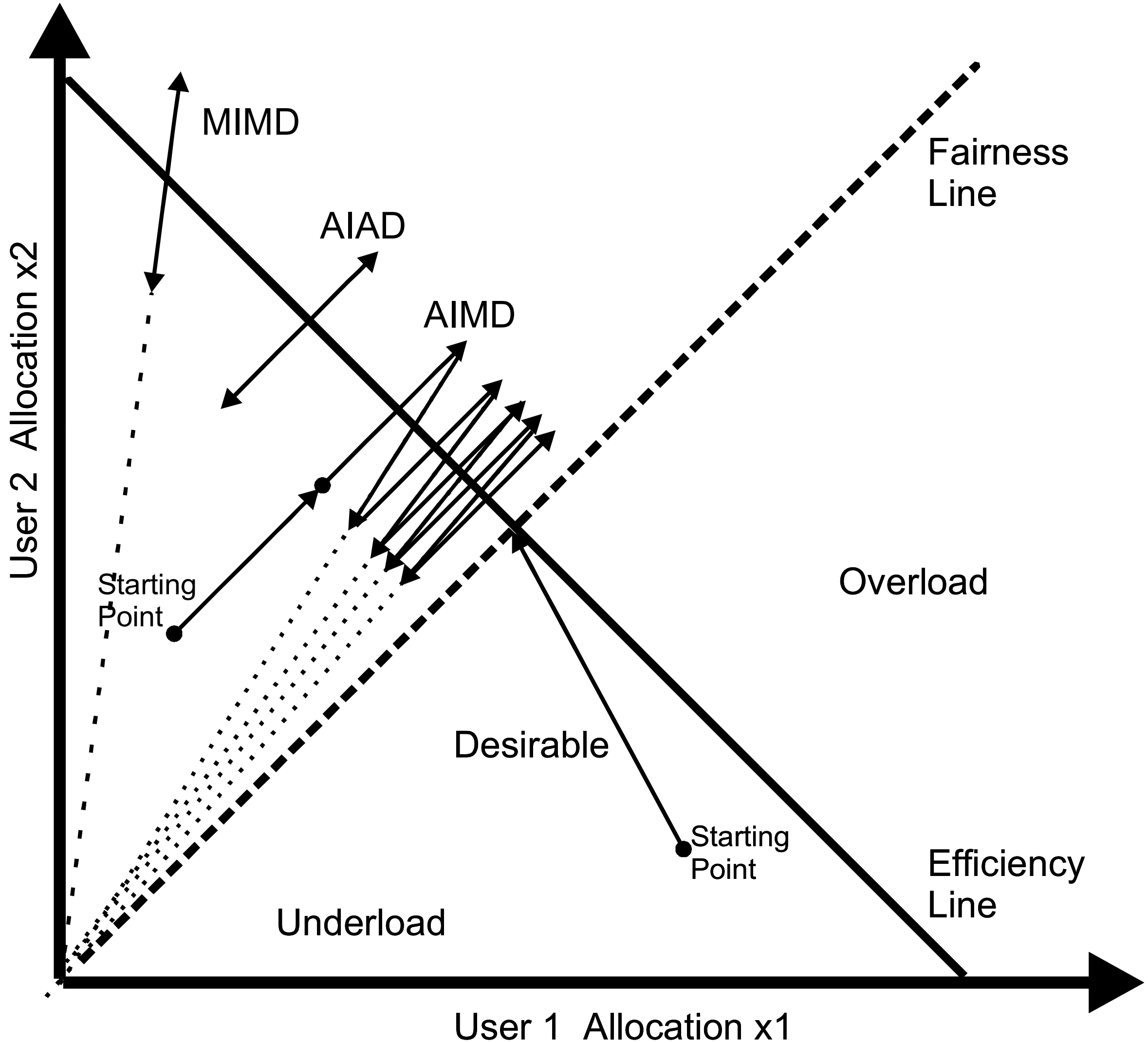
Internet congestion control: TCP

- **1988:** "Congestion Avoidance and Control" (Jacobson/Karels) Combined congestion/flow control for TCP
- Goal: stability - in equilibrum, no packet is sent into the network until an old packet leaves
  - ack clocking, "conservation of packets" principle
  - possible through window based stop&go - behaviour
- Superposition of stable systems = stable -> network based on TCP with congestion control = stable
- Today, TCP dominates the Internet (WWW, ..) recent backbone measurement: 98% TCP traffic

AIMD Background



Some reasons for TCP stability

- Exponential backoff:
  "For a transport endpoint embedded in a network of unknown topology and with an unknown, unknowable and constantly changing population of competing conversations, only one scheme has any hope of working - exponential backoff - but a proof of this is beyond the scope of this paper."

- Conservation of packets:
  "The physics of flow predicts that systems with this property should be robust in the face of congestion."

- Additive Increase, Multiplicative Decrease:
  Not explicitely cited as a stability reason **<u>in *the* paper</u>**!
  - ...but in 1000's of other papers!

