Tambur: efficient loss recovery for videoconferencing via streaming codes

Presented by **Michael Rudow** at NSDI '23

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Carnegie Mellon University

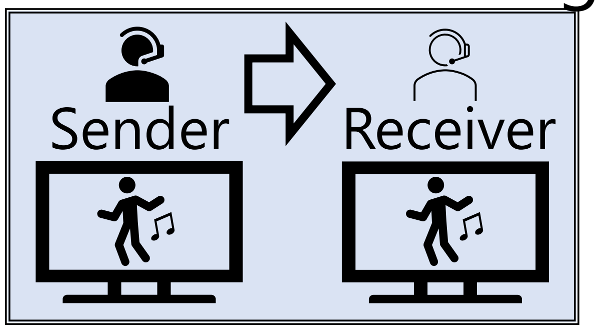
Microsoft



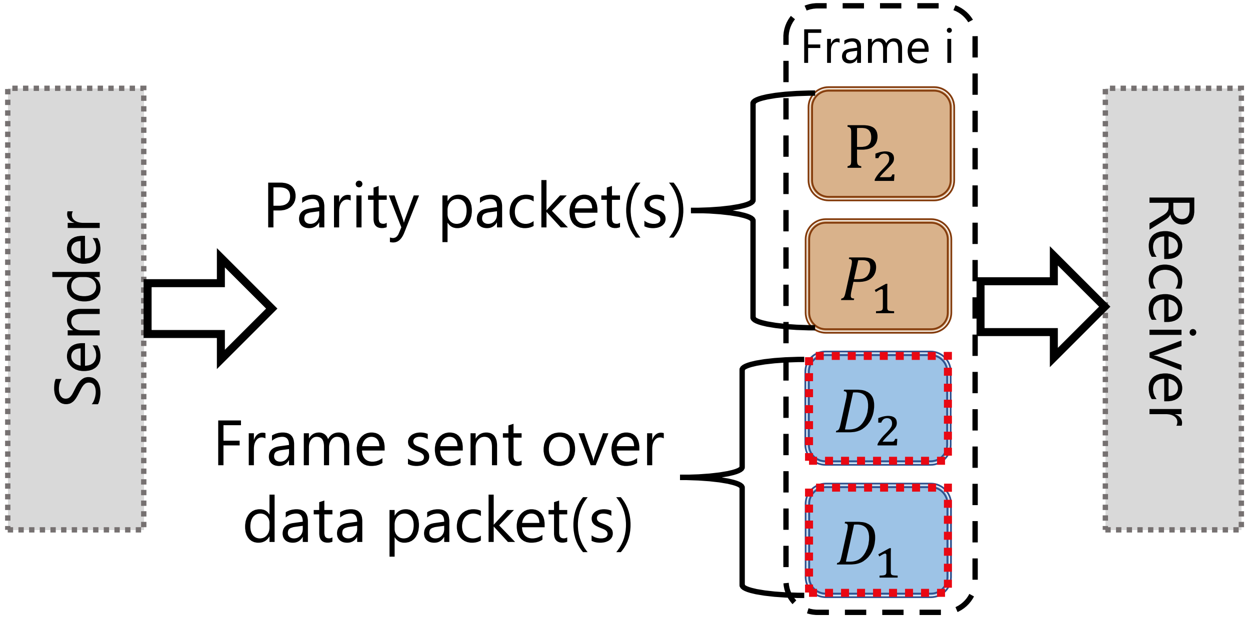



Motivation: packet loss reduces live-streaming QoE

- Streaming applications like videoconferencing (VC)



- Transmit sequence of video frames over a lossy network
  - Sending frame i



Low-latency packet loss recovery is needed!

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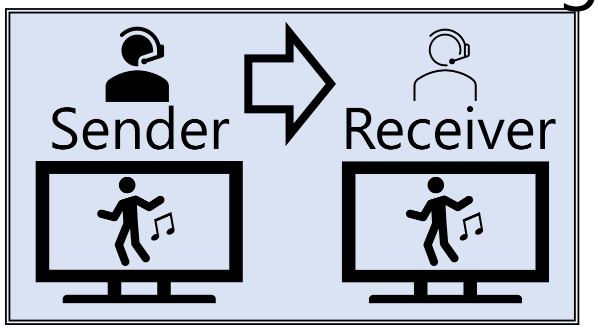

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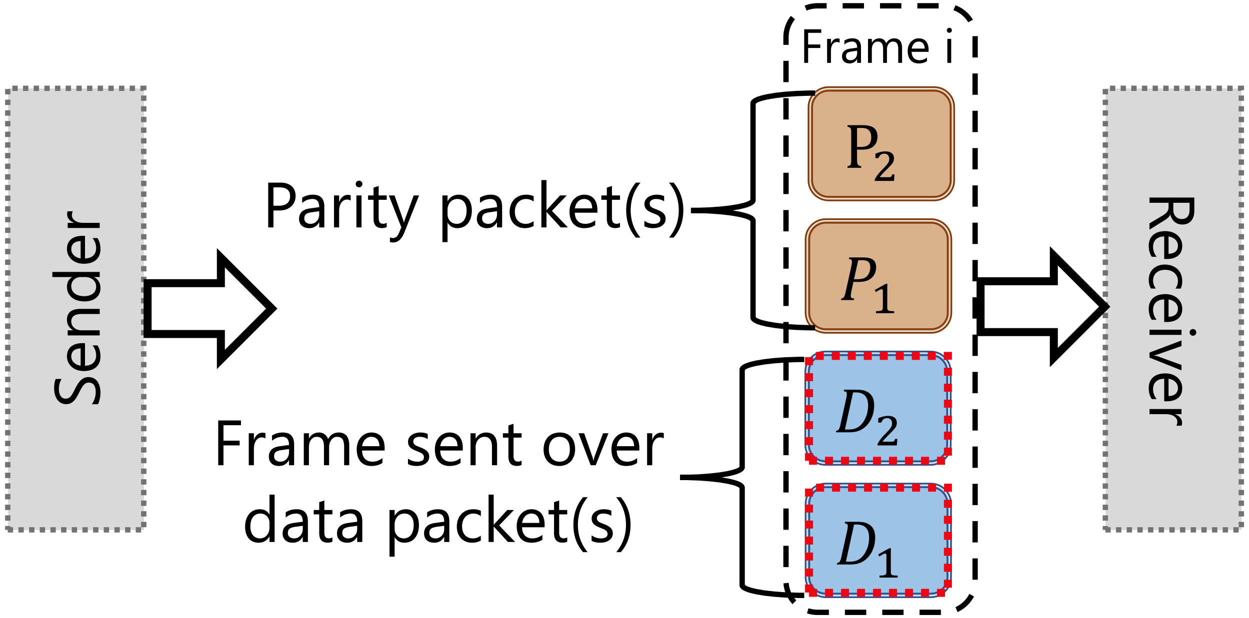

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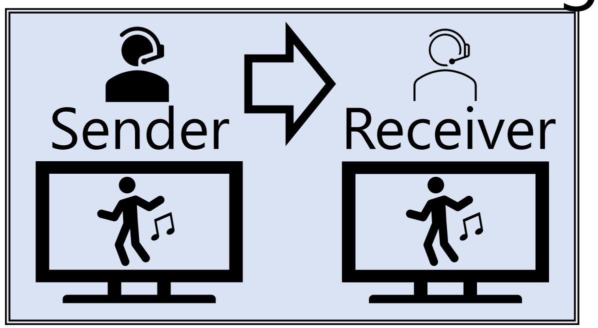

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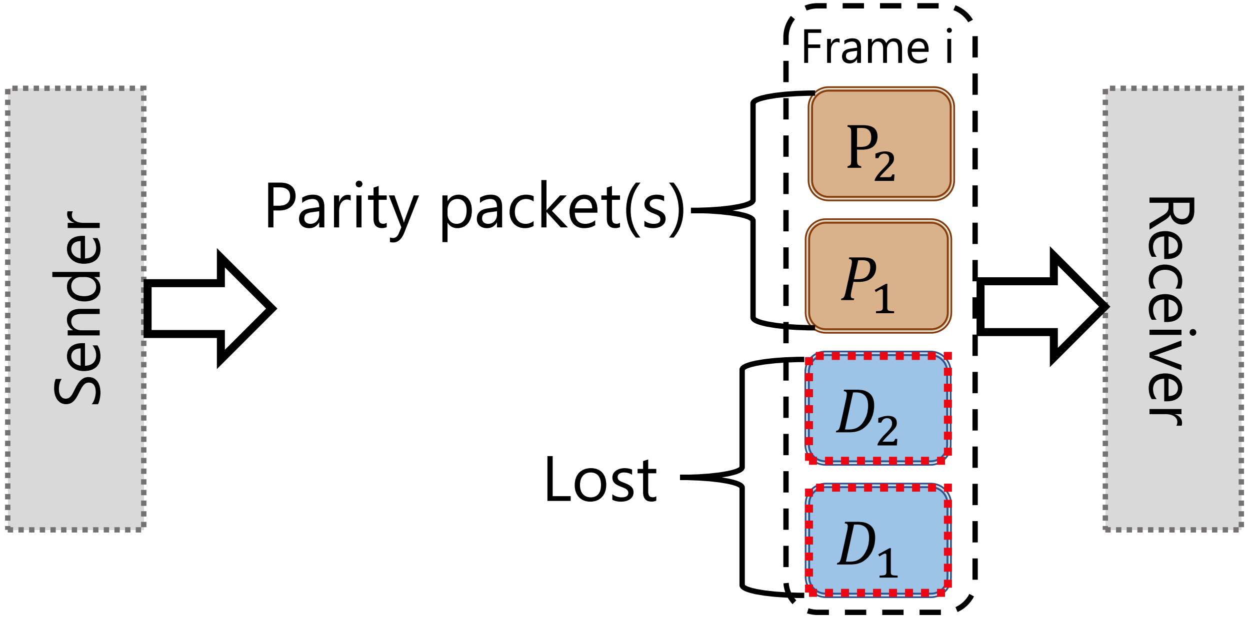

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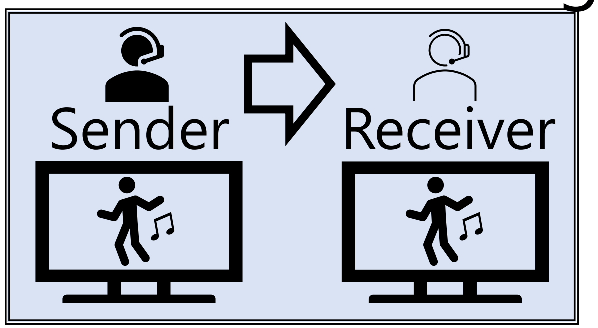

