

## 4. Simulation Results

### 4.1 KPI evaluated through simulations

#### 4.1.1 End-to-end quality

The end-to-end video quality is evaluated through multiple objective metrics, namely PSNR and SSIM. This choice allows to validate the numerical results reliably and, at the same time, to compare them easily with other future solutions. A subjective evaluation of the results obtained with the simulator will be also performed in parallel with the analysis of demonstration results and will be included in the final deliverable of the work package, namely D6.5 “Report on final validation”.

In addition to the quality metrics, the QoE is evaluated through the end-to-end delivery delay. Low latency enables real-time or quasi real-time services, characterized by the possibility for the remote specialist to provide an interactive support to the on-site staff.

#### 4.1.2 Transmission Quality

Radio link quality is evaluated by means of traditional metrics, such as achieved throughput and link delay. In particular, in the considered use cases we will focus on the performance across the LTE mobile radio access, as it constitutes the most critical wireless link.

### 4.2 Simulated scenarios

#### 4.2.1 Ambulance and emergency area

This scenario mainly focuses on multimedia communication between the ambulance on the move and hospital 2 where a doctor can utilise multimedia content from the ambulance for different purposes such as monitoring, coordination and/or consultation. The main video and medical data streams are originated by pieces of equipment on the ambulance and are received by a client at the hospital. At the hospital a doctor, using these video and data, can monitor the patient condition and coordinate the first-aid operations. Some medical devices may be connected via radio with the ambulance equipment. Since the ambulance is typically moving, sometimes also at high speed, a 4G link is supposed to be available to communicate with the hospital. The simulated scenario is illustrated in Figure 8.

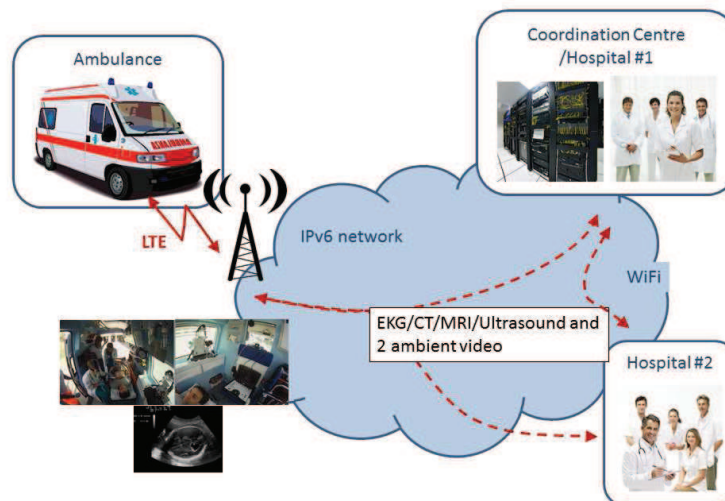


Figure 8 – Simulated scenario

The simulation is focused on transmission of two ambient video streams and one ultrasound sequence acquired inside the ambulance. In this way we simulate the medical assistance performed after loading the patient on the vehicle with the support of advanced tele-diagnosis services. The ambulance is assumed to be moving (most likely fast) towards the hospital.

The main characteristics of the options implemented in the Concerto simulator and evaluated in this user scenario are summarized in the following.

- **Ambulance** - The processing unit on-board the ambulance is responsible to collect and aggregate the multimedia streams from two ambient cameras and from one medical device available in the ambulance and to transmit them through the LTE link, toward the CC/hospital. To perform its processing, the ambulance exploits the ranking information received from the CC and the achievable throughput across the LTE link. The target video coding rates of the selected sources are jointly adapted to guarantee the optimal level of quality (in terms of PSNR and SSIM) compatible with the available radio resources and communicated to the application controller of each encoder. Two techniques have been compared to perform rate adaptation:
  - Equal rate (**ER**), in which the available throughput is equally divided by the number of sources selected for transmission, without taking into account the final video quality.
  - Quality fair (**QF**), in which the target source rates are jointly determined splitting the overall throughput not uniformly, but aiming at achieving the same video quality for two ambient streams. On the contrary, for the medical stream a higher quality is selected, in order to allow tele-diagnosis support.

