QoS Enhancement and Adaptation

Overview

- Jitter Adaptation
- Loss Recovery
- Source Rate control

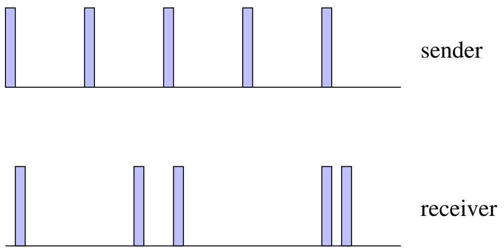
Network impairments

duplicates: not uncommon for multicast; generally harmless

packet loss: up to several percent ➠ audible clicks, loss of encoding state; not as bad for conferencing video

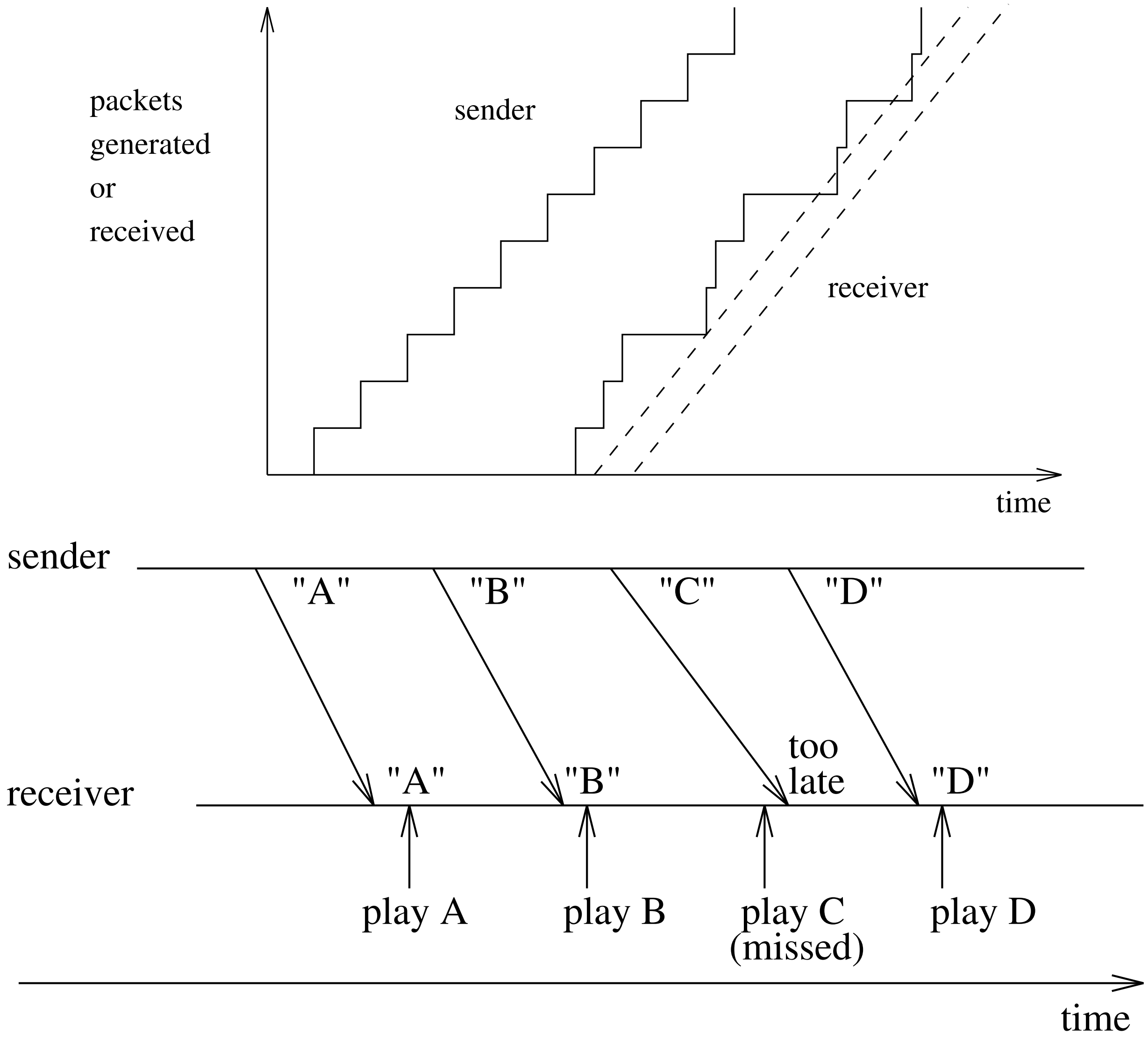
delay: due to transmission on slow links, propagation (5 μs/km), switching

delay jitter: arrival distortion:

