Challenges with Layering: a Video Conferencing Case Study

COS 316: Principles of Computer System Design

Lecture 12

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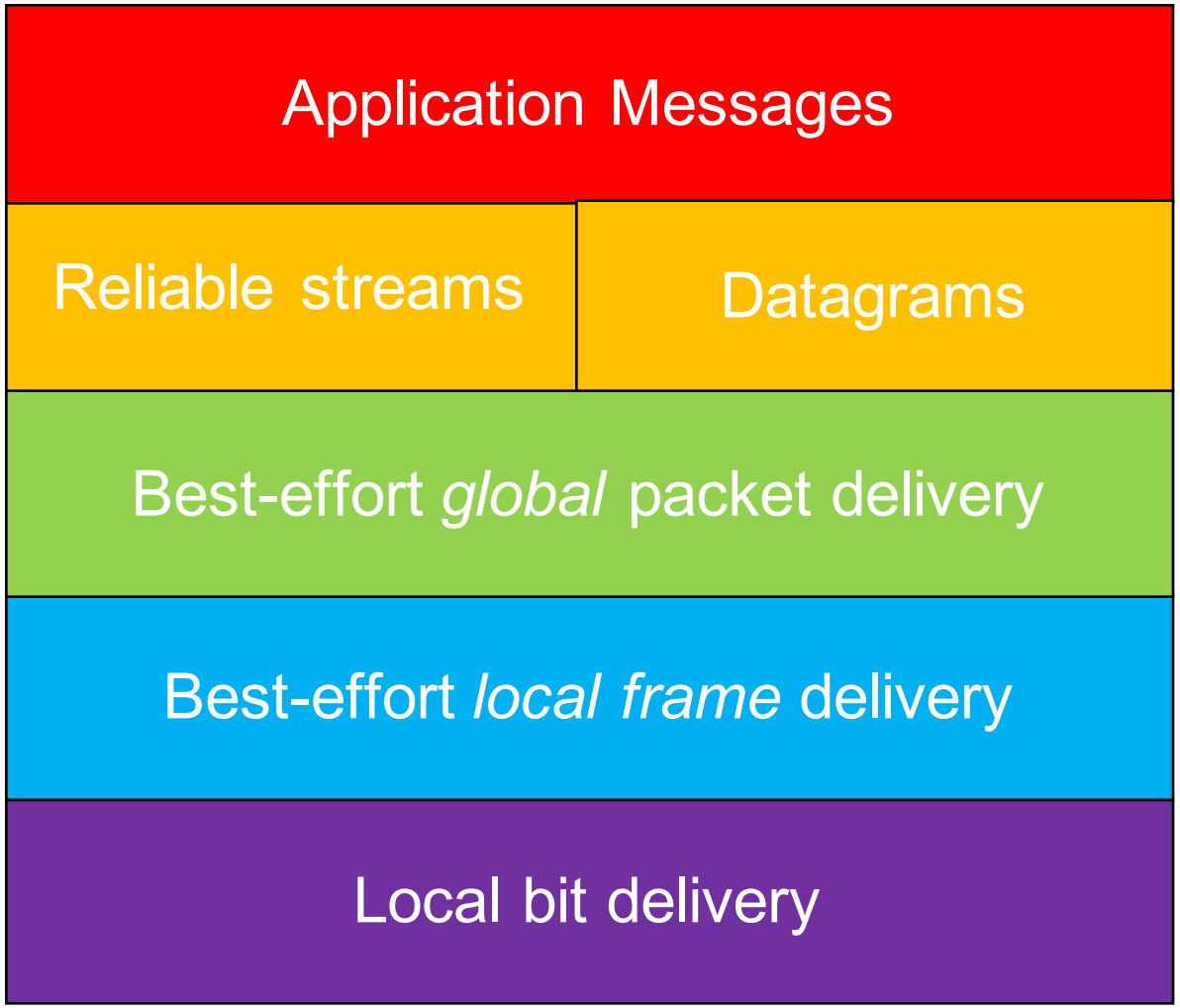
Modularity Through Layering

- Systems on systems on systems though layering
- Each layer hides complexity with abstraction
- Pro: simplifies design of each layer (and overall system)

Internet Protocol Layers



HTTP, SMTP, FTP, Skype, etc.

TCP, UDP

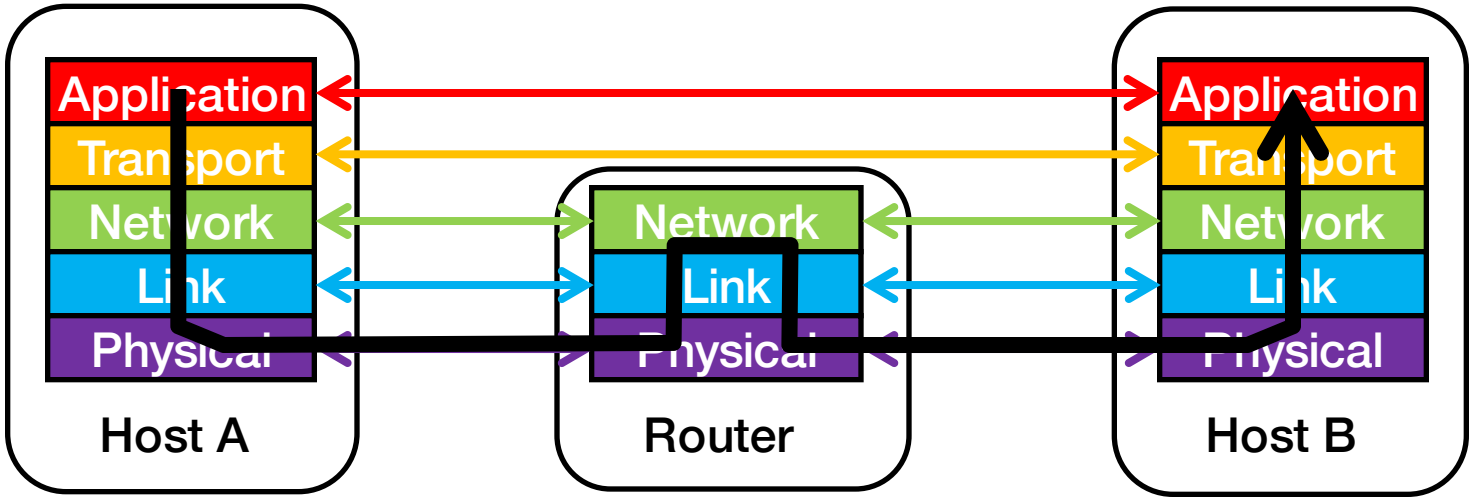
IP

Ethernet, WiFi, etc.

Coaxial cable, fiber optic cable, etc.

Physical communication

- Communication goes down to the **physical network**
- Then from **network** peer to peer
- Then up to the **relevant application**



Cons of Layering?

- Focus on generality rather than top-to-bottom optimality
- Lack of visibility across layers
- Potential lack of coordination across layers

Salsify:
Low-Latency Network Video Through Tighter Integration Between a Video Codec and a Transport Protocol

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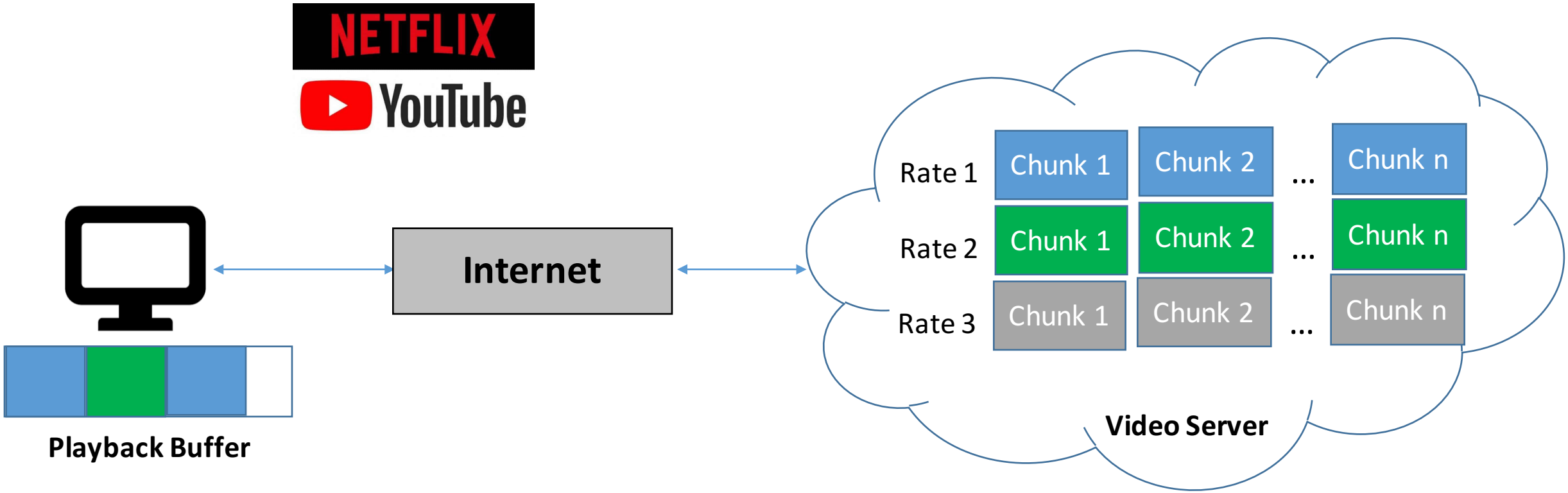
https://snr.stanford.edu/salsify