Together with the rate adaptation, the efficiency of packet-level forward error correction technique (PL-FEC) developed in WP4 based on non-binary LDPC codes have been evaluated in this scenario. Three cases have been evaluated:

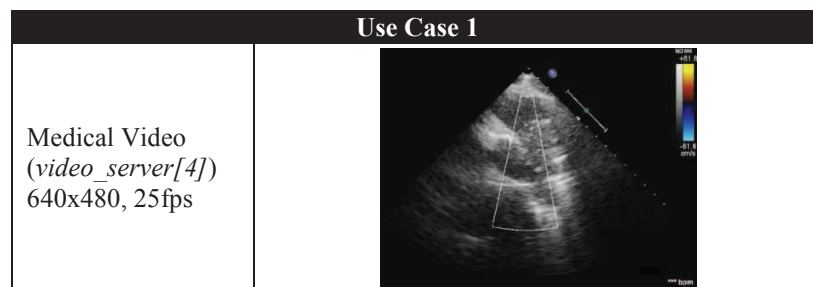
- PL-FEC is disabled with equal rate (ER) adaptation technique (benchmark)
  - PL-FEC is enabled with equal rate (ER) adaptation technique
  - PL-FEC is enabled with quality fair (QF) adaptation technique
- **LTE eNodeB** – Two scheduling and radio resource allocation techniques have been implemented in the simulator: a traditional proportional fair (**PF**) approach and the **GBR-ALRE** solution described in D4.3 [14]. In the latter case, the ambulance is treated by the LTE access network as a privileged user, and a guaranteed bitrate is provided by the system based on a low-complexity processing scheme fully compatible with the LTE uplink standard. The simulator takes into account the presence of additional non-Concerto users in the cell requiring bandwidth for their uplink transmissions. These techniques have been evaluated with the three cases of optimization techniques described above.
  - **Coordination Center** –In addition, this scenario is used for comparing RTP and HTTP-based streaming mainly in terms of delay achieved using HTTP-based streaming instead of RTP-based streaming. As HTTP streaming uses TCP, only near real-time streaming is possible to achieve. However, the use of TCP instead of UDP can offer reliability for critical medical data, which justifies its usage in some scenarios. In order to validate the trade-off between real-time requirements and smooth, error-free delivery, simulations of using HTTP-based streaming instead of RTP-based streaming from the coordination centre (or data base located in the CC/hospital) to hospital 2 (e.g. a specialist on the move with handheld device connected to the hospital WiFi wireless network) are carried out.



#### 4.2.1.1 Simulation Parameters

In this paragraph, the main simulation settings used for Use Case 1 are reported, with reference to different sections of the communication chain.

##### Video Sources:

The initial encoding rate for each camera has been set to 500kbps. The rate adaptation period has been set equal to the GOP size, namely 1s (25 frames).



Ambient Camera 5 ( <i>video_server[5]</i> ) 640x480, 25fps	
Ambient Camera 6 ( <i>video_server[6]</i> ) 640x480, 25fps	

Emergency Area	
Type of radio link	IEEE 802.11g
Channel model	ITU Commercial Area
Modulation	OFDM (QPSK on each subcarrier)
PHY FEC	Punctured convolutional code, $R_c=1/2$
Cyclic Prefix	1/16

