Latency-Aware Media Delivery through Software-Defined Networks

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Overview

- Low Latency Video Streaming
- Software-Defined Networking
  - 5G Broadcast through SDNs
  - Predictably-reliable Real-Time Transport[1]
  - Transparent Transmission Segmentation[2]
- Buffer Dynamics Stabilizer[3]

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3. Shuai, Yongtao; Herfet, Thorsten: "Improving User Experience in Low-Latency Adaptive Streaming by Stabilizing Buffer Dynamics", IEEE Consumer Communications and Networking Conference (CCNC), Las Vegas, January 2016

Video is going IP

- All domains are going IP
  - TelCos, BCAST con- & distribution (DOCSIS), production (Industrial Ethernet, TSN), even the holy grail automotive (OPEN)
  - Antipodes are synchronous, prioritized streaming (TSN) and dynamic adaptive streaming (DASH)

- Significant progress in theory and domains
  - Coding with Finite Block Length (Polyanskiy et.al. 2010)
    - Channel capacity with limited block length (no given time)
  - ITU-T Watch Report (Tactile Internet, 2014)
    - Extremely low latencies of 1 ms (no given reliability)

But…

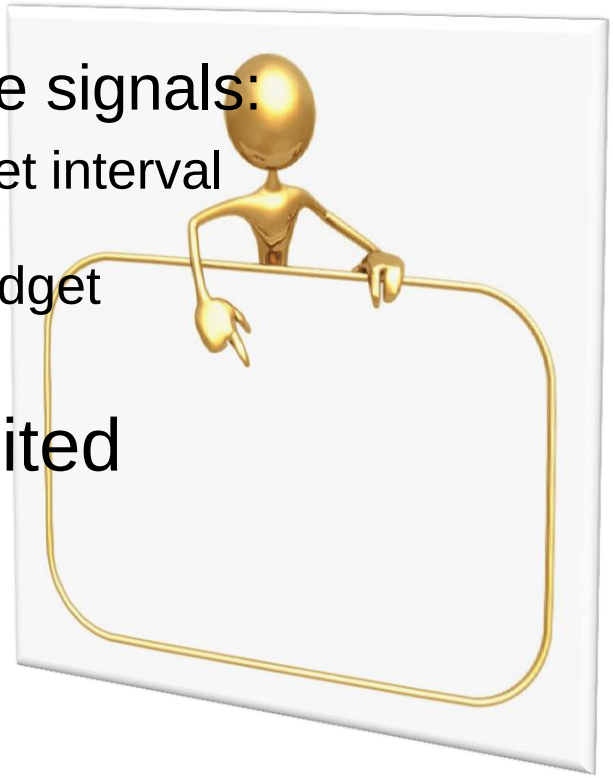
- Media Transmission requires (ITU-T Y.1541)
  - Predictable Delay
    - Application dependent
  - Predictable Reliability
    - Application dependent
- Coding delay essential
  - 64 kbit block length means:
    - 0.5 – n sec. for audio
    - 0.005 – 0.2 sec. video
- And additionally…

Class	IPTD	IPDV	IPLR	IPER	IPRR	Applications (examples)
0	100 ms	50 ms	1×10^{-3}	1×10^{-4}	–	Real-time, jitter sensitive, low delay, highly interactive
1	400 ms	50 ms	1×10^{-3}	1×10^{-4}	–	Real-time, jitter sensitive, medium delay, interactive
2	100 ms	U	1×10^{-3}	1×10^{-4}	–	Transaction data, low delay, highly interactive
3	400 ms	U	1×10^{-3}	1×10^{-4}	–	Transaction data, medium delay, interactive
4	1 s	U	1×10^{-3}	1×10^{-4}	–	Low loss
5	U	U	U	U	–	Best effort
6	100 ms	50 ms	1×10^{-5}	1×10^{-6}	1×10^{-6}	High bit rate, strictly low loss, low delay, highly interactive
7	400 ms	50 ms	1×10^{-5}	1×10^{-6}	1×10^{-6}	High bit rate, strictly low loss, medium delay, interactive

Notes – U: undefined
IPTD: IP Packet Transfer Delay
IPDV: IP Packet Delay Variation
IPLR: IP Packet Loss Rate
IPER: IP Packet Error Ratio
IPRR: IP Packet Reordering Ratio

Everything over IP

- IP makes things worse:
  - Not „*noisy*“ (AWGN) but „*lossy*“ (E[rasure]) channel
  - Complete IP packets get lost due to noise, contention and/or queuing
  - IP packet rate low even for high rate signals:
    - 4 Mbps SD video has ~2.5 ms IP packet interval (assumed 7 MPEG-2 TS packet per IP packet)
    - Small blocks already consume time budget (40 packets / 100 ms)
- Channel capacity has to be revisited
  - Capacity under delay constraints
    - Time dependent minimum redundancy
  - Residual error rate tolerable



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Intro

- TV broadcasting and mobile broadband are undoubtedly essential parts of today's society. Both of them are now facing tremendous challenges to cope with the future demands.
- Video is expected to contribute ~70% of all the mobile traffic by 2018.
- Content distribution is expected to be the dominant contributor to the mobile data traffic demand, therefore content media distribution is being more and more present in everyday life communications.

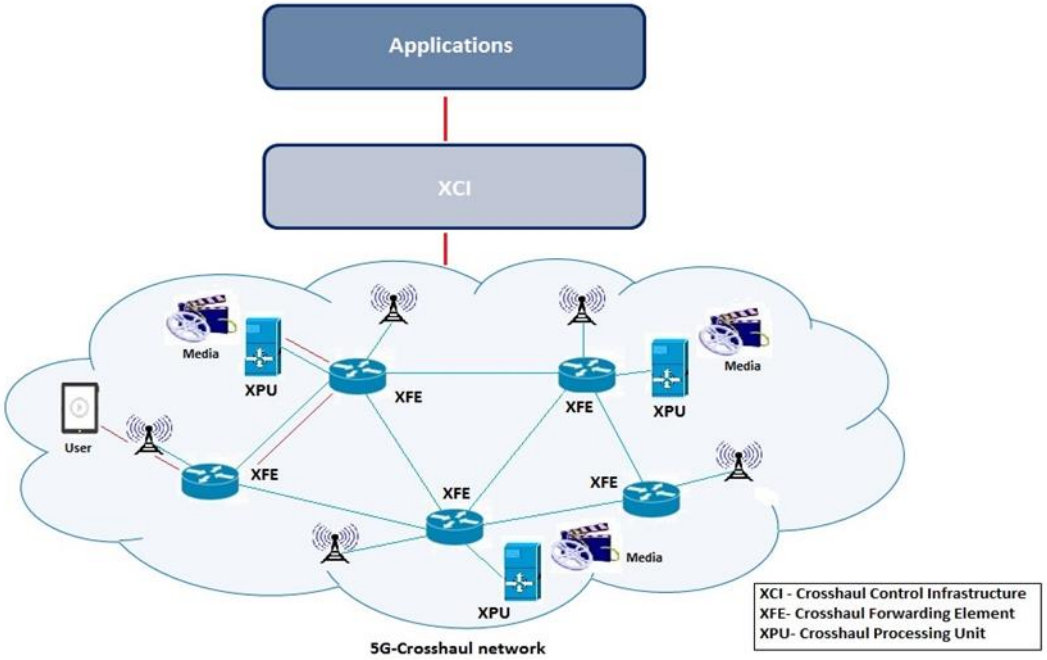
Demand

- The consumption is moving towards on-demand services so that anyone can access contents anytime, anywhere and regardless of the device type.
- And quality is going further very quickly, in most European countries people have become accustomed to HDTV and are now expecting even UHD..

- The trends of on-demand, mobile, and Ultra HD quality impose formidable challenges for TV and the delivery network of the future to be coped for 5G capabilities.

