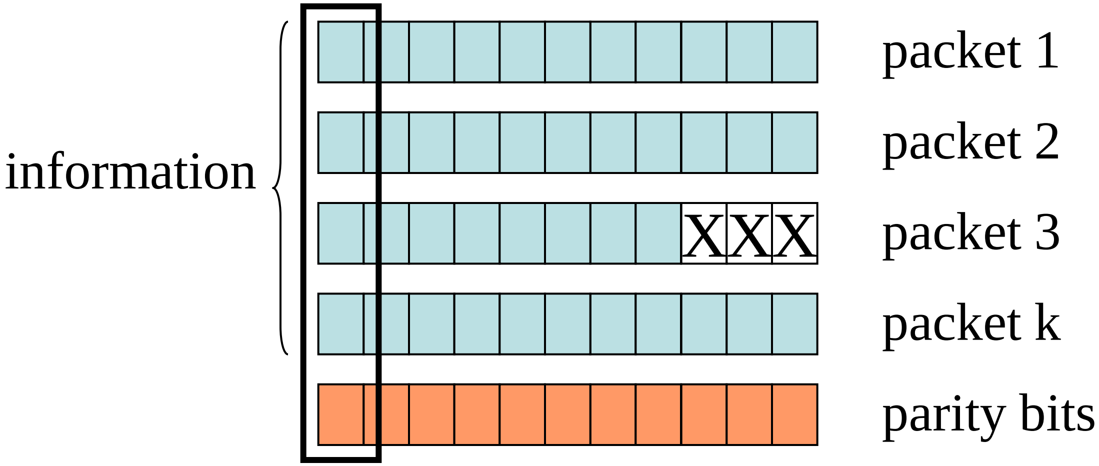
Forward Error Correction

- Media dependent FEC – parameters: n,k
  - add redundant copy to a consecutive packet
    - with low coding rate – media dependence
  - for low bitrate sources, with delay limitations
  - implemented in VoIP tools, IETF recommendation
  - for increased performance
    - increase the delay between the original and the copy (n=1,2,...)
    - add multiple copies (k=2,3,4...)



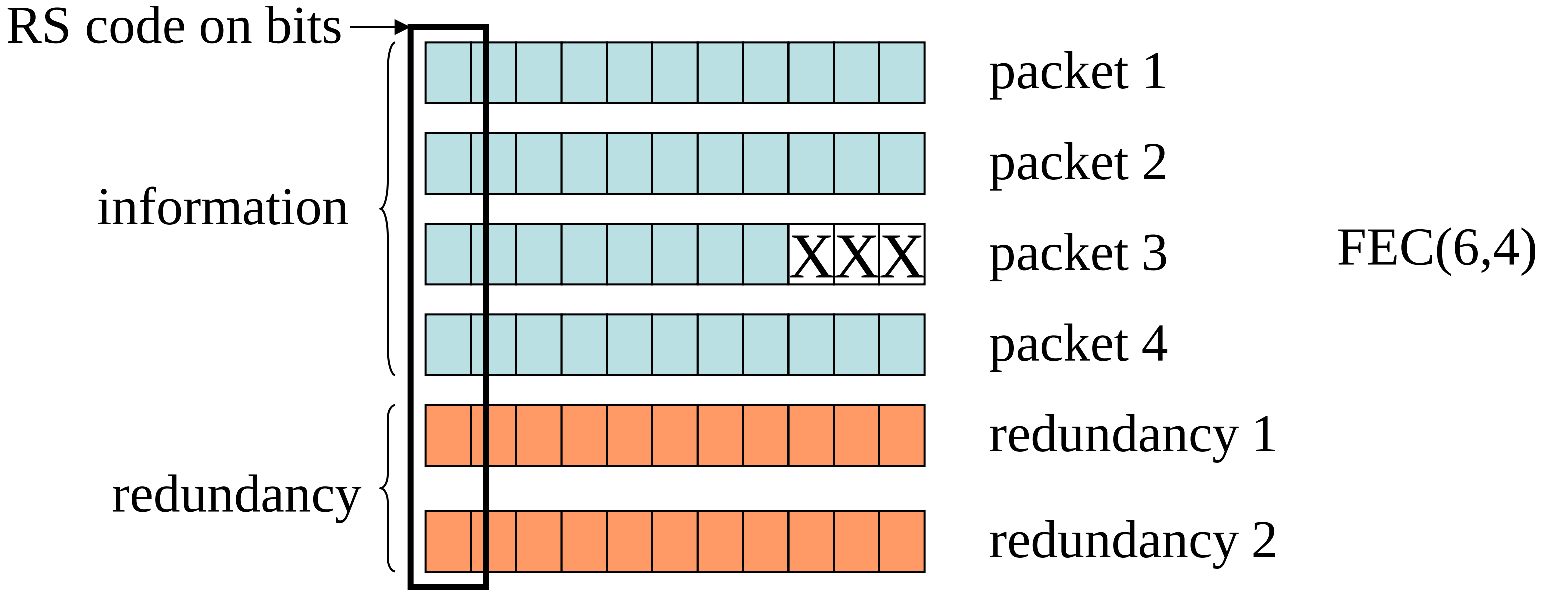
Forward Error Correction

- Media independent FEC (block codes)
  - apply error correction codes on a block of packets
  - e.g., parity bit – can regenerate one lost packet



