5G key technology enableRs for Emerging media COntent pRoDuction services

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Final trials and technology validation

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Abstract

Deliverable D5.3 describes in detail the final tests and trials of the 5G-RECORDS project after the first phase of development, integration and preliminary tests. During the second phase of the project, the trials focused on the validation of the 5G-RECORDS components and end-to-end (E2E) solutions in the context of the three use cases: live audio production, multiple cameras wireless studio and live immersive content production. The tests and trials were performed under real conditions to assess to what degree 5G fulfils the specific use-case KPIs and technical requirements. Several metrics and results have been collected to study to which degree 5G fulfils the technical KPIs and requirements of the project use cases in the context of professional content production.

[1] CO = Confidential, only members of the consortium (including the Commission Services)

PU = Public

Keywords

Final trials, Measurements, Testbeds, Results, KPIs analysis, Technology validation

Executive Summary

The document describes the final tests and trials performed to validate the 5G-RECORDS components and end-to-end (E2E) solutions demonstrated in the context of the three use cases: live audio production, multiple camera wireless studio and live immersive media production. 5G-RECORDS final trials have been designed to experimentally assess and validate the technical performance of 5G and media devices under real conditions and to ascertain to what degree 5G fulfils the technical performance requirements. After a first phase of development, integration and preliminary tests, the 5G-RECORDS use cases were finally tested in Sophia Antipolis, Aachen, Copenhagen and Madrid. Some of the measurements initially planned had to be adapted to the testbed due to challenges encountered during the second phase of the project.

The Live Audio Production use case (UC1) successfully demonstrated the integration of live audio production on network layer into multiple 5G testbeds. Efforts concentrated on extensive optimizations to reduce the latency in a state-of-the-art 5G system, namely, in the 5G-RECORDS UC1 5G disaggregated infrastructure. This use case aimed to study the performance of the E2E system when using a single UE, to measure deterministic audio streams through multiple 5G modems, and to conduct mobility tests to better understand the use-case KPIs in a more realistic environment. Also, UC1 team was able to collaborate with UC2 partners during the trial in Tivoli Garden to demonstrate the delivery of audio and video over the same 5G network, as well as to conduct latency measurement as part of the evaluation of the state-of-the-art 5G components.

The Multiple Camera Wireless Studio use case (UC2) provided satisfactory results of the trials and gave valuable information of the 5G networks and components uptake for media content production. The trials KPI analysis showed that content production, either being a local studio or remote scenario, can in fact be performed successfully, taking advantage of the 5G performance and features. The current glass-to-glass latency value of about 200 ms for the local studio and 2 seconds for the remote contribution is very close to the proposed limits. UC2 was able to test the 5G modem integrated with the encoder, which attached to the back, making all-in-one capturing, encoding and transmission worked as expected. Additionally, PTP offset values were measured to test frame-level synchronization, which is critical for the content production workflow. The network slicing functionality was also successfully tested, assuring a specified bandwidth versus non-prioritized users. Finally, both multi-cam, SMPTE 2110 video functionalities, remote audio communication and remote camera control were successfully validated, resulting in a good overall user experience.

The Live Immersive Production use case (UC3) final trial provided relevant information as result of all the work carried out in the project. It demonstrated the viability of a full end-to-end FVV Live deployment to stream and record an event over a 5G network. The trial was chiefly intended to bring the use case into a real environment. The event consisted in a live music performance by professional artists which was produced as a FVV service in real time and streamed to the final user. The FVV content was also recorded to demonstrate the FVV playback functionality of the system. The final trial allowed to validate every module, video tools and component developed in the project, and to track key metrics in real-time generating logs that could be monitored in real-time. Furthermore, Grafana dashboards were shown and monitored during the whole session. Two traffic slices were configured (Multimedia Gold and Best Effort) in two different network segments (5G RAN and Transport Network), covering delivery of the produced video from the Media Delivery VNF in the Delivery Edge Cloud to the End User in the trial site.

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List of Acronyms

Acronym	Term
5G	Fifth Generation
5GC	5G Core
5GS	5G System
ACR	Absolute Category Rating
AV	Audiovisual
BBU	Baseband Unit
BE	Best Effort (slice)
BW	Bandwidth
CDF	Cumulative Distribution Function
COTS	Commercial-off-the-shelf
CPE	Customer Premise Equipment
CRA	Continuous Random Access
CU	Central Unit
DL	Downlink
DoF	Depth of Field
DU	Distributed Unit
E2E	End-to-end
eMBB	Enhanced Mobile Broadband
FN	Free Navigation
FOWR	Few Observers With Repetitions
FPS	Frames Per Second
FR	Frequency Range
FTTP	Fiber-To-The-Home
FVV	Free-Viewpoint Video
G2G	Glass-to-Glass
GDR	Gradual Decoder Refresh
gNB	Next Generation Node B
GOP	Group of Pictures
GP	Guaranteed Performance
GTP	GPRS Tunnelling Protocol
HEVC	High Efficiency Video Coding
HLS	HTTP Live Streaming
HRT	Hypothetical Rendering Trajectories
HTTP	Hypertext Transfer Protocol
HW	Hardware
IDR	Instantaneous Decoder Refresh
IEM	In-ear monitoring
IP	Internet Protocol
IQR	Inter Quantile Range
KPI	Key Performance Indicator
LAP	Local Audio Processing
LTE	Long Term Evolution
MCR	Master Control Room
MCS	Modulation and Coding Scheme
MEC	Mobile Edge Computing
MG	Media Gateway
MOCG	Media Orchestration Control Gateway
mmW	Millimeter wave
MOC	Mean Opinion Score
MOCN	Multi-Operator Core Networks
MPR	Media Proxy

Acronym	Term
MPL	Media Player
MTP	Motion-to-Photon
NR	New Radio
NPN	Non-Public Network
NTP	Network Time Protocol
NSA	Non-Standalone
OAI	Open Air Interface
PCAP	Packet Capture
PDCP	Packet Data Convergence Protocol
PDU	Protocol Data Unit
PER	Packet Error Rate
PPS	Pulse-per-second
PTP	Precession Time Protocol
PVS	Processed Video Sequences
QoE	Quality of Experience
QoS	Quality-of-Service
QP	Quantization Parameters
RAN	Radio Access Network
RTP	Real-time Transport Packet
RTSP	Real-time Streaming Protocol
RTT	Round-Trip Time
RU	Radio Unit
SA	Stand-alone
SCS	Subcarrier Spacing
SDI	Serial Digital Interface
SDN	Software-Defined Networking
SDR	Software-defined Radio
SMPTE	Society of Motion Picture and Television Engineers
SNPN	Stand-alone Non-Public Network
SoC	System-on-Chip
SW	Software
TCP	Transmission Control Protocol
TDD	Time-Division Duplexing
TSC	Time Sensitive Communications
TS-DF	Time Stamped Delay Factor
TSN	Time Sensitive Networking
UC	Use Case
UE	User Equipment
UHF	Ultra-High Frequency
UL	Uplink
UPF	User Plane Function
URLLC	Ultra-Reliable Low Latency Communications
VNF	Virtual Network Function
VR	View Renderer
VLAN	Virtual Local Area Network

1 Introduction

5G-RECORDS is about the use of 5G components for professional audio-visual (AV) content production. This project aims to explore the opportunities that new 5G technology bring to the content production sector using primarily the so-called non-public networks (NPNs). To this end, the project has built on completely new 5G components and, also, enhanced devices developed within previous 5G-PPP projects, to deploy three end-to-end 5G testbeds across three use cases: live audio production (UC1), multiple camera wireless studio (UC2) and live immersive media production (UC3). These testbeds include 5G core network (5GC) and radio access network (RAN) technologies, as well as media orchestration and end devices.

After the development and integration of those components into the specific testbeds, final 5G-RECORDS efforts concentrate on the trialing, demonstration, and validation of the technology. The final trials of the three use cases were performed in specific AV content production scenarios where stringent technical requirements are in place, e.g., where content acquisition devices such as cameras and microphones are connected to a 5G network to convey live content.

The execution of the final trials consisted in the evaluation of the complete constitutive chain of the use cases, the appropriate configuration and optimization of the 5G infrastructure when the components are integrated together. Both technical laboratory tests and field trials in unique venues have been conducted to assess and showcase the performance of the 5G-RECORDS components. Key Performance Indicators (KPIs) and requirements such as reliable and sustainable throughput, latency and synchronization have been evaluated. Dedicated production and 5G equipment have been fine-tuned to understand the ability of the systems to fulfill the media production requirements.

1.1 Scope

This document concentrates on the final trials and technology validation of the 5G-RECORDS components and end-to-end (E2E) solutions demonstrated in the context of the three use cases. While deliverable D5.2 [1] focused on the initial stage of the trials, this document targets the outcomes of the final stage. It includes not only lab tests, but also field measurements (noted as UCX.B.Y, where "X" refers to the use case number and "Y" to the trial number).

The final trials have been designed to experimentally assess and validate the technical performance of 5G and media devices under real conditions. They also aim to ascertain to what degree 5G fulfils the technical performance requirements of the three use cases in the context of professional content production. For that, project partners have selected unique venues to deploy their trials, i.e., Sophia Antipolis for UC1, Aachen and Copenhagen (Tivoli Gardens) for UC2, and Madrid for UC3.

The specific objectives of the final stage of 5G-RECORDS trials are the following:

1. E2E trials execution, using the 5G components.
2. Demonstration and application of the 5G technologies involving those applicable to RAN, Core and the newly developed components in the content production sector to support the requirements and functionalities of each use case.
3. Testing and validating the 5G infrastructure capabilities against different configuration and deployment options.
4. Collection, visualization and detailed analysis of the measurement results obtained from the trials.
5. Validation of the 5G technologies and infrastructures against the targeted content production technical KPIs, including the identification of potential bottlenecks and providing, if needed, specific recommendations and proposals to standardisation bodies.

1.2 Objectives

- To present the outputs and major achievements of the final trials
- To report the measurement data obtained from the final trials
- To carry out the analysis of the measurements and present a detailed description of the results
- To assess the technical performance of deployed 5G infrastructures against the UC-specific KPIs, thus, to validate the operation and fulfillment of content production requirements.
- To validate the technology in the context of the considered professional content production scenarios.
- To provide insights and draw a conclusion.

1.3 Structure

The document is organized across the three use cases as follows:

- Section 2 describes the outcomes of the final trials related to the live audio production use case (UC1).
- Section 3 describes the outcomes of the trials related to the multiple camera wireless studio use case (UC2).
- Section 4 describes the outcomes of the trials related to the live immersive media production use case (UC3).
- Section 5 present the conclusions of the 5G-RECORDS final trials and technology validation.

Each section related to the three Use Cases (Section 2-4) presents *(i)* an update of the 5G testbed infrastructure including the risks assessment, *(ii)* the measurement results of the final tests and trials, *(iii)* the analysis of the use case KPIs, and *(iv)* the outcomes from the technology validation.

2 Use case 1: Live Audio Production

This chapter describes the outcomes of the final stage of trials in the context of use case 1 (UC1). The latest updates on the testbed architecture will be presented alongside new measurements results and a detailed KPI analysis. Also, the technology validation of this use case will be addressed.

2.1 Deployed testbed architecture

Figure 1 depicts the final architecture deployed at EURECOM site for UC1 5G disaggregated testbed, which is based on the architecture presented analytically in D3.1 [2].

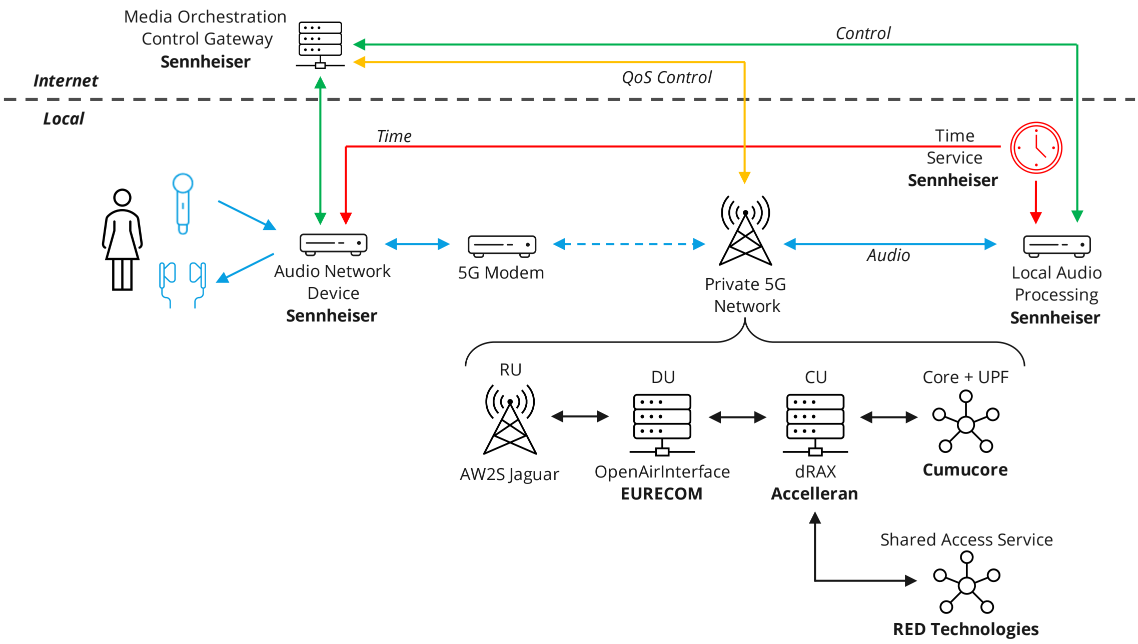


Figure 1: Final UC1 deployed testbed architecture

Compared to the previously reported initial testbed deployment (see D5.2 [1]) the final setup

- incorporates the disaggregated RAN components from Accelleran and Eurecom
- uses the Cumucore core network implementation
- integrates the Shared Access Client (part of dRAX) from Accelleran
- integrates the Shared Access Service from RED Technologies
- includes the interface for remote Quality-of-Service configuration

Figure 2 shows the radio unit, the modem, audio network device and local audio processing unit.

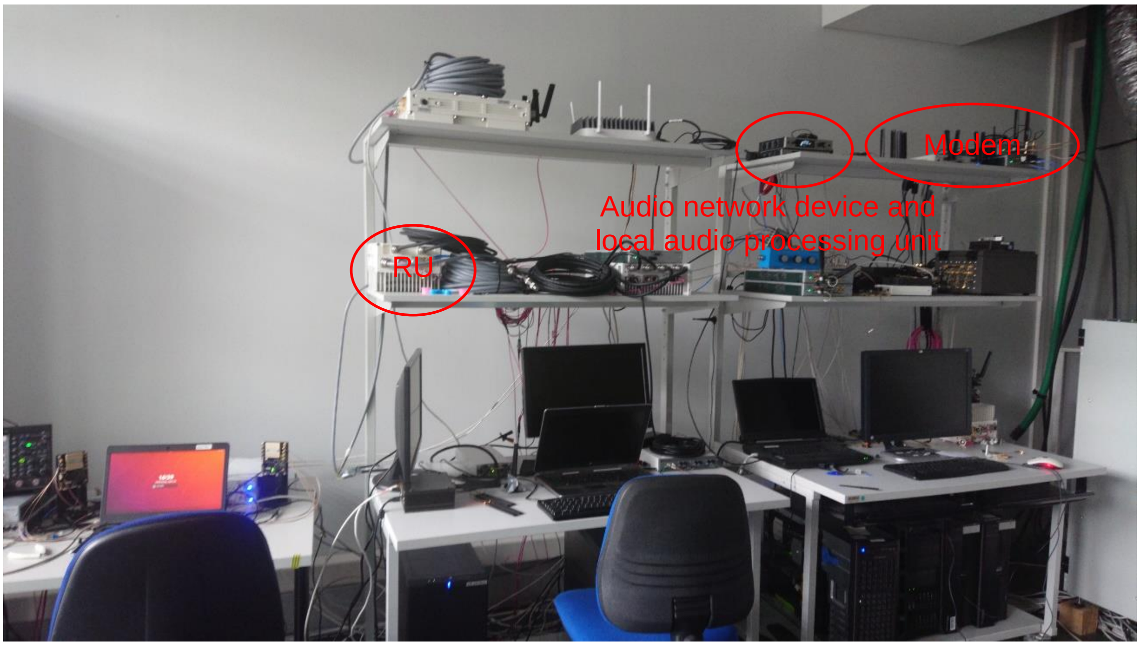


Figure 2: Lab room at Eurecom with radio unit, modem, audio network device and local audio processing unit

In Figure 3 a setup of multiple audio network devices is shown.

Figure 3: Lab setup with multiple audio network devices

2.1.1 Updates on measurements planning and network architecture

The following section describes updates on the measurement planning. Originally described in D5.2 [1].

Trial UC1.A.4 – Comparison monolithic OAI vs. disaggregated

Extensive latency optimizations with further reduced DL/UL pattern periodicity and increased sub-carrier spacing. Comparison of the monolithic OAI setup and the disaggregated 5GS including all partner components.

Update: Contrary to the original planning the team did not reduce the slot duration below 0.5 ms since other elements not related to the slot timing turned out to be the current bottleneck for latency (see measurement results in section below).

Trial UC1.B.1 and Trial UC1.B.2 – Up to 8 audio UEs

Validation of a concept to deliver low-latency audio to more than one audio UE.

Update: Due to limited modem availability up to three simultaneous audio UEs were tested.

Trial UC1.B.3 – Time synchronization over 5G

Comparison of delivery of synchronized time information via dedicated Ethernet cable, over-the-top and with TSN-based mechanisms.

Update: Due to unavailability of 5G components with supporting features and/or interfaces for time synchronization or TSN up until the end of the project it was not possible to evaluate distribution of time information to the extend initially planned.

It is also worth noting that the project has developed an API that can be used to set QoS parameters from outside of the 5G SA network. This API has been tested, but not used in the testbed because testing time was used to fine tune end-to-end performance.

2.1.2 Uncertainties and risk assessment

The availability and maturity of available 5G components remained a major uncertainty up until the final activities in UC1. Here, the COTS 5G modem was the most important external dependency. It is of central significance in the architecture of UC1 and has major influence on the support for specific features and achievable KPIs.

2.2 Measurements results

During the trials all UC1 partners put extensive effort into the optimization of each component to ultimately reduce the end-to-end latency. This section gives insight in some exemplary optimization steps that took place over the course of the project.

2.2.1 Deployment of UPF

During the trials it became obvious that some components introduced significant latency jitter into processing and forwarding of audio IP packets in the 5G system. *Figure 4* shows an exemplary measurement of application downlink latency, latency histogram and CDF. Jitter is significantly increasing over the course of the measurement time.

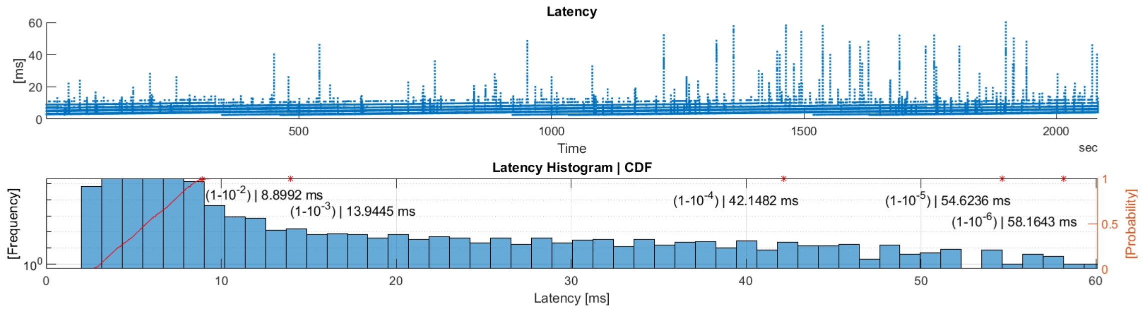


Figure 4: Application downlink Latency - non-real-time UPF deployment

By taking Wireshark captures at different steps in the processing chain it was possible to identify individual components as cause for jitter. Figure 5 shows exemplary the inter-packet time of audio packets after leaving the 5G UPF. At this stage of trials, the 5G UPF was deployed on a dedicated machine using a general-purpose operating system kernel.

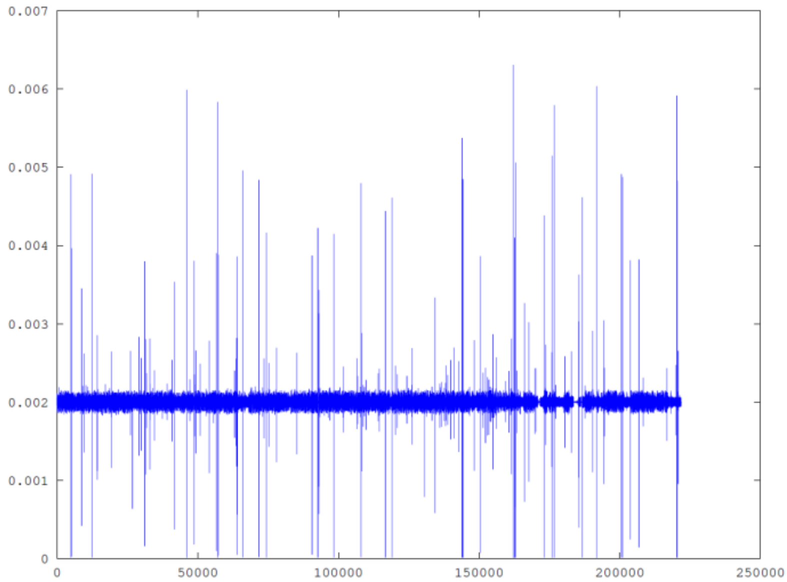


Figure 5: Inter-packet time after non-real-time UPF processing

By using a real-time kernel on that machine, the jitter introduced by the UPF could be significantly reduced as shown in Figure 6.

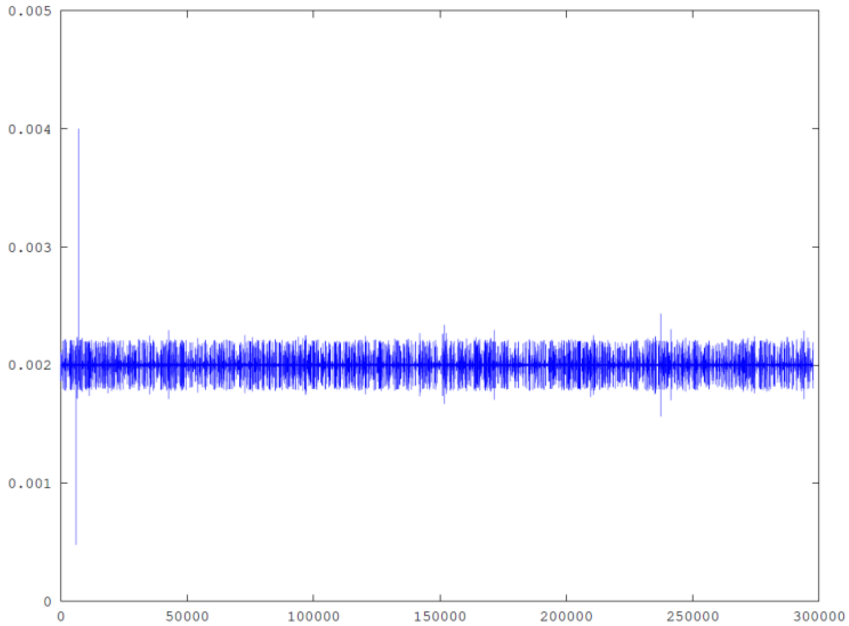


Figure 6: Inter-packet time after real-time UPF processing

2.2.2 Parameterization of CU

With the continued trials a specific behavior of the 5G CU became apparent. In the initial CU configuration setting that touches packet reordering led to the behavior that a lost packet caused to a high jitter event of about 100 ms for the following packets. This behavior was particularly clear in the mobility case due to the optimizations performed at the MAC layer of the DU in the absence of HARQ retransmissions. An exemplary measurement showing this behavior can be seen in *Figure 7*. The graphs show the latency, packet-loss, latency histogram and CDF of a measurement with many lost packets.

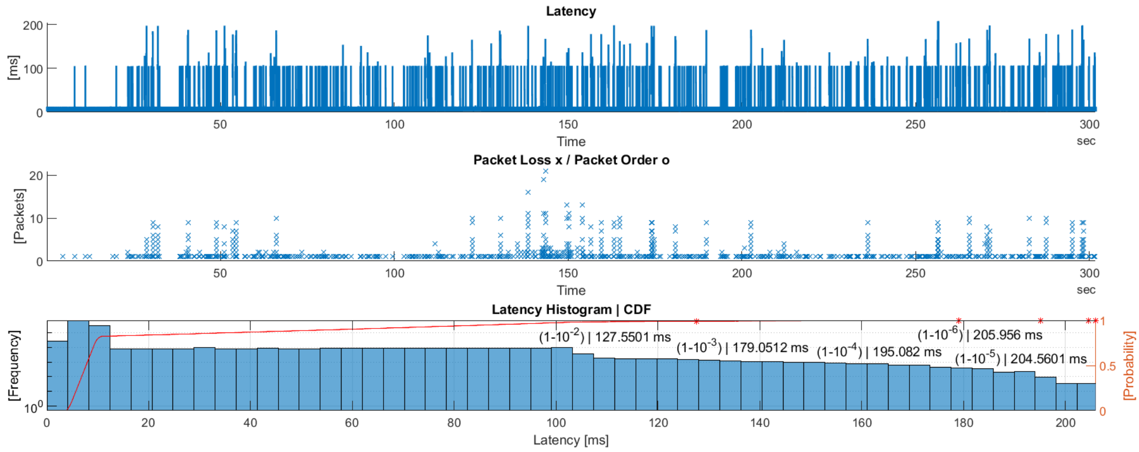


Figure 7: Application downlink latency with packet-loss

It was found that by configuring the PDCP reordering timer of the CU from the default value of 100 ms to 0 ms the jitter events could be removed. The default parameter caused that a lost packet would be tried to be fixed in case it was a reordering issue for about 100 ms. This delayed the following packets. Since we do not expect any packet reordering in this testbed it is a valid approach to reduce this timeout.

2.2.3 Latency with single audio UE

As mentioned before, extensive effort was put into the development of 5G components during 5G-RECORDS to optimize the E2E data-path for latency. Generally, very limited optimizations were possible with the 5G modem as it is a highly integrated off-the-shelf component. Over the course of the project, we used 5G modems (SIMcom 8200, Quectel RM510Q, Quectel RM520N) from different vendors with very similar results. This section presents the achieved latency with the 5G testbed optimized for a single audio UE.

Figure 8 presents the uplink latency from microphone audio network device to local audio processing. The measurement shows a test where audio was sent through the 5GS with a pace of 1 ms between IP packets. The 5GS operates with specific internal timing grids and periodicities for processing and over-the-air transmissions. The mismatch between audio and 5GS periodicities results in buffering of results in distinct latency lines as shown in the figure. The measurement was conducted in a static scenario where the modem was placed near the RU to ensure reliable transmissions without packet loss. The CDF-graph gives the latency for 99,9999 % of all packets at about 12 ms.

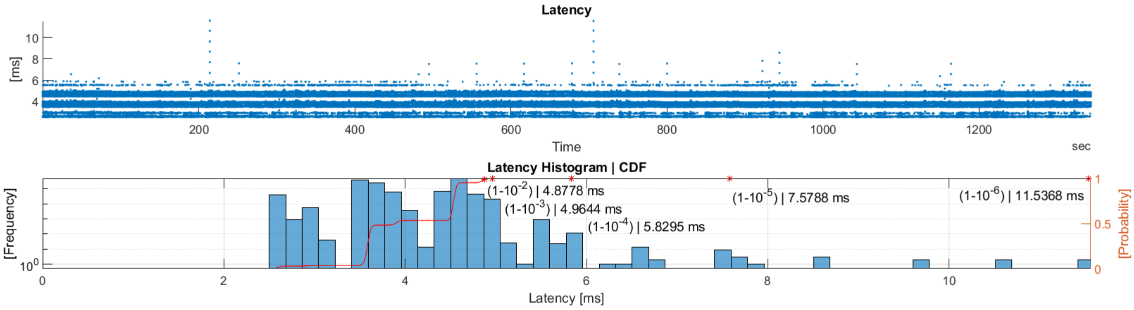


Figure 8 : Application uplink latency

Figure 9 shows the achieved downlink latency from local audio processing to IEM audio network device with a single UE optimization. Here, the one-way latency for 99,9999 % of all packets is at about 11 ms. From the top latency graph some distinct behaviors can be seen. First, there are more parallel lines present compared to uplink. And second, the lines are drifting. From several tests we know that this is related to the USB connection between modem SoC and embedded PC, which form the 5G modem. We found that the USB connection of the modem SoCs operates asynchronously with a specific periodicity of 6 ms, which overlays the 5GS timing behavior. We found no way to change the behavior with the available modem SoCs and the inherent asynchronous 6 ms latency, although the SoCs were labeled as URLLC capable.