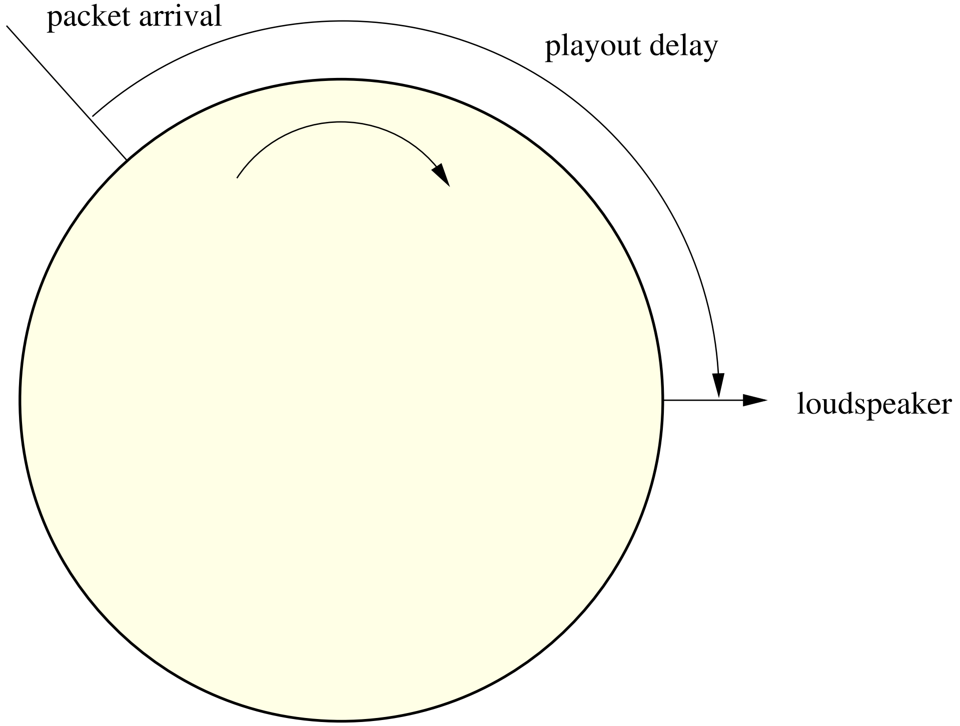


caused by queueing (resource contention) in nodes

Playout delay



Playout delay buffer



Playout delay

- only compensate for *variable* of delay δ
- packets may be lost, reordered, gaps ➠ need timestamp t
- low loss (recording, seminar) $\leftrightarrow$ low delay (telephony, discussion)
- achieve minimum possible playout delay ➠ adaptation
- can adjust delay D only at beginning of talkspurt

relative timing: play first packet after D ➠ what if first few packets bunched?
Example: $D = 200, t = 0, \delta = 50$ ➠ playout at 250

absolute timing: maintain fixed relationship
Example: playout at 200 regardless of arrival of first packet

complications: clocks are not synchronized (drift and offset); "reboot"
See NeVoT description for details

Handling packet loss

discover: gap in packet sequence (account for reordering)

retransmit: if enough time, ask for retransmission ➠ multicast dangerous!

forward error correction: like RAID ➠ transmit XOR of block of packets ➠ can recover one lost packet per block