"Proofs" of TCP stability

- AIMD:
  Chiu/Jain: diagram + algebraic proof homogeneous RTT case
- steady-state TCP model: window size ~ 1.22/sqrt(p)
  (p = packet loss)
- Johari/Tan, Massoulié, ..:
  - local stability, neglect details of TCP behaviour (fluid flow model, ..)
  - assumption:
    "queueing delays will eventually become small relative to propagation delays"
- Steven Low:
  - Duality model (based on utility function / F. Kelly, ..):
    Stability depends on delay, capacity, load and AQM !

Extended Use of Vector Diagrams

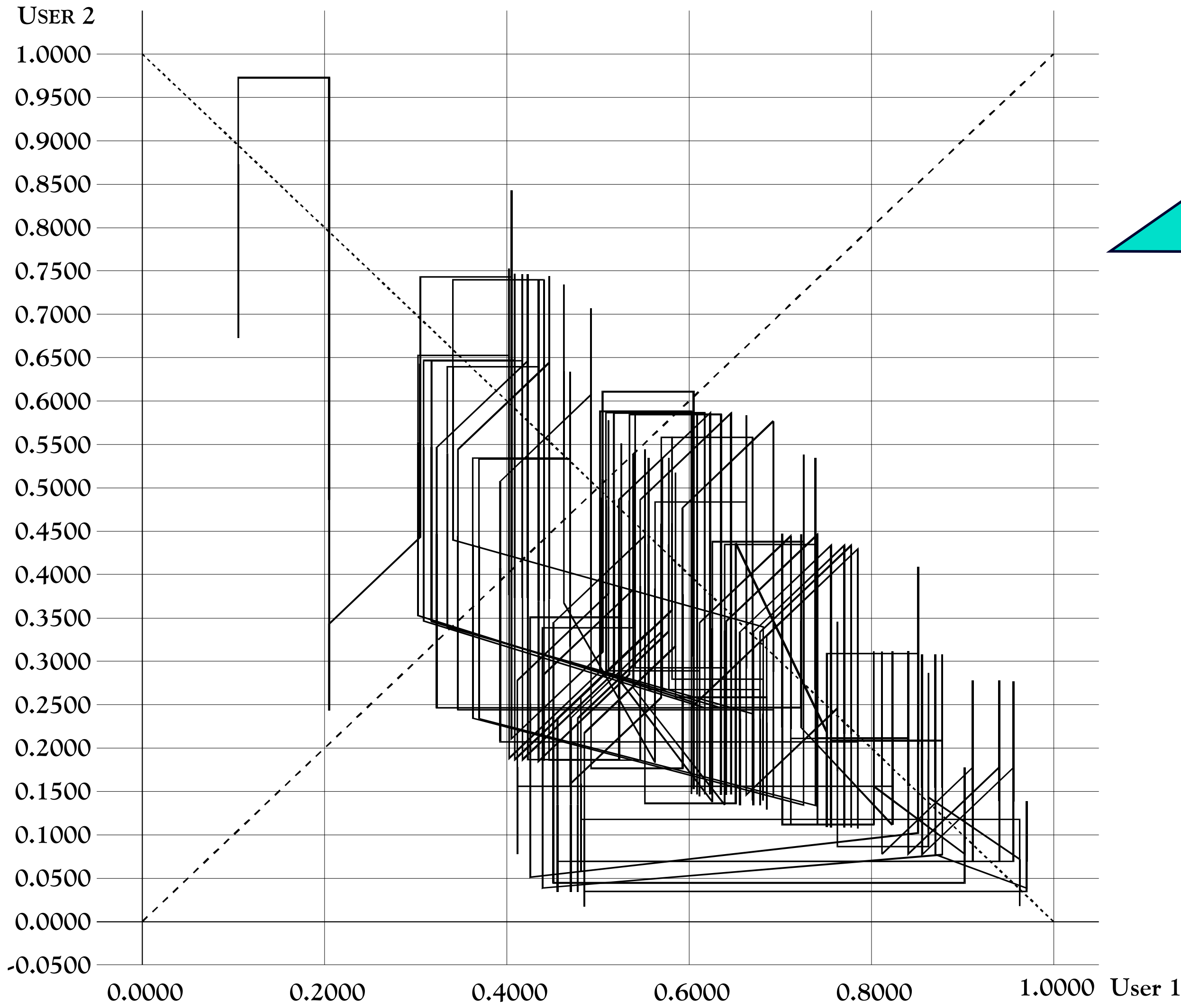
- Problem:
  - Stability analysis very complex
  - TCP-like mechanism design difficult

- Solution:
  - Extended use of vector diagrams!