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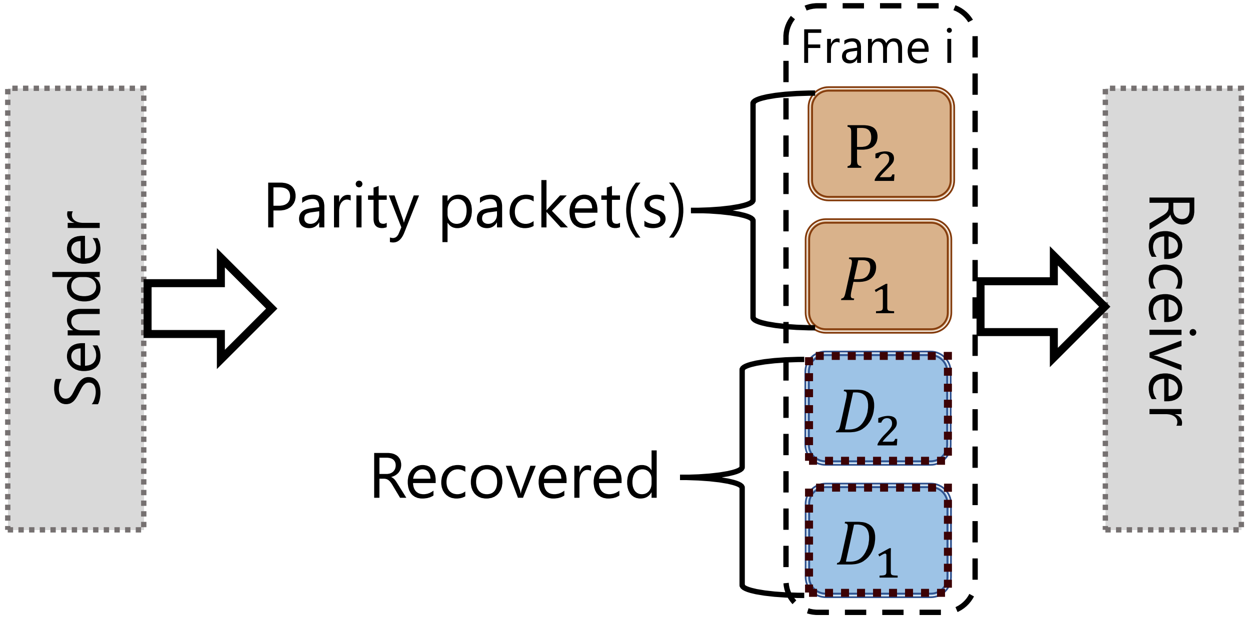


Low-latency packet loss recovery is needed!

Outline: improve VC QoE via streaming codes

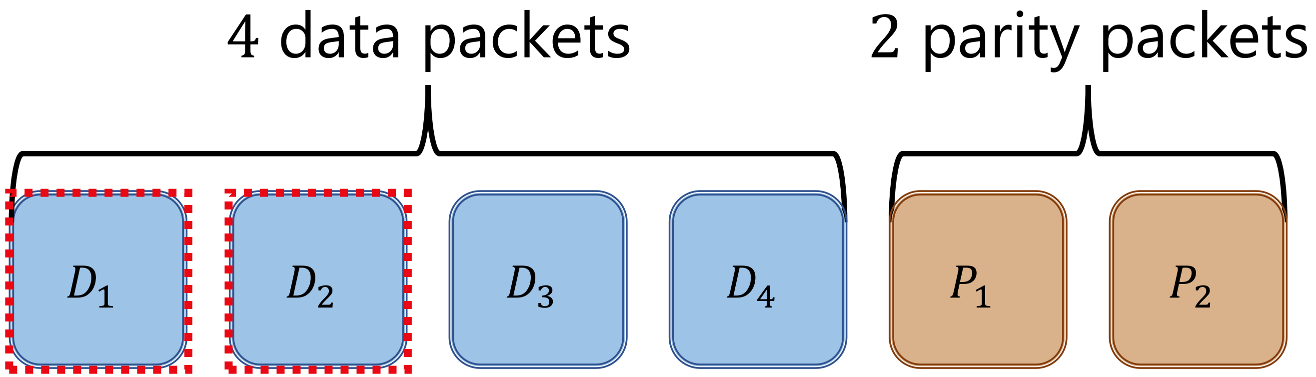
- **Problem**: conventional loss recovery sub-optimal QoE

- **Approach**: new streaming codes for low-latency loss recovery

- **Outcome:** improve key metrics of QoE like video freeze

Conventional loss-recovery is ill-suited to VC

- Retransmission has too high latency if high RTT (e.g., over long-distance)
- Replication requires a 100% BW overhead
- FEC in form of block codes widely used (e.g., by Teams)
  - Reed-Solomon (RS)



  - Traditional erasure codes use **sub-optimal BW** for VC, as we see next

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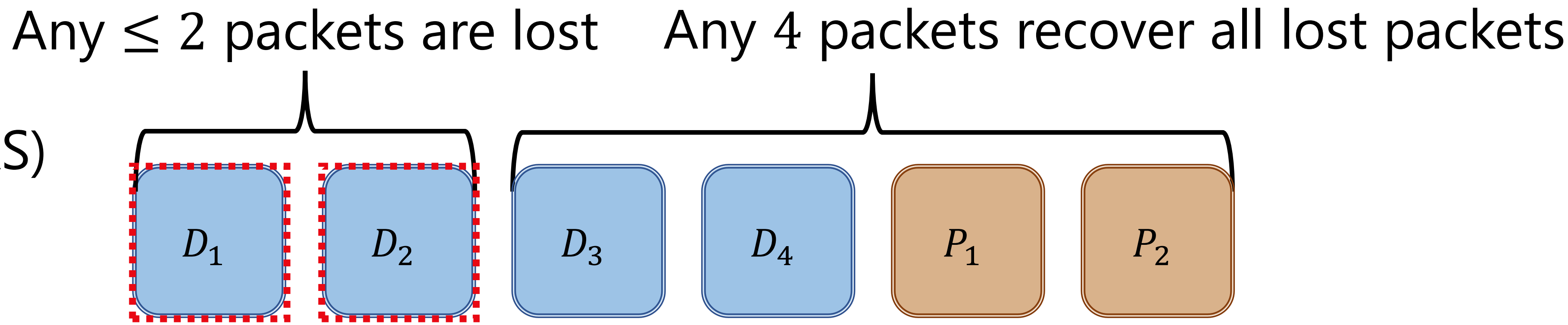

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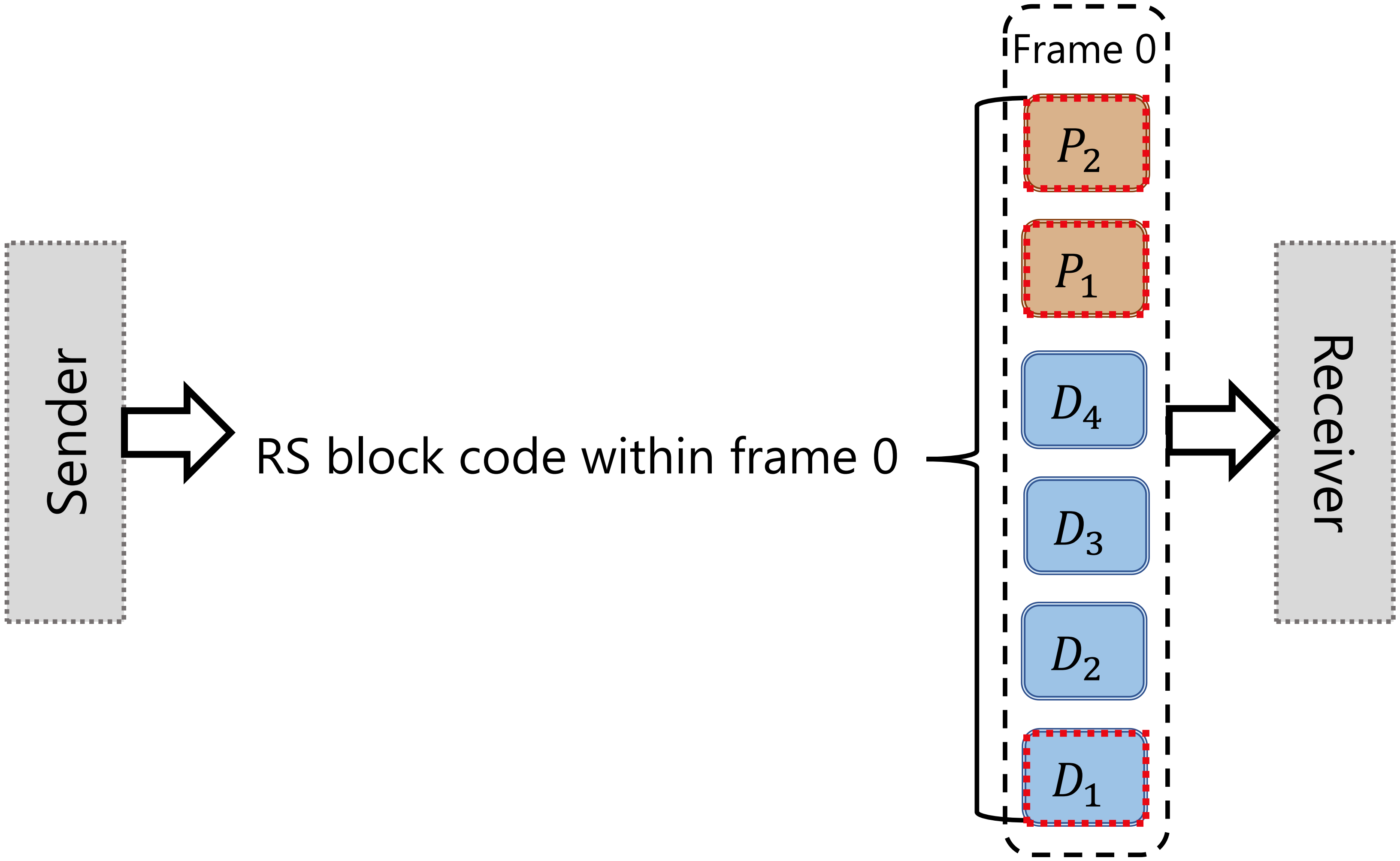
RS code within each frame wastes parity