redundancy: transmit low-fidelity version with delay

cover up: fill in waveform at receiver, e.g., based on prior and next block

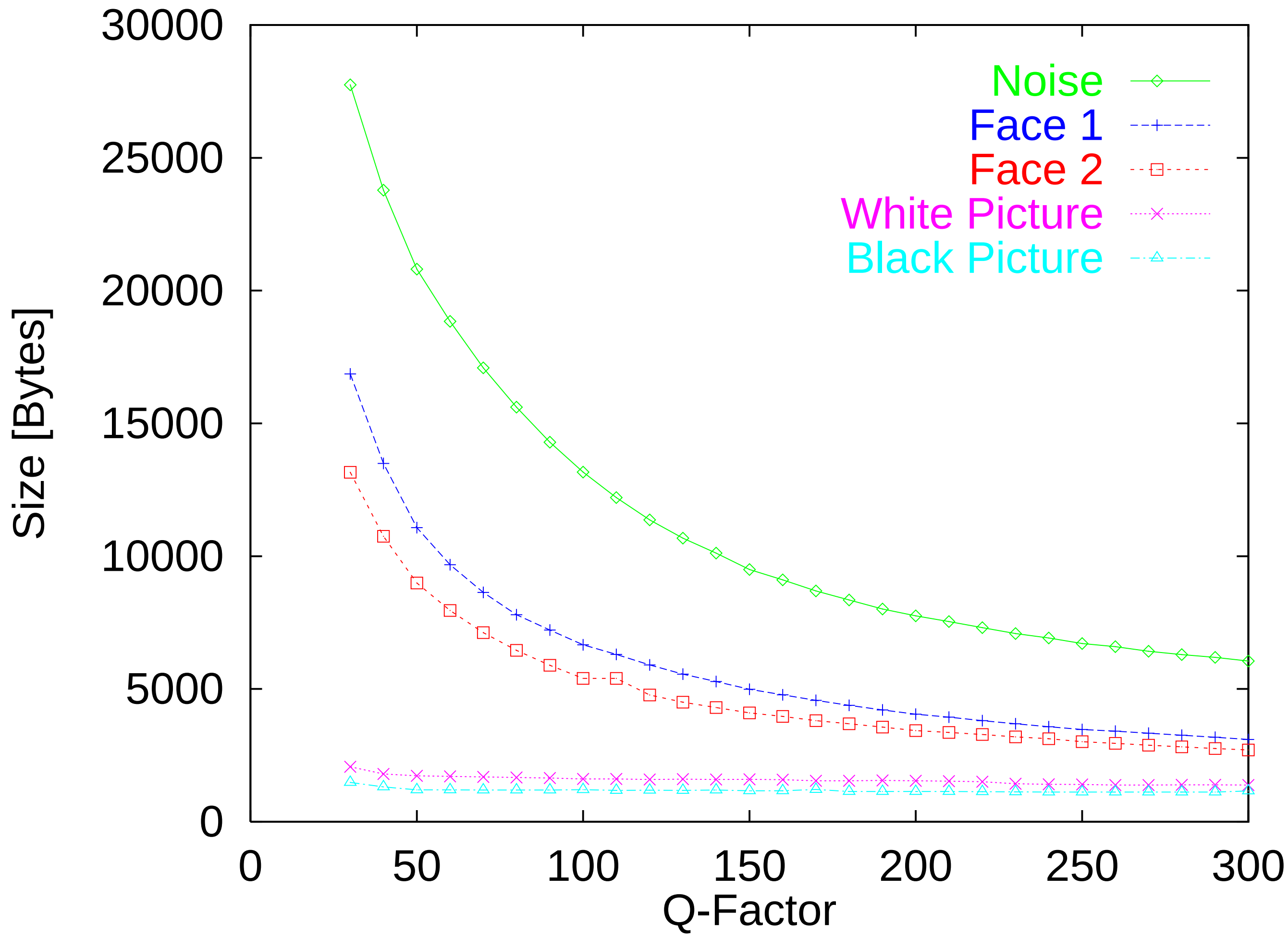
Source Adaptation

Multimedia applications can adjust their data rates:
Audio: encoding parameters (MPEG L3), encoding, sampling rate, mono/stereo

encoding	sampling rate	bit rate
LPC	8,000	5,600
GSM	8,000	13,200
DVI4	8,000	32,000
μ-law	8,000	64,000
DVI4	16,000	64,000
a range of DVI4 and MPEG L3		
L16 stereo	44,100	1,411,200

