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INF 3190

New Internet Standards

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Overview: a biased sample…

- Based on: “what goes on ‘under the hood’ today/tomorrow, with the things many of us are using?”
  - but also related to own research

  - … and some not-so-novel background needed!

Stream Control Transmission Protocol (SCTP)

- TCP, UDP do not satisfy all application needs

- SCTP evolved from, and is used for, IP telephony signaling
  - Like TCP: reliable, full-duplex connections
  - Unlike TCP and UDP: new delivery options that are particularly desirable for telephony signaling and multimedia applications

- TCP + features
  - Congestion control similar; some optional TCP mechanisms mandatory
  - Two basic types of enhancements: 1) Performance; 2) Robustness

SCTP services: SoA TCP + extras

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 (ALF)	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

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 containg 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

Application Level Framing (ALF)

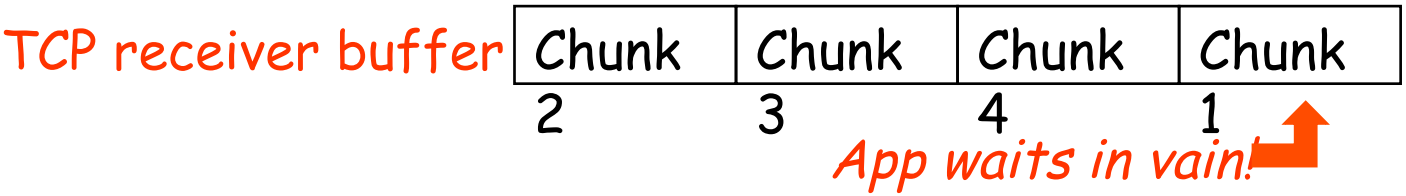
- Concept applied in RTP and SCTP
  - Byte stream (TCP) inefficient when packets are lost
  - Application may want logical data units (“chunks“)

- 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!

Unordered delivery & multistreaming

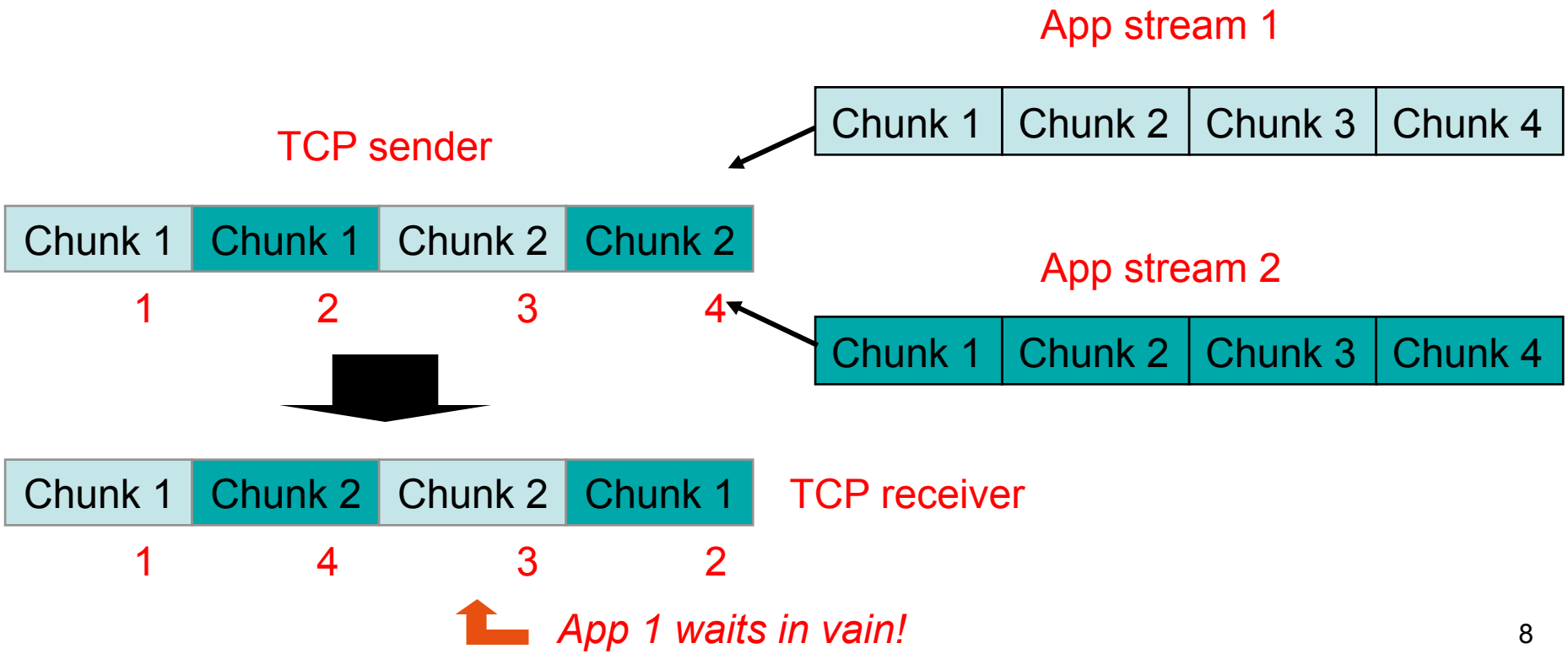
- Decoupling of reliable and ordered delivery
  - Unordered delivery: eliminate Head-Of-Line (HOL) blocking delay



