Experiences with Interactive Video Using TFRC

Alvaro Saurin, Colin Perkins

University of Glasgow, Department of Computing Science

Ladan Gharai

University of Southern California, Information Sciences Institute

QuickTime™ and a
TIFF (Uncompressed) decompressor
are needed to see this picture.

Talk Outline

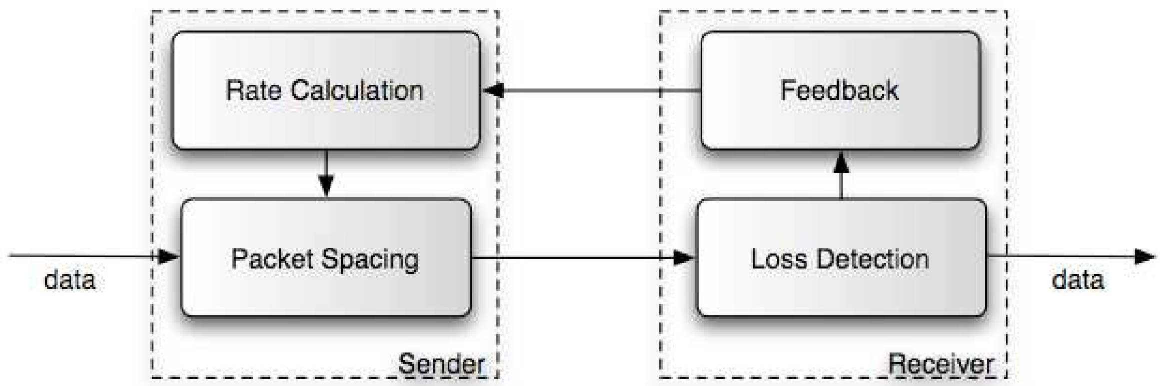
- Aims and objectives
- Implementation and performance of TFRC
- Implications for real-time video
  - Protocol issues
  - System design issues
  - Experimental results
- Open issues and implications for DCCP

Aims and Objectives

- Evaluate performance of interactive video conferencing systems running over congestion controlled transport
  - Implemented video conferencing tool
    - PAL/NTSC format video
    - Motion-JPEG compression ⇒ responsive, low compression delay
    - Typical data rate ~10s Mbps
  - User space implementation of TFRC, sending feedback within RTCP, data in modified RTP packets
    - draft-ietf-avt-tfrc-profile-05.txt
    - DCCP implementations not available when work started
    - Expect many results applicable to DCCP implementation, although a kernel implementation might have better timing characteristics
  - Experiments
    - Over Internet: Arlington, VA ↔ Glasgow ↔ Helsinki
    - Using local test bed (FreeBSD dummynet)