Adaptive applications

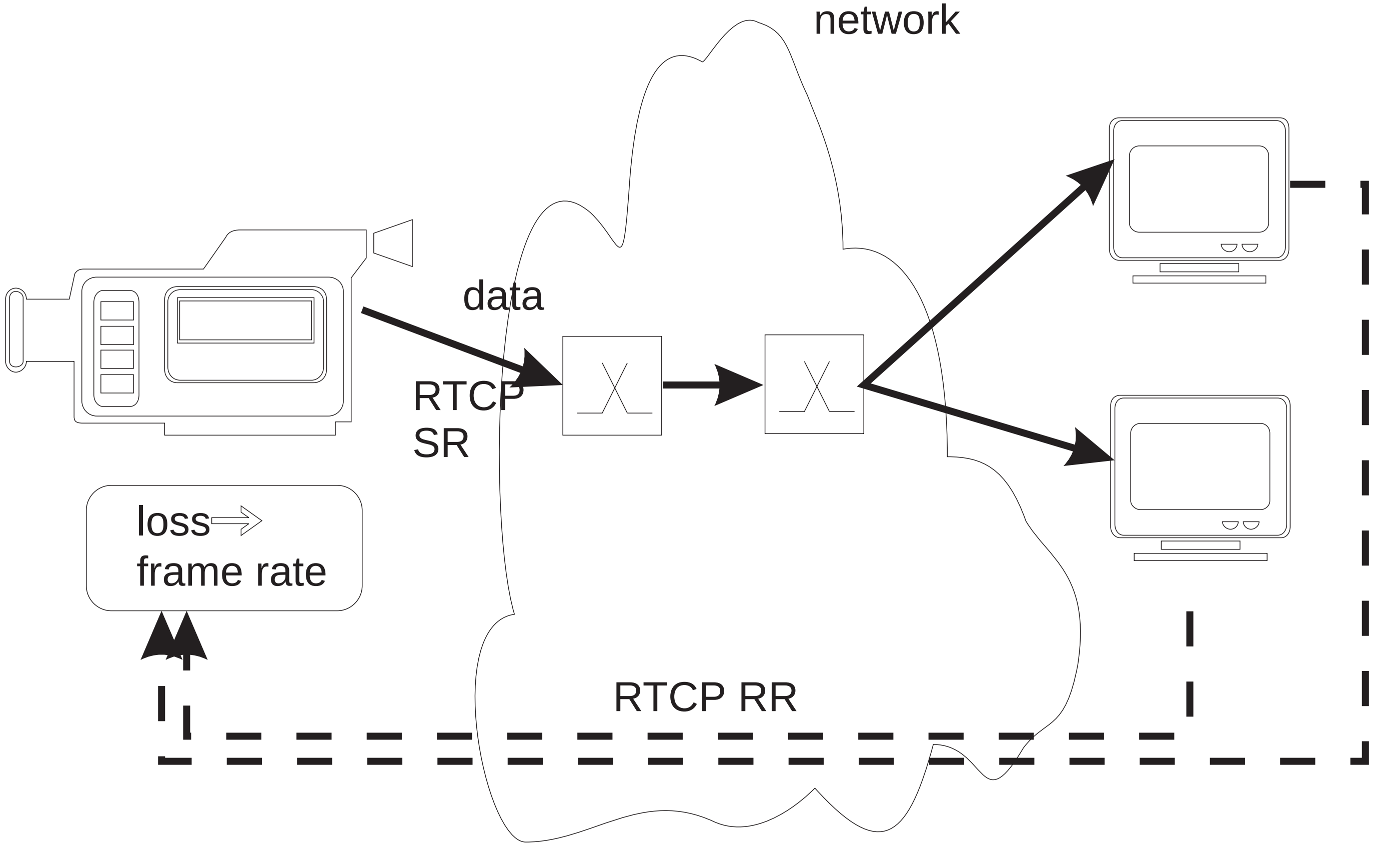
Video: frame rate, quantization, image resolution, encoding



