Implementing SCTP to support WebRTC data channels in pure Rust

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Boulder/Denver Rust Meetup

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About me

- Freelance software engineer
- Worked for many years in the consumer electronics industry
- C, C++, Linux device drivers, audio/video
- Long-time systems programmer interested in network protocols

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Outline

- **Goals**
- The problem domain
  - Peer-to-peer networking and WebRTC
  - SCTP: Stream Control Transport Protocol
- Implementing SCTP in Rust
  - Approach
  - Design
  - Implementation
  - Testing
  - The Future
- Demo time!

Goals

- Long-term: A set of crates that provides a complete solution for using WebRTC outside of a web browser, thus providing a peer-to-peer networking stack for devices and apps.
- Short-term: Support WebRTC data channels with a Rust implementation of the Stream Control Transport Protocol (SCTP).
- The stack should be pure Rust, with the exception of cryptography.

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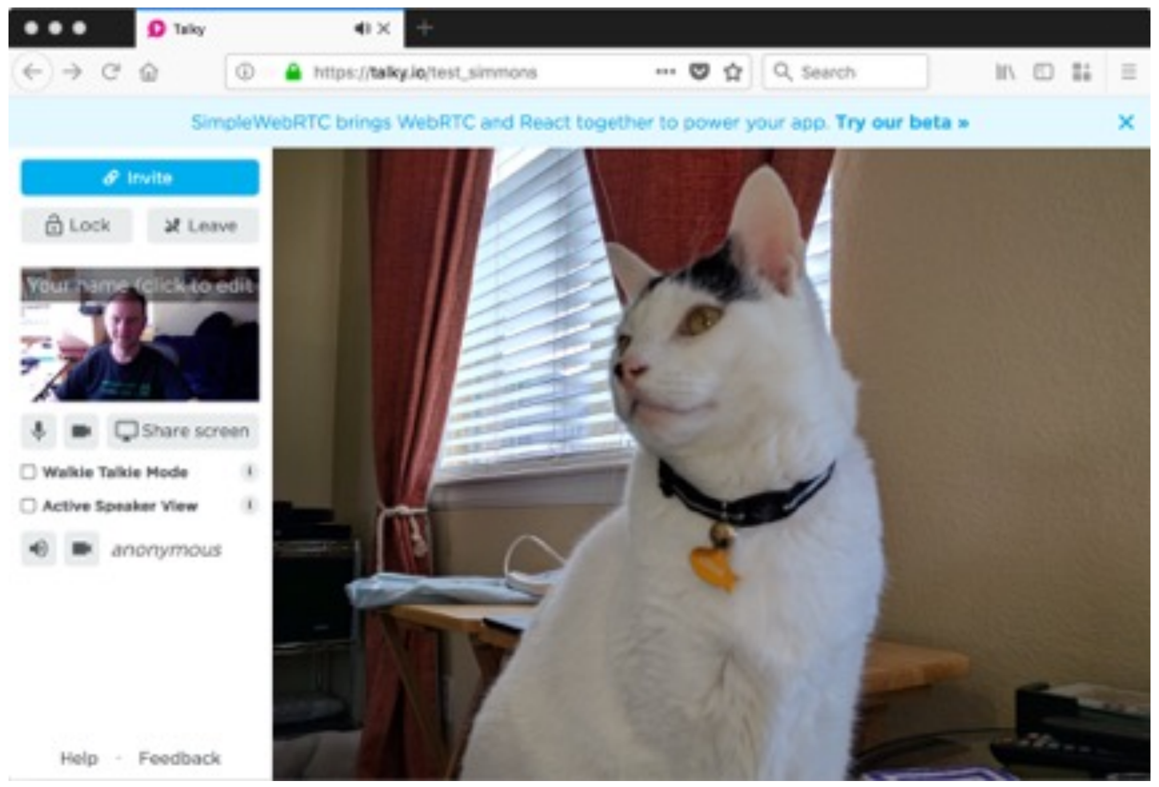
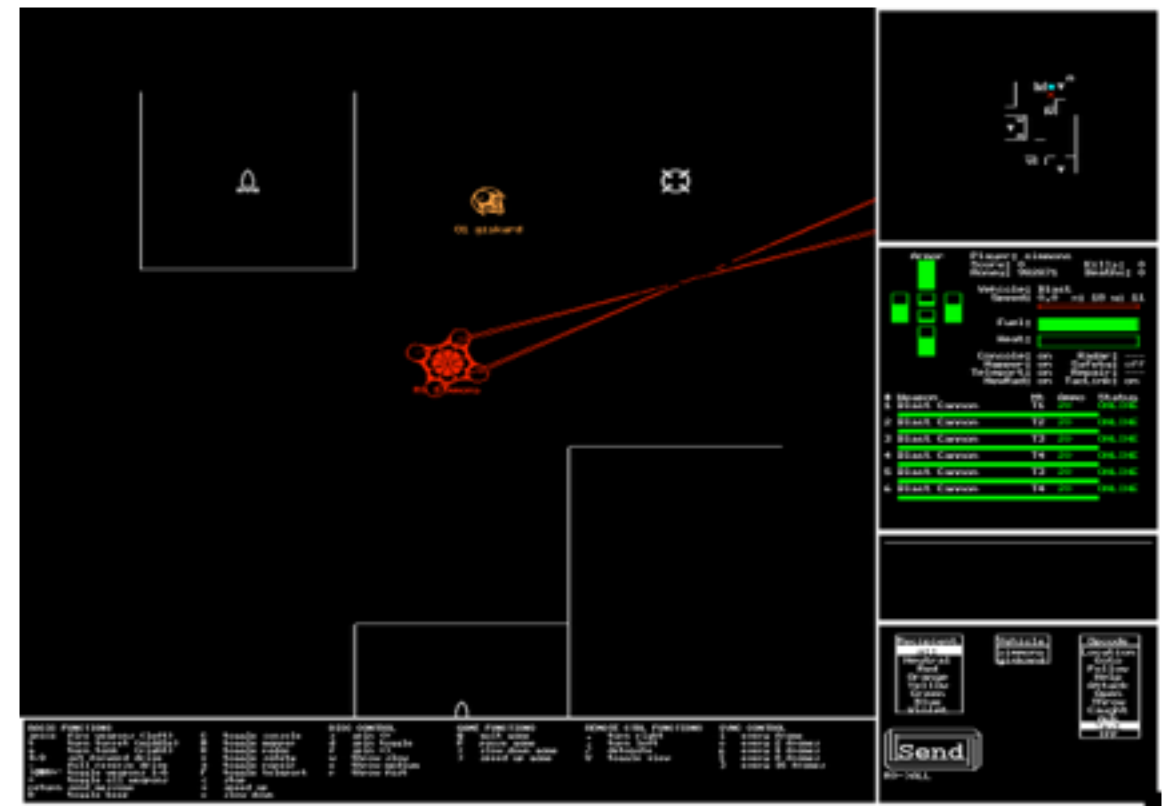
Why peer-to-peer networking?