Implementation

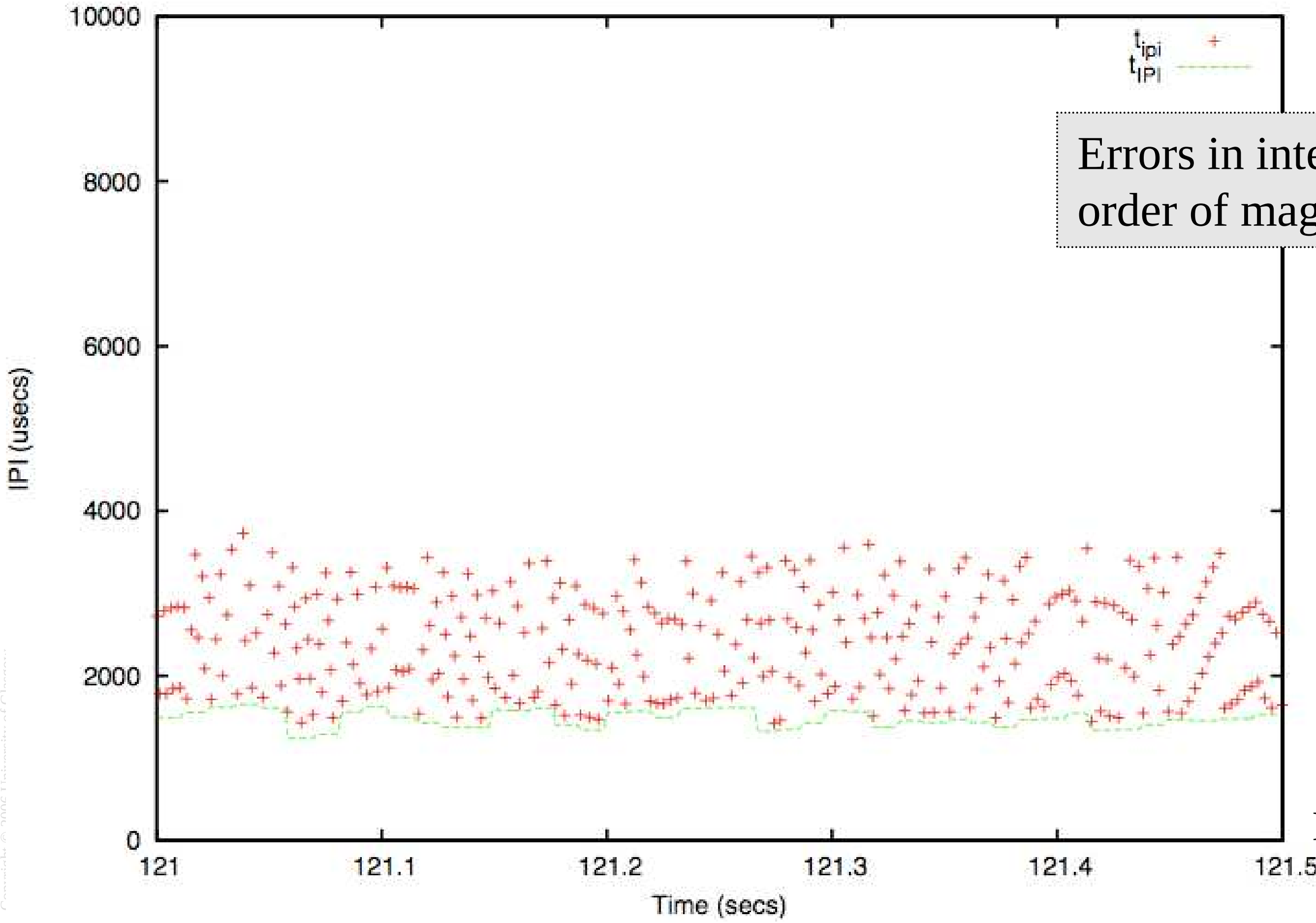
- TFRC implementation can be done at application level, part of existing RTP stack
- Four basic functions in feedback loop:



- Challenges:
  - Accurate packet spacing at sender
  - Timely feedback

Implementation: TFRC sender

- High performance video requires small inter-packet interval
- Difficult to accurately schedule packets
  - Due to inaccurate wakeup after sleep, thread scheduling issues



