Drawing the line between transport and network requirements

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IETF 86 ICCRG
6-Nov-2012

Goals of this presentation

- Ask some unsolved questions
- Stimulate thought
- Collect pointers to existing work
- Identify potential contributors
- Encourage interest in the revitalized IPPM WG

- Non-Goal: answering the questions at this point

IPPM Model Based Metrics

- Active IPPM Draft
  - draft-mathis-ippm-model-based-metrics-01.txt
  - Matt Mathis & Al Morton
  - A tractable way to characterize bulk performance
- It will implicitly divide responsibility
  - Bursts (e.g. IW10 and TSO)
    - TCP should send fewer or
    - The network should tolerate more
  - Reordering
    - TCP should tolerate more
    - Then network should cause fewer
  - Etc

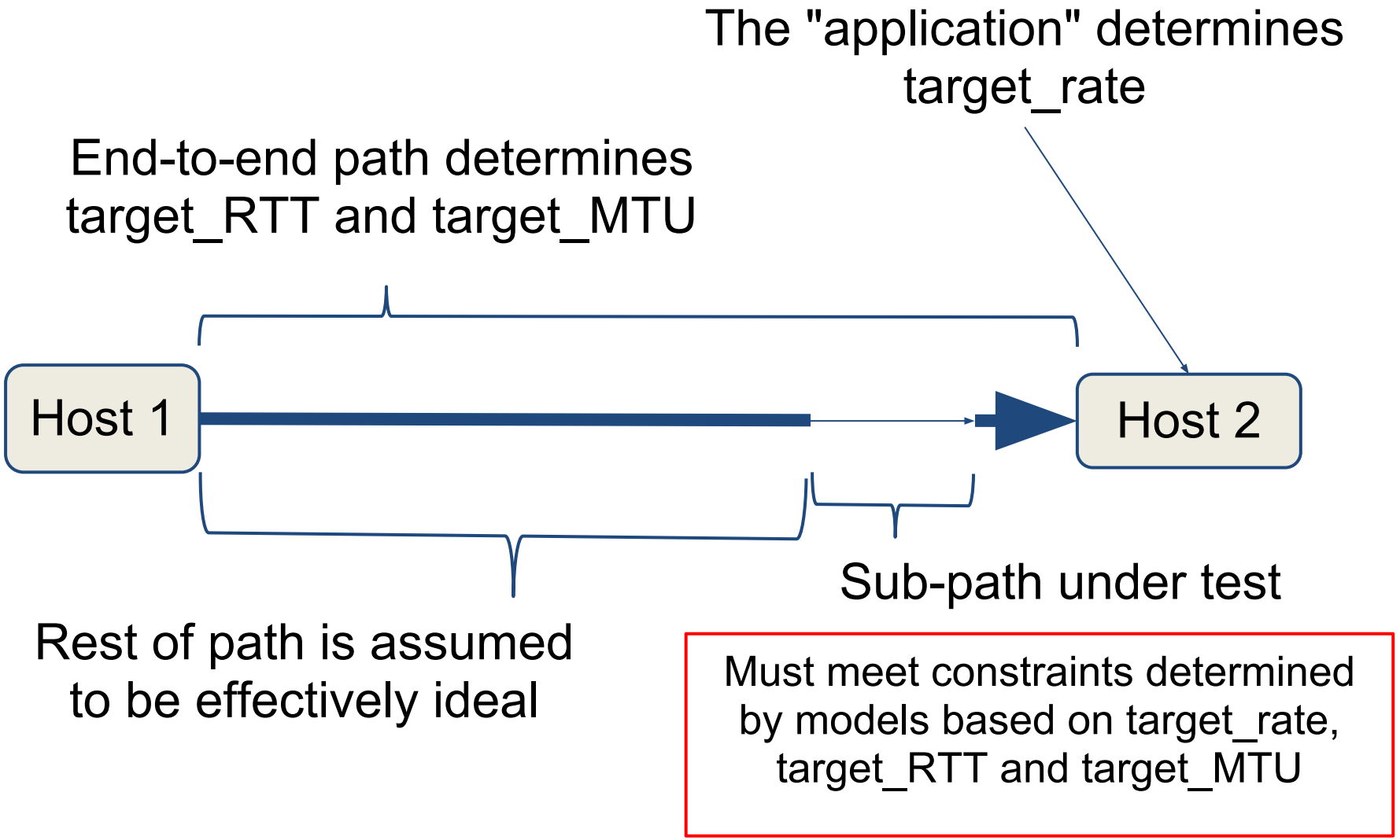
Bulk Transport Capacity is hard for a reason

- TCP and all transports are complicated control systems
  - TCP causes self inflicted congestion
  - Governed by equilibrium behavior
  - Changes in one parameter are offset by others
- Every component affects performance
  - All sections of the path
  - End systems & middle boxes (TCP quality)
  - Routing anomalies and path length
- The Meta-Heisenberg problem
  - TCP "stiffness" depends on RTT
  - The effects of "shared congestion" depend on
    - Bottlenecks and RTT of the other cross traffic
  - Can't generally measure cross traffic with 1 stream

