TCP Tuning

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TCP

- Transmission Control Protocol (TCP)
- One of the original core protocols of the Internet protocol suite (IP)
- >90% of the internet traffic
- Transport layer
- Delivery of a stream of bytes between
  - programs running on computers
  - connected to a local area network, intranet or the public Internet.
- TCP communication is:
  - Connection oriented
  - Reliable
  - Ordered
  - Error-checked
- Web browsers, mail servers, file transfer programs use TCP

Connection-Oriented

- A connection is established before any user data is transferred.
- If the connection cannot be established the user program is notified.
- If the connection is ever interrupted the user program(s) is notified.

Reliable

- TCP uses a sequence number to identify each byte of data.
- Sequence number identifies the order of the bytes sent
- Data can be reconstructed in order regardless:
  - Fragmentation
  - Disordering
  - Packet loss

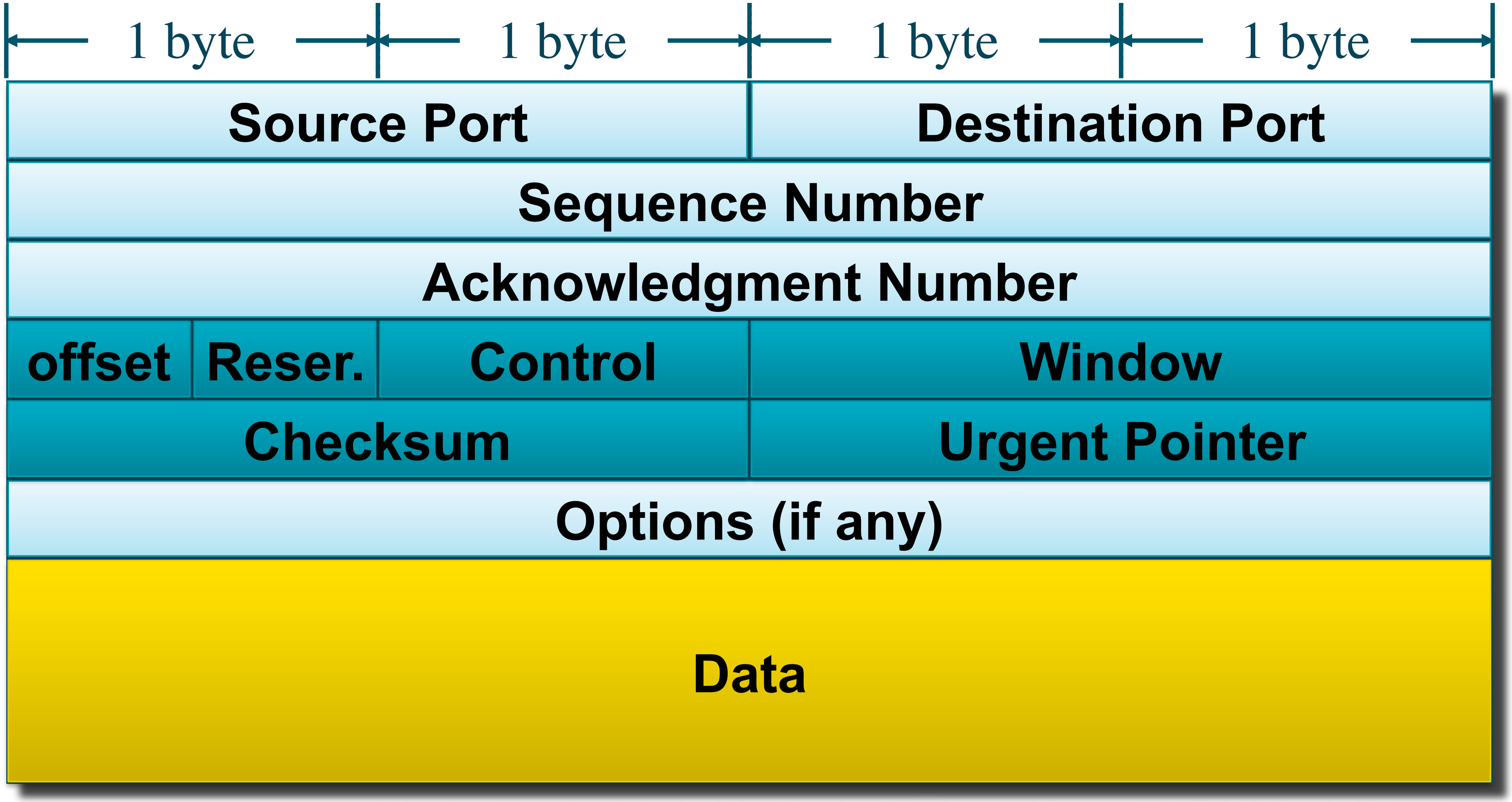
  that may occur during transmission.
- For every payload byte transmitted, the sequence number is incremented.

TCP Segments

- The block of data that TCP asks IP to deliver is called a *TCP segment*.

- Each segment contains:
  - Data
  - Control information

TCP Segment Format



Client Starts

- A client starts by sending a SYN segment with the following information:
  - Client's ISN (generated pseudo-randomly)
  - Maximum Receive Window for client.
  - Optionally (but usually) MSS (largest datagram accepted).