4G Link	
Type of radio link	LTE Uplink
Duplexing Mode	FDD
Modulation	OFDM (adaptive constellation based on AMC)
Multiple Access Type	SC-FDMA
Minimum Allocable Resource Block	1 PRB (=12 subcarriers x 7 OFDM symbols)
PHY FEC	Turbo code (adaptive $R_c$ based on AMC)
Cyclic Prefix	4.69us (short CP)
Adaptive Modulation and Coding (AMC)	Enabled
Radio Resource Management	PF / GBR-ALRE
Ambulance (Tx)	
<i>Channel model</i>	ITU Vehicular A
<i>Velocity</i>	50 km/h
<i>Max TX Power</i>	18dBm
<i>Antenna Gain</i>	2.0dBi
<i>Nr. of Antennas</i>	1
<i>Lognormal Shadowing sigma dB</i>	4dB
<i>Initial distance from the eNodeB</i>	500m
eNodeB (Rx)	
<i>Antenna Gain</i>	18.0dBi
<i>Nr. of Antennas</i>	2
<i>Cable Loss</i>	2.0dB
<i>Rx Noise Figure</i>	2.0dB
Additional Non Concerto Users (Tx)	
<i>Nr. of Non Concerto Users</i>	10
<i>Channel model</i>	ITU Pedestrian A
<i>Distance to the eNodeB</i>	Between 50m and 1500m

Hospital Area	
Type of radio link	IEEE 802.11g
Channel model	ITU Office Area
Modulation	OFDM (QPSK on each subcarrier)
PHY FEC	Punctured convolutional code, $R_c=1/2$
Doctor's UE velocity	3 km/h
Distance to the AP	Between 1.5m and 30m
Shadowing Sigma dB	Between 4dB and 6dB

**4.2.1.2 Use Case 1: Numerical Results and comparison**

The simulation campaign realized to evaluate the Use Case 1 consists in a series of tests taking into account different combinations of optimization techniques developed in the project focusing especially on PL-FEC and comparison of RTP and HTTP-based streaming in the final link between the coordination centre and the hospital. In Figure 9, the set of performed simulations is reported where the Sim.UCI.00 case corresponds to the system benchmark, representing what may be achieved nowadays with traditional technologies. We compare this system benchmark with different rate adaptation techniques (ER and QF) at the ambulance with PL-FEC enabled or disabled, different resource allocation techniques at LTE eNodeB (PF and GBR-ALRE) and either RTP or HTTP-based streaming from the CC to the hospital.

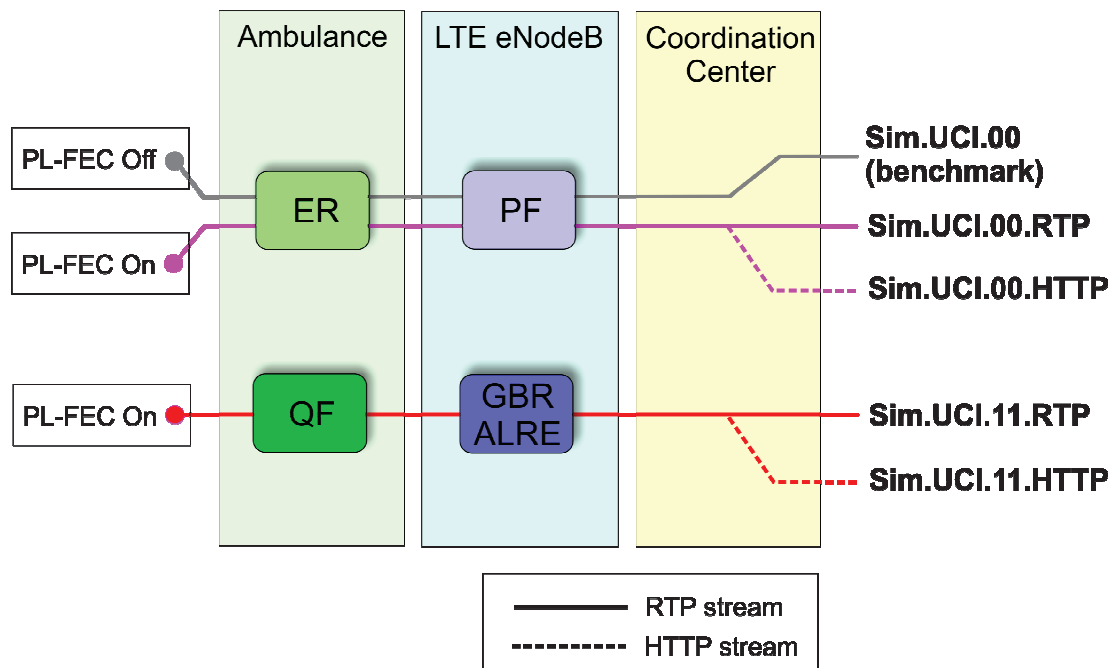


Figure 9 - Scheme of the simulation campaign realized for the Use Case 1.