Application control

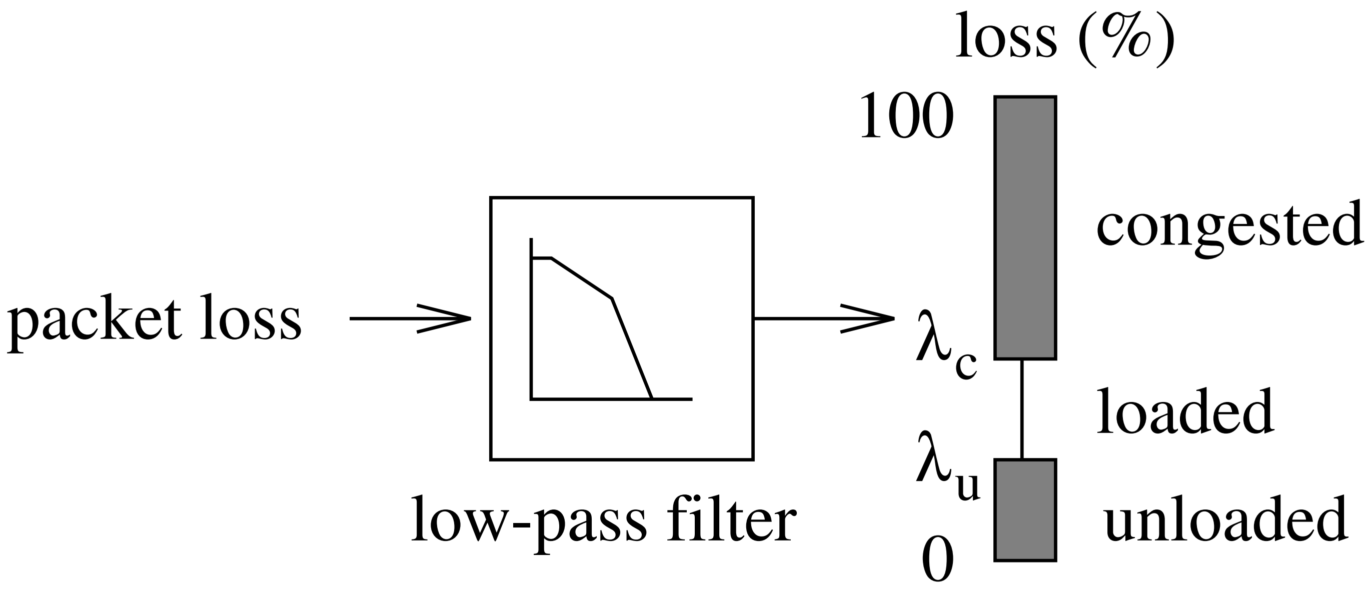
- networks with QoS guarantees:
  - QoS negotiation at call set-up time, network guarantees this quality
  - long call durations ➠ network load may change significantly
  - "wrong" initial allocation ➠ many rejected calls or low quality
- networks without QoS guarantees (= current Internet) or shared reserved link:
  - adapt application to available bandwidth
  - share bandwidth fairly with TCP?
  - lowest common demoninator ➠ combine with mixers, translators

End-to-End feedback control mechanism



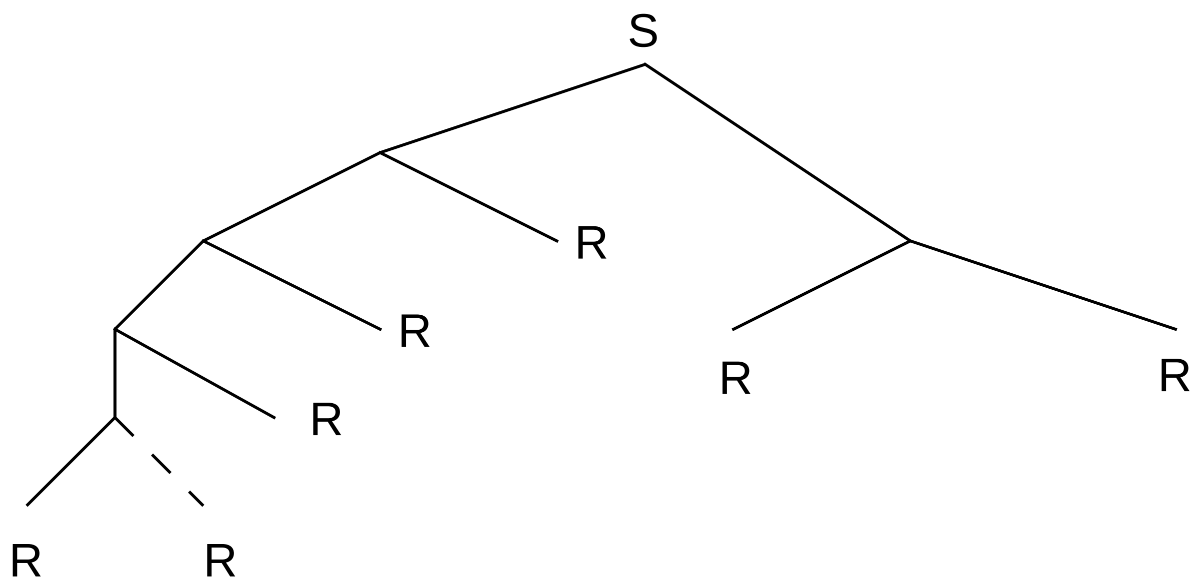
video application bandwidth is based on network feedback:
low losses ➠ slow bandwidth increase ➠ higher framerate
high losses ➠ bandwidth decrease ➠ lower framerate