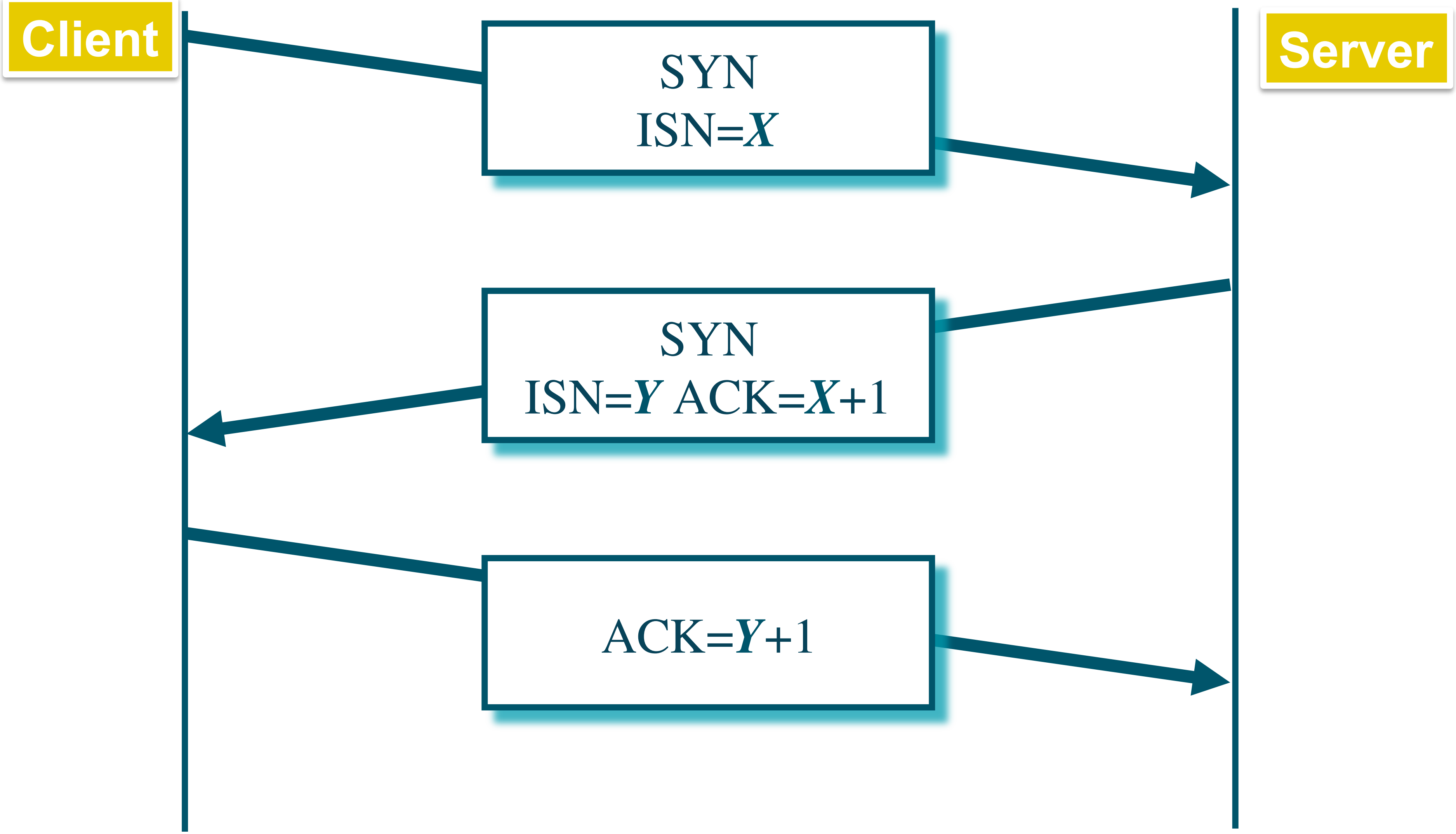
Server Response

- When a waiting server sees a new connection request, the server sends back a SYN segment with:
  - Server’s ISN (generated pseudo-randomly)
  - Request Number is Client ISN+1
  - Maximum Receive Window for server.
  - Optionally (but usually) MSS

Connection established!

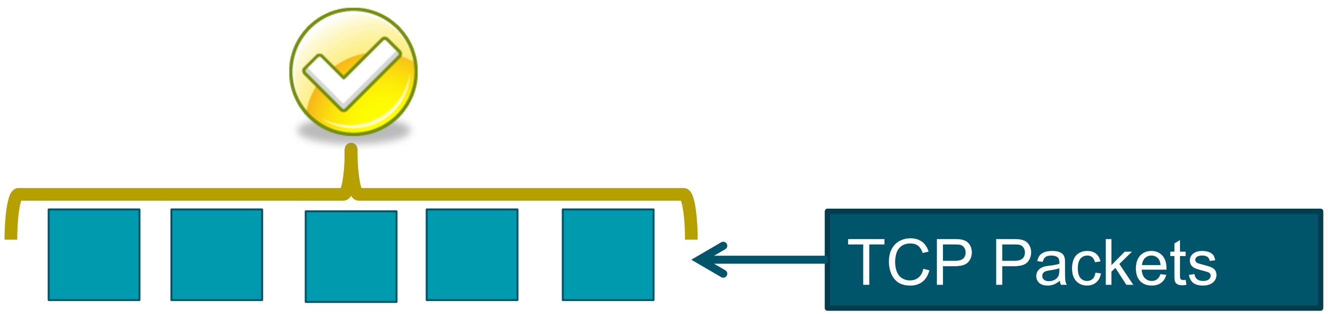
- When the Server’ s SYN is received, the client sends back an ACK with:
  - Acknowledgment Number is Server’s ISN+1

In blocks:



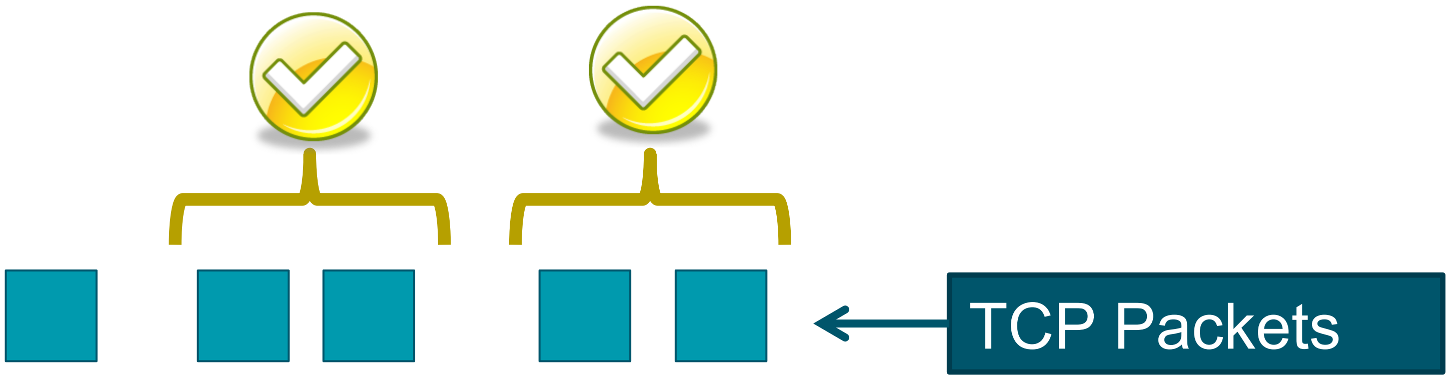
Cumulative acknowledgement

- Cumulative acknowledgment:
  - The receiver sends an acknowledgment when it has received **all** data preceding the acknowledged sequence number.
- Inefficient when packets are lost.
- Example:
  - *10,000 bytes are sent in 10 different TCP packets and*
  - *the first packet is lost during transmission.*
  - *The receiver cannot say that it received bytes 1,000 to 9,999 successfully*
  - *Thus the sender may then have to resend all 10,000 bytes.*



Selective acknowledgment

- Selective acknowledgment (SACK) option is defined in RFC 2018
- Acknowledge discontinuous blocks of packets received correctly
- The acknowledgement can specify a number of SACK blocks
- In the previous example above:
  - *The receiver would send SACK with sequence numbers 1000 and 9999.*
  - *The sender thus retransmits only the first packet, bytes 0 to 999.*