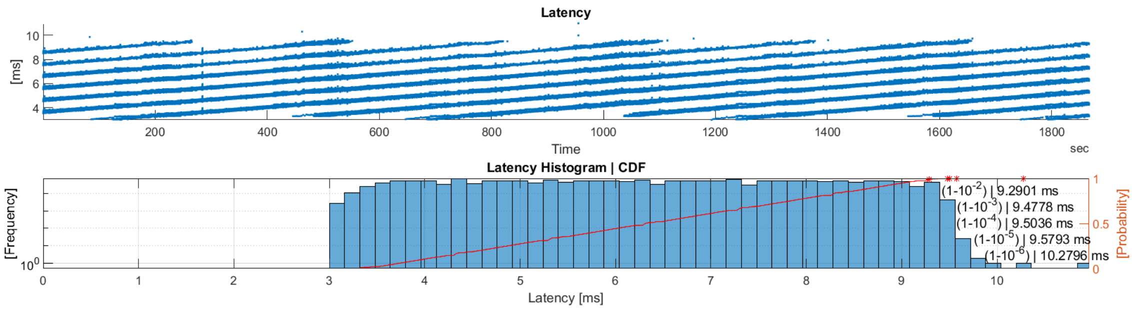


Figure 9 : Application downlink latency

The one-way latency results for uplink and downlink are similar in number for 99.9999 % of all packets. Looking at another operation point it becomes clear that the behavior of the two directions is different. Looking for example at 99.99 % of all packets the uplink latency is at about 6 ms, while the downlink latency is still at about 10 ms. The latency in uplink is determined by 5GS timing and jitter, while the downlink latency is determined by asynchronous processing and the USB connection.

2.2.4 Latency with support for multiple audio UEs

To approach the evaluation of the potential suitability of using a 5GS to deliver wireless audio for more than one UE, the partners of UC1 optimized the testbed with that aspect in focus. New trade-offs between latency and resource distribution were necessary. Nonetheless, through extensive effort it became possible for the first time to measure te

In Figure 10 an exemplary measurement from one microphone network audio interface to the local audio processing system is shown. The measurement shows the transmission of 30 minutes of continuous audio over the 5GS. For the first 10 minutes only a single 5G UE was transmitted, then an audio stream over a second UE was added. After 20 minutes a third stream was added.

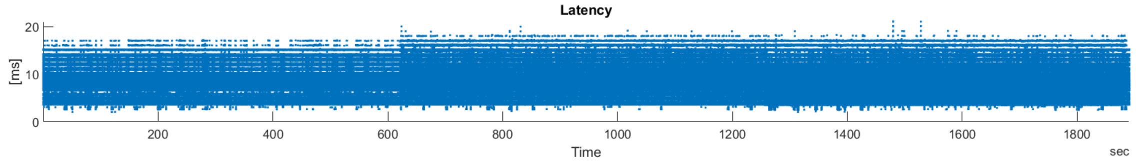


Figure 10: One-way uplink latency with multiple audio UEs in parallel

At the points in time when the additional audio streams are added, the jitter through the 5GS increases. Still, the latency for the large majority of packets through the systems remains the same at about 20 ms independent if one, two or three UEs. Compared to the implementation of the testbed optimized for a single UE, the latency is about double (see Figure 8).

2.2.5 Mobility test

In order to better understand KPI behavior in a more realistic use case environment some mobility tests were conducted. For that one UE was connected to the 5G network and moved around in the lab premises. Although the physical environment of the lab is not the same as on a live music stage some insight can still be gained from such tests on latency and packet error ratio in a more dynamic setup.

Figure 11 shows an exemplary setup of such a trial. The functions of the local 5G network are distributed in different rooms in the facilities. While the DU, CU, Core & UPF functions are installed at the server room, the RU is installed in the lab room. The light blue area indicates the approximate directionality of RU antenna. The LAP as well as the Audio Network Device that is part of the audio UE is installed in the lab room. The 5G modem that is part of the audio UE is moved on a trolley (see Figure 12) along the corridor as shown by the dotted arrow in Figure 11. The modem and the audio network device were connected with a LAN cable.

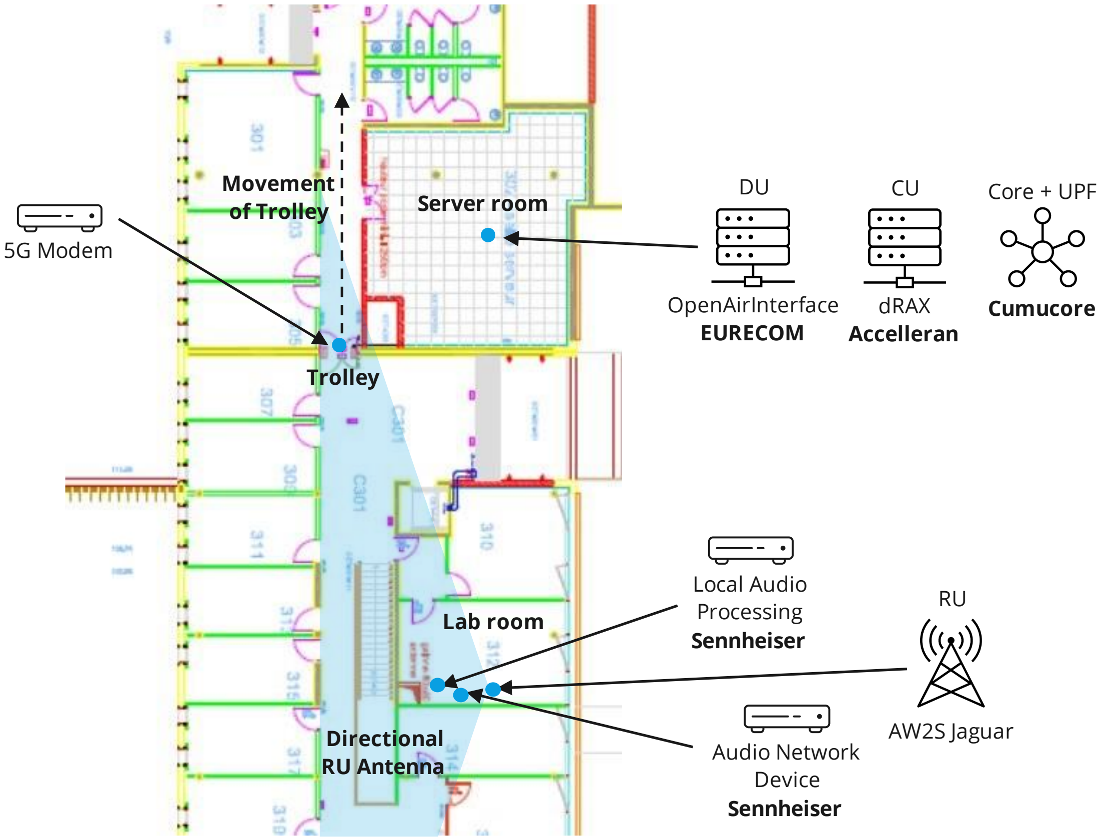


Figure 11 : Mobility trial setup

The 5G network is configured to use a static Modulation and Coding Scheme (MCS), (MCS 16). The RU is set for a TX power of 250 mW. The network operates with 20 MHz bandwidth in the 3.5 GHz band.

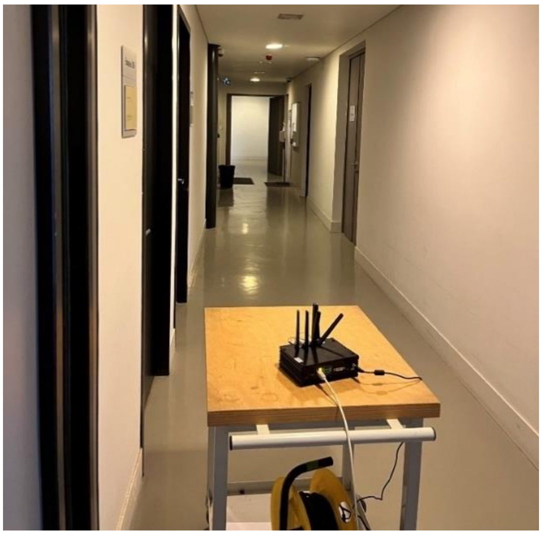


Figure 12 : Mobile audio UE modem on a trolley; corridor for mobility trial

For one of the mobility tests the modem was moved on a trolley along the corridor away from the RU (see Figure 11). For about 45 seconds the trolley was moved about 30 meters along the corridor as marked by the arrow. After that the trolley was moved back at about the same speed.

Figure 13 shows the results of this test. The top graph shows the latency of all downlink audio packets. As expected, the latency is stable over time since no retransmission mechanism was active. The second graph shows in blue the number of consecutive lost packets and in orange the buffer level of the audio playback buffer. It can be seen that the number of lost packets remains zero for the initial section of the path. At one point lost packets start to appear in increasing bursts while moving along the corridor. The effect is also visible in the third graph that shows the number of received packets per second. When no packets are lost, it should be 1000 (since 1000 packets were being sent per second). Around second 67 almost 100% of the packets are lost for a short period of time.

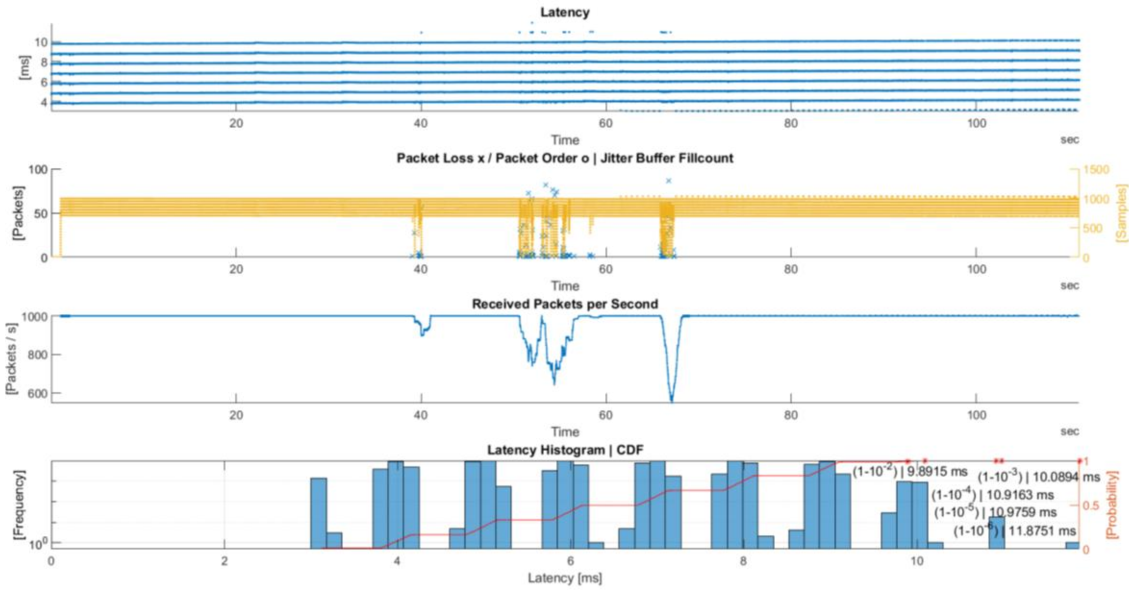


Figure 13: Downlink latency and packet loss during mobility trial

Depending on individual packet loss pattern, audio content and mitigation techniques observed packet loss would most likely have led to significant audible dropouts in the application.

2.3 KPI analysis

This section documents to which extent the use case KPIs were achieved during 5G-RECORDS, according to the trials and measurements. The use case KPIs are listed and explained in D2.1 [3].

2.3.1 Network latency

The network latency requirement describes the latency of a continuous stream of audio data packets from application layer on UE side to application layer on LAP for UL direction and the other way around for DL direction. The use case requires a network latency of below 1 ms one way.

To evaluate this KPI a measurement method is established that allows to measure the absolute latency of every network packet sent between two participating audio devices (UE/LAP). This method is described in detail in D4.1 [4].

Initial measurements in the testbed showed network latencies of above 100 ms. During multiple iterations we were able to reduce the one-way network latency in the disaggregated 5G testbed to about 10 ms, in downlink and uplink direction (see above).

2.3.2 Synchronicity

To ensure a low latency and high-quality audio transmission all audio devices need to be synchronized to a common clock with a maximum deviation of 500 ns.

In the wired part of the production network this can be achieved by using PTP over wired IP networks. For wireless devices it is required to synchronize with a similar accurate clock like the wired. A method to measure this KPI based on PPS signals of all components is developed and described in D4.1 [4].

Due to the fact that no 5G components with PTP support were available until the end of the project we were not able to perform measurements that would show the needed performance. Some short tests with available Release 15 equipment revealed that it was not possible for professional audio devices to lock timing over the 5GS, mainly due to high latencies, jitter and path asymmetries. As a comparison the PTP trials of UC2 could be taken into account. In [5] members of UC2 showed that they could achieve a timing accuracy of 3.5us within an mmWave 5G testbed that included features from release 16 and 17. Even though this was also a measurement in a private 5G network, it is unclear to what extent these results can be transferred to a non-mmWave system as required for UC1. As of today, it was not possible to achieve sufficient time synchronization over a state-of-the-art 5GS in UC1.

2.3.3 Packet error ratio

The use case requires a packet error ratio < 10^{-6} to ensure a high-quality audio production. Furthermore, the distribution of the errors plays a role when it comes to the effect on audio quality (compare D2.1 [3]).

In the initial trials a systematic error caused by a concurrency of TX preparation and RX made it impossible to evaluate a meaningful PER. This issue was solved for the final trials. Short measurements with stationary devices in a controlled lab environment showed that the system can work without any packet errors at this point in time. Some small mobility trials were conducted to touch on the topic.

At this point in time, it remains difficult to conclude about PER of a current 5GS in live stage environment. Packet error ratio is a parameter that is tightly bundled to latency. Both can typically be traded-off in different ways e.g., retransmissions. Since it was not possible to achieve the latency needed for this demanding use case, an operation point could not be fixed to make meaningful measurements for packet error ratio.

2.4 Technology validation outcome

This section gives a summary about the technology validation and remaining open questions for use case 1.

Technology validation

- Network integration of application successful into multiple private 5G networks.
- Disaggregated setup is on-par with monolithic implementation.
- 5G ecosystem is not mature for low latency use cases such as live audio production scenarios.
- Extensive effort was needed to optimize available 5G components for latency.
- Some URLLC-labeled off-the-shelf 5G components did not provide low-latency capable interfaces.
- Against previous assumptions the finally achieved latency is not only determined by the 5G radio timing grid (e.g., slot-length), but also significantly defined by implementation of interfaces and processing functions, and types of deployments.

- One-way latency at about 10 ms for a single audio UE in a private 5G network (microphone or IEM).
- One-way latency at about 20 ms with up to three audio UEs in a private 5G network (microphone or IEM).

Remaining gaps / open questions

- **Latency**: the one-way 1 ms latency requirement for live professional audio scenarios was not met with a state-of-the-art 5G system
- **Reliability and Efficiency**: since the latency requirement was not met yet, it remains an open question in what way the trade-off between latency, reliability and (spectral) efficiency can result in valid operation points in the use case context-
- **Synchronicity**: state-of-the-art 5G components do not yet support sufficient time synchronization or provide corresponding interfaces on application level-

To the final date of the 5G-RECORDS project it can be stated that significant effort is still needed to finally achieve the full set of requirements for live audio production scenarios. Low latency must be considered end-to-end. All components and interfaces on all layers in the full signal path need to be designed with low latency paradigms. This remains especially challenging in complex wireless connectivity systems with many individual components and standardized interfaces.

Nonetheless, 5G-RECORDS has shown that it was possible to integrate live audio production on network layer into multiple 5G testbeds and that the latency in a state-of-the-art 5G system can be reduced significantly with extensive optimizations.

2.5 Joint UC1+UC2 trial: Audio for local 5G TV production

The fact that different media use-cases and related partners are represented within the 5G-RECORDS consortium gave room to extended inter-use case exchange. One outcome of this was the opportunity for UC1 partners and components to be part of a UC2 trial to explore the possibility of using 5G in a local TV production to also deliver wireless audio.

The local TV production scenario focusses on the production of media content for live distribution. In the “Tivoli Garden” trial the goal was to produce a 15-minute live interview about the 5G-RECORDS project. The envisioned scene incorporated a moderator and multiple interviewees that answered questions about the project one after the other. To give the moderator and the interviewees sufficient flexibility to be able to walk through an area with multiple technical demonstrations the production required wireless microphones and wireless cameras to capture the scene, and a wireless in-ear-monitoring device for the moderator to be able to receive instructions or information from a director.

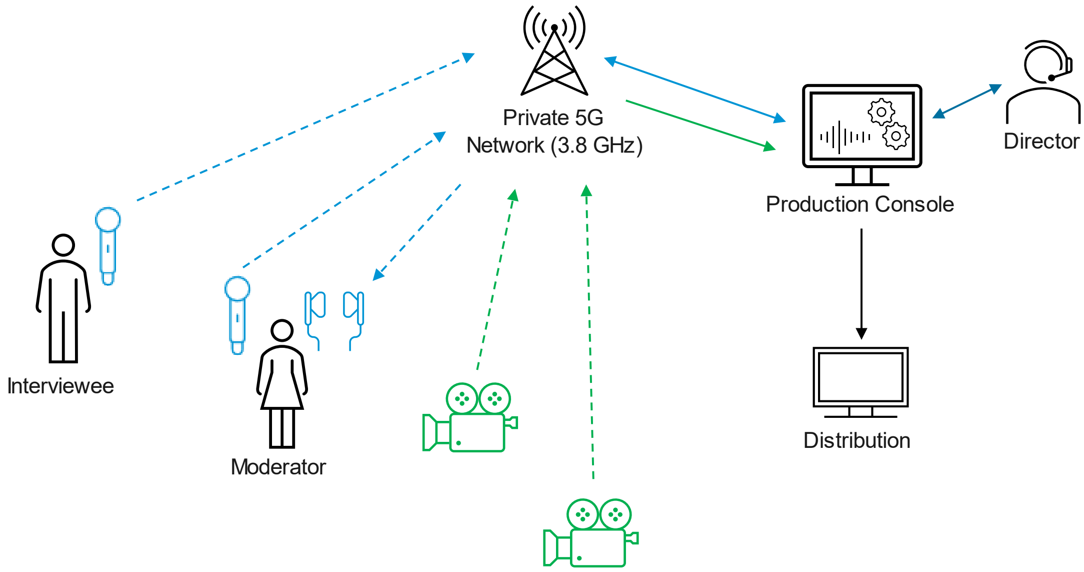


Figure 14: Overview local 5G TV production trial

Such a scenario could potentially benefit from using a private 5G network to connect media wireless media devices to the production console for processing and further distribution (see Figure 14). Possible advantages could be a more efficient workflow when only a single system-based wireless technology is deployed to connect all wireless applications in the TV production.

Compared to the live audio production scenario a local TV production poses different requirements on the 5G system. While the requirement for reliability is similar, the latency requirement for delivery of wireless audio is less strict. In the live audio production use case lowest latency is mandatory in the closed audio loop between microphone, mixing console and in-ear-monitoring device to provide sufficient self-feedback to an artist. In contrast the local TV production scenario in this trial does not incorporate such a self-feedback loop. Consequently, the latency requirement for wireless audio in this trial was only determined by the latency of video delivery. Since audio and video need to be synchronized for distribution, respective latencies must be aligned. This is typically done by adding artificial delay to the slower of both signal paths. To avoid the more complex addition of artificial delay to video the audio latency budget is limited by the video latency.

Figure 15 gives a detailed view of the architecture deployed in the local TV production trial with a focus on the audio components. Only some video components are shown where necessary to understand the overall signal path of audio.

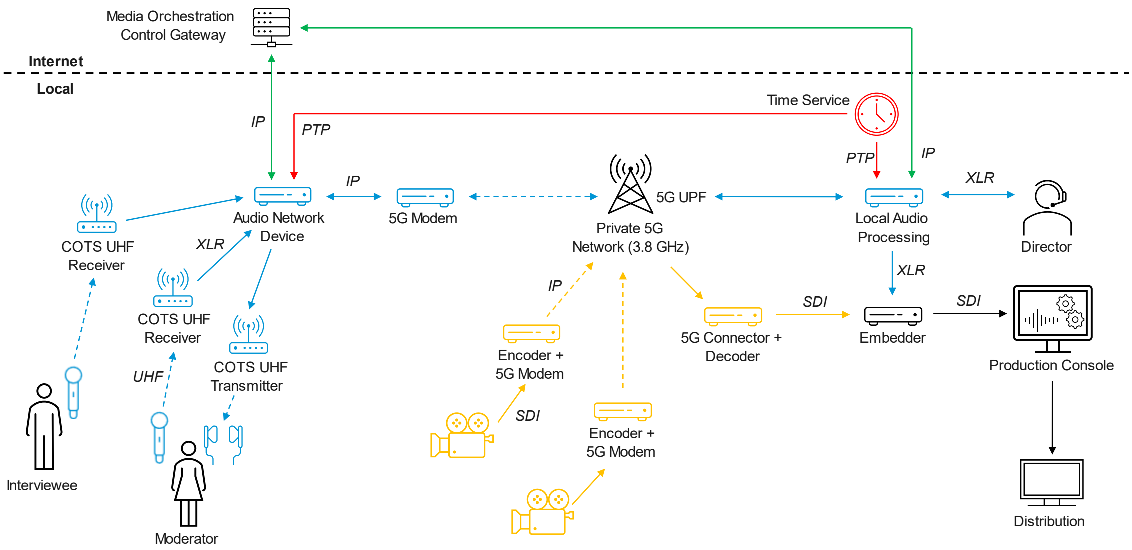


Figure 15: Detailed architecture audio in local 5G TV production trial

Central component of the trial was the private 5G network that connected the local media capture devices (microphones, cameras) and playback devices (IEM) to the production console for live distribution. Several UC1 components were integrated into the UC2 trial architecture to deliver end-to-end audio (see Table 1). Since the miniaturization and portability of components was not focus of 5G RECORDS COTS UHF-based wireless microphones and IEM devices were daisy-chained with the 5G system to give the moderator and interviewees a sufficient degree of flexibility.

Table 1: UC1 components integrated into UC2 trial

UC1 component	
Audio Network Device	Interface device between COTS UHF-based wireless microphones / IEM and 5G modem. The 5G modem was provided by UC2 partners.
Local Audio Processing	Mixing and interfacing between 5GS UPF and director and production console.
Time Service	Synchronization between audio network device and local audio processing. Delivered with a wired Ethernet connection. Basis for latency measurements.
Media Orchestration Control Gateway	Allowed off-premises remote control of audio network device and local audio processing.

Focus of the local 5G TV production trial was the demonstration of delivering audio and video over the same 5G network, as well as conducting latency measurement as part of the evaluation of the state-of-the-art 5G components. Latency was measured between audio network device and local audio processing based on the wired time synchronization. UC1 components allowed the continuous measurement of latency during preparation and live production. The audio transmission was running for more than 5 hours, while the live production took less than 30 minutes.

As introduced above, focus of the local 5G TV production trial was the demonstration of delivering audio and video over the same 5G network, as well as conducting latency measurement as part of the evaluation of the state-of-the-art 5G components. Latency was measured between audio network device and local audio processing based on the wired time synchronization. UC1 components allowed the continuous measurement of latency during preparation and live production. The audio transmission was running for more than 5 hours, while the live production took less than 30 minutes.

Figure 16 shows the one-way 5G uplink latency of every audio IP-packet measured from audio network device to local audio processing over the trial day. The 5GS did not provide low-latency quality-of-service to the audio or video application. Most packets are faster than 75 ms, some higher latency spikes occurred at irregular times. The cause of the latency spikes in the 5GS is unknown. Traffic was generally handled with a best-effort approach and the 5G testbed was not optimized for media applications. By chance no significant latency spike occurred during the live production.

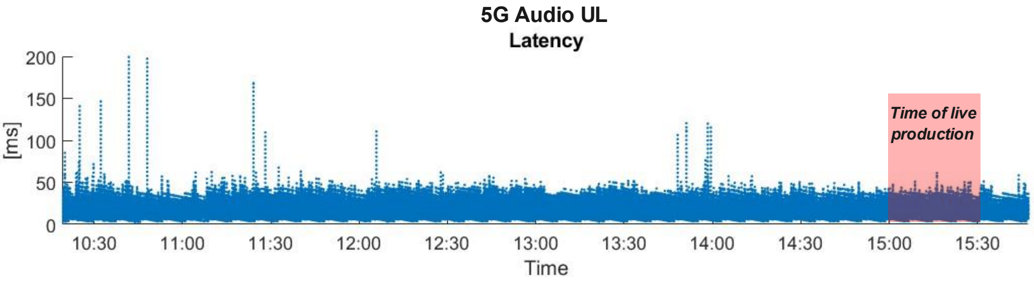


Figure 16: One-way 5G UL audio latency, measured from audio network device to local audio processing for more than 5 hours

2.6 Final Shared Access Spectrum validation

The initial version of the Shared Access Client component from Accelleran and the Shared Access Server component from RED Technologies were initially validated for Phase 1 interoperability as reported in D4.2 [6].

The final validation of an enhanced cloud native disaggregated Open RAN Shared Access Client component from Accelleran and an enhanced Shared Access Server component from RED Technologies using configured spectrum leases for the Sophia Antipolis geographical location were validated again as shown in Figure *17* and Figure *18*.

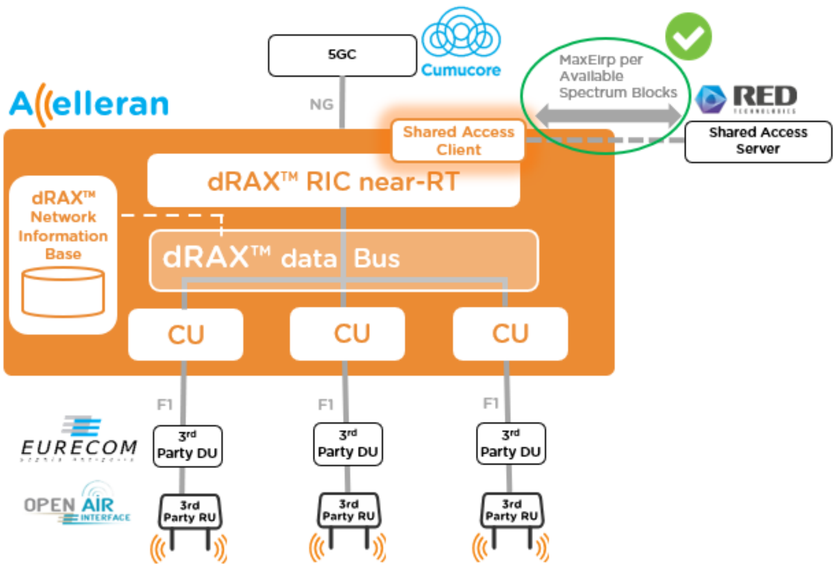


Figure 17: Shared Access client/server interoperability

User ID	ACC-TEST-USERID
Deployment	Outdoor
EIRP capability	30 dBm/10MHz
Latitude	43.614565 DD
Longitude	7.071114 DD
Number of grants	1

GRANT ID	3038c182-4c35-4a1b-8772-6c0f68d1d84a
Status	GRANTED
EIRP currently used	-26.00 dBm/MHz
Maximum allowed EIRP	-26.00dBm/MHz
Frequency range	3550.00 MHz to 3650.00 MHz

Figure 18: Spectrum lease granted in Eurecom Sophia Antipolis site

The scenarios validated were the following:

- Single step registration and grant acquisition
- Multiple step registration and grant acquisition
- Grant relinquishment and deregistration
- Grant suspension and identification of the maximum allowed transmission power to protect nearby users from interferences
- Multiple grant acquisition

Details of the actual procedures validated are presented in Annex A of this document.

3 Use case 2: Multiple Cameras Wireless Studio

This chapter describes the outcomes of the final stage of trials in the context of use case 2 for the **two scenarios** examined in the use case: the **wireless local production** and the **remote production**. The latest updates of the testbed architecture will be presented alongside new measurements results and a detailed KPI analysis. Also, the technology validation of this use case will be addressed.

3.1 Deployed testbed architecture

3.1.1 Updates on network architecture

This section presents an update on the network architecture of use case 2 scenarios for 5G-RECORDS tests and trials.

For the **local production** scenario, the following tests and trials were conducted:

(i) PTP measurements in Aachen (1st round, January 2022) [UC2.B.1]
(ii) PTP measurements in Aachen with GV LDX 150 (2nd round, May/June 2022) [UC2.B.2]
(iii) Final test session in Aachen (October 2022) [UC2.B.3]
(iv) MCR: local instance and MCR in the cloud (July & September 2022) [UC2.B.4]
(v) Mobility and handover in Aachen trial network (October 2022) [UC2.B.5]
(vi) Tests with Low Latency HEVC encoding (October 2022) [UC2.B.6]

For the **remote production** scenario, the following tests and trials were conducted:

(vii) Remote production tests in Aachen (Germany) [UC2.B.7]
(viii) Trials at UPV in Valencia, Spain, with their lab and Orange commercially (June 2022) [UC2.B.8]

Both scenarios were jointly showcased in the final UC2 trial:

(ix) Final trial in Copenhagen (Tivoli gardens, June 2022) [UC2.B.9]

3.1.1.1 PTP measurements in Aachen (1st round) [Local production – UC2.B.1]

<u>PTP performance</u>

Time synchronization between media sources can be challenging in 5G environments due to inherited jitter, introduced by scheduling and TDD, throughput variations, and path asymmetries (different capacity allocation for uplink and downlink), and varying latencies of the radio transmissions. In 5G-RECORDS UC2 we have explored the possibility of using IEEE 1588 PTP (Precision Time Protocol) over a 5G system for media production applications. 3GPP TS 23.501 Release 16 has introduced support for precise time synchronization, focusing initially on gPTP for TSN. The Time synchronization feature was extended (and renamed from Time-sensitive networking (TSN) to time-sensitive communication (TSC)) in 3GPP TS 23.501 Release 17, also supporting SMPTE Profile for the use of IEEE Std 1588 Precision Time Protocol in Professional Broadcast Applications ST 2059-2:2015. The 3GPP system can operate in three modes for time synchronization: PTP Relay (IEEE Std 802.1AS), as a Boundary Clock, or as Transparent Clock. For these tests, the Transparent clock mode was used.