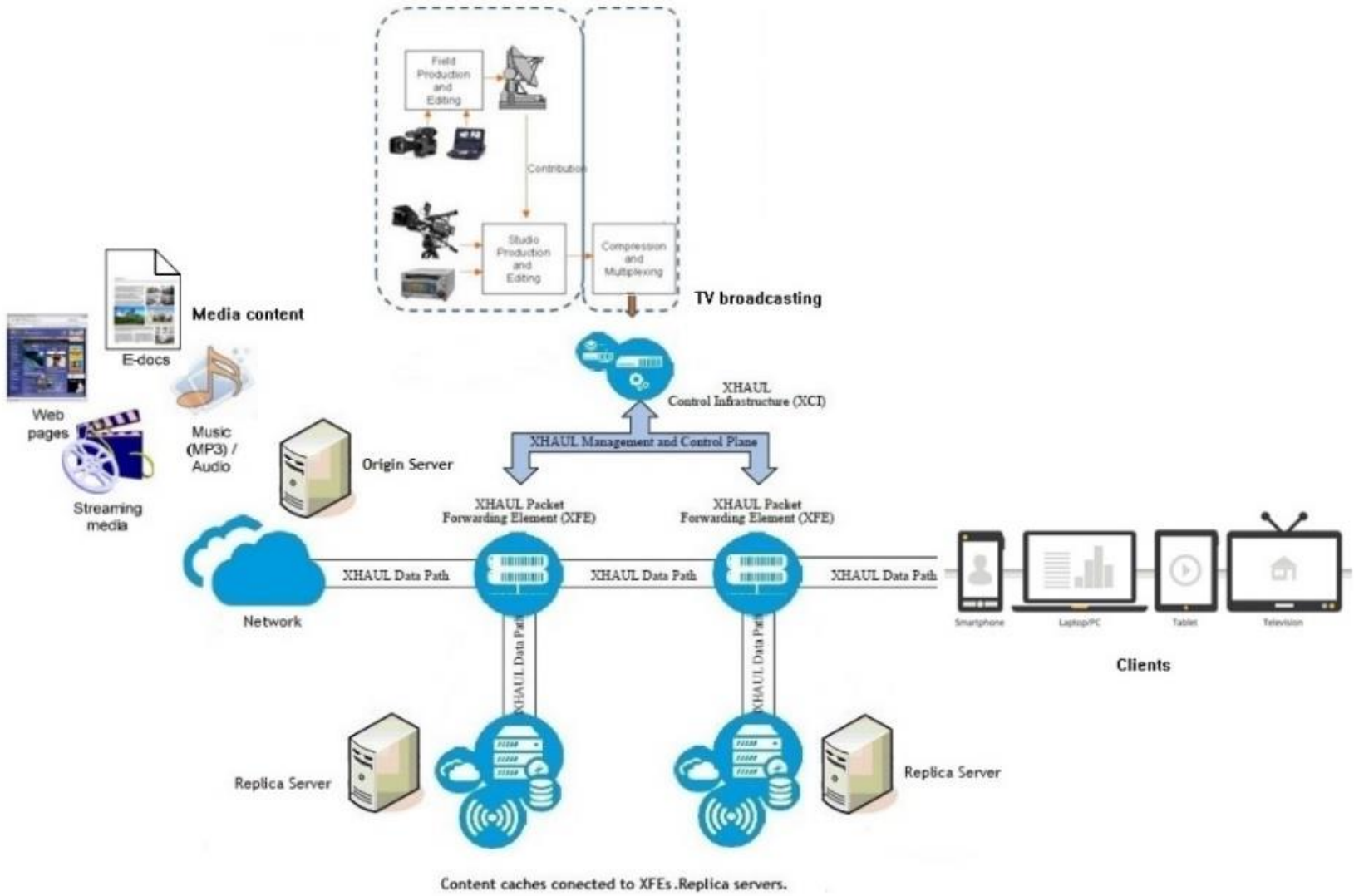
5G Media

- Media content distribution takes advantage of an adaptive and cost-efficient solution for the 5G transport network, integrating both fronthaul and backhaul segments.
- The envisioned solution requires a fully integrated and unified management of fronthaul/backhaul resources in a sharable, scalable and flexible way.



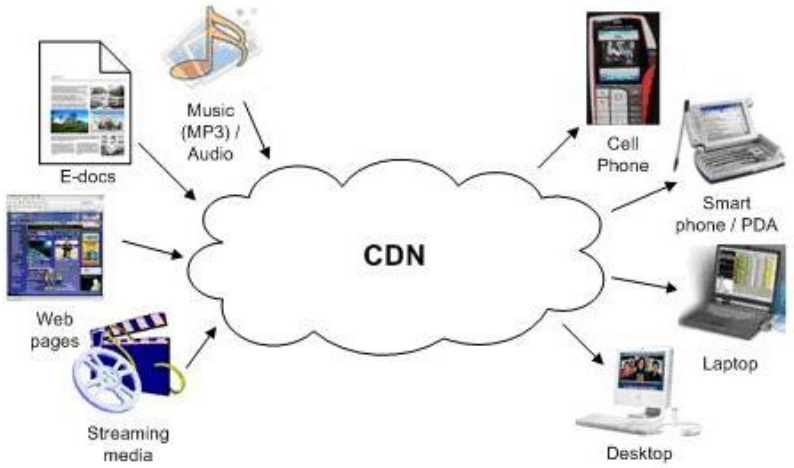
- The control and management of such an integrated transport network, will be based on the **SDN principles** and architecture defined by the Open Networking Foundation (ONF) and will adopt **Network Function Virtualisation** (NFV) concepts and mechanisms as well as aligned with the ETSI **Management and Orchestration** (MANO) **architecture** as a specific means to offer a subset of services of network, cloud and storage.

Media content distribution scenario



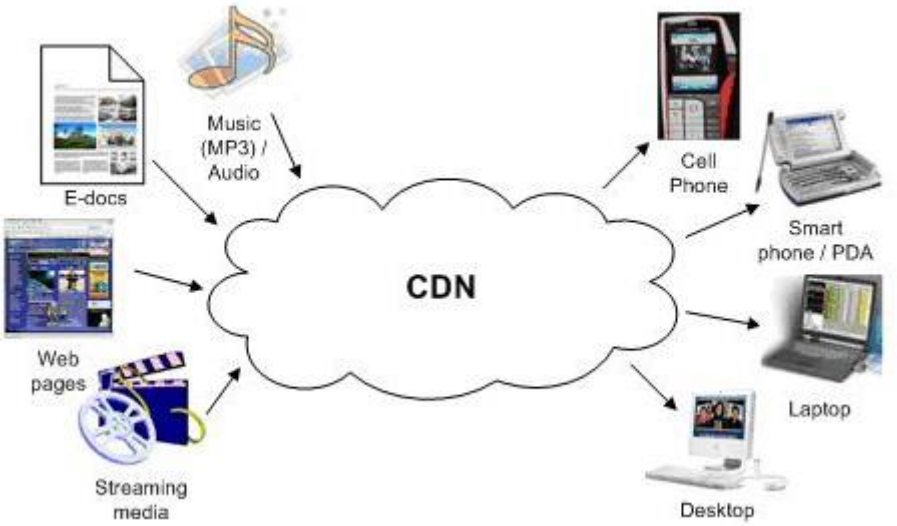
Content Delivery Networks

- A CDN is a combination of a content-delivery infrastructure, a request-routing infrastructure and a distribution infrastructure.