Errors in inter-packet spacing on same order of magnitude as the RTT

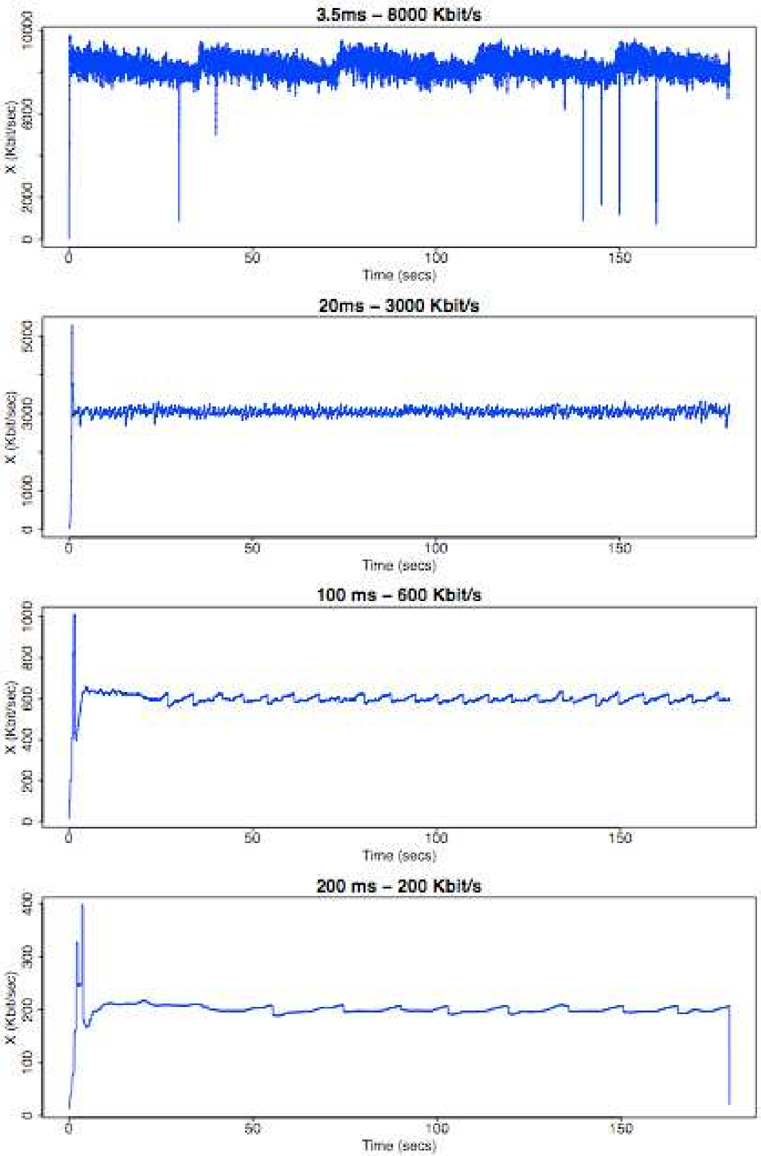
Example: 3.5ms RTT @ 8Mbps

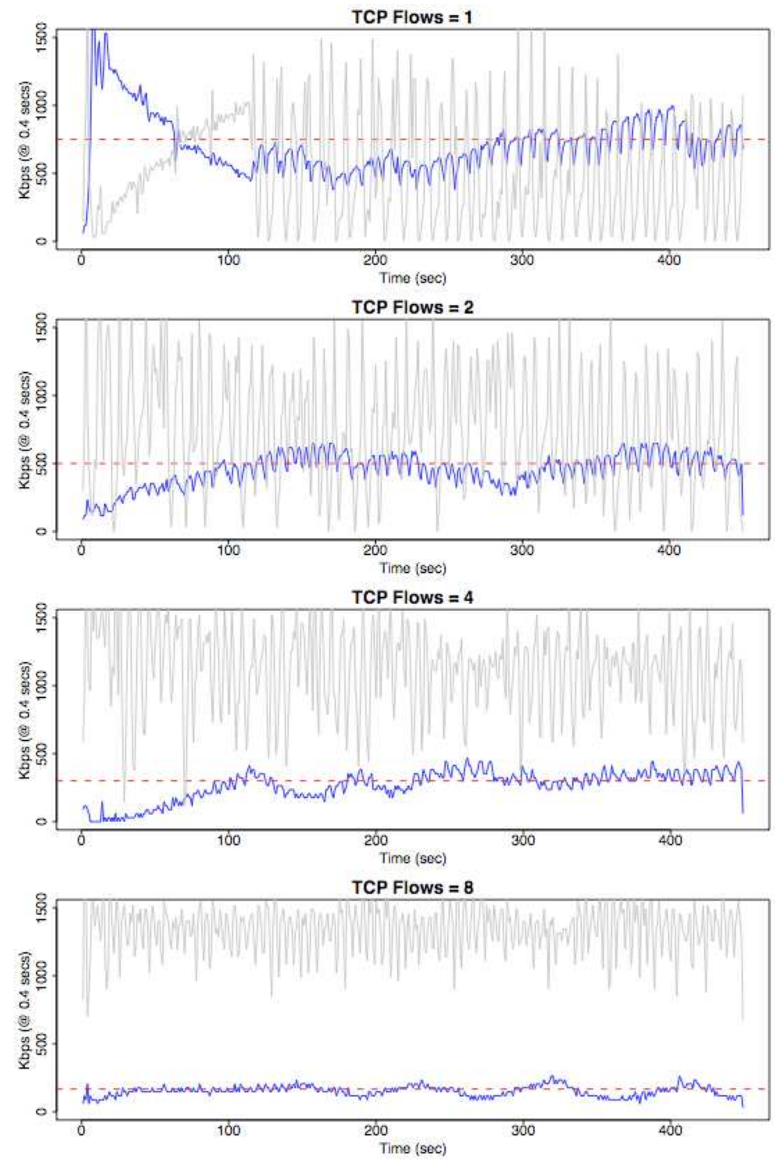
Implementation: TFRC receiver

- Similar issues with slow wakeup
  - System slow to schedule thread on expiry of feedback timer
  - 10ms wakeup latency not uncommon
  - Significantly delays feedback

