

Internet Transport Tomorrow: Introducing SCTP, UDP-Lite and DCCP

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Outline

1. Internet transport today: too much, or not enough
2. Internet transport tomorrow
 1. SCTP
 2. UDP-Lite
 3. DCCP

Transport layer problem statement

- Efficient transmission of data streams across the Internet
 - various sources, various destinations, various types of streams
- What is “efficient“?
 - terms: latency, end2end delay, jitter, bandwidth (nominal/available/bottleneck -), throughput, goodput, loss ratio, ..
 - general goals: high throughput (bits / second), low delay, jitter, loss ratio
- Note: Internet = TCP/IP based world-wide network
 - no assumptions about lower layers!
 - ignore CSMA/CD, CSMA/CA, token ring, baseband encoding, frame overhead, switches, etc. etc. !

Internet transport today: one size fits all

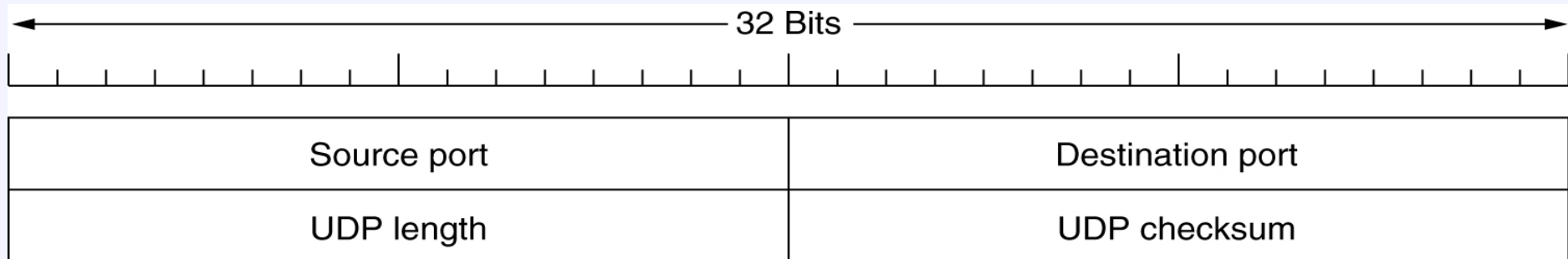
- UDP used for sporadic messages (DNS) and some special apps
- TCP used for everything else
 - in 2003, approximately 83 % according to:
Marina Fomenkov, Ken Keys, David Moore and k claffy, "Longitudinal study of Internet traffic in 1998-2003", CAIDA technical report, available from <http://www.caida.org/outreach/papers/2003/nlanr/>
 - backbone measurement from 2000 said 98% ⇒ UDP usage growing
- Original Internet proposition:
IP over everything, everything over IP
- Today's reality:
IP over everything, almost everything over TCP, and the rest over UDP

What TCP does for you (roughly)

- UDP features: multiplexing + protection against corruption
 - ports, checksum
- stream-based in-order delivery
 - segments are ordered according to sequence numbers
 - only consecutive bytes are delivered
- reliability
 - missing segments are detected (ACK is missing) and retransmitted
- flow control
 - receiver is protected against overload (window based)
- congestion control
 - network is protected against overload (window based)
 - protocol tries to fill available capacity
- connection handling
 - explicit establishment + teardown
- full-duplex communication
 - e.g., an ACK can be a data segment at the same time (piggybacking)

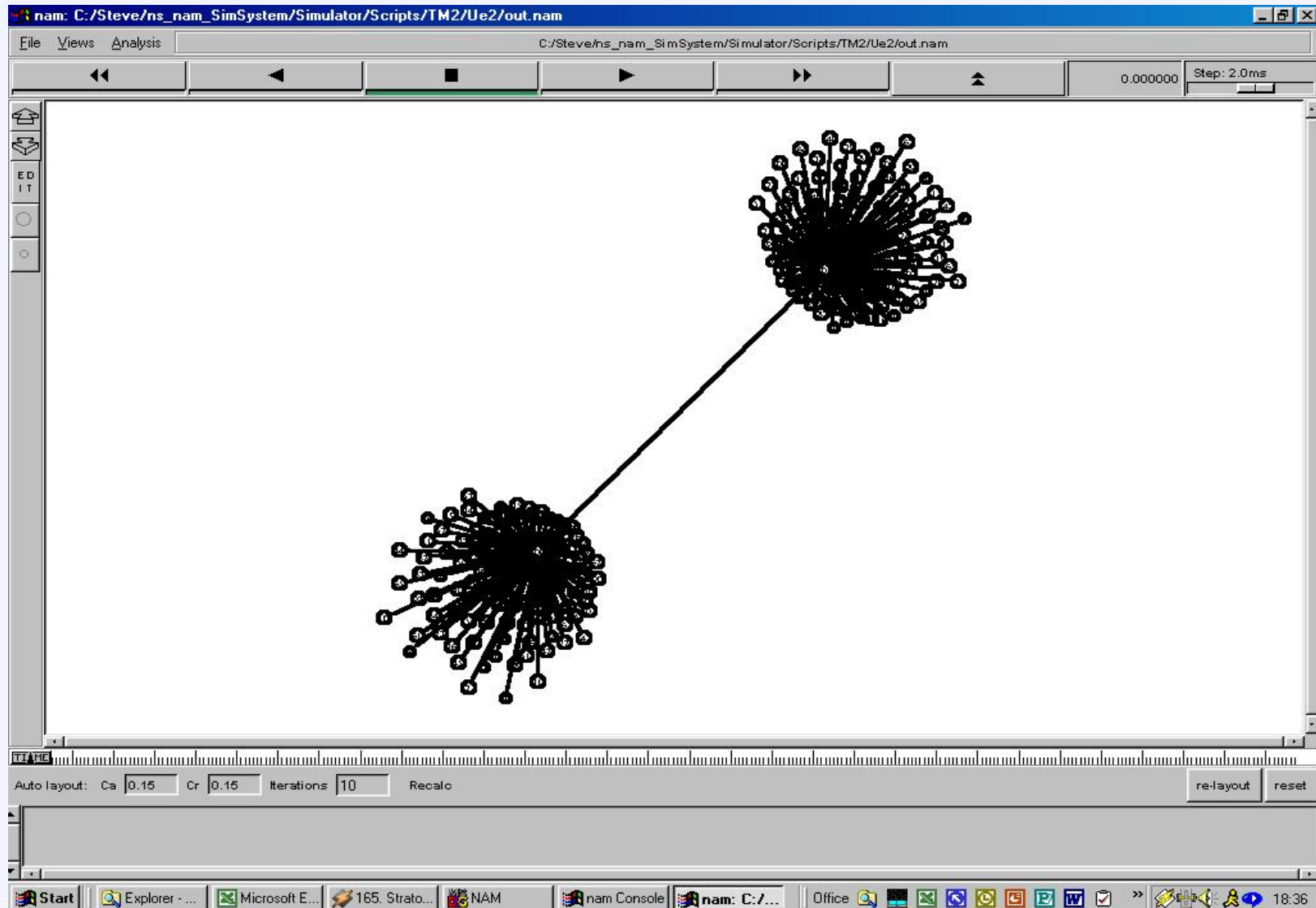
Are all these features
always appropriate?