The entire set of simulations reported in Figure 9 has been realized assuming small amount of transmission errors across the radio links (realistically simulated) and packet losses within the IPv6 core network ( $PLR=5E-3$ ).

The HTTP simulations are divided into two parts. In the first part, the medical ultrasound video is transmitted from the Emergency Area to the Coordination Center. In the second part, the received video is transcoded and forwarded to Hospital 2 using HTTP streaming (MPEG-DASH). For MPEG-DASH, the received RTP stream in the Coordination Center is multiplexed with one second segment length with one representation field presented in the playlist file (.mpd). Retransmission minimum timeout (*RTO*) value was set to 300ms in order to avoid long TCP pauses.

The analysis of the simulation results has been carried out considering two steps. In the first one, we evaluated the performance of the proposed PL-FEC schemes (see 3.11). As it is shown in Figure 9, in the benchmark case the PL-

FEC module is deactivated while in Sim.UCI.00.RTP and Sim.UCI.11.RTP it is operating in order to prevent packet drops during the wireless transmission and losses during the IP routing and delivery.

In the second analysis step, all the simulations are compared (benchmark, RTP-based and HTTP-based): the end-to-end performance and final video quality are evaluated considering specifically the Medical ultrasound stream (*video\_server [4]*) as it constitutes the most important video sequence.

**PL-FEC Performance**

Figure 10 and Figure 11 show the received video quality, in terms of PSNR and SSIM for the RTP-based simulations. In the bottom right sub-figures, the mean PSNR and SSIM value is depicted for each user.

In the benchmark case, it is noticeable how, in presence of transmission errors and packet dropping, the final quality can be dramatically compromised. On the contrary, the application of a PL-FEC scheme based on a non-binary LDPC code developed in WP4 allows to reach satisfying level of quality recovering the packet losses. In the reported simulations, we designed a LDPC code built on GF(4), with  $k=2048$  and  $n=3072$  (i.e.  $Rc=2/3$ ). However, in Sim.UCI.00.RTP the medical video does not obtain a sufficient level of quality to be used for medical purposes such as tele-diagnosis. Minimum tolerable values for diagnostic evaluations are 35 dB of PSNR and 0.95 of SSIM. Finally, the combination of the PL-FEC strategy and the proposed quality fairness and GBR-ALRE criteria provides the suitable protection from packet losses and at the same time guarantees higher level of quality for the medical user and fair quality for the remaining ambient videos.

