

Impact of IW10 on Interactive Real-Time Communication

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- Any burst of back-to-back packets tends to create transient queue
 - Initial window is significant contributor of back-to-back bursts
 - IW effect tends to intensify when large number of parallel connections is used (e.g., Web traffic)
- In a common case bottleneck resides in the access network
 - Quite often capacity of the access network is not higher than few Mbps

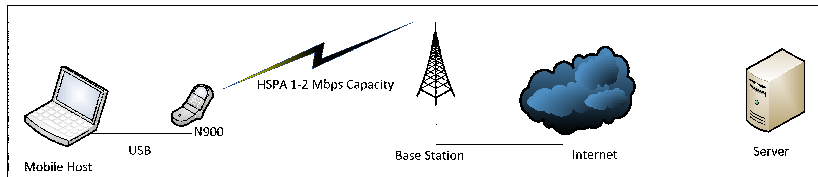


Figure: Test Environment

- 16kbps CBR flow, 20ms sending interval
- 1-6 short TCP flows with total size of 372kB, start 10..12 seconds after CBR
- TCP variant has SACK, delayed ACK and limited transmit enabled.

- A "pure loss" when a packet does not reach the receiver.
- A "delayed loss" when a packet is delayed more than a codec jitter buffer can handle.
- The "base delay" is the minimum one-way delay of the CBR packets sent n seconds before the start of the observation period. The default value of n is taken as 2 seconds and it is a configurable parameter.
- "Jitter buffer" is the delay one waits before playing. This is also a configurable parameter (e.g., 40ms, 60ms, 80ms, 100ms)
- If jitter exceeds the jitter buffer, the packet is considered to be lost ("delayed loss").

- The metric categorizes the quality of audio based on loss periods (number of consecutive losses, either pure or delayed loss).
- Extract the loss periods and provide the overall distribution of packets in such loss periods (e.g., how many loss periods are ≥ 40 ms, ≥ 80 ms, etc).
- Loss period mapped to non-linear scale (0-5)
 - 0 = packet was not lost
 - 1 = Only single loss (20msec lost), no adjacent packets were lost
 - 2 = This packet is part of 2-3 pkts consecutive loss period (40-60ms lost)
 - 3 = This packet is part of 4-5 pkts consecutive loss period (80-100ms lost)
 - 4 = This packet is part of 6-9 pkts consecutive loss period (120-180ms lost)
 - 5 = This packet is part of 10 or more pkts consecutive loss burst (200ms+ lost)