- Analyze actual results (from simulation or real life measurements)

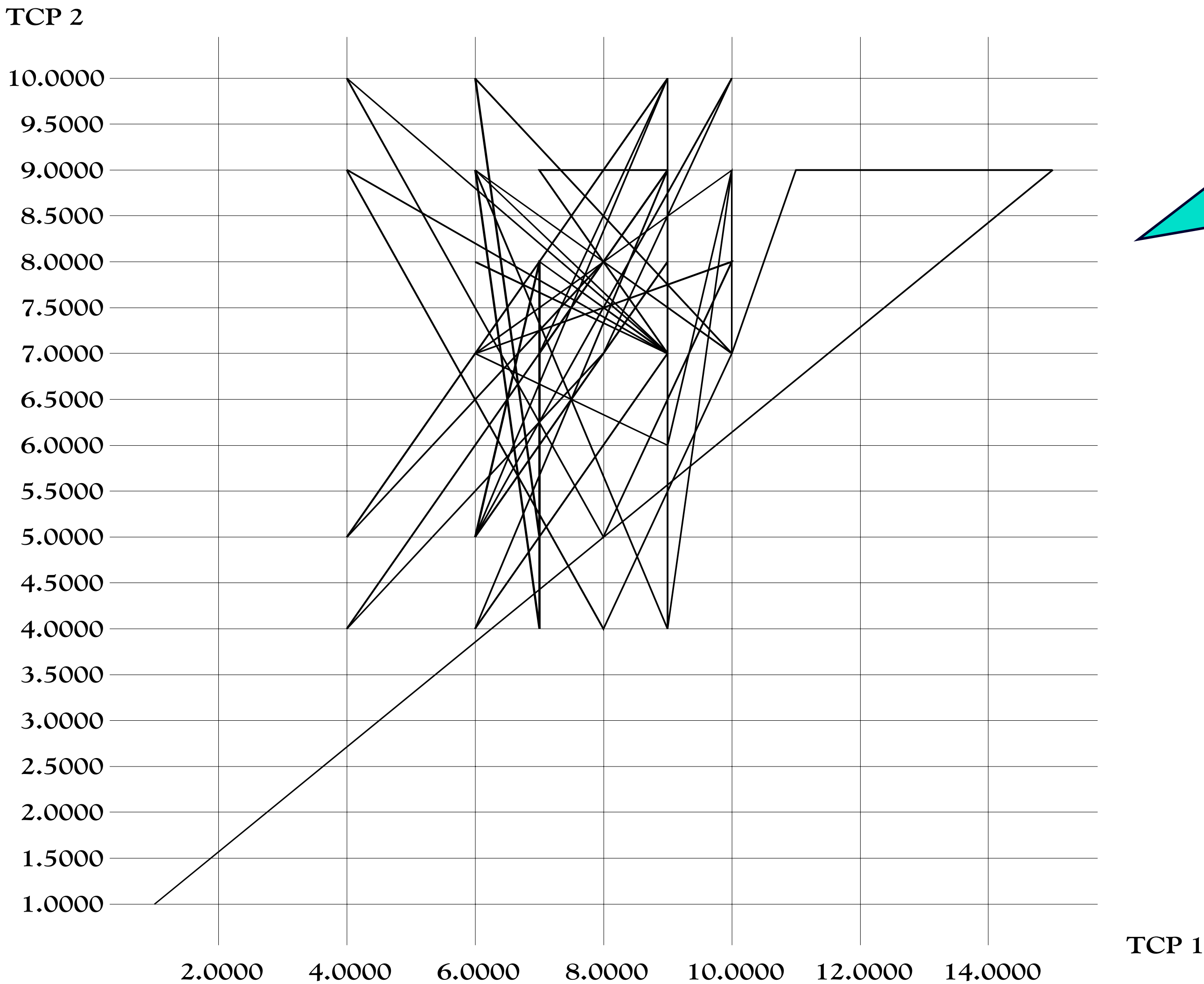
- Instead of just explaining a concept, design in the diagram space
  - Necessary simplifications may even be less dramatic!

How Stable is AIMD / async. RTT?



- Simple simulation (no queues, ..)
- RTT: 7 vs. 2
- AI=0.1, MD=0.5
- Simul. time=175

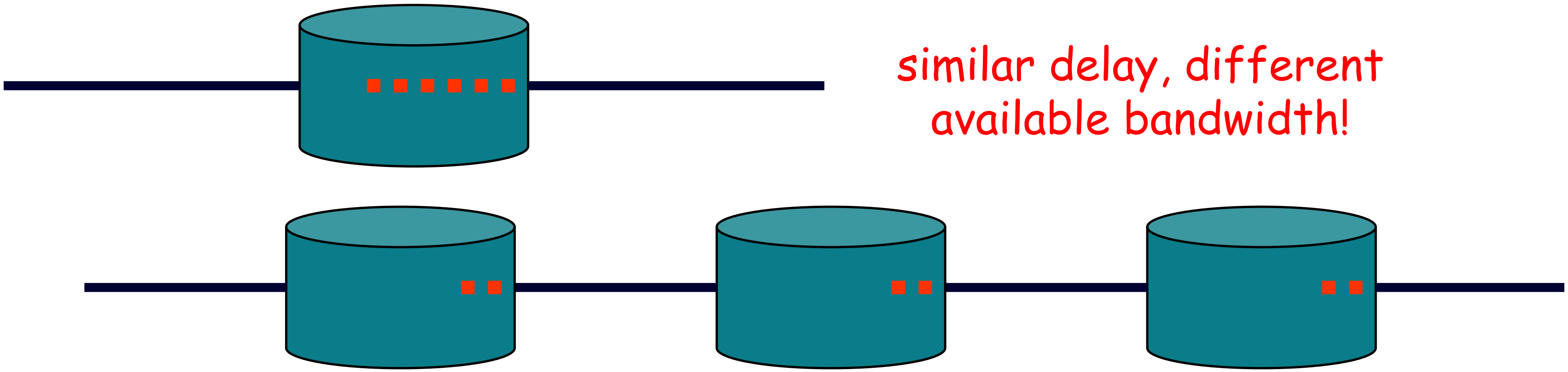
How is AIMD distorted in TCP?