- Timing inaccuracy in sender and receiver poses a *significant* challenge to stable TFRC implementation

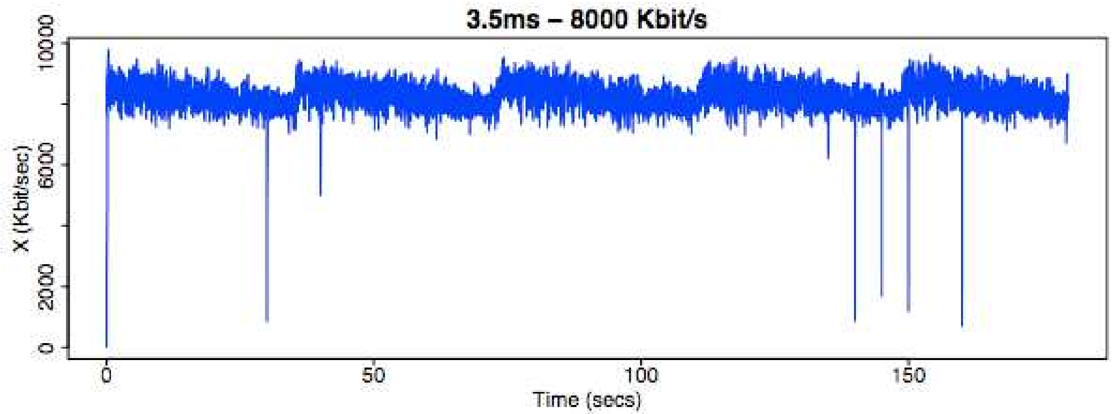
Experimental Performance: TFRC





Experimental Performance: TFRC

- Observe poor stability with short RTT:



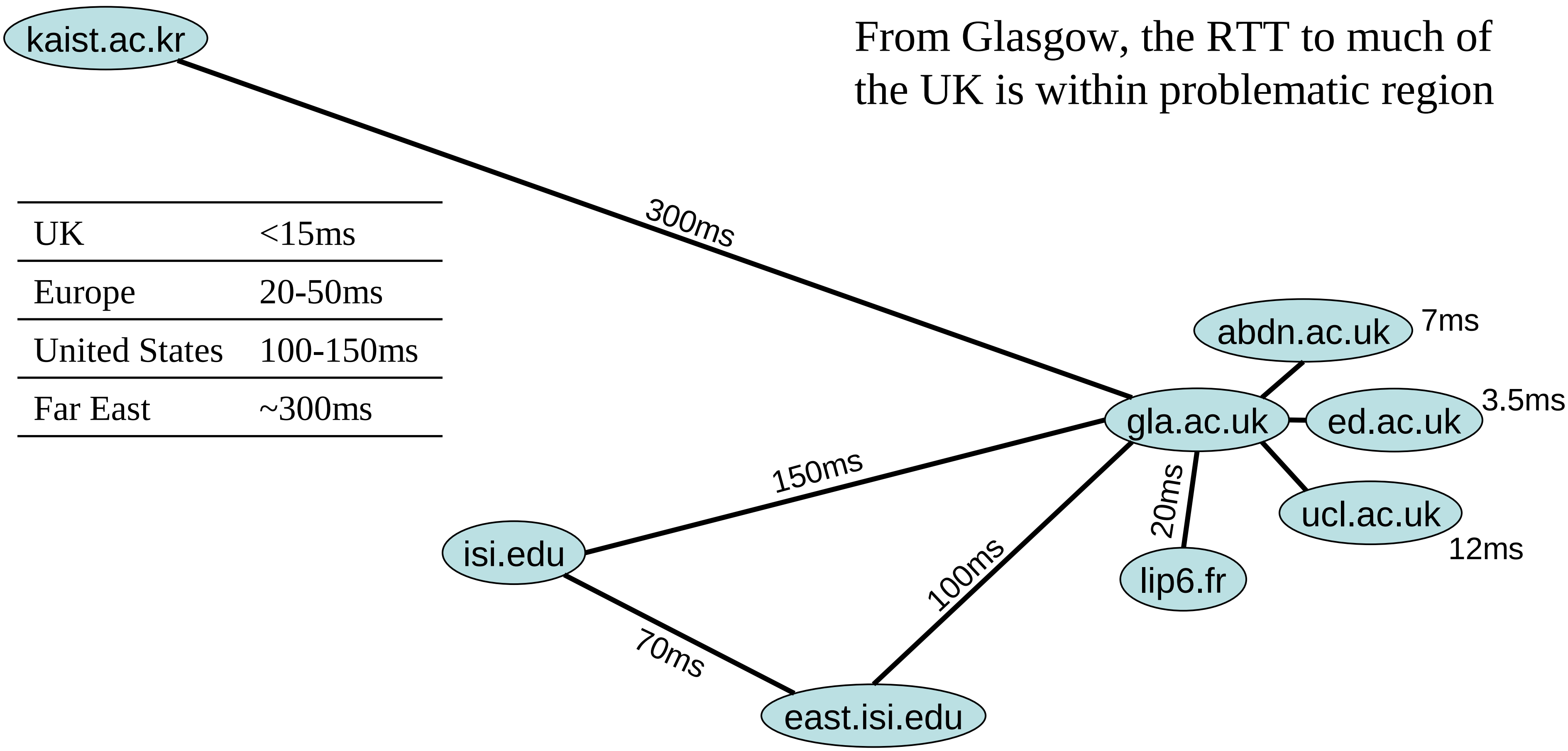
- Issues:
  - Bursty sending behaviour
    - Packets sent in bursts spaced around wakeup intervals
    - Degenerates into something similar to a window-based approach
    - May be simpler just to use a window based protocol?
  - Slow feedback
    - With 10ms wakeup latency and 3.5ms RTT, possible for feedback to be delayed >2RTT due to inaccuracies
    - Will force sender to halve sending rate

