# Flow Control

An Engineering Approach to Computer Networking

# Flow control problem

- Consider file transfer
- Sender sends a stream of packets representing fragments of a file
- Sender should try to match rate at which receiver and network can process data
- Can't send too slow or too fast
- Too slow
  - wastes time
- Too fast
  - can lead to buffer overflow
- How to find the correct rate?

# Other considerations

- Simplicity
- Overhead
- Scaling
- Fairness
- Stability

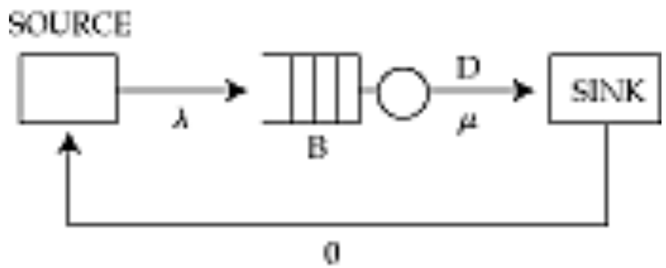
- Many interesting tradeoffs
  - overhead for stability
  - simplicity for unfairness

# Where?

- Usually at transport layer
- Also, in some cases, in datalink layer

# Model

- Source, sink, server, service rate, bottleneck, round trip time



# Classification

- Open loop
  - Source describes its desired flow rate
  - Network *admits* call
  - Source sends at this rate
- Closed loop
  - Source monitors available service rate
    - Explicit or implicit
  - Sends at this rate
  - Due to speed of light delay, errors are bound to occur
- Hybrid
  - Source asks for some minimum rate
  - But can send more, if available

# Open loop flow control

- Two phases to flow
  - Call setup
  - Data transmission
- Call setup
  - Network prescribes parameters
  - User chooses parameter values
  - Network admits or denies call
- Data transmission
  - User sends within parameter range
  - Network *polices* users
  - Scheduling policies give user QoS

# Hard problems

- Choosing a descriptor at a source
- Choosing a scheduling discipline at intermediate network elements
- Admitting calls so that their performance objectives are met (*call admission control*).

# Traffic descriptors

- Usually an *envelope*
  - Constrains worst case behavior
- Three uses
  - Basis for traffic contract
  - Input to *regulator*
  - Input to *policer*

# Descriptor requirements

- Representativity
  - adequately describes flow, so that network does not reserve too little or too much resource
- Verifiability
  - verify that descriptor holds
- Preservability
  - Doesn’t change inside the network
- Usability
  - Easy to describe and use for admission control

# Examples

- Representative, verifiable, but not useble
  - Time series of interarrival times
- Verifiable, preservable, and useable, but not representative
  - peak rate

# Some common descriptors

- Peak rate
- Average rate
- Linear bounded arrival process (LBAP)

# Peak rate

- Highest ‘rate’ at which a source can send data
- Two ways to compute it
- For networks with fixed-size packets
  - min inter-packet spacing
- For networks with variable-size packets
  - highest rate over *all* intervals of a particular duration
- Regulator for fixed-size packets
  - timer set on packet transmission
  - if timer expires, send packet, if any
- Problem
  - sensitive to extremes