- ns-2 simulator
- TCP Tahoe
- equal RTT
- 1 bottleneck link

Problems with TCP

- TCP over wireless: checksum error -> packet drop misinterpretation
- TCP over "long fat pipes": large bandwidth*delay product
  - long time to reach equilibrium, MD = dramatic!
- TCP reaches equilibrium, but not a stable point
  - fluctuations lead to regular packet drops & reduced throughput
  - fluctuations not feasible for streaming multimedia apps
- TCP Vegas: interpret delay as congestion ... but:



similar delay, different available bandwidth!

Enhancement idea: ATM ABR

- TCP -> TCP/RED -> TCP/RED/ECN -> TCP/RED/Multilevel ECN -> ...
- logical consequence: „ECN“ with fine granularity (explicitely ask for congestion information)

True for all mechanisms which use binary feedback!

- Routers know more about congestion. TCP-AI is like guessing and reacting when it is already too late.
- Idea related to ATM Available Bit Rate (ABR) service:
  - Resource management (RM) cells query the network for ECN flag & Explicit Rate information
  - framework for **sophisticated ER calculations** -> 1000s of papers

Often:
Fairness through flow counting
-> not scalable!

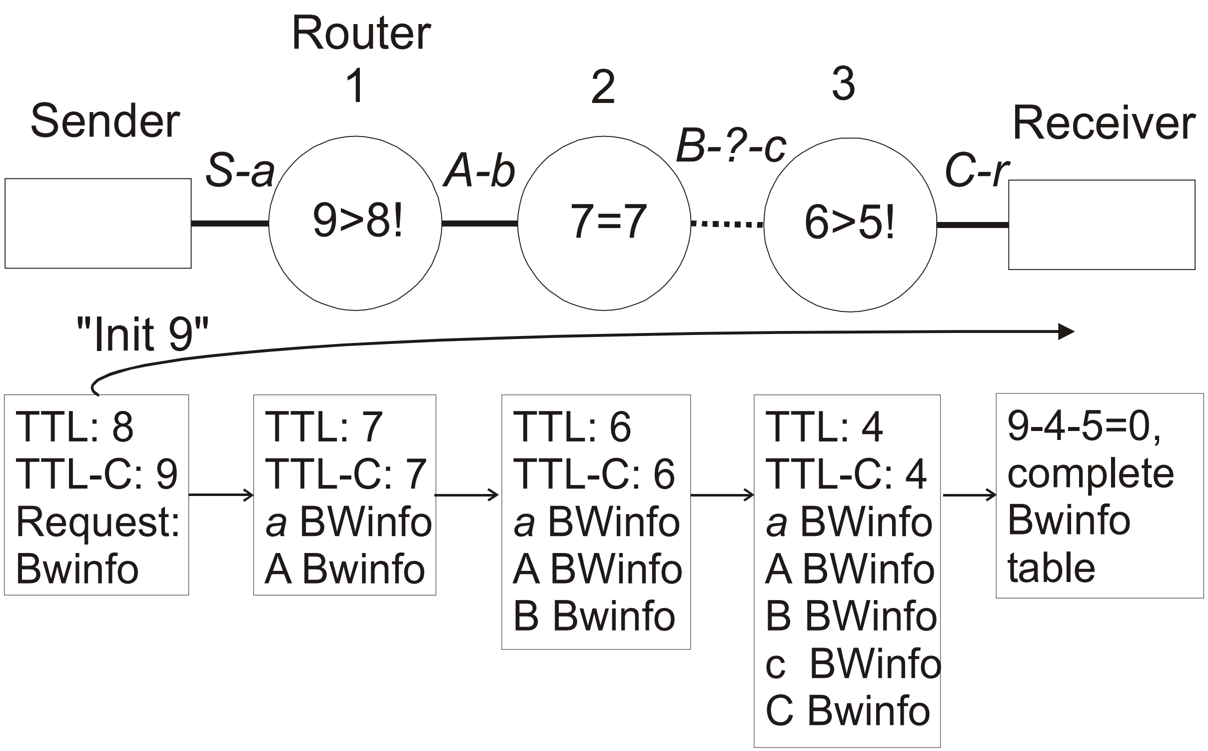
PTP: The Performance Transparency Protocol (draft-welzl-ptp-05.txt)

- "Generic" ECN / BECN - to carry traffic information (e.g. queue length, ..)
- **Stateless & simple -> scalable!**
- Only every 2nd router needed for full functionality
  - If less routers support it, results are an upper limit
- Available Bandwidth Determination:
  - nominal bandwidth ("ifSpeed") + 2* (address + traffic counter ("if(In/Out)Octets") + timestamp) = available bandwidth
- two modes: "forward packet stamping" / "direct reply" (not for available bandwidth (byte counters))

No problems w/ wireless links unless combined with packet loss!