Buffering

- TCP works by:
  - buffering data at sender and receiver

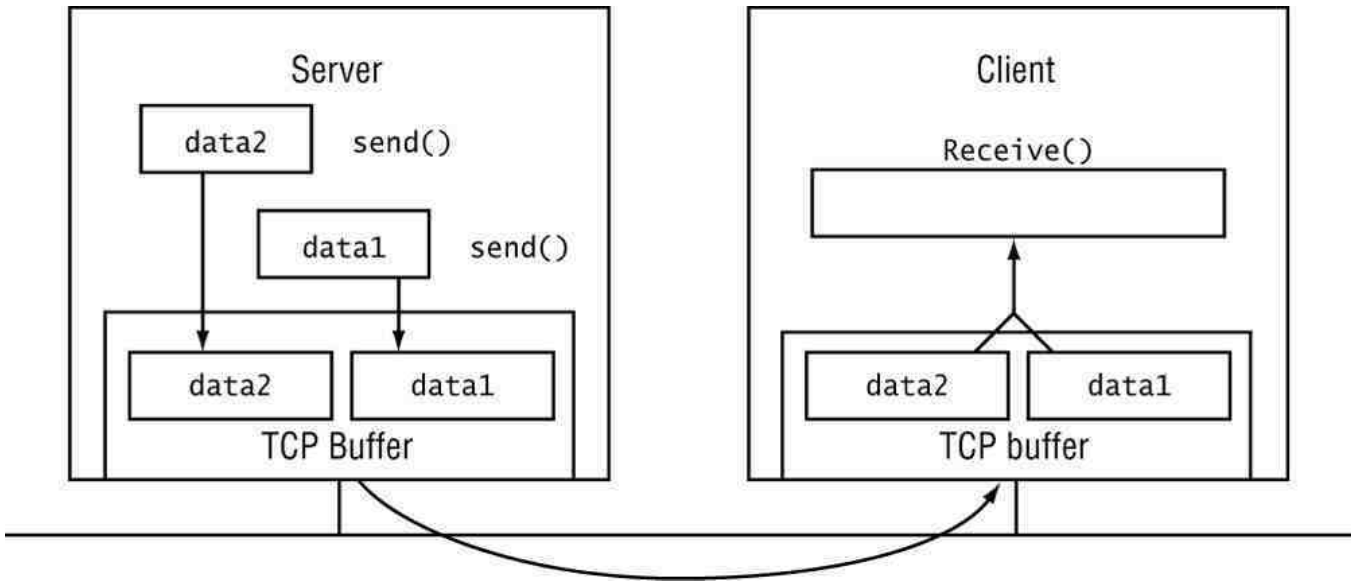


Image source: http://codeidol.com/img/csharp-network/f0502_0.jpg

Dynamic transmission tuning

- Data to send are temporarily stored in a send buffer
  - where it stays until the data is ACK'd.
- The TCP layer won't accept data from the application unless there is buffer space.
- Both the client and server announce how much buffer space remains
  - Window field in a TCP segment, with every ACK
  - TCP can know when it is time to send a datagram.

Flow control

- Limits the sender rate to guarantee reliable delivery.
- Avoid flooding
- The receiver continually hints the sender on how much data can be received
- When the receiving host buffer fills
  - the next ack contains a 0 in the window size
  - this stop transfer and allow the data in the buffer to be processed.

TCP Tuning

- Adjust the network congestion avoidance parameters for TCP
- Typically used over high-bandwidth, high-latency networks
  - Long-haul links (Long Fat Networks)
  - Intercontinental circuits
- Well-tuned networks can perform up to many times faster

Tuning buffers

- Most operating systems limit the amount of system memory that can be used by a TCP connection.
- Maximum TCP Buffer (Memory) space.
- Default max values are typically too small for network measurement and troubleshooting purposes.
- Linux (as many OSes) supports separate send and receive buffer limits
- Buffer limits can be adjusted by
  - The user
  - The application
  - Other mechanisms
- within the maximum memory limits above.

BDP Bandwidth Delay Product

- BDP=Bandwidth * Latency
- Number of bytes in flight to fill path
- Max number of un-acknowledged packets on the wire
- Max number of simultaneous bits in transit between the transmitter and the receiver.
- High performance networks have very large BDPs.

TCP receive buffer

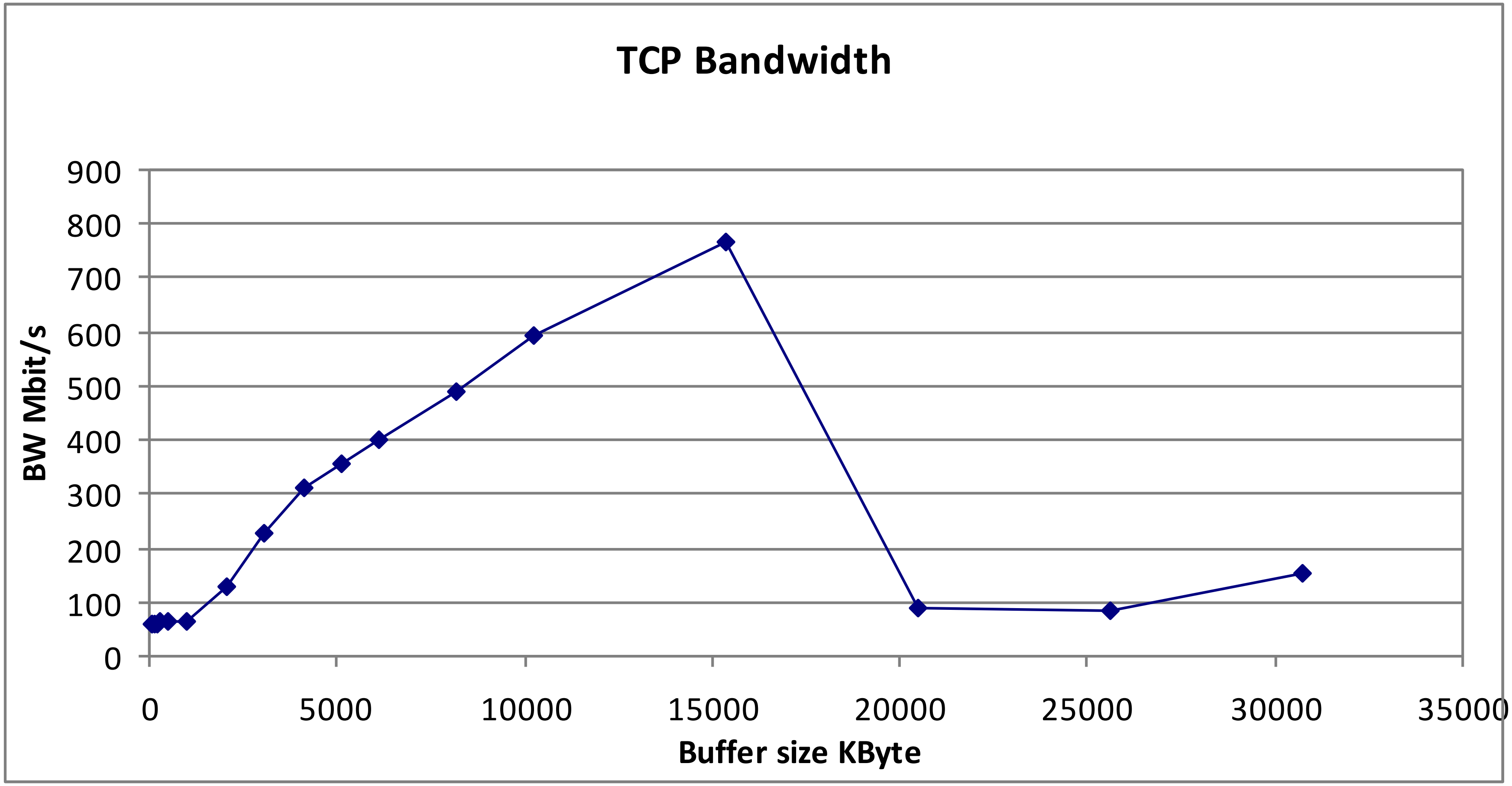
- Amount of data that a computer can store without acknowledging the sender.
- It can limit throughput
  - even if there is no packet loss in the network!
- TCP transmits data up to the buffer size before waiting for the ack
- Therefore the full bandwidth of the network may not always get used.