- Technical reasons
  - Lower latency
  - Reduce bandwidth costs
- Political reasons
  - Theoretically possible to build overlay networks for privacy, anonymity and "routing around censorship" (e.g. I2P)
  - Self-hosting to reduce dependency on third parties
- Creative reasons
  - Experimenting with novel decentralized systems (e.g. distributed hash tables (DHTs), cryptocurrencies)
  - Enable the development of network applications that have yet to be imagined

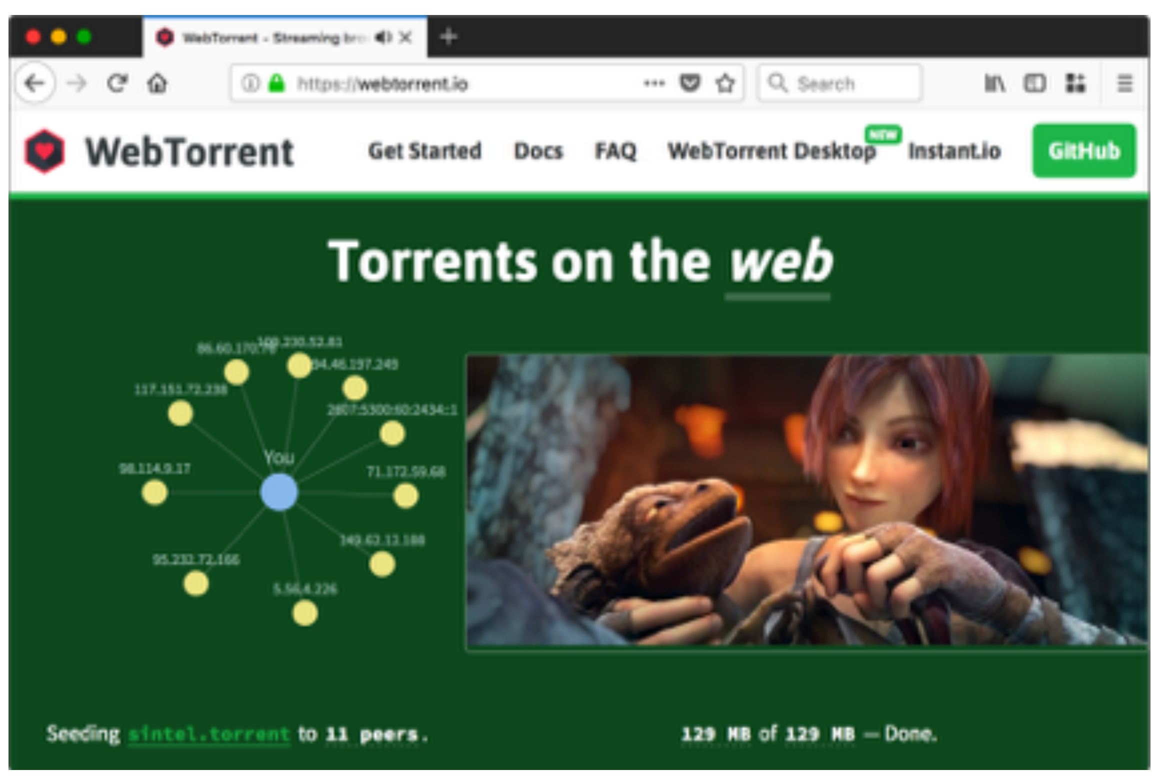
Peer-to-peer applications



Conferencing



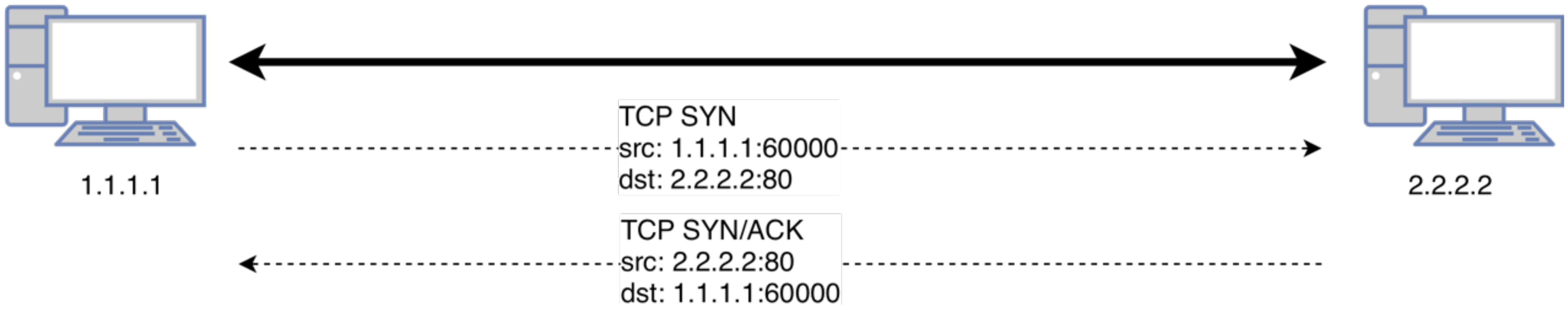
Multiplayer games



File transfers

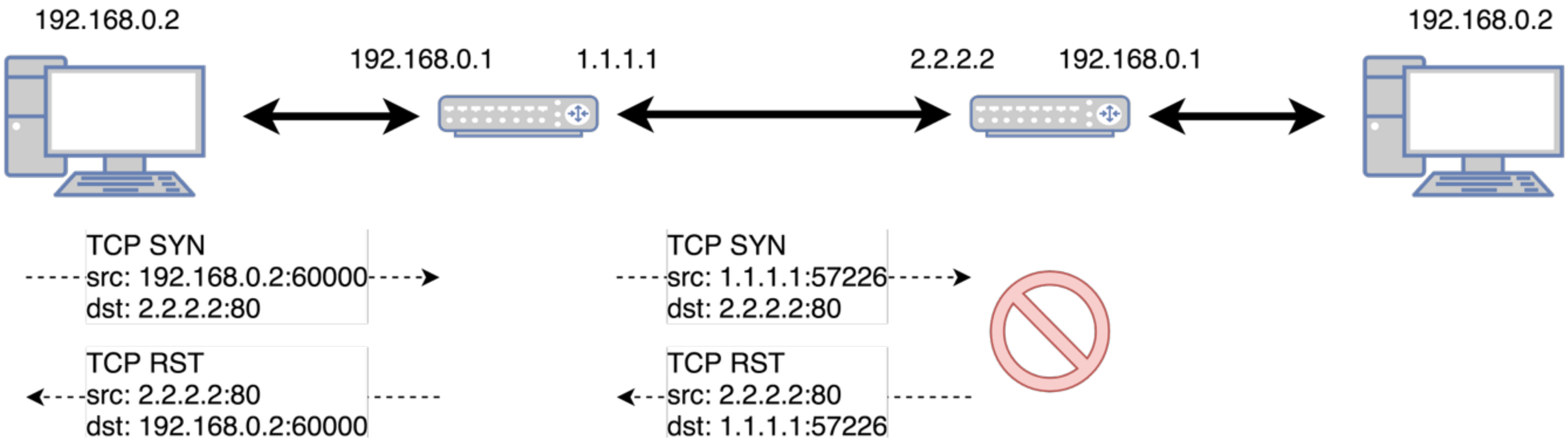
Security cameras

NAT Traversal (1/3)