UDP, however...

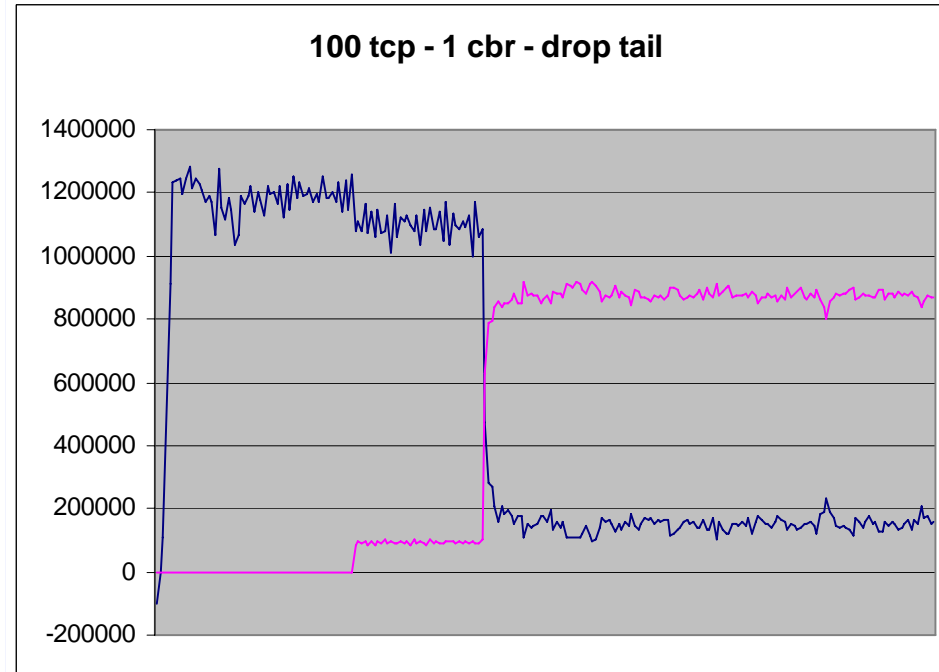
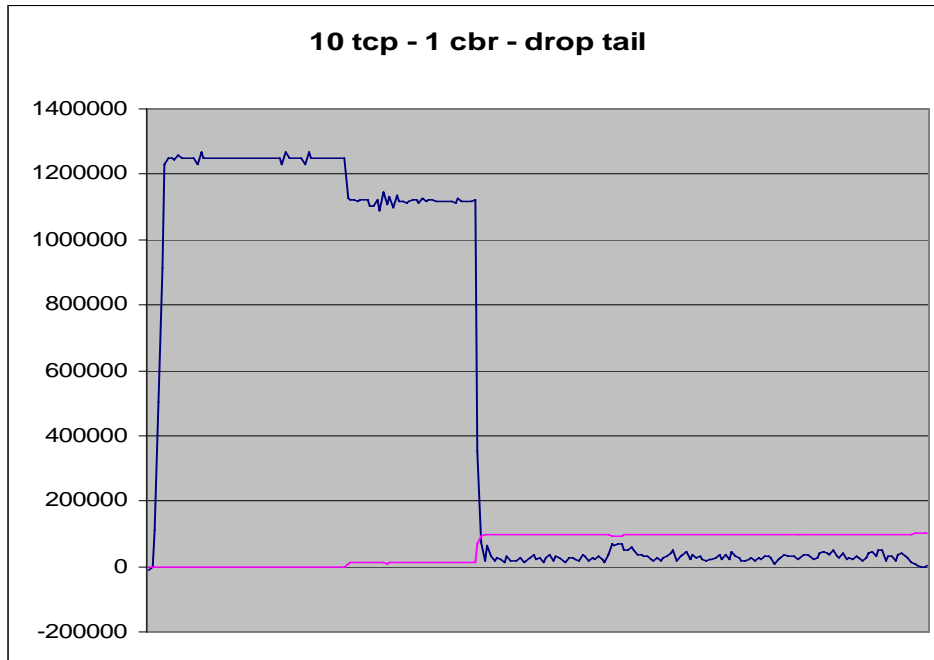


- RFC 768: **three pages!**
- IP + 2 features:
 - Multiplexing (ports)
 - Checksum
- Used by apps which want **unreliable, timely delivery**
 - e.g. VoIP: significant delay = ☹ ... but some noise = ☺
- **No congestion control**
 - fine for SNMP, DNS, ..