$$Throughput \leq TCP\ Receive\ buffer / RTT$$

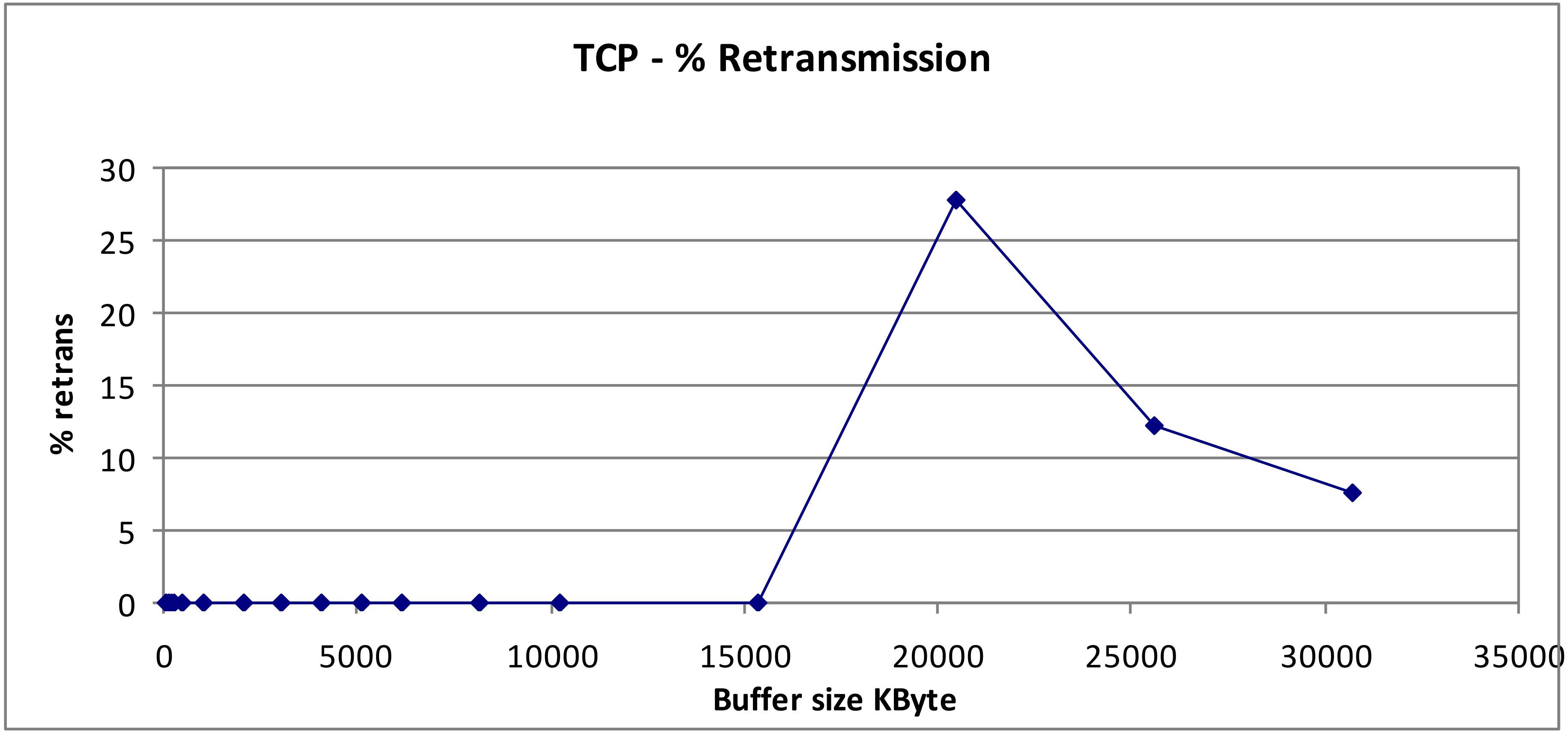
Optimising buffers

- TCP receiver and sender buffers needs tuning
- They should be ideally equal to BDP to achieve maximum throughput
- The sending side should also allocate the same amount of memory
- After data has been sent on the network
  - the sending side must hold it in memory until it has been ack'd
  - If the receiver is far away, acks will take a long time to arrive.
  - If the send memory is small, it can saturate and block transmission.