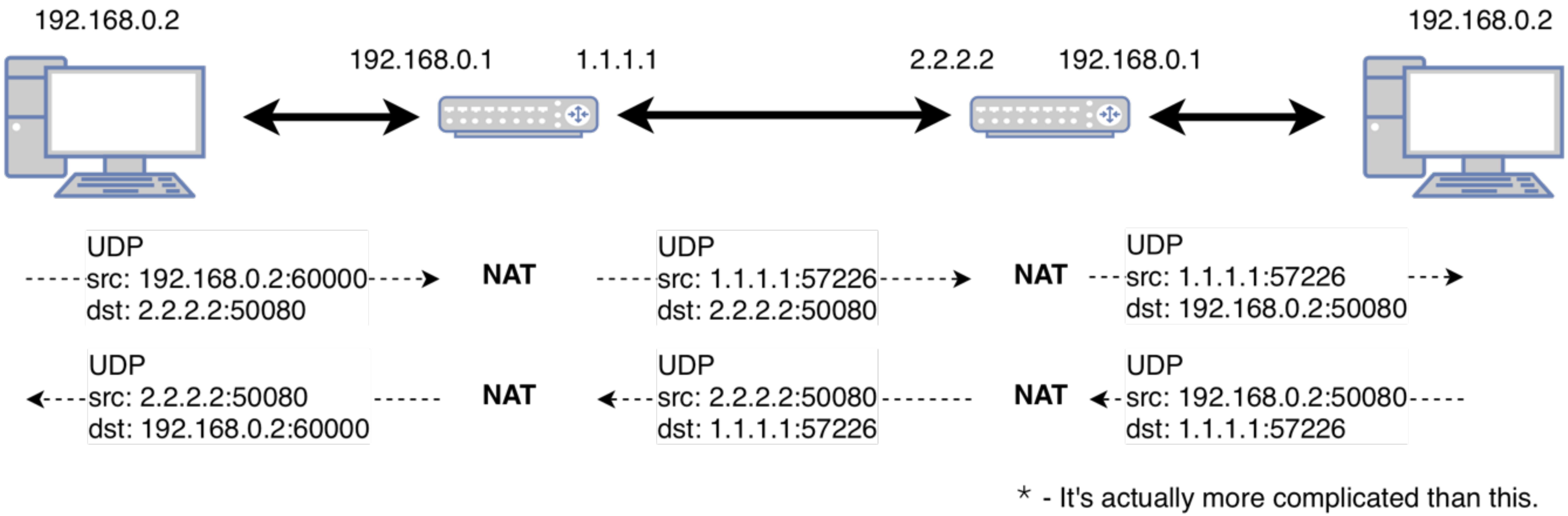
In an ideal world, we could connect to peers as simply as we connect to servers.

NAT Traversal (2/3)



- Consumer broadband routers use Network Address Translation (NAT) to provide private address spaces, translate addresses, and protect the network with simple stateful firewalls.
- Direct communication between hosts is problematic when each is behind a NAT.

NAT Traversal (3/3)



- The UDP hole punching technique works around NATs by each host independently initiating communication, thus establishing the required NAT mappings on both routers.
- Unfortunately, this means re-inventing the world on top of UDP.
- “TCP simultaneous open” is a thing, but is unreliable in practice.

IETF standardization of NAT Traversal

- Bespoke NAT traversal systems were invented and re-invented for years (e.g. Skype, circa 2003).
- In recent years, the Internet Engineering Task Force (IETF) has developed standards for NAT traversal.
  - STUN (RFC 3489, 2003) - Discover one's public IP and NAT situation.
  - TURN (RFC 5766, 2010) - Relay traffic as a fallback.
  - ICE (RFC 5245, 2010) - A system for negotiating address candidates and NAT-traversed UDP flow.
- NAT traversal is a kludge, but at least now it's a standardized kludge.

Enter WebRTC

- From my perspective as a systems programmer, WebRTC bundles all the best-known methods for peer-to-peer networking, and provides a common target for interoperability.

- Invented at Google in 2011, standardized by the W3C and IETF (RFC 7478 and others, 2015-)

- Like many things that start with the word "Web", WebRTC has ambitions beyond web browsers.

W3C

- Browser concerns
- Media access
- API

IETF

- Protocol specifications
- Network concerns

WebRTC

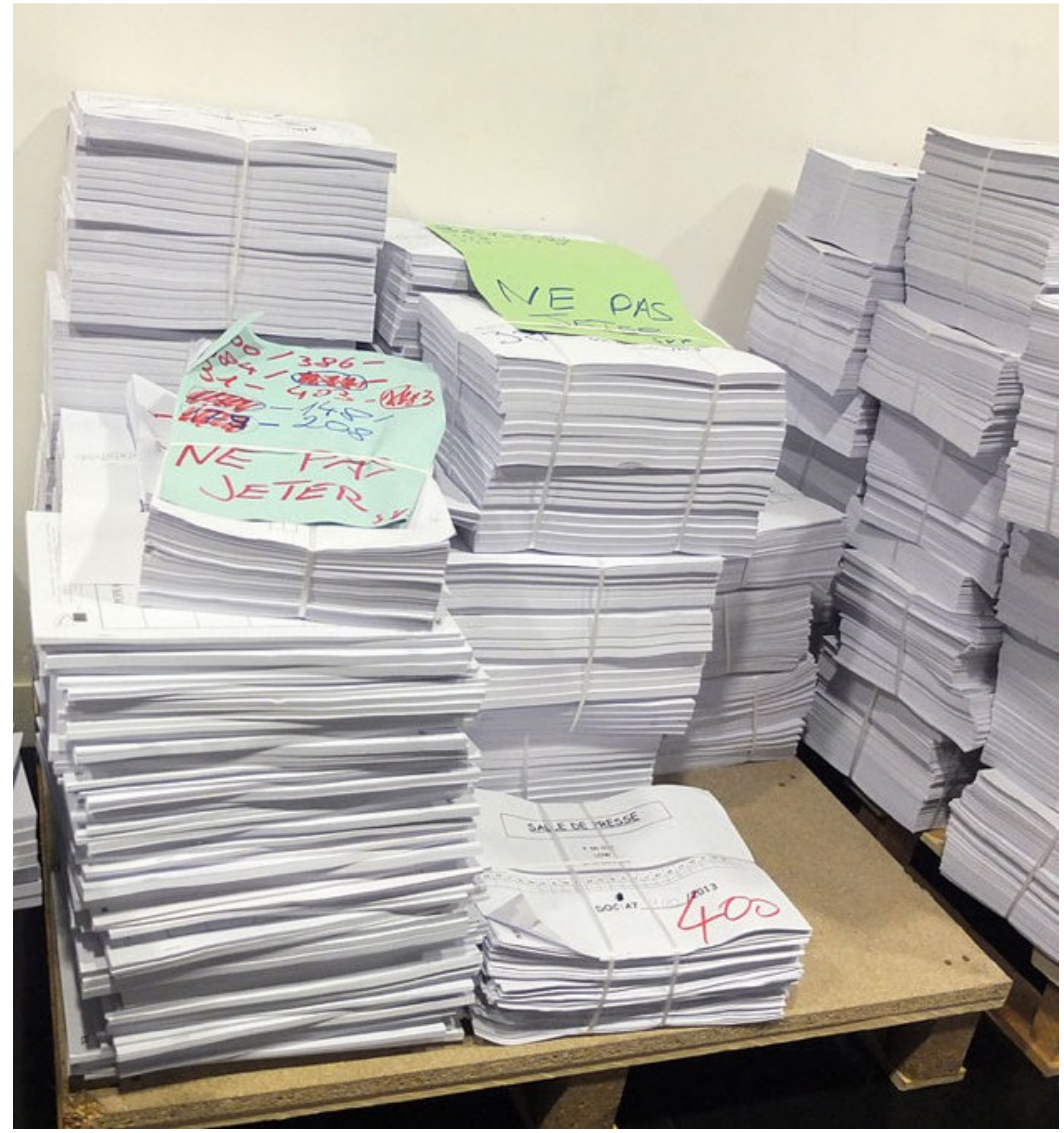
Re-inventing TCP

- We need to re-invent everything we lost when we ditched TCP in favor of UDP hole punching.
  - Reliability (when desired)
  - Ordered delivery (when desired)
  - Congestion control
- One solution would be TCP-over-UDP.
- WebRTC instead picked SCTP as a “better TCP”.