- Have found stability difficult to achieve with RTT < 10-20ms

Network Round Trip Times

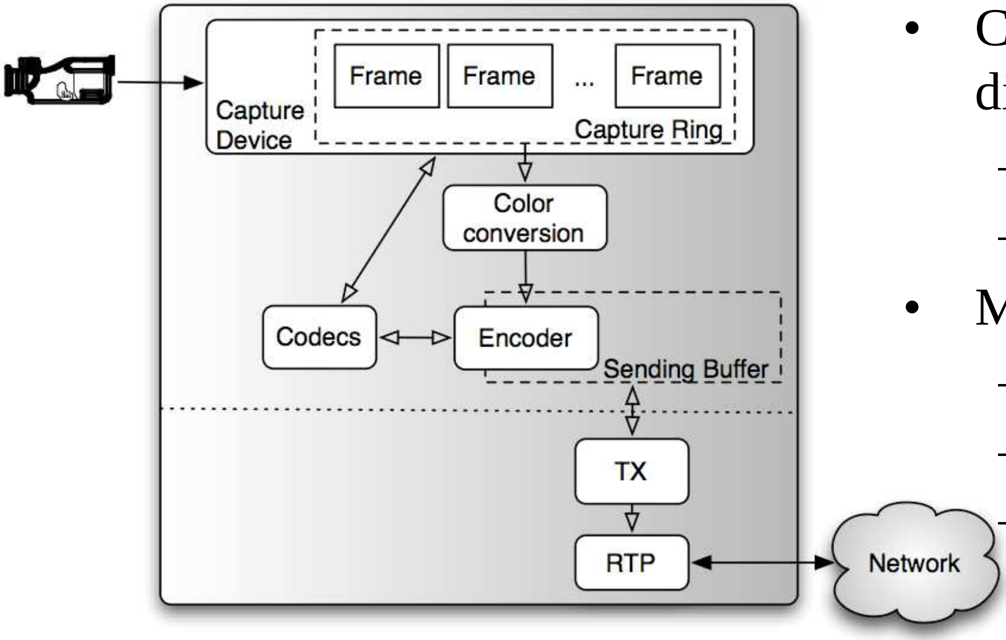
From Glasgow, the RTT to much of the UK is within problematic region

UK	<15ms
Europe	20-50ms
United States	100-150ms
Far East	~300ms



- Straight forward to add smoothing to protocol
  - Reduces responsiveness and fairness to TCP
  - Kernel implementation of TFRC likely more accurate timing ⇒ smoother