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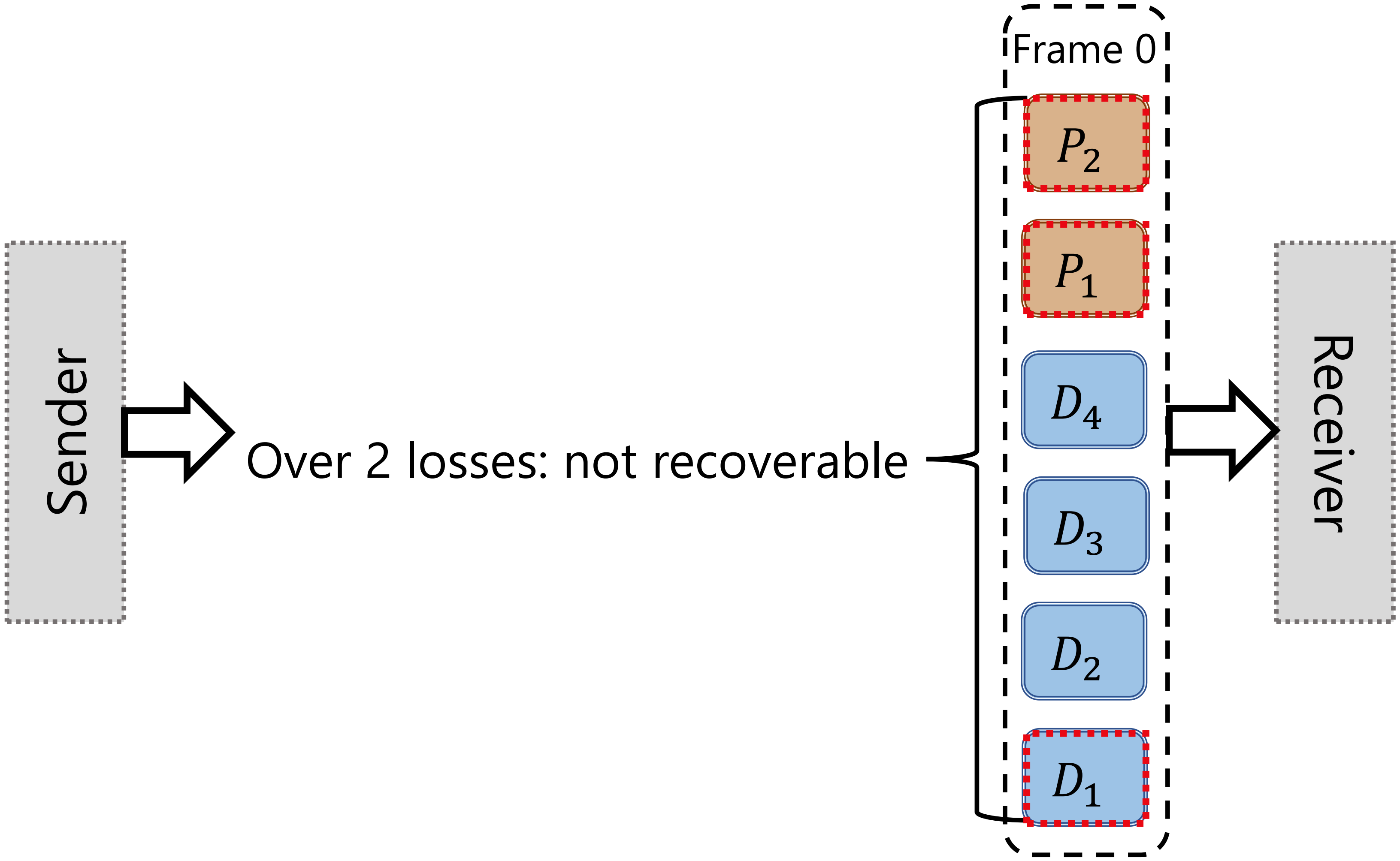
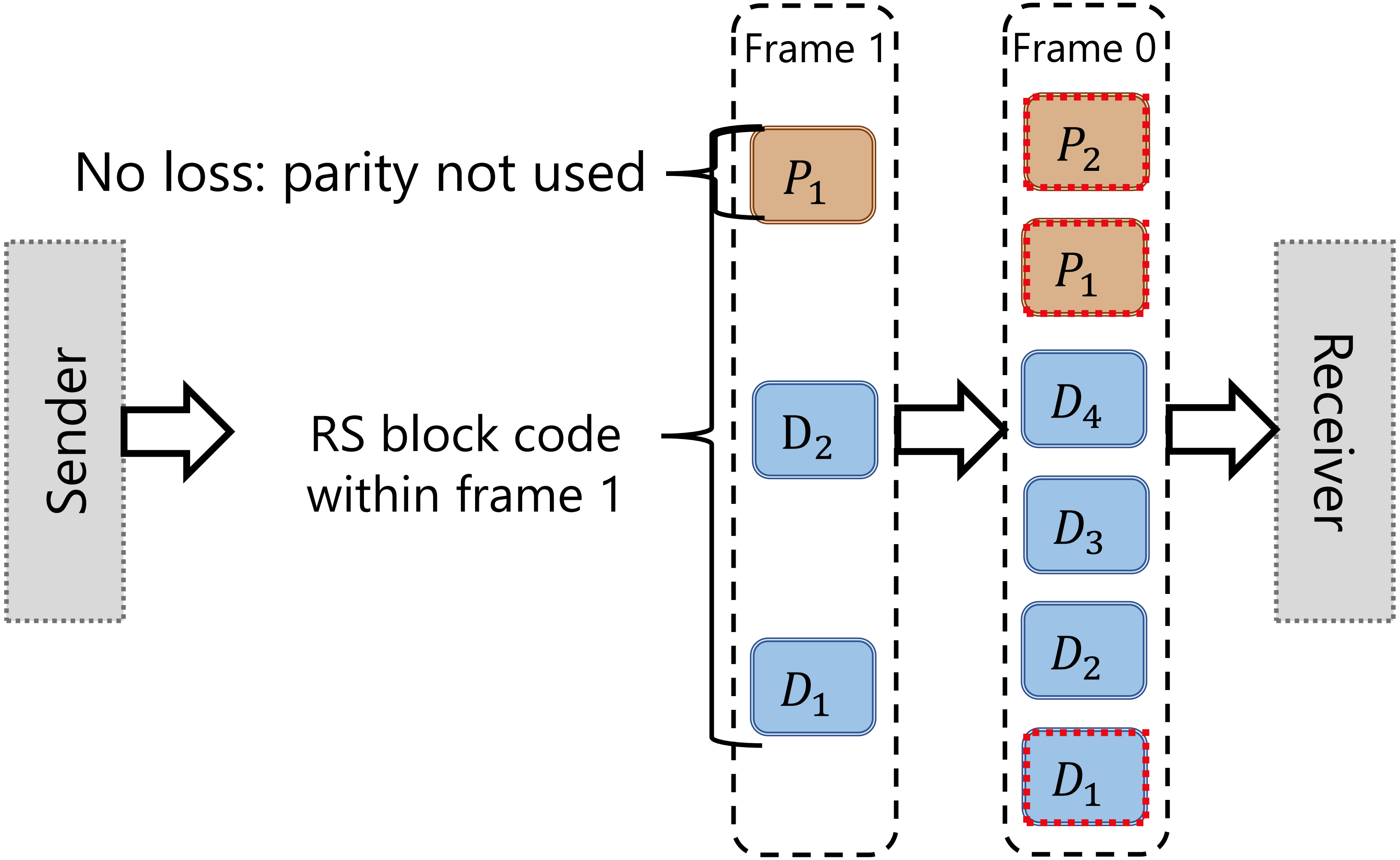


RS code within each frame wastes parity

Drawbacks:

Wasted parity for frame 1 not useful for frame 0

Freeze: frame 1 not playable without frame 0

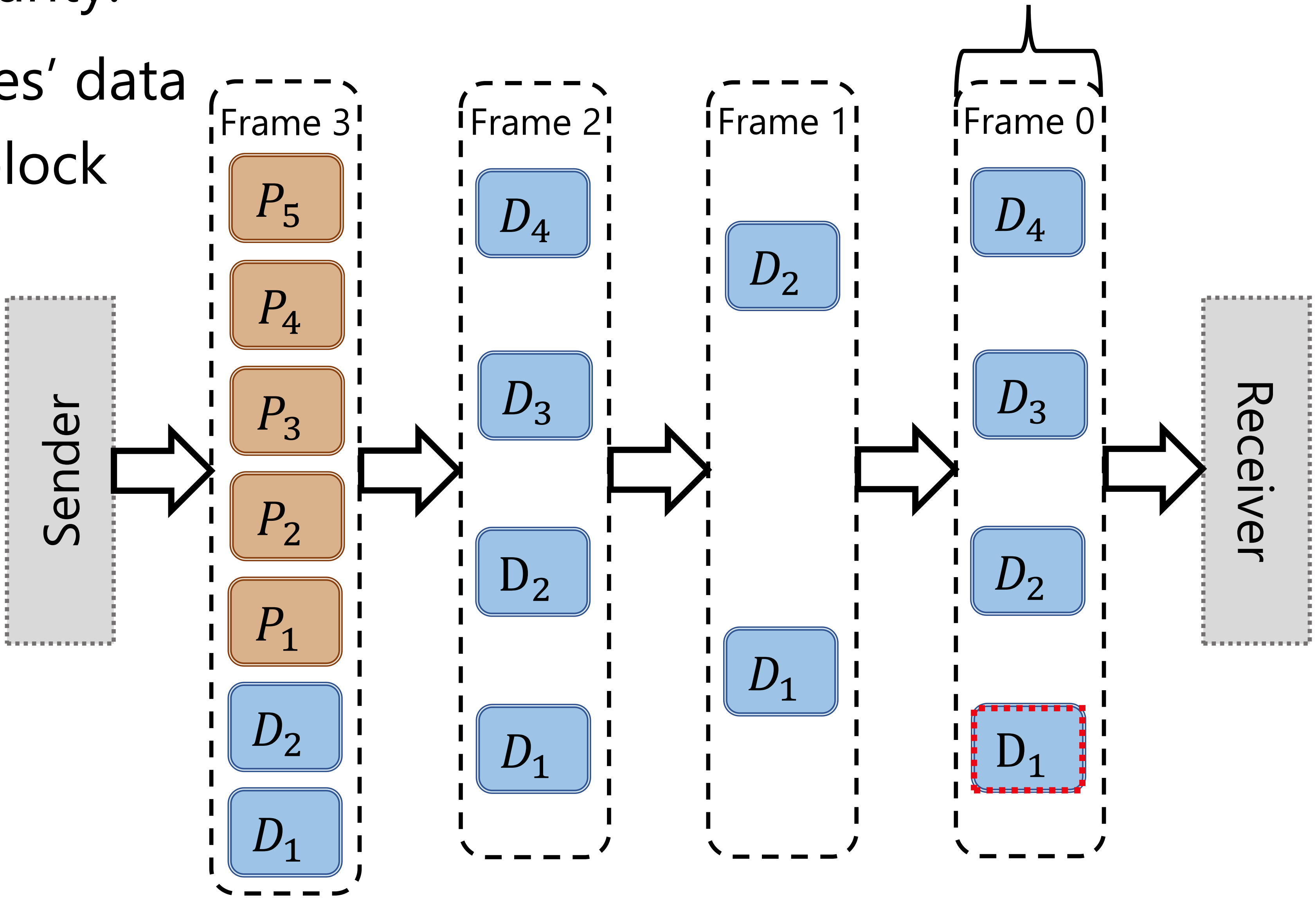


RS across frames costs latency and spikes BW

Quick fix for wasted parity:

Block code for 4 frames' data

Parity sent at end of block



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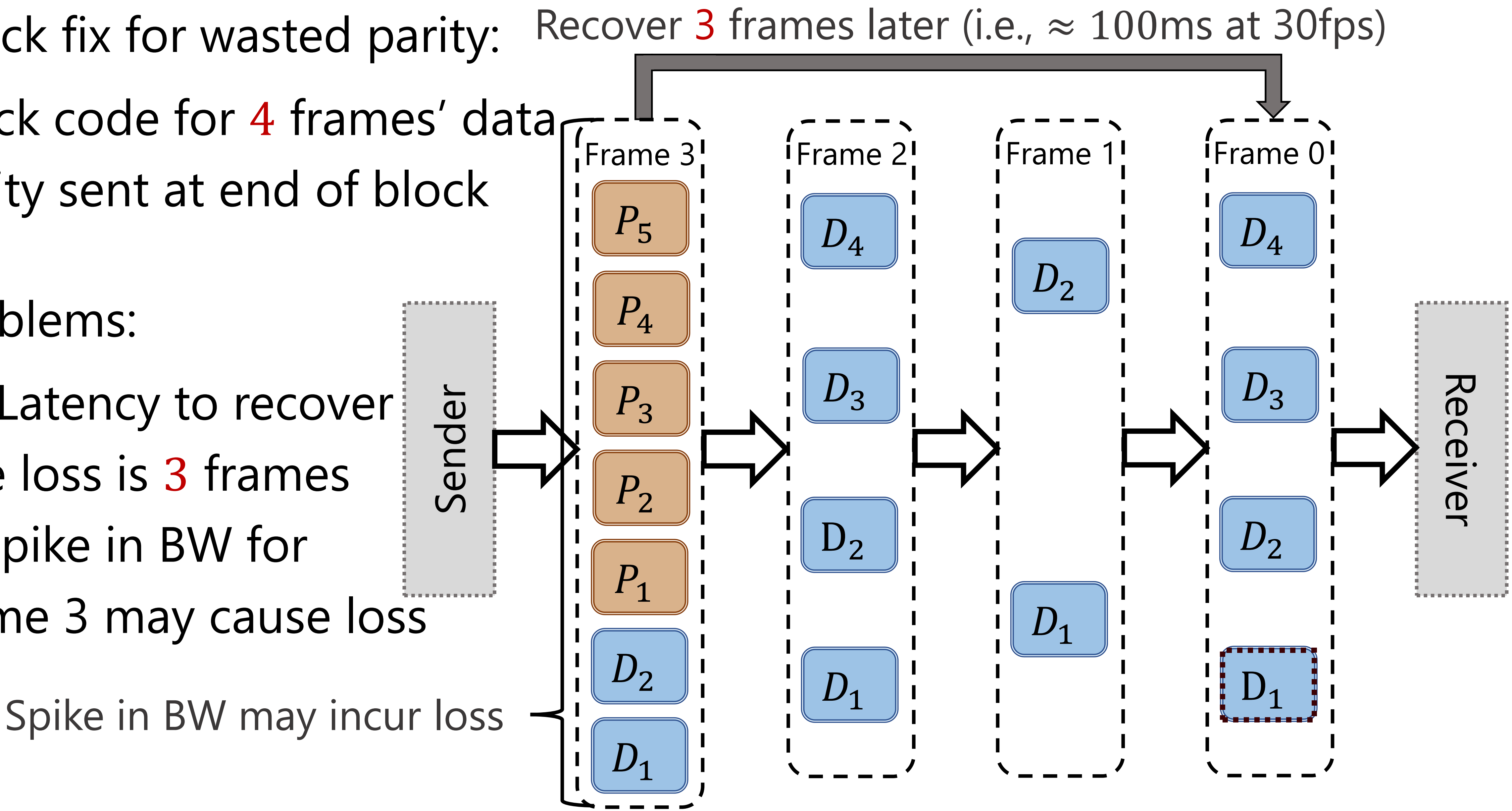
Problems:

1. Latency to recover one loss is 3 frames
2. Spike in BW for frame 3 may cause loss

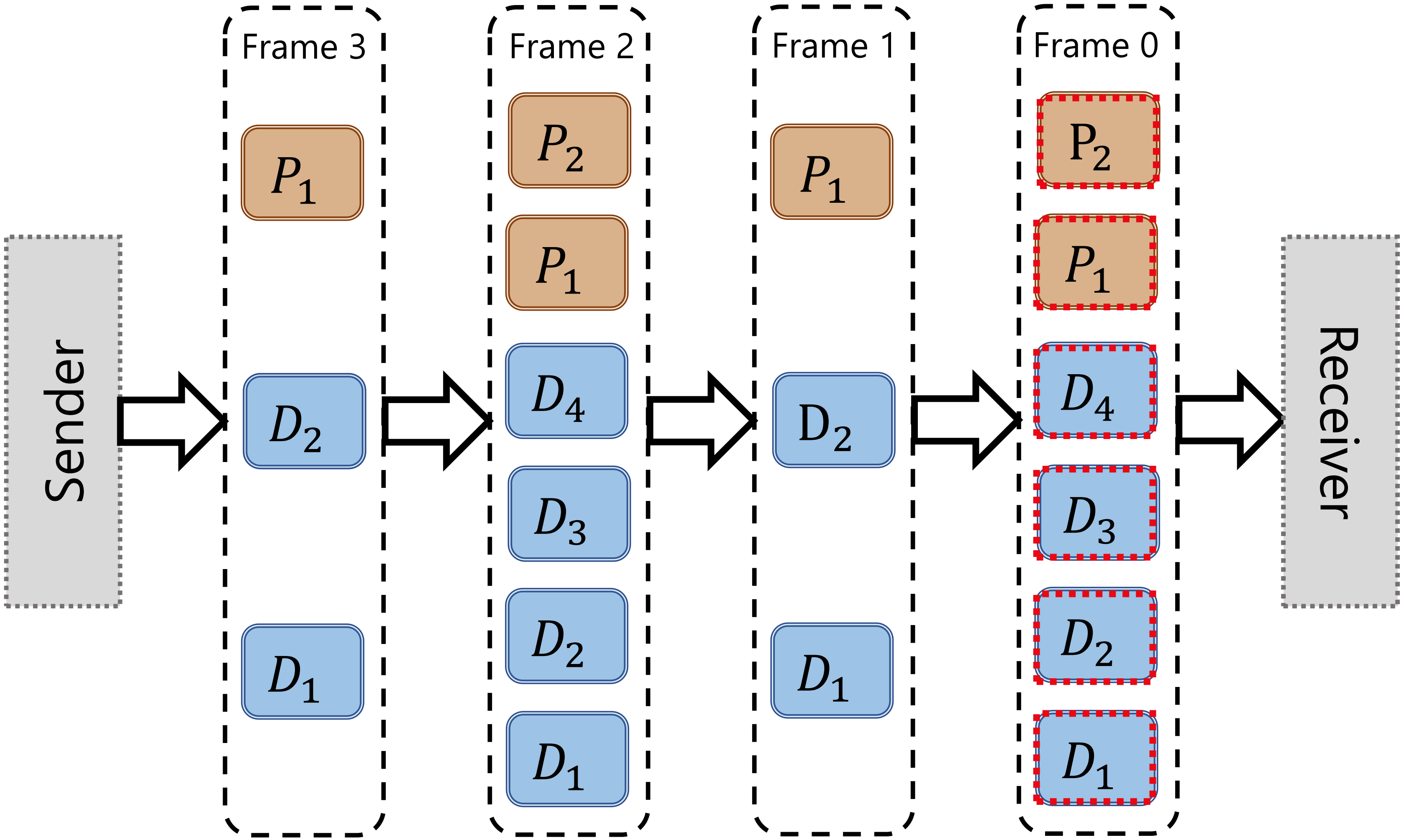


Streaming codes: bandwidth-efficient loss recovery

- Problem: RS codes sub-optimal for live communication: BW and latency
  - Block codes over 2 or 3 frames trades off these metrics
  - Our goal: fast recovery for one loss without wasting parity

- Streaming codes designed for following live-communication model
  - Latency: recover each frame within τ extra frames

Latency in # of frames to reflect end-to-end latency



Latency in # of frames to reflect end-to-end latency

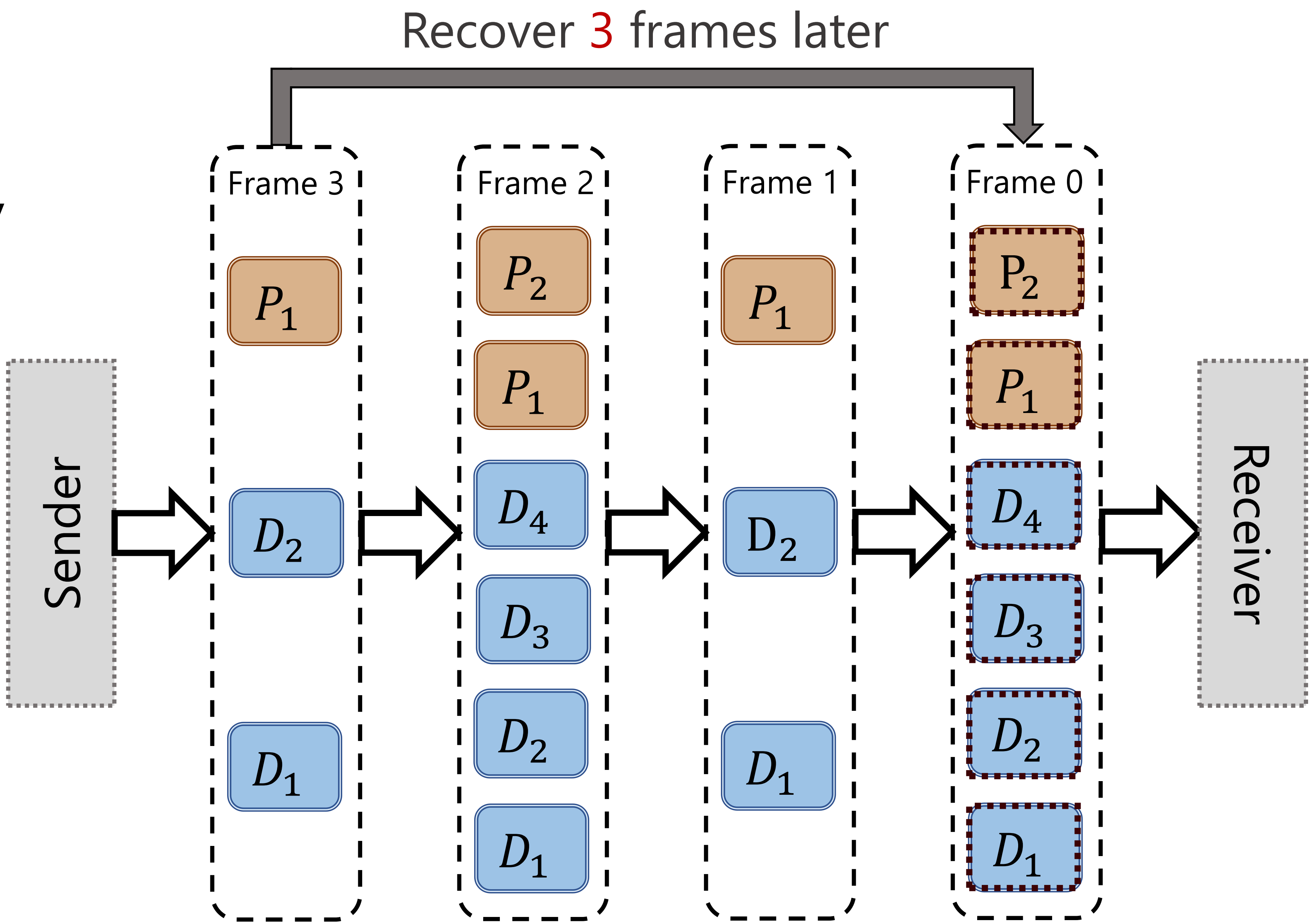
Suppose the call has

- 30 fps
- 50ms one-way delay

End-to-end latency:

$\approx 3 \cdot 33.3 + 50$

$= 150ms$



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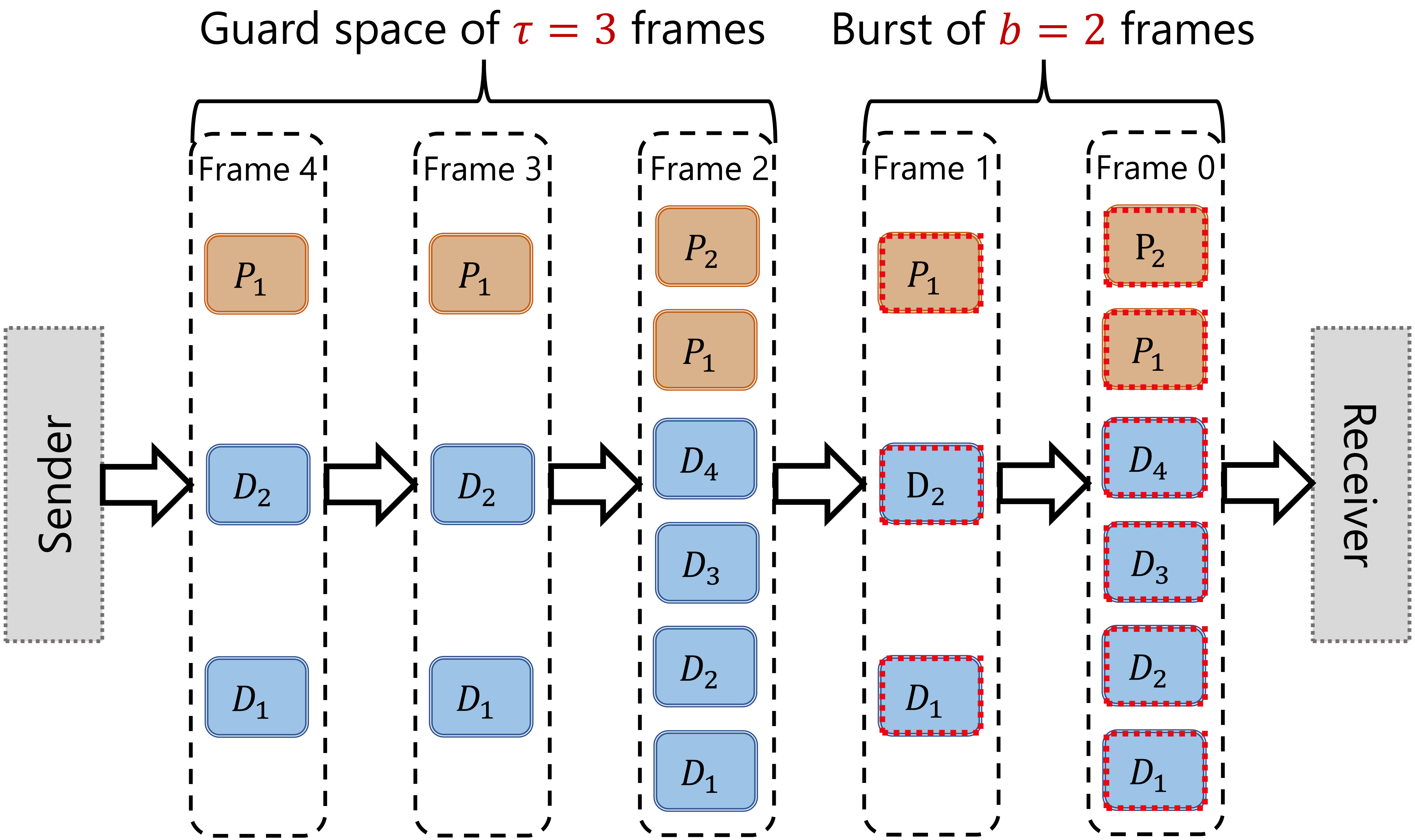
  - Burst: at most b consecutive lossy frames, then
  - Guard space: at least τ consecutive frames with no losses

Loss model of bursts followed by guard spaces



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- Streaming codes work by
  - Sending parity packets within each frame and computed over multiple frames to
  - **Sequentially recover** lost frames of burst each at their deadlines
  - As opposed to *simultaneously recovering* all lost packets (e.g., of a block)

Streaming codes: challenges

- Suitability over real-world losses unknown

- Gaps between theory and practice, including
  - Drop all packets of a frame
  - Never loss in guard space

- Not yet assessed for impact on the QoE

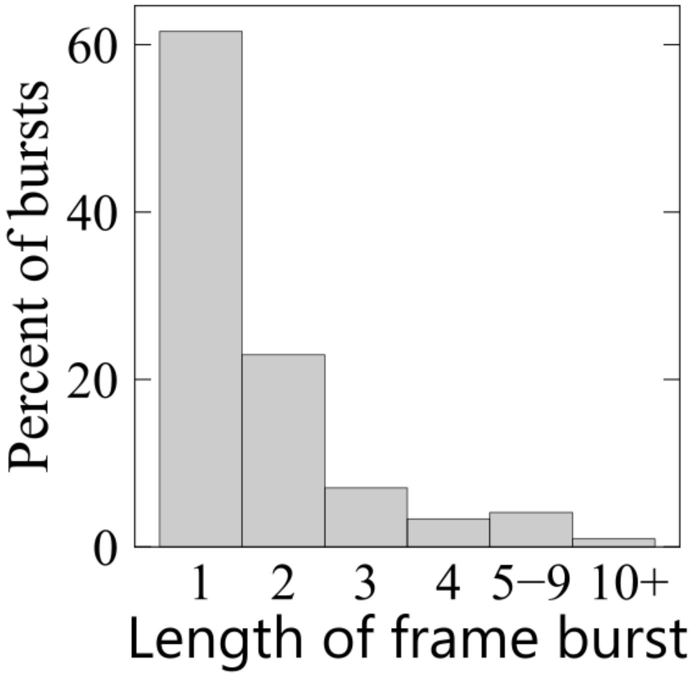
Analysis of traces from Teams video calls

- ≈9700 traces from two-week random sample Microsoft Teams 1:1 calls

- Burst losses are characterized by
  - Number of consecutive frames with at least one lost packet
  - Fraction of packets lost in a burst over multiple frames

- Guard spaces need only exceed τ to enable loss recovery
  - Set $\tau = 3$ to cap the latency at ≈ 150 ms at 30 fps with a 50 ms one-way delay

Losses suited to streaming codes… if address gaps



- Many burst losses of 2 – 4 frames determine parity needed
- No clear worst-case value, b

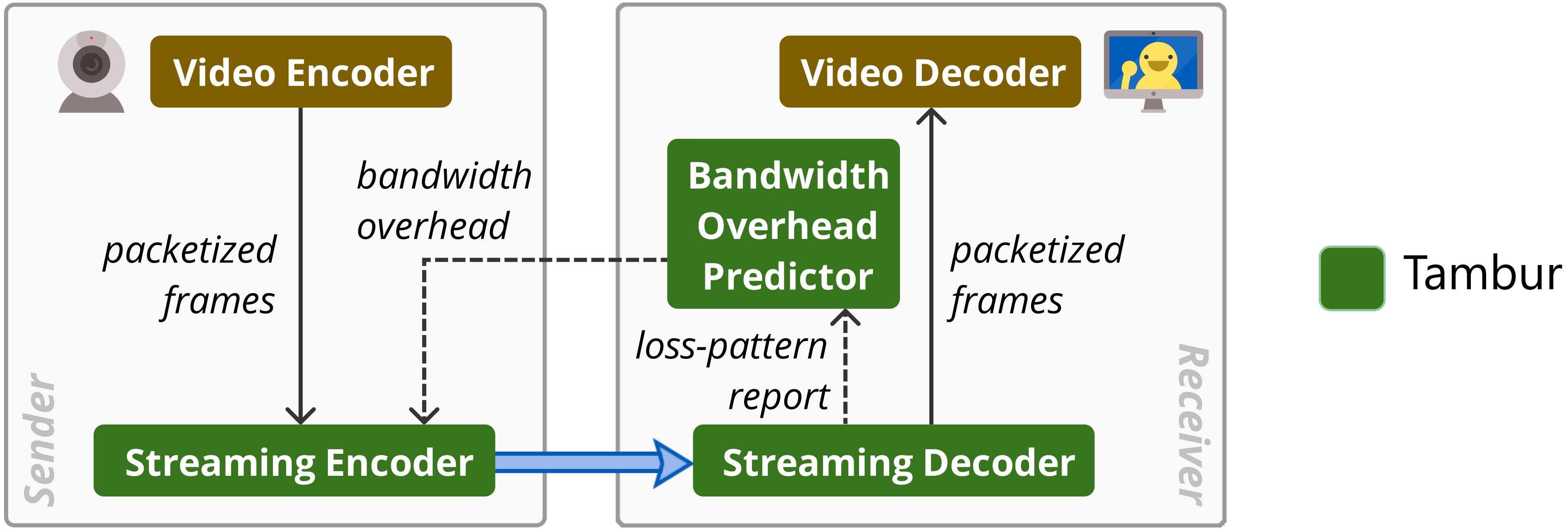
Fraction of packets lost in multi-frame burst

- Varies from just over 0 to 1
- Model of all packets lost is pessimistic

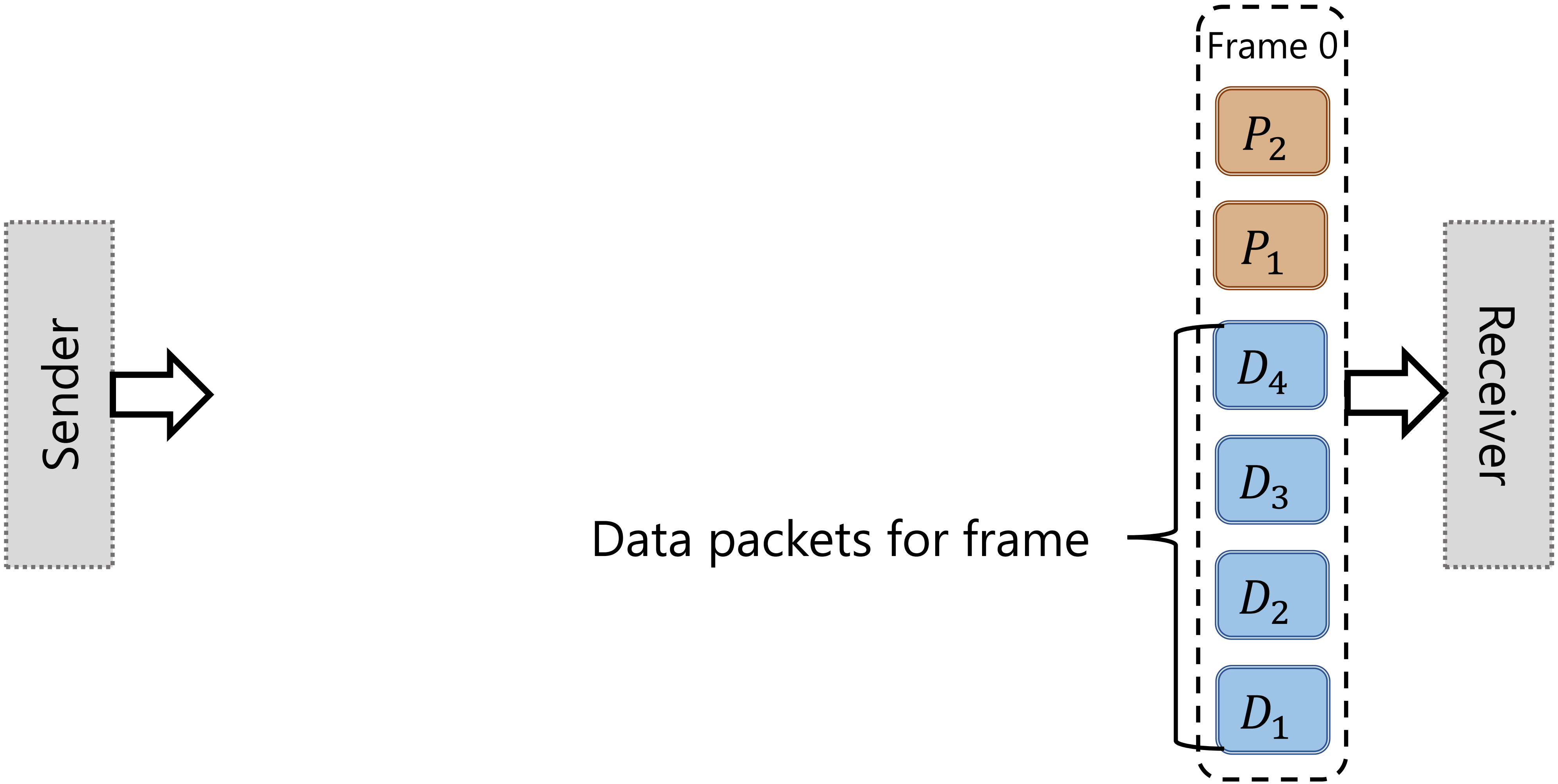
Guard spaces are common, but sometimes losses occur in guard space

