Types of Network Delay

Network Startup Resource Center
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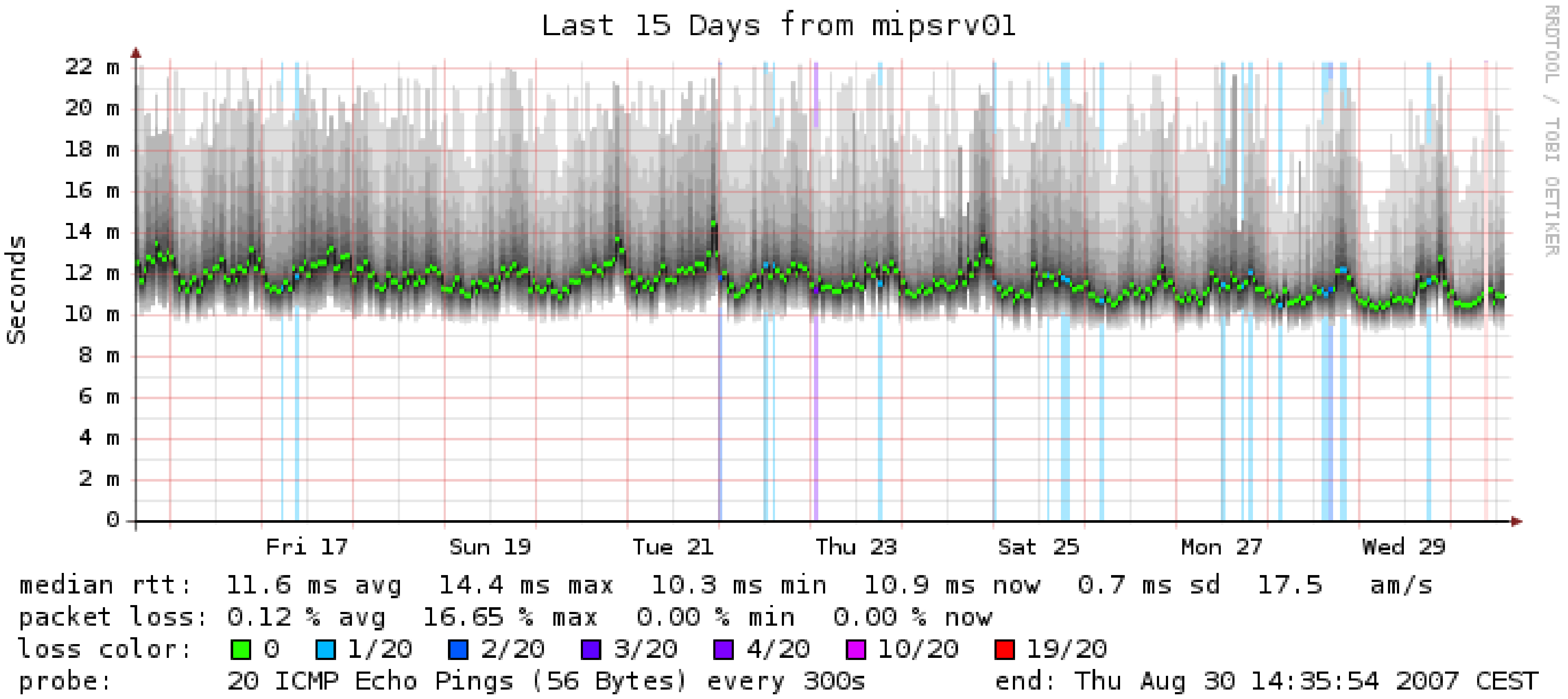
End-to-End Delay

The time required to transmit a packet along its <u>entire path</u>

Created by an application, handed over to the OS, passed to a network card (NIC), encoded, transmitted over a physical medium (copper, fibre, air), received by an intermediate device (switch, router), analyzed, retransmitted over another medium, etc.

The most common measurement uses *ping* for total round-trip-time (RTT).