- Support for multiple data streams (per-stream ordered delivery)
  - Stream sequence number (SSN) preserves order *within* streams
  - no order preserved *between* streams

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 ⇔ SCTP association
  - 2 IP addresses, 2 port numbers ⇔ 2 sets of IP addresses, 2 port numbers
- Primary goal: robustness
  - automatically switch hosts upon failure
  - eliminates effect of long routing reconvergence time
  - Now also CMT (Concurrent Multipath Transport) as with MPTCP; more later...
- TCP: no “keepalive“ messages when connection idle
- SCTP monitors reachability via ACKs of data chunks and heartbeat chunks

Google's SPDY (basis for HTTP/2.0, many similarities)

- Goal: reduce page load time
- Reuses HTTP semantics
  - But changes how data is written to the network (e.g. no ascii protocol!)
  - retains all features including cookies, Content-Encoding negotiations etc.
- Universal encryption
  - SPDY is negotiated over SSL/TLS, thus operates exclusively over a secure channel

SPDY /2

- Header compression
- Server Push/Hint
  - Servers could proactively push resources to clients (e.g. scripts and images that will be required)
  - Or can send hints advising clients to pre-fetch content
- Content prioritization
  - Client can specify the preferred order in which resources should be transferred

SPDY Multiplexing

- Multiplexing
  - Persistent connection as in HTTP/1.1
    Reason: allow TCP to increase its window
    (most web flows terminate in slow start)
  - But: in HTTP/1.1, sequence determined by client
  - Client does not know which requests take long (e.g. database lookups, ..); can cause HOL delay !
  - SPDY multiplexes frames onto the TCP connection
    - TCP can still cause RTT-timescale HOL blocking delay
    - Google's solution: QUIC / UDP…

MPTCP

- Many hosts are nowadays multihomed
  - Smartphones (WiFi + 3G), data centers
  - Why not use both connections at once?

- Cannot know where bottleneck is
  - If it is shared by the two connections, they should appear (be as aggressive) as only one connection
  - MPTCP changes congestion avoidance “increase” parameter to “divide” aggression accordingly
    - but instead of being “½ TCP”, tries to send as much as possible over least congested path
    - Least congested = largest window achieved; hence, increase in proportion to window