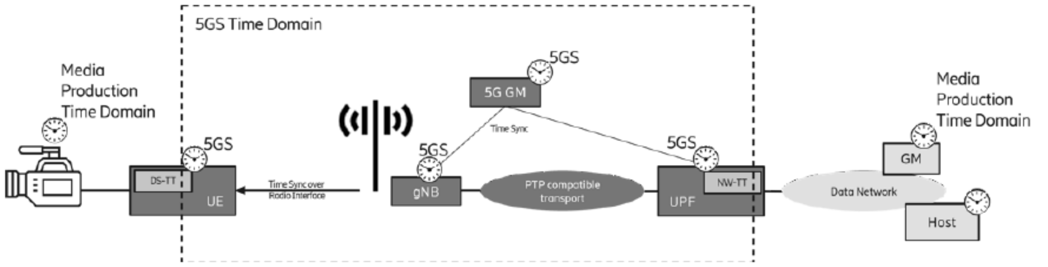


Figure 19: Timing Architecture

The blocks DS-TT and NW-TT (see Figure 19) handle the PTP specific message processing inserting an ingress timestamp when a PTP message enters the 5G system (5GS) and calculating the residence time when the PTP message is leaving the 5G system. The NW-TT and the DS-TT function within the UE must be precisely synchronized to the same time domain to determine the residence time. The 5GS clock is used as the time domain.

<u>Testbed setup:</u>

The testbed operates within Frequency Range 2, i.e., at 28GHz (mmWave) and with a 200MHz bandwidth. It is configured with a 1:1 TDD pattern, which means the scheduler can split radio resources equally between the uplink and the downlink. The subcarrier spacing is set to 120KHz, resulting in a time slot duration of 0.125ms. This configuration will result in low latency because the UE gets a slot for transmission every short period of time. The testbed network can provide a consistent RTT of ~2ms. For PTP support, the URLLC testbed acts as an end-to-end transparent clock. This PTP clock was added in Release 17 to support time synchronization for media production use-cases as defined by SMPTE ST 2059-2:2015.

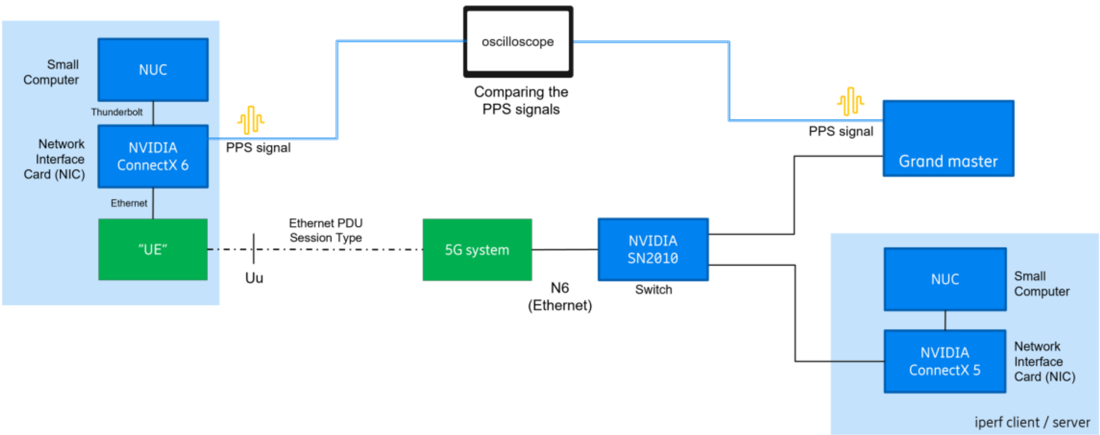


Figure 20: UC2 Aachen testbed

The Tektronix SPG8000A was used as the main PTP clock providing grandmaster capabilities to the system. The standard SMPTE ST 2059-2:2015 PTP profile was used, except for the communication method set to Unicast. The SPG8000A was locked to GPS using a rooftop antenna. The reference output of SPG8000A was connected to the oscilloscope using a 50 Ohm cable and was configured for PPS out.

The NVIDIA SN2010 switch was used in a non-PTP aware state and therefore was simply forwarding the IP packets. The QoS was not enabled on the switch. 1000BASE-T Copper RJ-45 SFPs from FS were used in the switch. Intel NUCs running Ubuntu 20.04 with NVIDIA NICs were used as PTP clients and iperf endpoints. The ptp4l library from *linuxptp* version 3.1.1 suite was used as PTP client. For precise iperf measurements, the system clocks of Intel NUCs were also synchronized to PTP through built-in NICs with PTP-capable Intel I210-AT Ethernet controllers and using the phc2sys library. All NTP clients and services were disabled to avoid interfering with the PTP time transfer function. The NVIDIA/Mellanox OFED driver version 5.5-1.0.3.2 was used with the real-time clock, and ports timestamping explicitly enabled and recommended kernel configuration applied. The PPS output on ConnectX6 DX PPS was enabled for the appropriate built-in clock and set to a 1- second interval using *testptp.c* library compiled for the kernel used. The PPS output of the NIC was connected to an oscilloscope with a 50 Ohm cable. The second NUC with ConnectX-5 NIC was used exclusively as the second iperf endpoint.

The measurement results of these tests are presented in Section 3.2.1.

3.1.1.2 PTP measurements in Aachen (with GV LDX 150, 2nd round) [Local production - UC2.B.2]

The main goal of this test was to see if the PTP accuracy over URLLC 5G testbed would be enough to transport ST 2110-22 stream using JPEG-XS codec natively supported by GV LDX150 camera. At the same time, we used this occasion to measure PTP accuracy once again with a different PTP client (the GV camera).

For these tests the same testbed was used as described in the section above GV camera connected to the UE instead of the NUC and with no PPS being used.

The measurement results of these tests are presented in Section 3.2.2.

3.1.1.3 Final test session in Aachen [Local production - UC2.B.3]

Phase 2 of the final stage took place in Aachen in the first week of April 2022 for the multiple-camera wireless studio scenario, while the remote production scenario took place during the last week of March 2022. The session aims to validate the integration of all the available components with the 5G test network in Ericsson, Aachen lab.

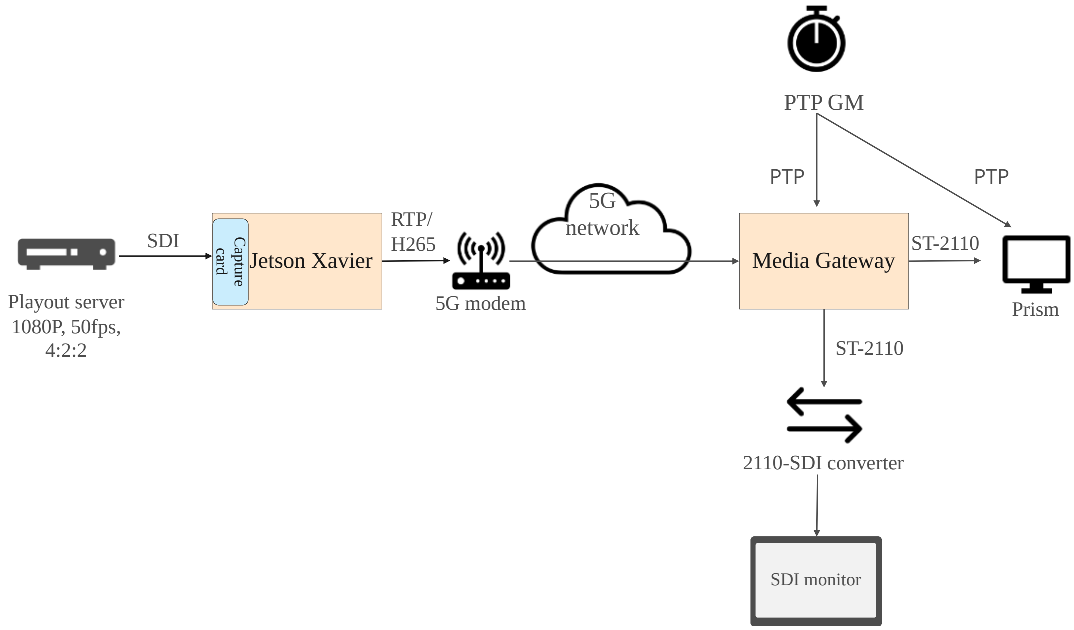


Figure 21: The component architecture validated during the final testing phase

Figure 21 demonstrates the architecture of the validated components for the multiple-camera wireless studio scenario. The tested components are the 5G network, the 5G modem from Fivecomm, the encoder unit (Jetson Xavier), and the Media Gateway. The components' individual validation was discussed in D4.2 in section 3.1.8 [6].

During this phase, the team dedicated a detailed session to study the media traffic behavior over the 5G network. Table 2 shows the used bitrates, the Frames Per second (FPS) and the GOP structure. The team generated 5 tests using the video sequence provided by TV2 with a bitrate between 10 - 50Mbps and 50 FPS with 1 second GOP, as can be seen in Figure 22.

```
gst-launch-1.0 v4l2src device=/dev/video0 ! nvvidconv ! "video/x-raw(memory:NVMM), width=
1280,height=720,format=NV12, framerate=50/1" ! nvv4l2h265enc bitrate=20000000 maxperf-
enable=true preset-level=1 control-rate=1 iframeinterval=50 MeasureEncoderLatency=true !
'video/x-h265, stream-format=(string)byte-stream' ! h265parse config-interval=-1 !
rtph265pay ! udpsink host=10.85.197.249 port=5000 sync=0
```

Figure 22: Gst pipeline used for testing

Table 2: Tested bitrates

Test ID	Bitrate	FPS
Test 1	10 Mbps	50 FPS (50/1 GOP)
Test 2	20 Mbps	
Test 3	30 Mbps	
Test 4	40 Mbps	
Test 5	50 Mbps	

During the analysis, the team realized that different bitrates result in similar results. Therefore, the team focused on a single bitrate to explain the traffic behavior.

3.1.1.3.1 KPI Definitions:

In this section, the team defined the KPIs necessary for explaining the results. The newly defined KPIs are as follows (showed in Figure 23 and Figure 24):

3.1.1.3.1.1 Frame delay:

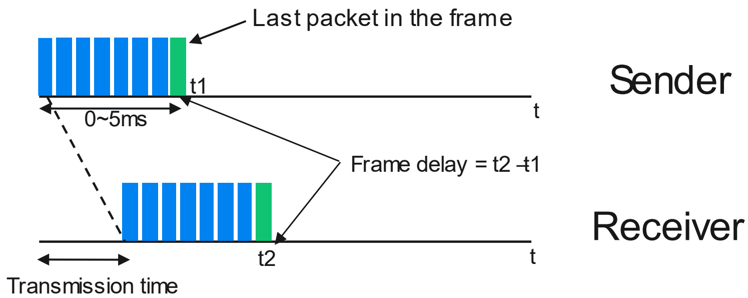


Figure 23: Frame delay

Frame delay is the time difference between receiving the last RTP packet in a frame “t2” and the availability of the same RTP packet at the sender “t1”. It can be noted that all the packets in the frame could be made available at the sender side at 0-5ms. The last packet in the frame is buffered at the sender until all preceding packets are in-flight.

3.1.1.3.1.2 Frame inter-arrival duration:

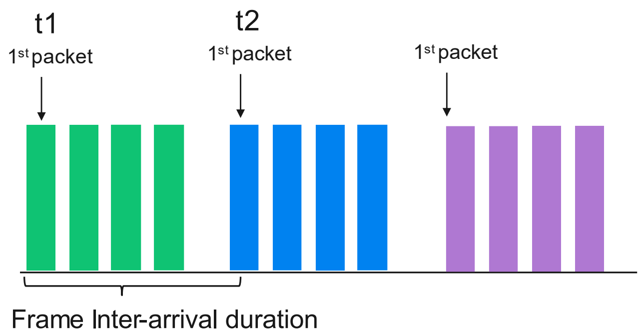


Figure 24: Frame interarrival duration

Frame inter-arrival duration is the time between receiving the first RTP packet in frame "t2" and receiving the first packet in the preceding frame "t1". This time should ideally reflect the FPS configured at the encoder. In our test, the encoder is configured to 50 FPS, so the frame inter-arrival duration should be 1/50 = 20ms.

3.1.1.3.2 Basic architecture

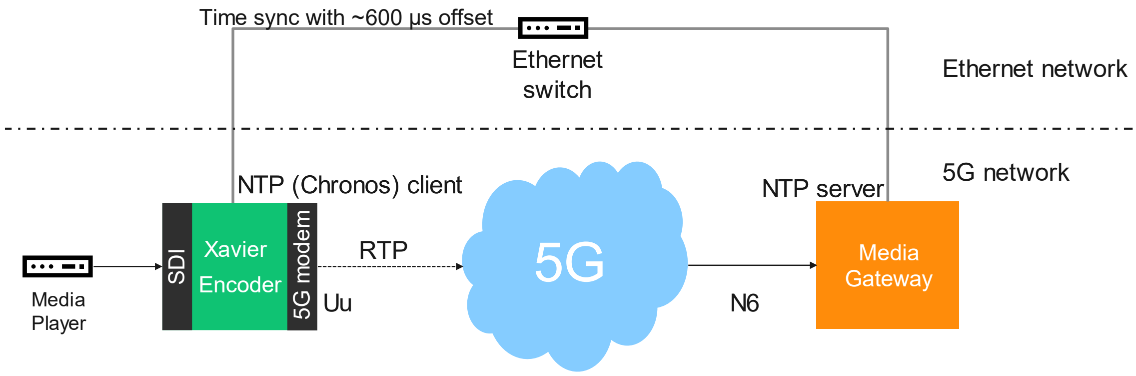


Figure 25: Video analysis architecture

Figure 25 shows the used simplified architecture used in the analysis. The encoder box is connected via SDI to the media player, which act as a live video source. It also has a 5G modem connected via USB interface. The media Gateway is connected to the N6 interface (local break-out). Both the encoder (UE) and the Media Gateway (local break-out server) are connected to each other over the 5G network radio interface.

To achieve an acceptable time synchronization between the encoder and the Media Gateway, a parallel ethernet network is used beside 5G, which is used only for NTP signaling. The Media Gateway hosts an NTP server and connected via another interface to an ethernet switch. While the encoder is connected via ethernet to the same switch.

The encoder runs an NTP client and request the time from the Media Gateway via the ethernet switch. The offset between the encoder and the Media Gateway is measured to be around ~600 µs.

The measurement results of these tests are presented in Section 3.2.3.

3.1.1.4 MCR: local instance and MCR in the cloud [Local production - UC2.B.4]

Apart from using the MCR in the trial in Tivoli (see section 3.1.1.8), EBU assessed its performance in the lab of both the local instance and the cloud instance.

The server used for local instance had an AJA Corvid 88 SDI card installed. So, first we recorded the output of the Test Signal Generator available in GV AMPP with Blackmagic Hyperdeck Studio 12G recorder. We played this in a loop and connected the output to the BlackMagic SmartView SDI monitor. Then from the output of the monitor we brought it into the MCR with the “SDI Input” application. Finally, with the “SDI Output” application we connected the PGM via AJA Corvid 88 card to the second screen of the SDI monitor.

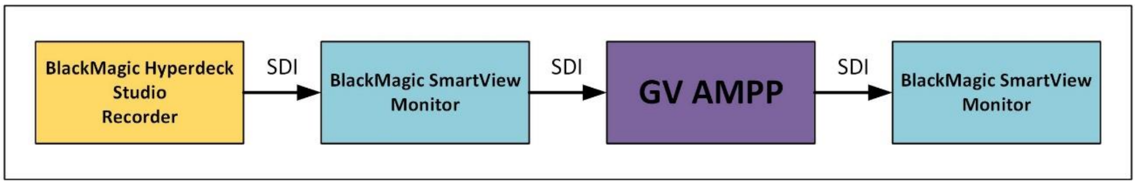


Figure 26: GV AMPP local instance test setup.

The signal was in 1080p50 format and the observed latency was measured to be 32 frames. There is a way to optimize that latency and one of the EBU members reported 13 to 16 frames of latency in their tests of GV AMPP, but our plan was to measure the baseline latency with default configuration. No additional signal processing was taking place in AMPP, which would otherwise add some more latency. Also, we believe that the latency of the other parts of the chain in this test is neglectable.

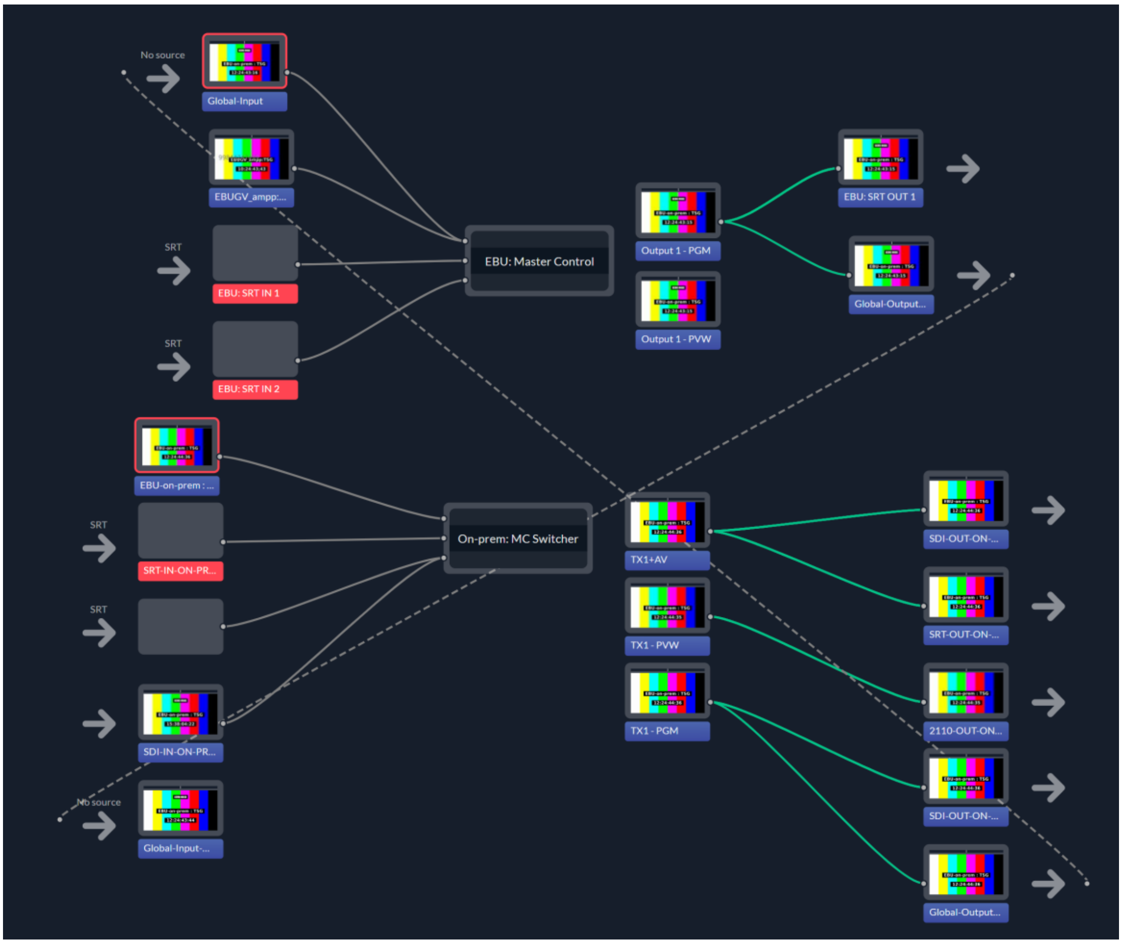


Figure 27: Routing diagram of local and cloud instance interconnected.

For assessing the performance of the cloud MCR we used the Test Signal Generator of the local instance, selecting it as PGM in the switcher and sending it to the cloud instance as a 15 Mbps stream using the “Global output” application available in GV AMPP. The cloud GV AMPP instance received this stream with “Global input” application. After switching this stream was sent back in a similar fashion (“Global output” in the cloud instance, “Global input” in the local instance). Two instances of “Flow monitoring” application were used on the local instance to see the both PGM stream (before being sent to the cloud) and the “Global input” stream after being received from the cloud.

The measurement results of these tests are presented in Section 3.2.4.

3.1.1.5 Mobility and handover in Aachen trial network [UC2.B.5]

The trial network described in D4.1 [4] was used to conduct mobility and handover tests. The test's target is to understand better how the video traffic will be impacted during handover and mobility. The trial network is an outdoor network with an NSA core and 5G-NR and LTE bands available. Our tests are conducted using the 5G-NR radio.

The network consists of multiple cells covering ~ 1KM of the outdoor area. The cell locations and the coverage direction are depicted Figure 28 and Figure 29.

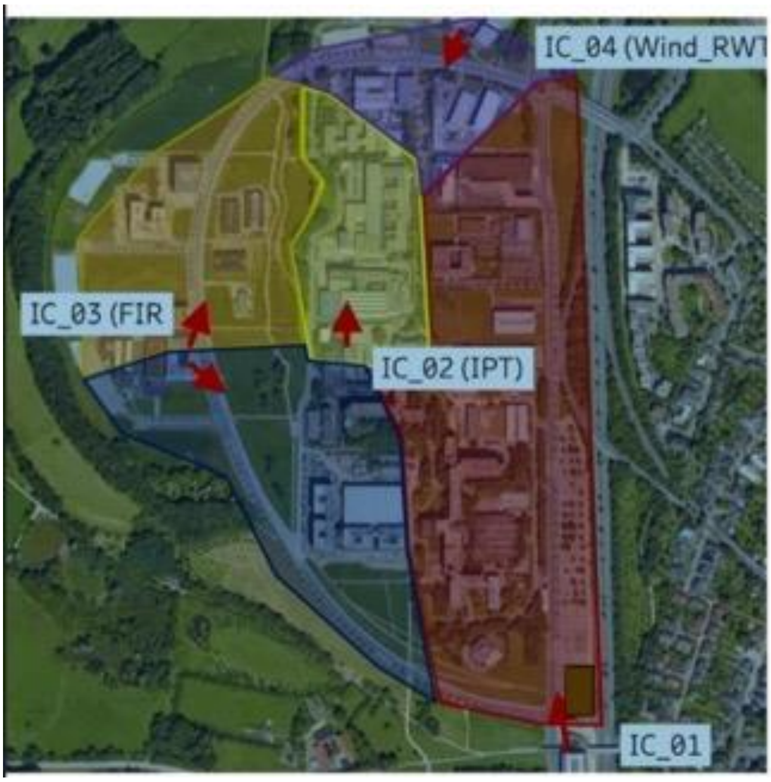


Figure 28: Cell coverage at IPT.

Figure 29: Antenna's location on the building.

The measurement results of these tests are presented in section 3.2.5.

3.1.1.6 Tests with Low Latency HEVC encoding (GDR) [UC2.B.6]

The HEVC codec contains several different tools. Within 5G-RECORDS, we looked at HEVC Gradual Decoder Refresh (GDR) as a tool for low latency. The concept is illustrated in the next two figures:

Figure 30 depicts a normal encoding at the example of 50 fps encoding, where the frame encoding takes 20ms. After the HEVC encoding of the video frame is finished, all data of the video frame are available for the transmission process over 5G.

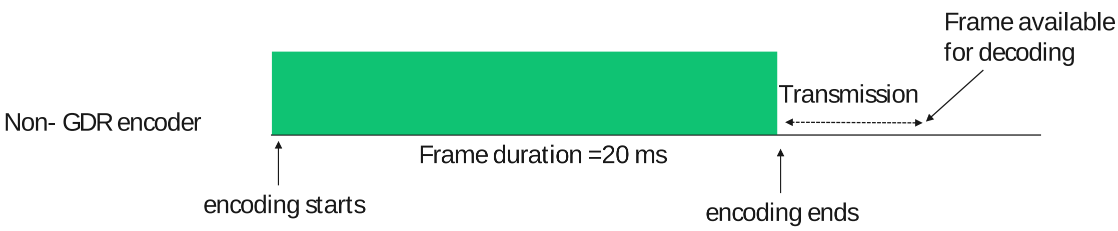


Figure 30: Normal HEVC encoding (Non-GDR)

Figure 31 illustrates the (simplified) concept of low latency HEVC encoding with GDR. Here, the video frame is subdivided into 8 slices. The encoding of each slide takes around 2.5 ms (i.e. 20 ms divided by 8).

Slices become available approx. every 2.5ms and the transmission process can be started. As consequence, the transmission of the data starts much earlier than in the normal HEVC encoding. The decoder can also start the decoding of each slice individually and store the decoded slices until all slices of a frame are available.

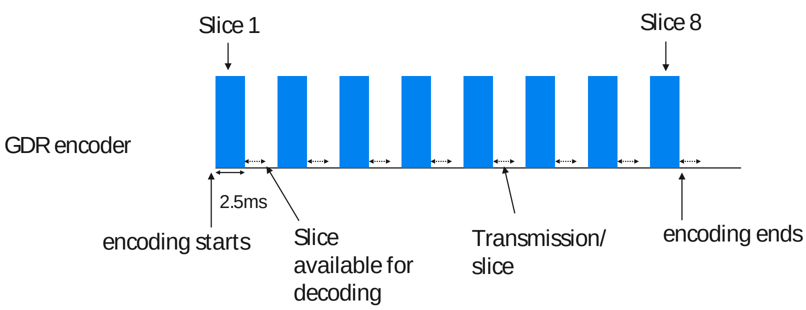


Figure 31: Low Latency HEVC encoding with GDR

The results of these tests are presented in section 3.2.6.

3.1.1.7 Remote production tests in Aachen [Remote production – UC2.B.7]

Before presenting the results from the different tests performed for the remote production scenario (i.e., tests in Aachen, Tivoli trial and UPV trial), it is worth describing some common and general aspects of all tests and trials for the remote production scenario.

The architecture of the remote production scenario includes LiveU LU800Pro with embedded Sierra Wireless 5G EM9190 modems as well as external 5G modems. A single or multiple live video feeds (multiple by using LiveU Blackmagic video player and splitter with RAI/LiveU video clips) was connected to the LU800Pro. It was configured to work with the relevant 5G networks. The transmission went through the 5G network, the public Internet, into the RAI lab in Turin (or a LiveU cloud server in some cases). End-to-end application-level performance measurements were collected by the LU800Pro, according to its application-level estimation based on communication with the LU2000 server ("MMH") on the receiving end. Some considerations are noted below:

- Performance parameters: UL BW, UL latency, UL loss rate. These are logged in snapshots every 5 seconds and then gathered and annualized offline.
- The LU800Pro maximum bandwidth when connected to a single feed is 30Mbps and when transmitting 4 simultaneous feeds – 60Mbps.
- UL congestion was done via a computer running iperf or similar that generated continuous traffic at specific rates and connected to the relevant network/slice via another modem.

It is worth noting that 2-modems bonding, i.e., when the LU800Pro uses two modems for transmitting its live, HEVC-encoded video, can be used to compare performance over the two links. When all is equal between the links, the LU800Pro splits the transmitted packets very evenly between the two. When there are performance issues with one of them, as the LU800-LU2000 application measure them, the LU800Pro will react and split the video differently. In extreme conditions such as the performances of both links are bad, the LU800 will reduce its HEVC encoder output to match the total available bandwidth according to its measurements.

The results shown for each trial are the significant highlights, but for the sake of space, not all test cases and their results are given below.

The measurement results of these tests are presented in section 3.2.7.

3.1.1.8 UPV trial testbed [Remote production - UC2.B.8]

The trial at UPV was done in two different testing environments:

a) Indoors at UPV lab, with commercial 5G NSA network provided by Orange and a 5G SA test network provided by Amarisoft.
b) Outdoors with the commercial 5G NSA network. All the tests were done over a wireless connection.

The test site is shown below. The UPV lab is located in the adjacent building to the gNB placement (< 50 meters). The distance between the gNB and the outdoor test site is around 520 meters.

Figure 32: Outdoor test site and NSA gNB placement[2] of LU-UPV trial.

While there were no slices available, the trial included the use of different TDD UL/DL patterns (different between the 5G NSA and 5G SA network) and 3 different channel bandwidths: 100MHz, 50MHz, 20Mhz in the case of the 5G SA. Configurations of x2 bonding vs no bonding were also tested.