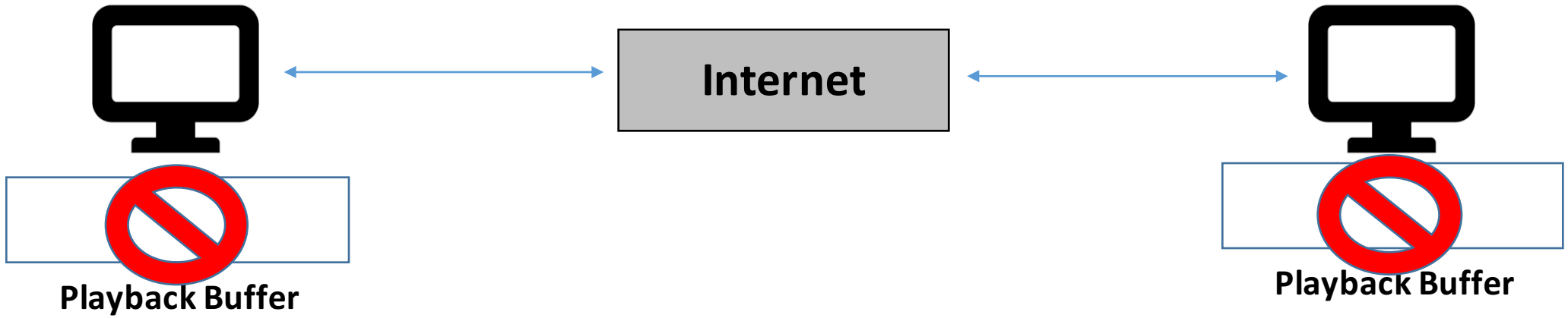
Stanford

Video Systems: Streaming



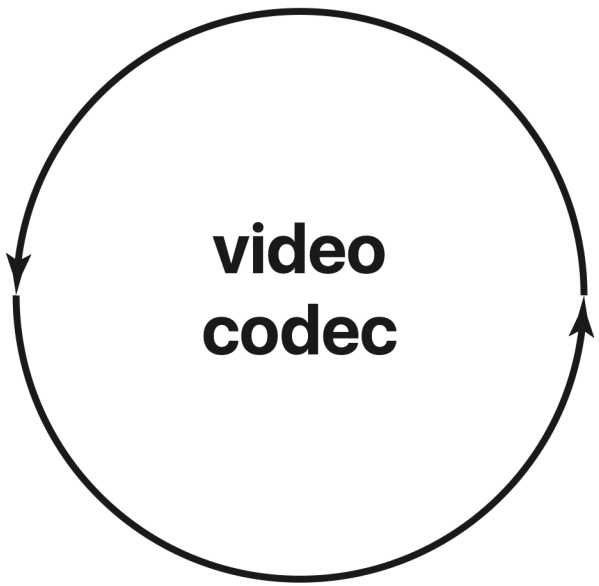
- Use **TCP**
  - Want reliability, not terribly latency-sensitive
- Video **encoded beforehand**
- Uses **playback buffer** (downloads ahead of time)
  - Can hide errors in network prediction

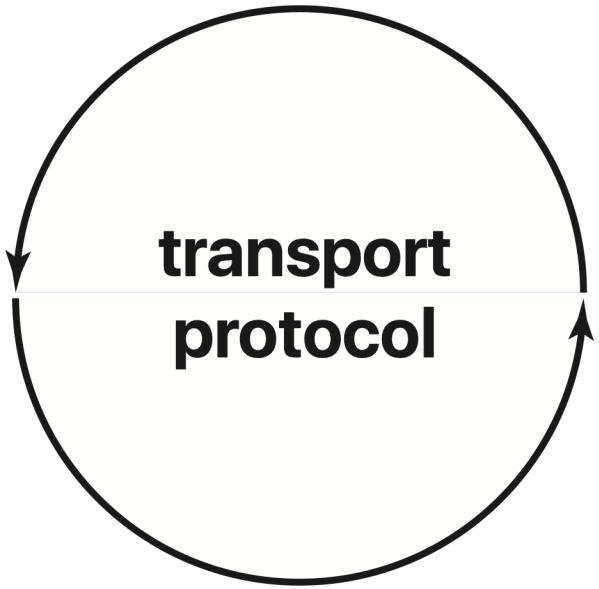
Video Systems: Conferencing



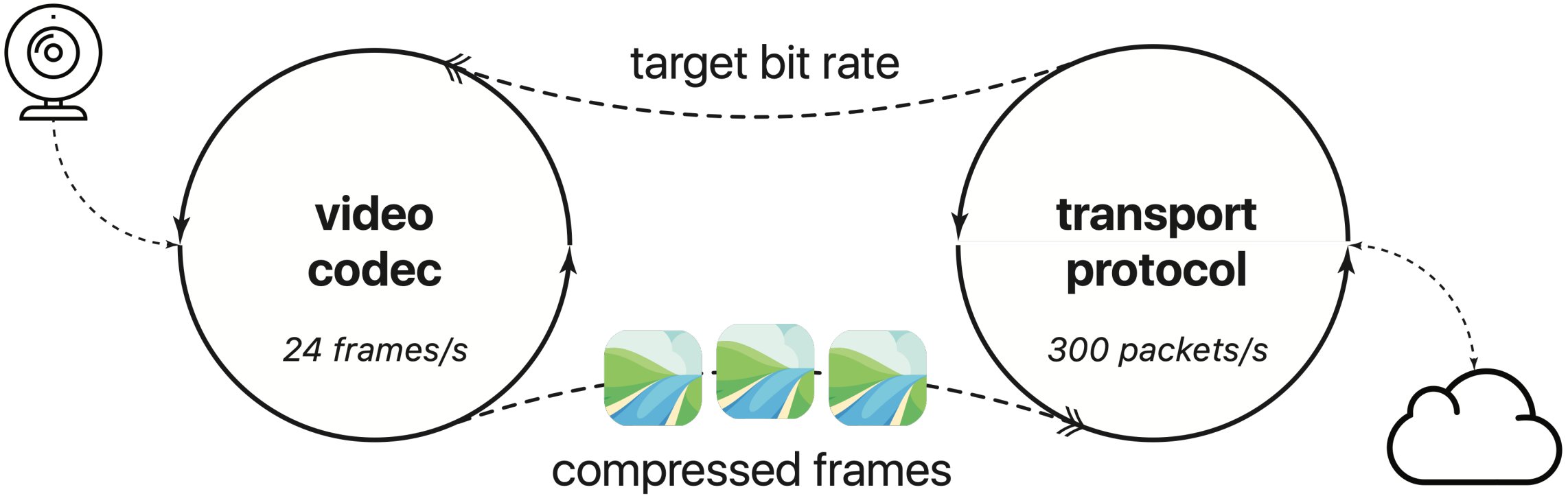
- Use **UDP**
  - Prioritize low latency over reliability
- Video **encoded on the fly**
- **No playback buffer**
  - Network prediction errors can be costly!