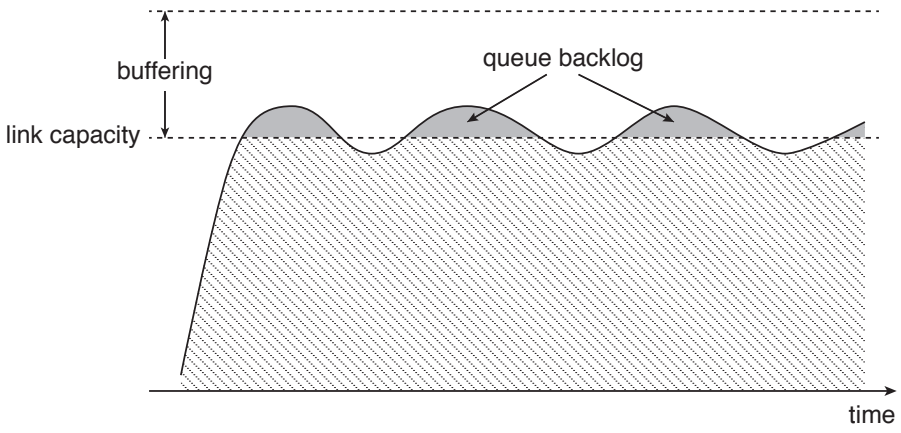
MPTCP /2

- Moving traffic away from congested links achieves “resource pooling”
  - A web server connected to two 100 Mbit/s links behaves roughly as if it had one 200 Mbit/s link
  - Only one host needs to be multi-homed

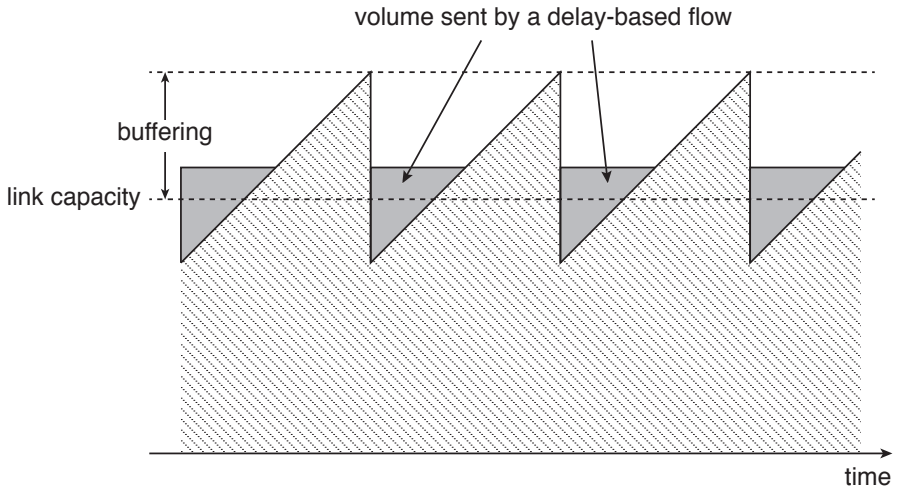
- Issues
  - Must look like TCP to work everywhere
    Minimal on-the-wire changes: new TCP option
  - Parallel paths can cause reordering ➔ delay in handing over data to application on receiver side

LEDBAT

- Try to send when others don't
  - For low-priority traffic that should not get in the way of other applications
  - Growing (assumption: queuing) delay = early congestion signal
  - Possible benefit: low delay
  - Encapsulation (how to embed in existing protocols) not (yet?) defined; implemented over UDP in BitTorrent



(a) Sending rate of a delay-based flow: ideal case.



(b) Coexistence of an LBE flow with a non-LBE one.

WebRTC / rtcweb

- Direct UDP communication between browsers (p2p)
  - Better latency & bandwidth, important for interactive communication (video, audio, online games, …)
  - Data channel (e.g. control data in games, file transfers, ..) uses SCTP in userspace (in browser)
  - Between browsers, there are many middleboxes; several tricks played (ICE / STUN / TURN protocols)
- Javascript API lets web designer control “peer connections”

- Congestion control under development; requirements:
  - Avoid queue: react to delay, yet interoperate with TCP
  - Detect shared bottlenecks, combine controls of flows