PTP: How does it work?

- Forward packet stamping:



Upper case:
outgoing interface

Lower case:
incoming interface

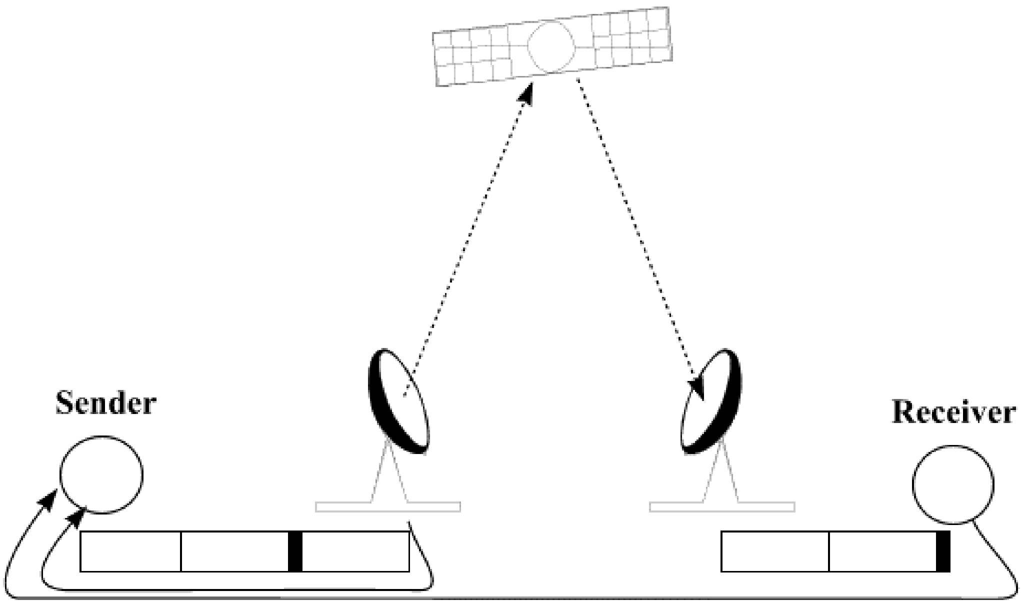
- The receiver can tell if the information is complete (DS Count, TTL)

PTP: How does it work? /2

- Direct reply:
  - Routers compare values; if requirements cannot be met...
    - the values are updated, the packet is redirected to the sender
  - Similar to RSVP reservation setup
  - Does not work with available bandwidth (traffic counter)
  - Advantageous for satellite links!

- Forward packet stamping satellite usage scenario:

  1st ground station acts as a receiver - only relies on PTP



Design algorithm

- find useful (closely related) ATM ABR mechanism
- start with simplifications, then expand the model
- A new mechanism must work for 2 users, equal RTT
  - simple analysis similar to Chiu/Jain (diagram + math)
- it must also work for heterogeneous RTTs
  - simulate using a simple diagram based simulator
  - analyze using extended vector diagram analysis
- it must also work for more users and more realistic scenarios
  - simulate with ns

The ATM ABR best match: CAPC

- "Congestion Avoidance with Proportional Control" (A. Barnhart 1994)
- Uses load factor LF: Input Rate IR / Target Rate R0
  - R0 e.g. 95% of nominal bandwidth, d = 1 - LF (available bandwidth)
- *"As long as the incoming rate is greater than R0, the desired rate, ERS will diminish at a rate that is proportional to the amount by which R0 is exceeded. Conversely, whenever the incoming rate is less than R0, ERS will increase."*
- for each new cell entering the queue:
  LF<=1: ERX = min(ERU, 1 + d*Rup) ... else ERX = max(ERF, 1 + d*Rdn)
  ERS = ERS*ERX
    - constants: Rup, Rdn define the speed of rate increase / decrease, ERU, ERF = upper / lower bound
    - different default values for LAN and WAN!

hint for RTT dependance!