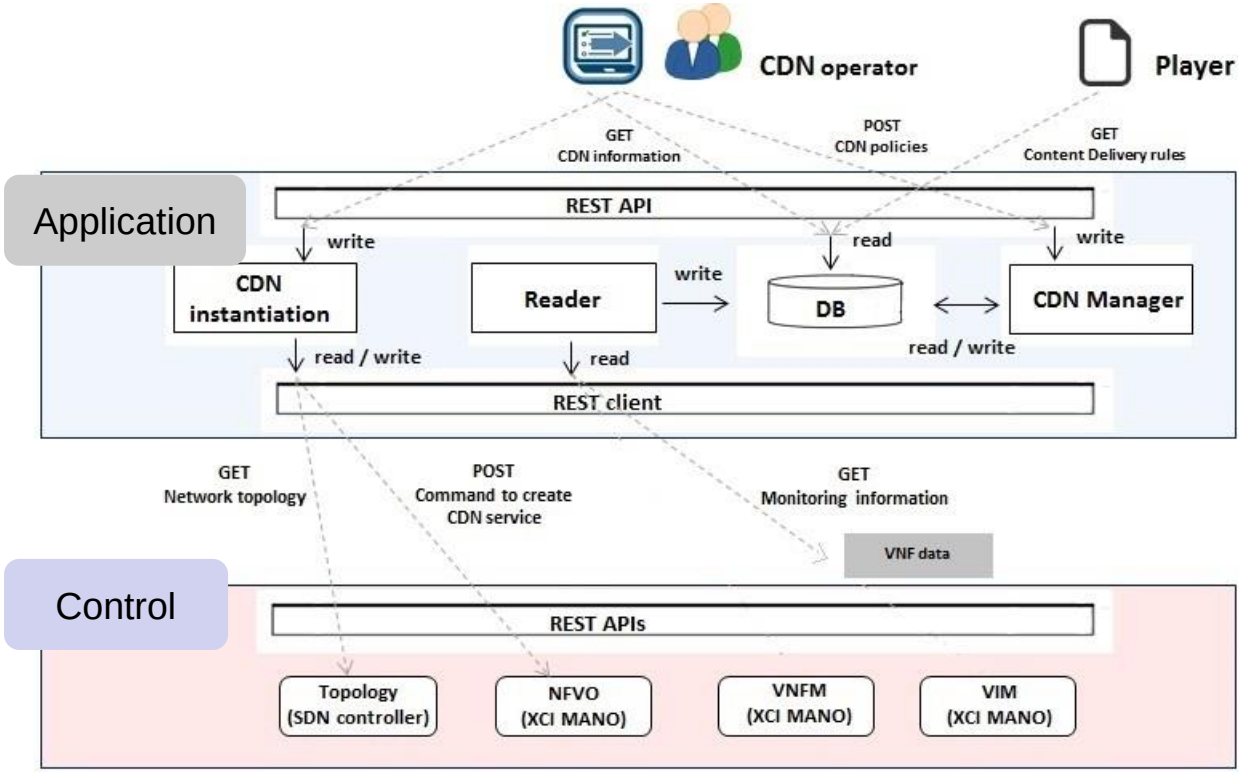
- Virtual CDNs will control the load balancing over several replica servers strategically placed at various locations in order to deal with massive content requests.
- This model will improve the content delivery maximizing bandwidth and improving accessibility through the CDN infrastructure according to the user demands.

Content Delivery Networks

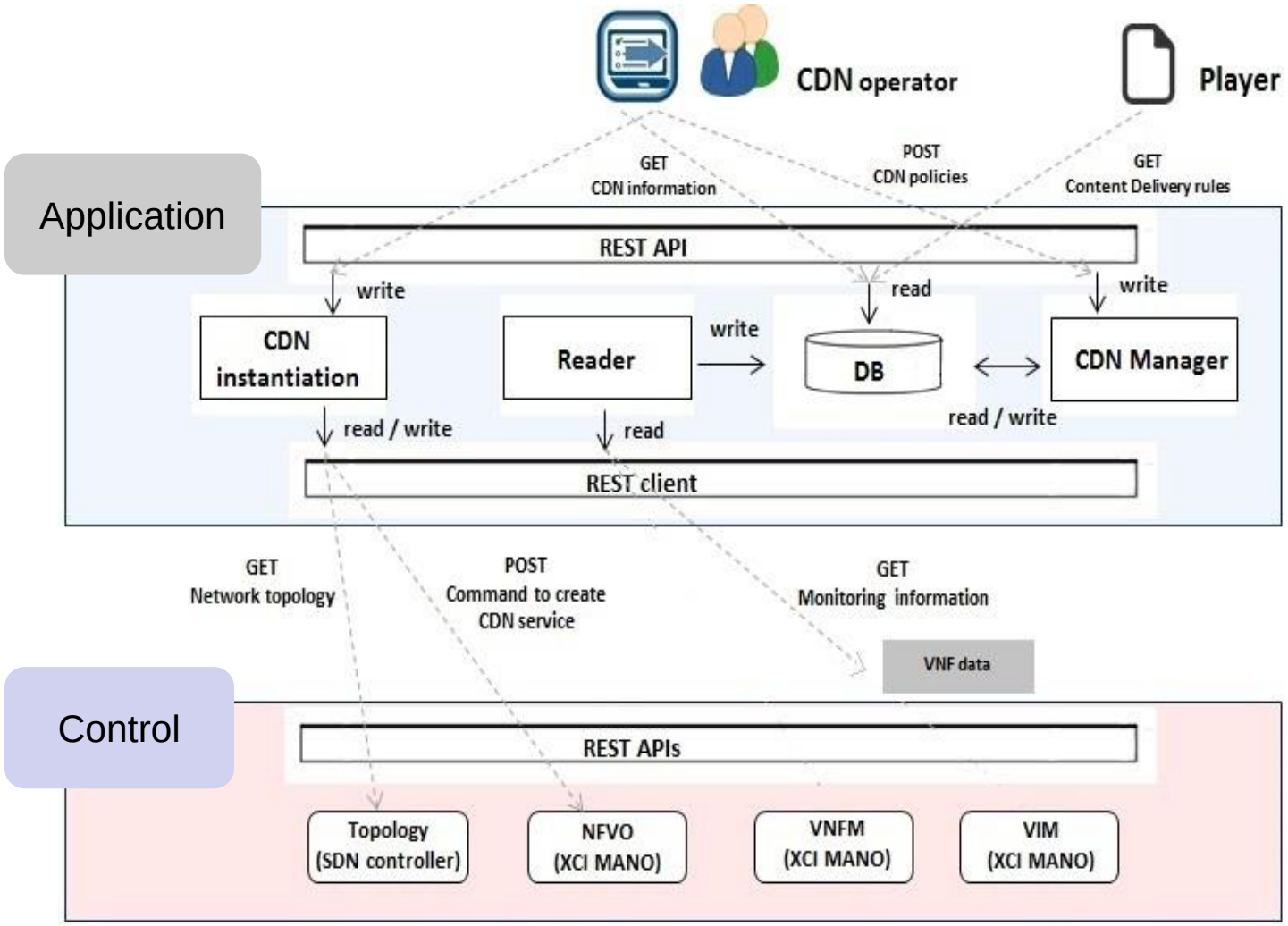


- The managing application running on the control plane will manage the entire CDN infrastructure and it will evaluate the system performance.
- It will receive monitoring information from the infrastructure (location, latency, packet-loss, bandwidth, server load) and information from the user about his location.

- Taking this information as input and applying optimization logic defined by the CDN operator based on the CDN and network metrics, the application strategically deploys and optimizes the replica servers in order to optimize resource utilization and deliver the content to the end users/content consumers with maximum possible QoE.