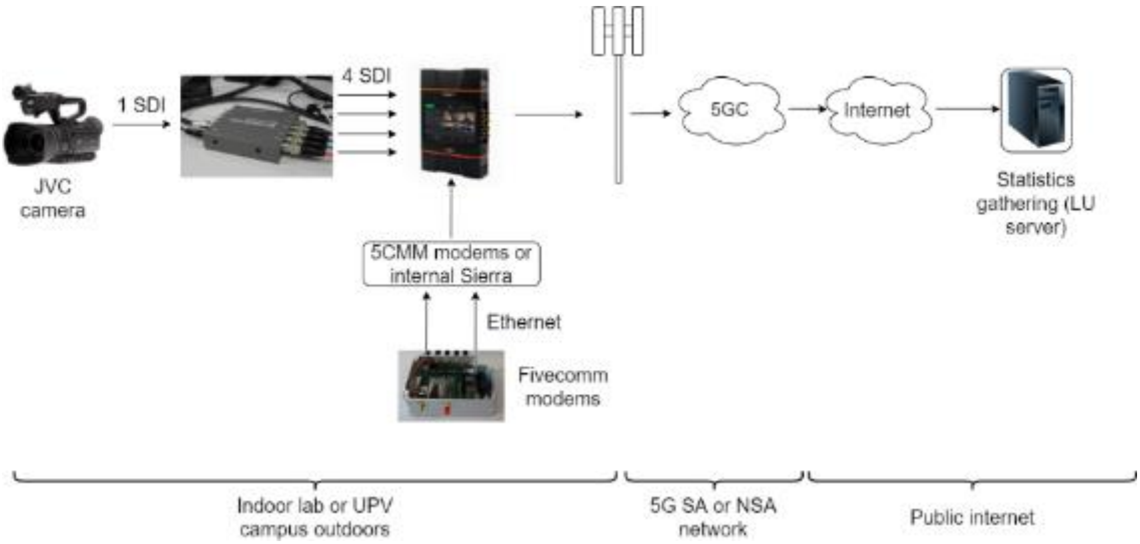


Figure 33: Tests setup for LU-UPV trial.

The indoor and outdoor setup and equipment is shown below. In this setup, an SDI camera was connected to an SDI distribution box to obtain a replicated 4 streams. This was done to make the LU unit work at the top of its capabilities, with 4 input SDI streams that carry uncompressed audio and video. Tests were made with the internal Sierra modems or with the Fivecomm modems connected via Ethernet to the unit. The internal or external modems connect to the 5G network wirelessly. For the SA and NSA networks, the utilized SIM cards are different. The internet access is provided to the modems via 3GPP PDU session, and the DNN utilized depended on the utilized SIM card.

[2] The UPV lab is not shown in this picture, but it is adjacent to the gNB placement.

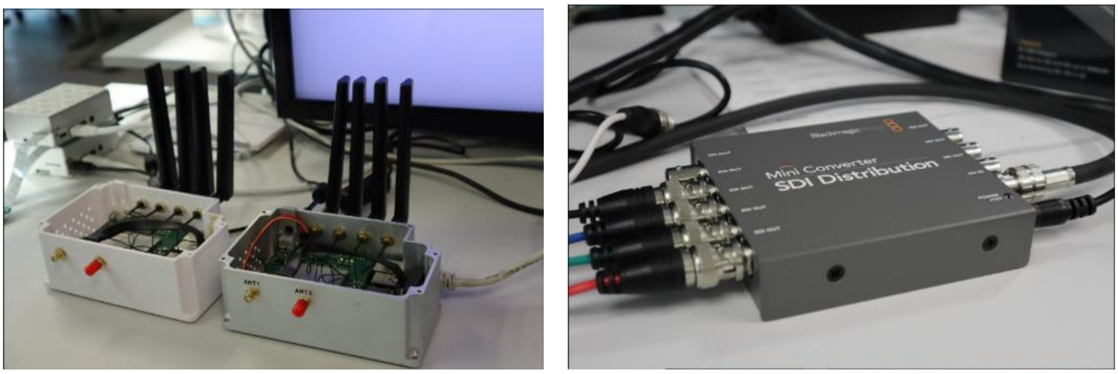


Figure 34: (a) Fivecomm modems for single and bonding scenarios. (b) SDI distribution device to quadruplicate a single SDI stream from the camera for LU-UPV trial.

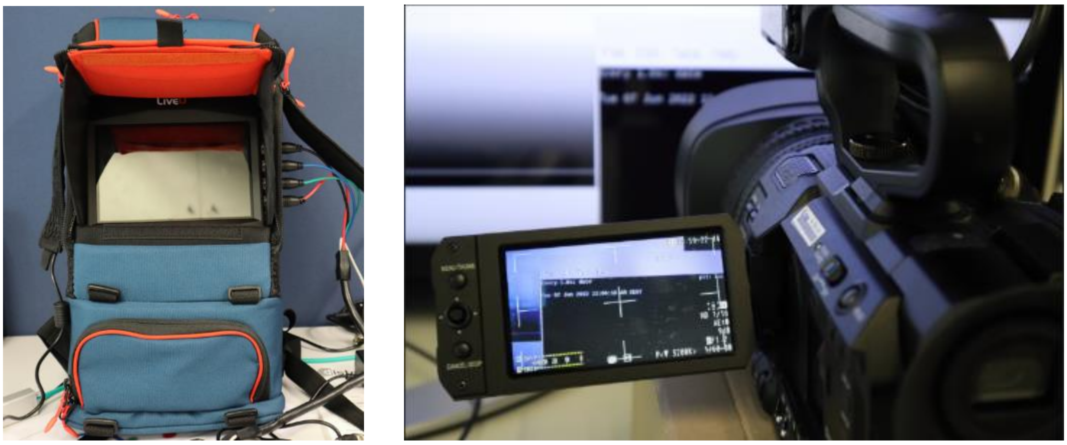


Figure 35: (a) LU800 unit. (b) JVC camera used for the LU-UPV trial.

Figure 36: Outdoor testing setup for UPV-LU trials.

Following, the test planning performed is shown:

Table 3. Test planning for LU-UPV trial.

Test ID	Type	Comments	Unit	Modem
T1	**NSA Benchmark Sierra**	x4 SDI streams 1080p60, 60Mbps max, commercial 5G NSA, bonding x2 SIMs	LU800	Sierra internal

T2	**NSA Single Sierra**	x4 SDI streams 1080p60, 60Mbps max, commercial 5G NSA, no bonding	LU800	Sierra internal
T3	**NSA Benchmark Fivecomm**	x4 SDI streams 1080p60, 60Mbps max, commercial 5G NSA, bonding x2 SIMs	LU800	Fivecomm external
T4	**NSA Single Fivecomm**	x4 SDI streams 1080p60, 60Mbps max, commercial 5G NSA, no bonding	LU800	Fivecomm external
T5	**Benchmark SA Bonding Sierra**	x4 SDI streams 1080p60, 60Mbps max, lab SA 5G network, x2 bonding, 30KHz SCS, (30/70% of UL/DL pattern), 100MHz BW	LU800	Sierra internal
T6	**Benchmark SA Single Sierra**	x4 SDI streams 1080p60, 60Mbps max, lab SA 5G network, no bonding, 30KHz SCS, (30/70% of UL/DL pattern), 100MHz BW	LU800	Sierra internal
T7	**Reduced BW sierra single**	x4 SDI streams 1080p60, 60Mbps max, lab SA 5G network, no bonding, 30KHz SCS, (30/70% of UL/DL pattern), 20MHz BW	LU800	Sierra internal
T8	**Reduced BW sierra bonding**	x4 SDI streams 1080p60, 60Mbps max, lab SA 5G network, 30KHz SCS, (30/70% of UL/DL pattern), 20MHz BW, x2 bonding	LU800	Sierra internal
T9	**Background traffic with UE Iperf, bonding**	x4 SDI streams 1080p60, 60Mbps max, lab SA 5G network, x2 bonding, 30KHz SCS, (30/70% of UL/DL pattern), 100MHz BW, background traffic with another UE doing Iperf to Amarisoft	LU800	Sierra internal
T10	**Background traffic with UE Iperf, single**	x4 SDI streams 1080p60, 60Mbps max, lab SA 5G network, no bonding, 30KHz SCS, (30/70% of UL/DL pattern), 100MHz BW, background traffic with another UE doing Iperf to Amarisoft	LU800	Sierra internal
T11	**Amarisoft UL benchmark**	x4 SDI streams 1080p60, 60Mbps max, lab SA 5G network, x2 bonding, 30KHz SCS, (70/30% of UL/DL pattern), 100MHz BW	LU800	Sierra internal
T12	**Benchmark SA Orange network single**	x4 SDI streams 1080p60, 60Mbps max, Orange SA network, no bonding	LU800	Fivecomm external
T13	**Benchmark SA Orange network bonding**	x4 SDI streams 1080p60, 60Mbps max, Orange SA network, x2 bonding	LU800	Fivecomm external
T14	**Outside production without bonding**	x4 SDI streams 1080p60, 60Mbps max, Orange NSA network, no bonding	LU800	Sierra internal
T15	**Outside production with bonding**	x4 SDI streams 1080p60, 60Mbps max, Orange NSA network, x2 bonding	LU800	Sierra internal

The measurement results of these tests are presented in section 3.2.9.

3.1.1.9 Final trial in Copenhagen (Tivoli gardens) [Local and remote production - UC2.B.9]

In July 2022, UC2 partners met for the first time in Copenhagen for the execution of the final trial including an end-to-end production chain. Both local and remote production scenarios were here showcased. Hosted by the Danish broadcaster TV2 at its studio facility within the famous Tivoli amusement park in Copenhagen, the setup was centered around a 5G non-public network (5G SNPN) for wireless cameras and microphones. It included the broadcast live transmission of an interview with one host and two guests.

Figure 37: Overview of the UC2 Tivoli trial. Media gateway and cloud production (left) and video cameras with 5G interface units (right).

Two cameras and a microphone were connected via a private 5G network to the media gateway specifically designed and developed within the project and a local GV AMPP instance connected to the TV2 gallery in the Tivoli part (see Figure 38). For the details about the contribution link, see the next section. This one focused on the multi-camera live production.

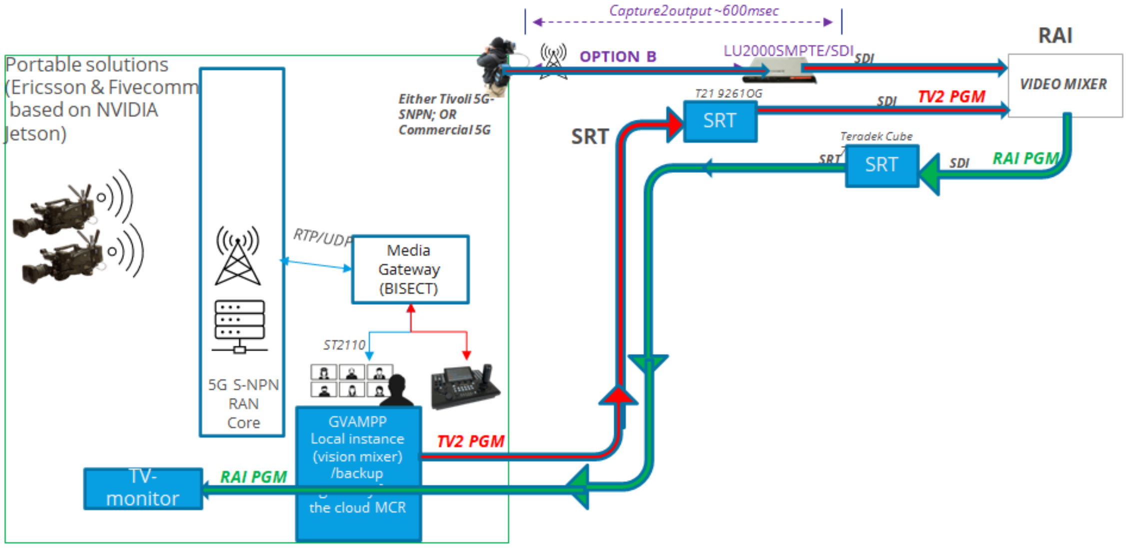


Figure 38: Setup during the Tivoli trial.

The list of components used for the trial is reported in Table 4.

Table 4: List of components used in Tivoli trials

	Component	Provider
Camera unit interface (portable)	NVIDIA Jetson Xavier used as codec integrated with the 5G S-NPN modem in a portable solution	Partners: Ericsson and Fivecomm
Baseline 5G modem	5G S-NPN modem	Partner: Fivecomm
Network	Portable S-NPN	Third party: Neutral wireless/Strathclyde University

Media Gateway	NVIDIA Jetson Xavier for media decoding/encoding, transport protocols remapping/translation	Partner: BISECT
MCR	GV AMPP local instance responsible for decoding the signals for the TV2 gallery	Partner: EBU Third party: Grass Valley

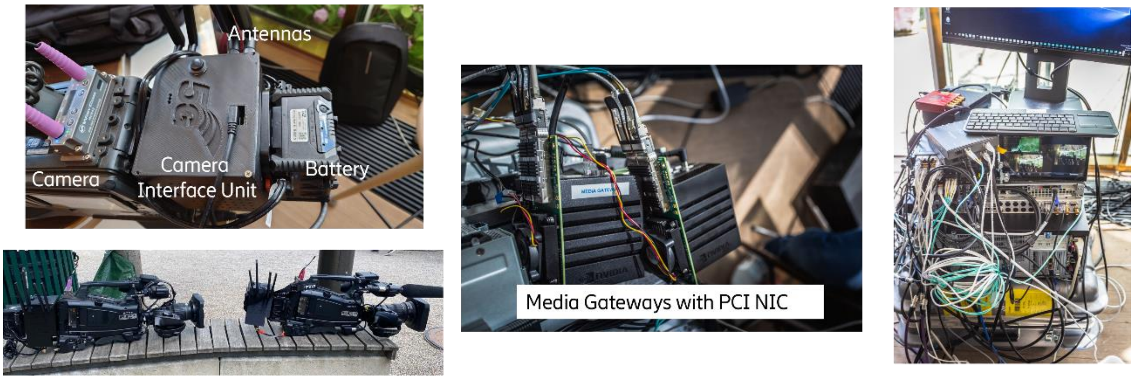


Figure 39: Main components integrated in the trial (local production scenario).

The Camera Interface Unit is formed by the 5G modem developed by Fivecomm (only the 5G module board), connected via USB to an NVIDIA Jetson Xavier (provided by EBU) and an SDI card that is in turn connected to the Jetson and used to capture the video from the camera. The NVIDIA Jetson board in the camera interface unit was used to encode the input signal (1080p50) with HEVC at 50Mbps and at 20Mbps. Note that the 5G modem is additionally connected to an external button, which is used to power on and off all components.

To make the solution portable and to be plugged into the professional camera, a 3D case was designed and printed. The case comes not only with an external button to power up the solution as explained, but also with external SMA connectors for mid-band 5G antennas, 3 LEDS for monitoring the status of the 5G modem, and 2 V-locks that are used to plug an external battery and the entire solution to the camera. The Camera Interface Unit was mounted on the back of the cameras provided by TV2.

Figure 40 shows another perspective of the 5G camera interface unit transmitting live over the 5G network.

Figure 40: 5G camera unit interface being used during the trial.

The HEVC stream was mapped into RTP and transmitted through the Fivecomm 5G modem to the 5G network working with these parameters:

Table 5: 5G network main parameters.

Parameters	
Band	n77
Center Frequency	3,870MHz
Bandwidth	100 MHz
No. DL Antenna	2
No. UL Antenna	2
TDD Config	2 DL and 7 UL slots (5ms switching period)

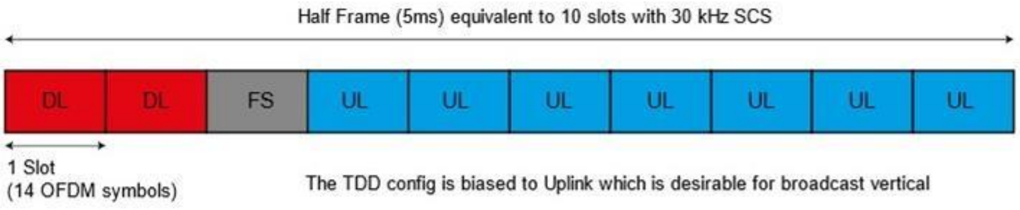


Figure 41: TDD configuration used in the trial.

The two RTP streams (one for each camera) were mapped into two SRT streams by the media gateway and passed to the MCR. The MCR decoded the two stream and forwarded the uncompressed SDI signals to the vision mixer in the TV2 gallery.

The ST 2110 conversion was not available in the media gateway at that time but was implemented straight after the trial.

The event went live for 15 minutes and transmitted to a conference room in Copenhagen where a 5G seminar organized by TV2 was taking place.

The Packet capture (PCAP) file was recorded during the live transmission with the intention to replicate the setup a second time in the laboratory with the ST 2110 feature included and working.

The Media Gateway was configured to receive the UDP streams coming from both cameras and to rewrap them as SRT, which was delivered to the GV AMPP. In parallel, the MG also decoded and presented both streams, in order to provide confidence monitoring.

The MG was monitoring the input streams, both in terms of bit rate and stream integrity. No packet losses were registered on either of the input streams and the bit rate was within the expected bounds.

The MG was configured in order to introduce as minimum a latency as possible and yet produce a valid MPEG TS stream that was subsequently mapped onto SRT. The MG was locked to PTP and, although the CIU clocks were not synchronized with it, the measurements showed that there was no significant jitter or wander.

In terms of confidence monitoring, the MG exposed the video streams via its WebRTC interface, which allowed the video to be displayed on a browser. We measured a consistent glass-to-glass delay between 180 and 200 ms, from the camera to the browser monitor running on a laptop computer. The MG was also configured to decode and display the streams on the MG console.

BISECT performed compatibility, performance and stability measurements for all the supported combinations of formats and protocols.

Running on a Jetson AGX Xavier with a NVIDIA ConnectX NIC, the MG was able to sustain 2 video streams and at least 2 audio streams on each direction (camera to studio and vice-versa), for any format combination.

Compatibility was tested using open-source tools such as GStreamer, FFmpeg, OBC and VLC, as well as commercial products such as GV AMPP, Larix Broadcaster and NVIDIA Rivermax.

All combinations were tested for stability by running a stream for at least 24h.

The latency was consistent across all formats, with values similar to the ones measured in the trials (around 200ms), expect for SRT and RIST which, inherently, introduce an additional, configurable, latency.

3.1.1.9.1 Remote production:

Figure 42 and Figure 43 show the the remote contribution scenario based on LiveU equipment. The LU800Pro was connected to the TV2 camera on a tower outside the Tivoli Garden, using a 5G mmWave CPE and antenna installed on the tower outside, directing point-to-point to the 5gmmWave directional point some 500 m away. While it was also using a 5G SNPN which was set at that area, the LU800Pro was configured to prioritize working with the 5G mmWave by using the LU800 prioritizing functionality to prioritize to the max the RJ45 Ethernet port where the mmWave CPE was connected to over all other links.

Both networks were provided by TDC. The mmWave used N258, 800MHz bandwidth. The radio equipment was based using Ericsson equipment and Inseego pole mounted outdoor CPE.

Figure 42: Remote production installation scenario at Tivoli Garden.

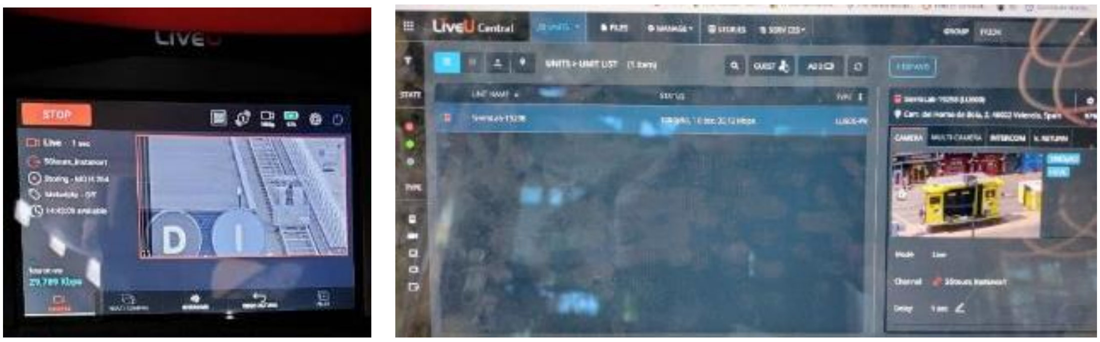


Figure 43: LU800Pro screenshots from remote production trial at Tivoli Garden.

In addition, a basic remote-control room is set up within RAI laboratories in Turin, to evaluate the usability of the remote production scenario. Figure 44 partially shows the temporary RAI lab setup.

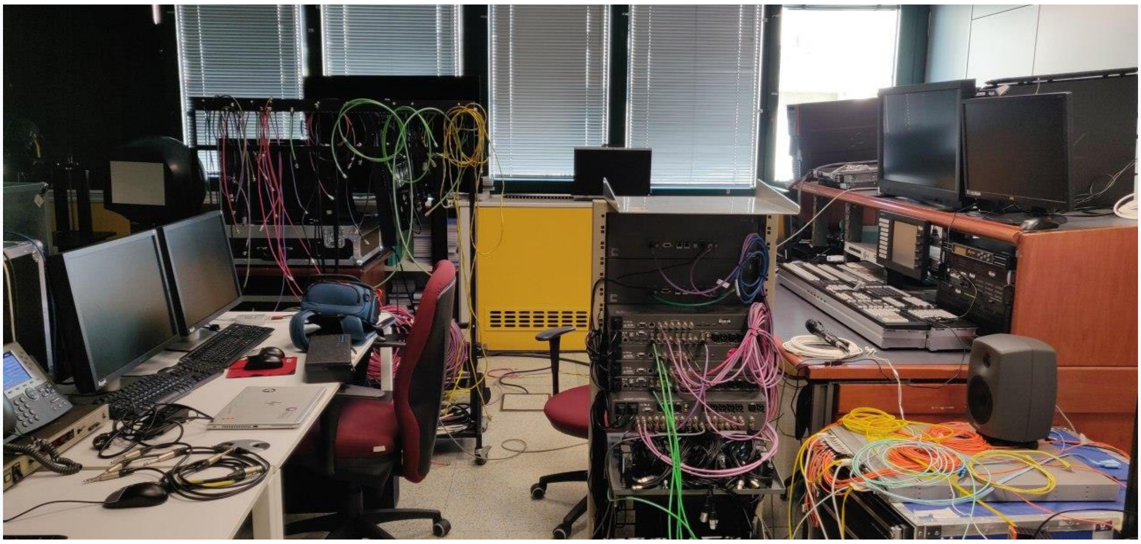


Figure 44: RAI control room.

The RAI control room is equipped with:

- A video mixer BlackMagic;
- A portable video matrix; for internal signal distribution among equipment
- A SRT decoder T21 9261OG; used for the Tivoli's video feed reception
- A SRT encoder Teradek Cube 755; used to send the RAI's PGM back to Tivoli
- LU2000-SMPTE receiver; for the contribution video feed coming from LU800
- Two SDI players BlackMagic Hyperdeck; one used for insertion of RAI logo, and the other one used to play a looped clip.

Figure 38 also shows the video flows among the different locations. Two video signals are sent from Tivoli to the RAI control room in Turin via public Internet: the TV2 PGM and the LU800 contribution feed.

The PGM is produced by the local GV AMPP instance and sent as SRT stream via fixed public network to Turin. It is then converted into SDI for mixer processing.

The contribution feed instead, is sent to Turin via a commercial NSA 5G network present on site in Tivoli. It is then retrieved from output of the LU2000-SMPTE in SDI format.

During the Tivoli live event produced locally by TV2, a simultaneous remote RAI PGM is produced inside RAI lab sent back to Tivoli for E2E latency measurement.

The measurement results of this trial are presented in section 3.2.10.

3.1.2 Uncertainties and risk assessment

The main uncertainties in UC2 were related to the availability of a portable 5G enabled camera and a portable SA network capable to operate in the Danish frequency spectrum. For this reason, both Fivecomm and Ericsson collaborated and devoted time to deploy a compact solution including the modem and encoding board that was plugged on the back of broadcast TV cameras as described in the sections above. This deployment was not included in the initial Grant Agreement as Image Matters was the partner initially responsible for that. During the preparation calls for the live trial in Aachen, Ericsson checked the possibility to provide a portable setup on top of the lab and trial network in Aachen. Unfortunately, at that time, the Ericsson portable 5G network, which was used later during the IBC demonstration, did not have radio units that could operate in the range 3,81 GHz – 3,91 GHz (100MHz), band n77 available in Copenhagen. To overcome the impasse, BBC connected the consortium with Neutral wireless who provide the SA portable setup in Tivoli.

Another uncertainty was whether the LU800Pro will work with the mmWave CPE as its external modem. It was impossible to mitigate or pre-test because of the availability of this mmWave CPE and the full mmWave 5G network. However, the LU800Pro did work with the mmWave 5G CPE and network as external modem over the LU800 Ethernet port.

3.2 Measurements results

This section will present the results obtained from the trials described in the previous section for the local production and the remote production scenarios.

3.2.1 PTP measurements in Aachen (1st round) [Local production - UC2.B.1]

This section presents the PTP measurement results conducted in Aachen (see section 3.1.1.1).

3.2.1.1 *Test 1: Measurements without timing assistance or PTP client's parameters tweaking*

This test set the baseline. This is the PTP accuracy that one can get from a 5G network with similar properties to the testbed used without tweaking the parameters of the PTP client and without getting information regarding the residence time of PTP packets traversing the 5G network.

The general parameters of the test are as follows:

Parameters – Test 1	
Residence time provided	No
PTP servo used	Linear regression
Advanced servo parameters tweaking in PTP client	No
Number of data points	1577

The main results are the following:

- Average PTP accuracy: 116653 ns (~117 µs)
- Median PTP accuracy: 116036 ns (~117 µs)

PPS measurements results:

- Number of data points: 12

- Average offset: 152 μs

3.2.1.2 Test 2: Measurements with timing assistance and PTP client's parameters tweaking

This was the test with the best results. This is the PTP accuracy that one can get from a 5G network with similar properties to the testbed used by tweaking the advanced PI servo parameters of the PTP client and, most importantly, using the feature where the residence time of PTP packets traversing the 5G network is provided to PTP endpoints via the correction field of the PTP packets.

The general parameters of the test are as follows:

Parameters – Test 2	
Residence time provided	Yes
PTP servo used	PI
Advanced servo parameters tweaking in PTP client	Yes
Number of data points	932

The main results are the following:

- Average PTP accuracy: 3451 ns (~3,5 μs)
- Median PTP accuracy: 2832 ns (~2,8 μs)

PPS measurements results:

- Number of data points: 20
- Average offset: 4,066 μs

3.2.1.3 Test 3: Measurements with timing assistance but no PTP client's parameters tweaking

This test was performed to see how much difference the PTP client's servo and advanced parameters tweaking make.

The general parameters of the test are as follows:

Parameters – Test 3	
Residence time provided	Yes
PTP servo used	Linear regression
Advanced servo parameters tweaking in PTP client	No
Number of data points	449

The main results are the following:

- Average PTP accuracy: 3633 ns (~3,6 μs)
- Median PTP accuracy: 3106 ns (~3,1 μs)

PPS measurements results:

- Number of data points: 10
- Average offset: 4,756 μs

3.2.1.4 Test 4: Measurements under load - in the presence of network congestion

This test was performed to see if the PTP accuracy is degraded if the 5G network is loaded significantly. 150 Mbps of uplink traffic and 120 Mbps of downlink traffic were generated.

The general parameters of the test are as follows:

Parameters – Test 4	
Residence time provided	Yes

PTP servo used	Linear regression
Advanced servo parameters tweaking in PTP client	No
Number of data points	737

The main results are the following:

- Average PTP accuracy: 23358 ns (~23,4 μs)
- Median PTP accuracy: 23350 ns (~23,4 μs)

PPS measurements results.

- Number of data points: 10
- Average offset: 13,56 μs

The evaluation showed that a synchronization accuracy of less than 4 μs can be achieved with residence time measured and reported by the 5G system compared to ~120-150 μs achieved with the TSN features disabled. The accuracy difference between the linear regression servo and the PI servo with additional tweaks was neglectable. These results have proved the possibility of using such a system for conventional media production over 5G, where the genlock or sampling references are generated locally by a media device receiving the timing information from a network via PTP. While this accuracy may theoretically be sufficient for high performance uncompressed media over IP solutions like the ST 2110, and it can theoretically be evaluated if it is enough for the precise packet pacing required by the ST 2110-21, it is unlikely that high-performance uncompressed video will be transported over a 5G network due to available spectrum being a too scared resource.

3.2.2 PTP measurements in Aachen with GV LDX 150 (2nd round) [Local production - UC2.B.2]

This section presents the PTP measurement results conducted in Aachen with GV LDX 150 (see section 3.1.1.2).

The PTP accuracy results were close to what was observed earlier with ptp4l client.

The main results are the following:

- Average PTP accuracy: 6772 ns (~6,8 μs)
- Median PTP accuracy: 5948 ns (~5,9 μs)

As of ST 2110-22 with JPEG-XS it was determined that because JPEG-XS is an ultra-low latency codec, it requires average PTP accuracy of 500 ns or lower with spikes of no more than 1 μs, so with the current URLLC testbed it was not possible to achieve that and it was not possible to transport this stream.

3.2.3 Final test session in Aachen [Local production - UC2.B.3]

This section presents the final test session measurements in Aachen (see section 3.1.1.3).