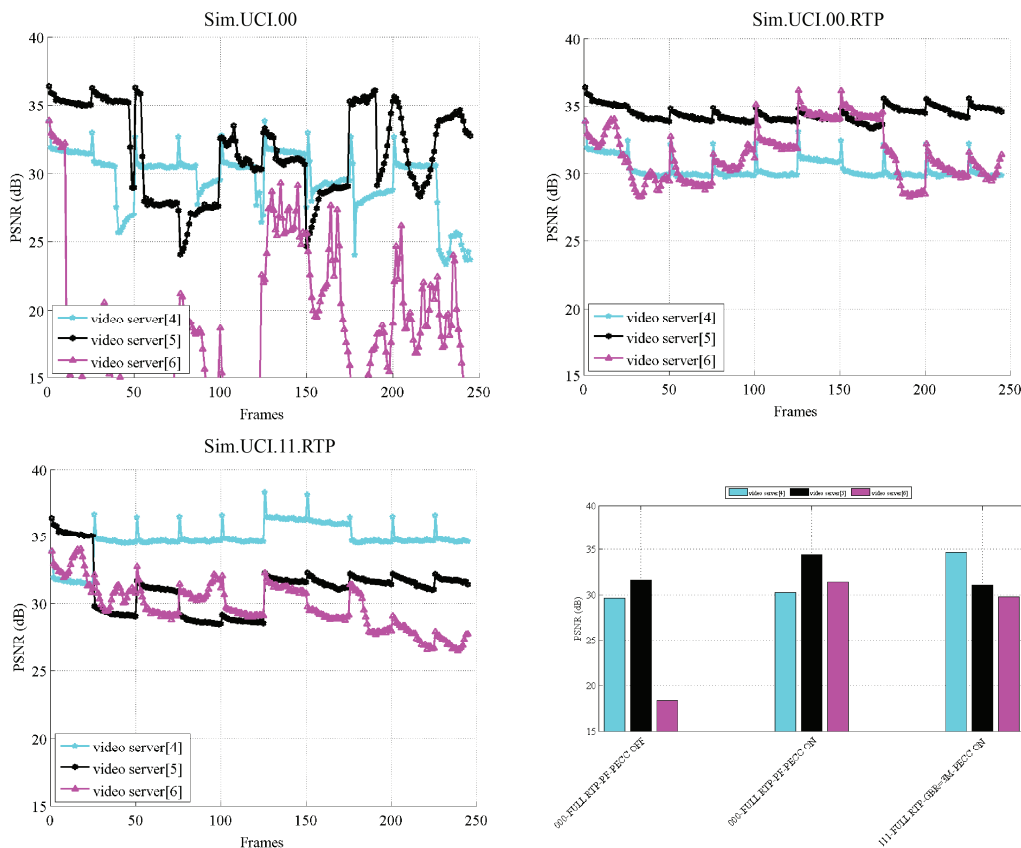


Figure 10 - Sim.UCI – End-to-end PSNR for RTP-based simulations. Comparison of results obtained with and without PL-FEC protection.