Tambur: a new communication paradigm for VC

- Design Tambur by combining
  - New streaming codes (shown shortly)
  - Lightweight binary classifier instead of b and τ set parity size (see paper)
    - Match existing system's parity size or reduce it by 50%



Tambur recovers with bounded latency



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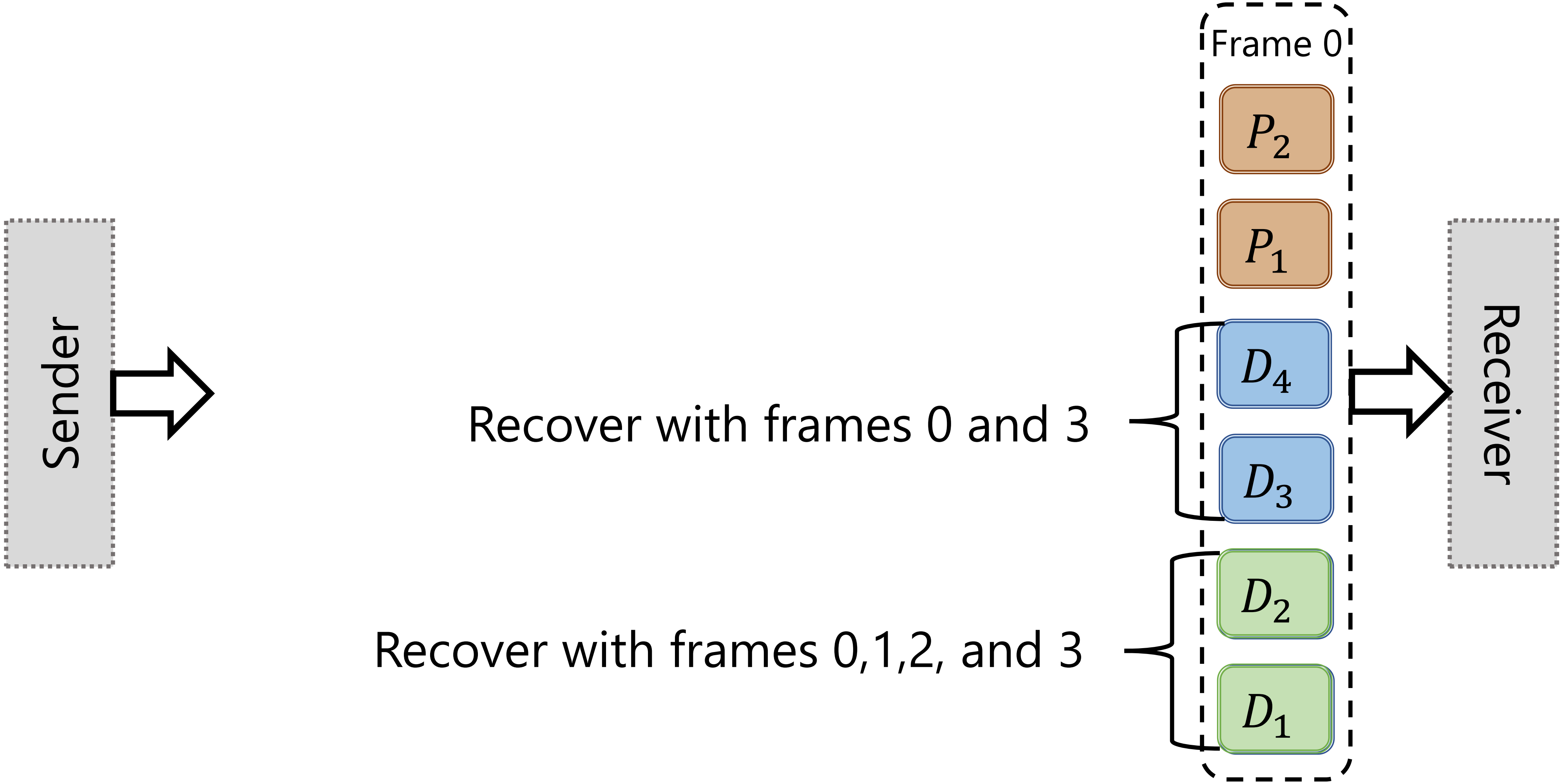

Recover with frames 0 and 3

Recover with frames 0,1,2, and 3

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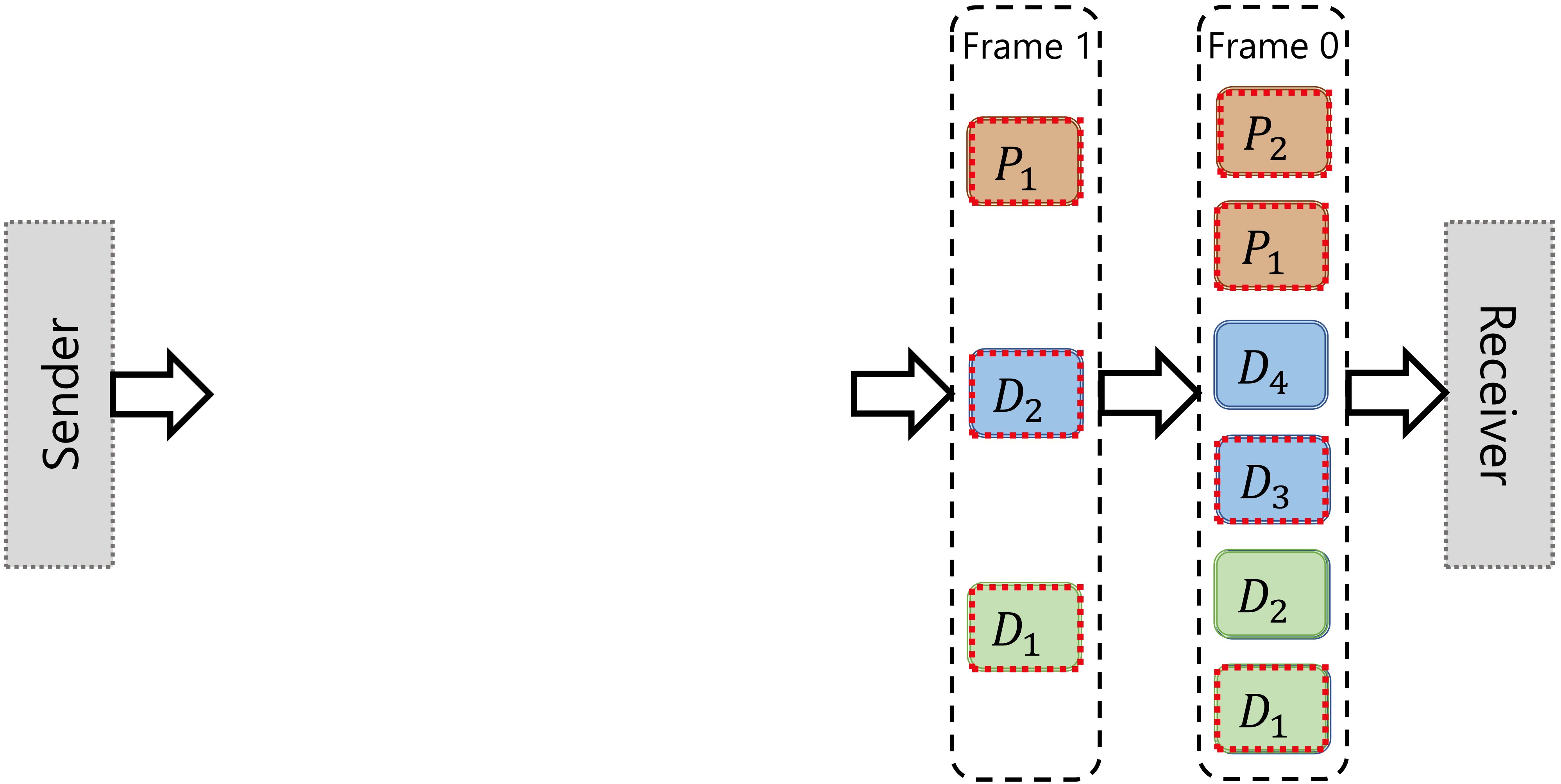