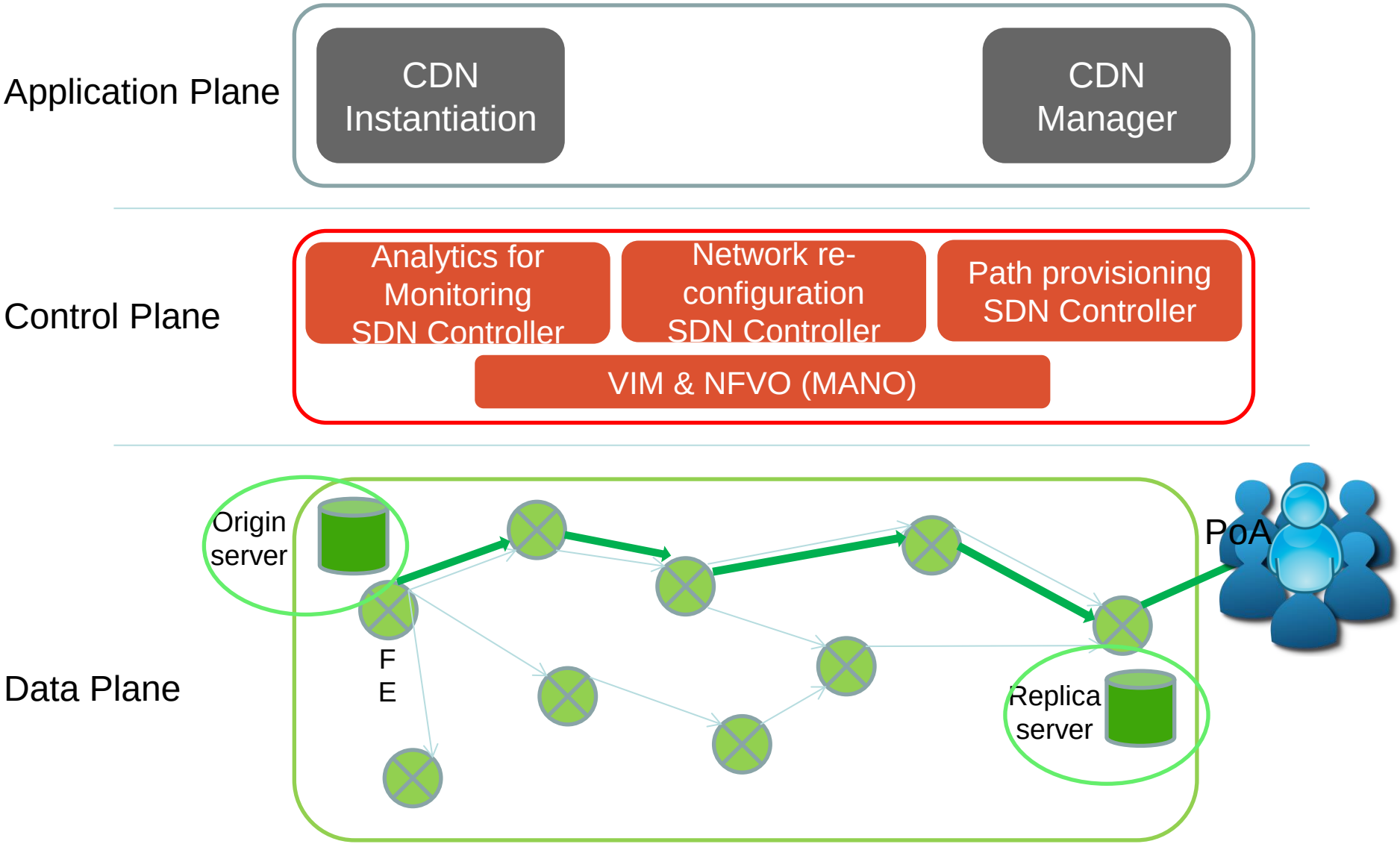
Content Delivery Networks



Content Delivery Networks

- The content delivery infrastructure will be implemented via networks of content caches -replica servers- which are deployed close to the FEs across the network topology.
- The request-routing and distribution infrastructure functions will be enforced through optimal content routing and delivery based on information from 5G network.

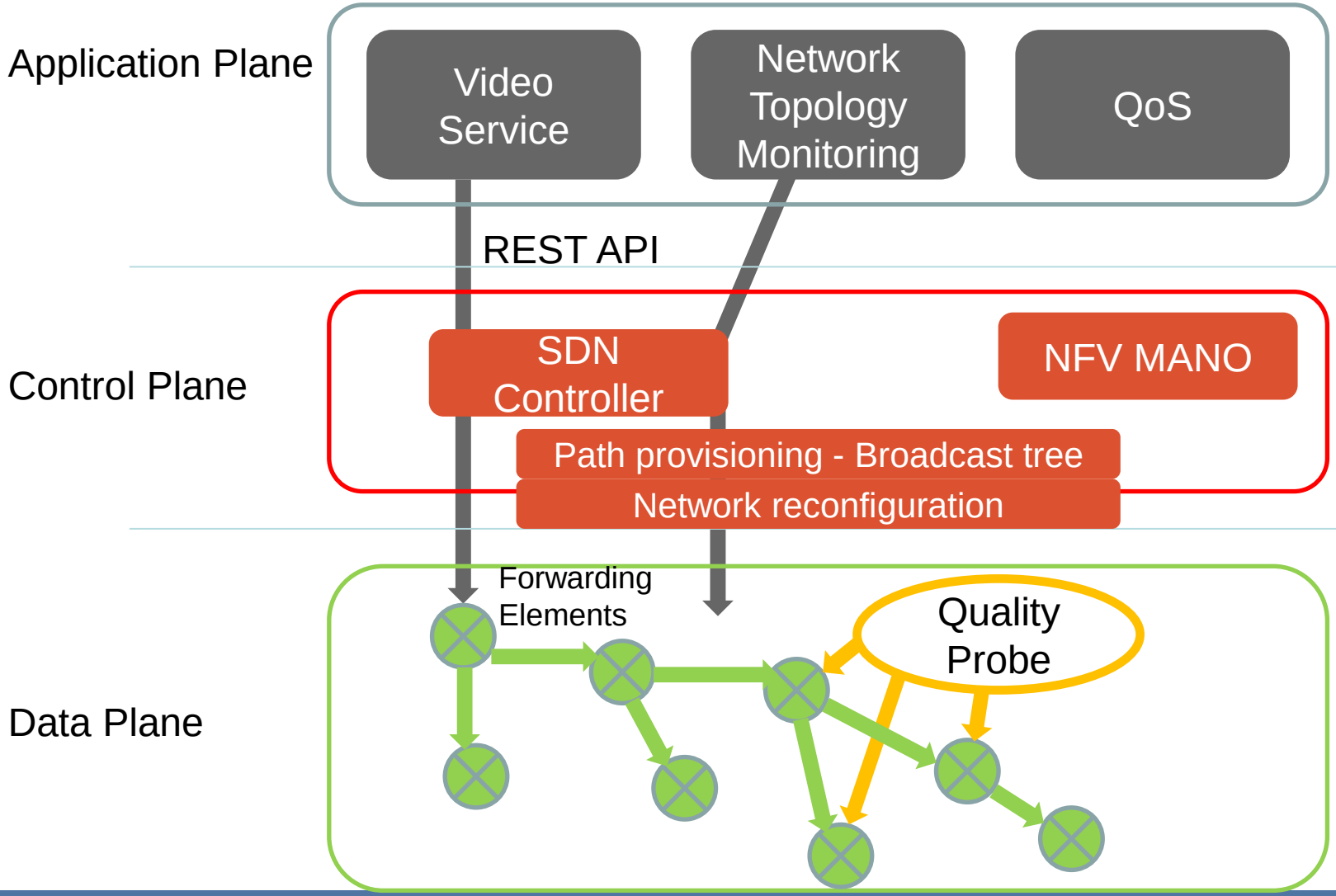
Content Delivery Networks



Broadcast solution

- Media/TV broadcasting & multicasting services utilizing the 5G using the same network with a controlled quality and offered as a Broadcast-as-a-service
- The focus is on minimizing both the cost and the spectrum consumption. Increase cost-effectiveness of transport technologies for media distribution
- Useful to demonstrate the feasibility of 5G objectives such as front and backhaul-integrated (control/planning) applications
- QoS based decision on the service provision (routing and quality adaptation) enables real-time decision