3.2.3.1 Delay analysis:

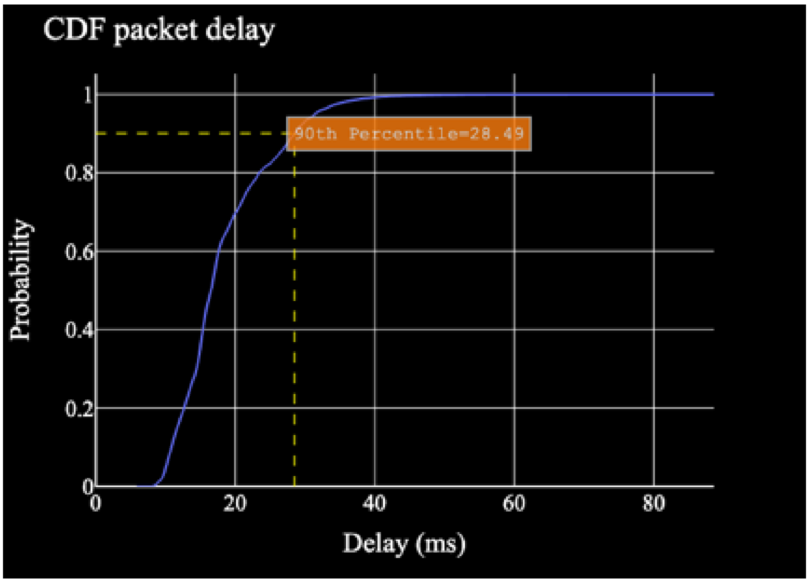


Figure 45: CDF for packet delay

Figure 45 demonstrates the CDF delay for the received RTP packets. It can be noted that the average delay falls below 20 ms, while the 90th percentile is 28.49 ms. Given that the Media Gateway requires 2 frames as a buffer (i.e., 40ms). The measured latency values can guarantee a smooth playout of the stream without the buffer depletion at the Media Gateway. Those values are also in line with the G2G latency measurements discussed at D4.2 [6] section 3.3.3. We can also notice that the 99th percentile could reach above 40 ms and given that the network conditions are stable, The team have made further investigation to understand the reason behind the latency spikes.

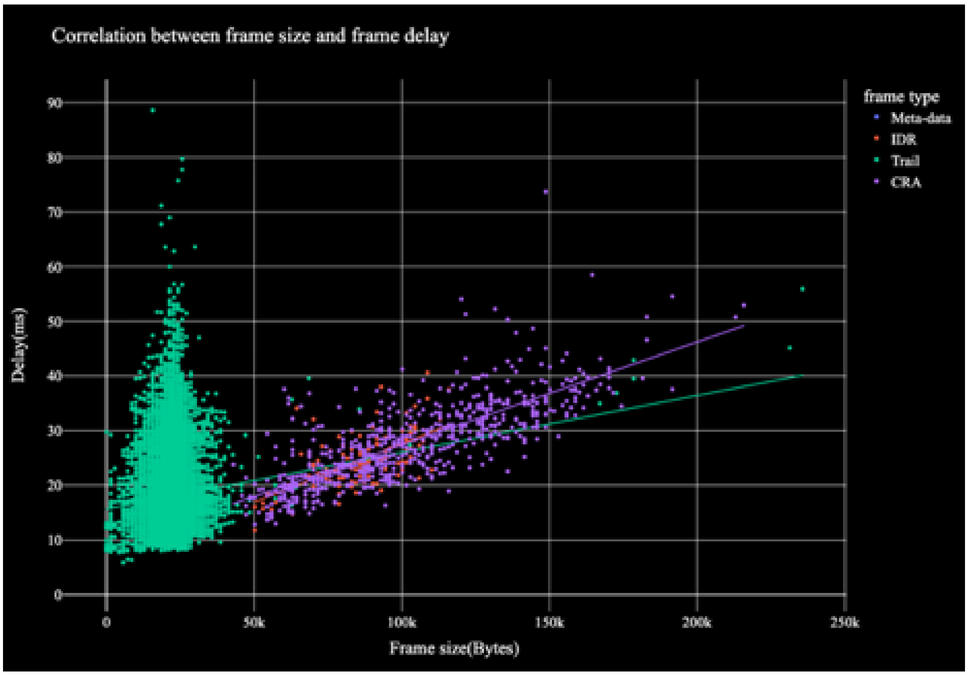


Figure 46: Scatter plot for frame size and frame delay

Figure 46 depicts the relation between the frame size and the frame delay. In this plot, multiple types of frames plotted can be seen: IDR & CRA frames (key frames) and Meta-data and Trail frames. It is known that the key frames are larger than trail frames, since they carry more information about the Group of Pictures (GOP). We can conclude from the scatter plot that there is an uptrend correlation between the frame size and its delay. So smaller frames experience less delay. Such conclusion is logical because larger frames consist of more RTP packets and requires more time to be transmitted. However, it can be also noted that some trail frames have experienced high latency despite its small size. Very few frames in the graph with more than 60ms delay can be seen. Such rare encounter can cause disruption to the playout and incase of varying network conditions those encounters could increase.

To get a better understanding of the frames that experienced high latency, all the received packets were plotted with their delay against time. Figure 47 shows that packet latency starts to build up at the beginning of each GOP. The GOP starts with metadata followed by IDR or CRA frames, and it can be seen from the figure that the metadata always has a delay of below 10ms, and then packet delay increases until the next GOP start.

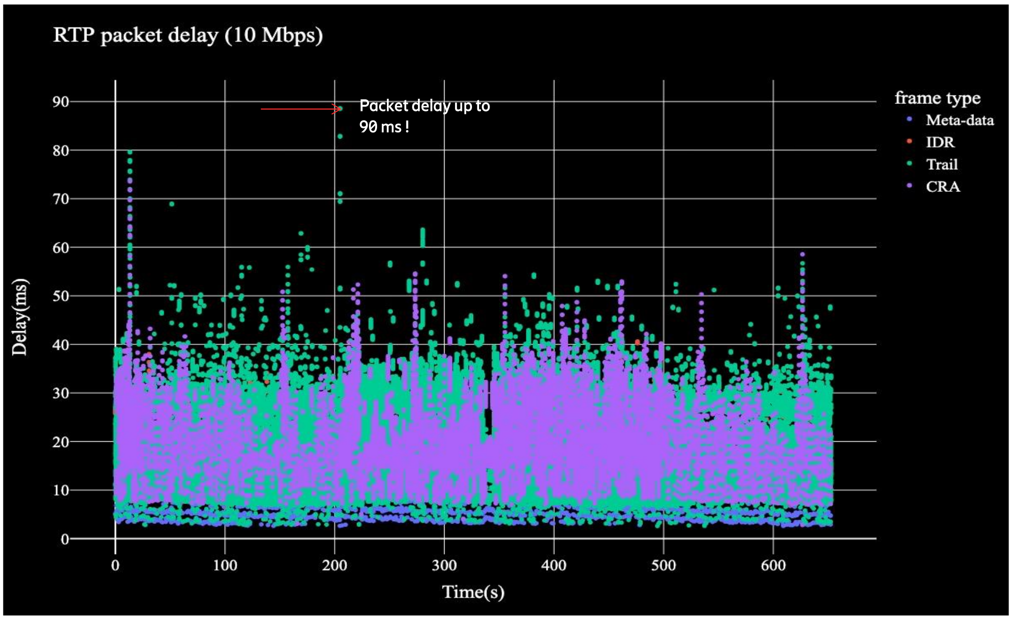


Figure 47: Time sequence of packets against delay

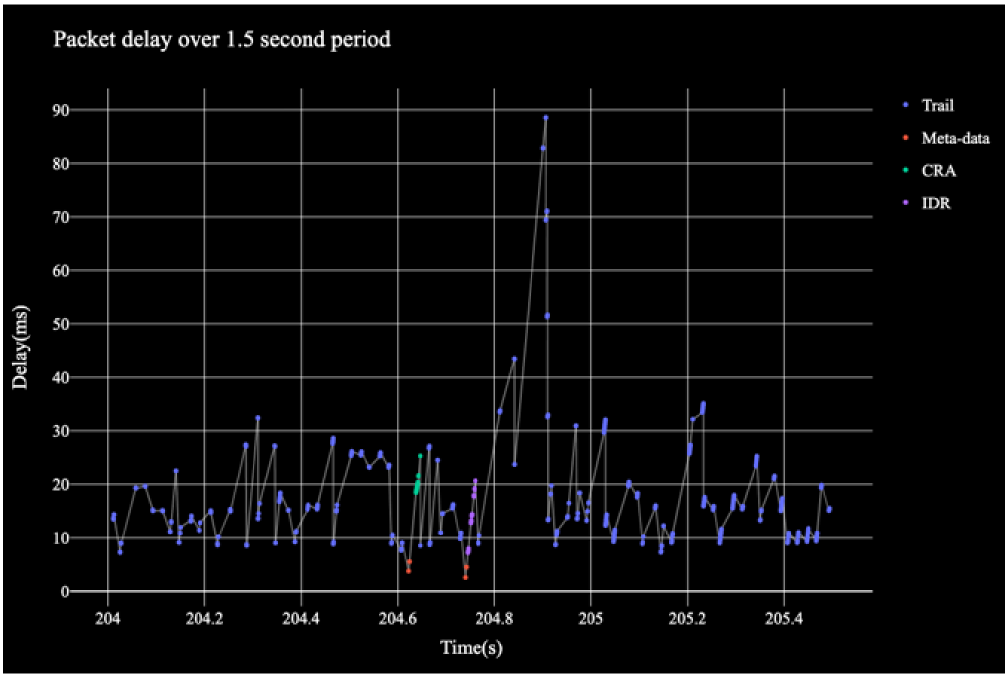


Figure 48: Packet delay for a single GOP

At Figure 48 we zoom on a single GOP to better visualize how latency builds up within a GOP. We can see that at ~204.6 s the GOP starts with the key frame (CRA) and then in the same GOP another key frame (IDR) is inserted. Afterwards the subsequent packets latency starts increasing because two large frames are sent within a GOP, so the sender buffer unexpectedly increases, and delay also increases.

Despite configuring the encoder to 50/1 GOP (i.e., 1 key frame every 1 second), Figure 49 shows that the encoder inserts a CRA frame every 1 second (which is expected) but it also inserts an IDR frame every 5 seconds with a duration of 120ms (i.e., 6 frames) after inserting the CRA frame. This encoder behavior is unexpected and could cause unnecessarily delay build-up in the stream.

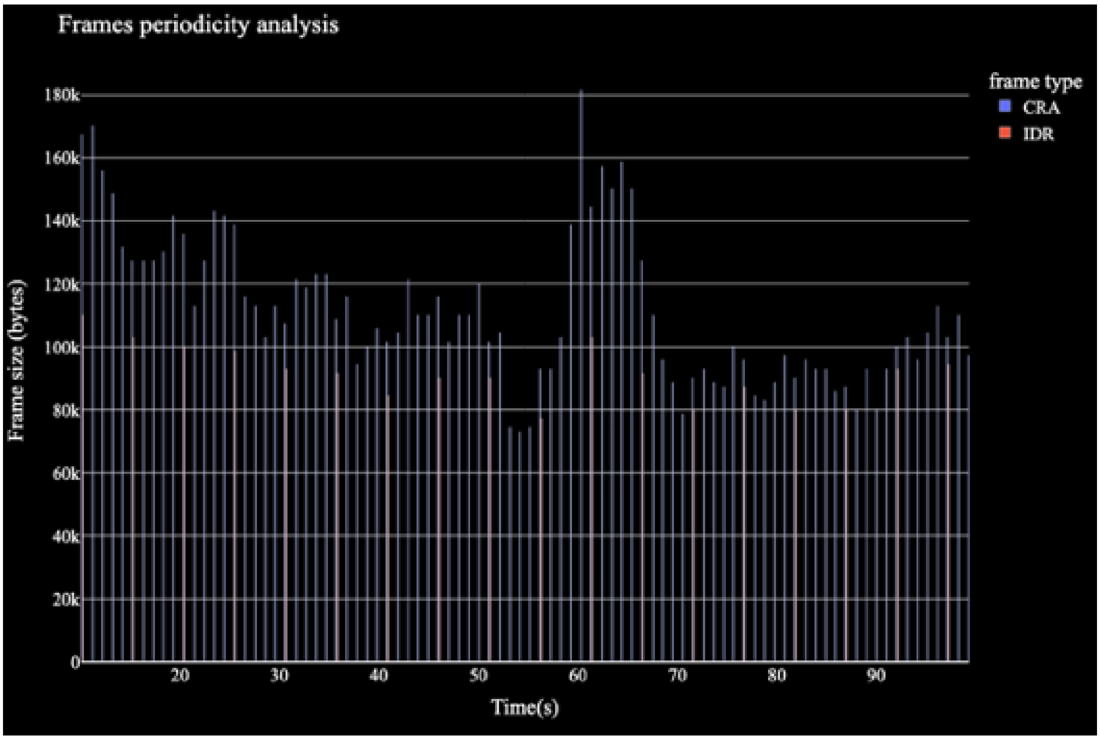


Figure 49: Key frames generation pattern

From the above section it can be concluded that the encoder configuration can cause unexpected latency spikes during the stream.

To have a closer look at the encoder behavior, the team plotted the boxplot for the frame inter-arrival duration both at the sender side (encoder) and the receiver side (Media Gateway) (i.e., After receiving frames over 5G).

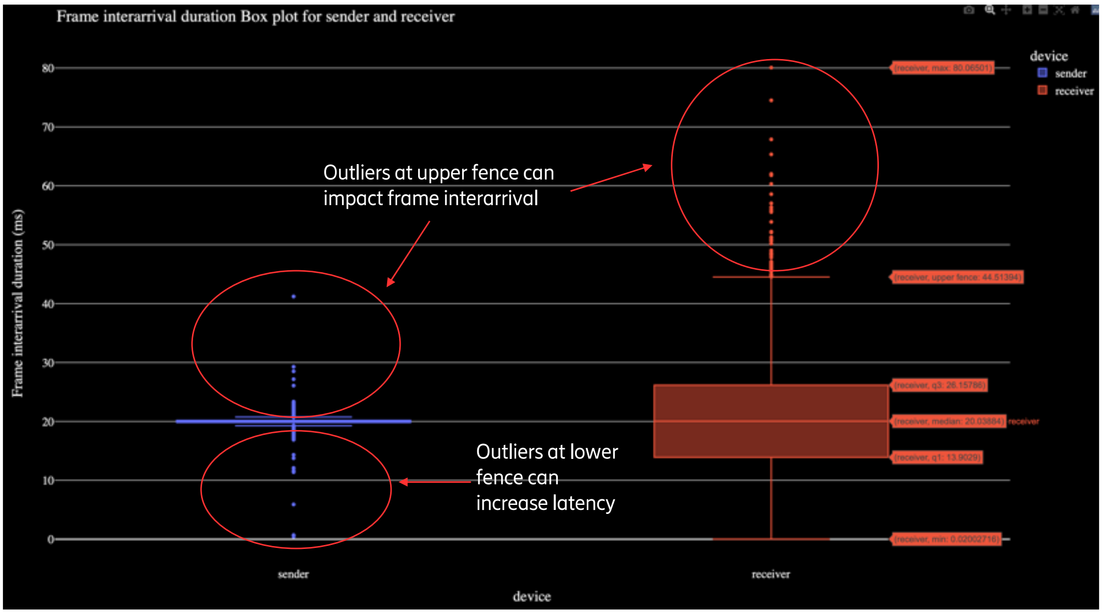


Figure 50: Boxplot for inter-arrival duration of frames both at sender and receiver

Figure 50 shows that the Inter Quantile Range (IQR) of the frame interarrival duration at the sender side is at around 20ms, which is an expected result for the encoder configuration (50 FPS). It can also be seen that the IQR at the receiver side has widened with around 6-7ms which is an acceptable value after sending the frames over the 5G radio. However, it can be noticed that the encoder has introduced some jitter in the frame generation, some frames (outliers) have been generated by the encoder at 40-80 ms and some frames are generated with interval of below 10 ms. That means that the encoder has generated 2 frames at the same time which could increase the load on the network.

- Ideal scenario:

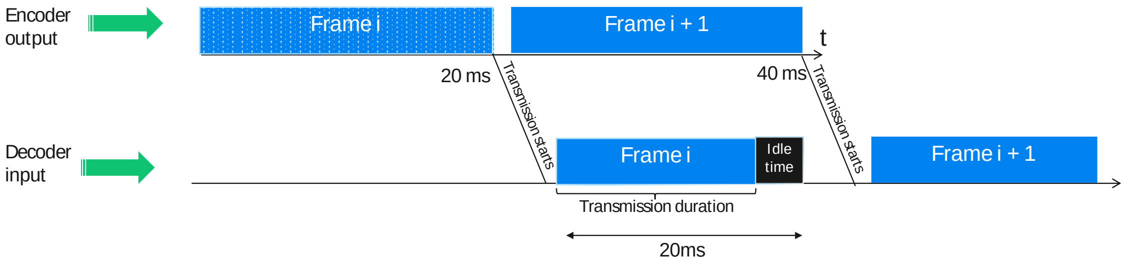


Figure 51: Ideal scenario for frame generation at the encoder

Figure 51 shows the ideal scenario on how the encoder should generate frames and how the frames are handled by the 5G network. At a 50fps, the encoder generates a full frame every 20ms and stays idle for 20 ms until the next frame arrives. During the idle 20 ms the network starts sending the RTP packets of the last generated frame. The network must send the frame before the next frame availability, or the frames will pile up in the sender buffer waiting for the transmission. If the network has higher capacity than the video bitrate, the network will send the frame and stay idle waiting for the next available frame.

- Jittery encoder scenario:

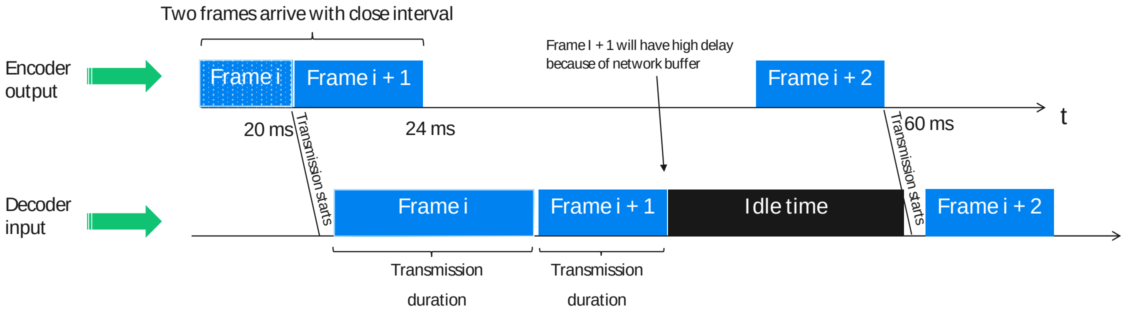


Figure 52: Jittery encoder scenario for frame generation

Figure 52 shows the case of the encoder that was discussed in the boxplot (Figure 50). In this scenario, the encoder makes “Frame I” available for transmission, and after 4ms (instead of 20ms) it also makes “Frame i+1” available. The network transmits “Frame i” while “Frame i+1” is buffered at the sender side waiting for transmission. When the frame delay is measured, “Frame i+1” will be considered as a frame that suffered from high delay because it was buffered behind “Frame i”, while in fact the "Frame i+1” has arrived earlier than it should. The same scenario is applied to “Frame i +2” which is generated later relative to “Frame i+1”, while in fact it was generated on-time.

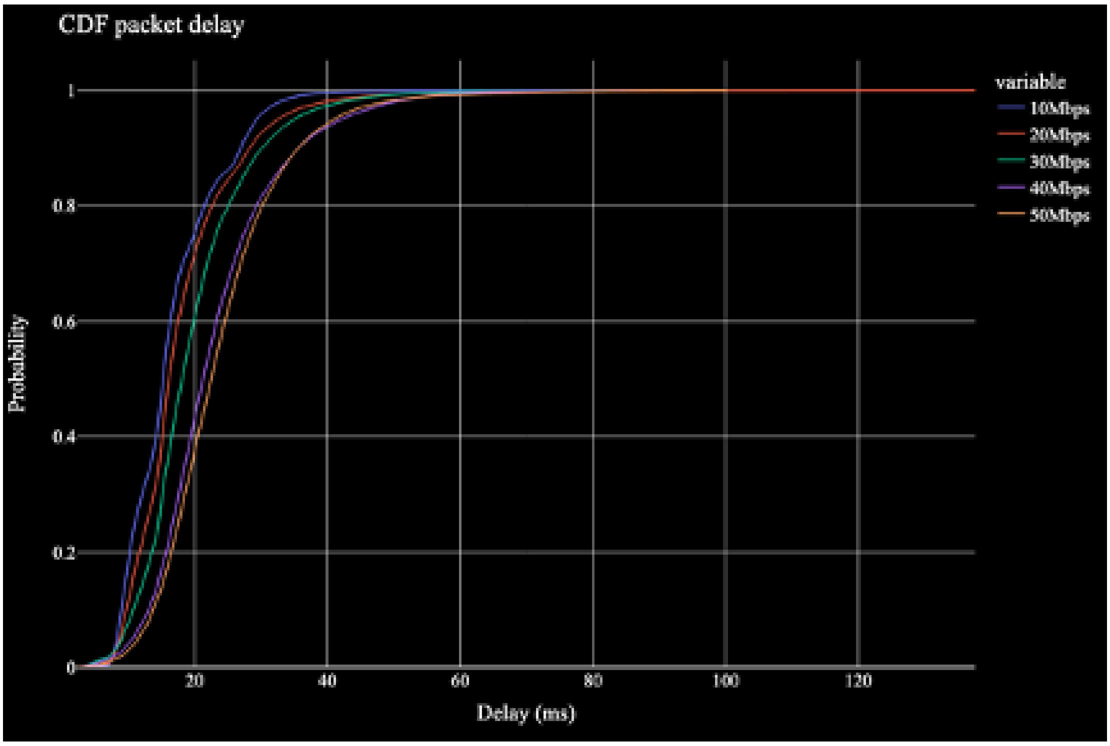


Figure 53: Packet delay CDF for all test cases

Figure 53 shows the packet delay CDF for all the test cases. We can see that the average delay increases with increasing the bitrate. However, average latency figures are still below the defined threshold.

It can be concluded that the encoder behavior has a strong impact on the measured KPIs and a professional encoder with deterministic performance can aid in the stream stability.

3.2.3.2 Packet loss and reliability analysis

During this test session we analyzed the network reliability by running a video stream for continuous 10 hours. We then measured the packet loss and the packet delay. We used a 20 Mbps bitrate stream.

Figure 54 shows that only 11 packets were lost after 5 hours of video streaming. While the rest of the session didn't show any packet latency. Those values fulfill the BER KPI defined in D2.1 [3].

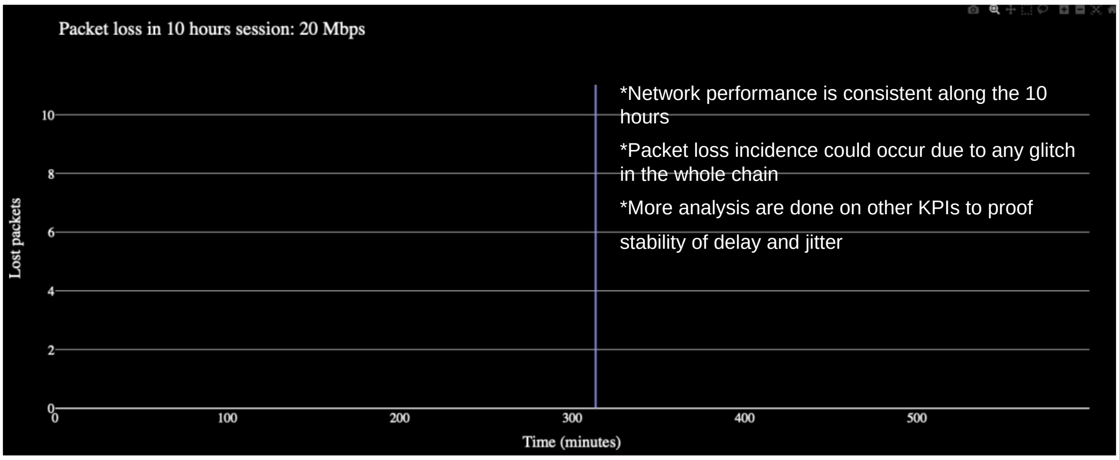


Figure 54. Packet loss analysis

3.2.4 MCR: local instance and MCR in the cloud [Local production - UC2.B.4]

This section presents the measurement results for the local instance and Master Control Room (MCR) in the cloud tests (see section 3.1.1.4).

Figure below shows the GV AMPP local instance latency measurement.

Figure 55: GV AMPP local instance latency measurement.

Figure below shows the cloud MCR latency measurement. The resulting latency observed was 47 frames with the signal format used being 1080p50.

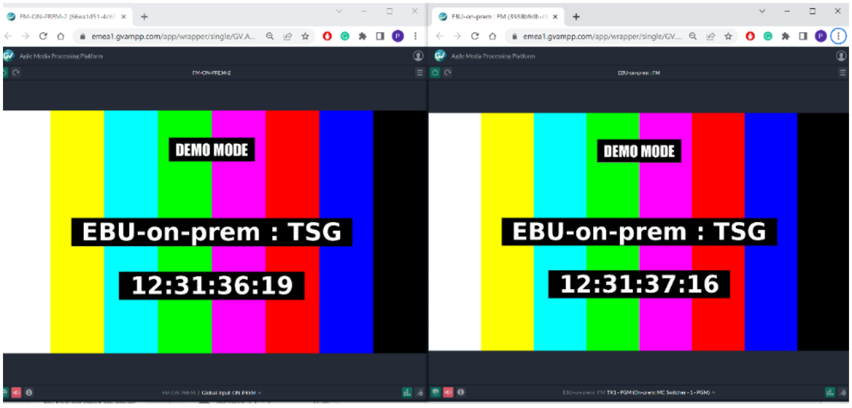


Figure 56: Latency measurement with default configuration.

This Global output application has a “Low Latency” configuration setting. With using it both ways the observed latency went down to 32 frames.

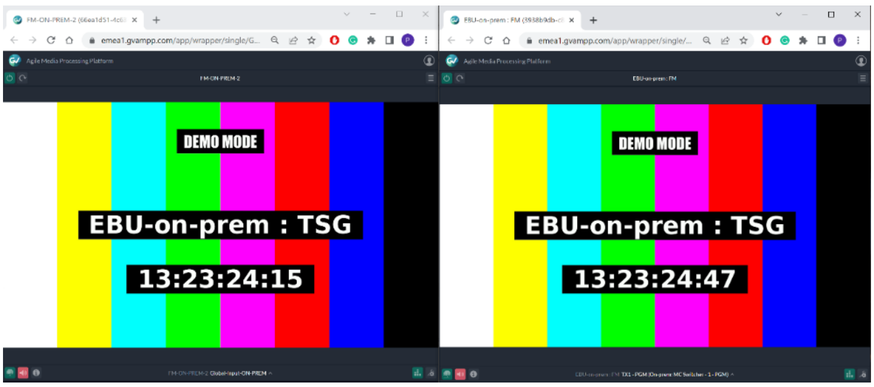


Figure 57: Latency measurement with Low Latency configuration.

3.2.5 Mobility and handover in Aachen trial network [Local production – UC2.B.5]

This section presents the measurement results of the mobility and handover tests (see section 3.1.1.5).

The team conducted 6 tests with pre-encoded video generated from the same video sequence provided by TV2. The video is streamed using RTP from the Raspberry Pi 4 to an edge server connected to the trial network's local break-out. The video duration is ~8 minutes. The mobility tests were conducted using a kick-scooter with an average speed of 15 km/h.

Table 6: Tests for the UC2 mobility and handover Aachen trial network.

Test ID	Bitrate (Mbps)	Type	FPS
Test 1	10	Mobility	50 (1 frame / 20ms)
Test 2	20	Mobility	
Test 3	50	Mobility	
Test 4	10	Stationary	
Test 5	20	Stationary	
Test 6	50	Stationary	

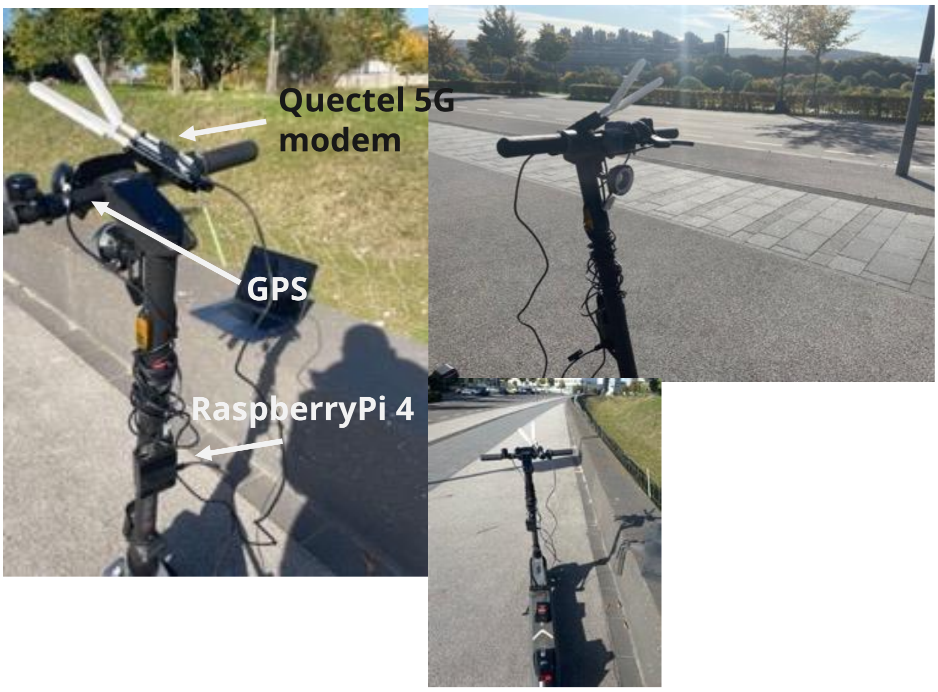


Figure 58: Scooter setup used during the mobility tests in Aachen.