CBR Quality with 40ms Jitter Buffer, 50 replications

Quality metric for Audio+1 short TCP flow with jitter buffer of 40 ms

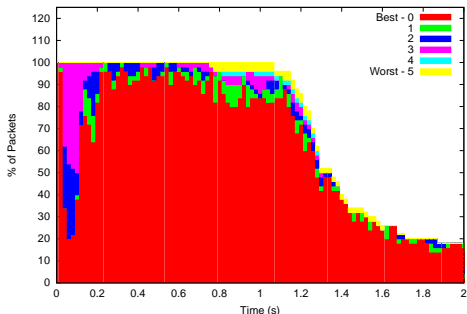


Figure: Audio + 1 short TCP flow, IW3

Quality metric for Audio+1 short TCP flow with jitter buffer of 40 ms

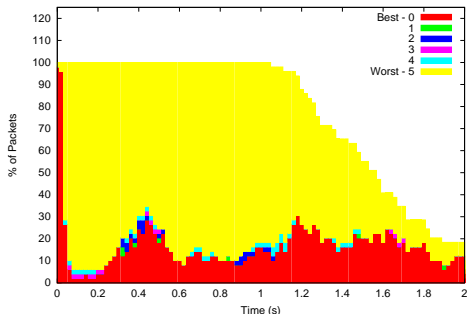


Figure: Audio + 1 short TCP flow, IW10

CBR Quality with 40ms Jitter Buffer, 50 replications

Quality metric for Audio+2 short TCP flows with jitter buffer of 40 ms

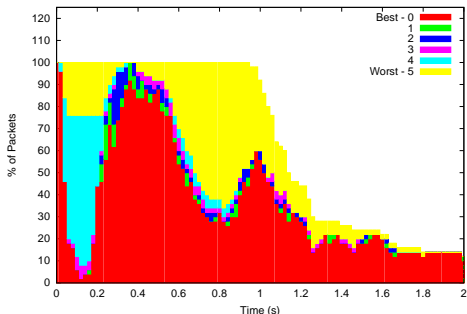


Figure: Audio + 2 short TCP flows, IW3

Quality metric for Audio+2 short TCP flows with jitter buffer of 40 ms

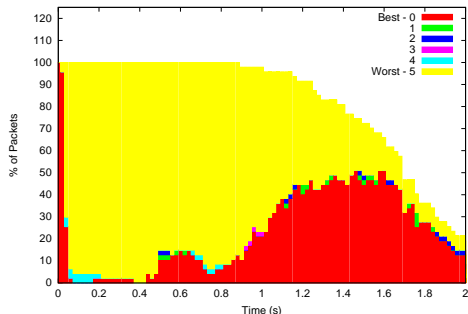


Figure: Audio + 2 short TCP flows, IW10

CBR Quality with 40ms Jitter Buffer, 50 replications

Quality metric for Audio+6 short TCP flows with jitter buffer of 40 ms

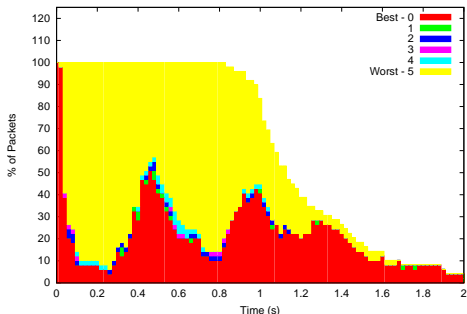


Figure: Audio + 6 short TCP flows,
IW3

Quality metric for Audio+6 short TCP flows with jitter buffer of 40 ms

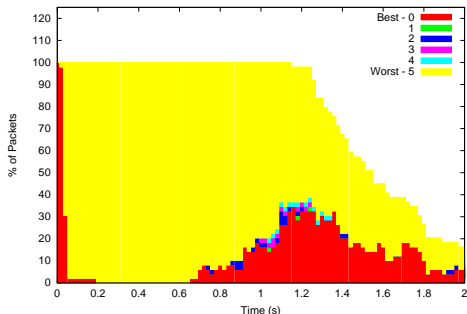


Figure: Audio + 6 short TCP flows,
IW10

CBR Quality with 100ms Jitter Buffer, 50 replications