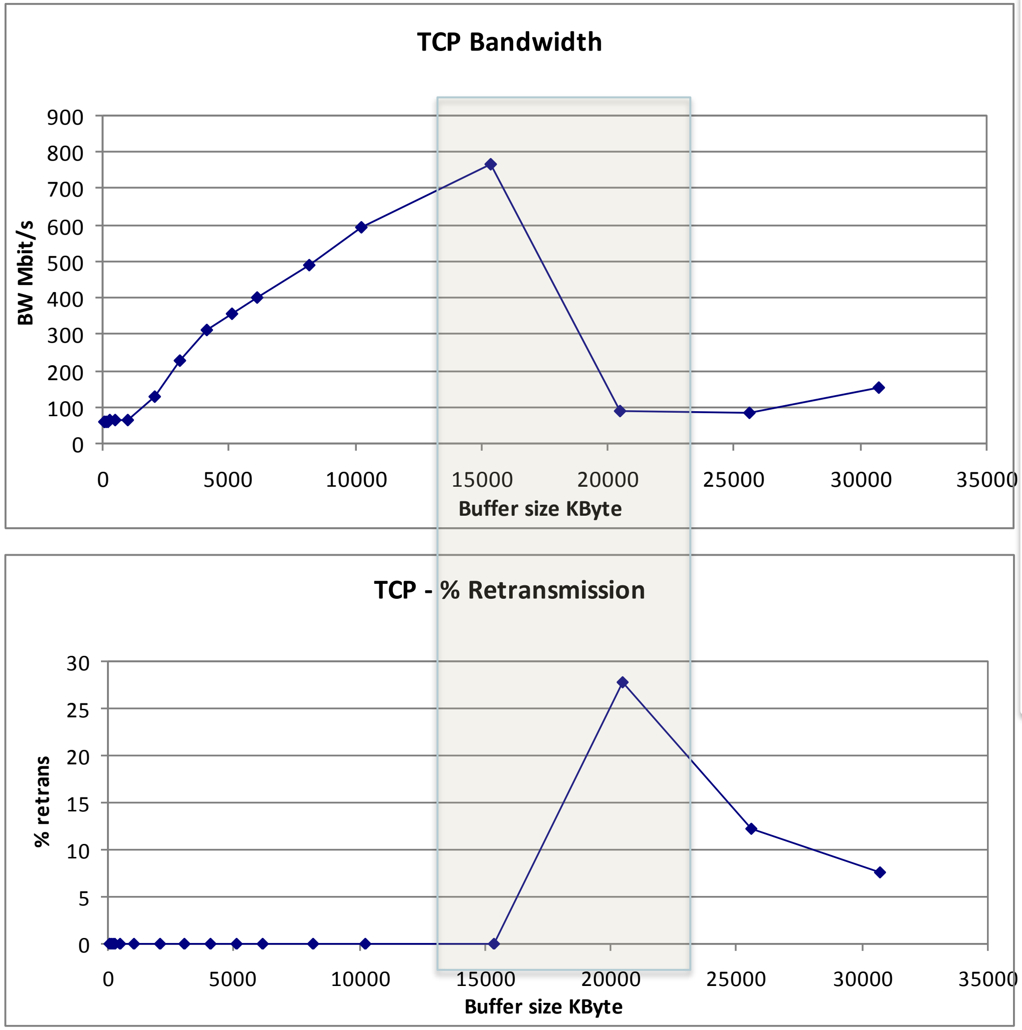
Madrid-Mumbai Bandwidth



Madrid-Mumbai Retransmission



BDP as optimal buffer parameter



Bandwidth increases with buffer size until it reaches BDP

RTT ~168ms
Bandwidth limit to 1GE interface
➔ ~20 Mbytes.

Checking send and receive buffers

- To check the current value type either:

```
$ sysctl net.core.rmem_max
net.core.rmem_max = 65535
$ sysctl net.core.wmem_max
net.core.wmem_max = 65535
```

- or

```
$ cat /proc/sys/net/core/rmem_max
65535
$ cat /proc/sys/net/core/wmem_max
65535
```

Setting send and receive buffers

- To change those value simply type:

```
sysctl -w net.core.rmem_max=33554432
sysctl -w net.core.wmem_max=33554432
```

- In this example the value 32MByte has been chosen:

  32 x 1024 x 1024 = 33554432 Byte

Autotuning buffers

- Automatically tunes the TCP receive window size for each individual connection
- Based on BDP and rate at which the application reads data from the connection
- Linux autotuning TCP buffer limits can be also tuned
- Arrays of three values:
  - minimum, initial and maximum buffer size.
- Used to:
  - Set the bounds on autotuning
  - Balance memory usage while under memory stress.
- Controls on the actual memory usage (not just TCP window size)
  - So it includes memory used by the socket data structures
- The maximum values have to be larger than the BDP
- Example: for a BDP of the order of 20MB, we can chose 32MB

Check and set autotuning buffers

- To check the TCP autotuning buffers we can use sysctl:

```
$ sysctl net.ipv4.tcp_rmem
4096    87380   65535
$ sysctl net.ipv4.tcp_wmem
4096    87380   65535
```

- It is best to set it to some optimal value for typical small flows.
- Excessively large initial buffer waste memory and can even hurt performance.
- To set them:

```
$ sysctl -w net.ipv4.tcp_rmem="4096    87380
33554432"
$ sysctl -w net.ipv4.tcp_wmem="4096    87380
33554432"
```

Checking and enabling autotuning

- TCP autotuning is normally enabled by default.
- To check type:

```
$ sysctl net.ipv4.tcp_moderate_rcvbuf
1
```

or

```
$ cat /proc/sys/net/ipv4/tcp_moderate_rcvbuf
1
```

- If the parameter *tcp_moderate_rcvbuf* is present and has value 1 then autotuning is enabled.
- With autotuning, the receiver buffer size (and TCP window size) is dynamically updated (autotuned) for each connection
- If not enabled, it is possible to enabled it by typing:

```
$ sysctl -w net.ipv4.tcp_moderate_rcvbuf=1
```

Interface queue length

- Improvement at NIC driver level
- Increase the size of the interface queue. To do this, run the following command.

```
$ ifconfig eth0 txqueuelen 1000
```

- TXQueueLen: max size of packets that can be buffered on the egress queue of a linux net interface.
- Higher queues: more packets can be buffered and hence not lost.
- In TCP, an overflow of this queue will cause loss
  - TCP will enter in the congestion control mode

Additional tuning

- Verify that the following variables are all set to the default value of 1

```
net.ipv4.tcp_window_scaling
net.ipv4.tcp_timestamps
net.ipv4.tcp_sack
```

Otherwise set them using

```
$ sysctl -w net.ipv4.tcp_window_scaling = 1
$ sysctl -w net.ipv4.tcp_timestamps = 1
$ sysctl -w net.ipv4.tcp_sack = 1
```

What not to change

- We suggest not to adjust *tcp_mem* unless there is some specific need.
- It is an array that determines how the system balances the total network buffer space
  - against all other LOWMEM memory usage.
- Initialized at boot time to appropriate fractions of the available system memory.
- In the same way there is normally no need to adjust *rmem_default* or *wmem_default*
  - These are the default buffer sizes for non-TCP sockets (e.g. unix domain and UDP sockets).

Congestion window and slow start

- **Congestion window**:
  - Estimation how much congestion there is between sender and receiver
  - It is maintained at the sender
- **Slow start**: increase the congestion window after a connection is initialized and after a timeout.
  - It starts with a window of 1 maximum segment size (MSS).
  - For every packet acknowledged, the congestion window increases by 1 MSS
  - The congestion window effectively doubles for every round trip time (RTT).
  - Actually not so slow…

TCP Congestion control

- Initially one algorithm available Reno
- Linear increment of the congestion window
- It typically drops to half the size when a packet is lost
- Starting from Linux 2.6.7, alternative congestion control algorithms were implemented
  - recover quickly from packet loss on high-speed and high BDP networks.
- The choice of congestion control options is selected when the kernel is built.