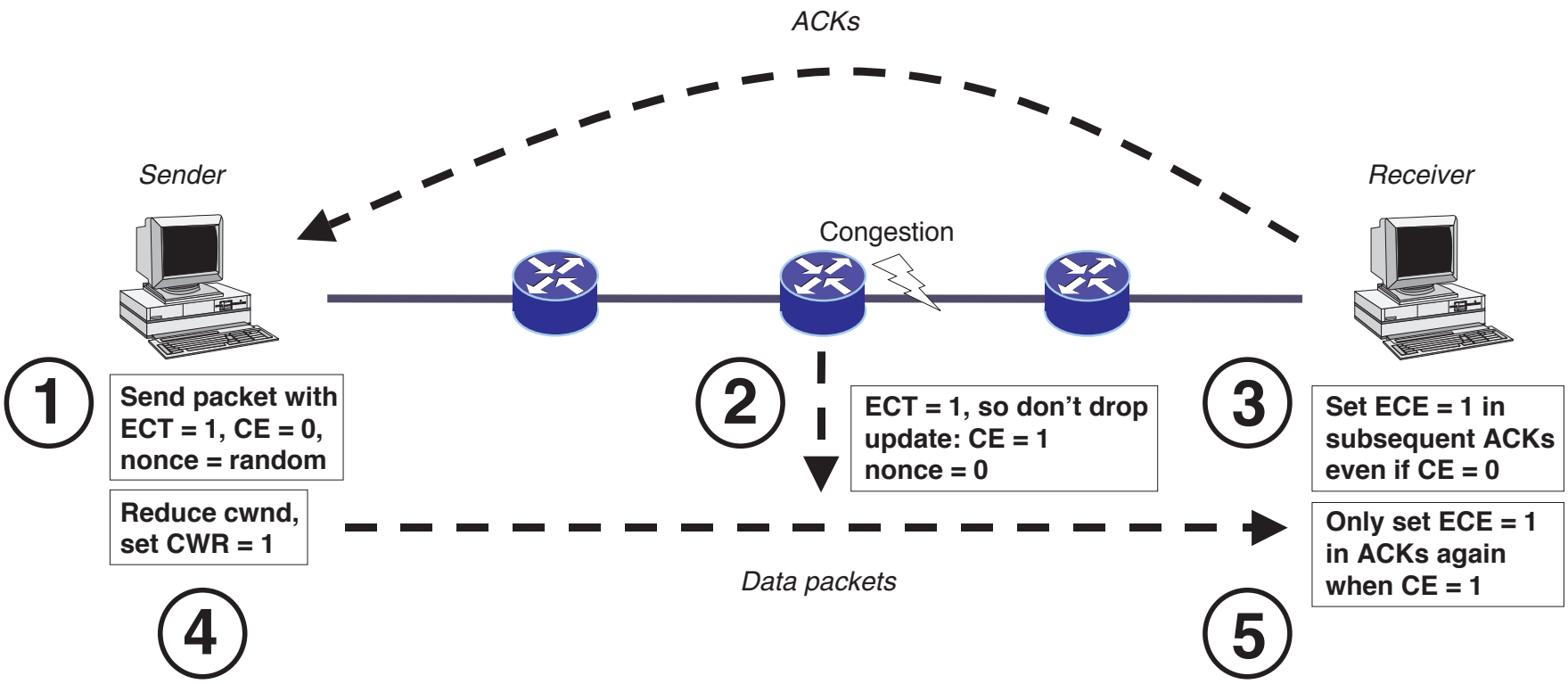
Explicit Congestion Notification (ECN)

- AQM: Instead of dropping, set a bit
- Receiver tells sender about it; sender behaves as if packet dropped
  ⇒actual communication between end nodes and the network
- Note: ECN = true congestion signal (i.e. clearly not corruption)

- Typical incentives:
  - sender = server; efficiently use connection, fairly distribute bandwidth
    - use ECN as it was designed
  - receiver = client; goal = high throughput, does not care about others
    - ignore ECN flag, do not inform sender about it

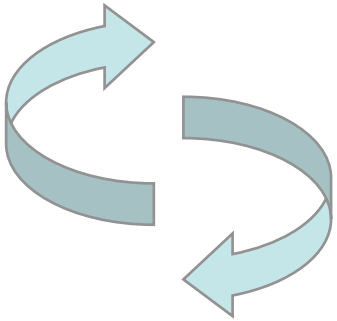
- Shouldn‘t be possible for receiver to lie about ECN when it was set!
  - Solution: nonce = random number from sender, deleted by router
  - Sender believes „no congestion“ iff correct nonce is sent back

