# Multipath variants for congestion control in context of WebRTC

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# Motivation

- Increased demand for real-time communication [1]
- WebRTC dominant technology for video streaming [2]
- Single path limitations

Discord

Google Meet

Microsoft Teams

WhatsApp

zoom

Jitsi Meet

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# Research goals

- Survey existing congestion control algorithms [3]
- Explore benefits of multipath transmission in WebRTC
- Evaluate the performance across various metrics

# WebRTC & Congestion control

- Peer-to-peer communication over UDP
- No built-in congestion control in UDP
- Existing algorithms: GCC, SCReAM, NADA [5][6][7]
- Limitations: single-path assumptions

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# Why Multipath?

- Send across multiple network routes at the same time [4]
- Advantages:
  - Increased resilience
  - Redundancy and fault tolerance
  - Efficient bandwidth use
  - Seamless mobility for mobile devices

# Multipath in practice

## MPTCP: Multipath TCP [8]

- Built into the protocol
- Multipath subflows

## UDP lacks native congestion control support

Congestion control is not built into the protocol itself

## Challenges

- Packet reordering
- Integration with existing architectures
- Path selection strategies

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# Methodology

- Create a realistic multipath simulation scenario
- Implement a custom multipath capable NADA that can switch between multiple path selection strategies
- Compare the behaviour of a classic NADA [7] and its multipath counterpart

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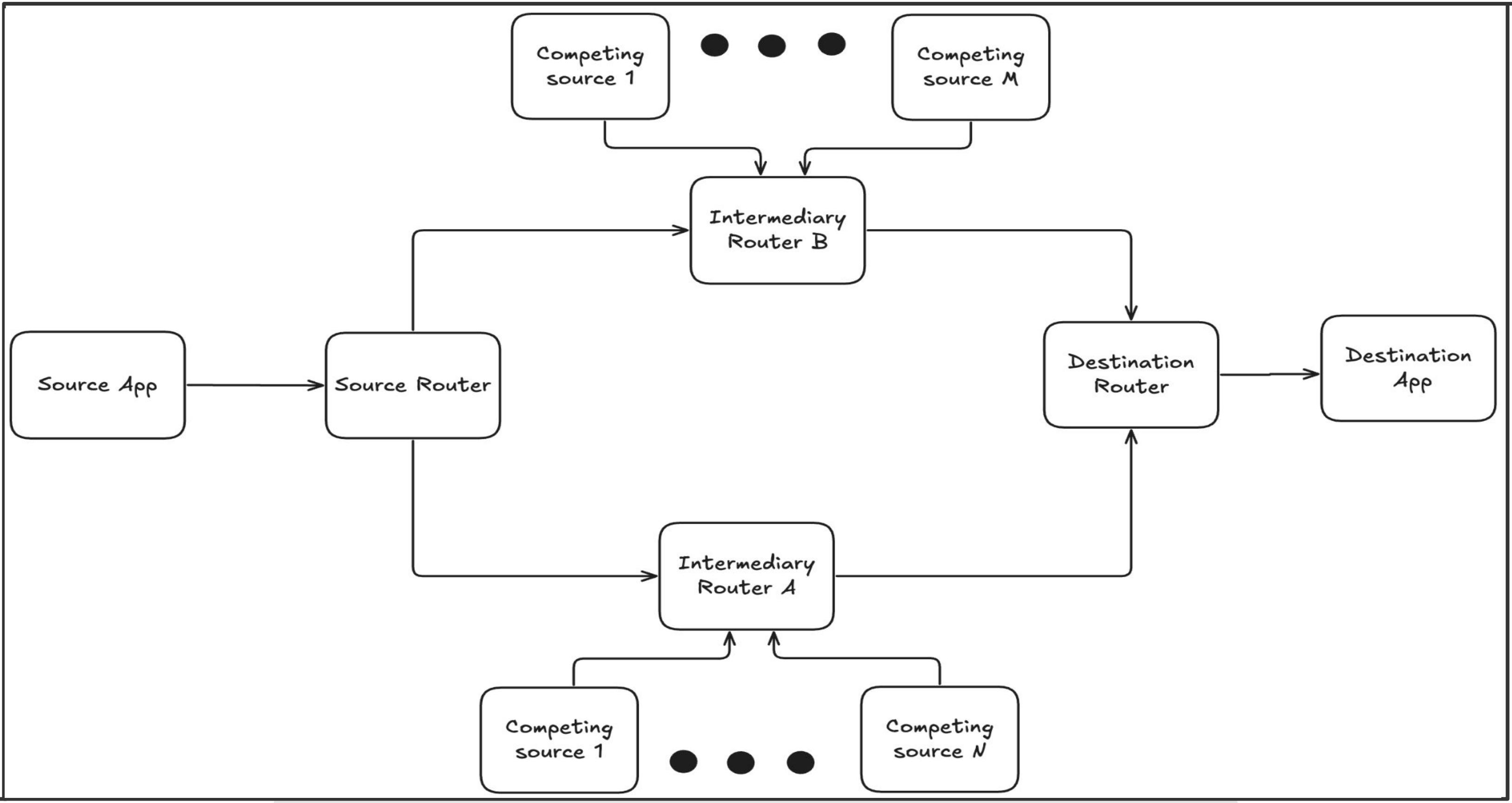
# System architecture