Some congestion control examples

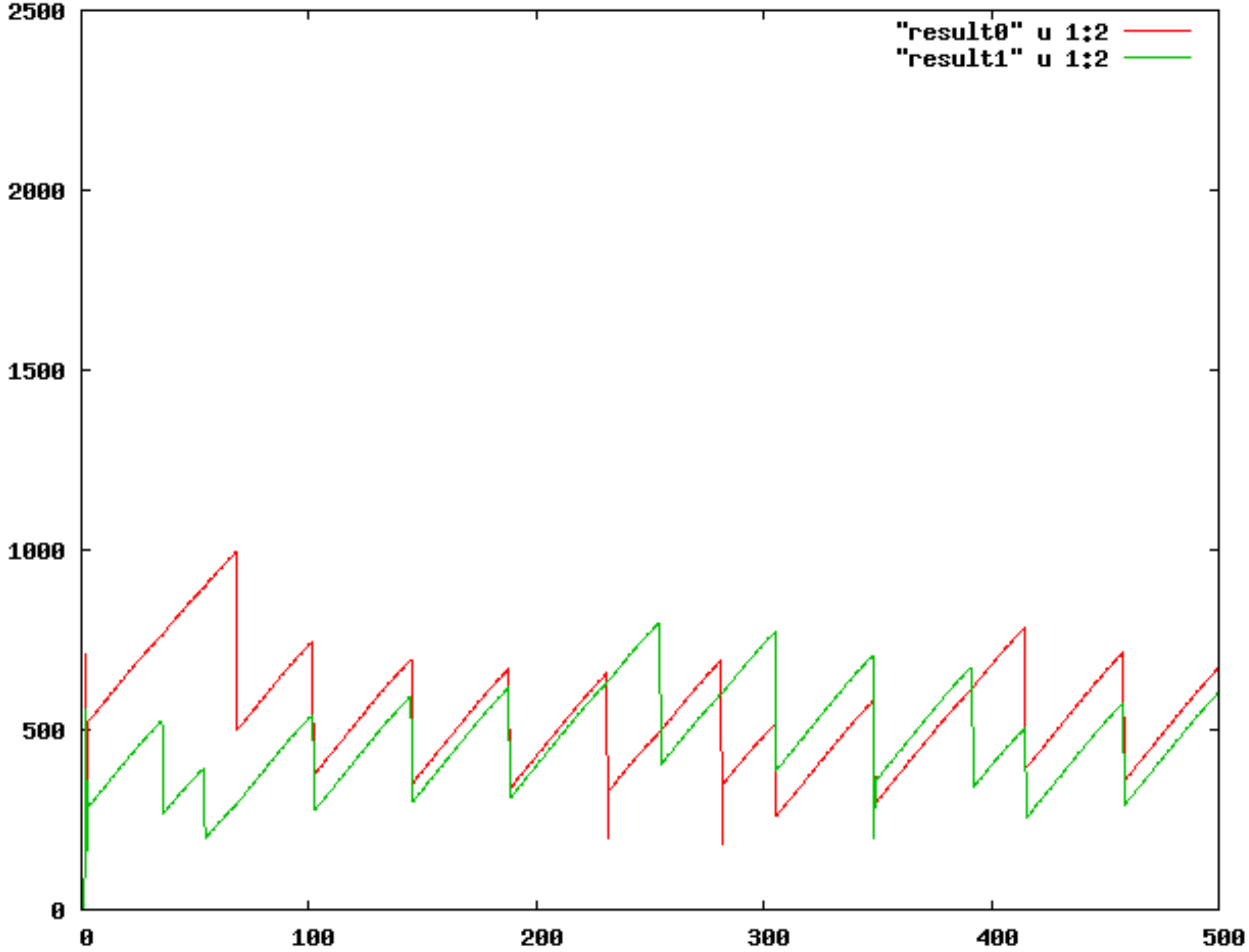
The following are some of the options are available in the 2.6 kernel:

- **reno**: Traditional TCP used by almost all other OSes (default with old Linux kernel).
  - It adjusts congestion window based on packet loss.
  - The slow start has an additive Increase window on each Ack and
  - a Multiplicative Decrease on loss
- **cubic**: Faster (cubic function) recovery on packet loss
  - Efficient for high-BDP network
- **bic**: Combines two schemes called additive increase and binary search increase.
  - It promises fairness as well as good scalability.
  - Under small congestion windows, binary search increase is designed to provide TCP friendliness.
  - Default congestion-control in many Linux distribution

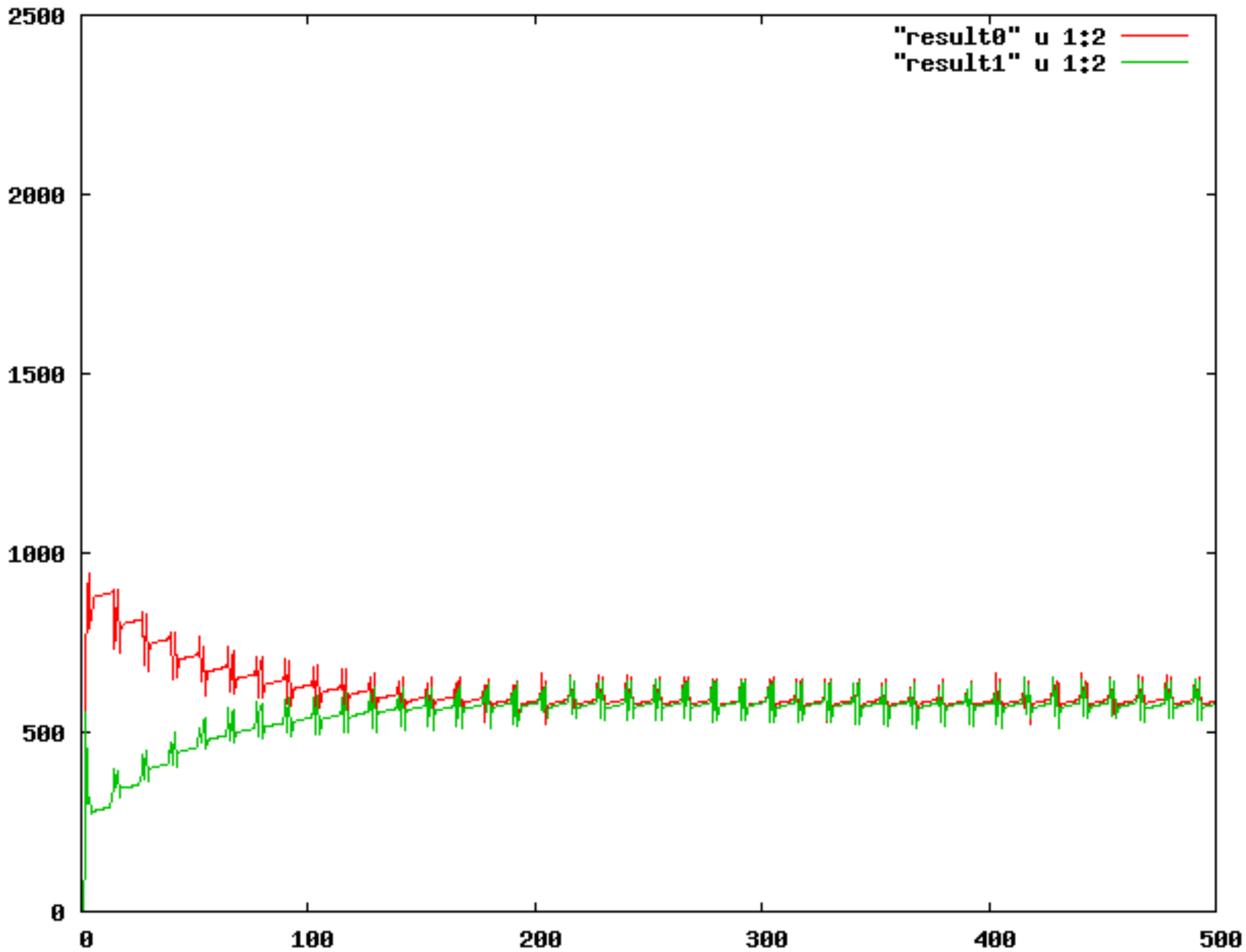
Some congestion control examples Cont.