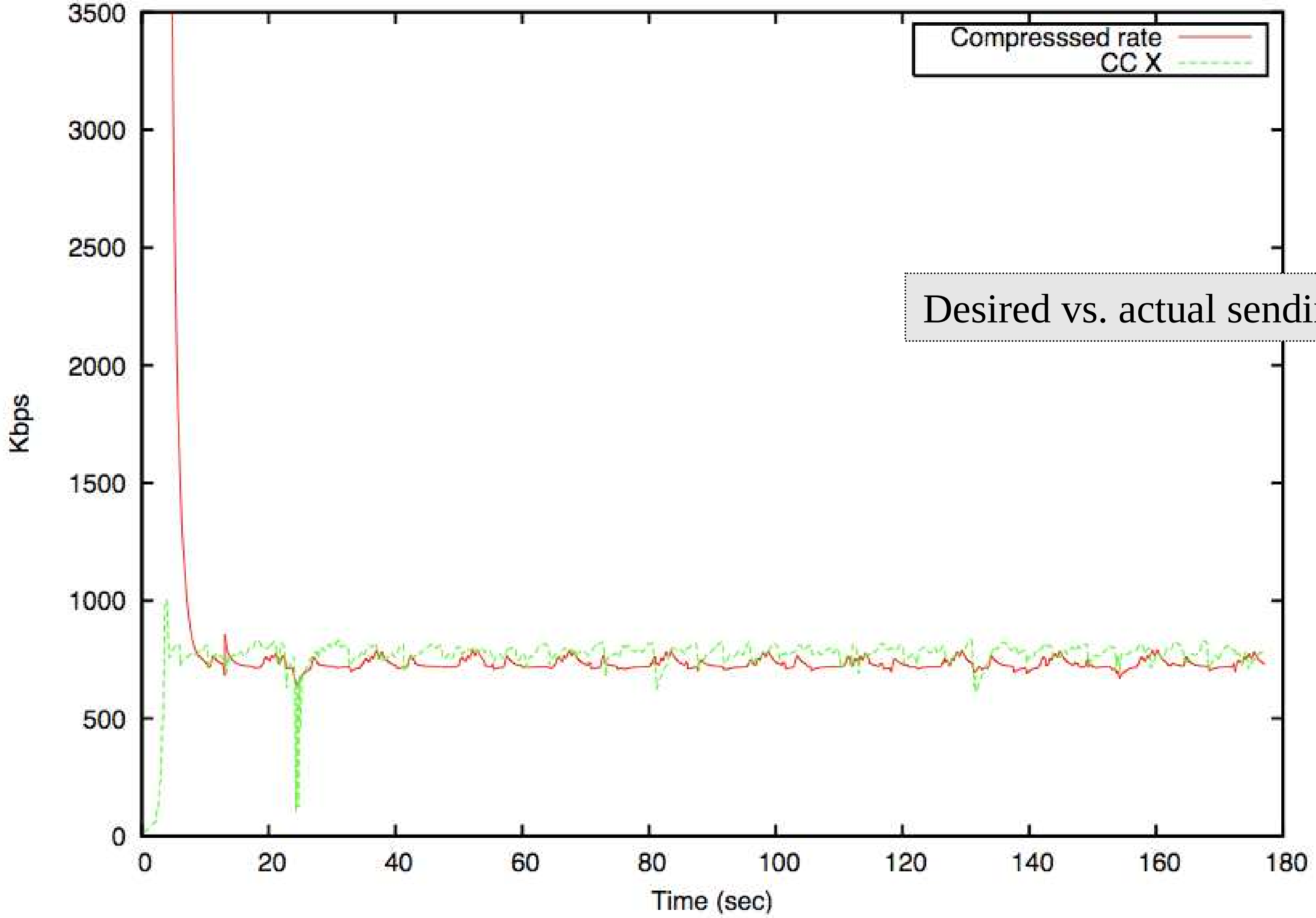
Implementation: Video Transmission



- Capture and transmission operate on different time scales
  - Slow bursts of arrivals from codec
  - Fast, smoothly paced, transmission
- Mismatched adaptation rates
  - TFRC $\Rightarrow$ O(round-trip time)
  - Codec $\Rightarrow$ O(inter-frame time)
  - Relies on buffering to align rates, varies codec rate

- Capture and encoding process causes timing problems:
  - Capture DMA operation can disrupt other bus accesses
  - Encoding uses significant amounts of processor time
    - M-JPEG currently, other codecs likely much worse
    - Linux general purpose scheduler barely adequate to get predictable thread scheduling in this environment; real-time scheduler difficult to tune/debug
- Sender dynamics difficult to tune and debug