Historical Measurement of RTT



Courtesy https://oss.oetiker.ch/smokeping/doc/reading.en.html

- What is this telling us?
- We need to understand the sources of delay

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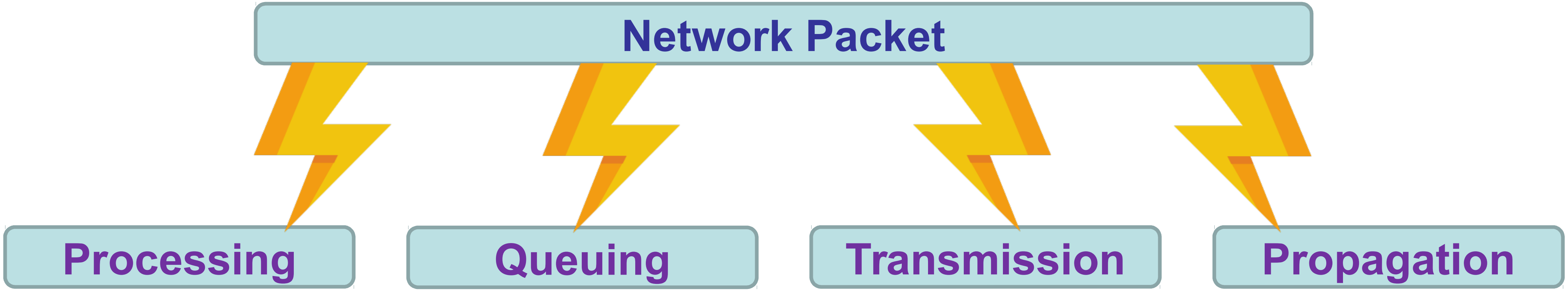
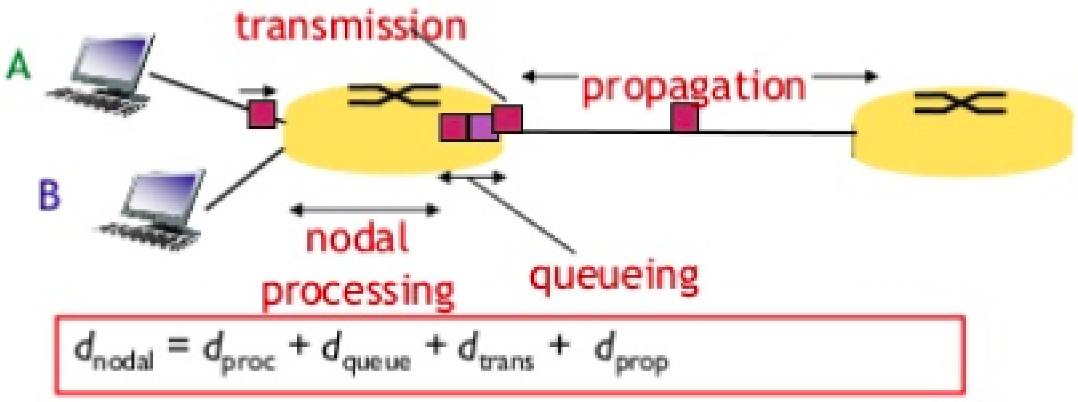
Four Causes of Delay





Courtesy https://www.slideshare.net/maamirfarooq/lec-4packet-delay-layered-architecture

Processing Delay

Time required by intermediate routers to decide where to forward the packet, update TTL, perform header checksum calculations

(Note: most modern routers handle packet forwarding in hardware at full line rate)

Plus:

Time for the far end to process the ICMP echo request and generate a response

Queuing Delay

The time a packet is enqueued while the link is busy sending other packets

- This is a statistical function and depends on the arrival times of other packets
- QoS configurations may prioritize some types of traffic over others
- (In practice, that means multiple queues, and different packets are assigned to different queues)

Transmission Delay

The time required to push all the bits in a packet on the transmission medium in use

N=Number of bits in packet,
R=transmission rate (bits per second)
t=time in milliseconds

t = N/R

Example: Transmit 1500 bytes (12000 bits) using FastEthernet (100Mbps)

$t = 12000/1 \times 10^8 = 0.12$ milliseconds

Propagation Delay

- Once a bit is 'pushed' on to the transmission medium, the time required for the bit to propagate to the other end of its <u>physical path</u>
- For a given medium, the velocity of propagation is usually constant (some fraction of the speed of light)
- The longer the path, the longer the delay

For x = distance, v = propagation velocity

t = x/v

Transmission vs. Propagation

Can be confusing at first

Consider this example:

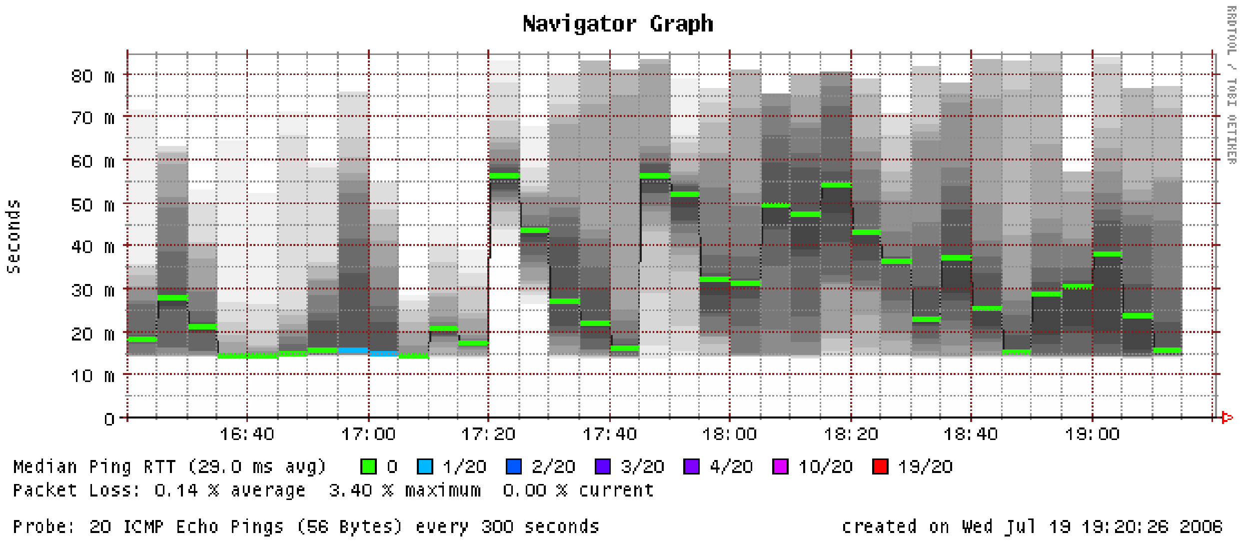
Two 100 Mbps circuits

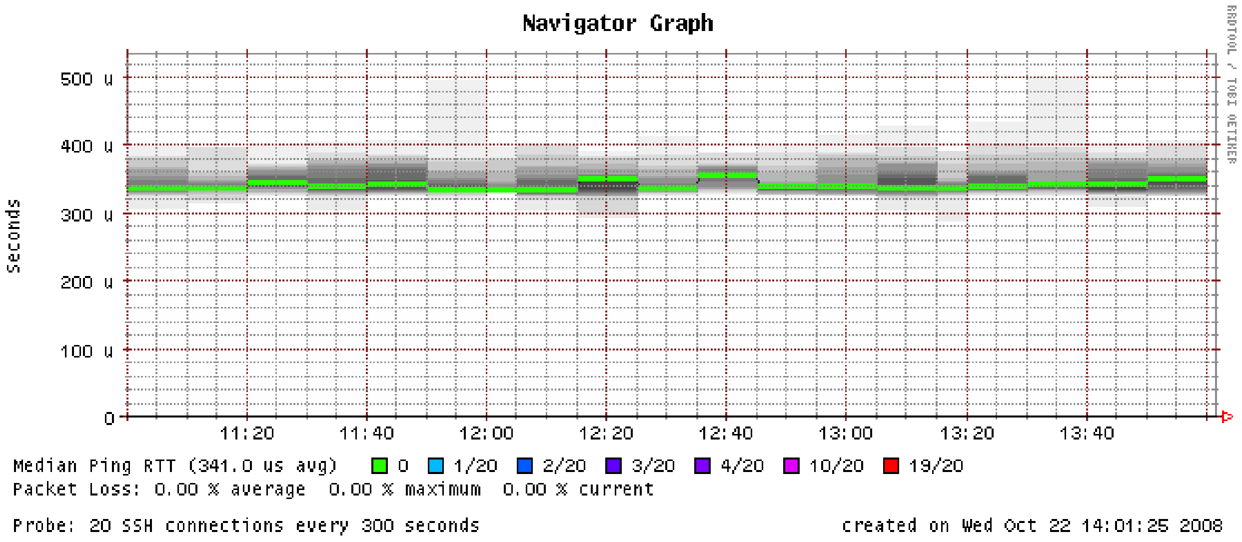
- 1 km of optic fiber

- Via satellite with a distance of 35,000 km between the base and the satellite

For two packets of the same size which will have the larger transmission delay? Propagation delay?