Broadcast solution

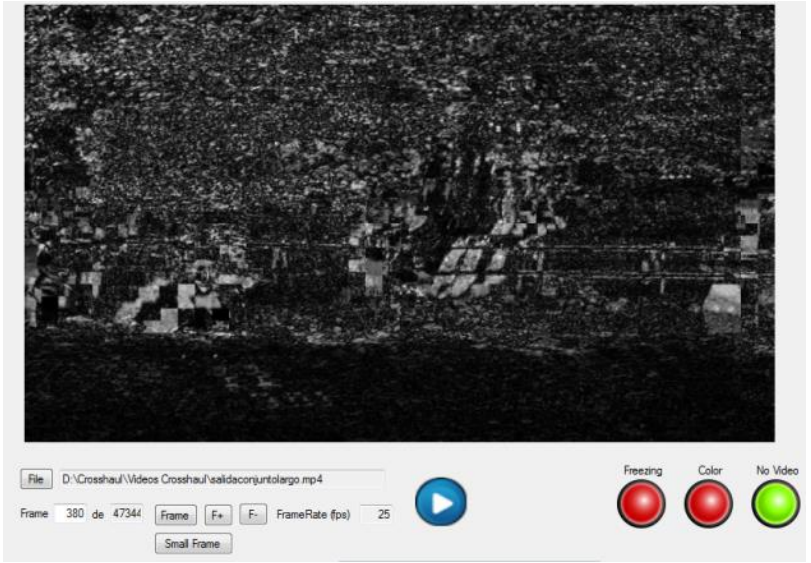


Quality Probe

- **Monitoring routing** is important to understand reachability and network paths that can affect network performance, leading to high packet loss or latency. Loss – Latency – Jitter
- **Network performance** is typically measured by the success of IP forwarding, the time and variation for packets to make it to the destination, and the theoretical capacity and actual bandwidth available.
  - Packet loss
  - Latency and jitter
  - Bandwidth
  - Undersized Path MTU and oversized TCP MSS

Quality Probe

- **Video quality assessment** is used for detecting transmission problems on the image. Frozen frames, black frames and NR metrics are used to feed back the service



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ONE: UdS Topology

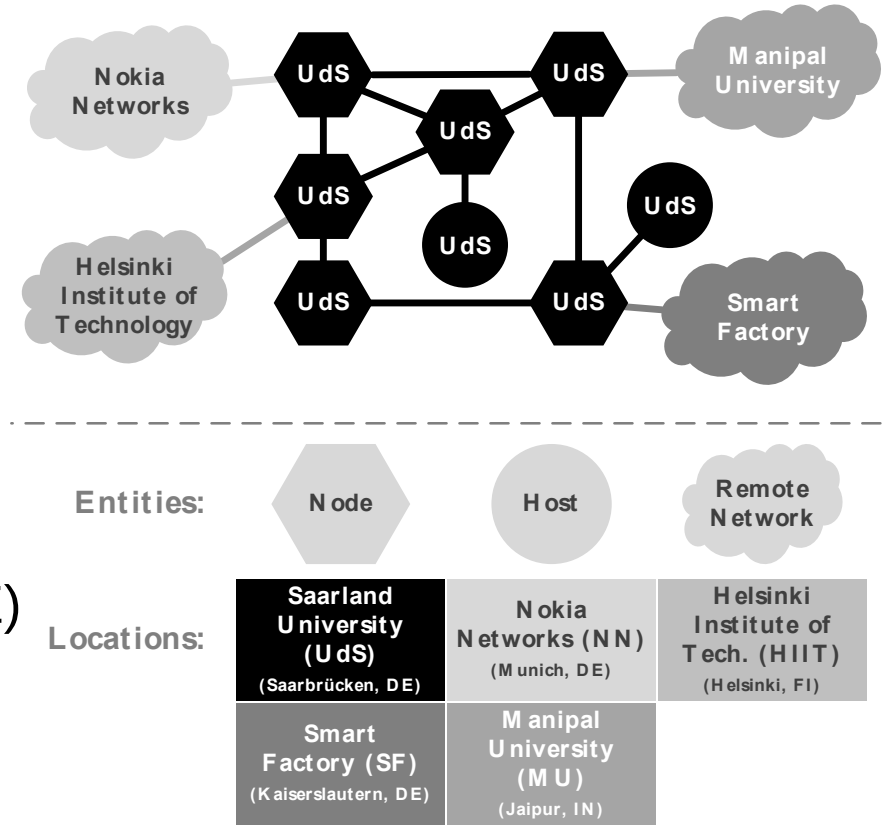
Devices

- Gateway (Orchestration Master, Firewall, VPN Endpoint).
- Nodes (Switching Units).
- Devices (End-Hosts).
- Relays (Tx Optimization).

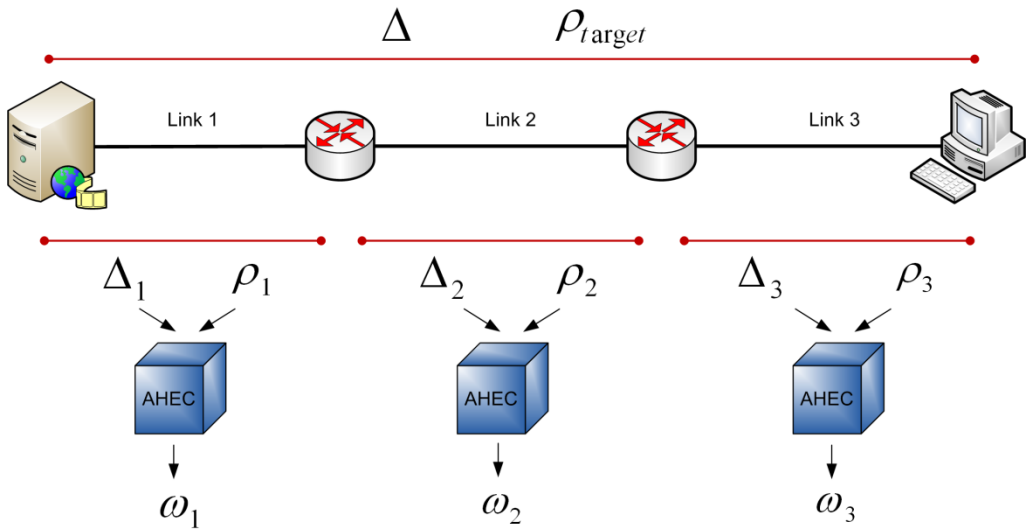
Partners

- SmartFactory (Kaiserslautern, DE)
- Nokia Networks (Munich, DE)
- HIIT (Helsinki, FI)
- Manipal University (IN)

