Model Based Metrics

IPPM Working Group IETF 90,

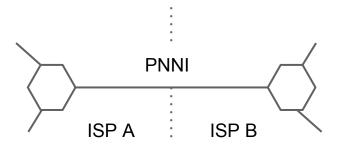
Matt Mathis <mattmathis@google.com> Al Morton <acmorton@att.com>



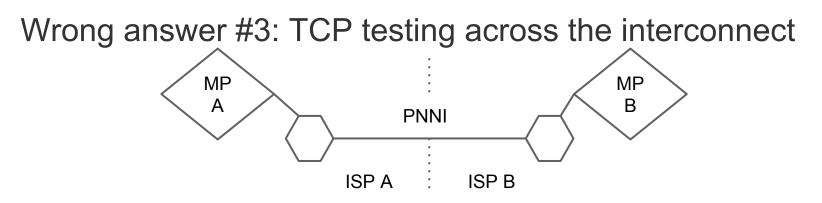
Outline

- Usage example (motivation)
- Next steps

Problem: Capacity planning at interconnects

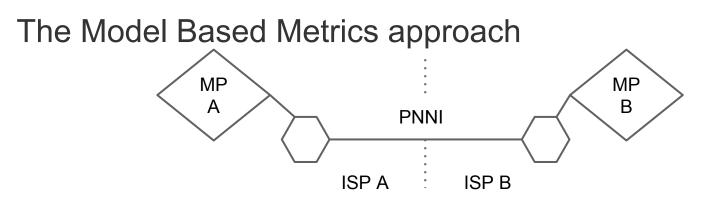


- Want to understand the performance of a busy interconnect
 - Does it have sufficient headroom?
 - Is there a problem with the link?
- Vague/experience based approaches
 - Use a rule of thumb for maximum utilization (e.g. 80%)
 - Theory comes from "effective bottleneck capacity"
 - Useless for non-load related problems
 - \circ Look for "flat tops" in the traffic data
 - But for 5 minutes averages, this is way too late



- Easy measurement
 - MP to MP TCP tests: netperf, iperf, nettest, etc
 - RTT is tiny, TCP grabs as much BW as it can
 - Measured performance is too good and irrelevant
 - Loss rate reflects self inflicted congestion (See BTC framework)
 - Furthermore the test itself adversely impacts users

It is common knowledge that this is a bad idea....



- Generate test traffic that mimics TCP and application behavior
 - Primarily determined by the presumed rest of the path and app
 - Presumed rest of path determines the target RTT
 - \circ Open loop
 - insensitive to the behavior of path under test
 - Test path length doesn't change the traffic
 - Except by diffusion/reclocking
- Gather delivery statistics (loss, jitter)
 - Evaluate against a performance model
 - Success criteria depends on target performance
 - Not the details of the test path

Streaming HD Video example (Follows from the draft)

- Target end-to-end parameters
 - o target_rate = 2.5 Mb/s
 - target_RTT = 50 mS
 - target_MTU = 1500 Bytes
 - Header_overhead = 64 Bytes
- Computed end-to-end parameters (from model)
 - o target_pipe_size = 11 packets
 - o target_run_length = 363 packets
 - e.g. one loss every 33 round trips
- Apportionment assumptions
 - For smooth traffic, the IXC contributes at most 10% of the loss
 - For bursty traffic, the IXC contributes at most 50% of the loss

HD Video tests

- Smooth
 - One 1500 byte packet every 4.5 mS
 - Loss rate must be below 1 in 3630 packets

• Bursts

- (11) 1500 byte packets, every 50 mS
- Loss rate must be below 1 in 66 packets (1 in 6 busts)
- Engineering (bench) tests
 - Onset of congestion must not cause loss of TCP self clock
 - e.g. AQM like progressive onset of losses

Next Steps

- New draft: MBM forStreaming HD video
 - Fully specified As concrete as possible
 - Built on existing metrics
 - E.g. OWAMP, etc
 - Field tests
 - Measurement Lab and elsewhere
- Research paper
 - Move some of the background from the current draft
 - Measurement results and other non-RFC material
- Hold current draft, pending input from above
 - Scope changes
 - Clarifications

Questions?