# Average rate

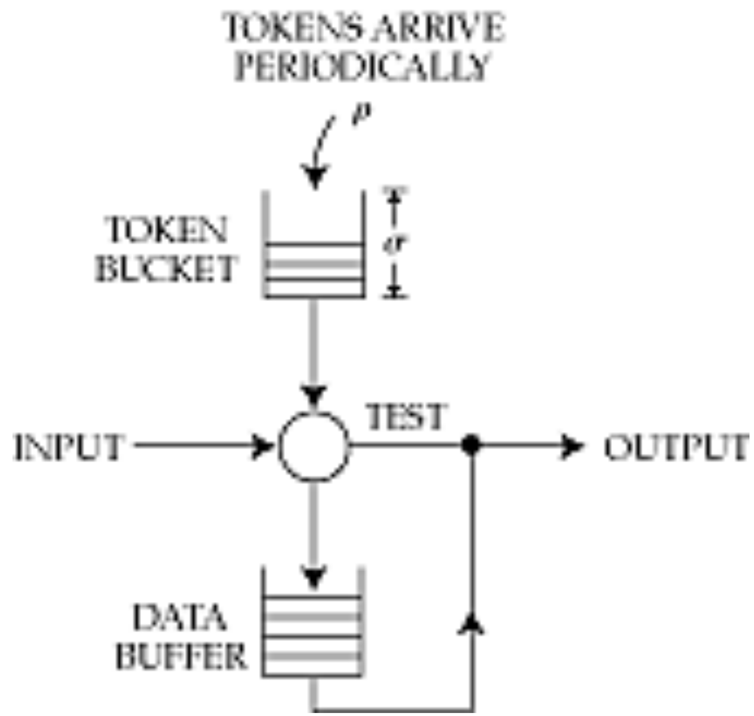
- Rate over some time period (*window*)
- Less susceptible to outliers
- Parameters: *t* and *a*
- Two types: jumping window and moving window
- Jumping window
  - over consecutive intervals of length *t*, only *a* bits sent
  - regulator reinitializes every interval
- Moving window
  - over all intervals of length *t*, only *a* bits sent
  - regulator forgets packet sent more than *t* seconds ago

# Linear Bounded Arrival Process

- Source bounds # bits sent in any time interval by a linear function of time
- the number of bits transmitted in any active interval of length $t$ is less than $rt + s$
- $r$ is the long term rate
- $s$ is the burst limit
- insensitive to outliers

# Leaky bucket

- A regulator for an LBAP
- Token bucket fills up at rate *r*
- Largest # tokens < *s*



# Variants

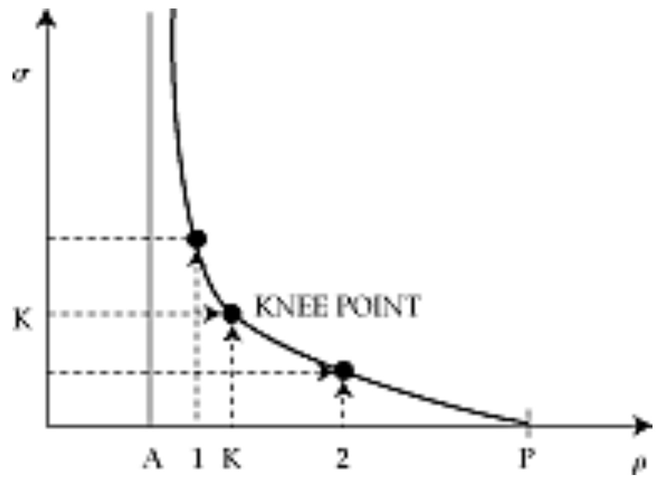
- Token and data buckets
  - Sum is what matters
- Peak rate regulator

# Choosing LBAP parameters

- Tradeoff between *r* and *s*
- Minimal descriptor
  - doesn't simultaneously have smaller *r* and *s*
  - presumably costs less
- How to choose minimal descriptor?
- Three way tradeoff
  - choice of *s* (data bucket size)
  - loss rate
  - choice of *r*

# Choosing minimal parameters

- Keeping loss rate the same
  - if *s* is more, *r* is less (smoothing)
  - for each *r* we have least *s*
- Choose knee of curve



# LBAP

- Popular in practice and in academia
  - sort of representative
  - verifiable
  - sort of preservable
  - sort of usable
- Problems with multiple time scale traffic
  - large burst messes up things

# Open loop vs. closed loop

- Open loop
  - describe traffic
  - network admits/reserves resources
  - regulation/policing
- Closed loop
  - can’t describe traffic or network doesn’t support reservation
  - monitor available bandwidth
    - perhaps allocated using emulation of Generalized Processor Sharing (GPS - see later under Scheduling)
  - adapt to it
  - if not done properly either
    - too much loss
    - unnecessary delay

# Taxonomy

- First generation
  - ignores network state
  - only match receiver
- Second generation
  - responsive to state
  - three choices
    - State measurement
      - explicit or implicit
    - Control
      - flow control window size or rate
    - Point of control
      - endpoint or within network