- We operate ON@UdS

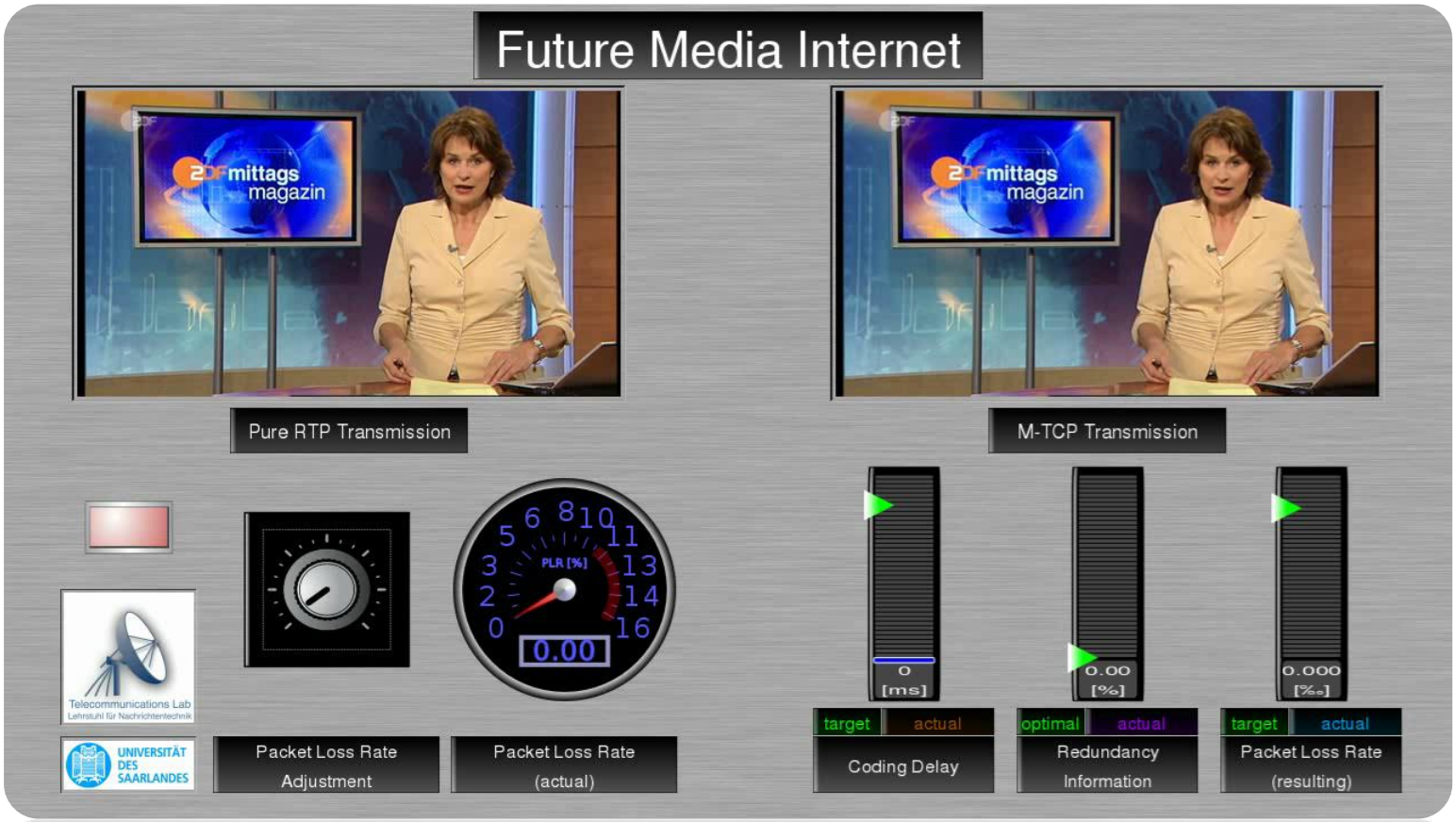


- Multi-Link / Multi-Hop
  - uses individual segment properties
  - intermediate nodes act as error correction relay
  - apply AHEC as atomic unit
  - Expect significant coding gain



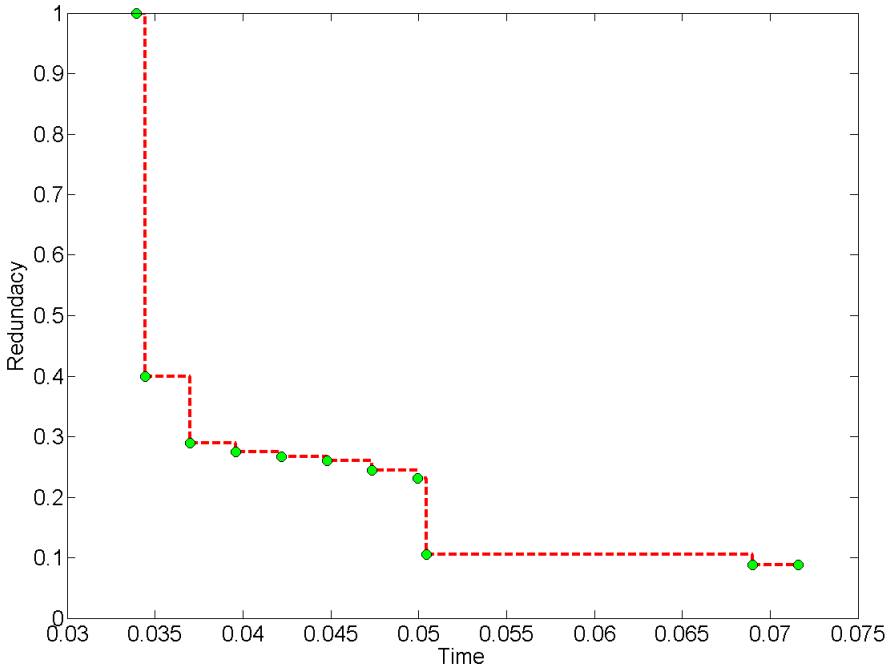
Error Correction in „Overlay Mode“

PRRT Demo


Future Media Internet
Pure RTP Transmission
M-TCP Transmission
Telecommunications Lab
Lehrstuhl für Nachrichtentechnik
UNIVERSITÄT DES SAARLANDES
PLR [%]
0.00
0 [ms]
0.00 [%]
0.000 [‰]
target
actual
optimal
actual
target
actual
Packet Loss Rate Adjustment
Packet Loss Rate (actual)
Coding Delay
Redundancy Information
Packet Loss Rate (resulting)

Time vs. Redundancy

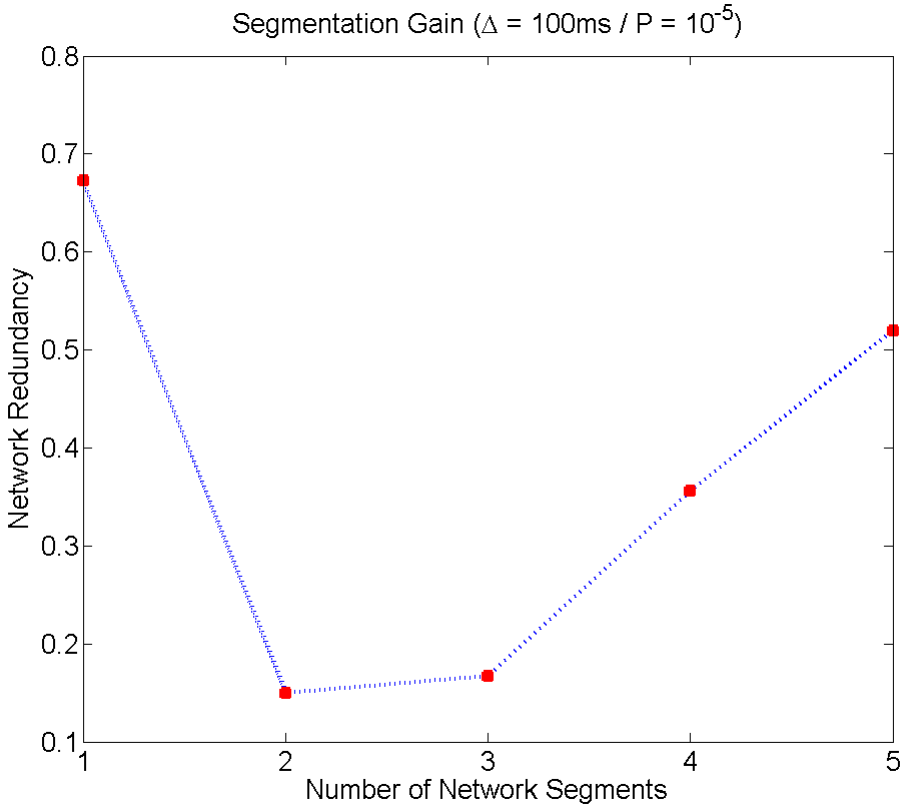
- Typical redundancy/coding-time behavior of a link:



  - More time leads to a higher coding efficiency (Shannon Theorem)
  - Monotonously decreasing shape
  - Discrete values (time/redundancy)

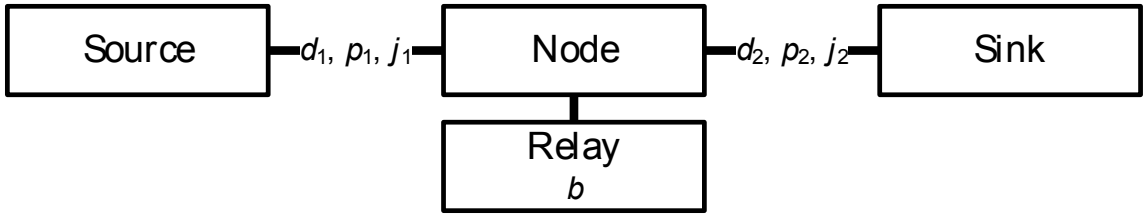
Saturation

- Bathtube shape of overall coding gain due to:
  - Available time per segment decreases linearly …
- … whereas …
  - Redundancy per link increases super-linearly



TTS: Evaluations

Setup



Results

- *Metric*: Transmission time [ms] for a TCP data stream.

