net 20: Transport Layer — Implemented

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Tame the Net

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Overview



Motivation

Datenflusssteuerung

Reliability

Cryptography



TCP–Flow Control: "Buffer Bloat"

- TCP as "carefree protocol" is not even remotely real-time capable, so far from "carefree" for media use
- UDP is only a "easy" access to raw IP, and otherwise "do it yourself"
- The SSL–PKI with their "honest Achmeds" as certification authorities
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- Routing address length changed to 128 bits
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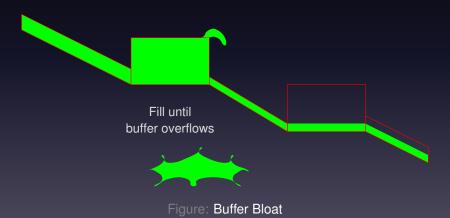


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Status: TCP Flow Control



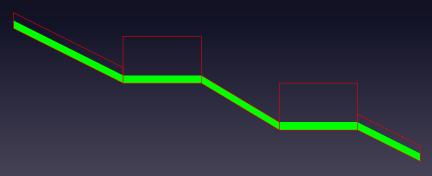
 TCP fills the buffer, until a packet has to be dropped, instead of reducing rate before. Name of the symptom: "Buffer bloat". But buffering is essential for good network performance.



Alternatives?



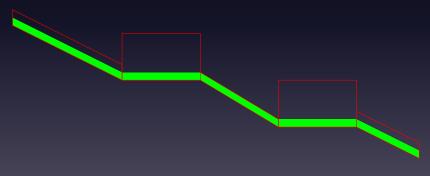
- LEDBAT tries to achieve a low, constant delay: Works, but not good on fairness
- CurveCP has a similar approach, which is not even documented (but DAN BERNSTEIN's code is by definition "obvious")
- Therefore, something new has to be done



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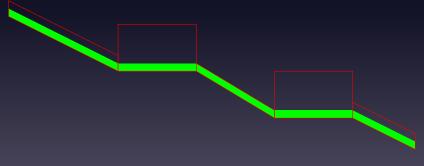


Figure: That's how proper flow control should look like



- Retransmits are making the situation worse in case of congestions and therefore should be avoided
- Riddle: How big should the buffer be, under the assumption that the bandwidth is used optimally, the bottleneck is on the other side of the connection, and a second data stream is opened up?
- Answer: about half the round trip delay, which are inevitably filled before any reaction is possible
 Buffers are good, but you shouldn't fill them up to the brim
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- Buffers are good, but you shouldn't fill them up to the brim
- The problem is inherent in the TCP protocol, but since Windows XP did not provide window scaling, the per-connection buffer limit was 64k for most connections on the Internet for quite a long time.