Conversion for packet nets: CADPC

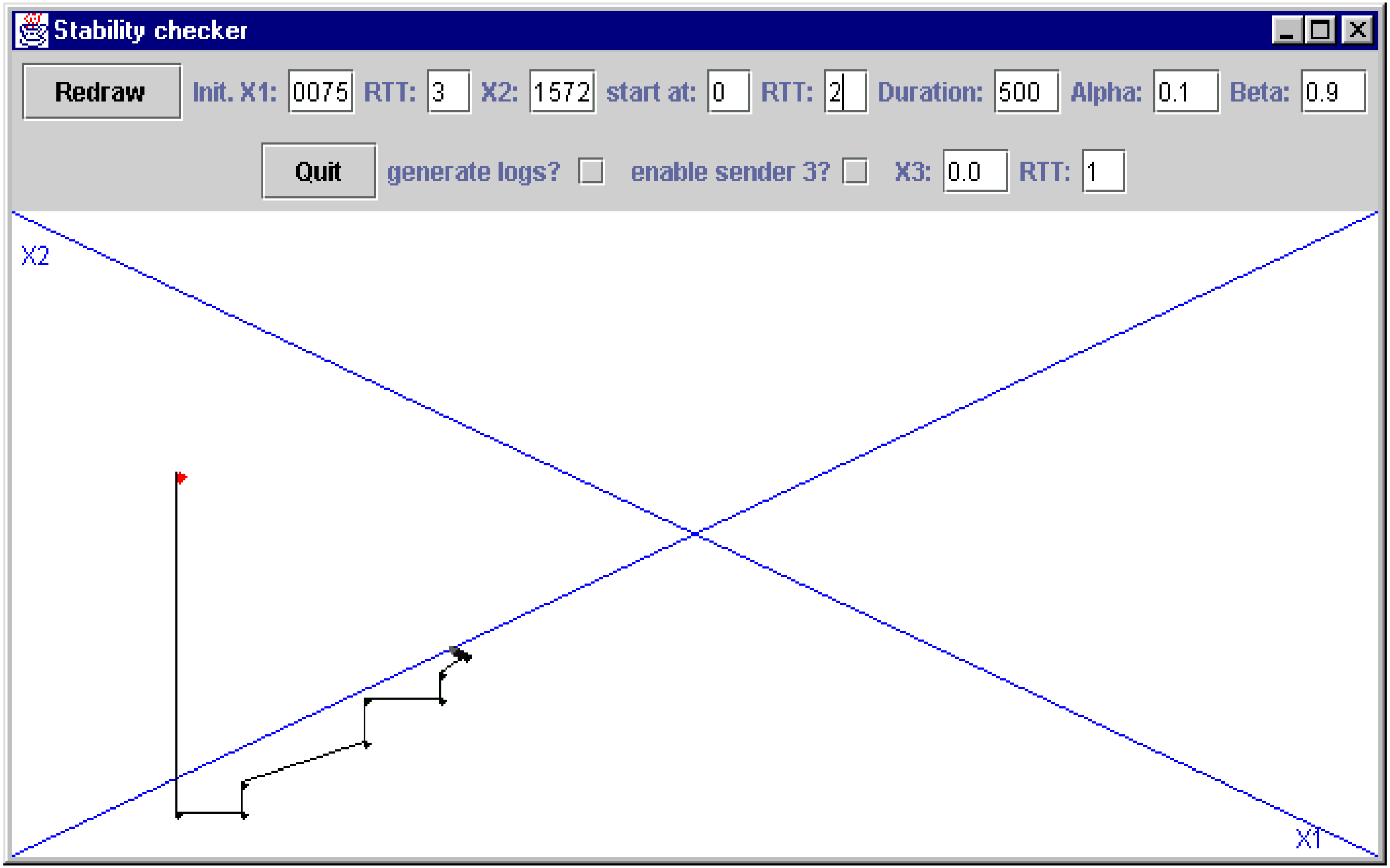
- “Congestion Avoidance with Distributed Proportional Control“
- Only ask for current load, do calculations at sender
- Fairness: sender 1 RTT = x * sender 2 RTT
  - sender 1 needs to calculate ERS x times -> ERS = ERS*(ERX^x)
- Result in Chiu-Jain-diagram-simulator:
  - max-min fairness approximated for limited rtt variations (trade-off between Rup, Rdn and allowed RTT's)
  - not good enough, but gives a hint that weighing the rate changes with the user's properties can lead to max-min fairness!

Note: **multiplicative!**

CADPC Design

- Idea:
  - relate user's current rate to the state of the system! (also in LDA+)
    Thought: in the Chiu-Jain-diagram, if the rate increase is indirectly proportional to the user's current rate, the rates will equalize.
- erx = 1 + rup * (1.0 - myRate/traffic)
  does not work! e.g. synchronous case -> myRate/traffic = constant!
- Solution:
  - erx = 1 + rup * (1.0 - myRate/d)
    relationship between user's rate and available rate keeps changing!
- Enhancement:
  - dependence on rup not desirable; rate changes should be proportional to the current load -> use d instead of rup!

CADPC vector diagram analysis



CADPC synchronous case analysis

- **Final formula per user:**
  d = traffic / r0;
  erx = 1 + d * (1.0 - myRate/d);
  ers = ers*erx;

- Combined:
  $x_i(t)$ = rate of user i, n users

$$x_i(t+1) = x_i(t)\left(1 + d\left[1 - \frac{x_i(t)}{1 - \frac{\sum_{j=1}^{n} x_j(t)}{r_0}}\right]\right)$$