- **hstcp:** An adaptive algorithm that:
  - Increases its additive increase parameter and
  - decreases its decrease parameter in relation to the current congestion window size.
- **vegas:** It measure bandwidth based on RTT and adjust congestion window on bandwidth
- **westwood:** optimized for lossy networks. The focus in on wireless networks (where packet loss does not necessarily mean congestion).
- **htcp**: Hamilton TCP: Optimized congestion control algorithm for high speed networks with high latency (LFN: Long Fat Networks).
  - Hamilton TCP increases the rate of additive increase as the time since the previous loss increases.
  - This avoids the problem of making flows more aggressive if their windows are already large (cubic).

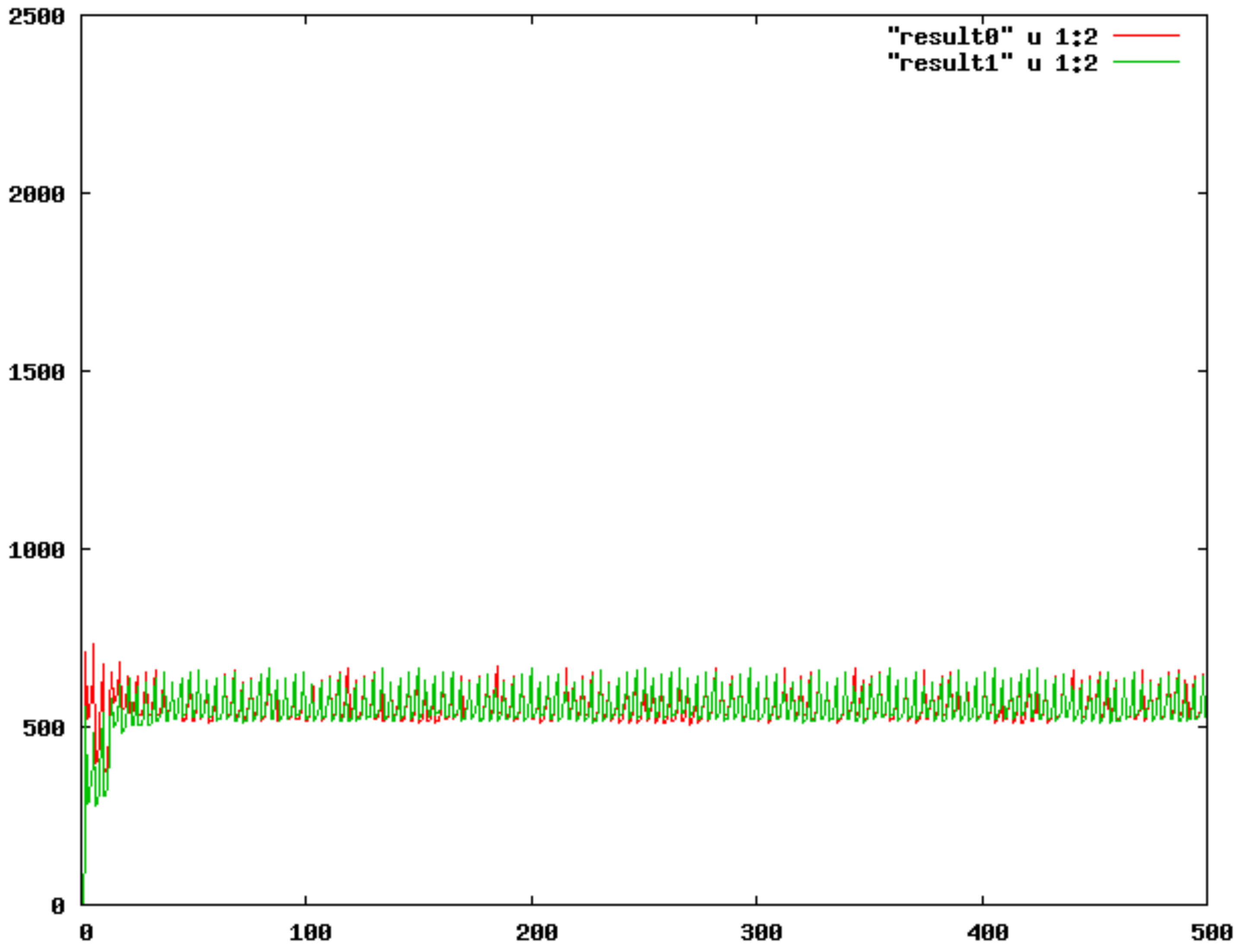
Congestion control: Reno with two flows



Congestion control: BIC with two flows



Congestion control: Hamilton with two flows



Checking and setting congestion control

- To get a list of congestion control algorithms that are available in your kernel, run:

```
$ sysctl net.ipv4.tcp_available_congestion_control
net.ipv4.tcp_available_congestion_control = cubic
reno bic
```

- To know which is the congestion control in use

```
$ sysctl net.ipv4.tcp_congestion_control
reno
```

- To set the congestion control

```
sysctl -w net.ipv4.tcp_congestion_control=cubic
```

Final considerations

- *Large MTUs*:
  - Linux host is configured to use 9K MTUs
  - But the connection is using 1500 byte packets,
  - ➔ then it is actually needed 9/1.5 = 6 times more buffer space in order to fill the pipe.
  - ➔ In fact some device drivers only allocate memory in power of two sizes, so it could be even needed 16/1.5 = 11 times more

Final considerations (cont.)

- *Very large BDP paths*:
  - For very large BDP links (>20 MB) there could be some Linux SACK implementation problem.
  - Too many packets in flight when it gets a SACK event
    - ➔ *too long to locate the SACKed packet*
    - ➔ *TCP timeout and CWND goes back to 1 packet.*
  - Restricting the TCP buffer size to about 12 MB seems to avoid this problem
    - *but clearly limits the total throughput.*
  - Another solution is to disable SACK