net2o Flow Control





Figure: Measure the bottleneck using a burst of packets

Client Measures, Server Sets Rate 🦀



Client recores the *time* of the first and last packet in a burst, and calculates the achieved rate for received packets, extrapolating to the achievable rate including the dropped packets. This results in the requested rate.

```
: calc-rate ( -- )
 delta-ticks @ tick-init 1+ acks @ */
 lit, set-rate;
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Server would simply use this rate

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: set-rate ( rate -- ) ns/burst ! :
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Fairness means that concurrent connections achieve about the same data rate, sharing the same line in a fair way.

 Ideally, a router/switch would schedule buffered packets round-robin, giving each connection a fair share of the bandwidth. That would change the calculated rate appropriately, and also be a big relieve for current TCP buffer bloat symptoms, as each connection would have its private buffer to fill up.

Unfortunately, routers use a single FIFO policy for all connections

Finding a sufficiently stable algorithm to provide fairness

connections, achievable rate changes are not only related to traffic.



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- Unfortunately, routers use a single FIFO policy for all connections
- Finding a sufficiently stable algorithm to provide fairness
- We want to adopt to new situations as fast as possible, there's no point in anything slow. Especially on wireless connections, achievable rate changes are not only related to traffic.

net2o Flow Control — Fair Router 🛛 🦀



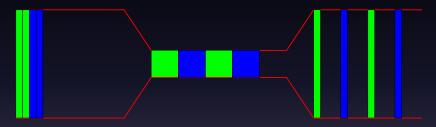


Figure: Fair queuing results in correct measurement of available bandwidth

net2o Flow Control — FIFO Router 🦀



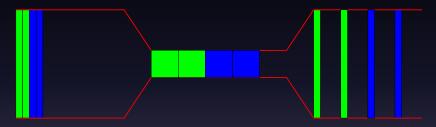


Figure: Unfair FIFO queuing results in twice the available bandwidth calculated



- To improve stability of unfair queued packets, we need to improve that P regulator (proportional to measured rate) to a full PID regulator
- The integral part is the accumulated slack (in the buffer), which we want to keep low, and the D part is growing/reducing this slack from one measurement to the next
 - We use both parts to decrease the sending rate, and thereby achieve better fairness
- The I part is used to exponentially lengthen the rate Δt

$$s_{exp} = 2^{\frac{slack}{T}}$$
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Fairness D



- To measure the differential term, we measure how much the slack grows (a Δt value) from the first to the last burst we do for one measurement cycle (4 bursts by default, first packet to first packet of each burst)
- This is multiplied by the total packets in flight (head of the sender queue vs. acknowledged packet), divided by the packets within the measured interval
- A low-pass filter is applied to the obtained D to prevent from speeding up too fast, with one round trip delay as time constant
- max(slacks)/10ms is used to determine how aggressive this algorithm is
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VDSL



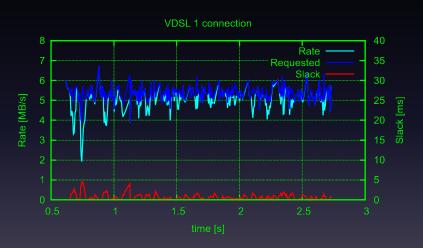


Figure: One connection on a VDSL

VDSL, Congestion



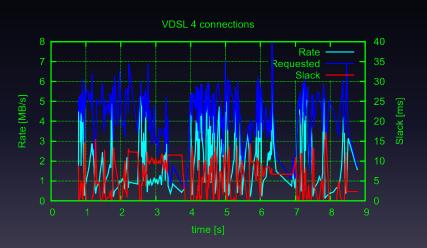


Figure: One of four connections on a VDSL

Unreliable Air Cable (WLAN)



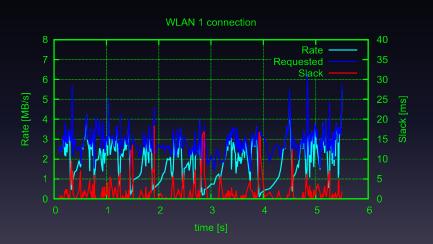


Figure: Single connection using WLAN

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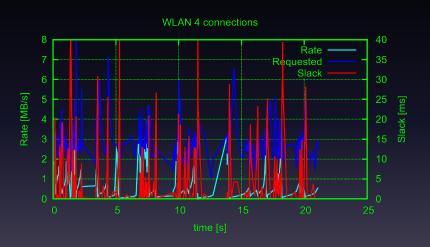


Figure: One of four connections using WLAN



- Packet ordering is dealt with the address each packet carries
- The receiver tracks received packets in two alternating bitmaps
- Received packets are marked as received in the active bitmap
- The other bitmap is filled up, until the bitmaps are swapped (twice per round trip delay RTD)
- Wait one RTD for retransmits
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Authentication The sender of the data can be identified

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- Elliptic Curve Cryptography doesn't base on large number factoring (as hard to solve problems), but on natural logarithms of elliptic curves
- Security level of Curve25519 corresponds to 128 bits in a symmetric key — that's sufficient today
- Curve25519 has a very efficient implementation
 It is optimized for 1:1 connections
- Each participant "multiplies" his secret key with the public key of the other side, both products are identical



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At the moment, I'm using Wurstkessel as symmetric encryption, even though there hasn't been a thorough review:

- Wurstkessel provides en/decryption and authentication in a single pass, computing a key-dependent secure hash
- Thus a single run of Wurstkessel solves all three tasks: confidentiality, integrity, and authentication.
 - 1 the data is encrypted
 - (2) the correct hash proves its integrity
 - 3 the hash can only be calculated knowing the key, therefore proving the authentication of the sender
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Hidden Initialization Vectors



- No key reuse allowed (only for retransmissions), otherwise a known–plaintext attack is possible
- Usual approach: initialization vector (IV) transmitted with each packet
- Disadvantage: Overhead and the "other" part of the key is known to the attacker
- Solution: Generate the IVs using a PRNG (with Wurstkessel in PRNG mode) on both sides — these IVs are "shared secrets". Only the seed for the PRNG is transmitted, and used together with the shared key to

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- Hierarchical Certification Authorities (e.g. SSL): The trust is delegated to "notaries", i.e. the CAs, which then must be trustworthy (all of them, since each CA can create a certificate for anybody). The server is certified, i.e. the user knows that he can trust this connection as much as the worst of those 600 CAs.
- Peer to Peer (e.g. PGP): trust is obtained through a "web of trust", i.e. you either trust directly or by using several people you trust. It is not sufficient to corrupt a single person in your trust network to obtain trust.
- 3 Observing changes (e.g. SSH): trust is reiterated by repeated contacts, and as long as keys don't change, trust is assumed.



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- Peer to Peer (e.g. PGP): trust is obtained through a "web of trust", i.e. you either trust directly or by using several people you trust. It is not sufficient to corrupt a single person in your trust network to obtain trust.
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For Further Reading



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Fossil Repository und Wiki

http://fossil.net2o.de/