TCP vs. UDP: a simple simulation example

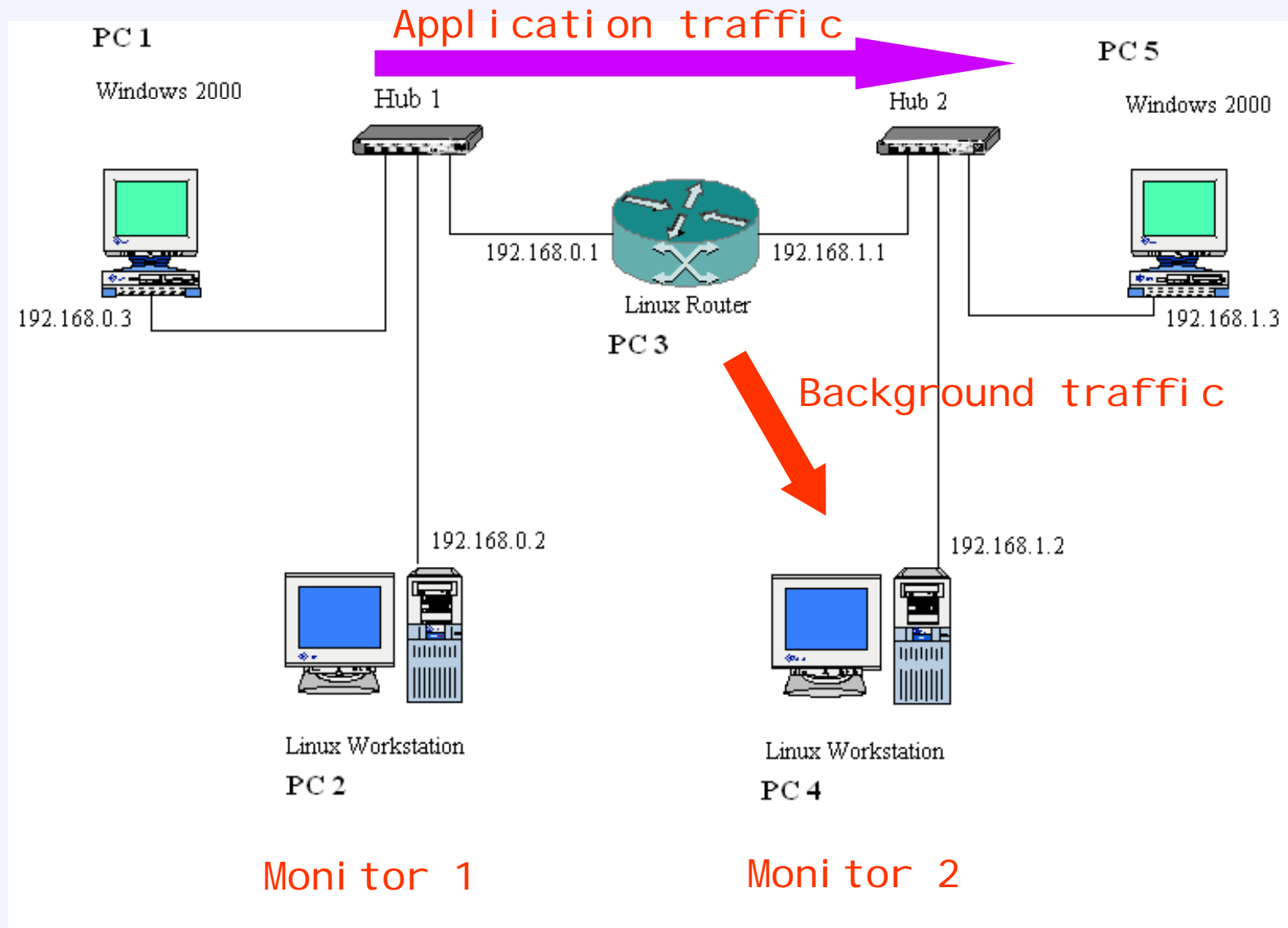


It doesn't look good

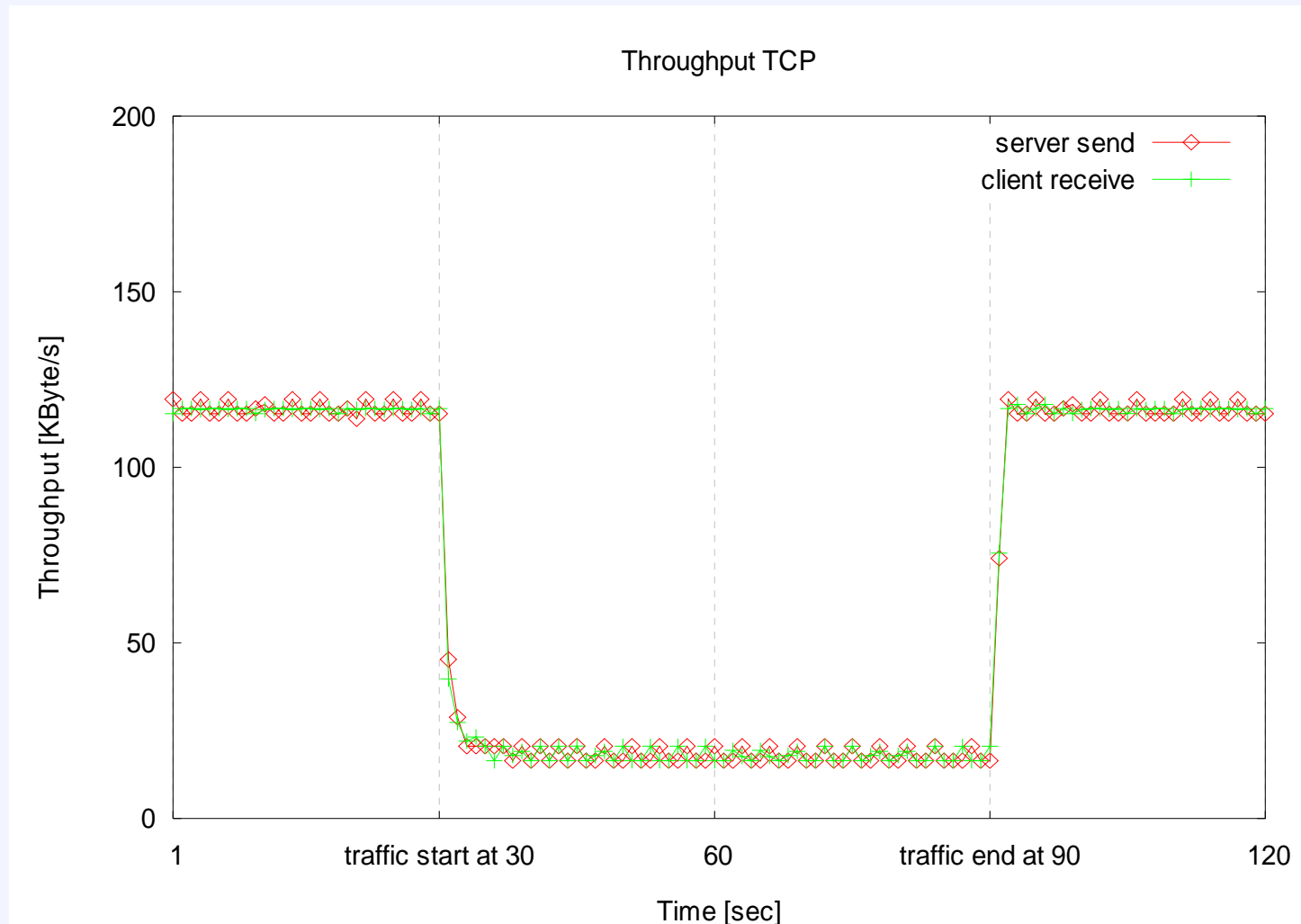


- For more details, see:
Promoting the Use of End-to-End Congestion Control in the Internet.
Floyd, S., and Fall, K..
IEEE/ACM Transactions on Networking, August 1999.