Specification graph to support data channels

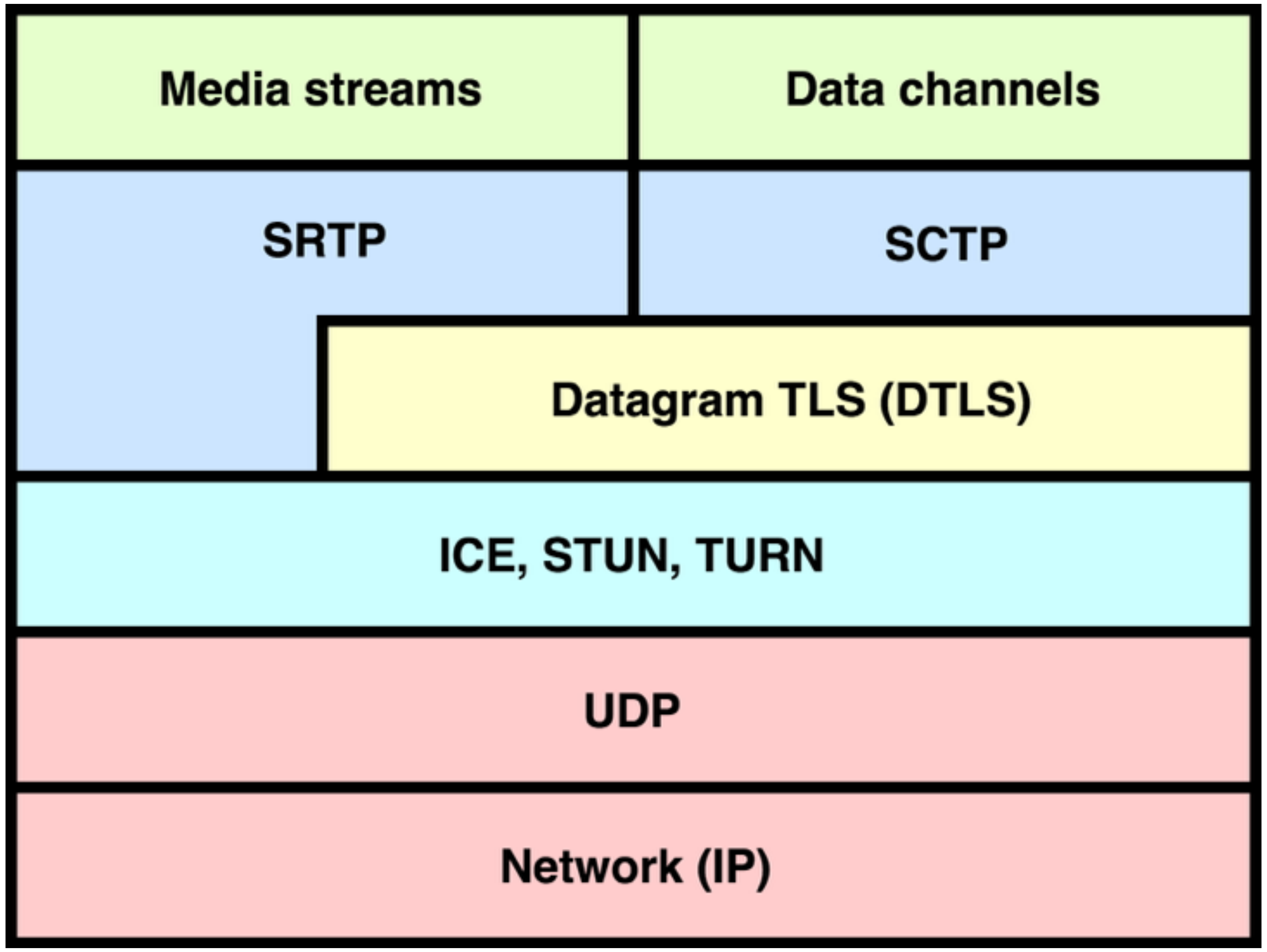
RFC 4960: Stream Control Transmission Protocol
RFC 3758: SCTP Partial Reliability Extension
RFC 7496: Additional Policies for the Partially Reliable SCTP Extension
RFC 6525: SCTP Stream Reconfiguration
RFC 5061: Dynamic Address Reconfiguration
RFC 4820: Padding Chunk and Parameter for SCTP
RFC 8260: Stream Schedulers and User Message Interleaving for SCTP
RFC 8445: Interactive Connectivity Establishment (ICE)
RFC 5389: Session Traversal Utilities for NAT (STUN)
RFC 5766: Traversal Using Relays around NAT (TURN)
RFC 4566: SDP: Session Description Protocol
RFC 3264: An Offer/Answer Model with SDP
draft-ietf-rtcweb-data-channel-13: WebRTC Data Channels
draft-ietf-rtcweb-data-protocol-09: WebRTC Data Channel Establishment Protocol
draft-ietf-mmusic-ice-sip-sdp-21: Session Description Protocol (SDP) Offer/Answer procedures for ICE
draft-ietf-rtcweb-jsep-24: JavaScript Session Establishment Protocol
draft-ietf-mmusic-sctp-sdp-26: SDP Offer/Answer Procedures For SCTP over DTLS Transport
draft-ietf-rtcweb-transports-17: Transports for WebRTC
draft-ietf-ice-trickle-21: Trickle ICE: Incremental Provisioning of Candidates for the ICE Protocol