Forward Error Correction

- Media independent FEC (block codes)
  - e.g., Reed-Solomon codes
  - FEC(n,k): k information packets, n-k redundant packets
  - all packets reconstructed if at least k received, otherwise no reconstruction
  - redundancy rate: (n-k)/n



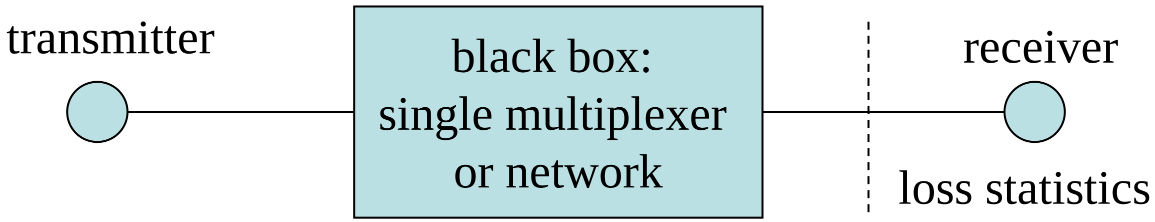
Forward Error Correction

- For both schemes:
  - performance depends both on the average packet loss probability and on the distribution of packet losses
    - consecutive packet losses (media dependent case)
    - number of lost packets in a block of n packets P(j,n) (media independent case)
  - experience: packet losses in the Internet are correlated

- Modeling the loss process at the receiver
  - detailed queuing model
  - Bernoulli model – each packet gets lost with the same probability
  - Gilbert models (reading)
  - Requirement: calculate information loss probability for simple cases

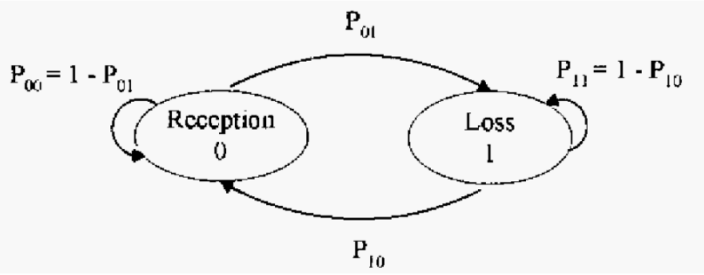
Modeling the loss process

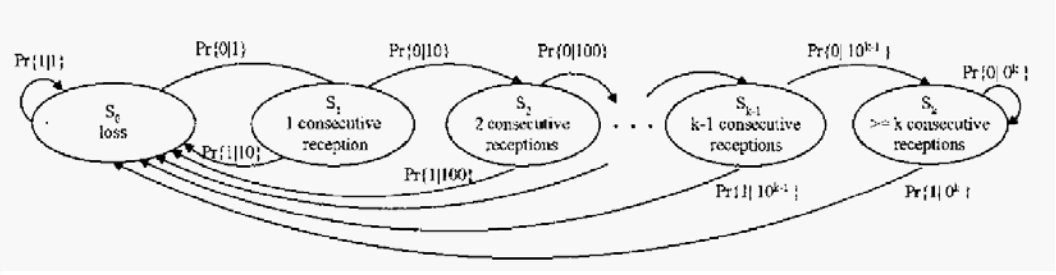
- Queuing models
  - modeling the reasons of packet losses
  - e.g., congestion: queuing networks with finite queues, appropriate arrival and service processes
  - or at least one queue, at the bottleneck link of the connection
  - needs packet arrival process and packet length distribution
  - can be exact, but very complex
- Black-box models:
  - do not model the reason, just the outcome (discrete time process)
  - simple, but not that accurate
  - Bernoulli and Gilbert models



Modeling the loss process

- Bernoulli model
  - All packets get lost with the same probability, losses are independent
- Gilbert model – basic version
  - To model the burstiness of the loss process
  - Two state discrete time Markov chain
  - Parameters to calculate:
    - Steady state packet loss probability
    - Average loss burst length
    - When does the Gilbert model reduce to Bernoulli model?
- Extended Gilbert model (reception run lengths)





Modeling the loss process
The Bernoulli example – Group work

- Bernoulli model, loss probability *p*
- Media dependent FEC(n,k)
  - n: redundant copy in the nth packet
  - k: altogether k copies (1 original, k-1 copies)

  Calculate P(information is lost) for
    - n=1, k=2
    - n, k=2
    - n=1, k
- Media independent FEC(n,k) – block code with k information packets
  - P(j,n)=P(j packet lost out of n)
  - P(losses despite redundancy)
  - N=E[lost packets in block after reconstruction]
  - I=E[lost information packets in a block after reconstruction]
  - P(information is lost)

The accuracy of loss models

- Reading assignment: The accuracy of Gilbert models ...
- Packet loss process - Gilbert models for a single multiplexer
  - Single source
  - Multiple sources
- Performance of Forward Error Correction (media independent)
  - RS(n,k): k information packets, and additional n-k parity packets
  - Steps of modeling
    - P(j,n) – Probability of loosing j out of n packets
    - Number of lost packets given j
    - Average number of lost packets in a block
    - Average number of lost information packets in a block
    - Probability of loosing an information packet

Summary

- Multimedia transmission
  - File download, off-line and on-line streaming
  - Delay and loss control
- Delay control: Playout buffer control
  - Playout buffer modeling with discrete time MC
- Loss control: FEC
  - Media-dependent and media independent
  - FEC modeling with Gilbert models