Today's systems combine two *(loosely-coupled)* components

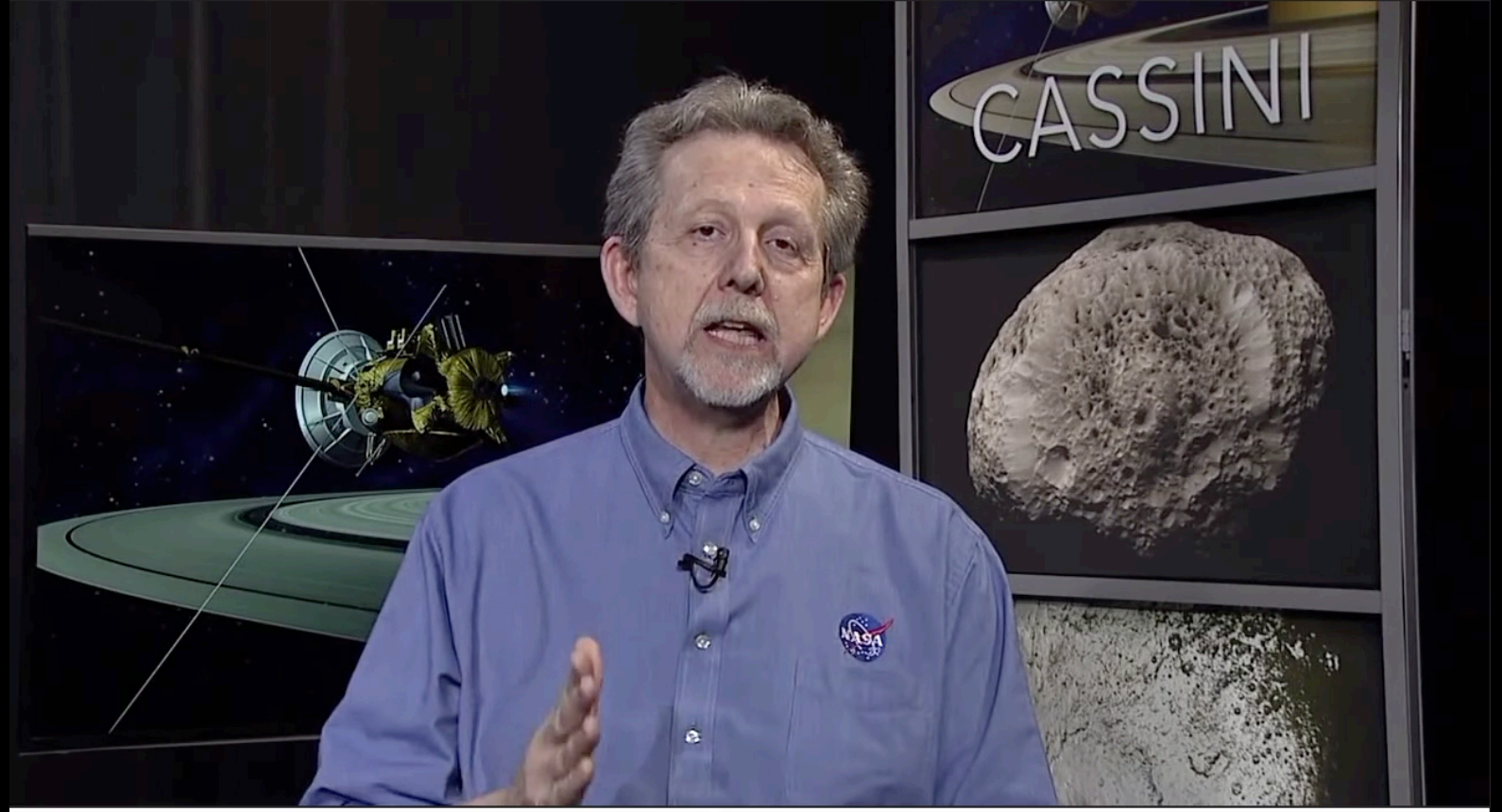


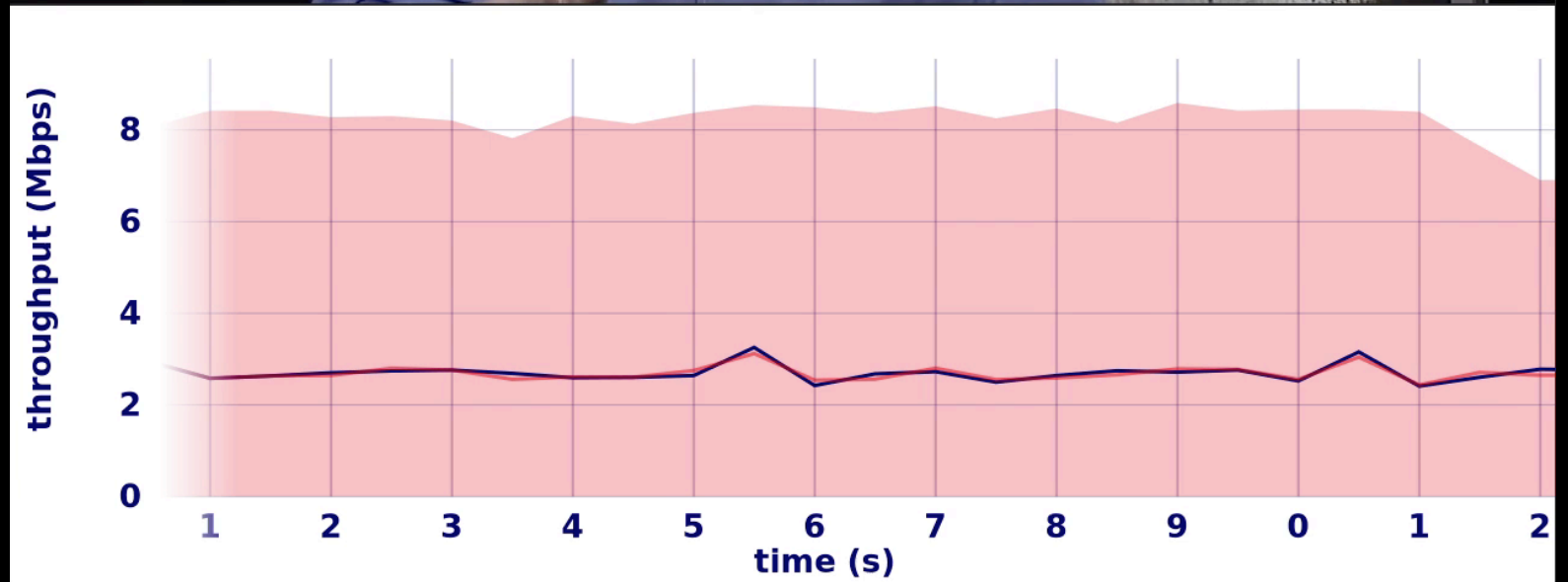


Two distinct modules, two separate control loops

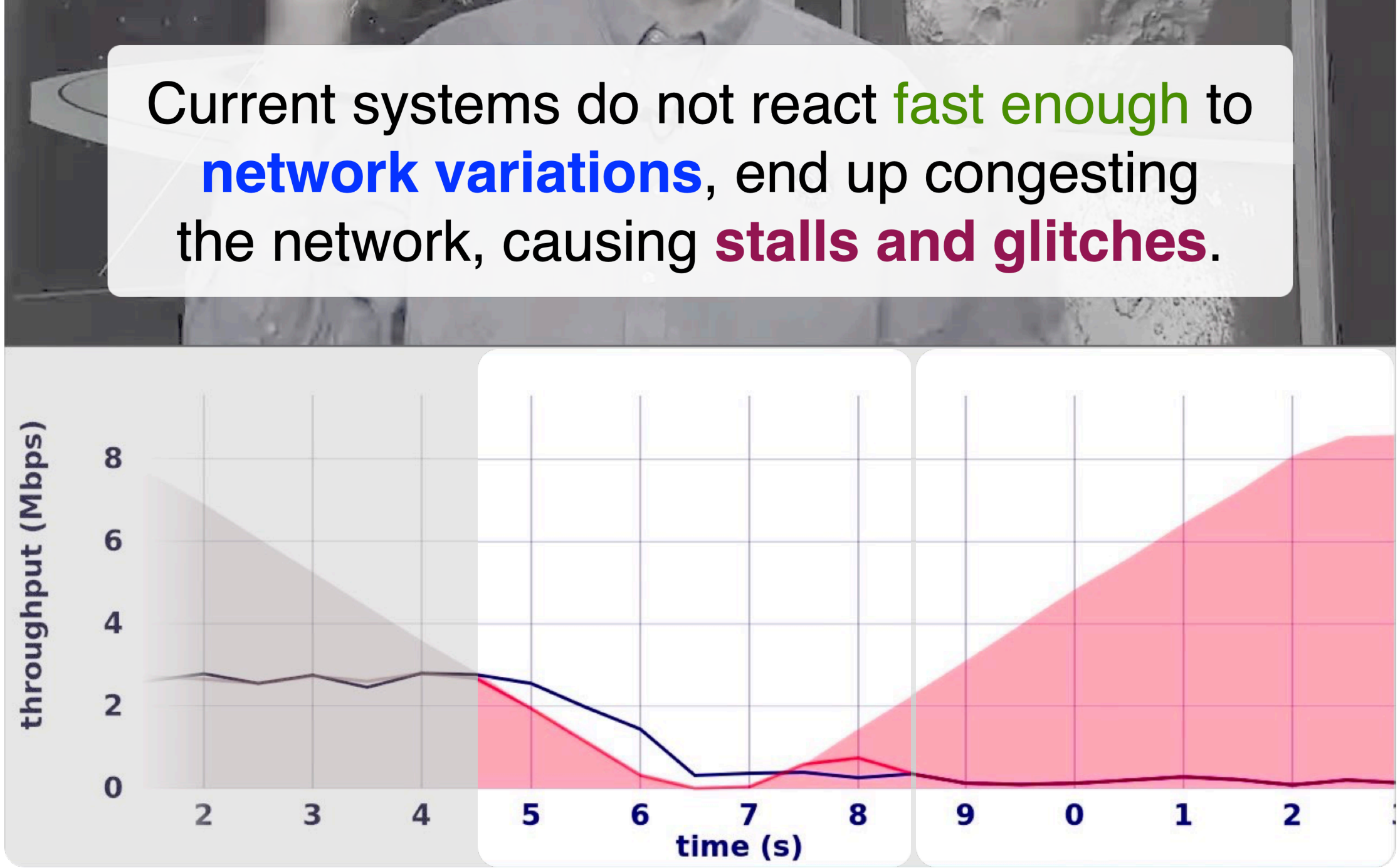


Existing conferencing systems already break layering we discussed! → expose network forecasts





WebRTC
(Chrome 65)


Current systems do not react fast enough to
network variations, end up congesting
the network, causing stalls and glitches.
throughput (Mbps)
8
6
4
2
0
2
3
4
5
6
7
8
9
0
1
2
time (s)

The problem: codec and transport are *too* decoupled

- The codec can only respond to changes in target bit rate over *coarse time intervals.*
  - Individual frames may cause packet loss/queueing.

- The transport has little control over what codec produces.