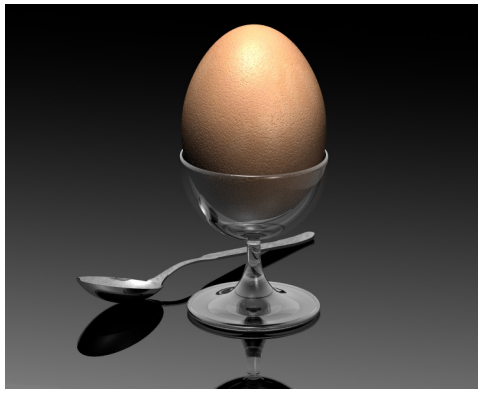
ECN in action



- Nonce provided by bit combination:
  - ECT(0): ECT=1, CE=0; ECT(1): ECT=0, CE=1
- Nonce usage specification experimental, suggestions to replace: ECT1 could mean "give me a different queuing behavior"

The Internet-deployment vicious circle





Application developer:

- wants to max. revenue
- Use new protocol: effort (-$) unless the protocol works everywhere (maybe ++$)

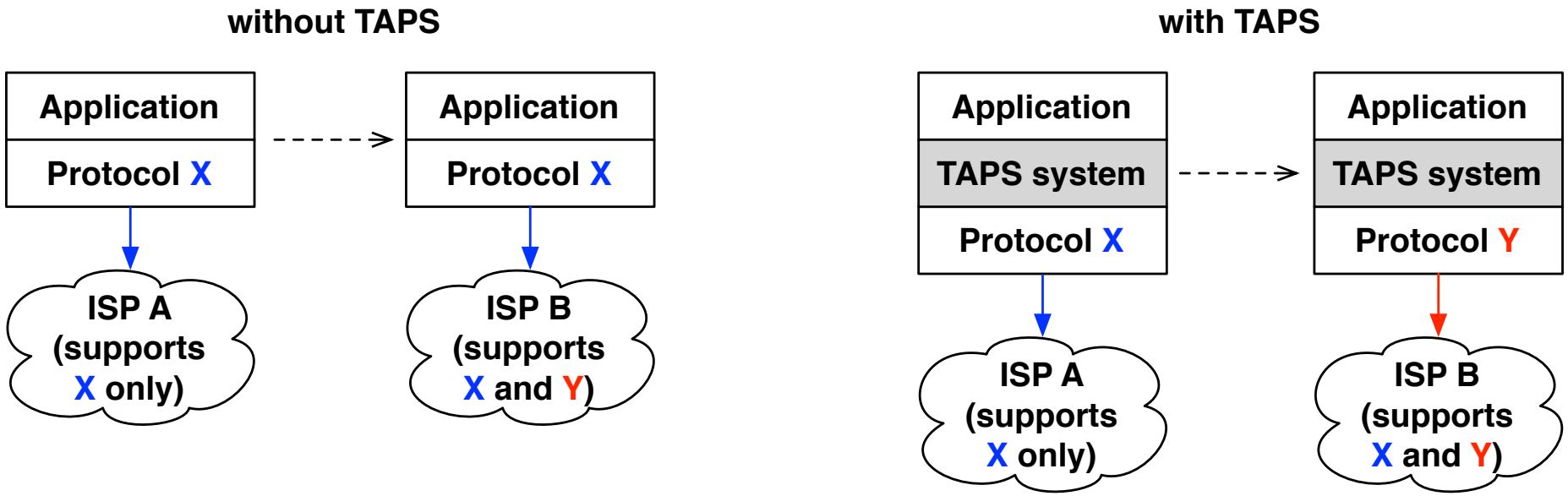
OS developer, middlebox designer / maintainer:

- wants to max. revenue
- Support new protocol alone: effort or risk (-$) unless the protocol is beneficially used by applications (maybe ++$)

Consequences of this vicious circle

- Efforts to make everything look like standard TCP
  - MPTCP, Minion (partially in Apple)
- Efforts to build whole protocols over UDP
  - LEDBAT (BitTorrent), QUIC (Google), RTMFP (Adobe), Skype (proprietary), etc.
- Possible solution: change interface to transport layer (let applications specify service, not protocol)
  - ongoing efforts in the IETF (TAPS WG)

Transport Services (TAPS)



EC project “NEAT“ (https://www.neat-project.org) develops TAPS-conformant implementation of a transport system