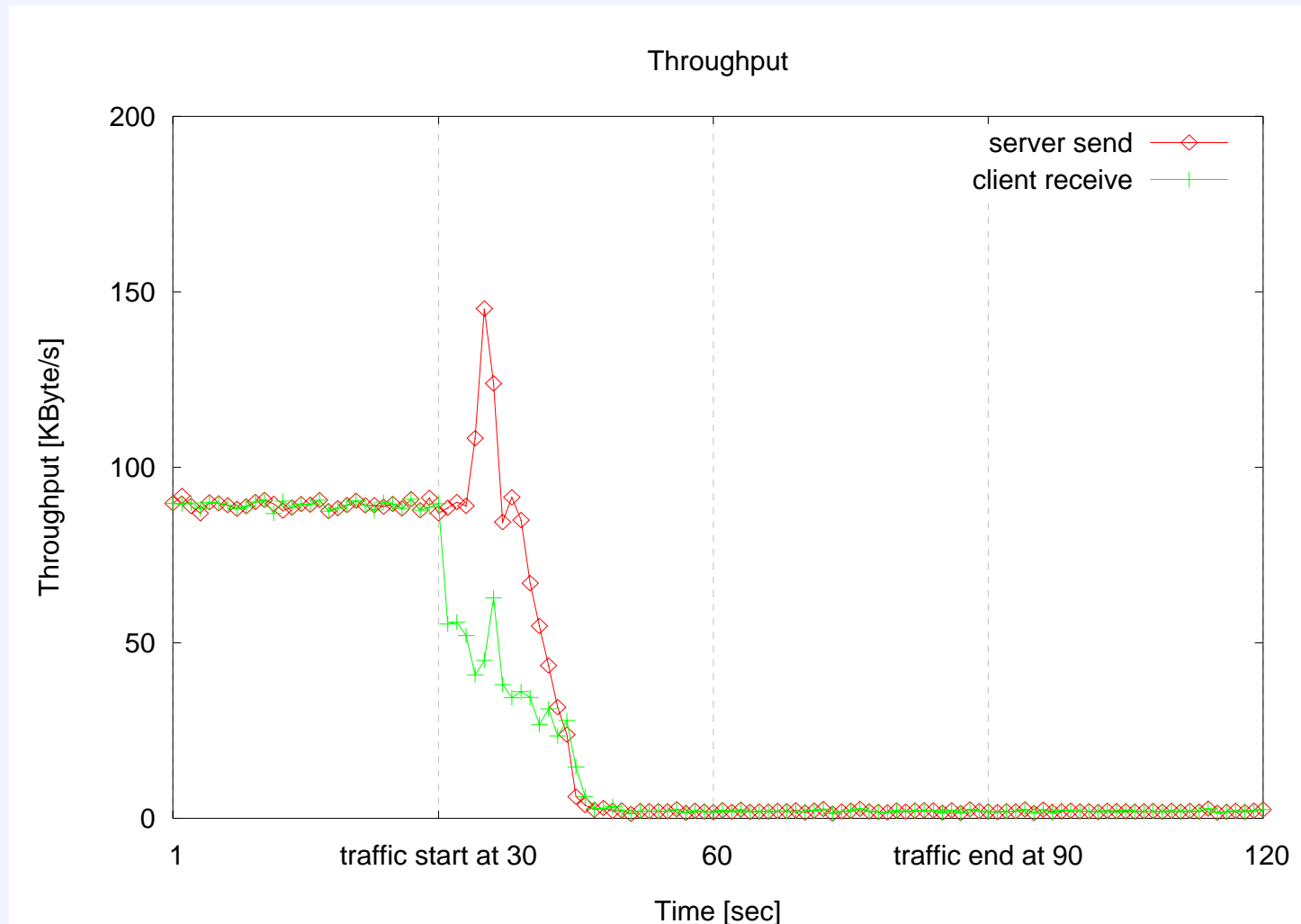
Real behavior of today's apps



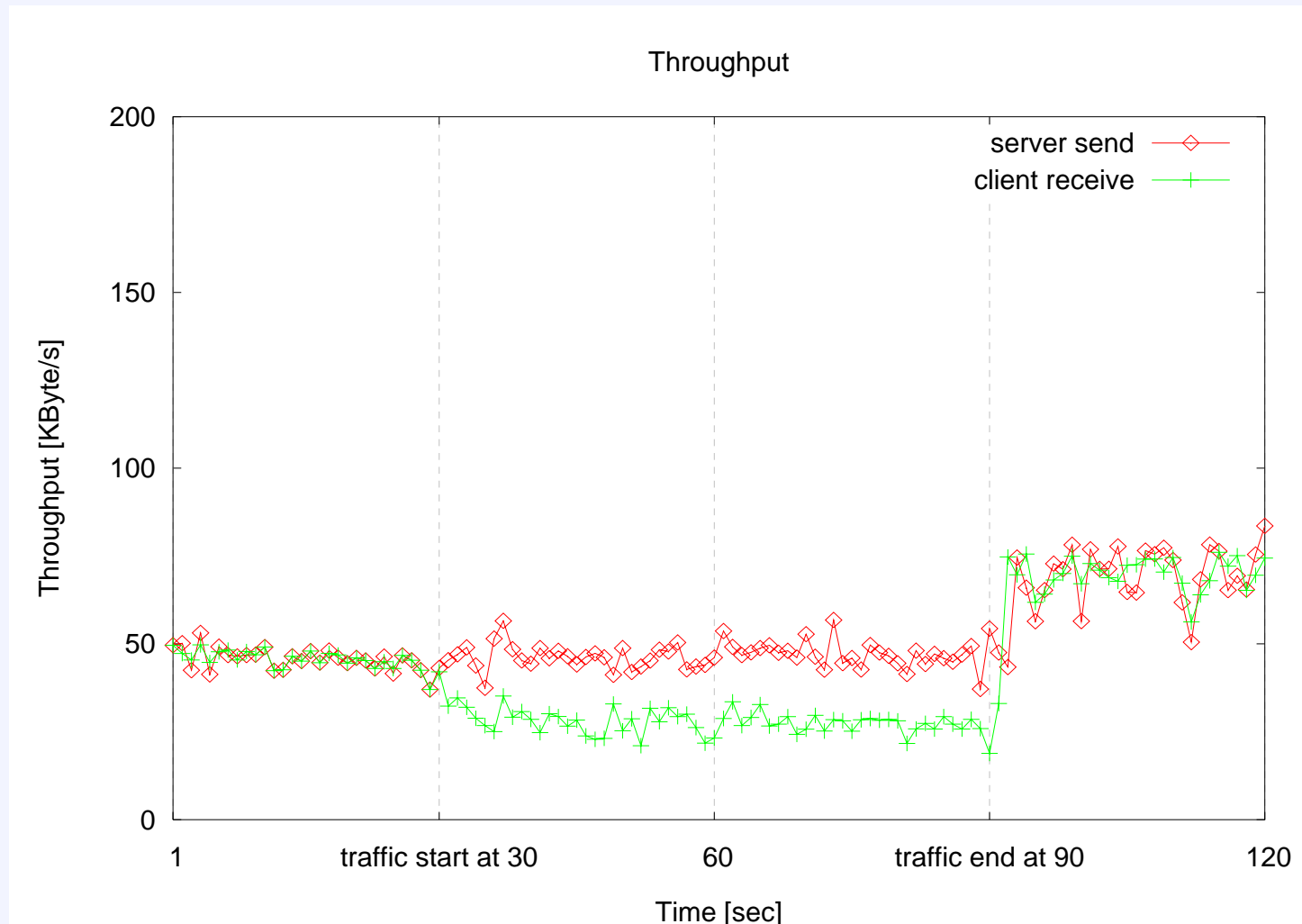
TCP (the way it should be)