# Explicit vs. Implicit

- Explicit
  - Network tells source its current rate
  - Better control
  - More overhead
- Implicit
  - Endpoint figures out rate by looking at network
  - Less overhead
- Ideally, want overhead of implicit with effectiveness of explicit

# Flow control window

- Recall error control window
- Largest number of packet outstanding (sent but not acked)
- If endpoint has sent all packets in window, it must wait => slows down its rate
- Thus, window provides *both* error control and flow control
- This is called *transmission* window
- Coupling can be a problem
  - Few buffers at receiver => slow rate!

# Window vs. rate

- In adaptive rate, we directly control rate
- Needs a timer per connection
- Plusses for window
  - no need for fine-grained timer
  - self-limiting
- Plusses for rate
  - better control (finer grain)
  - no coupling of flow control and error control
- Rate control must be careful to avoid overhead and sending too much

# Hop-by-hop vs. end-to-end

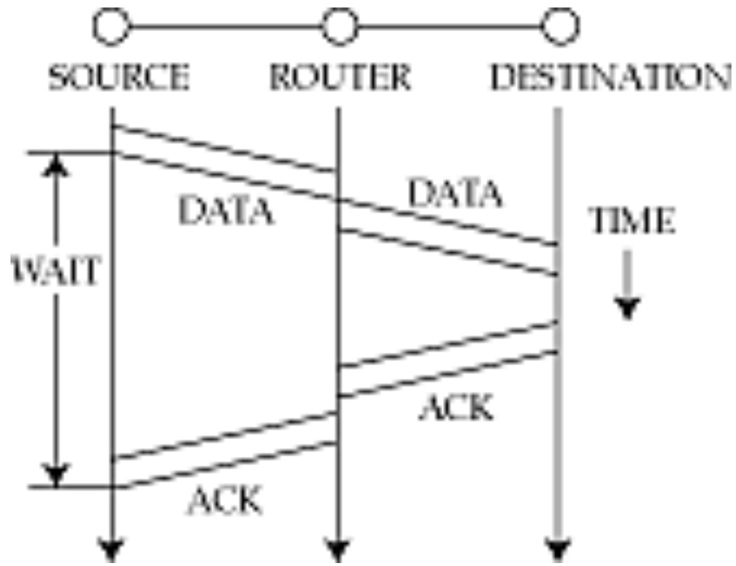
- Hop-by-hop
  - first generation flow control at each link
    - next server = sink
  - easy to implement
- End-to-end
  - sender matches all the servers on its path
- Plusses for hop-by-hop
  - simpler
  - distributes overflow
  - better control
- Plusses for end-to-end
  - cheaper

# On-off

- Receiver gives ON and OFF signals
- If ON, send at full speed
- If OFF, stop
- OK when RTT is small
- What if OFF is lost?
- Bursty
- Used in serial lines or LANs

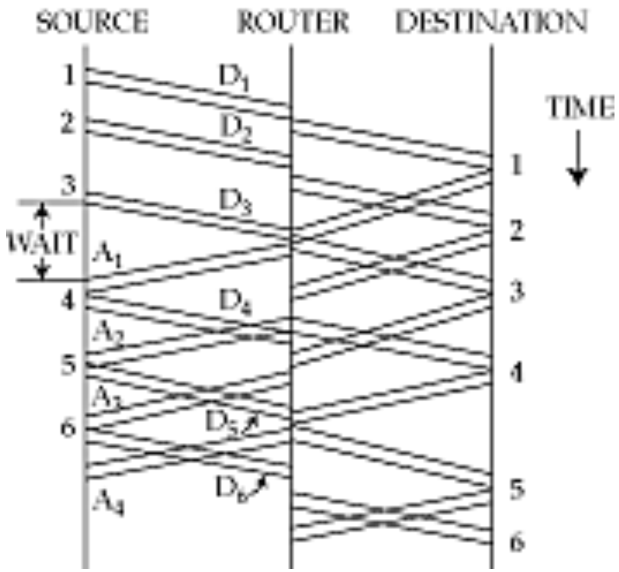
# Stop and Wait

- Send a packet
- Wait for ack before sending next packet



# Static window

- Stop and wait can send at most one pkt per RTT
- Here, we allow multiple packets per RTT (= transmission window)