3.2.5.1 Results:

3.2.5.1.1 10 Mpbs

Figure 59 shows the coverage with colors across the 3 cells. The building in the middle carries the 2 antennas (red for cell 12 and Indigo for cell 13), while the modem attached on the north of the figure to cell 11 colored Violet. The modem goes over intra-cell handover while crossing in front of the building twice. While an inter-cell handover is detected between cell 12 and 11 for a brief time at the north. Since the modem was configured to the Auto mode, it has switched from 5G-NR band to the LTE band during the test (green dots).

Figure 59: Cell coverage during test for 10 Mbps.

Figure 60 shows the throughput variation across the test. The throughput dropped in front of the building and then returned to higher values than 10Mbps. Afterward, it returned to 10Mbps. The reason for the drop is that the coverage under the building is weak because the antenna is at the top of the building and the UE is traveling very close to it. The area close to the antenna is known as the “NULL” area. The “NULL” area is more visible in the outdoor antennas because they are more directional to cover larger areas. However, this effect is minimized in indoor antennas. The UE attaches to the network via the reflections from the opposite buildings, which returns a weak signal. We can also detect in Figure 59 a spot in front of the building with no data. This spot has no data because the UE has lost connectivity because of the “NULL area” effect.

Another reason for the drop in the throughput is the UE has attached to the LTE signal at some points which has a smaller bandwidth (20 MHz) than the 5G-NR bandwidth (100 MHz).

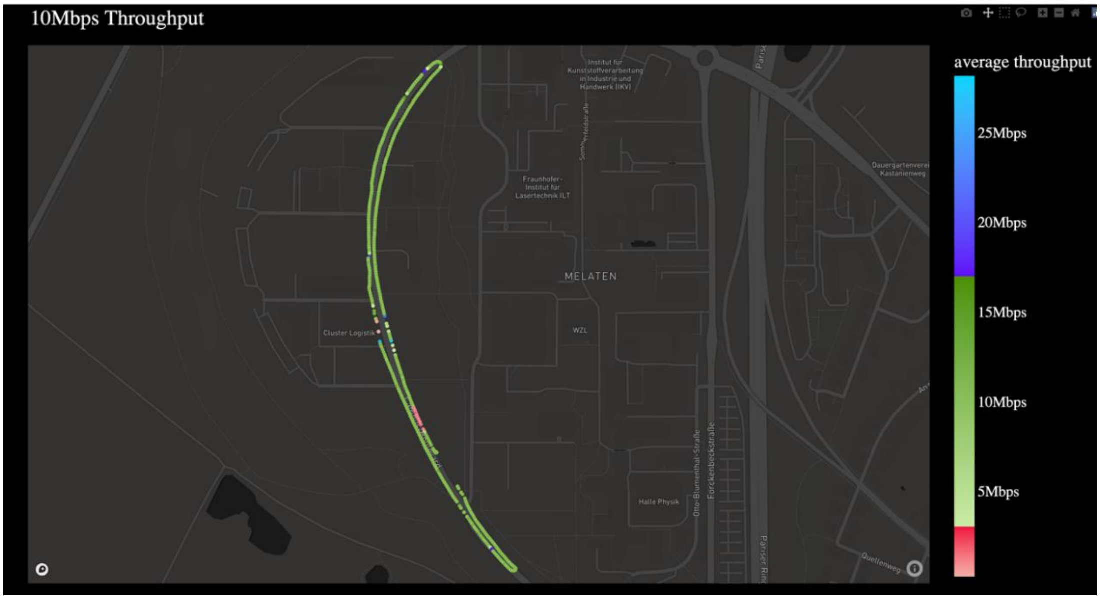


Figure 60: Throughput variation for 10 Mbps.

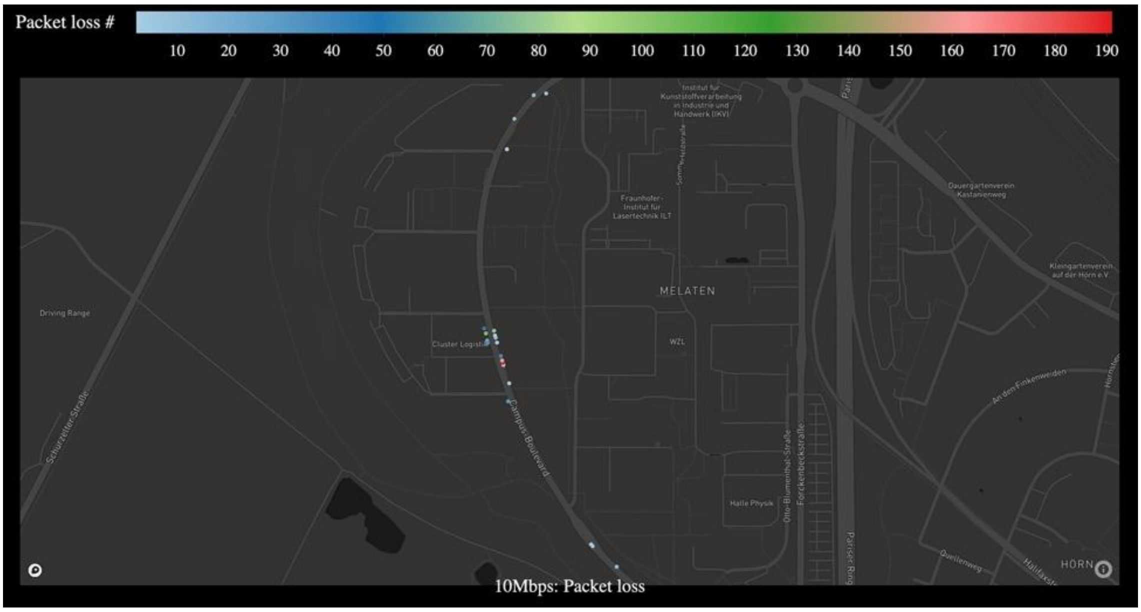


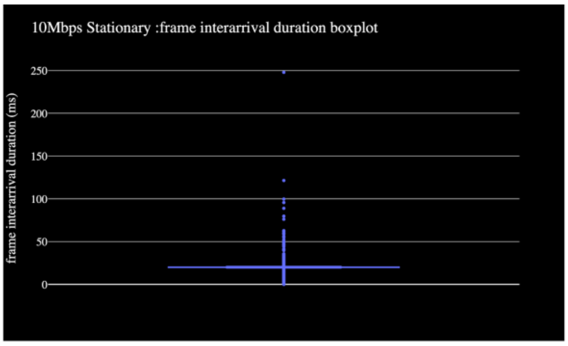
Figure 61: Packet loss at 10 Mbps.

Figure 61 is confirming the conclusion from the previous two plots. Most of the packet loss is detected under the building, while another packet loss is detected during the switch from the 5G to the LTE band. The stationary test has shown **no packet loss**.

Figure 62 shows the Frame inter-arrival duration across the map. The values remained around 30-15 ms along most of the test (green dots), while below the building we can detect values below 10ms, which is due to the fact that the modem buffers the frames and send it all once it gets a connection, resulting in high jitter due to the weak coverage caused by the “NULL area”.

Figure 62: Frame interarrival duration map for 10 Mbps.

Figure 63 shows a comparison between the frame inter-arrival duration for the stationary test and the mobility test. The stationary test has fewer outliers than the mobility test. This is because the stationary test is carried out from a good coverage area, while the mobility test had varying coverage (including the "NULL" area and the LTE area with less bandwidth). It can be concluded that radio network planning and ensuring that the media broadcasters reduce or avoid the "NULL" areas is important to guarantee a stable throughput, less jitter, and no packet loss.



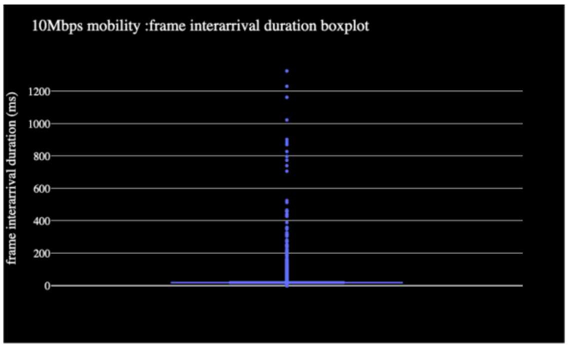


Figure 63: Frame inter-arrival duration for stationery and mobility tests.

The switching from 5G-NR to LTE has also contributed to the degradation in the quality at some points. Primarily because LTE coverage didn't have enough capacity for the 10Mbps traffic. The NSA system is introduced in 5G to help mobile operators to transition smoothly from LTE (EPC) to 5G (5GC), which means that the LTE band must be available for the control-plane. However, for 5G SNPN, it is expected that the enterprise will deploy SA system with 5G-NR directly. They will also ensure that the deployed network has enough capacity for their application.

3.2.5.1.2 20 Mbps

Figure 64 shows the coverage map for the 20 Mbps mobility test. The figure shows similar results to the 10 Mbps with the same 2 cells above the building but with a new cell at the south of the map (cell 16 in violet color). This is because the route of the test was different from the previous test (moving more towards south).

Figure 64: Cell coverage during the 20 Mbps test.

Figure 65 shows the throughput along the test. The throughput stayed at around 20 Mbps for the majority of the tests, while it dropped below 5Mbps Infront of the building.

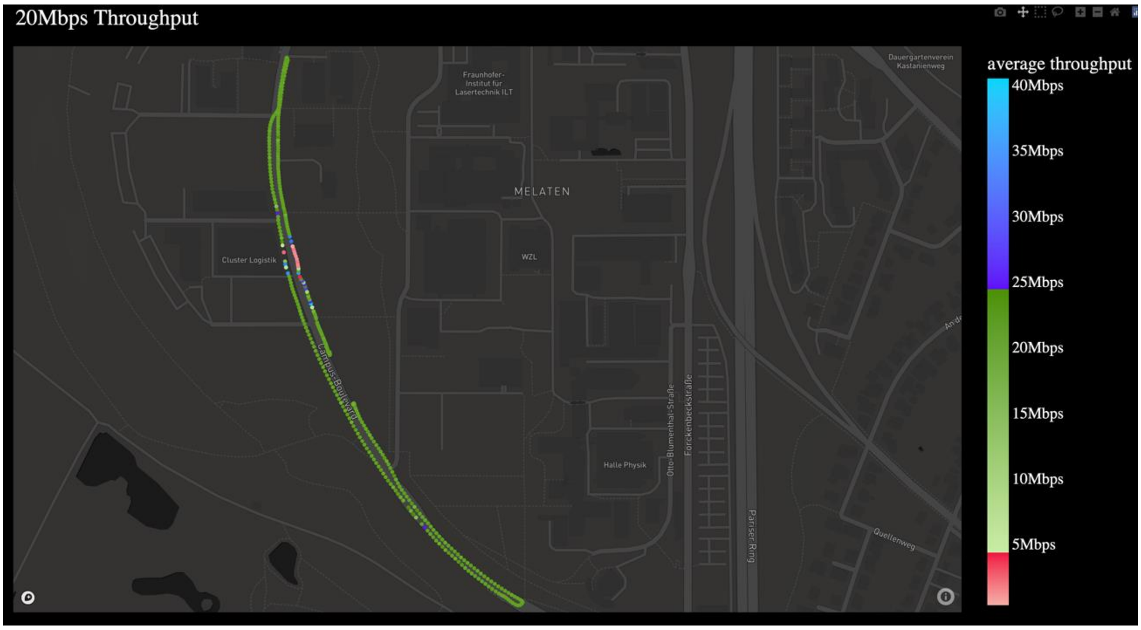


Figure 65: Throughput map for 20 Mbps.

Figure 66 shows the packet loss also in front of the building. While no packet loss is detected in the stationary test.

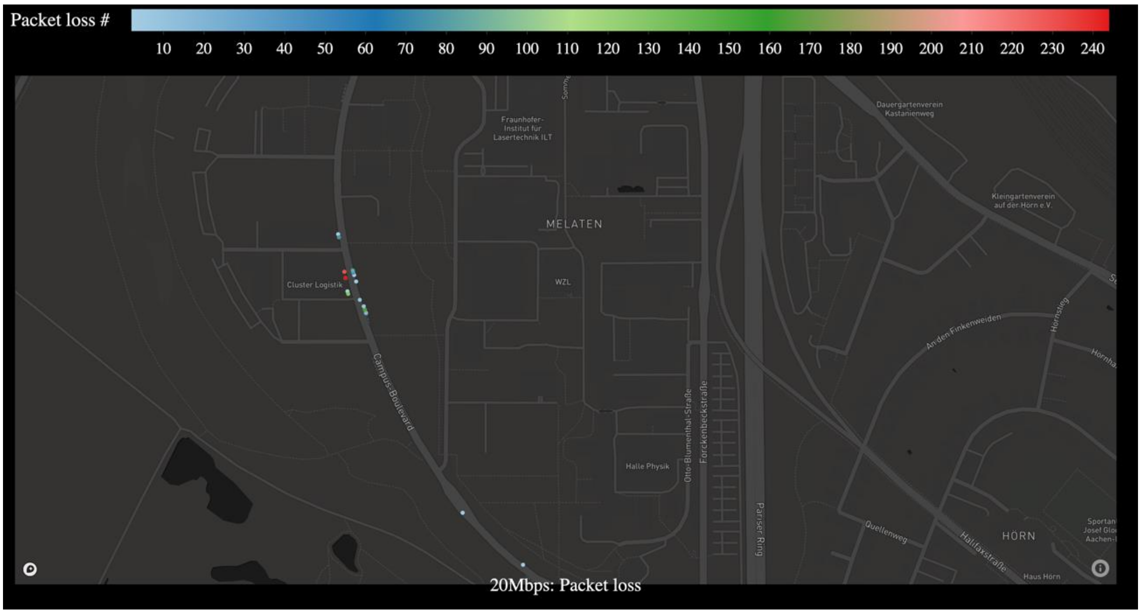


Figure 66: Packet loss for 20 Mbps test.

Figure 67 shows the frame interarrival duration which stayed around 25 ms for most of the test, while high jitter appeared in the “NULL” area in front of the building.

Figure 67: Frame inter-arrival duration map for 20 Mbps.

Figure 68 shows the box plot for the frame inter-arrival duration for both the mobility and the stationary test. The outliers for the stationary test are smaller and has less values than the mobility test.

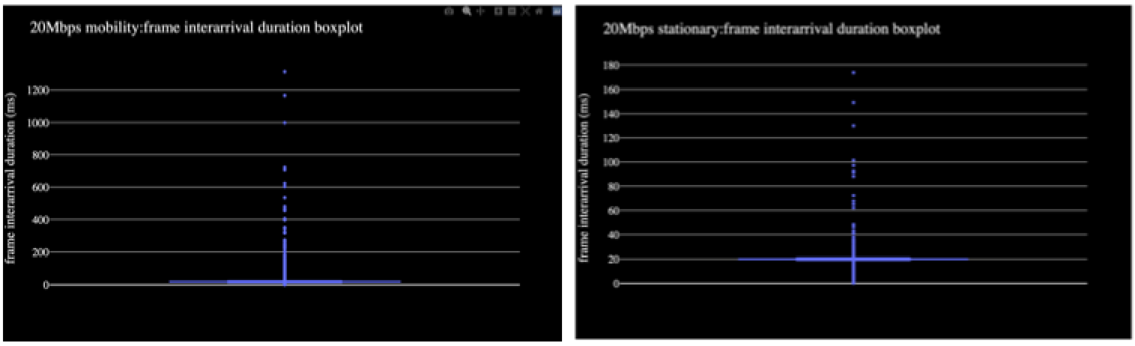


Figure 68: Frame inter-arrival duration boxplot for 20Mbps.

3.2.5.1.3 50 Mbps

Figure 69 shows the coverage of the cells during the test. We can see that all the 4 cells have been included this time.

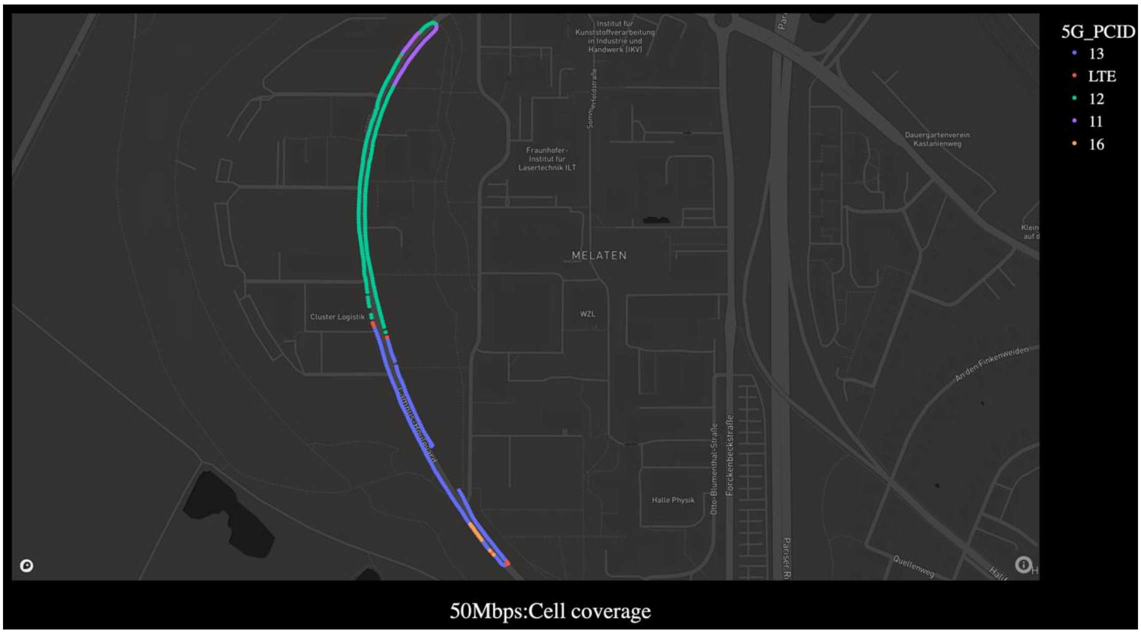


Figure 69: Coverage map for the 50 Mbps test

Figure 70 shows the throughput map for the 50 Mbps test. We can also see that the throughput is around 50 Mbps for the whole test, but it drops close to the building with the "NULL" area.

Figure 70: Throughput map for 50 Mbps.

Figure 71 shows the packet loss which is also concentrated close to the "NULL" area. **No packet loss is detected in the stationary test.**

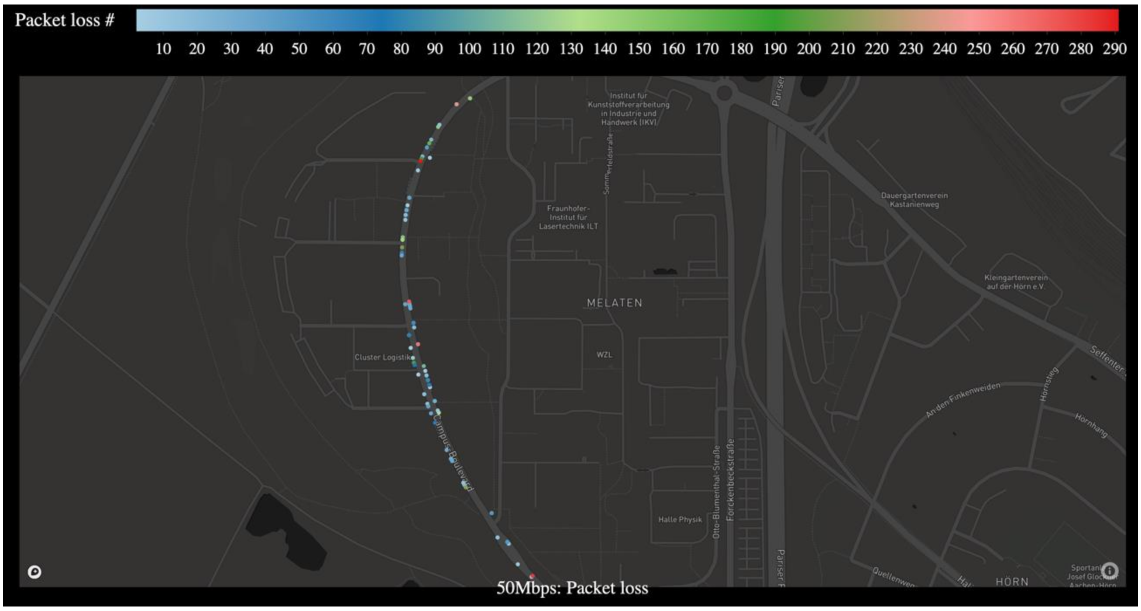


Figure 71: Packet loss at 50 Mbps.

Figure 72 shows the frame inter-arrival duration for the 50 Mbps. We can see that most of the frames stayed around 20 ms.

Figure 72: Frame inter-arrival duration map for 50 Mbps.

Figure 73 shows the frame inter-arrival duration boxplot for both stationary and mobility tests. The stationary has fewer outliers with less values.

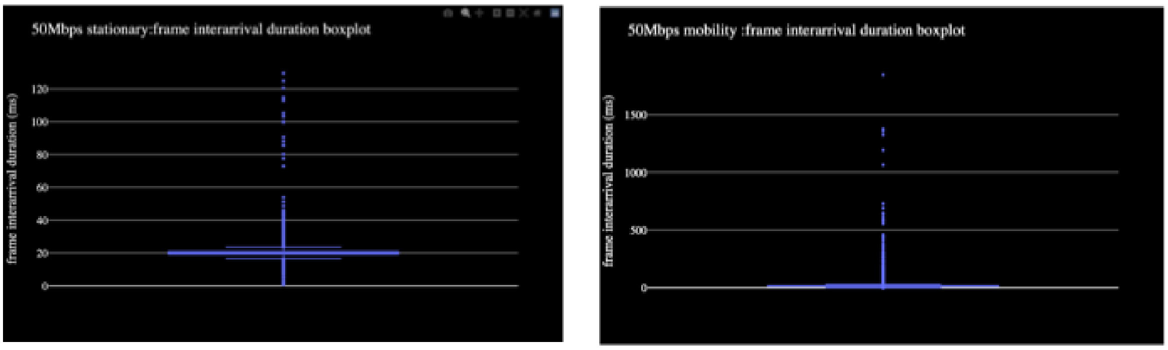


Figure 73: Frame inter-arrival duration boxplot for 50 Mbps.

3.2.6 Tests with Low Latency HEVC encoding (GDR) [UC2.B.6]

This section presents the results for the tests with Low Latency HEVC encoding (GDR) (see section 3.1.1.6).

Figure 74 evaluates the timing, at which an HEVC encoder provides individual GDR slices. The encoded vide sequence was captured at the sender side with tcpdump in a PCAP file. The test sequence in the PCAP file is 300s duration encoded with 50 fps HEVC at 50Mbps. Here, the encoding of a slice takes around 1 ms each.

The GST pipline for encoding the video sequence is :

```
omxh265enc num-slices=8 control-rate=low-latency gop-mode=low-delay-p target-bitrate=${brate}000 cpb-size=500 gdr-mode=horizontal initial-delay=250 periodicity-idr=240 filler-data=0 prefetch-buffer=true ! video/x-h265, alignment=nal ! …
```

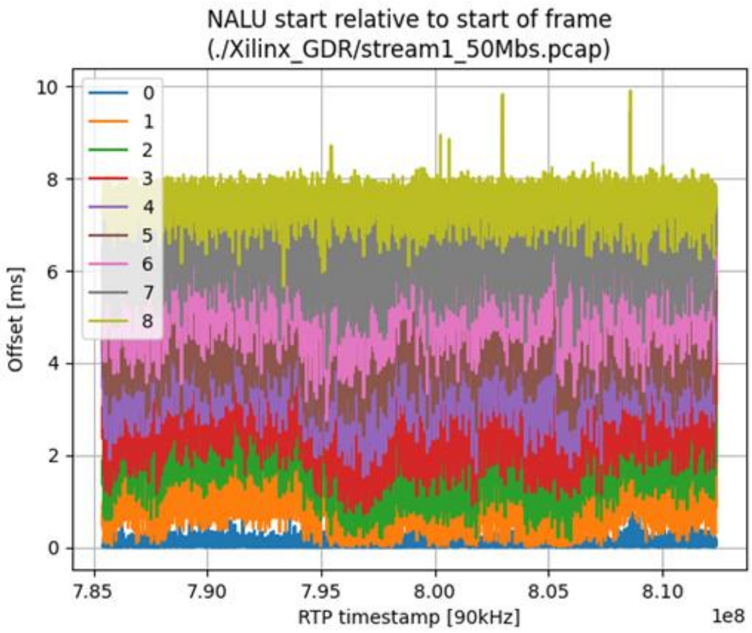


Figure 74: HEVC test sequence with GDR.

The histogram of the frame duration is depicted in Figure 75. It should that majority of HEVC slices are made available by the HEVC encoder 8ms after the first slice of the frame.

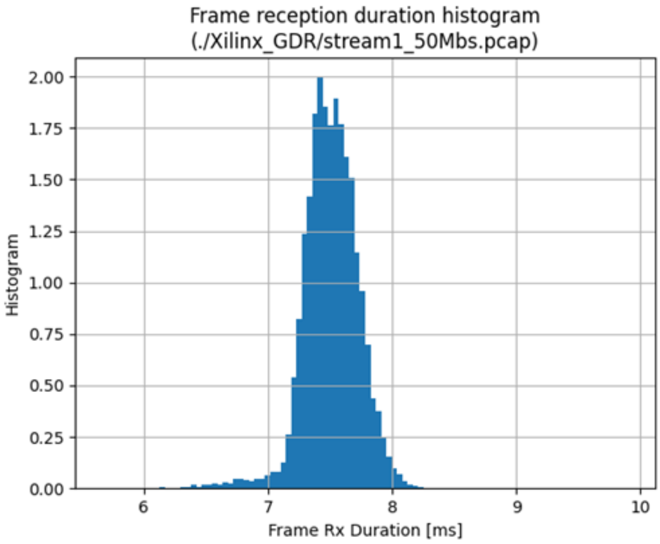


Figure 75: Frame Duration Histogram at the device (sender) side

The following figures illustrates the receiver side, i.e. how the video stream is received at the Media Gateway.

The 5G Network tests are conducted using a 5G Standalone (SA) setup, leveraging the band n78 (Industry Band) with a 100MHz carrier. The TDD patterns is DDSU. The 5G Network is deployed indoors and the Radio Antennas are mounted in the ceiling.

Figure 76 shows the effect of the 5G network on the video slice distribution. Again, the time offset between the reception of the first slice and subsequent slices of one frame is shown.

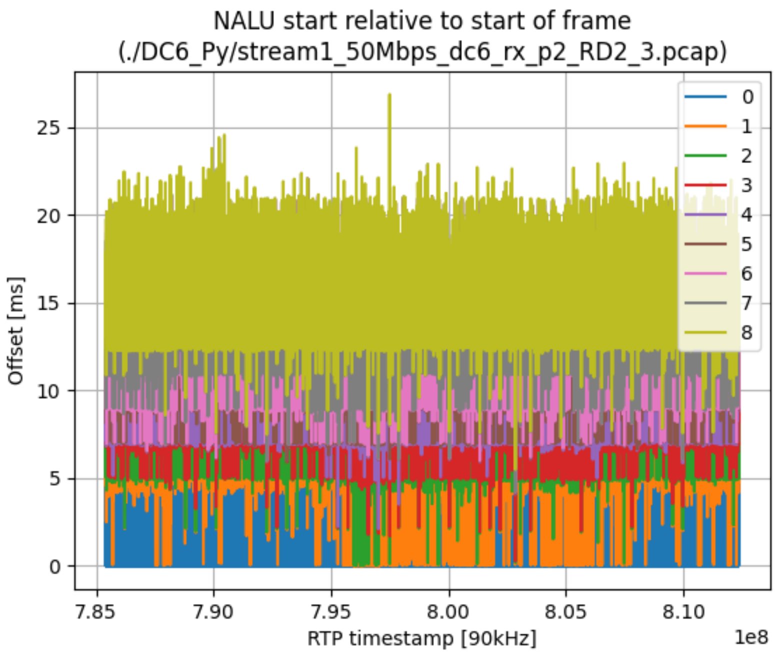


Figure 76: NALU delay at receiver side

Figure 77 depicts the Relative Transit Time of each frame and the Time Stamped Delay Factor (TS-DF). It shows that the receive only needs to have a de-jitter buffer of around 25ms.

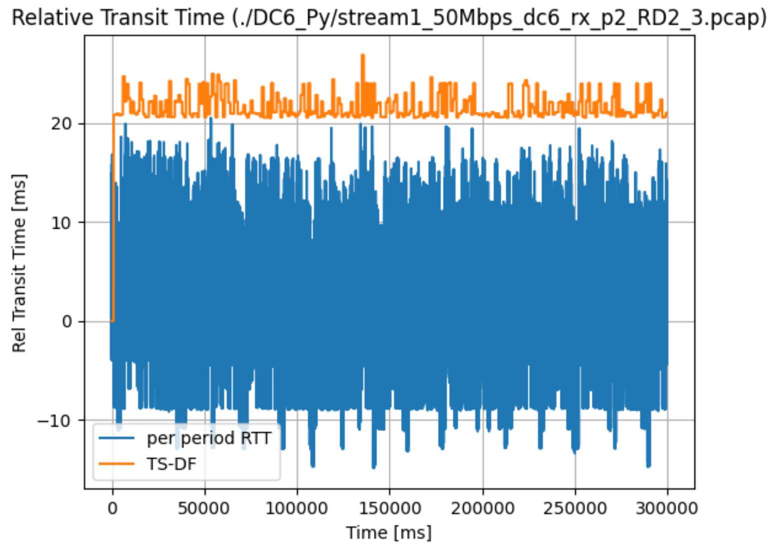


Figure 77: Relative Transmit Time and TS-DF (at receiver side)

Figure 78 illustrates the Frame Reception Duration (over time on the left and as histogram on the right). It shows that the variation is frame receptions is low, leading to a consistently low receiver buffer.