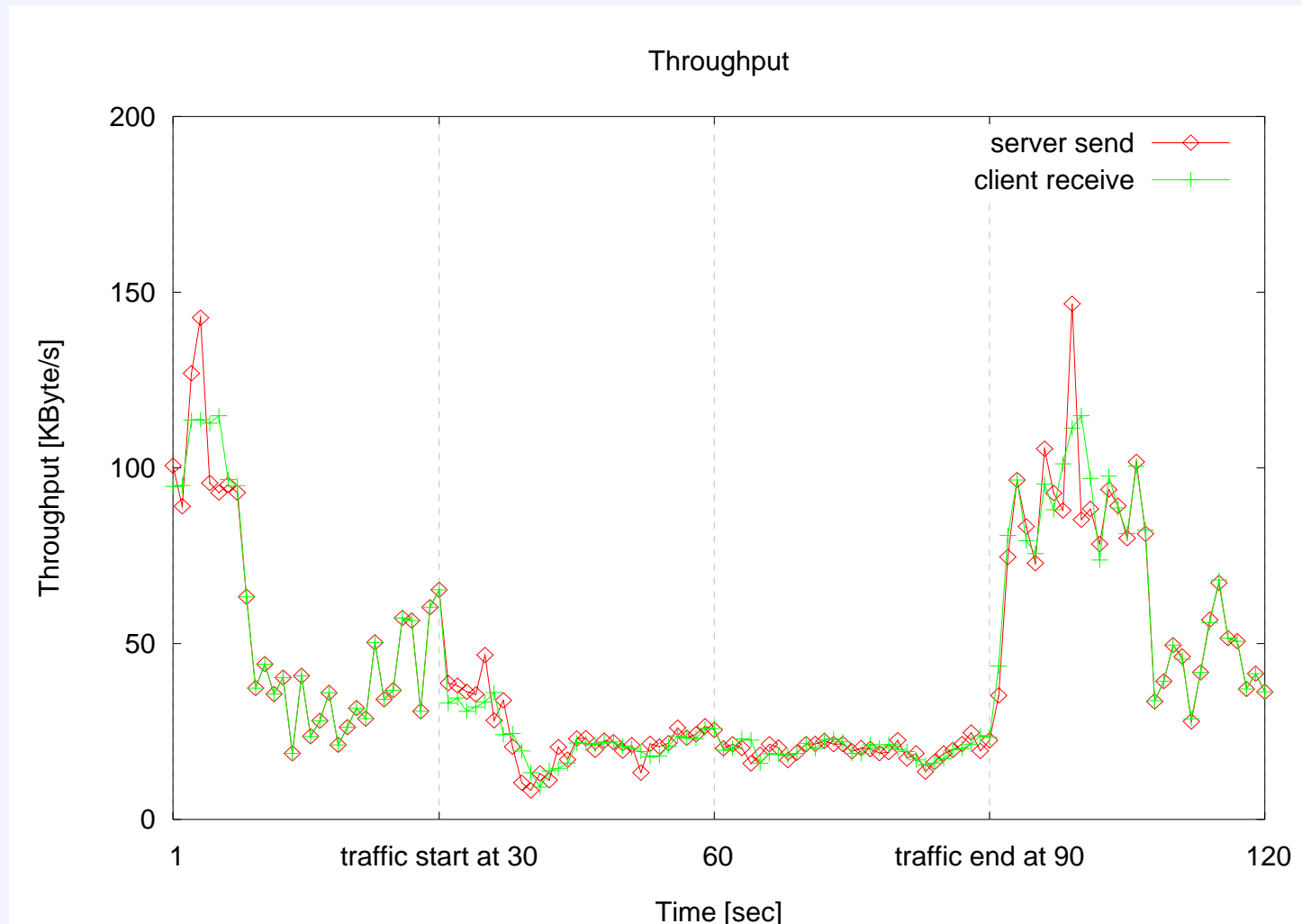
Streaming Video: RealPlayer



Streaming Video: Windows Media Player



Streaming Video: Quicktime



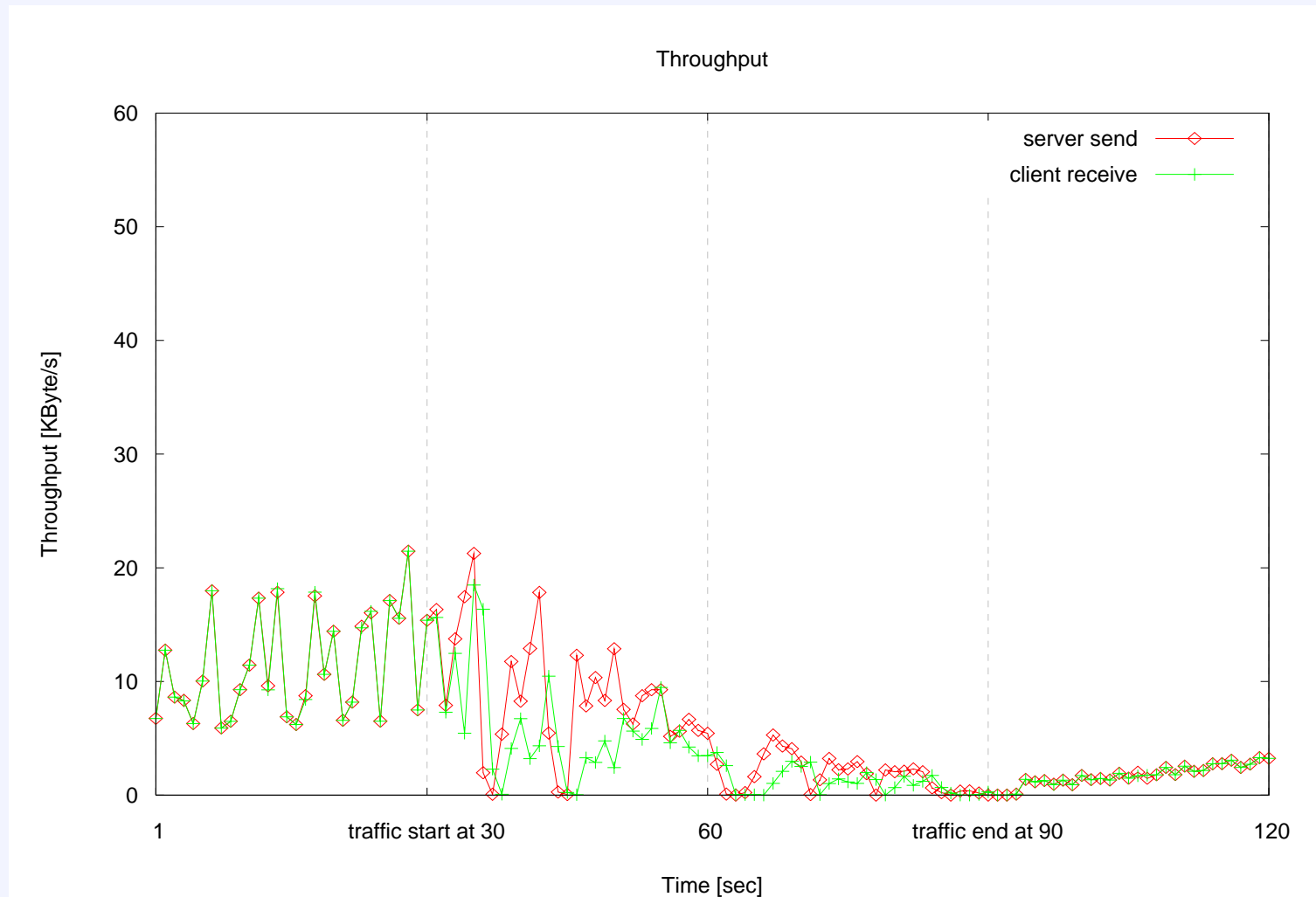
VoIP: MSN



VoIP: Skype



Video conferencing: iVisit



Observations

- Several other applications examined
 - ICQ, NetMeeting, AOL Instant Messenger, Roger Wilco, Jedi Knight II, Battlefield 1942, FIFA Football 2004, MotoGP2
- Often: congestion \Rightarrow increase rate
 - is this FEC?
 - often: rate increased by increasing packet size
 - note: packet size limits measurement granularity
- Many are unreactive
 - Some have quite a low rate, esp. VoIP and games
- Aggregate of unreactive low-rate flows = **dangerous!**
 - IAB Concerns Regarding Congestion Control for Voice Traffic in the Internet [RFC 3714]