845 pages!!!!1! 😨😧😦😱😫😩



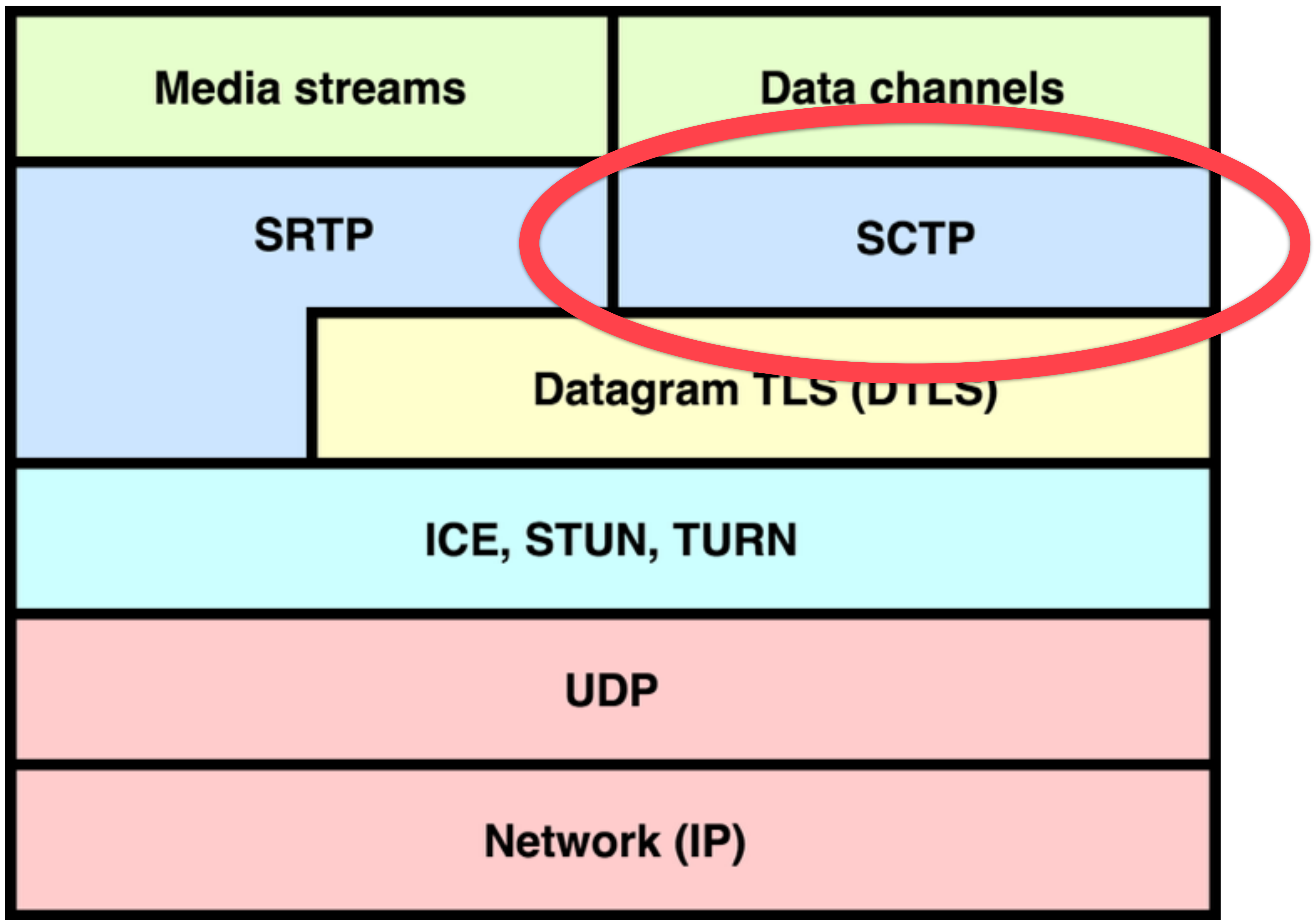
Picture: Kotivalo / Creative Commons

WebRTC protocol stack



Communicating with a peer is as simple as this.

WebRTC protocol stack



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What is SCTP?

- A transport-layer protocol that in theory is layered on top of IP, same as UDP or TCP.
- Invented in 2000 by the telecommunications industry.
    - When you make a phone call on your LTE phone, it's using SCTP to establish the call.
- More flexible than TCP: Configurable reliability, configurable ordered/unordered delivery, multiplexed streams, etc.
- Due to protocol ossification and poor support in OS and routers, SCTP has never been practical to use on the Internet as originally intended.
- WebRTC bypasses protocol ossification and OS support issues by encapsulating SCTP in UDP. (Actually, SCTP-over-DTLS-over-ICE-over-UDP.)

Why re-implement SCTP?

- Education
  - Experience with transport protocols
  - Experience with Rust networking (tokio, futures)
- Most everyone re-uses the same C-based SCTP implementation from FreeBSD (libusrsctp), and it's good to have options.
- A pure Rust implementation might (some day) prove more reliable than the C implementation.

Why not QUIC?

- It’s possible that QUIC may indeed eventually replace SCTP for WebRTC data needs.
- In its current form, QUIC may not provide the feature set needed for WebRTC. It may not be suitable for non-reliable or partially reliable data channels.
- There does appear to be work in progress on both the QUIC and WebRTC fronts to remove any obstacles to making QUIC a first-class citizen of the WebRTC world.
- Today, in 2018, you need SCTP for WebRTC. Current proposals for using QUIC in a WebRTC context seem to provide it as a separate bolt-on API, and not a direct replacement for the existing Data Channel feature.

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Set realistic goals

- Limit implementation to only the feature set need by WebRTC.
  - Don’t support multi-homing
  - Do support the SCTP extensions required by WebRTC
- Stick to stable/published tools and crates
- In the early stages, prefer boring code over cleverness
  - Use Box, .clone(), etc. for now and optimize later.
  - Simple, straightforward use of traits and lifetimes.
- Road map:
  1. Produce a minimal proof-of-concept to gain confidence
  2. Produce a correct implementation
  3. Optimize

Rusty tools

- Use nom to parse packets.
  - Is this really the most efficient way?
  - It would be nice to have a tool that would both parse and synthesize based on a format description.
- Tokio

Rapidly evolving crates

- `tokio-core → tokio`
  - Still using the legacy/deprecated tokio-core API
  - Tokio now defaults to a multi-threaded reactor, so futures must be thread-friendly (Rc → Arc, etc.)
- `futures 0.1 → ?`
  - Using the latest published futures crate — version 0.1
  - `futures` 0.2.x was briefly published, then yanked, to much drama.
  - When stable and published, migrate to 0.2, 0.3, async/await.
  - `futures::sync::mpsc` race condition in 0.1
- `tokio-timer 0.1 → 0.2`
  - 0.1 has 100ms clock granularity; 0.2 has 1ms granularity