- Simulations realised using Network Simulator 3 (ns-3) [9]
- Extended existing UDP client classes
- Simulate realistic video streams by implementing custom video packets
- Simulate a receiver application to consume the video stream

# Simulation topology

- Two paths: different bandwidth/latency
- Competing TCP flows to simulate real traffic and achieve statistical multiplexing over the network
- Wrapper classes over Source/Destination connections
- Custom video buffer and receiver app to simulate a video player


Competing source 1
Competing source M
Intermediary Router B
Source App
Source Router
Destination Router
Destination App
Intermediary Router A
Competing source 1
Competing source N

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# Path selection strategies

- Dynamic Weights
- Best Path
- Round Robin
- Frame-Aware
- Buffer-Aware

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# Congestion signals

**Agg signal**

| | |
|---|---|
| **Packet loss** | Packet loss is the primary congestion indicator |
| **One-way delay** | The one-way delay between sender indicates network queuing and potential congestion buildup |
| **Receiving / Sending rate** | NADA compares the receiving rate with the send rate to detect throughput reductions |
| **Explicit congestion notifications** | Packets are marked as congested by routers |

# Evaluated metrics



| | |
|---|---|
| **Throughput** | Measures the effective data rate at the receiver, calculated as total received bytes divided by simulation time (in Mbps) |
| **Delay** | The average time taken by packets to travel from sender to receiver, including all transmission, propagation, and queuing delays |
| **Packet loss** | The percentage of packets that were sent but never received at the destination |
| **Jitter** | Measures the variability in packet arrival times, indicating how uneven the delivery intervals are |
| **Buffer length** | Indicates how much video content (in seconds) is currently preloaded in the receiver's playback buffer |
| **Buffer underruns** | Counts the number of times the playback buffer is empty and causes a video freeze or rebuffering |
| **MOS (Mean Opinion Score)** | An estimated user experience score (0 to 5) derived from received video bitrate using a logistic mapping |
| **Stability** | Reflects the consistency of throughput over time; large deviations indicate unstable performance |
| **Efficiency** | Represents how many of the transmitted bytes actually reached the destination, indicating delivery effectiveness |

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# Results

**60.8% - 219.4%**

Throughput increase

**88.9%–99.9% reduction**

Packet loss

**Up to 86.7% lower**

Jitter on stable networks

**84.3 - 218.5%**

Variability in throughput

**0.01% – 0.88%**

Improvement in MOS score

**84.8 - 414.6%**

In buffer length (ms)

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# Trade-offs

- Increased throughput variability
- Adaptive strategies have increased jitter
- Path selection adds complex overhead

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# Future work

- Adaptive filtering and congestion prediction
- Real deploys on real networks and evaluating real user experience
- Testing on 5G/6G, IoT, edge computing scenarios
- Machine learning for strategy selection

# Conclusions

- The Multipath NADA approach improves real-time performance
- Modularity of the proposed architecture can easily be extended for future research
- Best for hybrid topologies
- Buffer-aware / Frame-aware are the most stable

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# Q&A