Conclusion

- TCP = too much
 - TCP++ (or rather TCP--) needed
- UDP = not enough
 - UDP++ needed
- We will see that, in fact, sometimes, even UDP = too much
 - UDP-- needed
- These gaps are filled by the new IETF transport protocols
 - TCP++ = SCTP
 - UDP++ = DCCP
 - UDP-- = UDP-Lite

Stream Control Transmission Protocol (SCTP)

Motivation

- TCP, UDP do not satisfy all application needs
- SCTP evolved from work on IP telephony signaling
 - Proposed IETF standard (RFC 2960)
 - Like TCP, it provides reliable, full-duplex connections
 - Unlike TCP and UDP, it offers new delivery options that are particularly desirable for telephony signaling and multimedia applications
- TCP + features
 - Congestion control similar; some optional mechanisms mandatory
 - Two basic types of enhancements:
 - performance
 - robustness

Overview of services and features

Services/Features	SCTP	TCP	UDP
• Full-duplex data transmission	yes	yes	yes
• Connection-oriented	yes	yes	no
• Reliable data transfer	yes	yes	no
• Unreliable data transfer	yes	no	yes
• Partially reliable data transfer	yes	no	no
• Ordered data delivery	yes	yes	no
• Unordered data delivery	yes	no	yes
• Flow and Congestion Control	yes	yes	no
• ECN support	yes	yes	no
• Selective acks	yes	yes	no
• Preservation of message boundaries	yes	no	yes
• PMTUD	yes	yes	no
• Application data fragmentation	yes	yes	no
• Multistreaming	yes	no	no
• Multihoming	yes	no	no
• Protection agains SYN flooding attack	yes	no	n/a
• Half-closed connections	no	yes	n/a

Packet format

- Unlike TCP, SCTP provides **message-oriented** data delivery service
 - key enabler for performance enhancements
- **Common header**; three basic functions:
 - Source and destination ports together with the IP addresses
 - Verification tag
 - Checksum: **CRC-32 instead of Adler-32**
- followed by **one or more chunks**
 - chunk header that identifies length, type, and any special flags
 - concatenated building blocks containing either control or data information
 - control chunks transfer information needed for association (connection) functionality and data chunks carry application layer data.
 - Current spec: 14 different Control Chunks for association establishment, termination, ACK, destination failure recovery, ECN, and error reporting
- Packet can contain several different chunk types
- SCTP is extensible

Performance enhancements

- Decoupling of **reliable** and **ordered** delivery
 - **Unordered delivery**: eliminate **head-of-line blocking** delay

TCP receiver buffer

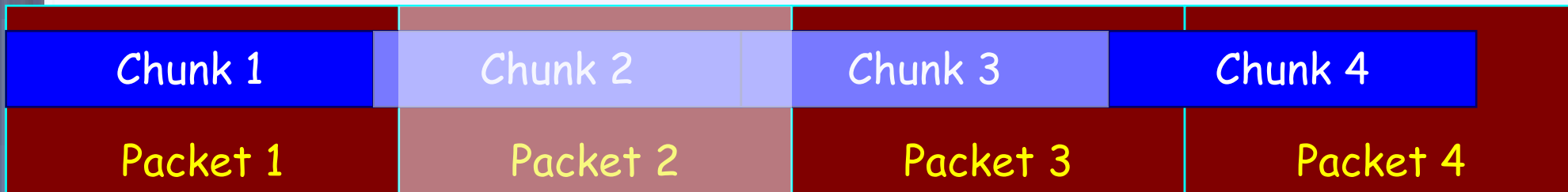
Chunk 2	Chunk 3	Chunk 4	Chunk 1
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App waits in vain! 

- Application Level Framing
- Support for **multiple data streams** (per-stream ordered delivery)
 - **Stream sequence number (SSN)** preserves order *within* streams
 - no order preserved *between* streams
 - per-stream flow control, per-association congestion control

Application Level Framing

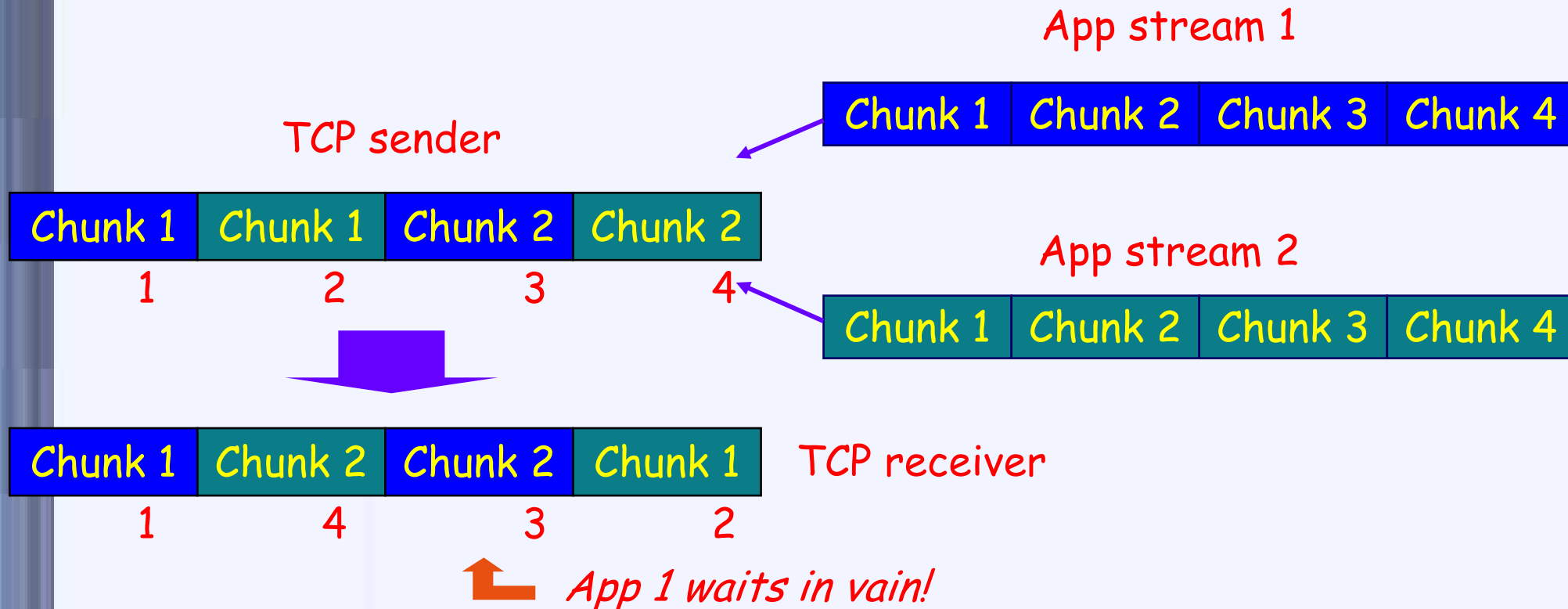
- TCP: byte stream oriented protocol
- Application may want logical data units ("chunks")
- Byte stream inefficient when packets are lost



- ALF: app chooses packet size = chunk size
packet 2 lost: no unnecessary data in packet 1,
use chunks 3 and 4 before retrans. 2 arrives
- 1 ADU (Application Data Unit) = multiple chunks -> ALF still more efficient!