Parameters

- Delays: d_1=50 ms, d_2=10 ms
- Jitter: j_1=5 ms, j_2=1 ms
- Loss: p_1=10^{-8}, p_2=3%

Consequences

- Mean and StdDev decrease (latencies decrease and jitter reduces).
- Median and minimum increase as there is more overhead.

Case	Mean	StdDev	Median	Min	Max
E2E	146.686	121.456	122.319	110.234	4753.730
TTS	140.427	104.601	123.371	110.660	5236.300
	+4.457%	+16.114%	-0.852%	-0.384%	-9.216%

Improvement for Delay

Case	d_1 (ms)	j_1 (ms)	p_1 (%)	d_2 (ms)	j_2 (ms)	p_2 (%)	b [Byte]	A_{12}	p
Buffer (Very Small)	50	5	1e-06	10	1	1e-06	16	0.065	<1e-04
Buffer (Small)	50	5	1e-06	10	1	1e-06	64	0.729	<1e-04
Buffer (Big)	50	5	1e-06	10	1	1e-06	1024	0.892	<1e-04
High Error Rate	50	5	1e-06	10	1	3	1024	0.917	<1e-04
No Errors	50	5	0	10	1	0	1024	0.901	<1e-04
High Jitter	50	50	1e-06	50	50	1e-06	1024	0.688	<1e-04
Natural	50	5	1e-04	5	1	1e-04	1024	0.841	<1e-04

- Scenario: Two Links, one split using 1 relay
- Parameters: Delay, Jitter, PLR, Relay Buffer Size
- Measured Metric: Transmission Time for all Data of a TCP Stream
- Performance Metric: A_{12} score (stochastic superiority[1])
  ($A_{12} < 0.5$: E2E better, $A_{12} > 0.5$: TTS better)

[1] A. Vargha and H. D. Delaney, **"A Critique and Improvement of the CL Common Language Effect Size Statistics of McGraw and Wong"**, Journal of Educational and Behavioral Statistics, vol. 25, no. 2, pp. 101–132, 2000.

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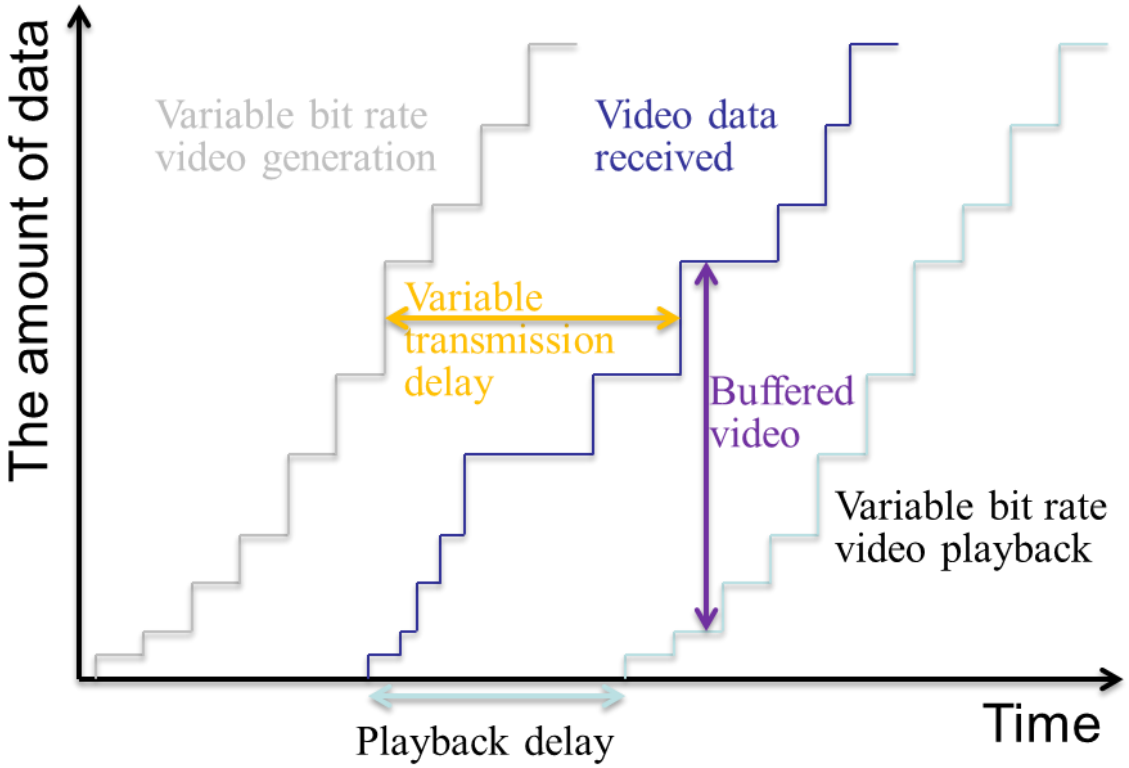
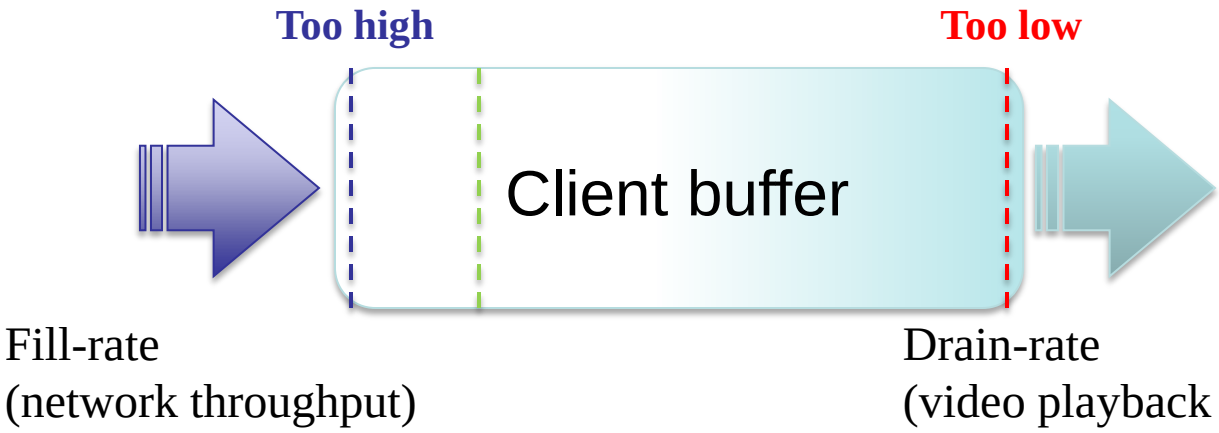
Buffering Delay



- Buffering Delay is buffered video in seconds.
- We achieve low-latency dynamic video streaming with buffering delays as low as the chunk-duration.

A Model of Buffer Dynamics



Quality Selection

Stabilizing the buffer to the desired level by regulating the drain-rate, i.e. by selecting a video bit rate for the chunk.