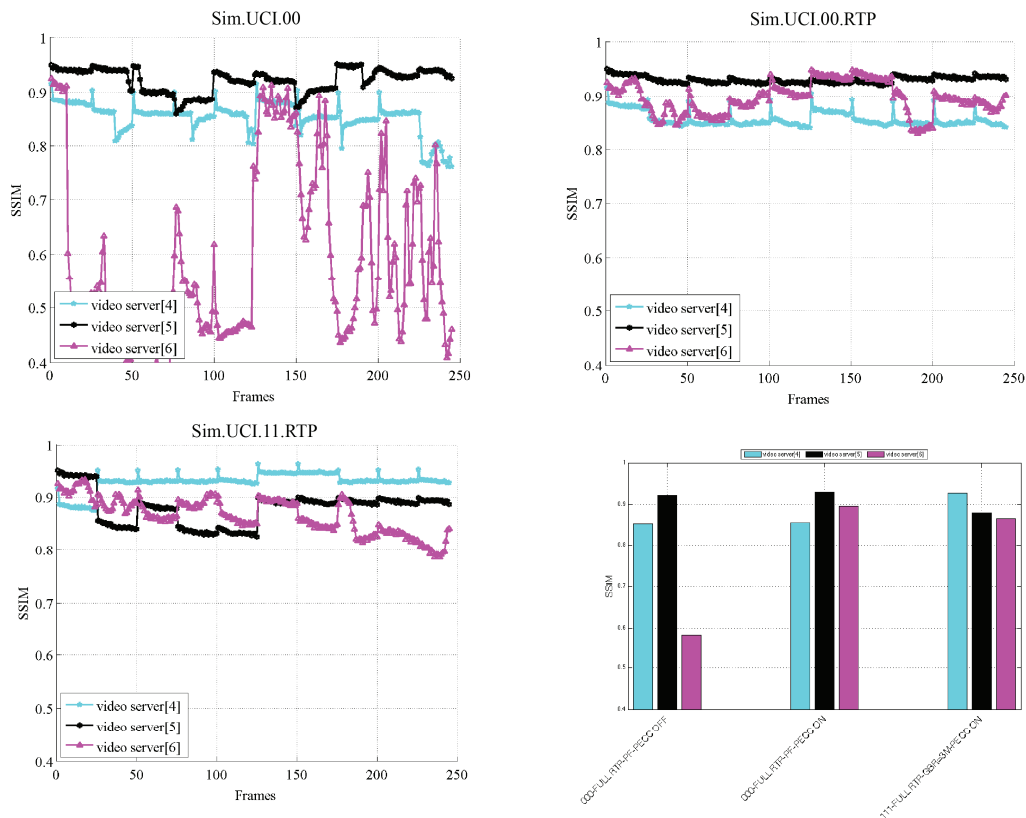


Figure 11 - Sim.UCI – End-to-end SSIM for RTP-based simulations. Comparison of results obtained with and without PL-FEC protection.

**End-to-end Performance**

In the following, we present the results concerning the end-to-end performance of the adaptive rate allocation for the transmitted video and comparison between the RTP and HTTP-based video streaming.

Figure 12 and Figure 13 show the total end-to-end quality and delay curves for the received video. The CONCERTO optimised techniques in Sim.UCI.11.RTP/HTTP provide the best final quality. The small difference between Sim.UCI.11(/00).RTP and Sim.UCI.11(/00).HTTP is explained due to transcoding phase in CC from RTP to MPEG-DASH. Also, the delay is the smallest for Sim.UCI.11.RTP with the highest quality. Naturally, the TCP characteristics and retransmissions extend the delays for RTP+HTTP simulations.

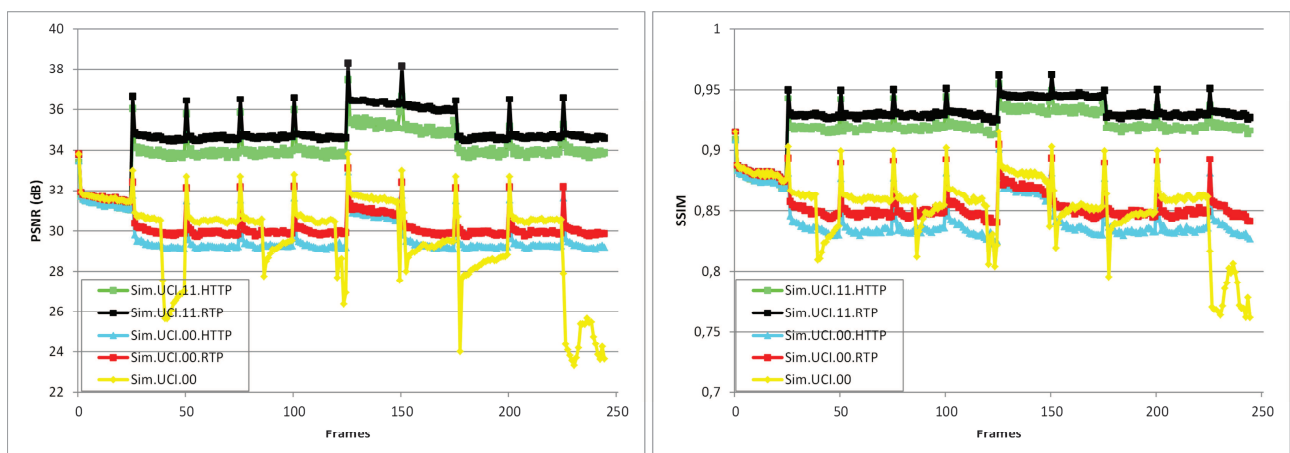


Figure 12 – Sim.UCI. PSNR and SSIM quality comparison for all the simulations in Use Case 1.