Jitter





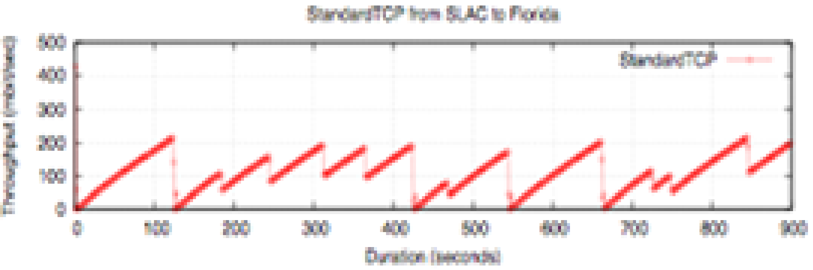
Obtain new top image. uS are no good

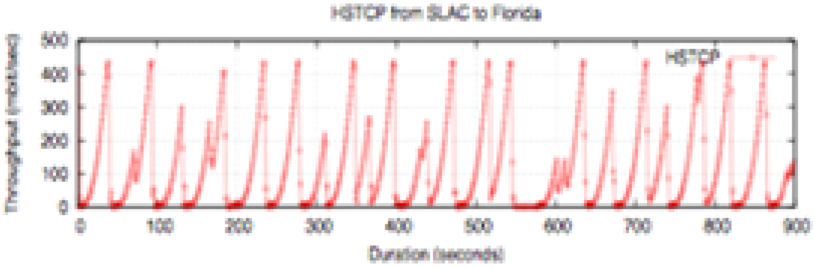
Questions about Jitter

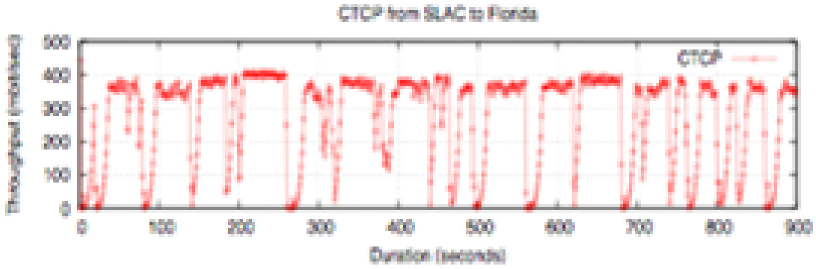
- We've seen four causes of delay. Which are constant for a given path and packet size, and which are variable?
- What applications are particularly sensitive to jitter?
- Those applications may apply extra buffering to smooth out jitter – why is that additional delay a problem?