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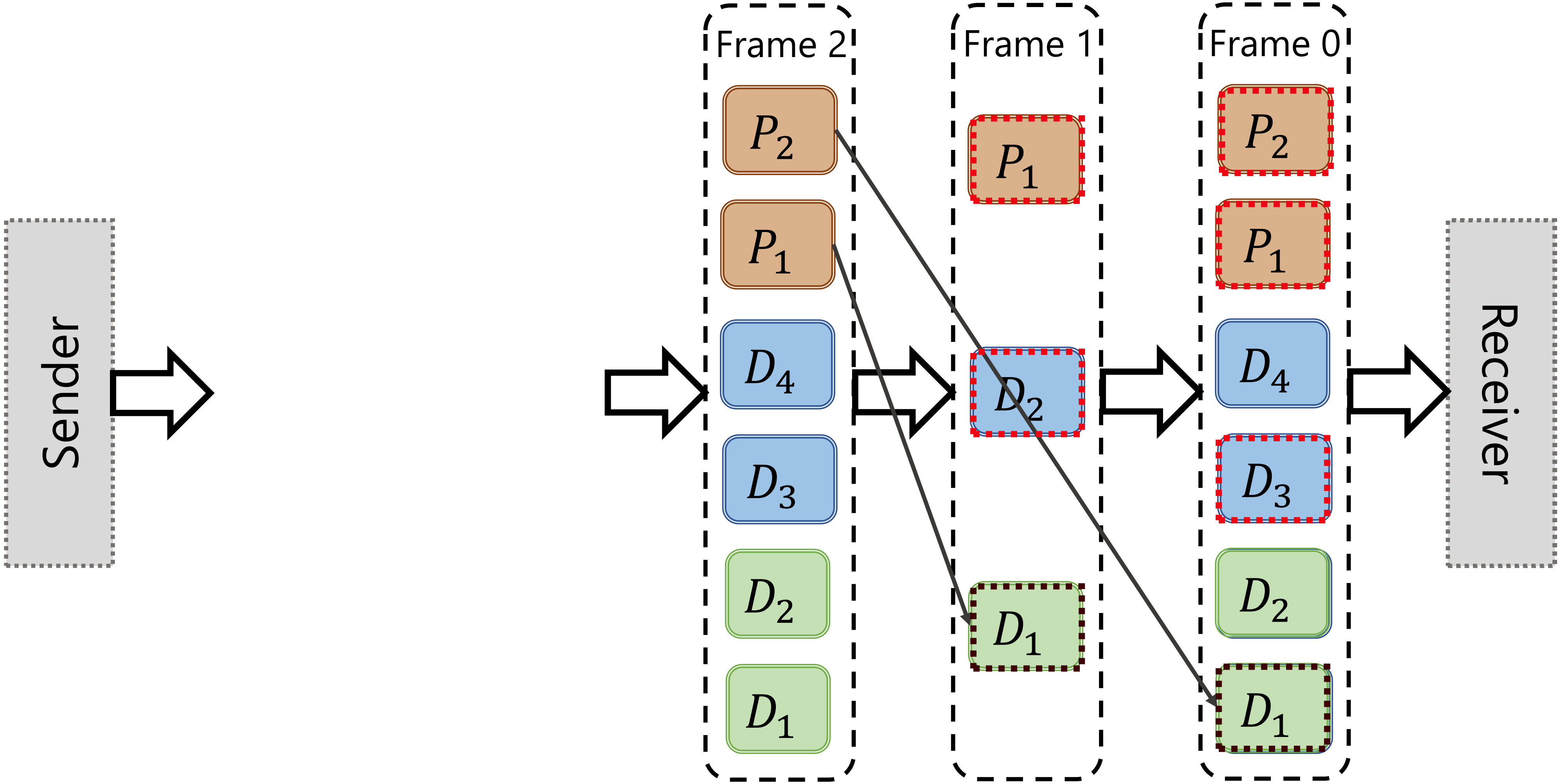

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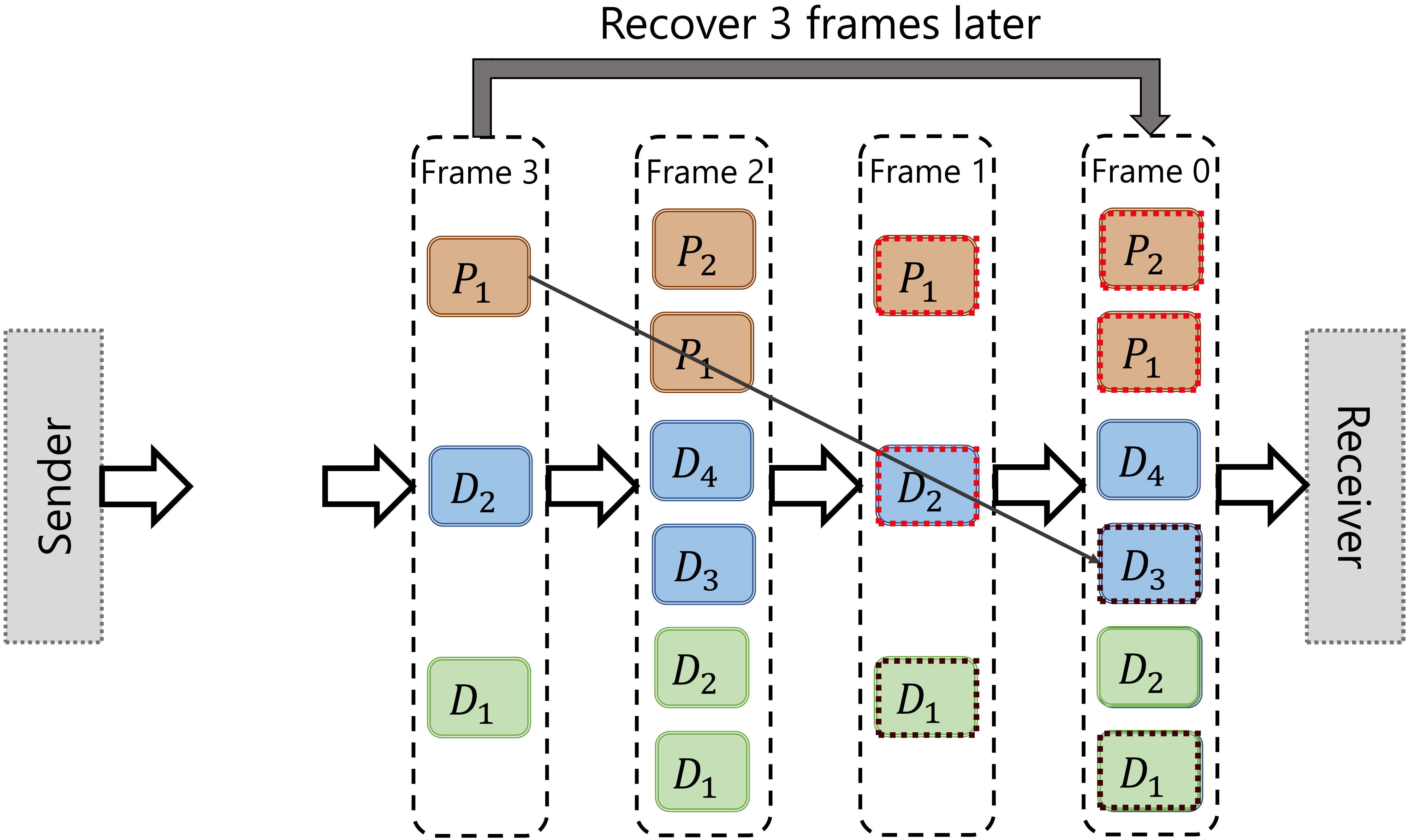

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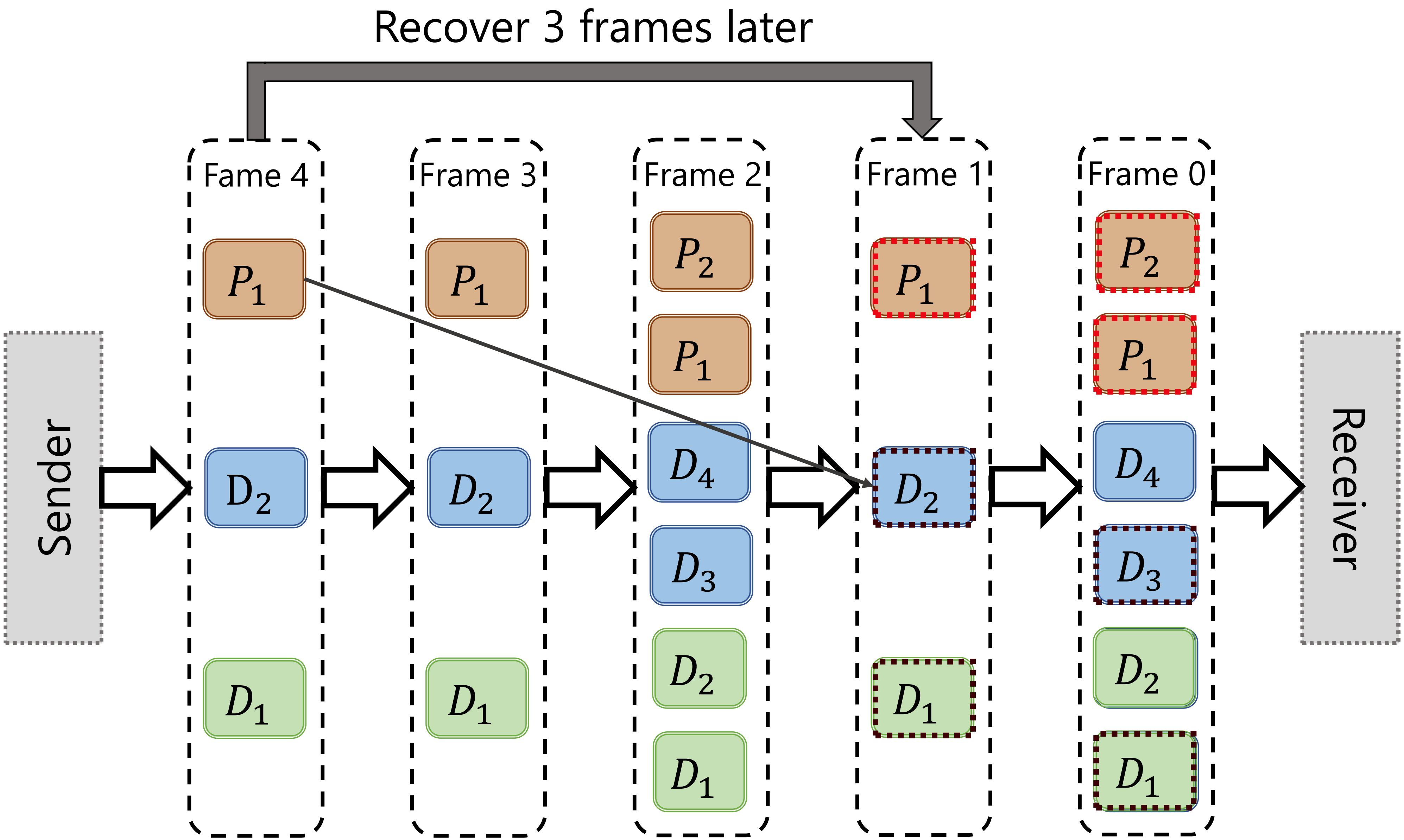
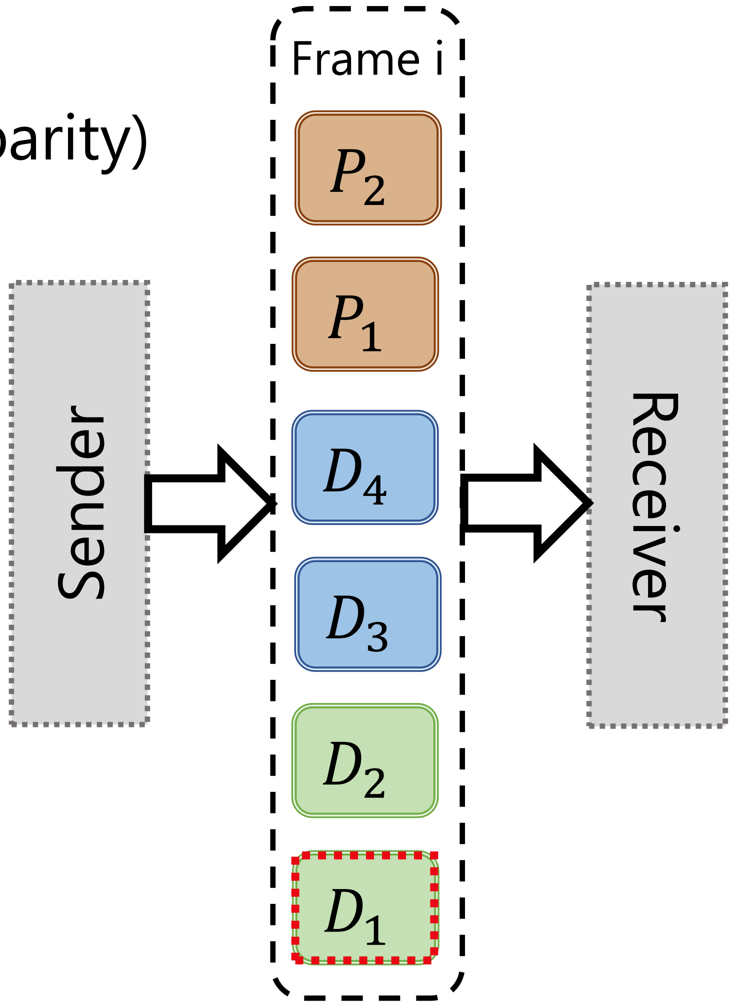