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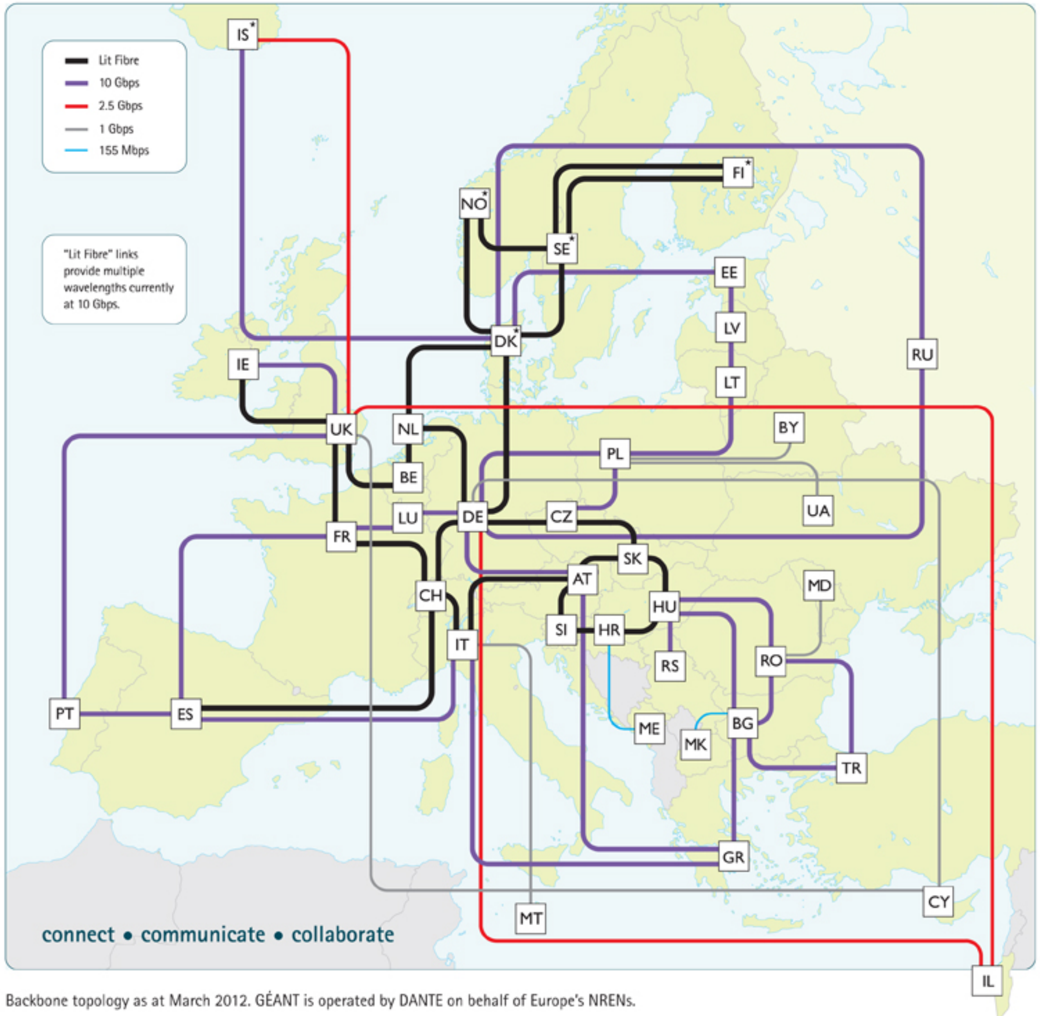

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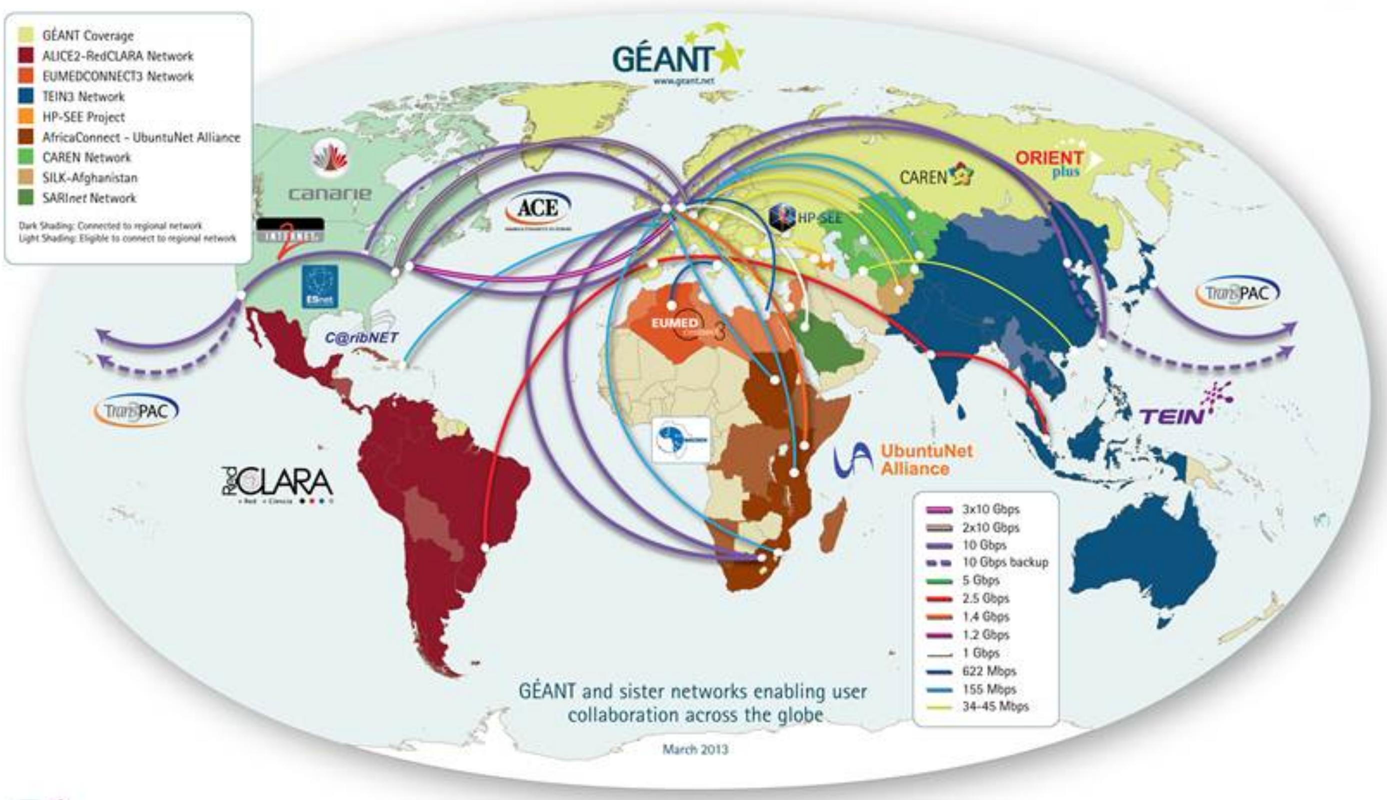

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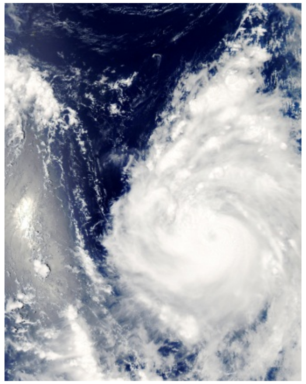




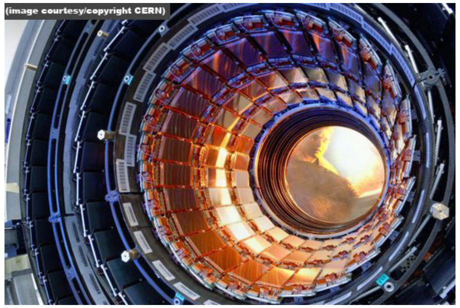

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