Multiple Data Streams

- Application may use multiple logical data streams
 - e.g. pictures in a web browser
- Common solution: multiple TCP connections
 - separate flow / congestion control, overhead (connection setup/teardown, ..)



Multihoming

- ...at transport layer! (i.e. transparent for apps, such as FTP)
- TCP connection \Leftrightarrow SCTP association
 - 2 IP addresses, 2 port numbers \Leftrightarrow 2 sets of IP addresses, 2 port numbers
- Goal: robustness
 - automatically switch hosts upon failure
 - eliminates effect of long routing reconvergence time
- TCP: no guarantee for “keepalive” messages when connection idle
- SCTP monitors each destination's reachability via ACKs of
 - data chunks
 - heartbeat chunks
- Note: SCTP uses multihoming for redundancy, not for load balancing!

Association phases

- Association establishment: 4-way handshake

- Host A sends INIT chunk to Host B
- Host B returns INIT-ACK containing a cookie
 - information that only Host B can verify
 - No memory is allocated at this point!
- Host A replies with COOKIE-ECHO chunk; may contain A's first data.
- Host B checks validity of cookie; association is established

Avoids SYN
flood attacks!

- Data transfer

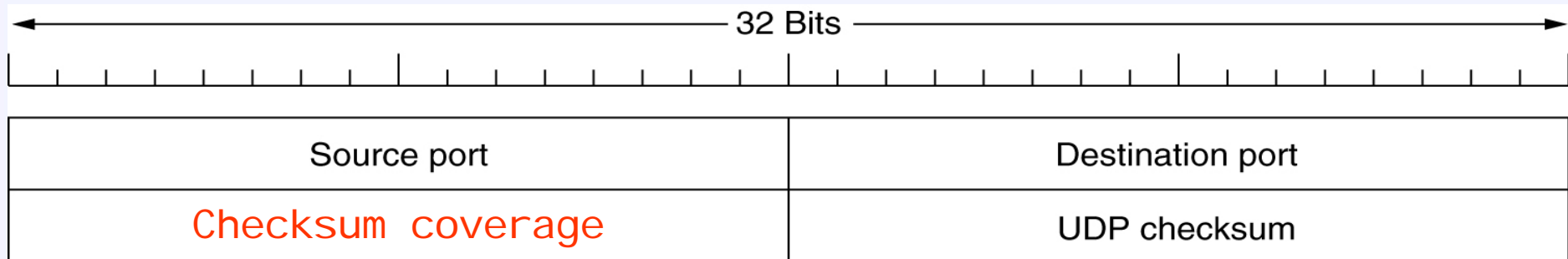
- SCTP assigns each chunk a unique Transmission Sequence Number (TSN)
- SCTP peers exchange starting TSN values during association establishment phase
- Message oriented data delivery; fragmented if larger than destination path MTU
- Can bundle messages < path MTU into a single packet and unbundle at receiver
- reliability through acks, retransmissions, and end-to-end checksum

- Association shutdown: 3-way handshake

- SHUTDOWN ⇒ SHUTDOWN-ACK ⇒ SHUTDOWN-COMPLETE
- Does not allow half-closed connections
(i.e. one end shuts down while the other end continues sending new data)

UDP-Lite

UDP-Lite

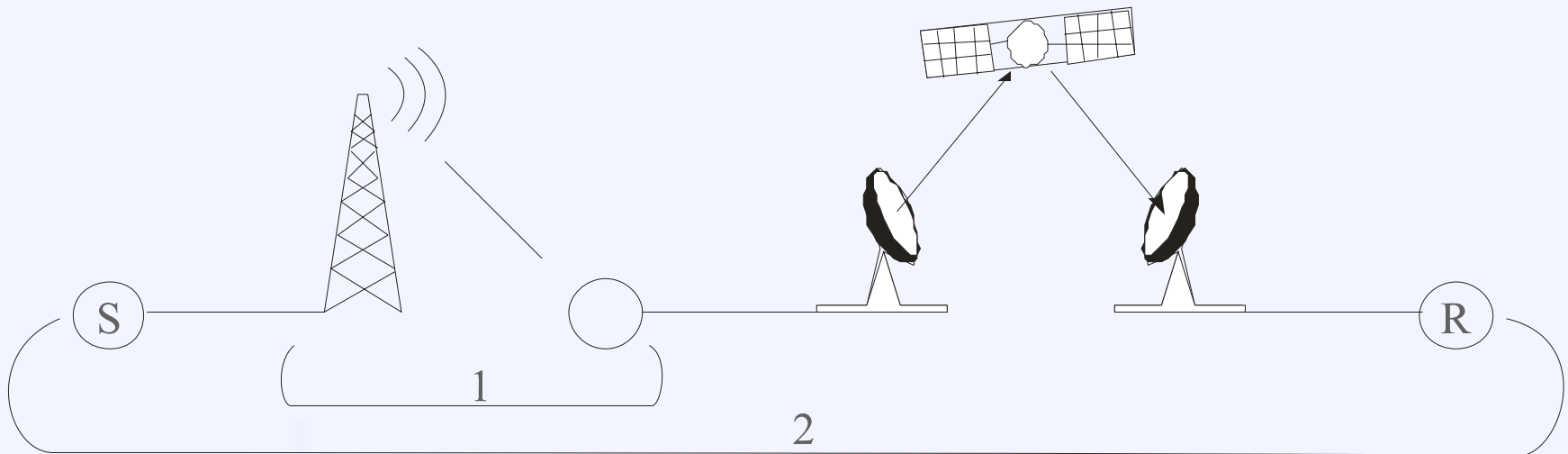


- **Checksum:** Adler-32 covering the whole packet
 - UDP: checksum field = 0 \Rightarrow no checksum at all - bad idea!
- **solution:** UDP-Lite (length := checksum coverage)
 - e.g. video codecs can cope with bit errors, but UDP throws whole packet away!
 - acceptable BER up to applications (complies with end-to-end arguments)
 - some data can be covered by checksum
 - apps can realize several or different checksums
- **Issues:**
 - apps can depend on lower layers (no more "IP over everything")
 - authentication requires data integrity - not given with UDP-Lite
 - handing over corrupt data is not always efficient - link layer should detect UDP-Lite

Inter-layer
communication
problem

Link layer ARQ

- Advantages:
 - potentially faster than end-to-end retransmits
 - operates on frames, not packets
 - could use knowledge that is not available at transport end points
- example scenario: control loop 1 much shorter than 2