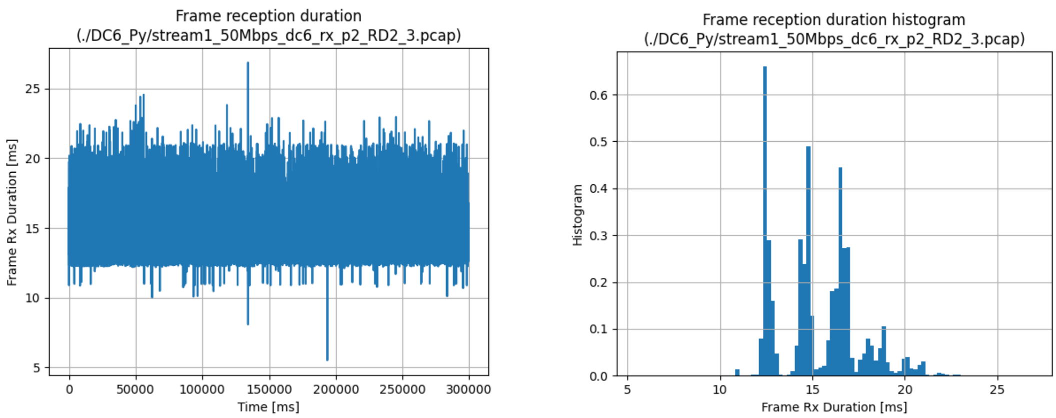


Figure 78: Frame Reception Duration (at receiver side)

3.2.7 Remote production tests in Aachen [Remote production – UC2.B.7]

This section presents the results from the tests and trials performed at Ericsson 5G upgraded lab in Aachen (see section 3.1.1.7). Their analysis was finished after deliverable D4.2 [6] was submitted.

These tests focused on the live professional video uplink transmission (‘contribution”) under various conditions and their combinations. Mainly for: single and dual bonded modems, using slices, at different simulated uplink congestion levels, with a commercial 5G NSA network.

Reference made to Sierrra Wireless or Fivecomm modems means the use of that modem by the LiveU LU800 video encoder-transmitter.

Expected network behaviours – shown in the graphs below:

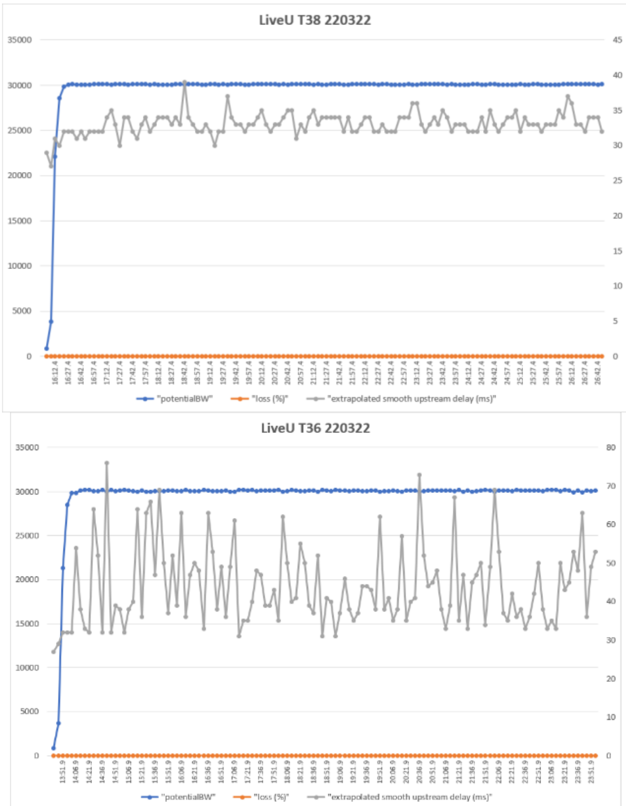
- GP[3] slice provides the guaranteed UL BW and latency when no other device is in use on any slice.
- Congesting eMBB[4] slice does not impact UL BW on the “GP” slice.
- Bonding adapts its live video transmission to changing conditions in any of the links.
- Bonding (by the LU800-LU2000) of different slices or networks, mitigates problems.

Unexpected network behaviours – shown in the graphs below:

- Congesting eMBB slice triples the UL latency on the “GP” slice
- Additional traffic or link on any slice, even eMBB slice, might impact the latency of the eMBB traffic – both absolute and jitter
- Seemingly, resources are reserved on the «GP» slice even when it’s empty and the resources are needed by the eMBB slice
- Commercial network and NPN cabled-in 5G SA lab network had very similar UL production (few tests)

3.2.7.1 Results for single feed, single modem, 5G with congestion, slices - LU800 with router or Fivecomm modem on GP slice (50% and 90% congestion on eMBB slice)

Results are shown in Figure 79.The UL BW of the 5G GP slice is not impacted, same as seen in other tests. The UL latency in the GP slice seems to remain more or less in the same range yet is adversely affected by the eMBB traffic in being much more erratic.



[3] GP slice: Guaranteed Performance, UL-oriented slice

[4] eMBB slice: best effort slice

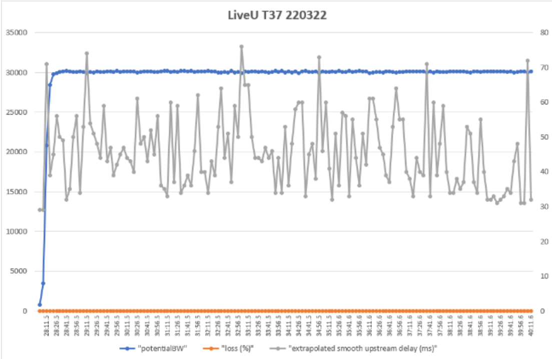


Figure 79. LU800 Bitrate, Loss,Latency with router or Fivecomm modem on GP slice - (a) No congestion, (b) 50 % congestion on eMBB, (c) 90% congestion on eMBB, from top to bottom

3.2.7.2 ***Results for single feed, single modem, 5G with congestion, slices – single Fivecomm modem on eMBB slice*** *(50% and 90% congestion on GP slice).*

Results are shown in Figure 80. The UL BW of the 5G eMBB is impacted slightly when congested 5G and more when 90% congested. The UL latency in the eMBB slice is high and unstable. Single Fivecomm modem, single video feed, LU80 max cap @ 30Mbps, LU800 on the eMBb/best effort slice and the network load is on the media/”guaranteed” slice.

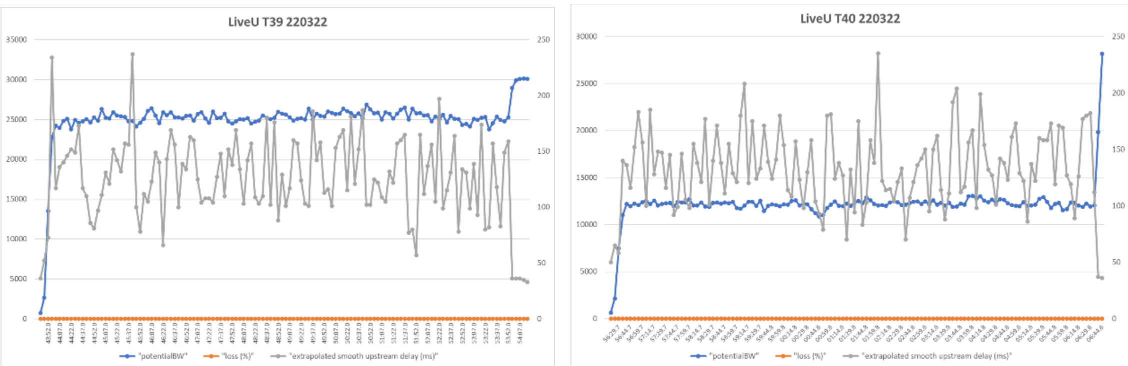


Figure 80: Bitrate, Loss, Latency of single modem on eMBB @50% slice load (left) and 90% slice load (right).

3.2.7.3 *Results for single modem eMBB with 50% congestion – Fivecomm modem with 50% UL congestion both on the eMBB slice.*

Results are shown in **Error! Reference source not found.**.Stable E2E transmission at reduced rate (50 Mbps vs. 60 Mbps) although the GP is empty. The 5G network adversely impacts the latency of at least one of the eMBB transmission (we do not measure the congestion).

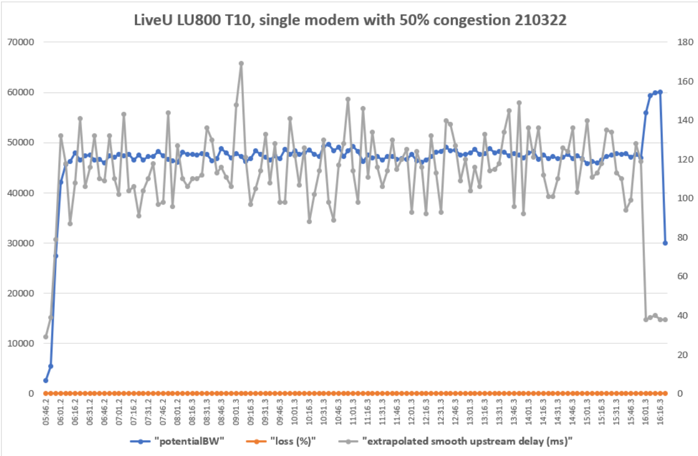


Figure 81. Sierra modem with 50% UL congestion both on the eMBB slice

3.2.7.4 Results for 2 modems basic bonding – No load, bonding the embedded Sierra Wireless + Fivecomm both on the GP slice

Results are shown in Figure 82. Stable maximum E2E transmission, evenly split between the two modems, low transmission latency, low loss rate.

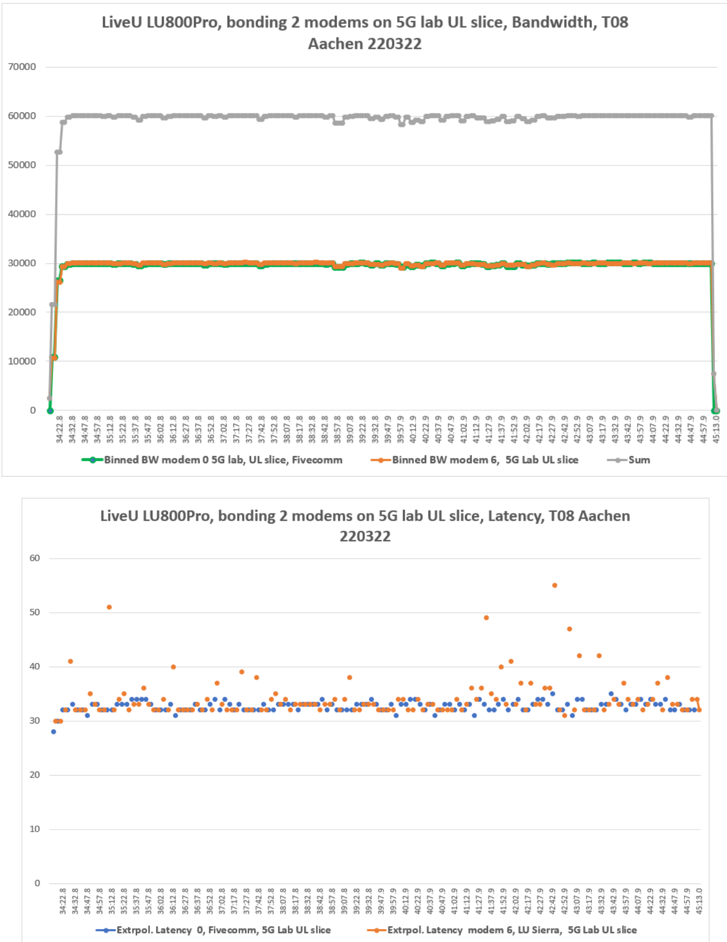


Figure 82. Results for 2 modems basic bonding over the same "GP" slice (Bitrate on the left and latency on the right)

3.2.7.5 Results for 2 modems bonding two slices – No load, bonding the embedded on the GP slice + Router on eMBB slice

Results are shown in Figure 83 and Figure 84.The 5G network adversely impacts the latency of the eMBB slice when serving first modem on GP slice. Still, bonding keeps a stable maximum E2E transmission, evenly split between the two modems. When performance of one modem drops a little, the LU800 increases the second in a mirror way.

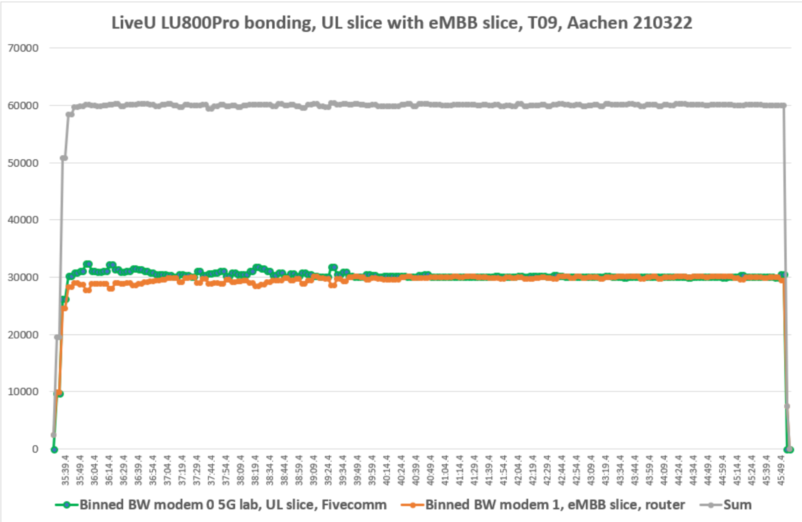


Figure 83: Bitrate of bonding 2 modems, one on media/"guaranteed" slice and one on eMBB/best effort slice

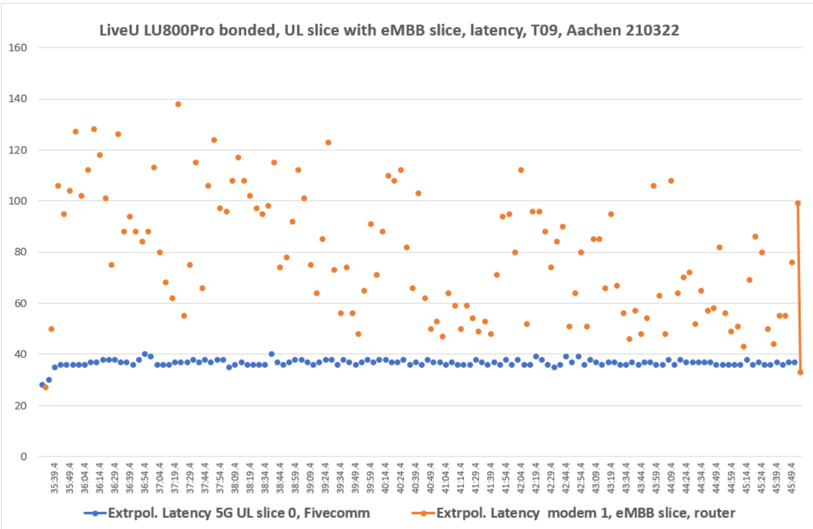


Figure 84: Latency of bonding 2 modems, one on media/"guaranteed" slice and one on eMBB/best effort slice

3.2.7.6 Results for 2 modems bonding - 5G and commercial with congestion – bonding Fivecomm modem on GP slice + SiW on commercial network, 50% congestion via eMBB

Results are shown in Figure 85 and Figure 86. The UL BW of the 5G GP slice is not impacted, same as seen in other tests. The UL latency (blue, right graph) of the 5G GP slice seems to increase and resemble that of the commercial. Bonding keeps a stable maximum E2D transmission, evenly split between the two modems.

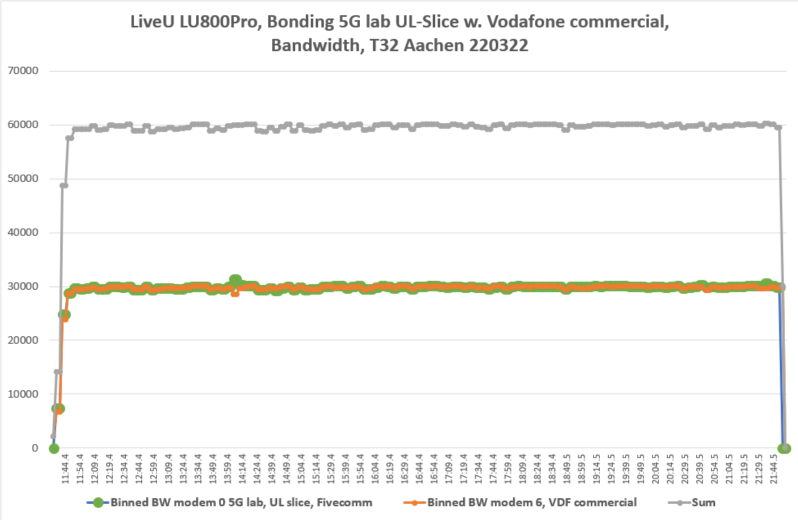


Figure 85. Bitrate of bonding 2 modems, one on media/"guaranteed" slice and one on commercial 4G network and 50% eMBB slice load

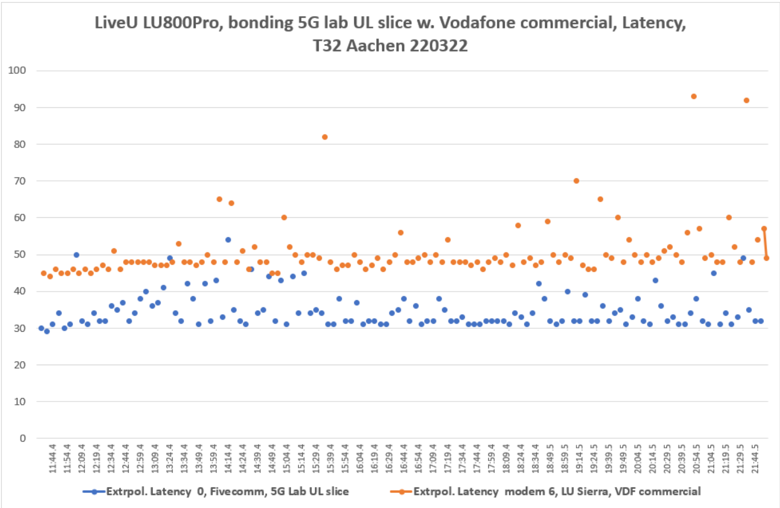


Figure 86: Latency of bonding 2 modems, one on media/"guaranteed" slice and one on commercial network and 50% eMBB slice load.

3.2.8 Video quality tests

In addition to what has been described so far, during one of the test sessions held in Aachen, RAI performed some video quality tests. The objective of these tests was to evaluate under what network conditions the video stream would be still usable for contribution/production purposes.

The test sequence, transmitted from Aachen multiple times under different network load conditions, was recorded in Turin in high quality (ProRes HQ) for subsequent quality analysis.

RAI asked to seven expert viewers to evaluate the quality of the video with respect to the original master clip, by giving a mark between 1 and 5. Clips were evaluated entirely, not scene-by-scene.

The network setup in Aachen labs is described below. Two end-user devices were used in this test: a LiveU LU800 encoder, connected to a modem, and a network congestor, connected to another different modem. Moreover, two network slices were configured: a

«Media slice» (with priority); and a «eMBB slice» (in best effort). At RX side, a single radio unit was used. Both modems were on the same cells competing for an available bandwidth of 100 Mbps.

Tests were done with different percentage of network congestion (50%, 90%), and with different mapping of LiveU encoder and congestor to different slices. Table 7 reports the specific network configuration used during each tests.

Table 7. Network configurations Remote Scenario UC2 - Aachen tests.

	UL Media slice				**eMBB slice**			
Test ID	Modem 1		Modem 2		Modem 1		Modem 2	
	User	Type	User	Type	User	Type	User	Type
T34-T35	Congestor @50%, 90%	Fivecomm or router	LU800	Sierra or Fivecomm or Router				
T36-T37	LU800	Fivecomm or router			Congestor @50%, 90%	Sierra or Fivecomm or Router		
T38	Congestor @0%							
T39-T40	Congestor @50%, 90%	Fivecomm or router			LU800	Sierra or Fivecomm or Router		

Test results are reported inside the following graphs (Figure *87*); test scores must be referred to Table 8, taken from ITU-R BT.500-13.

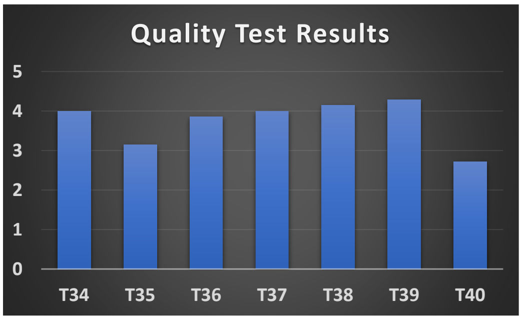


Figure 87: Quality test results from RAI *in* Aachen *[UC2-Remote Production].*

Table 8. ITU-R Quality and impairment scales – reference grade scale video quality evaluation

Five-grade scale	
Quality	Impairment
5 - Excellent	5 - Imperceptible
4 - Good	4 - Perceptible, but not annoying
3 - Fair	3 - Slightly annoying
2 - Poor	2 - Annoying
1 - Bad	1 - Very annoying

In general, results seem coherent with expectations, with minor exceptions.

In most cases, test results look very similar (T34, T36, T37, T38, T39). This is probably since, in this specific setup, 50% of congestion seems to be too little to make any impact to the video quality.

However, some quality degradation was noted during T35 and T40. This is somehow expected, since these tests were performed assuming the worst network condition, which specifically means absence of privileged slice for media and a high level of congestion.

Even if the average score is good for most of the performed tests, in few cases video artefacts were visible.

3.2.9 Trials at UPV [Remote production – UC2.B.8]

This section presents the results from trials at UPV (see section 3.1.1.8). These tests were designed to assess the performance of the network as well as the performance of the LU equipment in different scenarios and configurations. The NSA 5G commercial network is provided by Orange Spain, so no possible configuration of its parameters is possible. The Amarisoft 5G SA is an NPN lab deployment, so it has the possibility of configuring RAN parameters. There is a big number of variables that can be changed in this solution, but for compatibility reasons with the LU equipment, only basic parameters were changed, such as gNB bandwidth and bonding capabilities. The internal modem of the LU800 unit was tested against the Fivecomm modem to compare their performance.

Before presenting the results, it is worth indicating how to read the results.

- Transmitted UL BW is on the left axis
  - Usually the **blue/green** line
- UL Latency and loss rate are on the right axis
  - Usually the **grey** and **orange** lines respectively
- The max of the LU800Pro for single feed is ~30Mbps
- The max of the LU800Pro transmitting 4 HD feeds simulatenously is ~60Mbps

Following are the main results:

Obtained, expected network behaviours:

- Higher UL resources (with no load) enable more stable transmission also on single modem
- In real commercial network, outdoors, each link is unstable and max BW is not achieved
- Under same conditions, bonding (by the LU800Pro-LU2000) stabilizes the total UL BW while the network behaviour is erratic

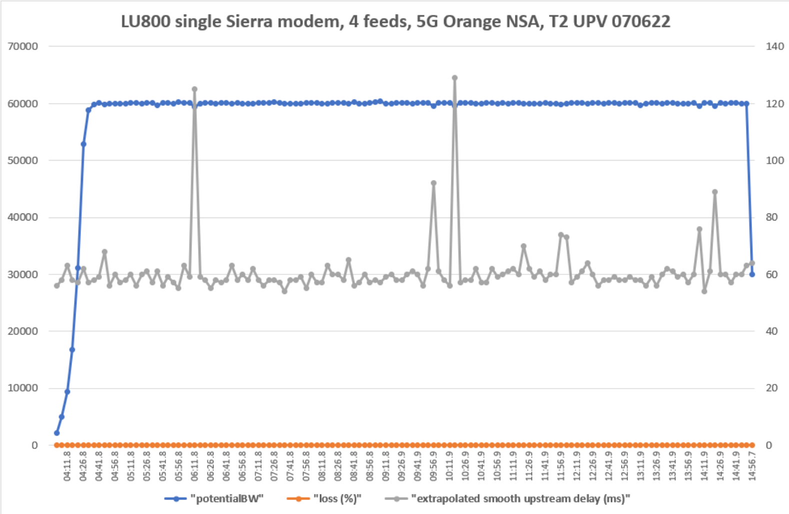
Obtained, unexpected network behaviours:

- Single link performance at 20MHz channel BW was more stable than at 100MHz
- Latencies and their instability increase significantly when UL resources are insufficient (from ~60 msec to >200msec)

Some of the obtained results are presented below:

3.2.9.1 Results for basic, single modem – Orange commercial 5G NSA, 60 Mbps maximum at LU800 without bonding

Results are shown in Figure 88 and Figure 89. Stable maximum E2E transmission; low, stable UL latency with occasional reasonable increase. The LU800Pro embedded Sierra Wireless modem performs well (top) and the external Fivecomm modem had issues (bottom) Figure 88.



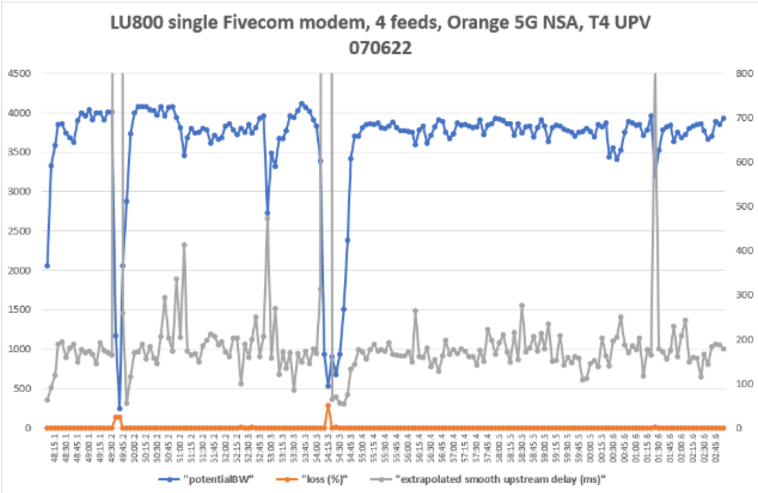


Figure 88: Benchmark single Sierra Wireless modem for LU-UPV trial.

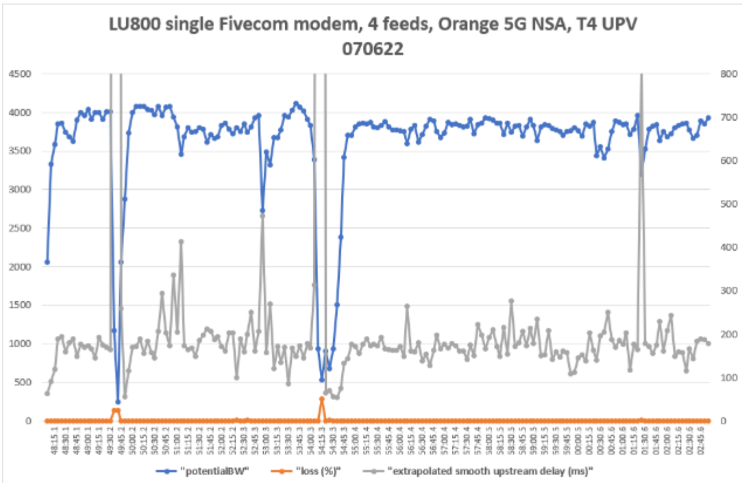


Figure 89: Benchmark single Fivecomm modem for LU-UPV trial

3.2.9.2 Results for basic, 2 modems bonded – Orange commercial 5G NSA, 60 Mbps maximum at LU800, 2 modems bonded

Results are shown in Figure 90, Figure 91, Figure 92 and Figure 93. Form first to last, stable maximum E2E transmission; low, stable UL latency with occasional reasonable increase. Bonded effectively splits the encoded video per each modem/link momentary performance. The LU800Pro embedded Sierra Wireless modem performs well. The external Fivecomm modem had issues.