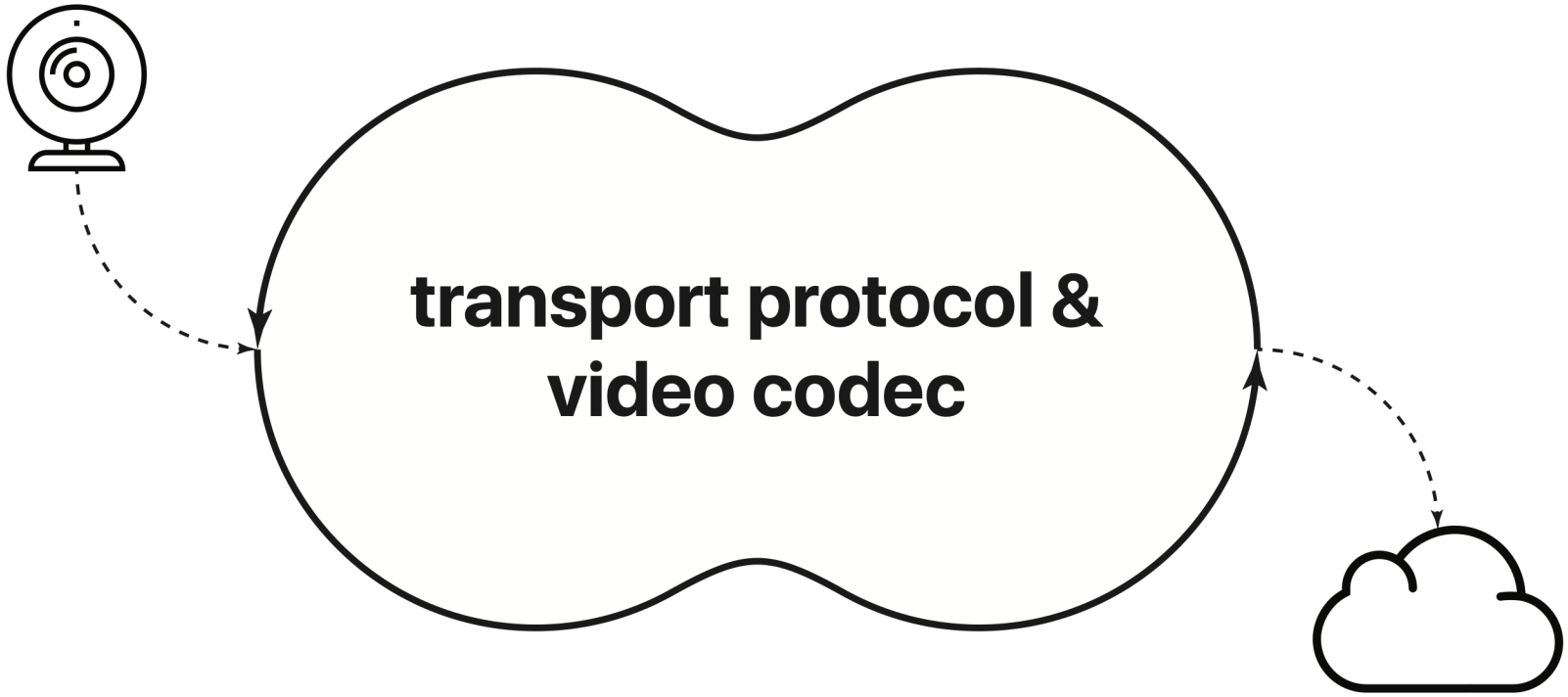
$\Rightarrow$ The resulting system is slow to react to network variations.

Enter *Salsify*

- Salsify is a new architecture for real-time Internet video.
  - Salsify tightly integrates a **video-aware transport protocol**, with a **functional video codec**, allowing it to **respond quickly to changing network conditions**.

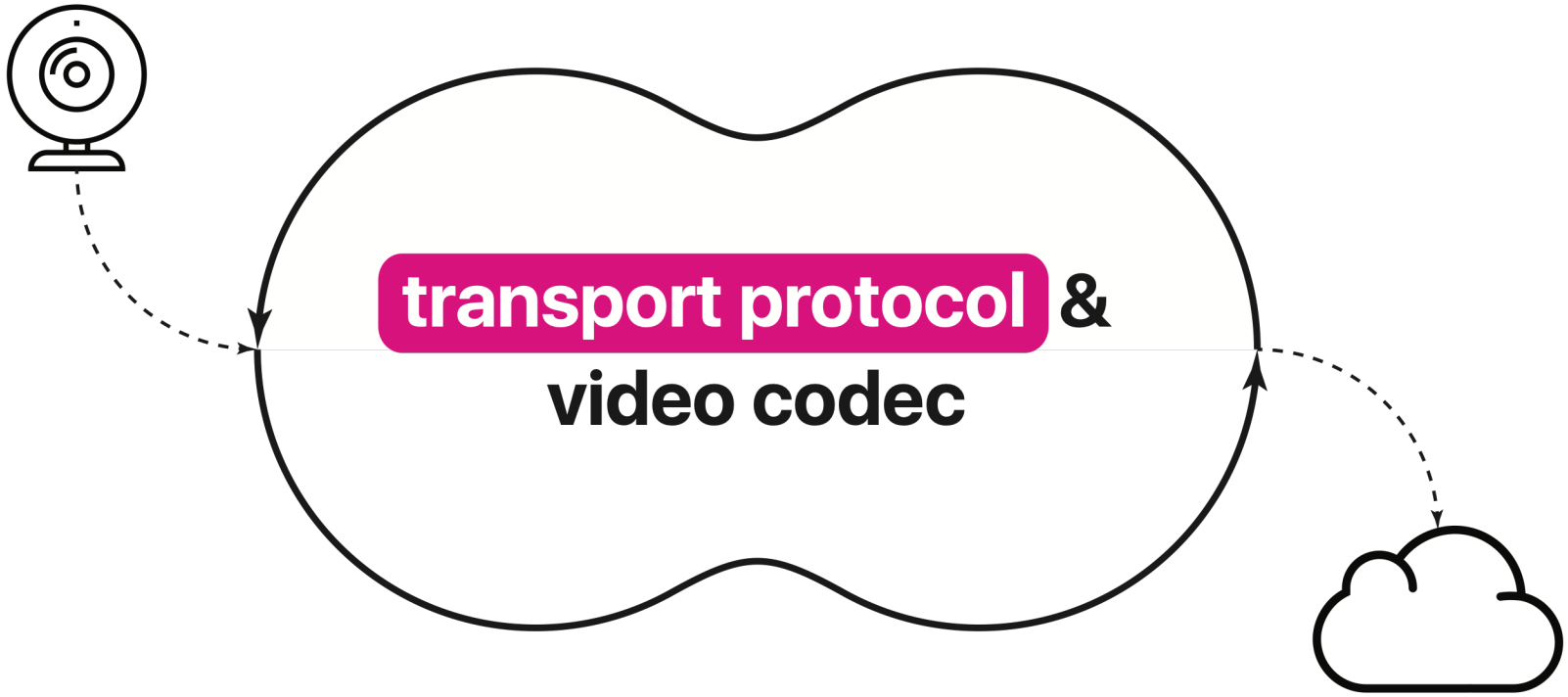
- Salsify achieves **4.6× lower p95-delay** and **2.1 dB SSIM higher visual quality** on average when compared with FaceTime, Hangouts, Skype, and WebRTC.

Salsify explores a more tightly-integrated design



Salsify's architecture:

Video-aware transport protocol



Video-aware transport protocol

“What should be the size of the next frame?”

*** without causing excessive delay**

- There’s no notion of bit rate, only the next frame size!
- Transport uses **packet inter-arrival time**, reported by the receiver.

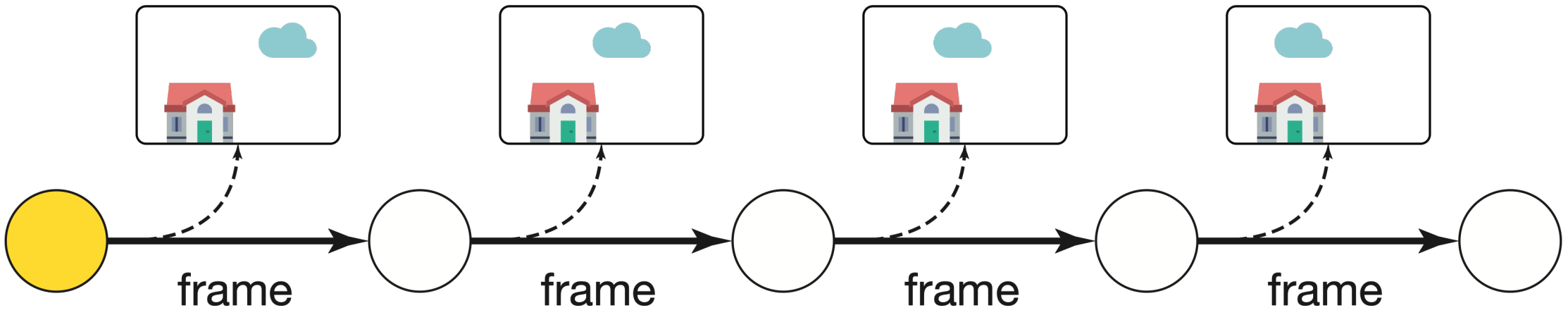
Transport tells us how big the next frame should be, but...

It’s challenging for **any codec** to choose the appropriate quality settings upfront to meet a **target size**—they tend to over-/undershoot the target.