Network state estimation and bandwidth adjustment



- loss information is filtered: $\lambda \leftarrow (1-\alpha)\lambda + \alpha b,\ \alpha \leq 1$
- linear regulator with deadzone
- multiplicative decrease if network is congested: $b_a \leftarrow \max\{b_a * \mu, b_{min}\},\ \mu < 1$
- additive increase if network is unloaded: $b_a \leftarrow \min\{b_a + \nu, b_{max}\}$

Multicast scalability



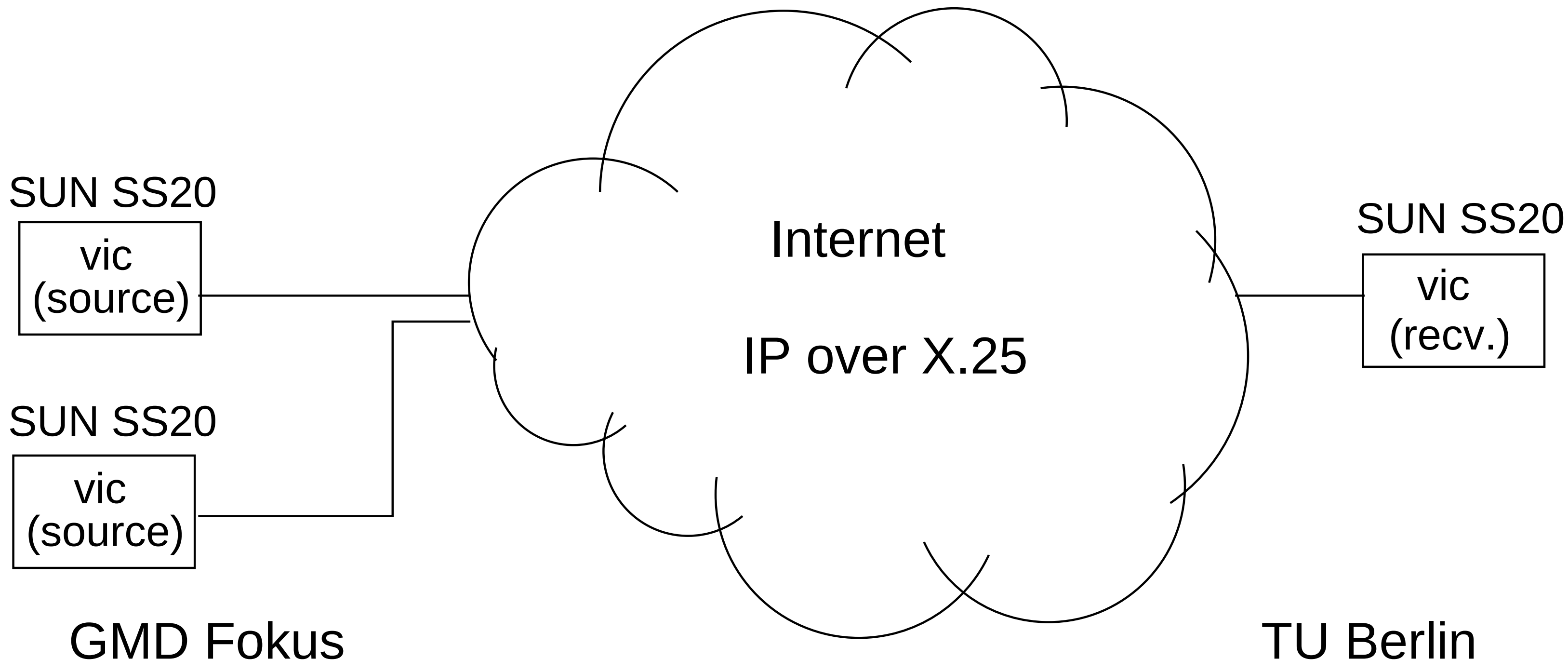
algorithm 1: adjust according to the worst-positioned receiver