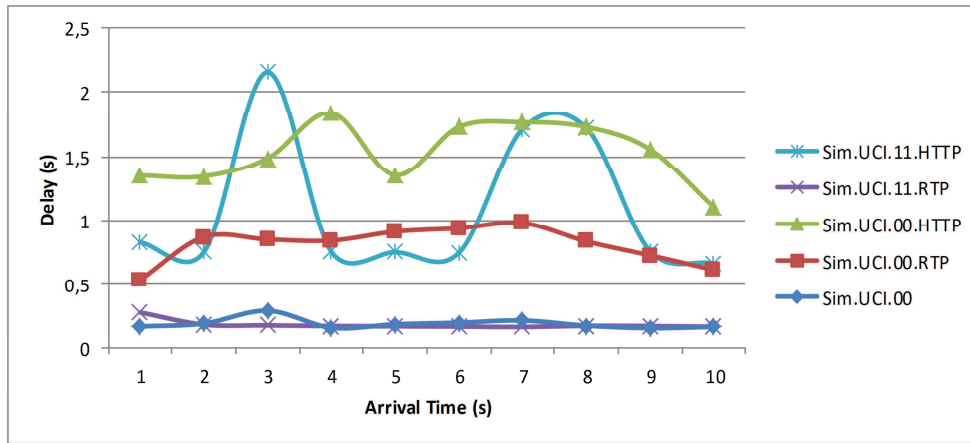


Figure 13 – Sim.UCI. Transmission delay comparison for all the simulations in Use Case 1.

**HTTP/TCP Performance**

Figure 14 and Figure 15 illustrate the TCP receiving rate and HTTP buffer status curves of the HTTP-based simulations taking place from the Coordination Center towards Hospital 2. Due to channel errors, TCP retransmissions and retransmission timeouts cause transmission delays for TCP. As a result, 1 second buffer is needed for smooth, continuous playback for the reference video and 2 second buffer for the adaptive CONCERTO technique. Thus, a higher video bitrate and quality for the latter case explains the need for longer playback buffer. As an outcome from HTTP simulations, we conclude that smaller HTTP delay is achieved for the lower quality video, but higher quality video can be obtained with slightly larger playback buffer. Naturally, the TCP characteristics already provide only near real-time video transmission, which means its use is designed especially for afterward playback from a video content server where a larger video buffer can be utilised.

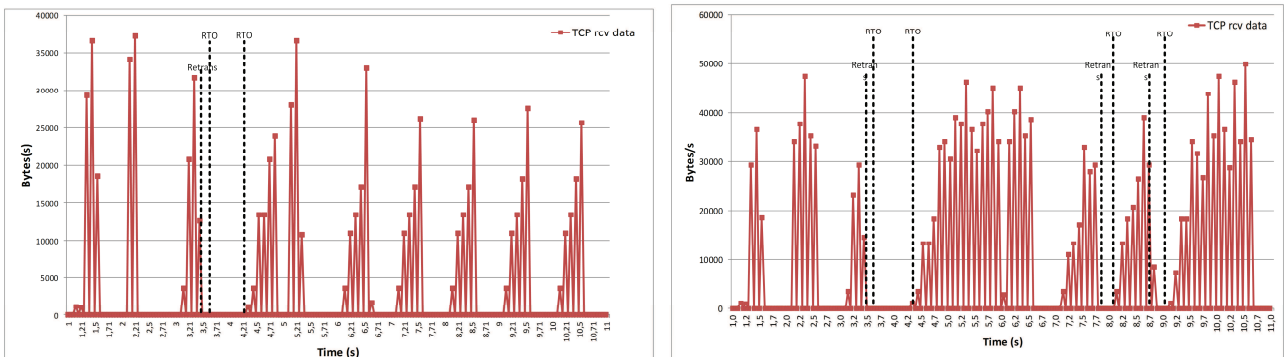


Figure 14 – Sim.UCI. TCP receiving rate, retransmissions and retransmission timeouts for Sim.UCI.00.HTTP (left) and Sim.UCI.11.HTTP (right).