How to get an accurate frame out of an inaccurate codec

- **Trial and error:** Encode with different quality settings, pick the one that fits.
  - ***Not possible with existing codecs.***

After encoding a frame, the encoder goes through a state transition that is impossible to undo



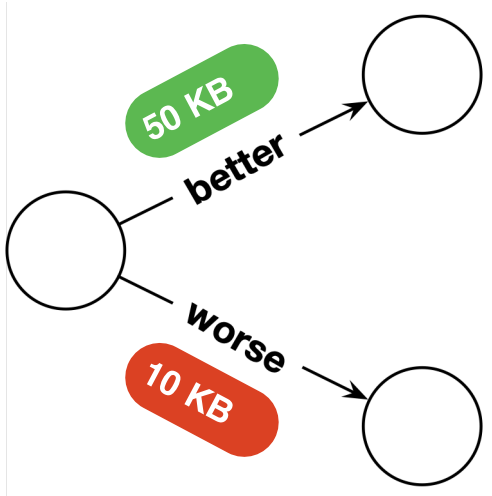
Functional video codec to the rescue

encode(*state*, ˋâ) → *state′*, frame

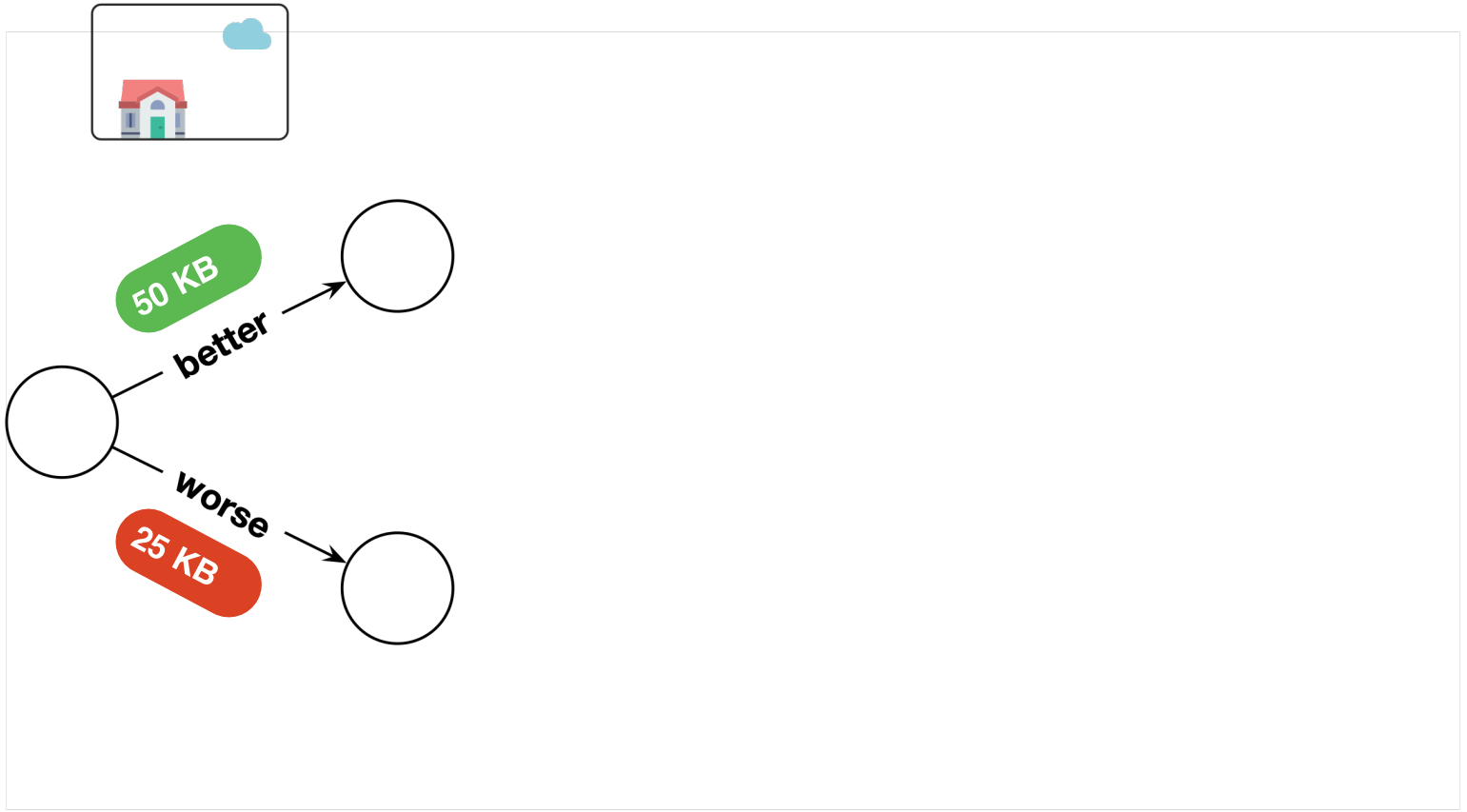
Salsify's functional video codec exposes the state that can be saved/restored.

Order two, pick the one that fits!

- Salsify's functional video codec can **explore different execution paths** without committing to them.
- For each frame, codec presents the transport with *three* options:
  - ▲ A slightly-higher-quality version,
  - ▼ A slightly-lower-quality version,
  - ✕ Discarding the frame.

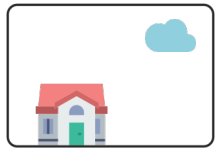


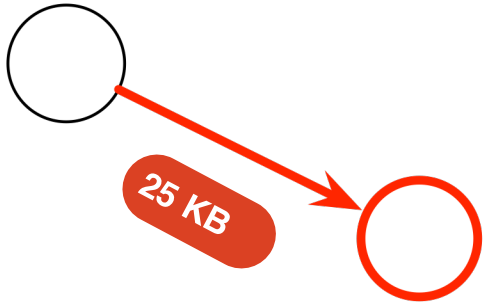
Codec → Transport
“Here’s two versions of the current frame.”



target frame size 30 KB

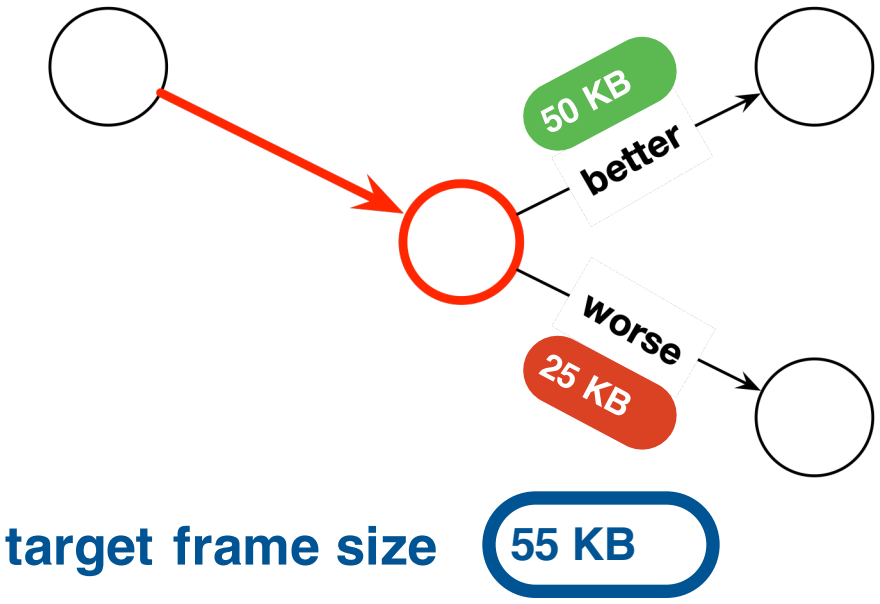
Transport → Codec
“I picked #2. Base the next frame on its state.”



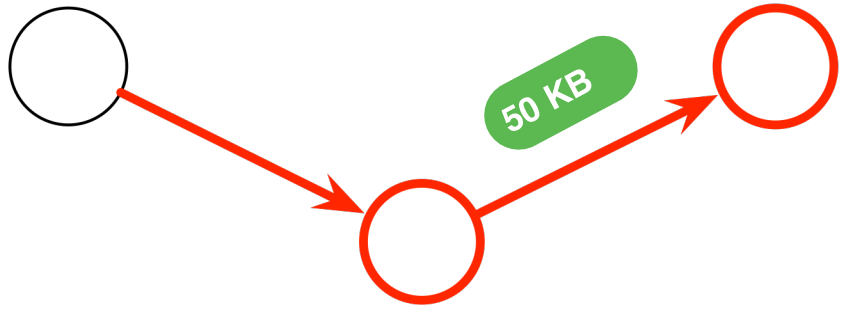


target frame size 30 KB

Codec → Transport
“Here’s two versions of the latest frame.”