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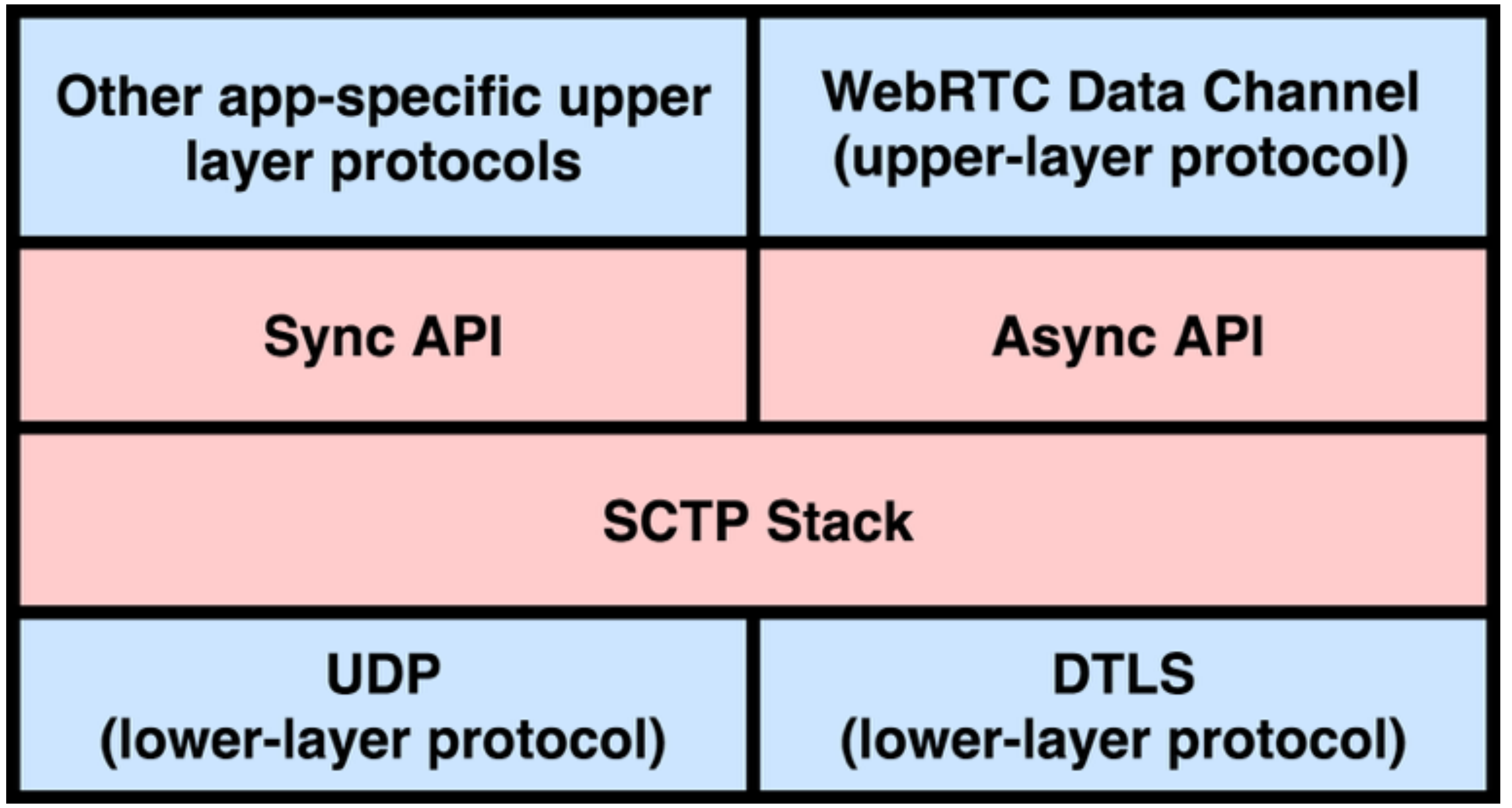
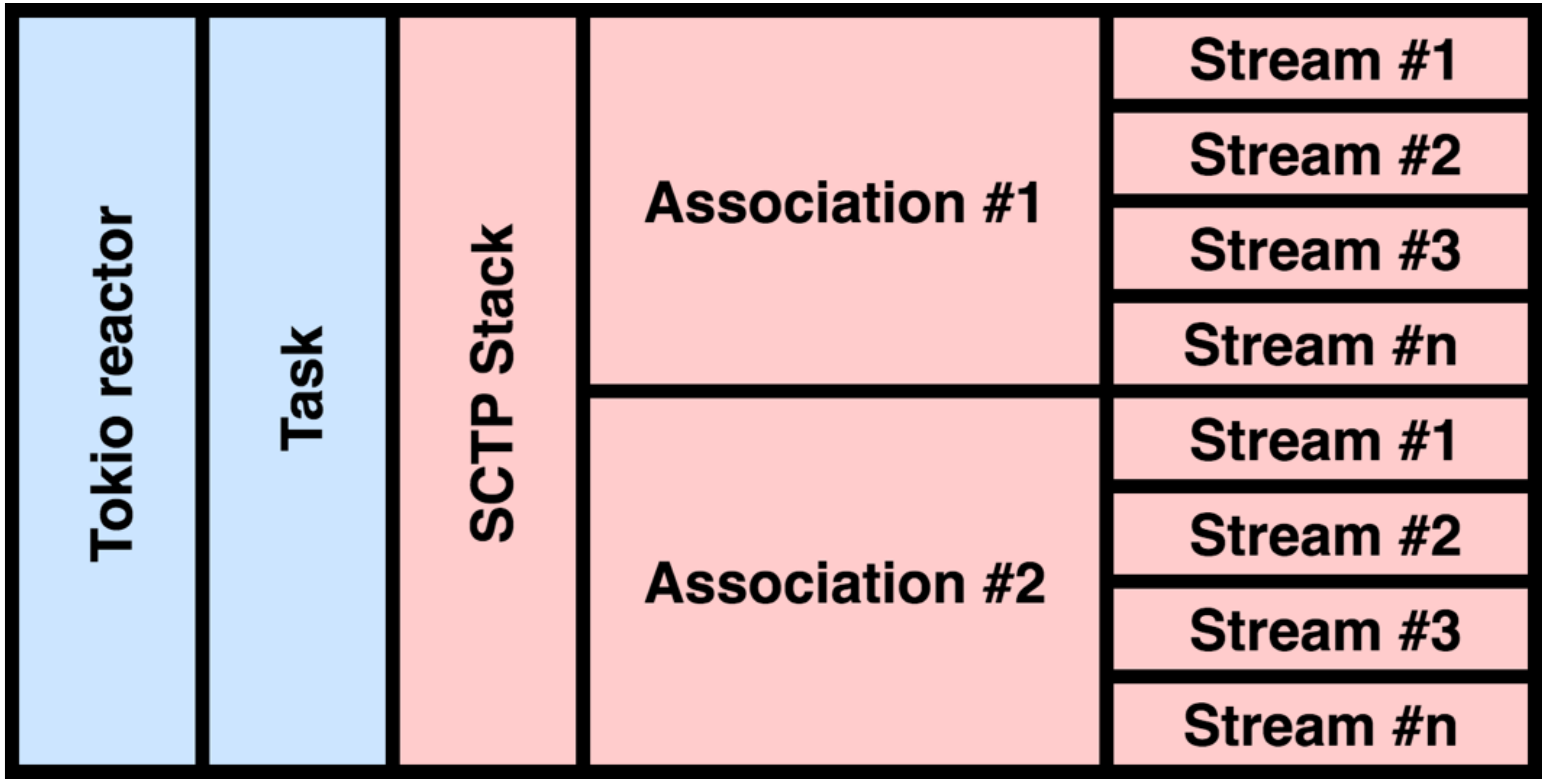
Stack design

- API
  - Asynchronous interface via Tokio
  - Synchronous API provided on top of the async API
- Configurable lower-layer protocol (LLP)
  - SCTP-over-UDP for testing with libusrsctp utilities
  - SCTP-over-DTLS-over-UDP for WebRTC



Futures design

- Currently, a single task is used for the SCTP stack and its association futures.