# What should window size be?

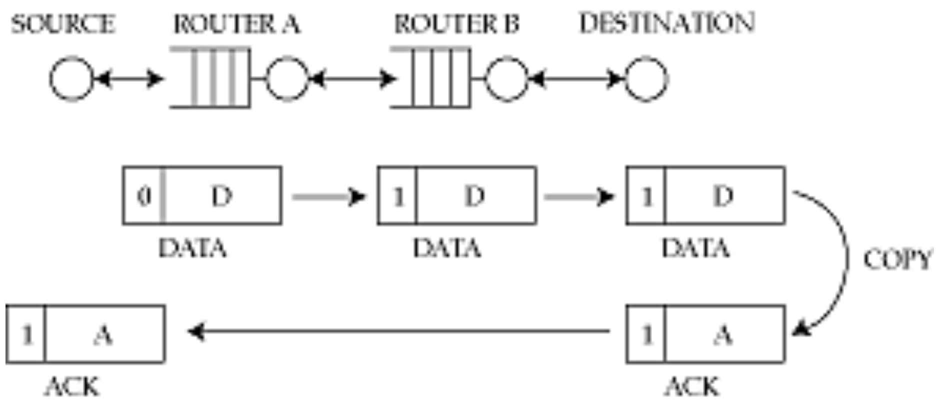
- Let bottleneck service rate along path = b pkts/sec
- Let round trip time = R sec
- Let flow control window = w packet
- Sending rate is w packets in R seconds = w/R
- To use bottleneck w/R > b => w > bR
- This is the *bandwidth delay product* or *optimal window size*

# Static window

- Works well if b and R are fixed
- But, bottleneck rate changes with time!
- Static choice of w can lead to problems
  - too small
  - too large
- So, need to adapt window
- Always try to get to the *current* optimal value

# DECbit flow control

- Intuition
  - every packet has a bit in header
  - intermediate routers set bit if queue has built up => source window is too large
  - sink copies bit to ack
  - if bits set, source reduces window size
  - in steady state, oscillate around optimal size

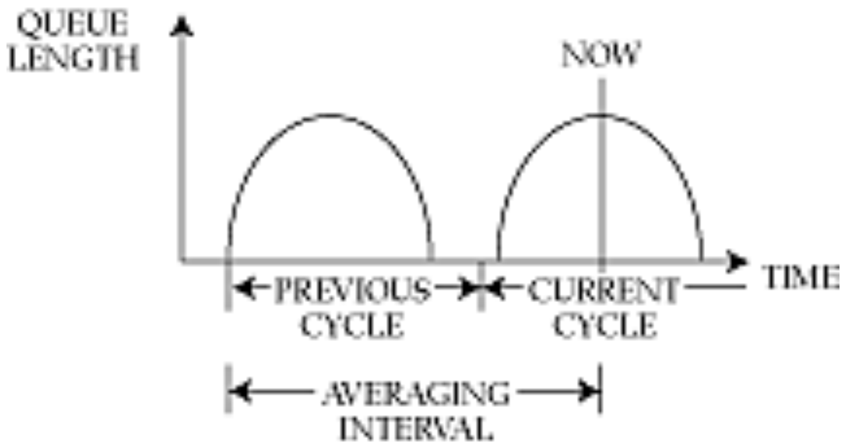


# DECbit

- When do bits get set?
- How does a source interpret them?

# DECbit details: router actions

- Measure *demand* and *mean queue length* of each source
- Computed over queue regeneration cycles
- Balance between sensitivity and stability



# Router actions

- If mean queue length > 1.0
  - set bits on sources whose demand exceeds fair share
- If it exceeds 2.0
  - set bits on everyone
  - panic!