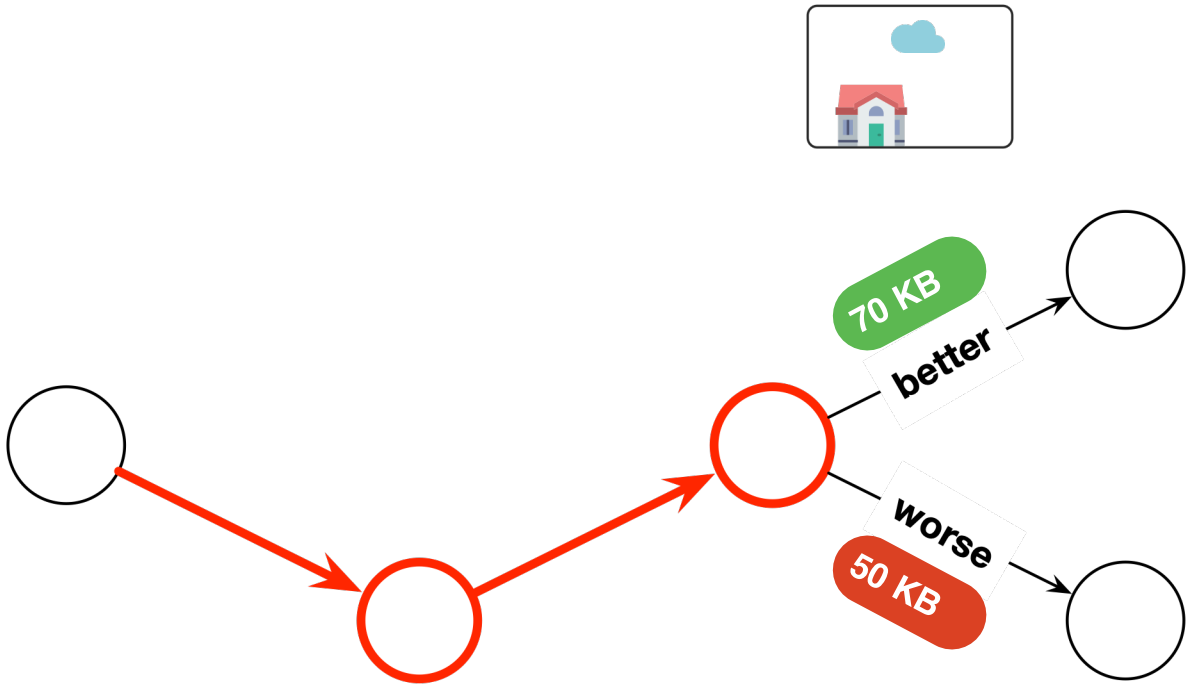


Transport ⟶ Codec
“I picked #1. Base next frame on its state.”



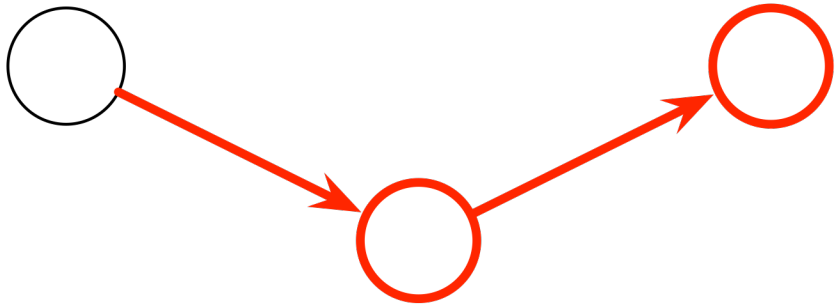
target frame size 55 KB

Codec → Transport
“Here’s two versions of the latest frame.”



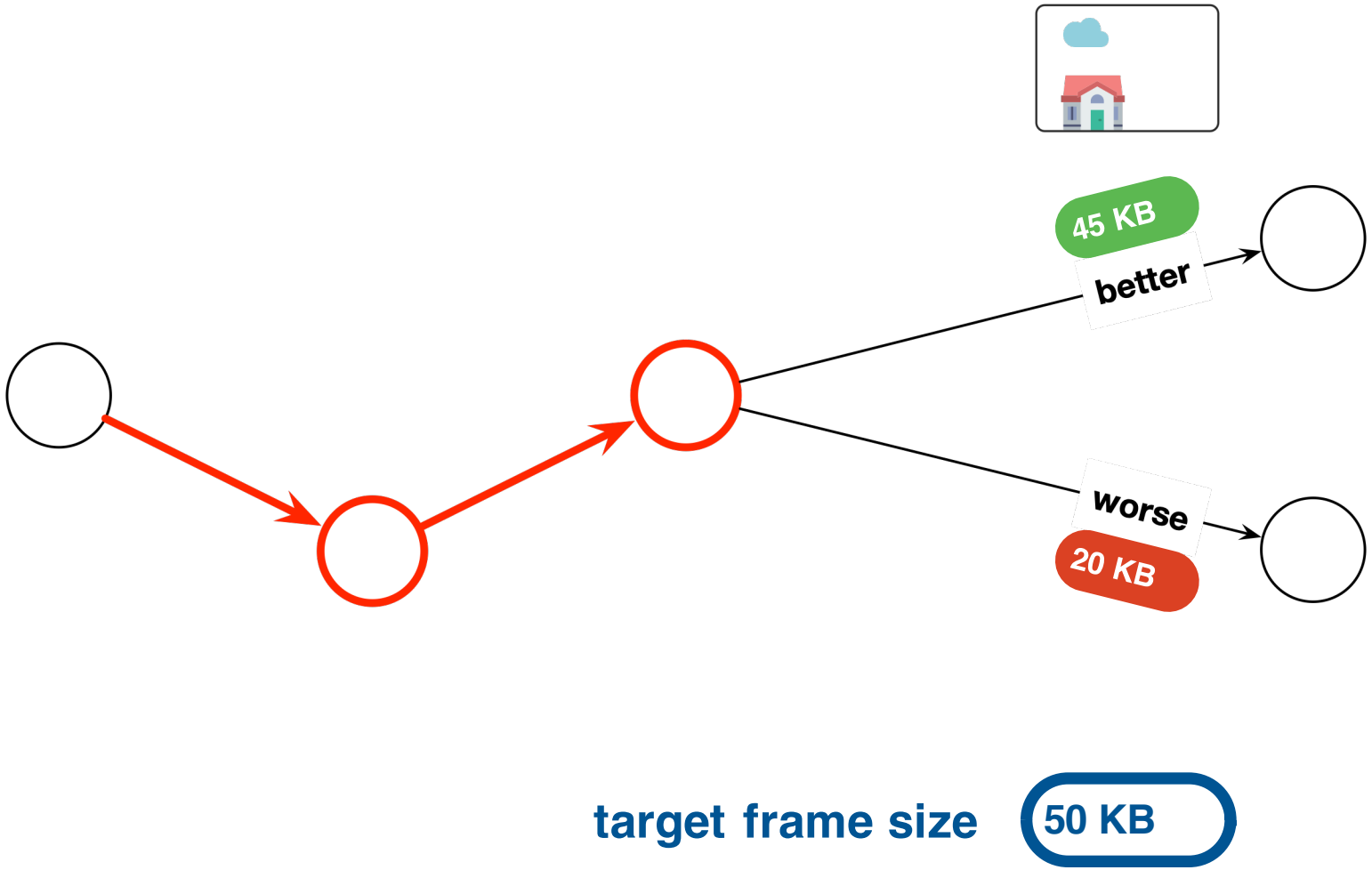
target frame size 5 KB

Transport ⟶ Codec
“Can’t send frames right now. Discard them.”