Quality metric for Audio+1 short TCP flow with jitter buffer of 100 ms

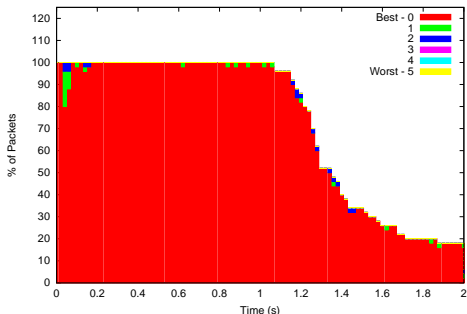


Figure: Audio + 1 short TCP flow, IW3

Quality metric for Audio+1 short TCP flow with jitter buffer of 100 ms

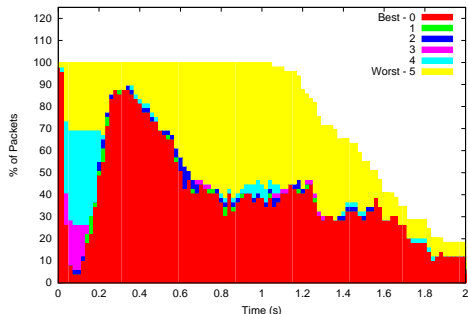


Figure: Audio + 1 short TCP flow, IW10

CBR Quality with 100ms Jitter Buffer, 50 replications

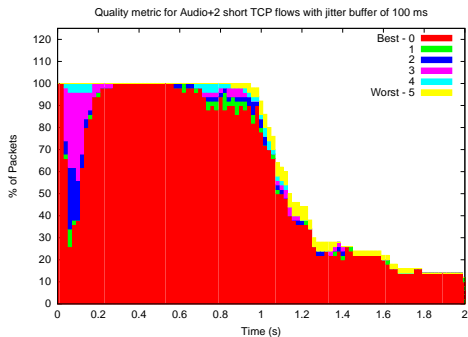


Figure: Audio + 2 short TCP flows, IW3

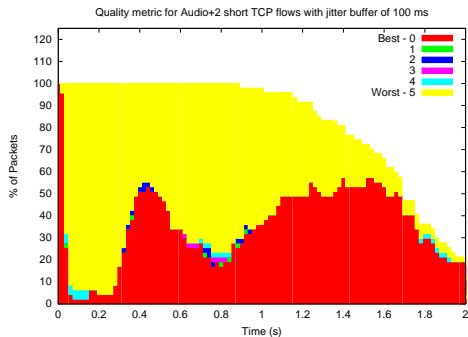


Figure: Audio + 2 short TCP flows, IW10

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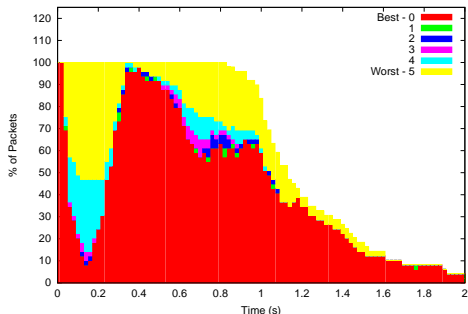


Figure: Audio + 6 short TCP flows,
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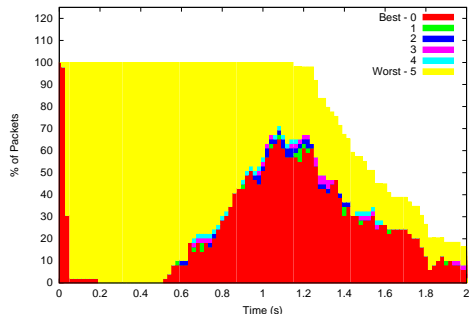


Figure: Audio + 6 short TCP flows,
IW10

Good Quality Level with Different Sized Jitter Buffers

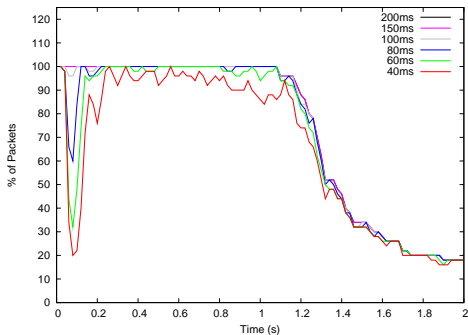


Figure: Audio + 1 short TCP flow, IW3

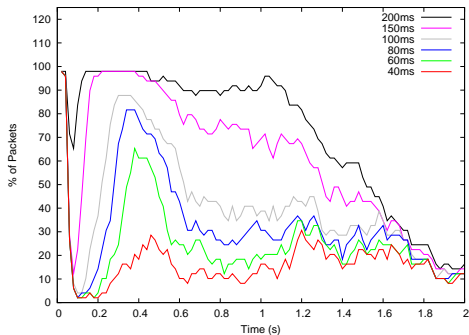


Figure: Audio + 1 short TCP flow, IW10

- Good quality here combines levels 0 (not lost) and 1 (no adjacent loss)

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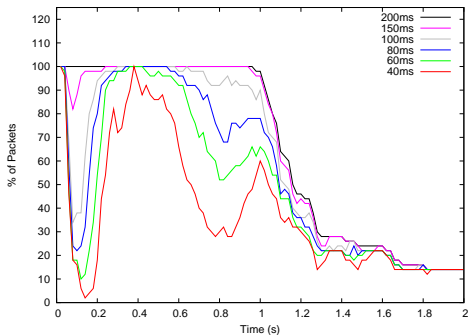