A Model of Buffer Dynamics

Fill-rate

$\frac{a(t)}{r(t)}$

Drain-rate

1

Express the buffer level in **seconds of video**.

$a(t)$: the throughput rate achieved at the time t

$r(t)$: the selected video bit rate at the time t

A Model of Buffer Dynamics

Compute the buffer level **every discrete chunk**.

$a[i]$	:	the throughput rate achieved during the reception of chunk i	$r[i]$	:	the selected video bit rate of chunk i
			$t[i]$	:	the reception duration of chunk i

A Model of Buffer Dynamics

Compute the buffer level **every discrete chunk**.

$a[i]$	:	the throughput rate achieved during the reception of chunk i	$r[i]$	:	the selected video bit rate of chunk i
			T_c	:	the chunk duration

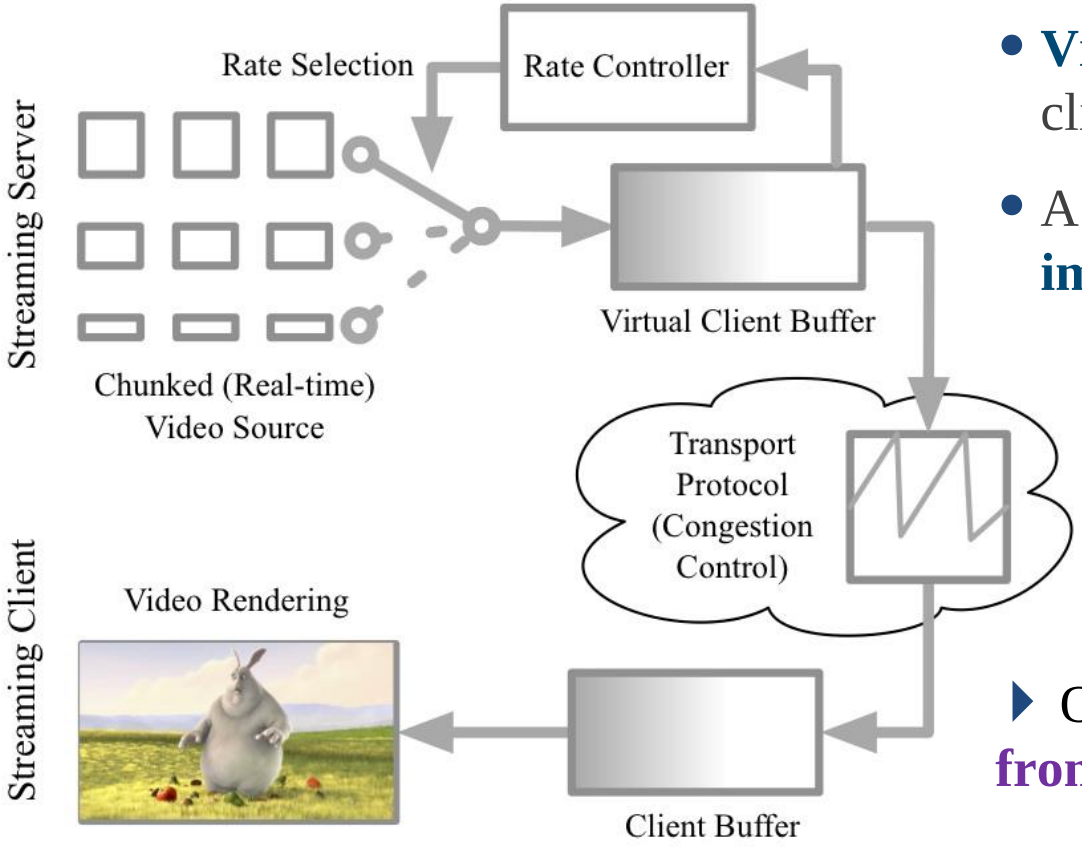
A Model of Buffer Dynamics

Current buffer level

$$b[i] = b[i-1] + T_c - T_c \cdot \frac{r[i]}{a[i]}$$

$b[i]$: the buffer level (in seconds) when the client finishes the reception of chunk i

Open-Loop rAte Control (OLAC)

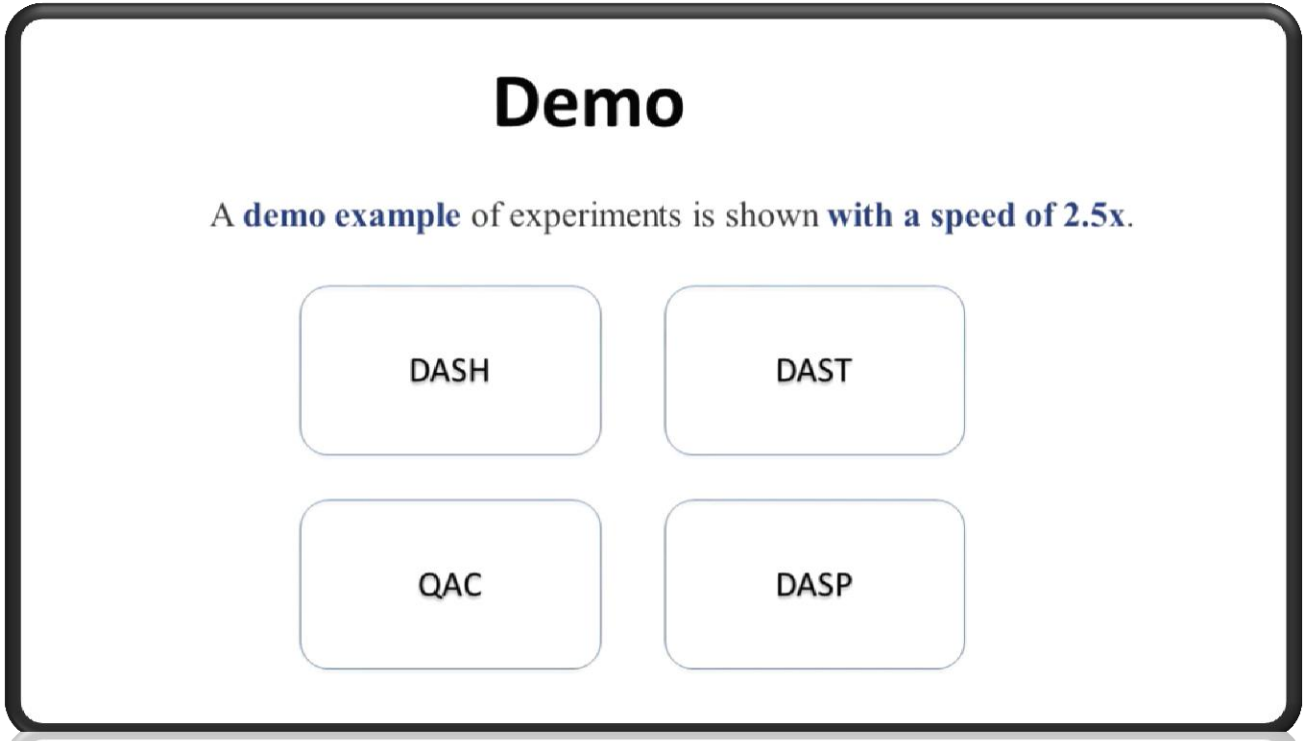


- **Virtual client buffer** simulates client buffer on the server.
- A rate control on the server offers **immediate feedback** from clients.

▸ Our buffer stabilization **benefits from OLAC**:

- **user-perceived quality** improves by up to 68%.

BDS Demo



https://youtu.be/G5R1Uj9fBLo

https://youtu.be/HQhlZWGPEYU

Summary

- Video is going IP (in all domains)
  - Low Latency is extremely important
  - Latency and reliability need to be balanced
- We introduced:
  - 5G Media Broadcast through SDNs
  - Predictably Reliable Real-Time Transport
  - Transparent Transmission Segmentation
    - Optimized link-segmentation
  - Buffer Dynamics Stabilizer
    - Buffer sizes as small as a single chunk size