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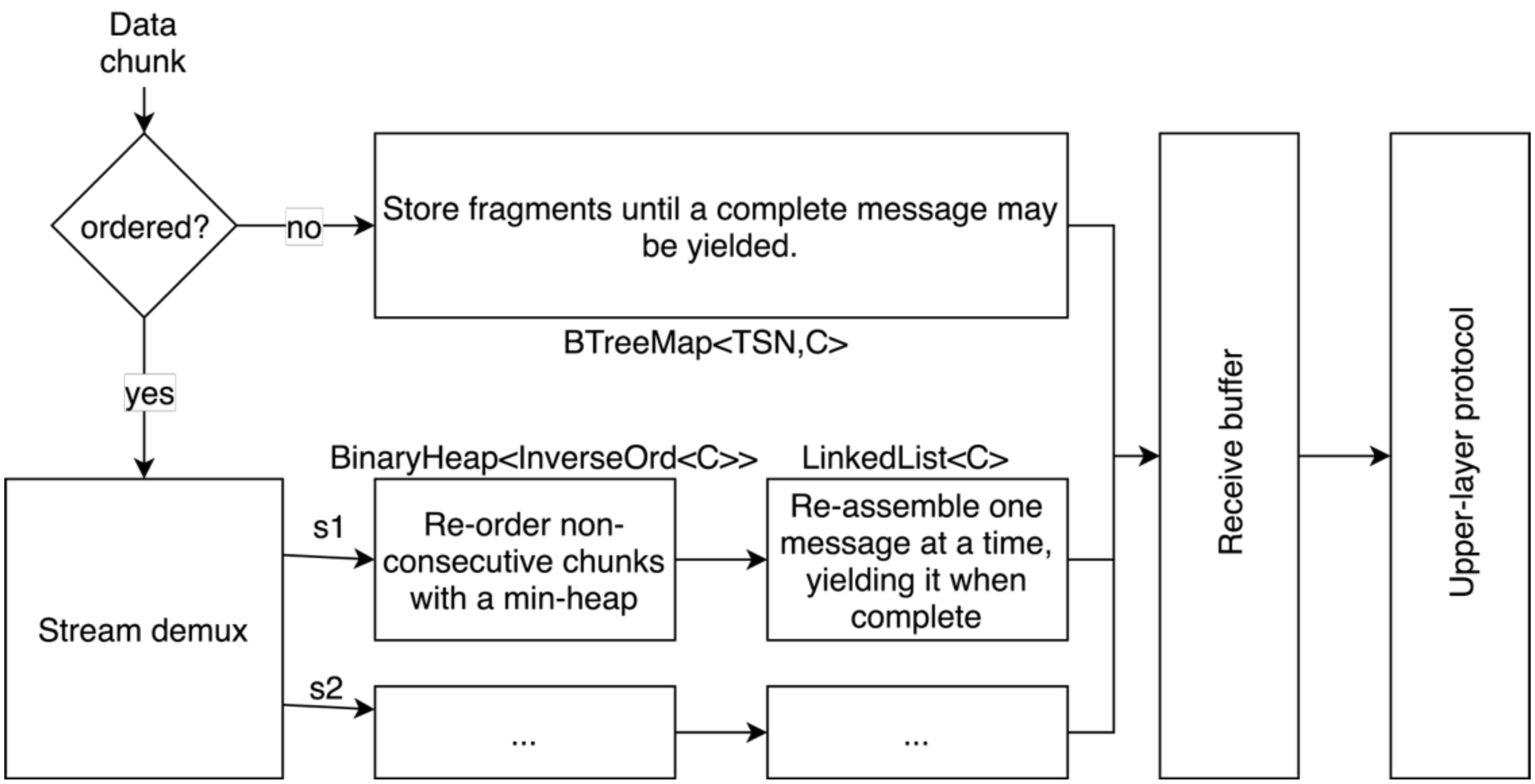
Associations

- Composing an association out of smaller futures or components is difficult due to the large and interconnected state.
- It's hard to break out cross-cutting concerns like retransmission and congestion control because they touch so many other parts of association state. (But this deserves more thinking!)
- Examples of association state:
  - SCTP association state (Established, CookieEcho, etc.)
  - Network 4-tuple (src/dst addresses/ports)
  - Stream counts
  - Queues: send, sent, receive, reassembly
  - Local and peer verification tags
  - Sending TSN, receiving TSN high water mark
  - Current calculated peer receiver window
  - Retransmission measurements
  - Numerous timers (timeouts, rtx, ...)
  - Many other things!

Message reassembly

- Multiplexing and optional ordering make reassembly a bit more complex than in TCP.



Memory efficiency

- The total payload size of all receive buffers (regardless of where they are enqueued) is tracked to enforce window size.
- Some buffers are reference counted for efficiency, but need to switch to the bytes crate.
- Memory for incoming packets is simply allocated on the heap.
- It turns out that even Linux heap-allocates for incoming data:

```c
224 	/* We do our best to align skb_shared_info on a separate cache
225 	 * line. It usually works because kmalloc(X > SMP_CACHE_BYTES) gives
226 	 * aligned memory blocks, unless SLUB/SLAB debug is enabled.
227 	 * Both skb->head and skb_shared_info are cache line aligned.
228 	 */
229 	size = SKB_DATA_ALIGN(size);
230 	size += SKB_DATA_ALIGN(sizeof(struct skb_shared_info));
231 	data = kmalloc_reserve(size, gfp_mask, node, &pfmemalloc);
```

— v4.12 net/core/skbuff.c line 231

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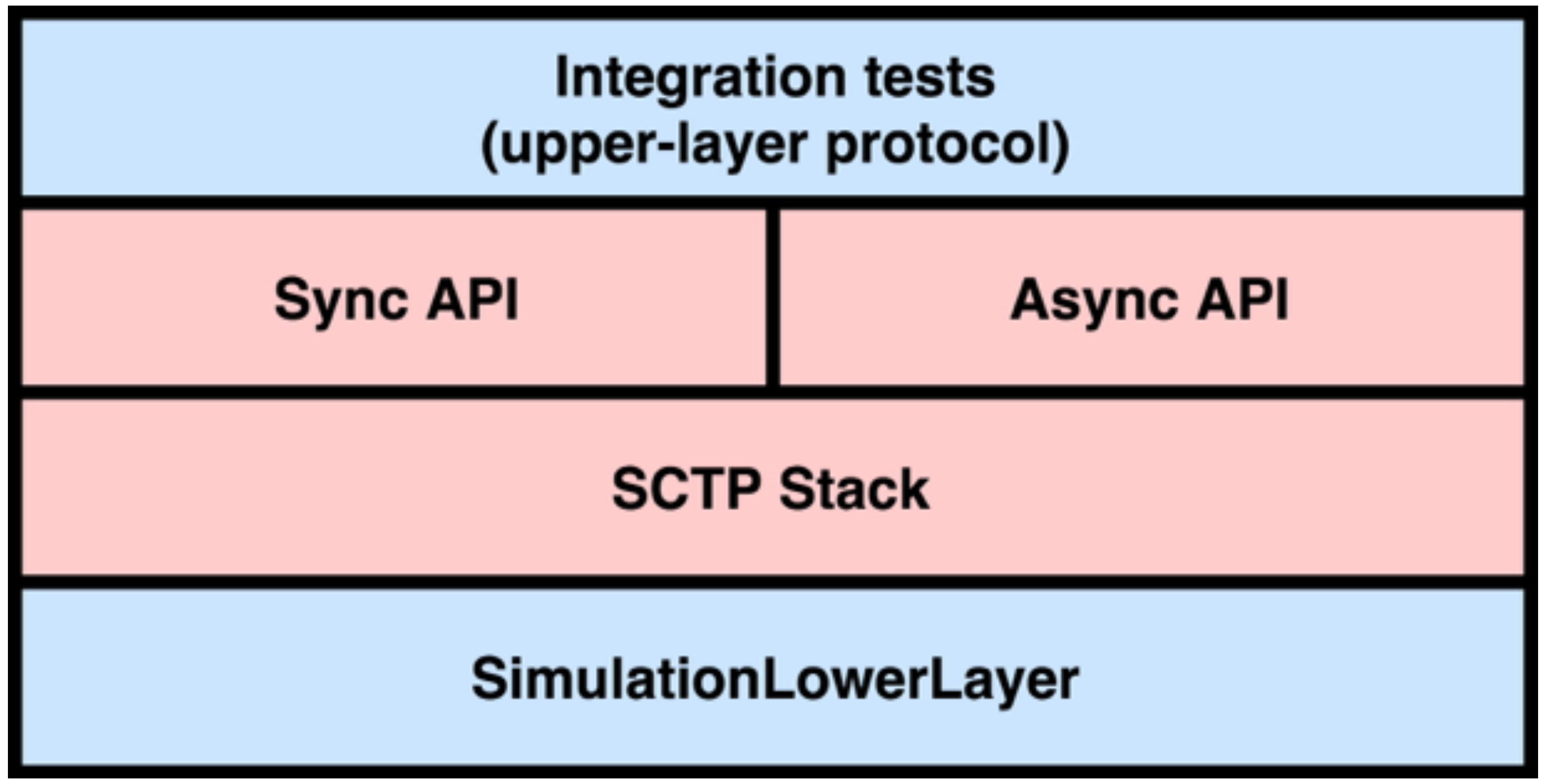
Unit tests