Figure: Audio + 2 short TCP flows, IW3

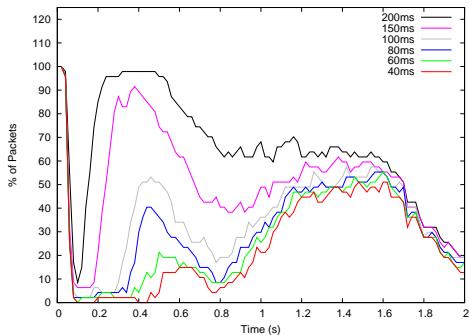


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Good Quality Level with Different Sized Jitter Buffers

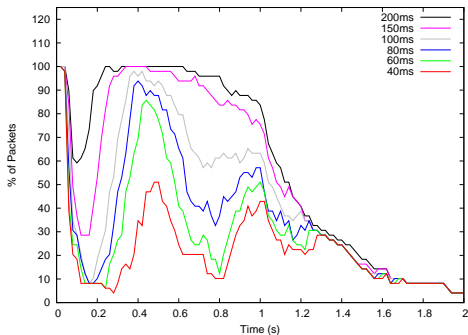


Figure: Audio + 6 short TCP flows, IW3

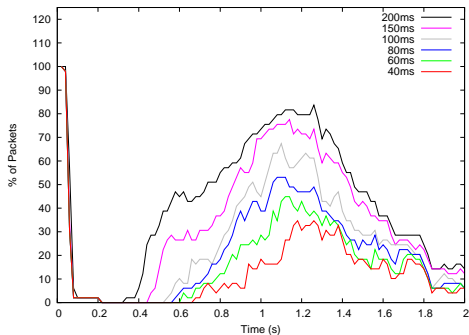
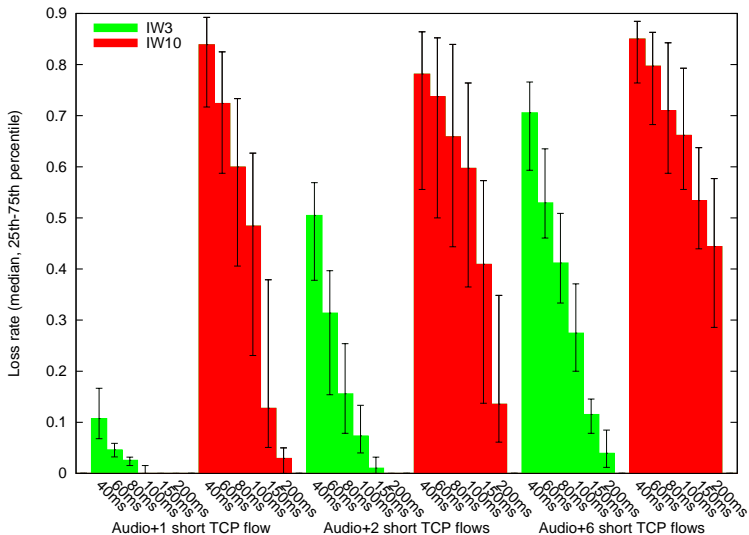


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Loss Rate with Different Jitter Buffer Sizes



- AQM is not (easy) solution
 - Off-the-shelf AQM solutions are too slow to react in time to help interactive communications, face deployment challenges if default parametrization is not good enough [Harsh RED: Improving RED for Limited Aggregate Traffic, AINA 2012]
 - An AQM designed specially slow start in mind?
- LEDBAT is not a solution
 - Web browsers and Web pages try to minimize page transfer times
 - Unlikely that LEDBAT (or like) would be enabled “by default”
 - Web transfers are not less than best-effort transfers by any means
 - The sender needs to apply LEDBAT
 - On-demand approach requires end-to-end signalling, challenging to deploy

- IW10 clearly more harmful to interactive flows than IW3 due to jitter triggered packet discarding
- Large number flows also with IW3 is bad

See also: Impact of TCP on Interactive Real-Time Communication, IAB CC workshop
[http://www.tschofenig.priv.at/cc-workshop/irtf_iab-ccirtcpaper9.pdf]