Model Based Metrics: A better way to do BTC

- Open Loop TCP congestion control
  - Prevent self inflicted congestion
  - Prevent circular dependencies between parameters
    - Data rate, loss rate, RTT
- Independently control traffic patterns
  - Defeat congestion control (generally slow down)
  - Mimic all typical TCP traffic (bursts, etc)
- Measure path properties section by section
  - Mostly losses
  - Compare to properties required per models
  - E2E path passes only if all sections pass all tests

The pieces (simplified)



A common context for all examples

- Target parameters:
  - 1 MByte/s bulk data over a path that is
  - 10 Mb/s raw capacity (~1.2 MByte/s)
    - More than the target!
  - 20 ms, 1500 Byte MTU, 64 byte headers
- Compute from Macroscopic Model
  - target_pipe_size
    - target_rate*target_RTT / (target_MTU-header_overhead)
    - 14 packets
  - reference_target_run_length (= 1/p)
    - (3/2)(target_pipe_size^2)
    - 274 packets
    - Same as p < 0.365%

A common context for all examples

- Target parameters:
  - 1 MByte/s bulk data over a path that is
  - 10 Mb/s raw capacity (~1.2 MByte/s)
  - 20 ms, 1500 Byte MTU, 64 byte headers
- Compute two additional (new) parameters:
  - Headway at target rate
    - target_headway = target_MTU*8/target_rate
    - target_headway = 1.5 mS
  - Headway at bottleneck rate
    - bottlenenck_headway = target_MTU*8/effective_rate
    - bottlenenck_headway = 1.2 mS

1) Baseline (CBR) performance test

- Measures basic data and loss rates
- Send one 1500 byte packet every 1.5 mS
  - 1 MByte/s target rate
  - Losses MUST be more than 274 packets apart
    - Otherwise "standard" Reno TCP can't fill the link
- Note that this is pass/fail
- Cool properties
  - Does not depend on sub-path RTT
  - Does not depend on measurement vantage
    - As long as rest of path is good enough
  - Run length bounds loss rate for entire path

Derating

- To some extent the model is subjective
  - And too conservative
  - What if TCP isn't standard Reno?
- Must permit some flexibility in the details
  - As TCP evolves
  - As the network evolves
  - The ID permits "derating"
- Actual test parameters must be documented
  - and justified relative to the targets
  - and proven to be sufficient
    - Meet the target goal over a derated network
- (ID will have) text about calibration and testing

2) Slowstart style burst test

- Mimic last RTT of a conventional TCP slowstart
  - Measure queue properties at the "constrained link"
- Send 4 packets every 2*bottleneck_headway (2.4 mS)
  - Builds a queue at bottleneck
  - Burst of 2*target_pipe_size (28 packets)
    - Peak queue will be target_pipe_size (14 packets)
    - (Test inconclusive if ACK are too early ->no queue)
  - Repeated bursts on 2*target_RTT headway
    - Below 14 packets, MUST meet target_run_lenght
    - Beyond 14 packets MAY derate
    - Beyond 28 packets (more?) loss rate SHOULD rise
      - To prevent excess queueing (bufferbloat)
      - THEORY or MODELS NEEDED

3a) Interface rate bursts caused by the server

- Full rate (e.g. 10 Gb/s) bursts from a server/tester
  - Note that these mostly stress the "front path"
    - Server up to the primary bottleneck
  - Typically not the same queue as the SS tests
    - Smaller bursts
    - Higher rate
- Caused by various application effects
  - 3 Packets: normal window increases, all states
  - 10 Packets: IW10
  - 44 Packets: TSO (only if cwnd is large enough)
  - Application or scheduler stalls
    - Any fraction of 2*target_pipe_size possible
    - Statistics scale: (target_rate) * (sched_quanta)

3b) Interface rate bursts caused elsewhere

- TCP sender reflects ACK bursts into the data
- Caused by:
  - ACK compression due to other traffic
  - Thinning/merging ACKs (network or receiver)
  - Compression due to channel allocation
    - E.g. Half duplex
  - Reordering etc of cumulative ACKs

- Clearly if the network caused the problem
  - TCP isn't likely to fix it
  - Even if it was a different network section

What burst tolerance should be ok?