Packet Loss

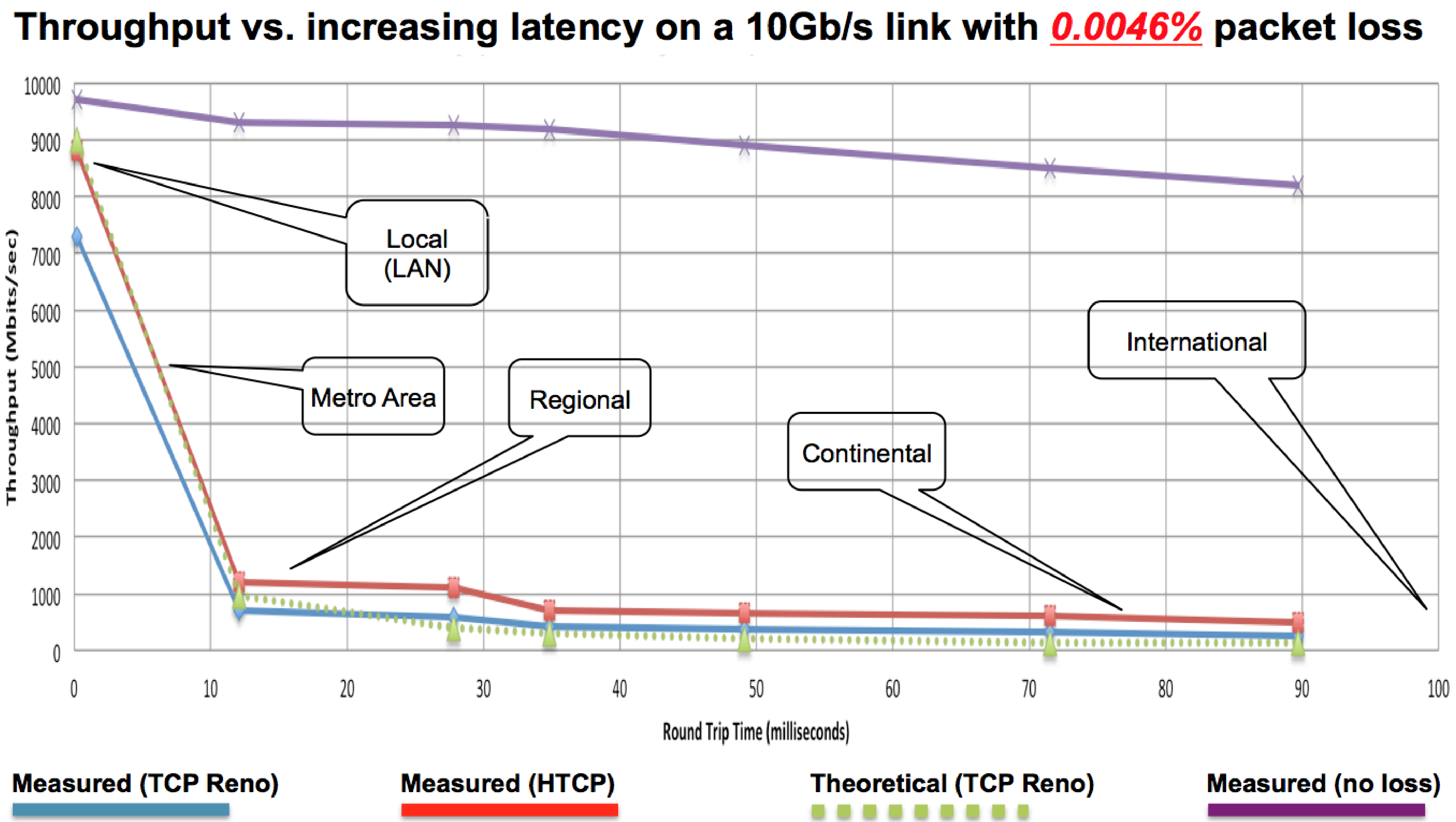
- Happens when buffers are full
- Packet loss, when necessary to correct, is done at the transport and application levels.
- Loss correction using packet retransmission can cause more congestion if flow control algorithms are not used.
    - Flow control: tell sender to back off sending and retransmit packets:







Bandwidth and Packet Loss



Questions

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Packet Loss

Causes of packet loss:

- Transmission errors
- Queue overflow (congestion)

Transmission Errors

"1" received as "0", or vice versa

- e.g. due to excess noise, poor connections, ...

Can be measured in terms of "bit error rate" (BER)

If one or more bits in a packet is corrupted, the whole packet is discarded

Retransmission of lost packets is the responsibility of higher layers (transport or application)