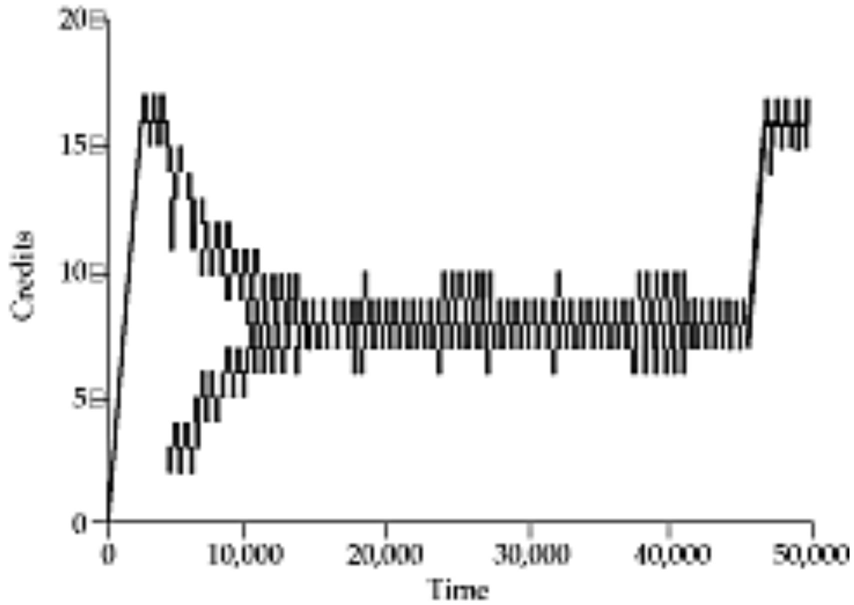
# Source actions

- Keep track of bits
- Can’t take control actions too fast!
- Wait for past change to take effect
- Measure bits over past + present window size
- If more than 50% set, then decrease window, else increase
- Additive increase, multiplicative decrease

# Evaluation

- Works with FIFO
  - but requires per-connection state (demand)
- Software
- But
  - assumes cooperation!
  - conservative window increase policy

# Sample trace



# TCP Flow Control

- Implicit
- Dynamic window
- End-to-end

- Very similar to DECbit, but
  - no support from routers
  - increase if no loss (usually detected using timeout)
  - window decrease on a timeout
  - additive increase multiplicative decrease

# TCP details

- Window starts at 1
- Increases exponentially for a while, then linearly
- Exponentially => doubles every RTT
- Linearly => increases by 1 every RTT
- During exponential phase, every ack results in window increase by 1
- During linear phase, window increases by 1 when # acks = window size
- Exponential phase is called *slow start*
- Linear phase is called *congestion avoidance*

# More TCP details

- On a loss, current window size is stored in a variable called *slow start threshold* or *ssthresh*
- Switch from exponential to linear (slow start to congestion avoidance) when window size reaches threshold
- Loss detected either with timeout or *fast retransmit* (duplicate cumulative acks)
- Two versions of TCP
  - Tahoe: in both cases, drop window to 1
  - Reno: on timeout, drop window to 1, and on fast retransmit drop window to half previous size (also, increase window on subsequent acks)