- There are lots of unit tests to cover the basics:
  - Parsing and synthesizing parameters, chunks, error causes, and packets.
  - Serial number arithmetic
  - Reassembly queues
  - Buffer management
- When testing with random values, seeded RNGs are used for reproducibility.
- More complex aspects such as the SCTP state machine cannot be easily tested in unit tests.

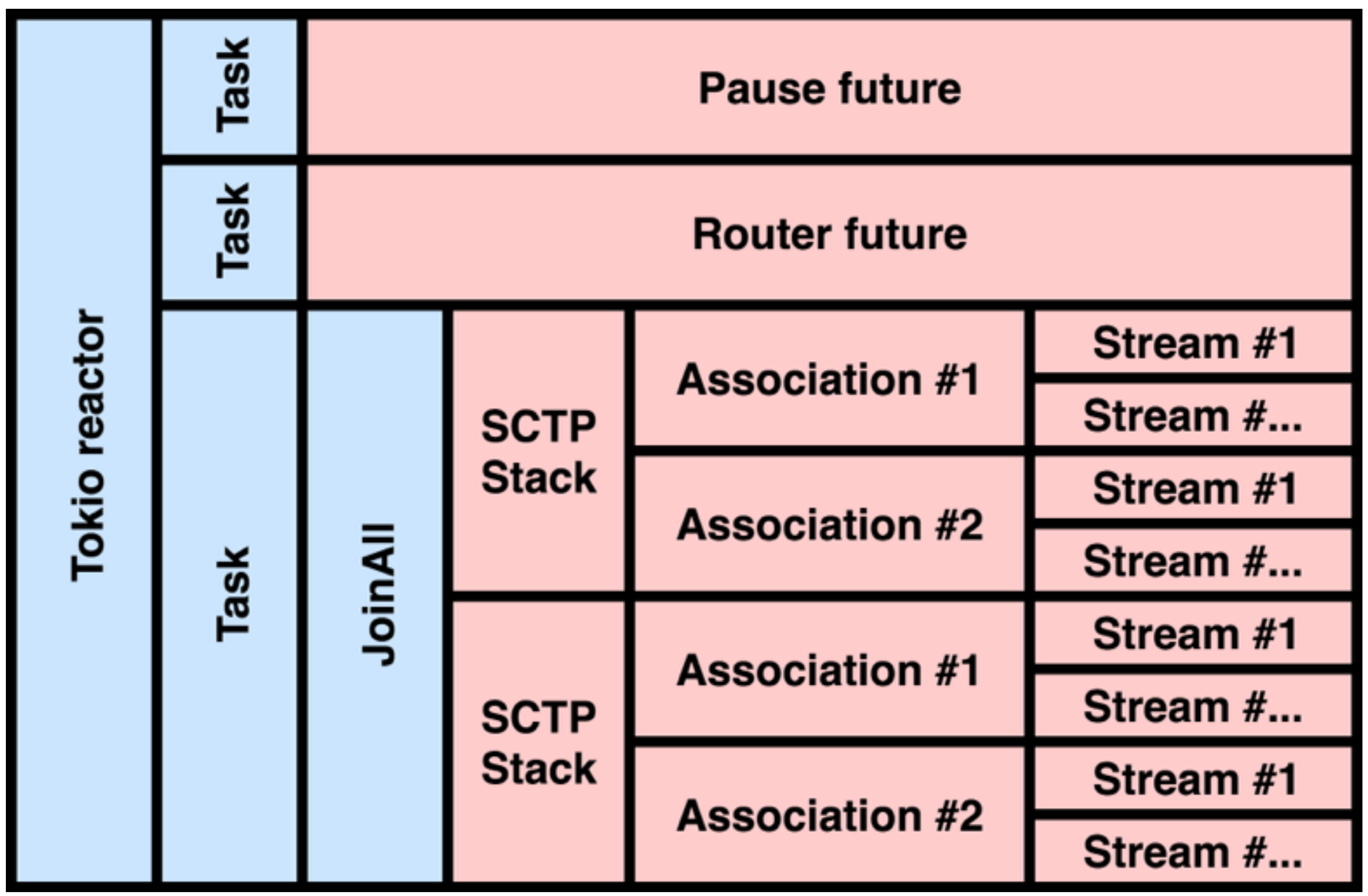
Simulation testing (1/2)

- A simulation framework supports integration tests where multiple stack instances communicate with each other.
- The downside to simulation tests is they only prove that the stack can interoperate with itself, not that it correctly implements SCTP.
- Simulations work by providing each stack with a custom lower-layer protocol that knows how to route packets in-memory:



Simulation testing (2/2)

- A pause/resume allows testing time-sensitive scenarios such as simultaneous shutdown.
- The arrangement of futures in the simulation looks like this:



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Risks

- Congestion control
  - If this isn't correct, it can cause grief for other network users.
- Bugs in transport protocols can leave systems vulnerable to denial-of-service attacks.


ZDNet

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Linux kernel bug: TCP flaw lets remote attackers stall devices with tiny DoS attack

'SegmentSmack' Linux bug gives a remote attacker the means to knock out a system with minimal traffic.

By Liam Tung | August 7, 2018 -- 10:16 GMT (03:16 PDT) | Topic: Security

Performance concerns

- Using UDP means performing a system call (and incurring the associated context switch costs) for every single datagram received and sent.
- Meltdown mitigations are likely to amplify this cost.
- Linux provides recvmmsg() and sendmmsg() system calls for operating on more than one datagram at a time. Maybe Mio could leverage this on Linux and any other operating systems that provide a similar API?

Debugging

- The complex nature of transport protocols allow for a great many bugs that the Rust compiler can't save us from, so testing and debugging will likely be a major effort even after functional completion.

Demo time!