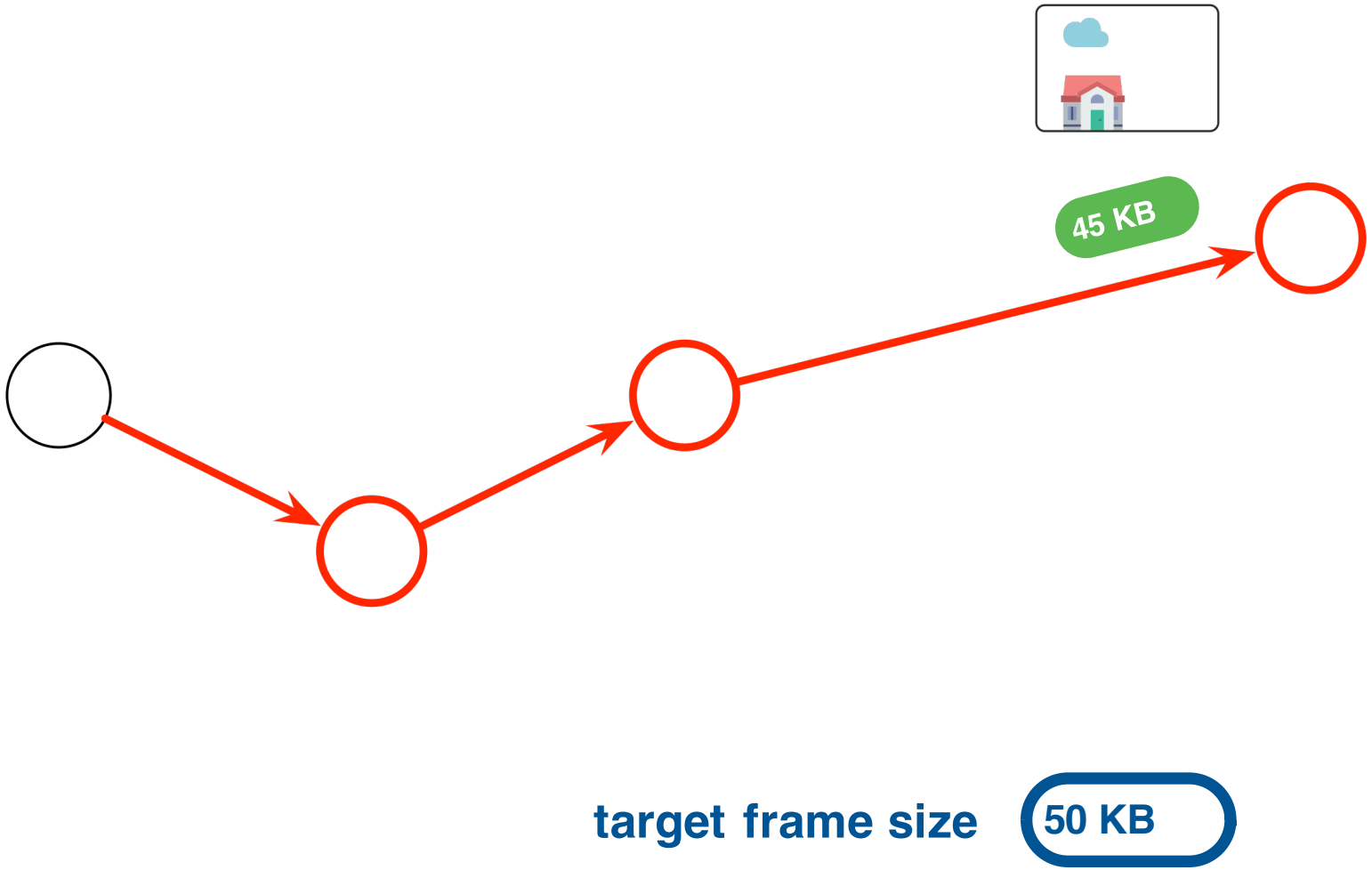


target frame size 5 KB

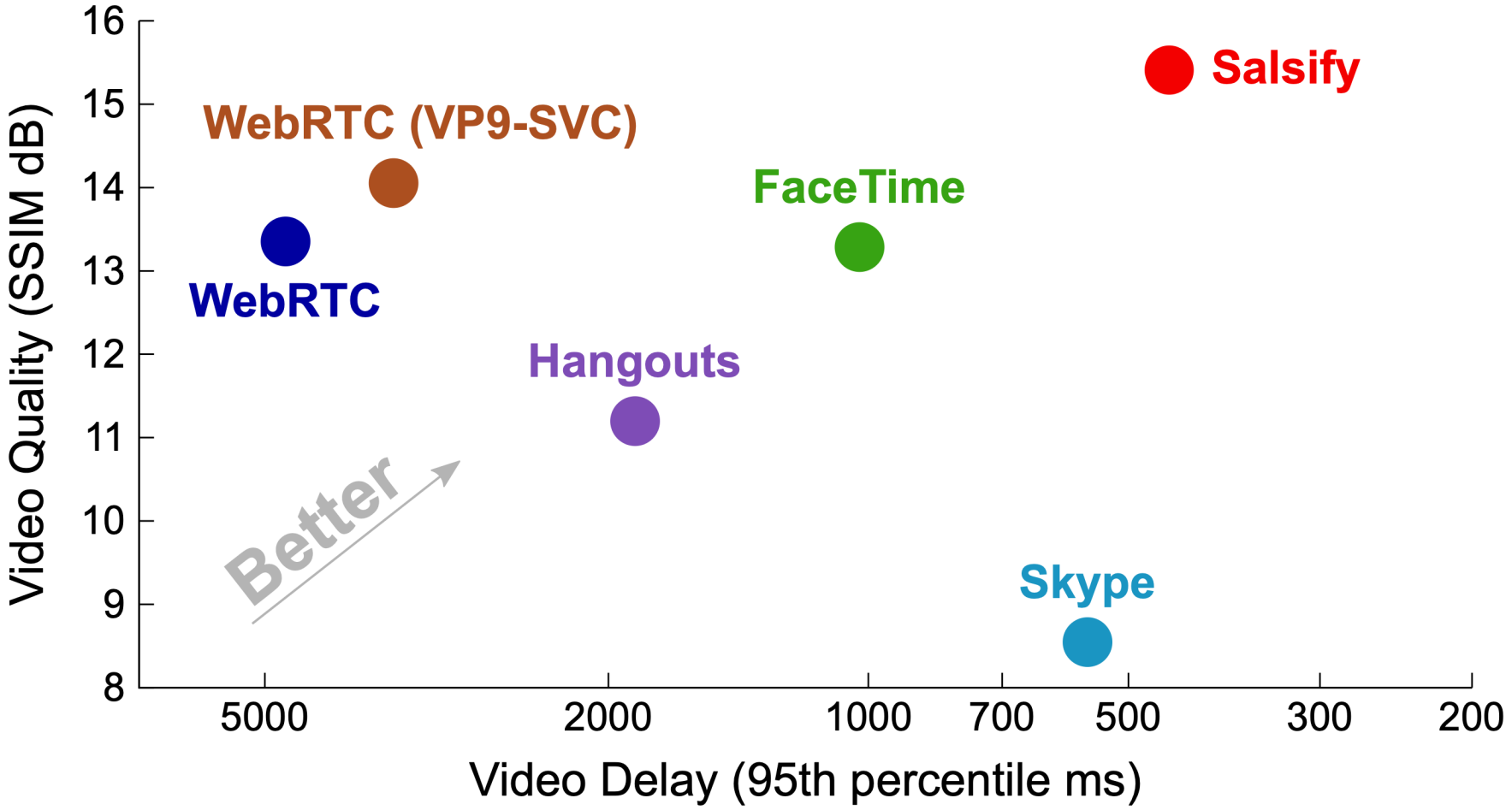
Codec → Transport
“Fine. Here’s 2 versions of the latest frame.”



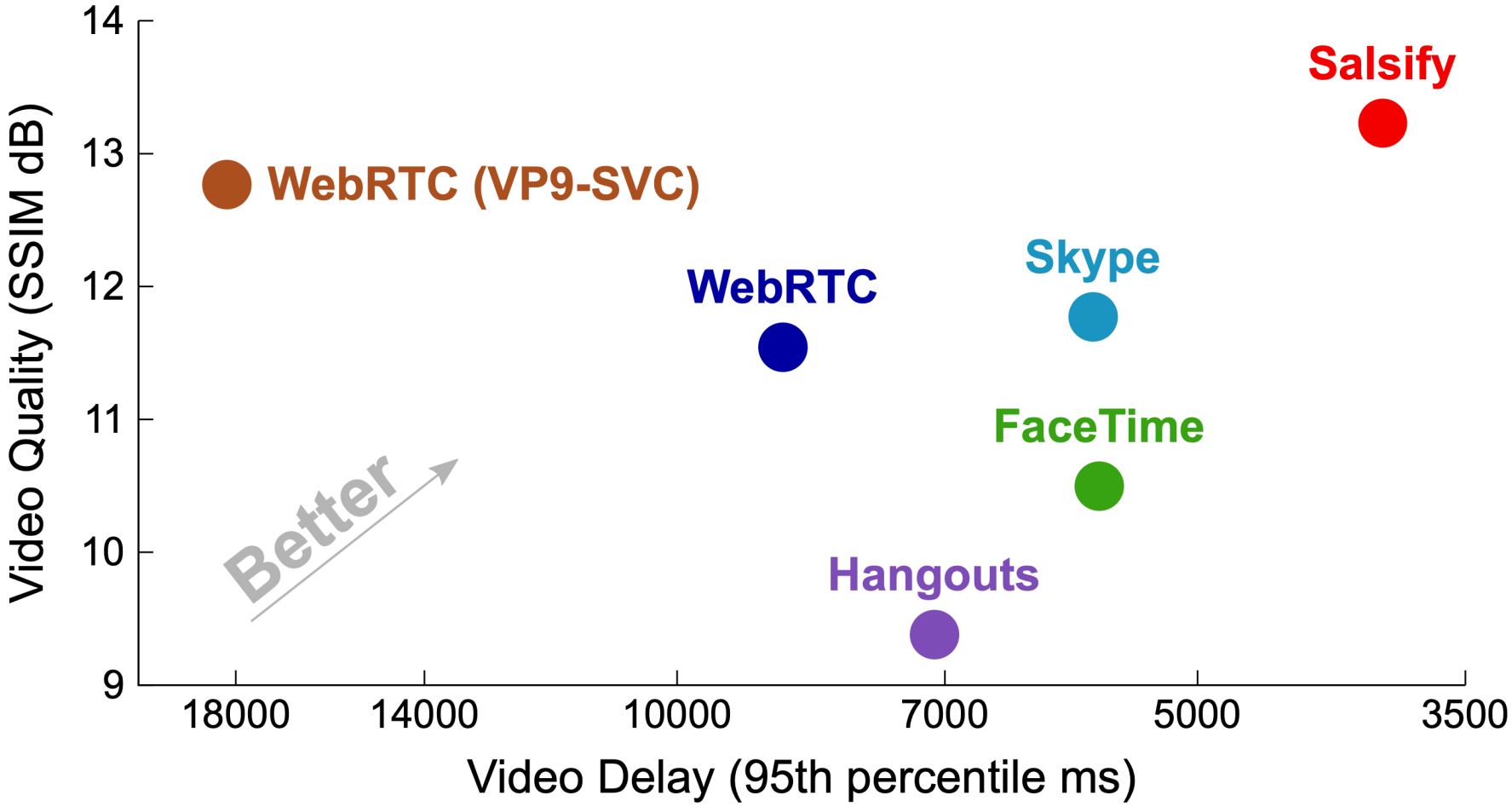
Transport ⟶ Codec
“I picked #1. Base next frame on its state.”