Tambur has minimal latency to recover rare losses

- Before: worst-case loss recovery
  - Leverage parity in guard space for recovery ✓
  - Unlike RS within each frame not recovering (waste parity)
- Now: address occasional losses
  - Loss recovery should have minimal latency
  - Unlike RS across 4 frames recovering 3 frames later



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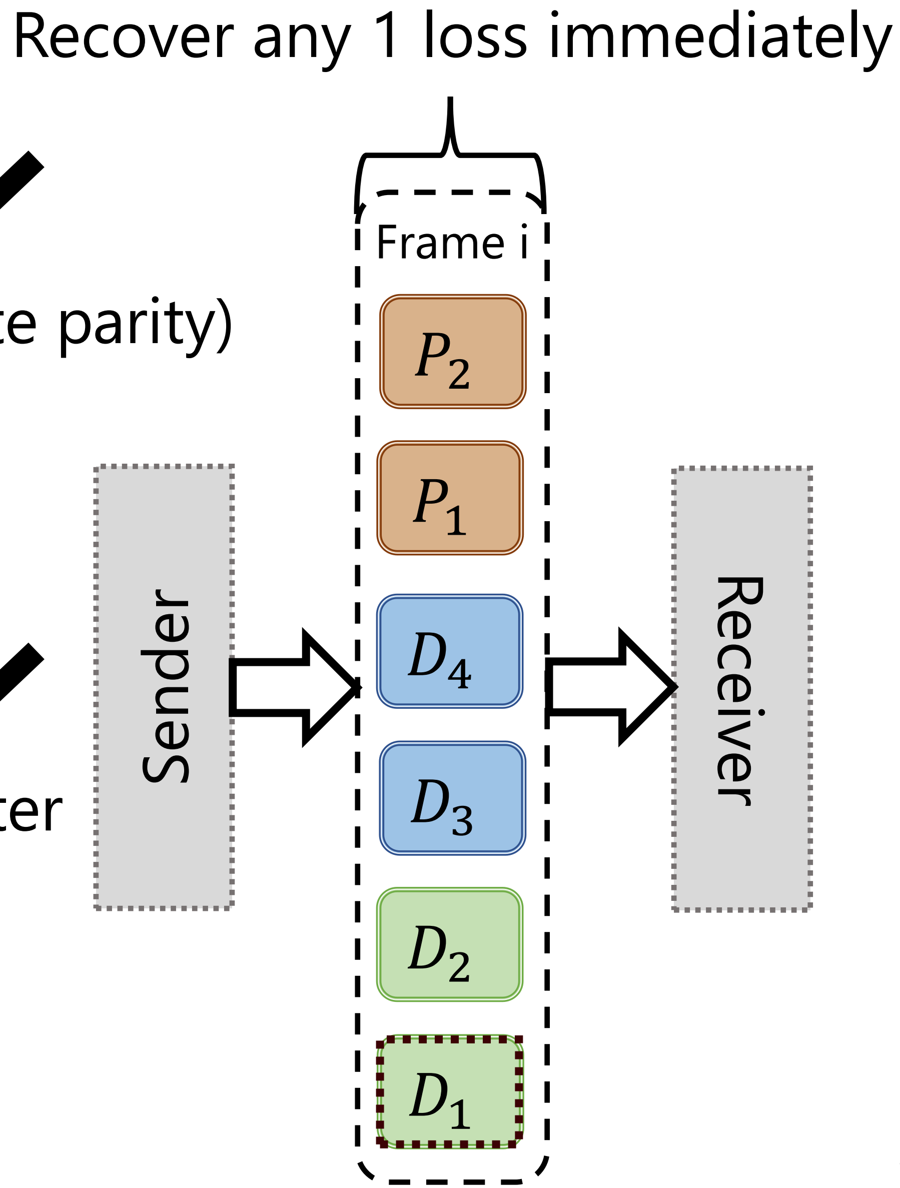
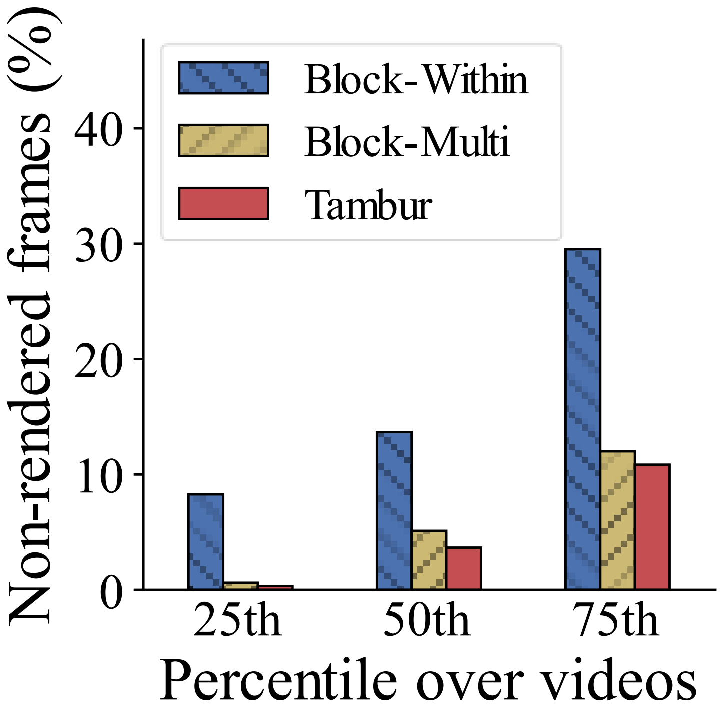


Online evaluation methodology

- Implement Tambur in C++ (https://github.com/Thesys-lab/tambur/)
- Integrate with Ringmaster (https://github.com/microsoft/ringmaster/)
  - Ringmaster is a VC platform for emulating 1:1 calls
- Compare to two standard baselines with **slightly extra parity**
  - Block-within—RS within each frame
  - Block-multi—RS across 4 frames
- Evaluate over 80 10-minute videos of varying bitrates
- Over Mahimahi and emulated networks (details in paper)

Tambur renders more frames at lower latency

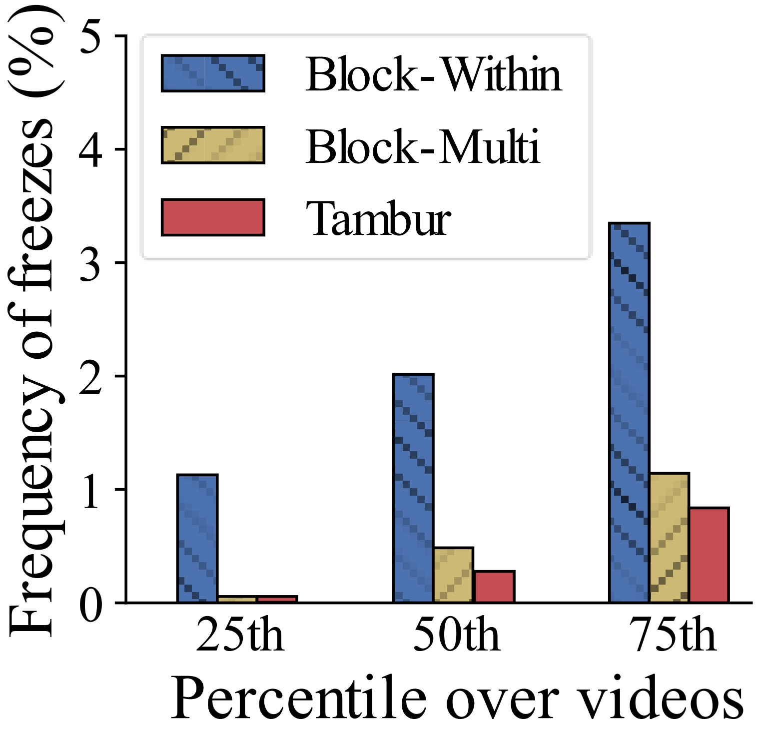
- Reasons for degrading QoE: not rendering frames or latency



- Fails to render **73% fewer frames** than Block-Within at median
- Fails to render **28% fewer frames** than Block-Multi at median
- 6.5 ms higher median latency than Block-within
- 18.9 ms lower median latency than Block-Multi

Tambur mitigates freeze frequency

- Freeze frequency crucial to mean opinion score (i.e., QoE)



- Freeze frequency **reduced by 78%** over Block-Within at median
- Freeze frequency **reduced by 26%** over Block-Multi at median

Takeaway: Tambur improves several key metrics of the QoE

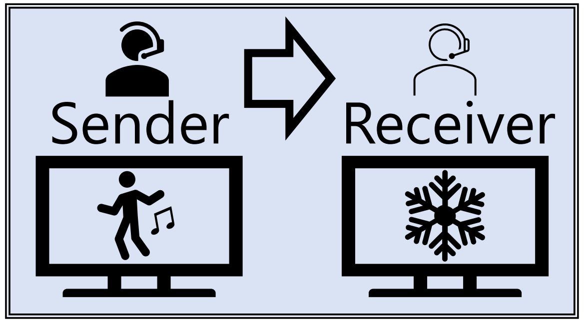
New interdisciplinary loss recovery VC

- **Challenge**: conventional loss-recovery sub-optimal videoconferencing

- **Approach:** build Tambur by designing new streaming codes + using ML

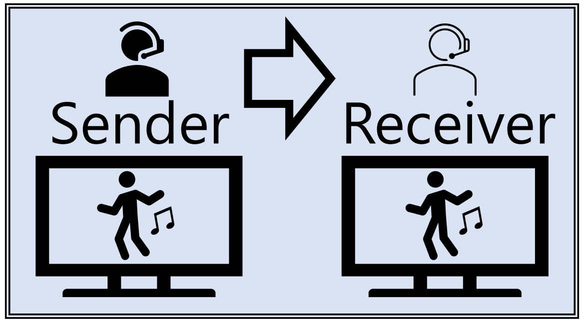
- **Outcome:**

Before



Eliminate 26% of freezes and 28% of rendering failures

After



This work was funded in part by an NSF grant (CCF1910813).