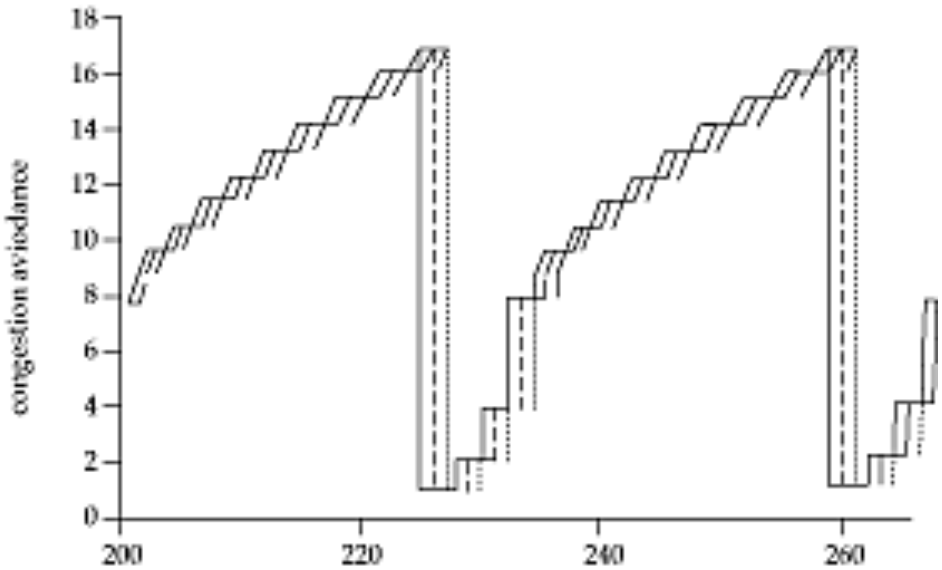
# TCP vs. DECbit

- Both use dynamic window flow control and additive-increase multiplicative decrease
- TCP uses implicit measurement of congestion
  - probe a black box
- Operates at the *cliff*
- Source does not filter information

# Evaluation

- Effective over a wide range of bandwidths
- A lot of operational experience
- Weaknesses
  - loss => overload? (wireless)
  - overload => self-blame, problem with FCFS
  - overload detected only on a loss
    - in steady state, source *induces* loss
  - needs at least bR/3 buffers per connection

# Sample trace



# TCP Vegas

- Expected throughput = transmission_window_size/propagation_delay
- Numerator: known
- Denominator: measure *smallest* RTT
- Also know *actual* throughput
- Difference = how much to reduce/increase rate
- Algorithm
  - send a special packet
  - on ack, compute expected and actual throughput
  - (expected - actual)* RTT packets in bottleneck buffer
  - adjust sending rate if this is too large
- Works better than TCP Reno

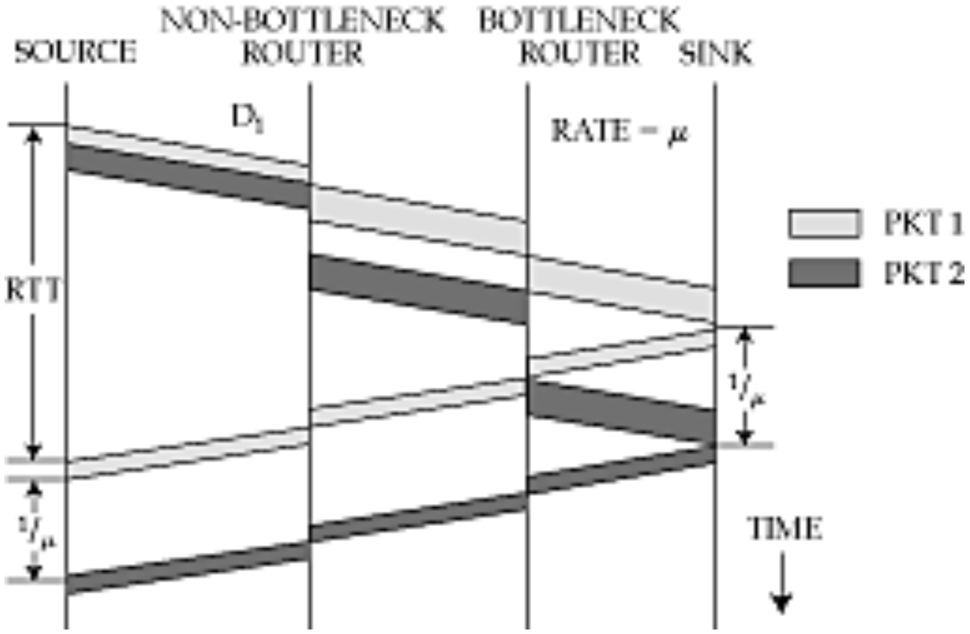
# NETBLT

- First rate-based flow control scheme
- Separates error control (window) and flow control (no *coupling*)
- So, losses and retransmissions do not affect the flow rate
- Application data sent as a series of buffers, each at a particular rate
- Rate = (burst size + burst rate) so granularity of control = burst
- Initially, no adjustment of rates
- Later, if received rate < sending rate, multiplicatively decrease rate
- Change rate only once per buffer => slow