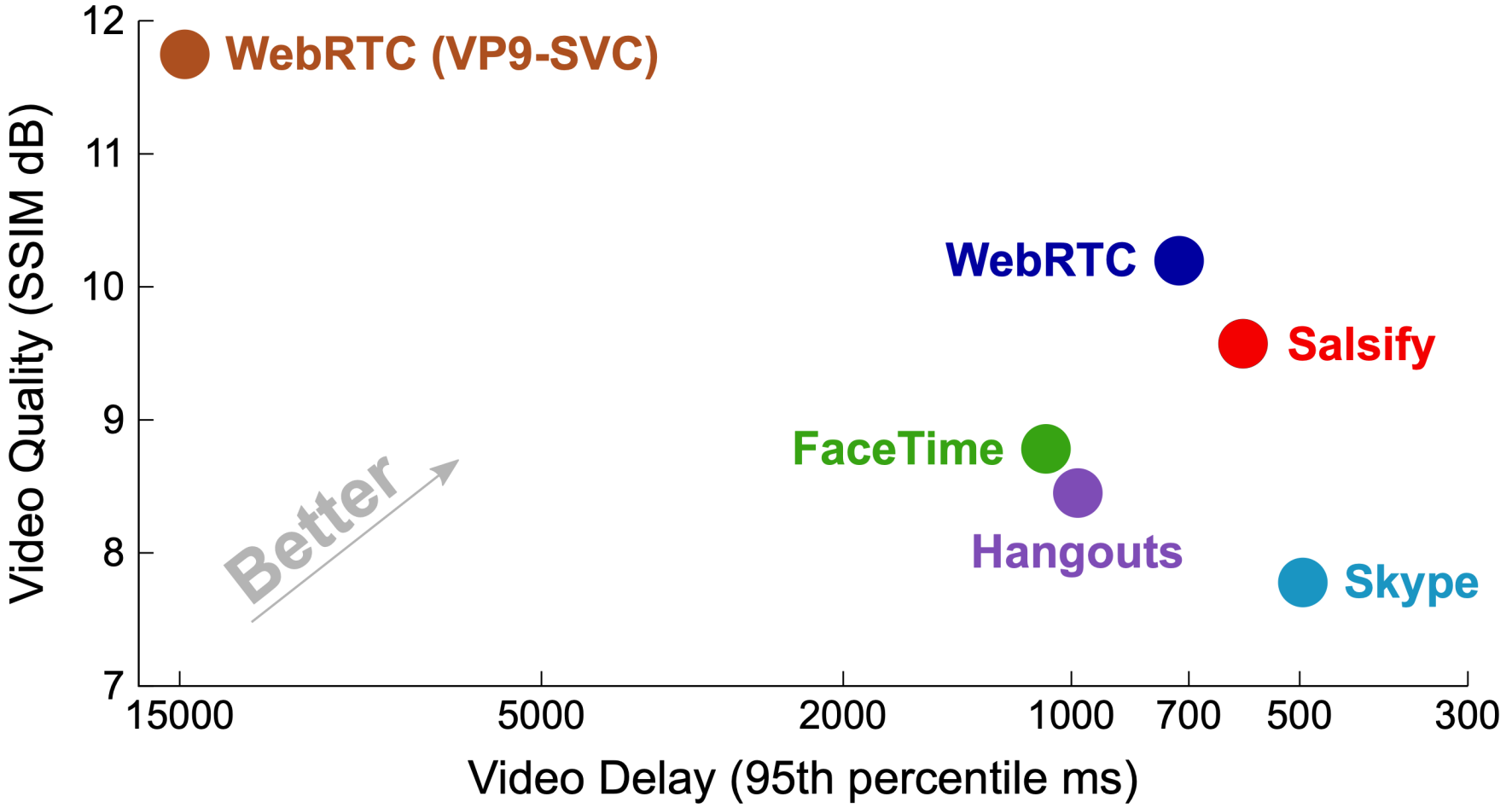
Evaluation results: AT&T LTE Trace



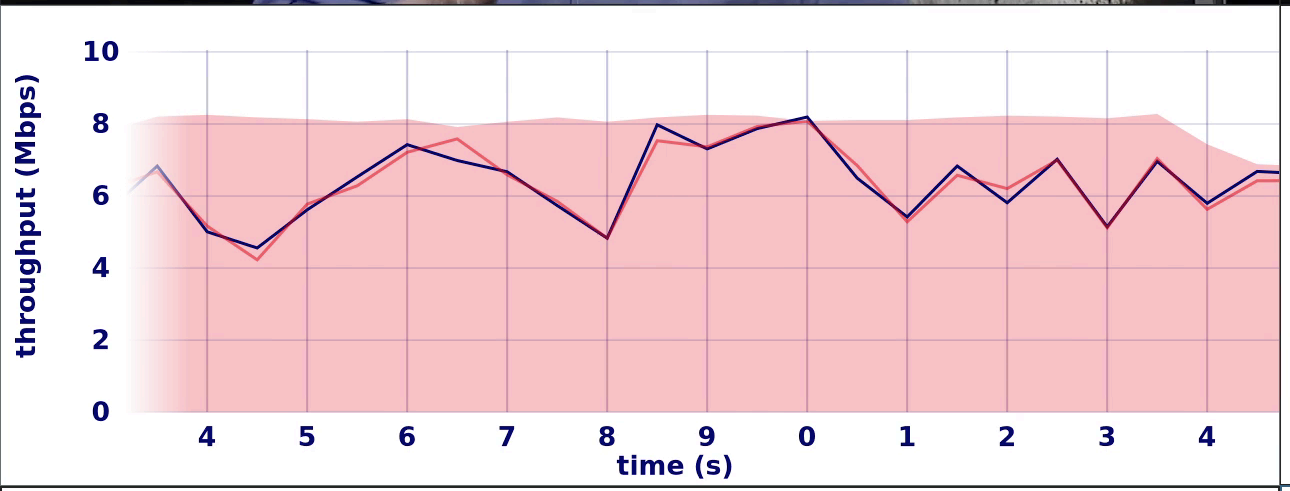
Evaluation results: T-Mobile UMTS Trace



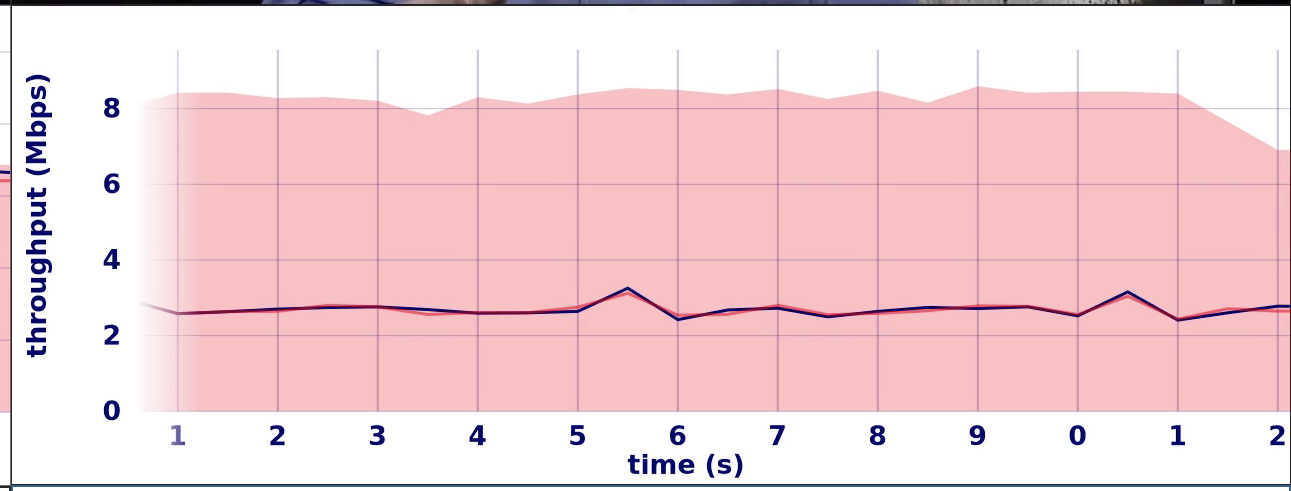
Evaluation results: **Emulated Wi-Fi (no variations, only loss)**



Network Variations



Salsify



WebRTC (Chrome 65)

Salsify, a new heart from old parts

- Individual component of Salsify are not exactly new:
  - The transport protocol is inspired by “packet pair” and “Sprout-EWMA”.
  - The video format, VP8, was finalized in 2008.
  - The functional video codec was described at NSDI’17.

- Salsify is a **new architecture** for real-time video that integrates these components in a way that responds quickly to network variations.

Improvements to *video codecs* may have reached the point of diminishing returns, but changes to the architecture of *video systems* can still yield significant benefits.

Takeaways

- Salsify is a new architecture for real-time Internet video.
  - Salsify tightly integrates a **video-aware transport protocol**, with a **functional video codec**, allowing it to **respond quickly to changing network conditions**.
- Salsify achieves **4.6× lower p95-delay** and **2.1 dB SSIM higher visual quality** on average when compared with FaceTime, Hangouts, Skype, and WebRTC.
- The code is open-source, and the paper and raw data are open-access:
  https://snr.stanford.edu/salsify

Breaking Layers!

- **Expose more information to application**
  - WebRTC (and other transport protocols like Sprout) share forecasts of what they think network can support based on ACKs, etc.

- **Coordinate the control loops**
  - Transport and application layers each have their own designs and flaws; jointly operating them helps mask those issues
  - 2 big techniques: (1) late-binding decisions and (2) test alternatives