Queue Overflow

Queues do not have infinite size

If a packet arrives when queue already full, it is dropped

Ultimately caused by insufficient capacity

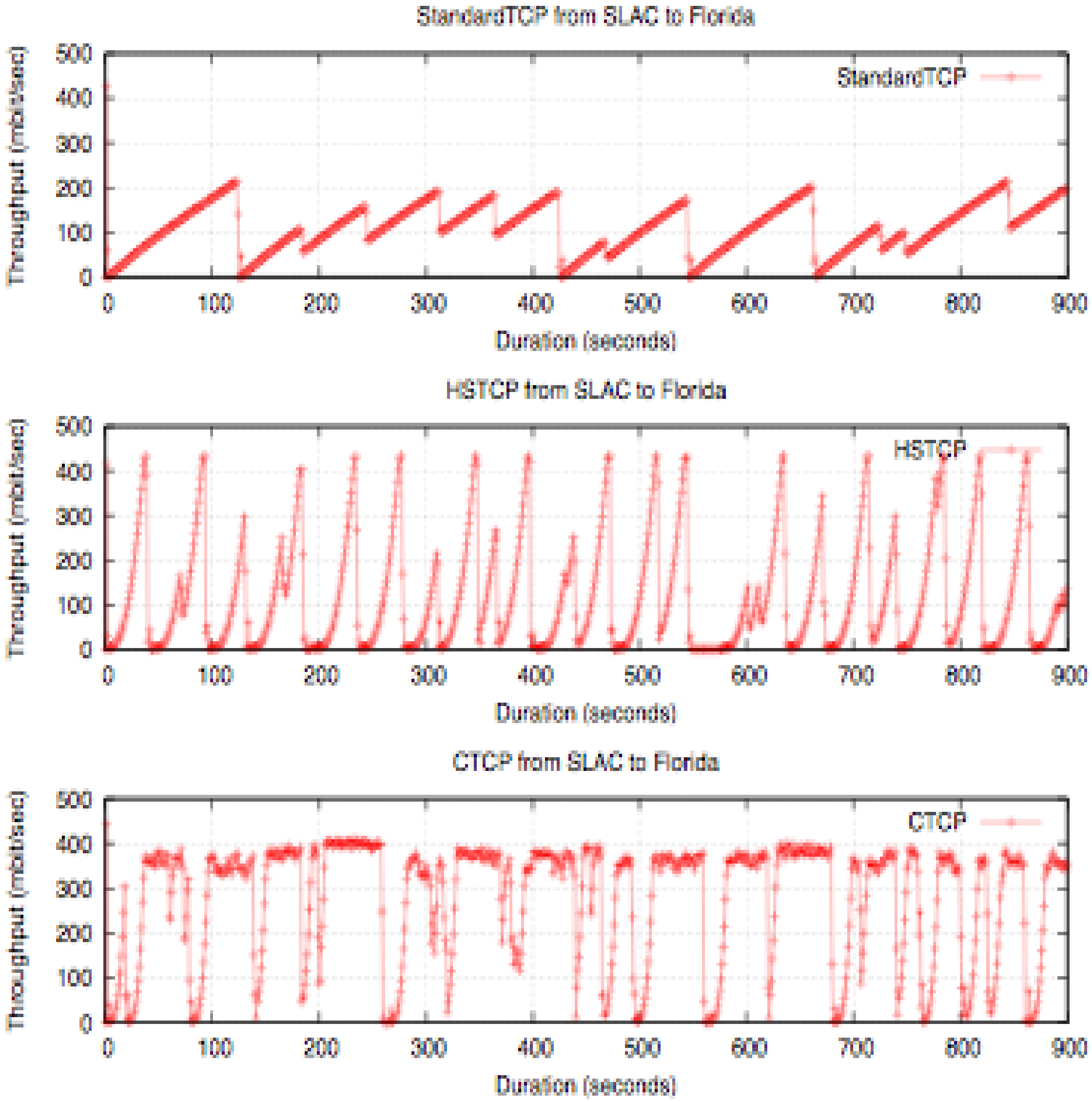
However, packet loss starts to occur before the link is 100% utilized, because of random distribution of arrival times

Retransmissions cause further demand and could lead to network collapse!

TCP and Congestrion Control

- TCP limits sending rate by means of a "congestion window"
- The congestion window starts small, and increases gradually while there is no packet loss
- Any detected packet loss causes the congestion window to shrink rapidly, so the sender sends more slowly

Different TCP Congestion Control Algorithms



Effects of TCP Congestion Control

- Network collapse is prevented
- "Fair sharing"
  - ✓ When there are multiple TCP streams, each one uses an approximately equal share of available bandwidth
- TCP detects congestion by <u>observing packet loss</u>
  - ✓ Newer TCP stacks also respond to "Explicit Congestion Notification" signals from routers: packets are marked when queues nearly full

TCP and Transmission Errors

- TCP cannot tell the difference between transmission errors and queue overflows!
- Hence transmission errors cause TCP to slow down too
- Formula for maximum throughput of TCP in the presence of packet loss:

$$\frac{MSS}{RTT \cdot \sqrt{p_{loss}}}$$

Example Calculation: LAN

- MSS = 1460 bytes
- RTT = 1ms = 0.001 seconds
- Packet loss = 2% = 0.02
- 1460 / (0.001 * √0.02)
  ≈ 10.3MB/sec = 82 Mbps
- Short RTT means packet loss does not have a huge impact on local transfers

Example Calculation: WAN

- MSS = 1460 bytes
- RTT = 150ms = 0.15 seconds
- Packet loss = 0.02% = 0.0002
- 1460 / (0.15 * √0.0002)
  ≈ 690KB/sec = 5.5 Mbps
- Loss of just *1 packet in 5,000* causes severe reduction of throughput when transferring across the Internet!

Measurement of Packet Loss

- Smokeping gives a coarse measurement
  (20 packets every 5 minutes => 5% loss detectable, but bursts may be missed)
- For more accurate measurement you need a tool like perfsonar / owamp
  - Standard configuration sends 10 packets per second continuously
  - Can detect packet loss of 0.17% over one minute, or 0.0028% over one hour
  - Separate measurements in each direction

Questions

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