algorithm 2: allow a fraction of the receivers to be congested

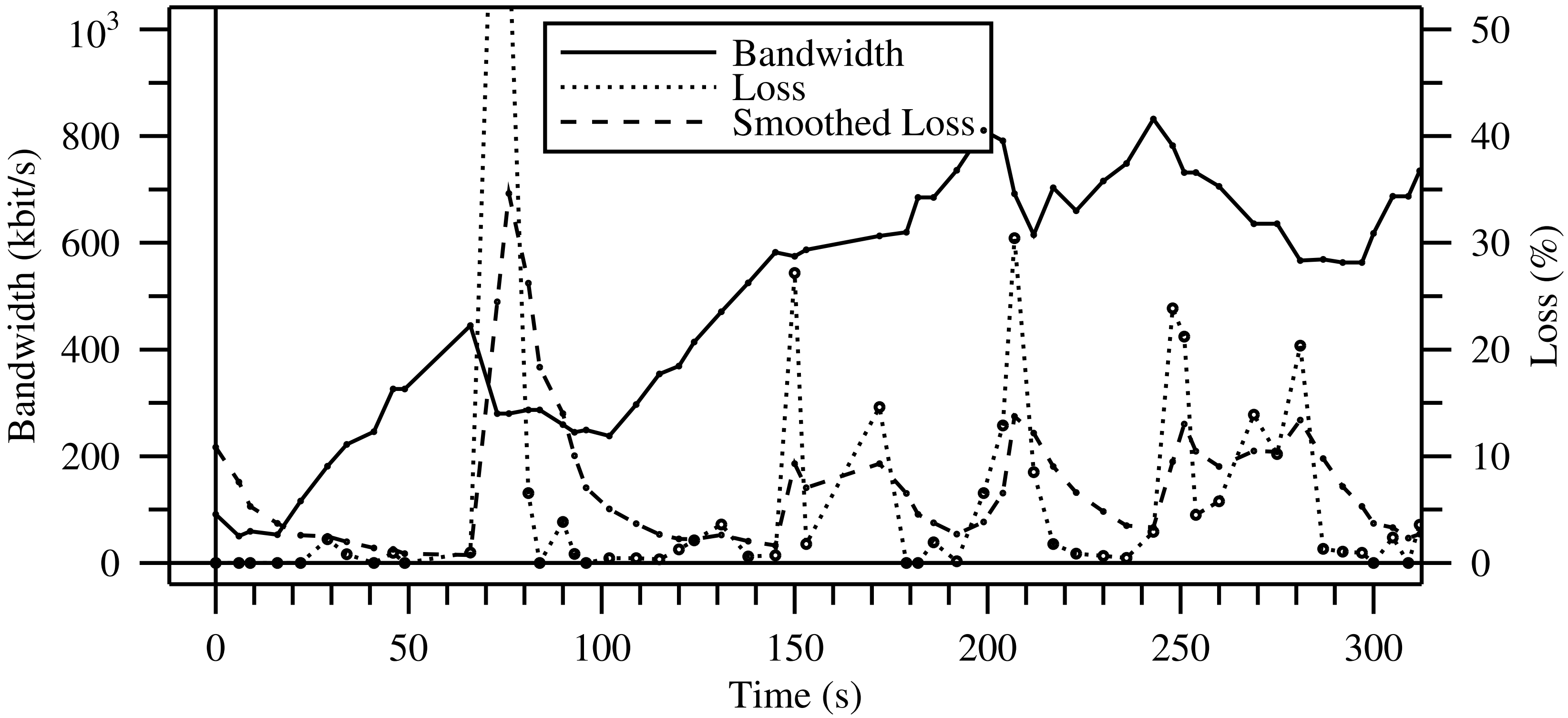
other solutions: video gateways, layered encodings

Internet scenario

- Measurements made on the 2 Mbit/s X.25 link between GMD Fokus and TU Berlin (5 hops distance)
- deadzone between 5% and 10%



Internet measurement



Observations

- jitter as loss predictor does not work well
- coexistence with controlled data applications
- use in ABR-like services?
- *no impact policy:* keep loss to within range observed without video application (monitor in transmission pauses)
- loss compensation can be dangerous ➠ ever higher loss fractions