Experimental Performance: Video

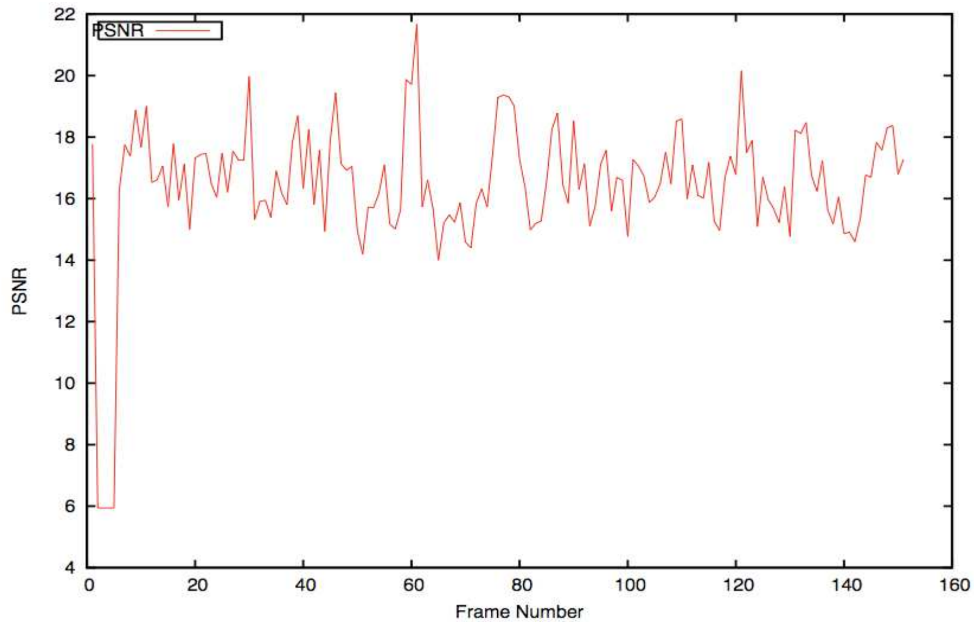


Desired vs. actual sending rate

100ms RTT, 800kbps bottleneck, 10 fps M-JPEG
Testing in dummynet

Best case: RTT and frame rate match

Experimental Performance: Video



- Poor man's video quality metric:
  - Peak Signal to Noise Ratio (PSNR)
  - Significant variation in quality over session lifetime
    - Changes in input source requires a variable output rate
    - Constrained to be smooth by TFRC ⇒ quality varies instead

- Also see packet losses due to rate limit at sending buffer
  - Could be solved by faster codec adaptation
  - But: requires codec that can change compression ratio *within* a frame
    - Effect on quality unclear; implementation challenge

Issues: Slow Start

- Slow start requires an application to send at a low initial rate, increasing exponentially each round-trip time where no loss is reported
  - Duration of slow start period depends on network conditions; unpredictable

- Video codec must be capable of such a rapid increase in sending rate whilst maintaining reasonable picture quality
  - Requires a highly scalable codec, capable of varying compression ratio on the order of network RTT
    - i.e. while coding a frame, since RTT likely doesn't match frame rate
    - Not clear this is feasible
  - Current implementation generates dummy data instead
    - Seems wasteful, but can cover call setup delay

Issues: Steady State

- Application required to send at a roughly constant rate, based on average loss rate observed
  - Transmission rate narrowly bounded
    - Large bursts above the prescribed rate must be avoided due to insufficient capacity; less aggressive senders will be “beaten down” by TCP traffic as consequence of the TFRC algorithm
    - Imposes constraints on when a codec can change its rate
    - Given sufficient buffering, and use of dummy data, is possible to meet rate constraints; not clear feasible for interactive systems
  - Difficult to accurately match transmission rate
    - Requires codec that can change rate on O(RTT) timescale
      - High frame rate; or codec that can vary compression within a frame
    - Requires accurate feedback timing
    - Problems with short RTT

Conclusions

- Initial experiments raise more questions than they answer
  - Likely possible to run video over TFRC, with more sophisticated codecs
    - Impact on perceptual quality of implied quality variation unclear
    - Likely easier as video quality, frame-rate and network bandwidth increase
  - Slow start very problematic
    - Codecs don't adapt in an appropriate way
  - Given difficulty in matching rate, and resulting bursty behaviour, not clear that window based congestion control wouldn't be more appropriate
    - To what extent is sending dummy data appropriate?

- DCCP a good base for experimentation
  - Not clear we understand problem sufficiently to give production quality advice on implementation of congestion controlled interactive video on TFRC