- General pattern:
  - No runlength derating for small burst sizes
  - Progressively more RL derating at larger burst sizes
  - It is a tradeoff between TCP and the network
    - Small bursts must be tolerated by the network
    - Network must tolerate network induced bursts
    - TCP should not cause large bursts
- NEED A MODEL
- Quick answer
  - We have been underestimating the impact of TSO

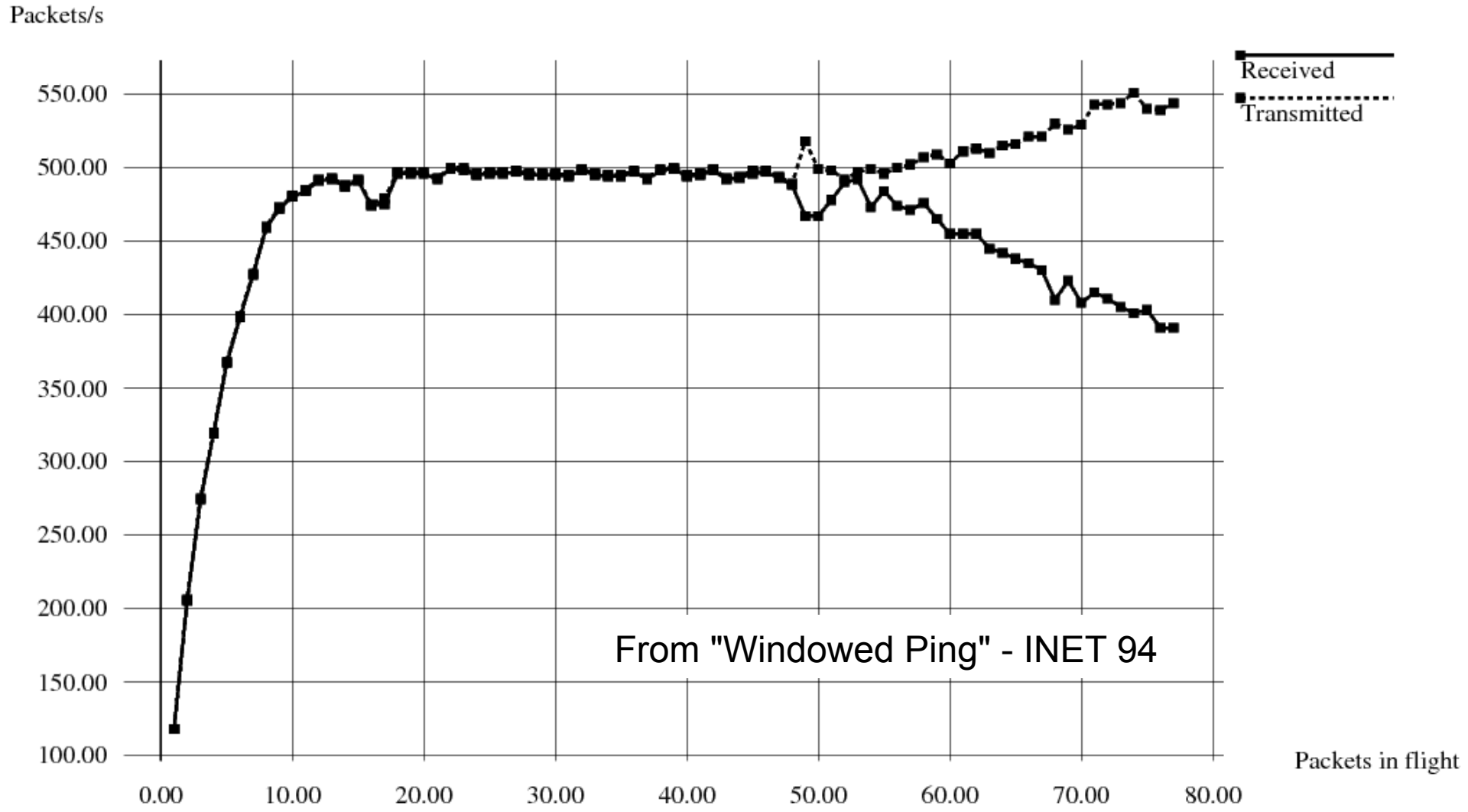
4) Tolerance to reordering

- OPEN QUESTION My specualtion
- Strict sequential switching costs Internet scale
  - Forced sequential processing
  - Less concurrency within chassis
  - No ECMP routing, even at the fabric level
  - Extra interlocks, controls or hashes
  - Mostly motivated by non-TCP applications
  - But TCP has it's limitations too
- How would it change net if reordering was common?
  - What is the opportunity cost of the current state?

5) Standing queue test

- Run approximately fixed window transport
- Gradually increment window
- Collect statistics on the onset of loss

Queuing example



Standing queue test

- Collect statistics on first loss
- Must not be before target_run_length
  - Otherwise TCP will not fill the link
- Must not happen at too large of queue
  - Direct measure of bufferbloat
  - How big is too big?
  - NO MODEL or THEORY

Open Questions

- During SS, how large of queue is too big?
  - Should there be an upper bound on the queue size?
- How large server rate bursts should:
  - the network tolerate?
  - TCP avoid?
- How much reordering should be ok?
- How long/much standing queue is ok?
  - Should there be an upper bound on triggering AQM?

- The end