(Special form of) logistic equation => **stable!**

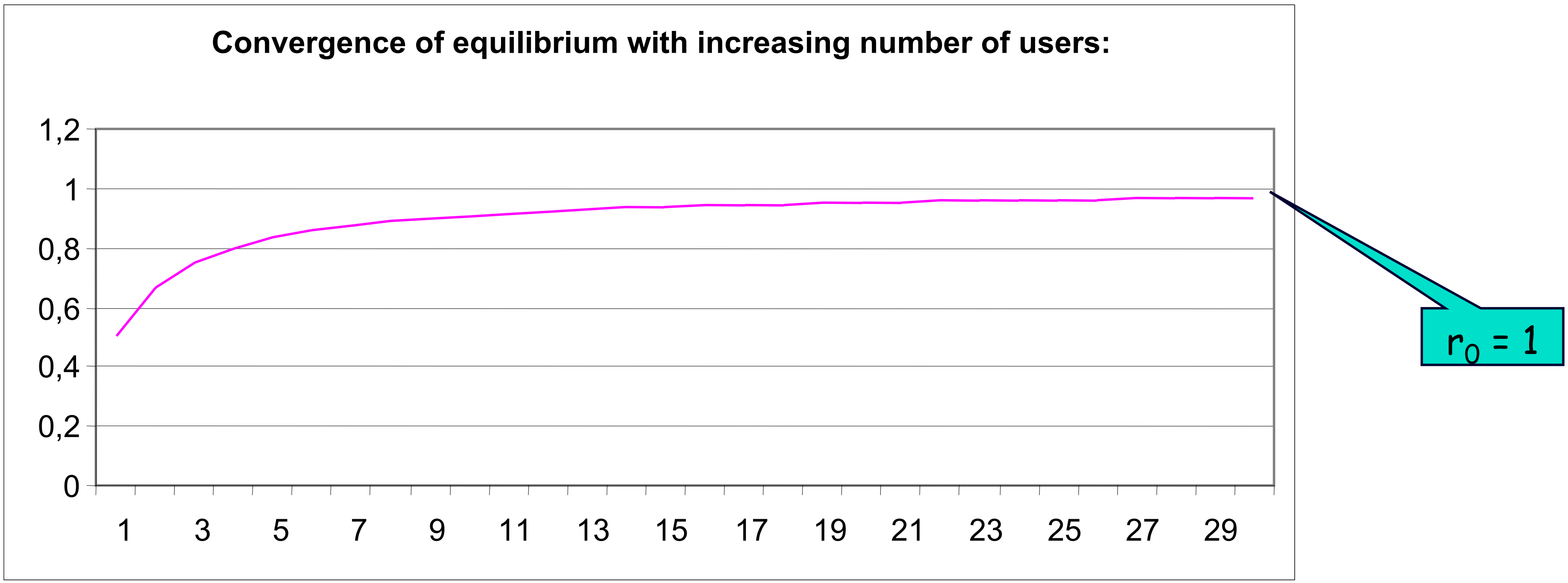
- after some straight-forward derivations:

$$x_i(t+1) = x_i(t)\left(r_0 + 1 - r_0 x_i(t) - \sum_{j=1}^{n} x_j(t)\right)$$

CADPC synchronous case analysis /2

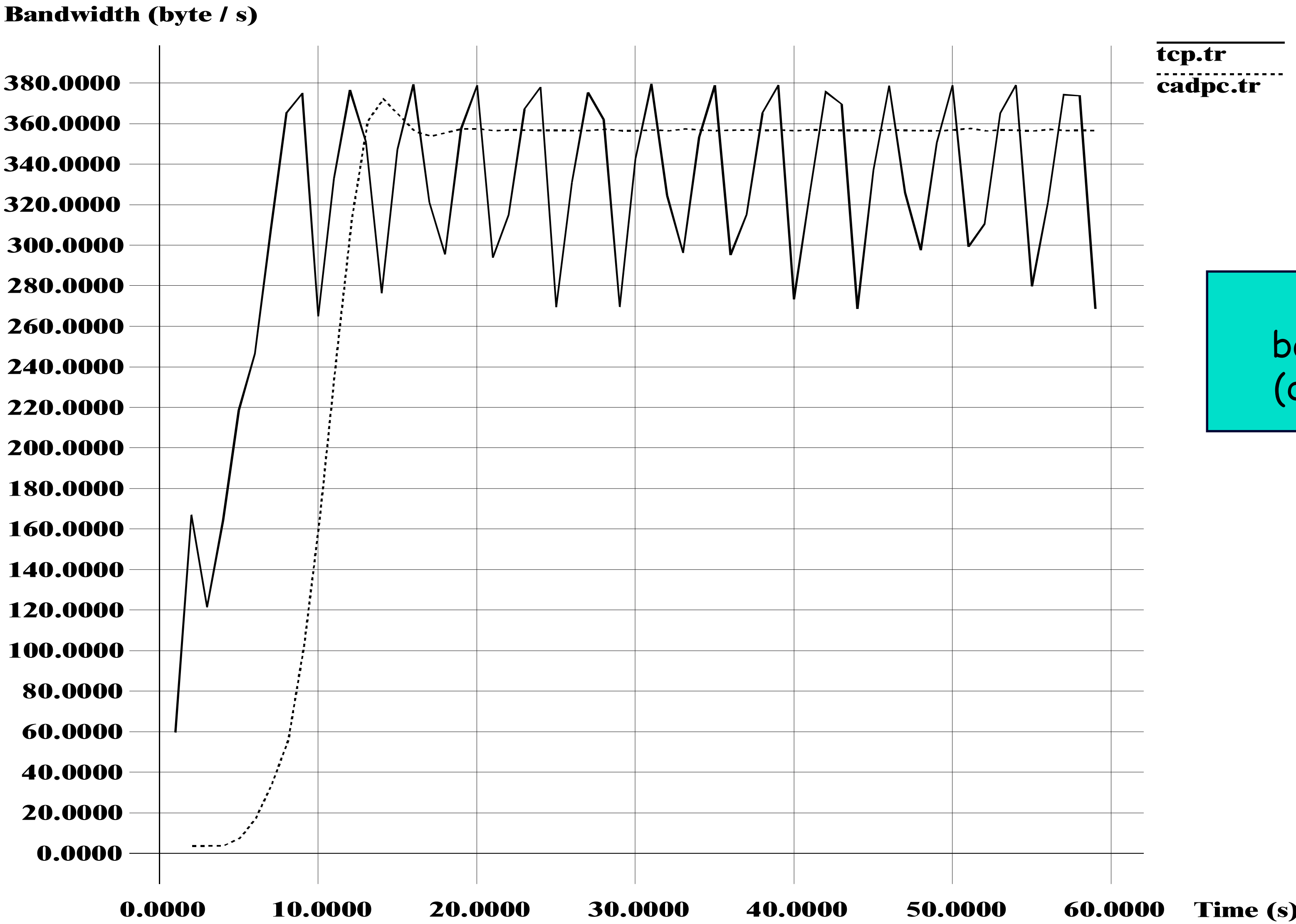
- Equilibrium: assume $x(t+1) = x(t)$

- leads to: $x(t) = r_0/(n+r_0)$
- traffic (n users): $n*x(t) = n*r_0/(n+r_0)$



ns simulation: 25 TCP / 25 CADPC

not simultaneously!



single bottleneck (dumbbell)

Results

- Implementation: r_0 normalized to 1 -> calc -> de-normalize
- 1 PTP packet every 4 RTTs, no other acks!
  - rate indeed converges to n/n+1
- **No** packet loss
- Very smooth rate, rapid convergence
- Not in the picture:
  - rapid convergence to almost perfect fairness
  - bg traffic: rapid backoff and recovery

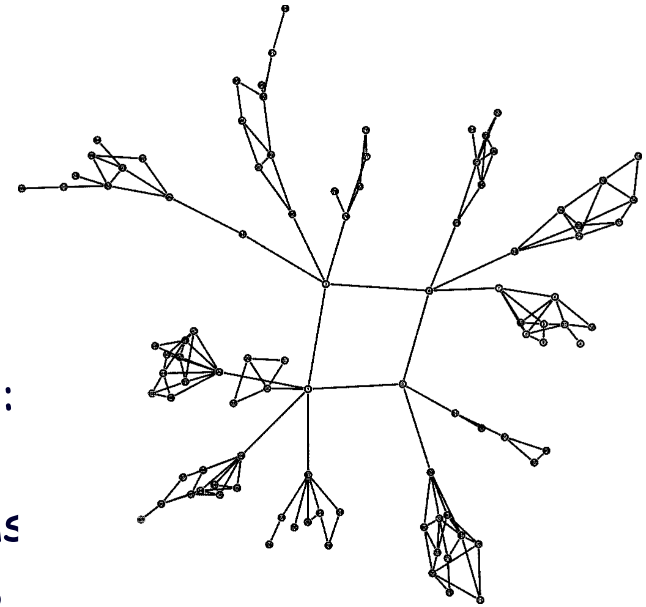
CADPC advantages

- Better stability than TCP
  - smooth rate advantageous for streaming media apps
- No problems with wireless links (no packet loss interpretation)
- Rare feedback - good in environments with long delay
  - rapid convergence & reaction - good in environments with a high bw*delay product
- Rate calculation independent of RTT => independent of position
  - **scalable!** if PTP = x% of generated traffic *n*, PTP scales O(*n*)
- Only (rare) PTP packets necessary to calculate rate
  - Satellite environments:
    do receiver's calculations at sat. base station and give earlier feedback
  - easier to differentiate pricing
  - easier to implement metering => traffic shaping, policing, admission control, ..

Deployment plans

- Problem: PTP needs router support
  - CADPC needs complete path information (every 2nd router)
- Possibilities:
  - QoS in a DiffServ class (QoS “in the small“):
    “we offer QoS & provide router support,
    you use CADPC and get a good result“
  - If CADPC works with non-greedy senders:
    edge2edge PTP signaling (TCP over CADPC)
    PTP supported traffic engineering
  - CADPC <=> TCP translation at edge routers?

Future work



- More ns simulations
  - CADPC vs. AIMD in vector diagram simulator: CADPC is much less aggressive
  - compare with TCP-friendly binary mechanisms
  - compare with other ER mechanisms PCP, ALS
- Extension to proportional fairness?
- CADPC implementation
  - PTP already available for Linux
  - compare with TCP, TFRC, RAP, ...
  - evaluate QoS