# Packet pair

- Improves basic ideas in NETBLT
  - better measurement of bottleneck
  - control based on prediction
  - finer granularity
- Assume all bottlenecks serve packets in round robin order
- Then, spacing between packets at receiver (= ack spacing) = 1/(rate of slowest server)
- If *all* data sent as paired packets, no distinction between data and probes
- Implicitly determine service rates if servers are round-robin-like

# Packet pair


SOURCE
NON-BOTTLENECK
ROUTER
BOTTLENECK
ROUTER
SINK
$D_1$
RATE = $\mu$
RTT
$1/\mu$
$1/\mu$
PKT 1
PKT 2
TIME

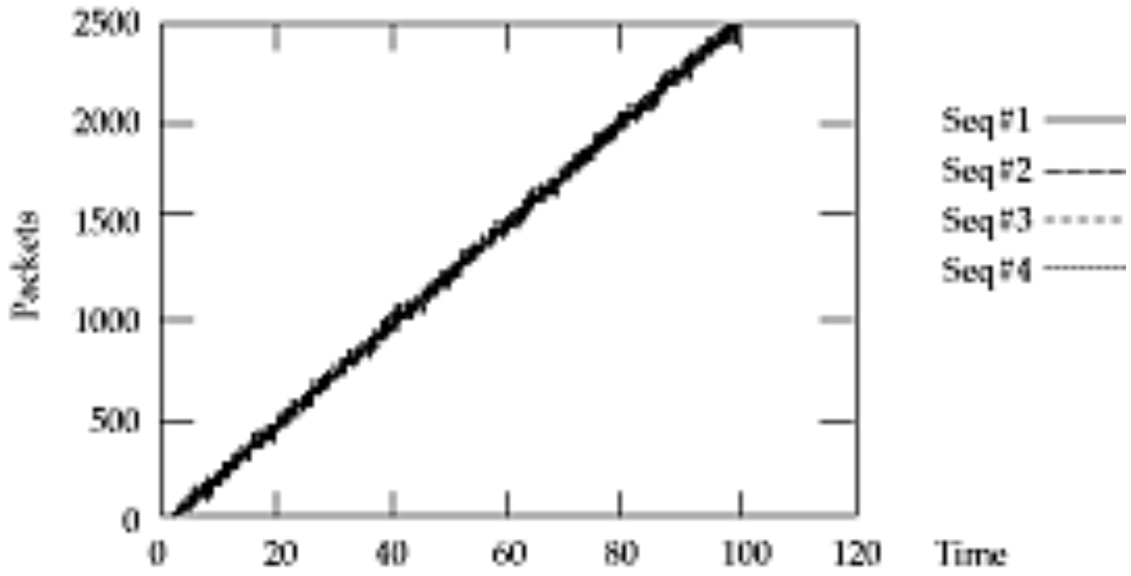
# Packet-pair details

- Acks give time series of service rates in the past
- We can use this to predict the next rate
- Exponential averager, with fuzzy rules to change the averaging factor
- Predicted rate feeds into flow control equation

# Packet-pair flow control

- Let X = # packets in bottleneck buffer
- S = # outstanding packets
- R = RTT
- b = bottleneck rate
- Then, X = S - Rb (assuming no losses)
- Let l = source rate
- l(k+1) = b(k+1) + (setpoint -X)/R

# Sample trace



# Comparison among closed-loop schemes

- On-off, stop-and-wait, static window, DECbit, TCP, NETBLT, Packet-pair
- Which is best? No simple answer
- Some rules of thumb
  - flow  control easier with RR scheduling
    - otherwise, assume cooperation, or police rates
  - explicit schemes are more robust
  - hop-by-hop schemes are more resposive, but more comples
  - try to separate error control and flow control
  - rate based schemes are inherently unstable unless well-engineered

# Hybrid flow control

- Source gets a minimum rate, but can use more
- All problems of both open loop and closed loop flow control
- Resource partitioning problem
  - what fraction can be reserved?
  - how?