Link Layer ARQ /2

- Disadvantages:
 - hides information (known corruption) from end points
 - TCP: increased delay \Rightarrow more conservative behavior
- Link layer ARQ can have varying degrees of persistence
- So what?
- Ideal choice would depend on individual end-to-end flows
- Thus, recommendation:
 - low persistence or disable (leave severe cases up to end points)
 - Give end points means to react properly (detect corruption)

Further details:
RFC 3366

Datagram Congestion Control Protocol (DCCP)

Motivation

- Some apps want **unreliable, timely delivery**
 - e.g. VoIP: significant delay = ☹ ... but some noise = ☺
- **UDP: no congestion control**
- **Unresponsive long-lived applications**
 - endanger others (congestion collapse)
 - may hinder themselves (queuing delay, loss, ..)
- **Implementing congestion control is difficult**
 - illustrated by lots of faulty TCP implementations
 - may require precise timers; should be placed in kernel

DCCP fundamentals

- Congestion control for unreliable communication
 - in the OS, where it belongs
- Well-defined framework for [TCP-friendly] mechanisms
- Roughly:

Not an explicit DCCP requirement, but a current IETF requirement

DCCP = TCP - (bytestream semantics, reliability)
= UDP + (congestion control with ECN, handshakes, ACKs)

- Main specification does not contain congestion control mechanisms
 - CCID definitions (e.g. TCP-like, TFRC, TFRC for VoIP)
- IETF status: working group, several Internet-drafts, thorough review
 - RFCs published in March 2006

What DCCP does for you (roughly)

- Multiplexing + protection against corruption
 - ports, checksum (UDP-Lite ++)
- Connection setup and teardown
 - even though unreliable! one reason: middlebox traversal
- Feature negotiation mechanism
 - Features are variables such as CCID ("Congestion Control ID")
- Reliable ACKs \Rightarrow knowledge about congestion on ACK path
 - ACKs have sequence numbers
 - ACKs are transmitted (receiver) until ACKed by sender (ACKs of ACKs)
- Full duplex communication
 - Each sender/receiver pair is a half-connection; can even use different CCIDs!
- Some security mechanisms, several options

Packet format

2 Variants; different sequence no. length, detection via X flag

Source Port			Destination Port	
Data Offset		CCVal	CsCov	Checksum
Res	Type	X = 1	Reserved	Sequence Number (high bits)
Sequence Number (low bits)				

Source Port			Destination Port	
Data Offset		CCVal	CsCov	Checksum
Res	Type	X = 0	Sequence Number (low bits)	

- Generic header with 4-bit **type** field
 - indicates following subheader
 - only one subheader per packet, not several as with SCTP chunks

Separate header / payload checksums

- Available as “Data Checksum option” in DCCP
 - Also suggested for TCP, but not (yet?) accepted
 - Note: **partial checksums** useless in TCP (reliable transmission of erroneous data?)
- Differentiate corruption / congestion
 - Checksum covers all
 - Error could be in header
 - Impossible to notify sender (seqno, ports, ..)
 - Checksum fails in header only
 - Bad luck
 - Checksum fails in payload only, ECN = 0
 - Inform sender of corruption
 - No need to react as if congestion
 - Still react (keeping high rate + high BER = bad idea) ⇒ experimental!
 - Checksum fails in payload only, ECN = 1
 - Clear sign of congestion

Additional options

- **Data Dropped:** indicate differentdrop events in receiver (differentiate: not received by app / not received by stack)
 - removed from buffer because receiver is too slow
 - received but unusable because corrupt (Data Checksum option)
- **Slow receiver:** simple flow control
- **ACK vector:** SACK (runlength encoded)
- **Init Cookie:** protection against SYN floods
- **Timestamp, Elapsed Time:** RTT estimation aids
- **Mandatory:** next option must be supported
- **Feature negotiation:** Change L/R, Confirm L/R

Classifying DCCP applications

- Congestion control trade-off (selfish single-flow view):

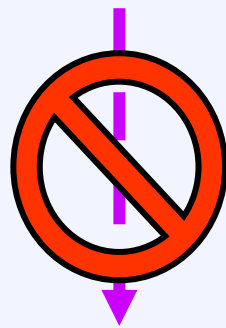
- + reduced loss

- necessary to adapt rate

- Use sender buffer, drain it with varying rate

- Change encoding

Trade-off: sender buffer size (=delay) vs. frequency of encoding changes



VoIP,
Games

Videoconf.

Sweet spot?

Streaming Media

Delay sensitive

Delay insensitive

Is TCP the ideal protocol for one-way streaming media?

- Perhaps! Let's consider what happens...
- Remember: we're at the "buffering" side of the spectrum
 - Buffers (delay) don't matter
 - User perception studies of adaptive multimedia apps have shown that users dislike permanent encoding changes (big surprise :-)

⇒ no need for a smooth rate!
- **Little loss case**: TCP retransmissions won't hurt
- **Heavy loss case**:
- DCCP: 1, ~~2~~, ~~3~~, 4, ~~5~~, ~~6~~, 7, ~~8~~, ~~9~~, 10...
- TCP: (assume window = 3): 1, ~~2~~, ~~3~~, 2, ~~3~~, ~~4~~, 3, ~~4~~, ~~5~~, 4...
 - Application would detect: 4 out of 10 expected packets arrived
⇒ should reduce rate
 - Is receiving 1, 4, 7, 10 instead of 1, 2, 3, 4 really such a big benefit?
 - Or is it just a matter of properly reacting?
 - In RealPlayer and MediaPlayer, TCP can be used for streaming... seems to work well (also in YouTube!)

DCCP usage: incentive considerations

- Benefits from DCCP (perspective of a single application) limited
- Compare them with reasons not to use DCCP
 - programming effort, especially if updating a working application
 - common deployment problems of new protocol with firewalls etc.
- What if dramatically better performance is required to convince app programmers to use it?
- Can be attained using “penalty boxes” - but:
 - requires such boxes to be widely used
 - will only happen if beneficial for ISP:
financial loss from unresponsive UDP traffic > financial loss from customers whose UDP application doesn't work anymore
 - requires many applications to use DCCP
 - chicken-egg problem!

References

- Michael Welzl: "Network Congestion Control: Managing Internet Traffic", John Wiley & Sons, July 2005.
- Randall R. Stewart, Qiaobing Xie: "Stream Control Transmission Protocols (SCTP)", Addison-Wesley Professional 2002.
- Key RFCs (main protocol specifications):
 - [SCTP](#): RFC 2960; [UDP-Lite](#): RFC 3828; [DCCP](#): RFC 4340
- Recommended URLs:
 - [SCTP, UDP-Lite](#):
 - <http://www.ietf.org/html.charters/tsvwg-charter.html>
 - [SCTP](#):
 - <http://www.sctp.org/>
 - http://tdrwww.exp-math.uni-essen.de/inhalt/forschung/sctp_fb/
 - [DCCP](#):
 - <http://www.ietf.org/html.charters/dccp-charter.html>
 - <http://www.icir.org/kohler/dccp/>