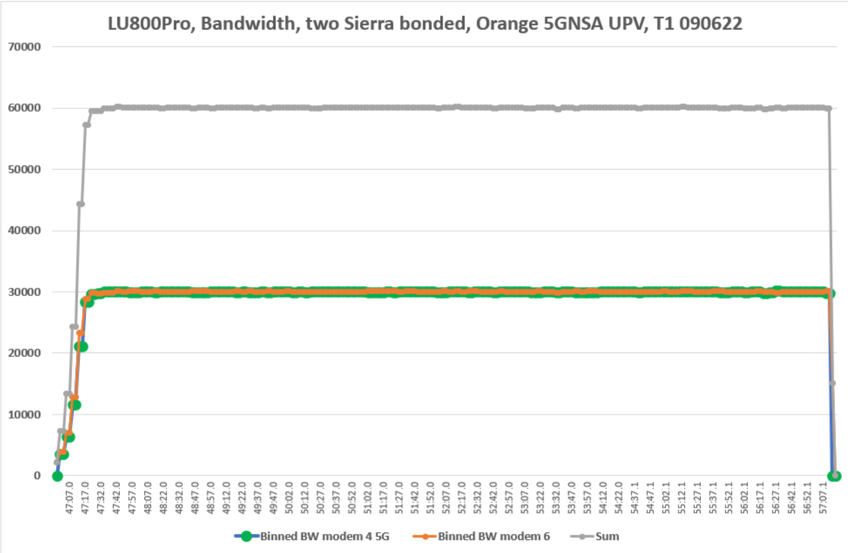


Figure 90: Benchmark, Bitrate of bonding 2 Sierra modems, NSA, LU-UPV trial

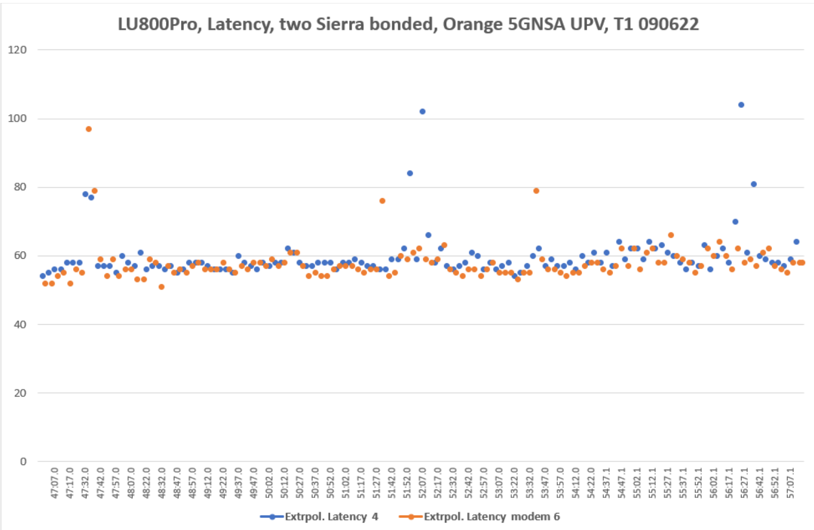


Figure 91: Benchmark, Latency of bonding 2 Sierra modems, NSA, LU-UPV trial

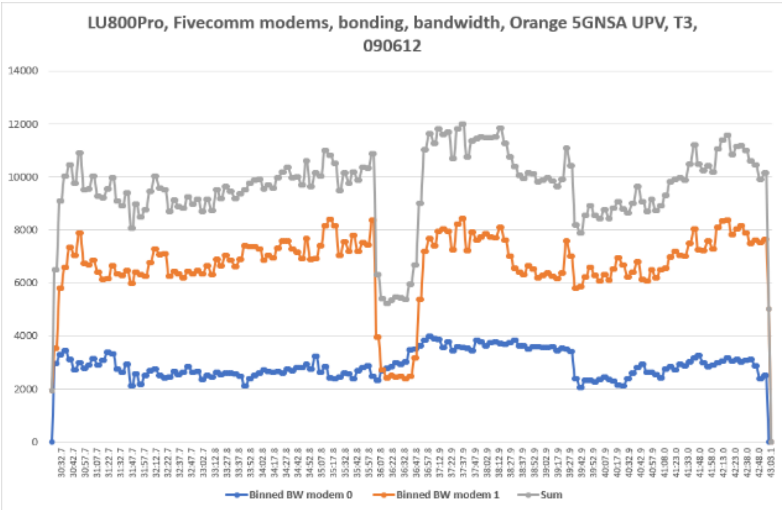


Figure 92: Benchmark, Bitrate of bonding 2 Fivecomm modems, NSA, LU-UPV trial.

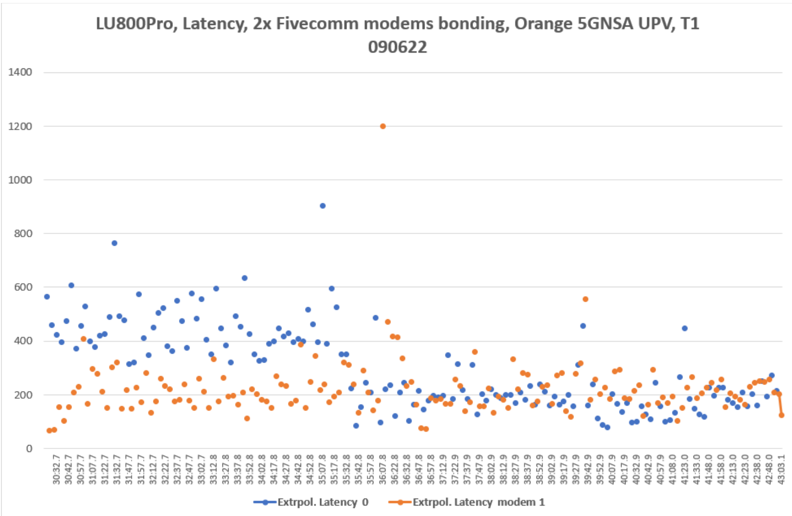


Figure 93: Benchmark, Latency of bonding 2 Fivecomm modems, NSA, LU-UPV trial

3.2.9.3 Results for single and bonded, low UL – UPV lab 5G-SA, 60 Mbps maximum at LU800

Network configuration: 30 kHz SCS, 100 MHz BW, 30/70% of UL/DL pattern

Results are shown in Figure 94, Figure 95 and Figure 96. From first to last, single modem unstable and not-maxed BW and high unstable latency. Bonding stabilizes each modem and the total BW at higher overall, while UL latency remains network-erratic.

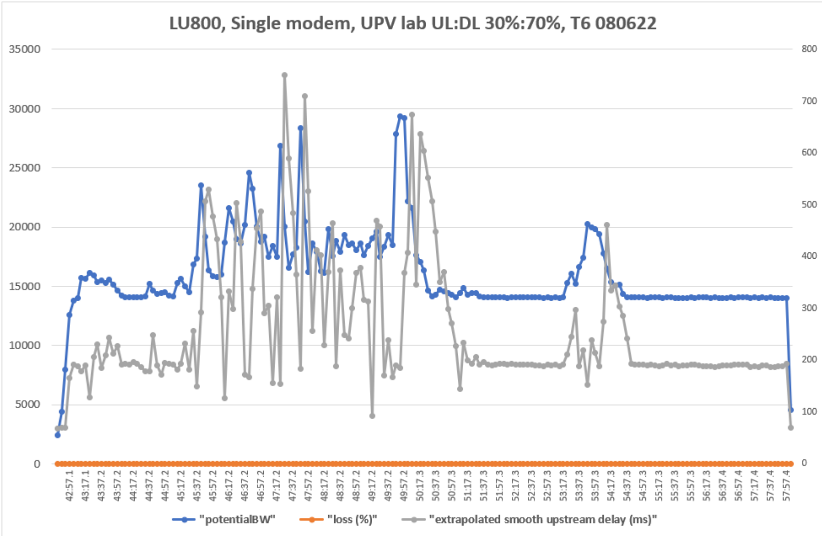


Figure 94: Bitrate, loss, latency of single modem, 30:70 UL:DL, SA, LU-UPV trial

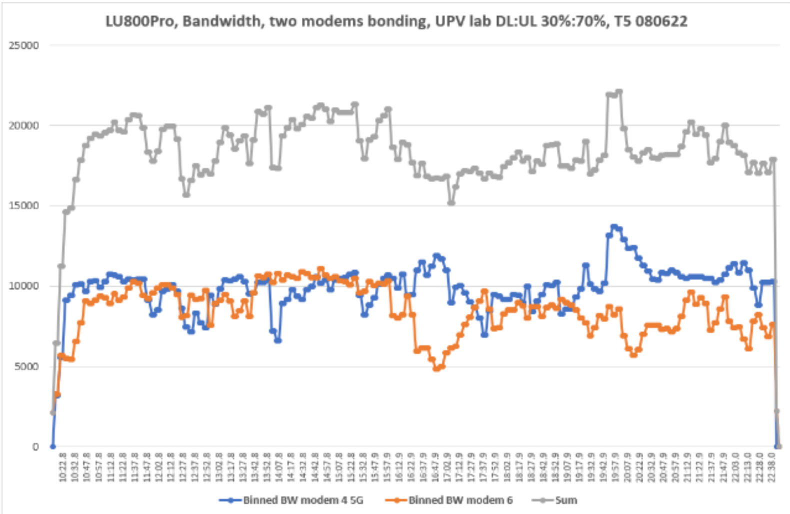


Figure 95: Bitrate of bonding 2 modems, 30:70 UL:DL, SA, LU-UPV trial

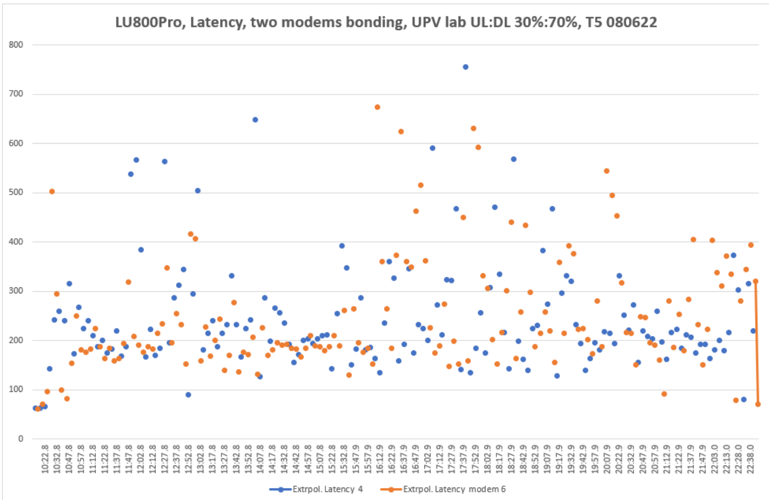


Figure 96. Latency of bonding 2 modems, 30:70 UL:DL, SA, LU-UPV trial

3.2.9.4 Results for single and bonded, very low UL – UPV lab 5G-SA, 60 Mbps maximum at LU800

Network configuration: 30 kHz SCS, 20 MHz BW, 30/70% of UL/DL pattern.

Results are shown in Figure 97, Figure 98 and Figure 99.Single modem more stable than over 100 MHz and high, unstable latency (Figure 97). Bonding stabilizes each modem and the total BW at higher overall, while UL latency remains network-erratic.

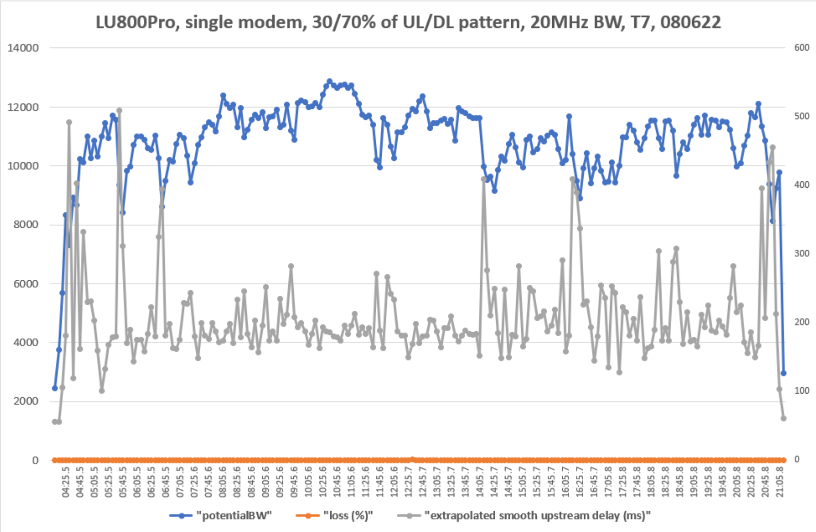


Figure 97. Bitrate, loss, latency of single modem, 30:70 UL:DL, 20MHz channel, SA, LU-UPV trial

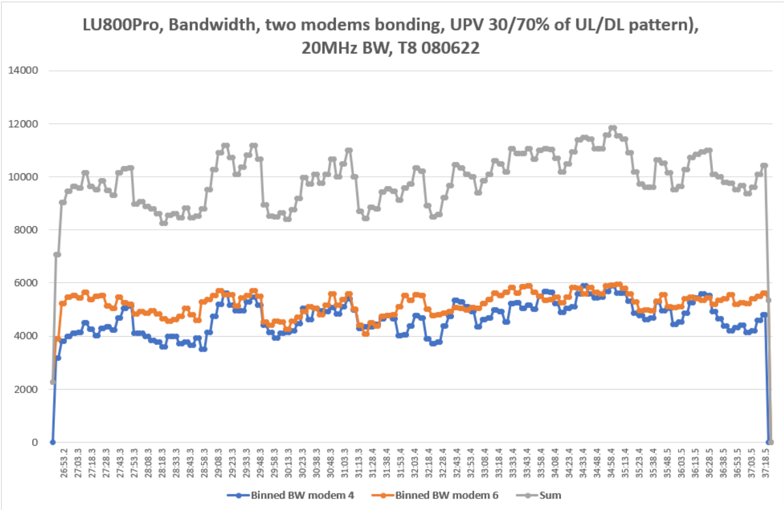


Figure 98: Bitrate of bonding 2 modems, 30:70 UL:DL, 20MHz channel, SA, LU-UPV trial

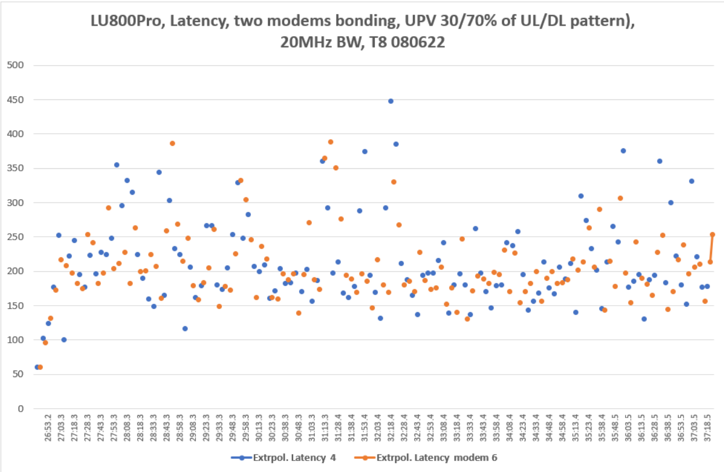


Figure 99. Latency of bonding 2 modems, 30:70 UL:DL, 20MHz channel, SA, LU-UPV trial

3.2.9.5 Results for single and bonded, unrealistic high UL – UPV lab 5G-SA, 60 Mbps maximum at LU800.

Network configuration: 30 kHz SCS, 100 MHz BW, 70/30% of UL/DL pattern.

Results are shown in Figure 100 and Figure 101.Bonding very stable with high BW, stable low latency – on each modem and combined. Assuming on single modem, results should be similar.

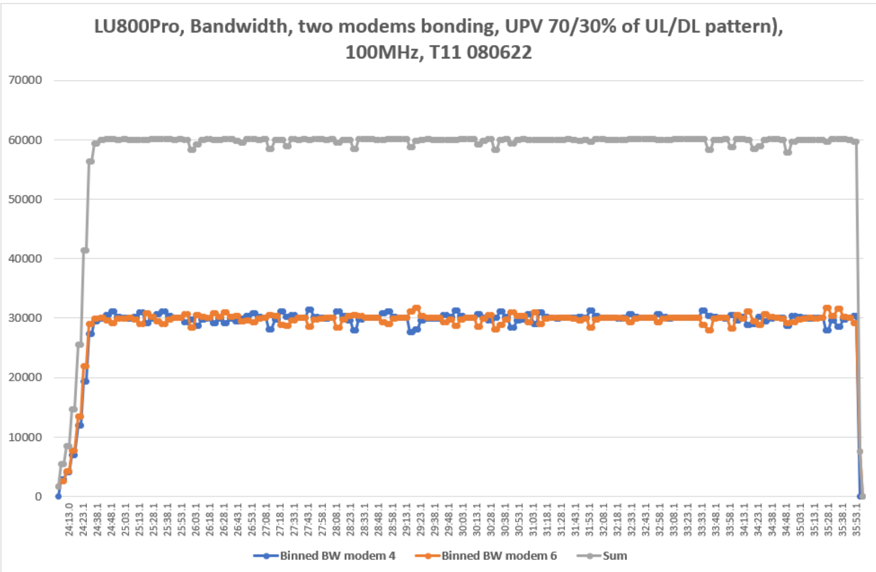


Figure 100: Bitrate of bonding 2 modems, 70:30 UL:DL, 100MHz, SA, LU-UPV trial

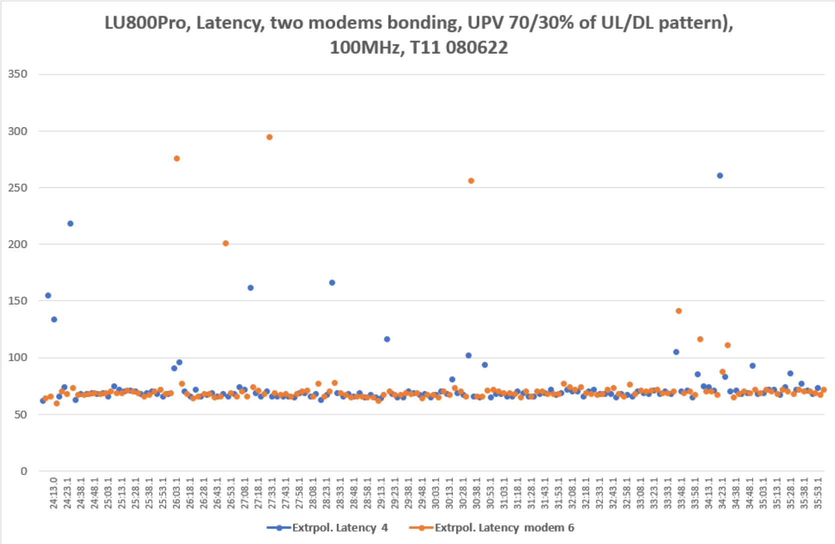


Figure 101: Latency of bonding 2 modems, 70:30 UL:DL, 100MHz, SA, LU-UPV trial

3.2.9.6 Results for single and bonded, commercial outdoors – Orange 5G-NSA outdoors 60 Mbps maximum at LU800

Single modem more stable than over 100 MHz and high unstable latency (Figure 102). Bonding stabilizes the total BW at the max overall, while UL latency and per modem performance remain network-erratic.

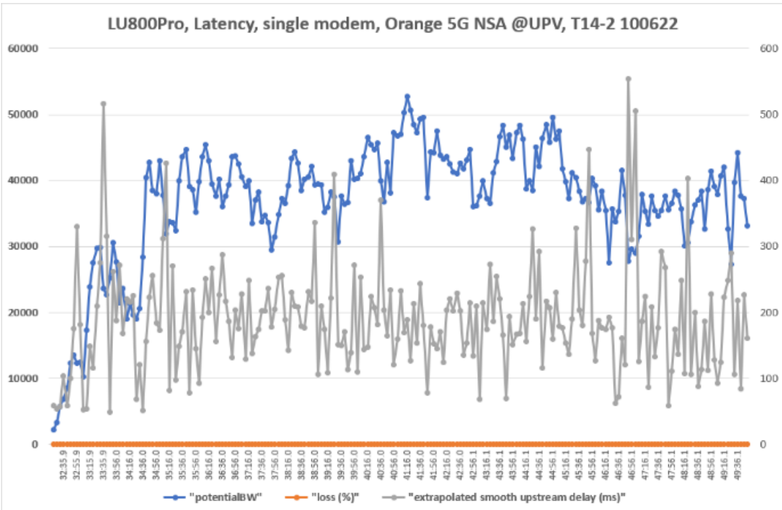


Figure 102: Bitrate, loss, latency of single modems, Orange NSA, LU-UPV trial

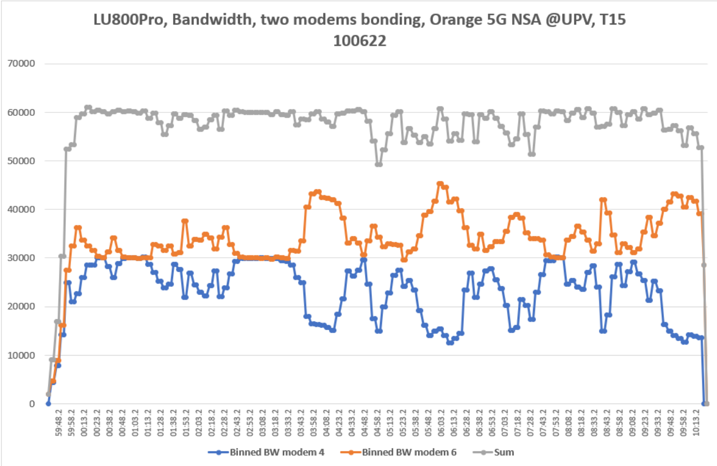


Figure 103: Bitrate of bonding 2 modems, Orange NSA, LU-UPV trials

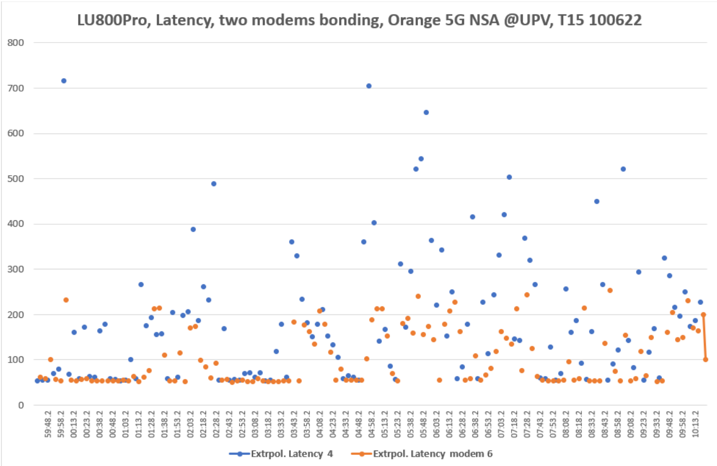


Figure 104: Latency of bonding 2 modems, Orange NSA, LU-UPV trial

From these tests, several conclusions can be extracted:

1. The external Fivecomm modems manifested low and unstable performance in all the benchmarking scenarios, so in advanced tests only the LU800 embedded Sierra Wireless modems were used.
2. Bonding effectively splits the encoded video, evenly when conditions of both links are similar and compensating for fluctuations in any single one of them with the other, as long as the other is capable of supporting that.
3. The commercial 5G NSA connections of single and dual links delivered good performance that enabled the LU800Pro to transmit 4 streams simultaneously, each at 15Mbps, for a total of 60MBps UL over dual-modems bonding.
   The single and dual links performance of the indoors 5G NSA lab network at 20MHz band were insufficient for a full utilization of the LU800UL and demonstrated instability in the BW and latency, resulting at a 10MBs-12Mbps UL capability with High and jittery latency. At 100MHz, this network showed the difference between 30%:70% UL:DL TDD patterns to 70%:30% UL:DL patterns, where the first was less stable in terms of BW and latency.
4. Only when the bandwidth and % of UL resources are increased and bonding is used, the SA network can provide similar capabilities to the NSA network in single modem configuration (stable 60 Mbps).
5. In outdoor scenarios, farther from the gNB, single modem configuration can’t achieve the maximum performance of 60 Mbps, but with the addition of bonding,

this problem is eliminated. Hence bonding is demonstrated to provide high quality video even when the networks are not congested, yet at a distance from the tower as the signal weakens and/or S/Eo is lower.

3.2.10 Final trial in Copenhagen (Tivoli gardens) [Local and remote production - UC2.B.9]

The results of this subsection correspond to the Tivoli trial described in Section 3.1.1.9.

3.2.10.1 PCAP analysis

In order to measure the total glass to glass latency during the trial at Tivoli, an app was used that gives a clock accurate to 1/60 of a second. By filming the clock via the connected 5G camera and displaying that on a monitor a photograph was taken with both the original clock and the image on the screen. From here it is possible to determine a latency figure of the whole chain by calculating the difference between the two clocks. In this instance it was measured this to be 180 ms (see Figure 105), which includes latency introduced by the camera, encoder, wireless transmission, decoder and display.

The PCAP file recorded in Tivoli was analyzed by Ericsson.

- Port 6000: Camera 1 with Traffic Shaping
- Port 16000: Camera 2 with FIFO
- HEVC @ 1920x1080p50 (hvc1)
- 20 Mbit/s for each camera – no QoS (see Fig. 76)
- 180ms glass to glass
- Duration: ~41min
- Total number of frames on port 6000: 123150
- Total number of frames on port 16000: 123148
- No Packet Losses (i.e. continuous RTP sequence number sequence)

Figure 105: Glass to glass latency, showing 180ms latency during the Tivoli Trial.

Figure 106 shows the measured bitrate obtained during the trial over time.

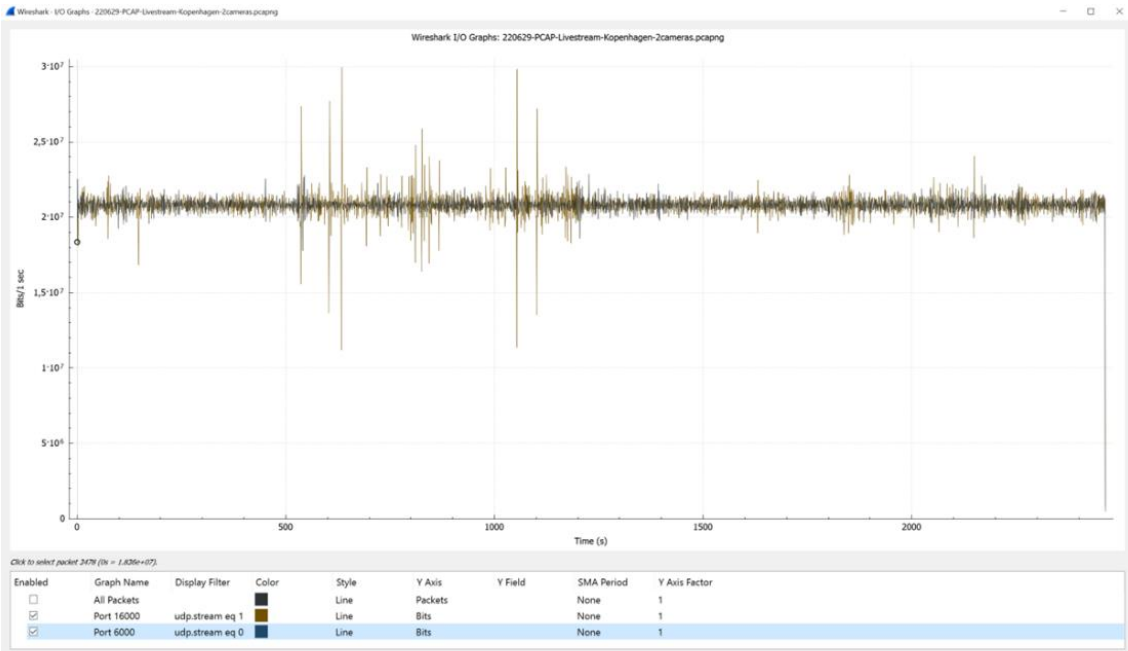


Figure 106: Bit-rate curve for the HEVC streams from the Tivoli Trial.

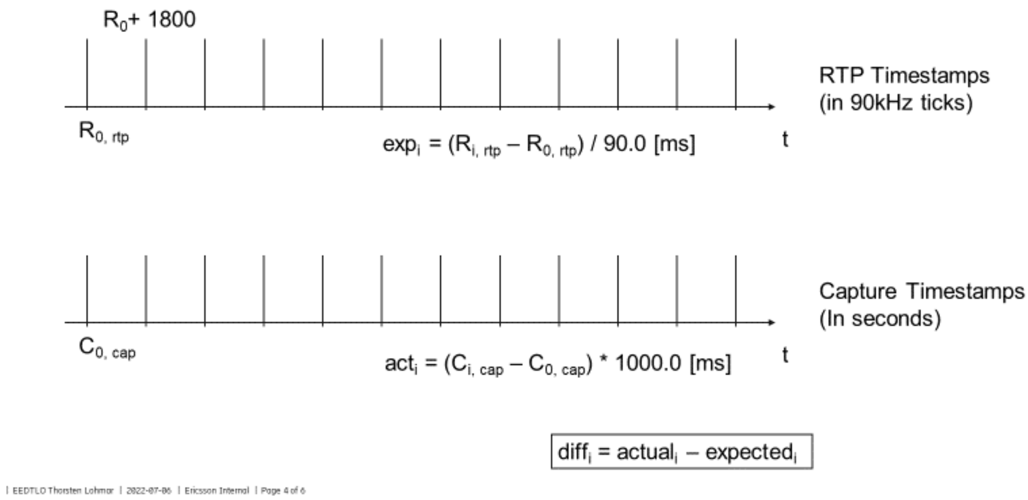


Figure 107: Frame arrival variation KPI.

The frame arrival variation KPI defined in Figure 107 was also measured. The frame arrival variation is very similar to the definition of the Relative Transit Time of the Timestamped Delay Factor (TS-DF).

Figure 108 and Figure 110 below presents the frame arrival variation and frame reception duration:

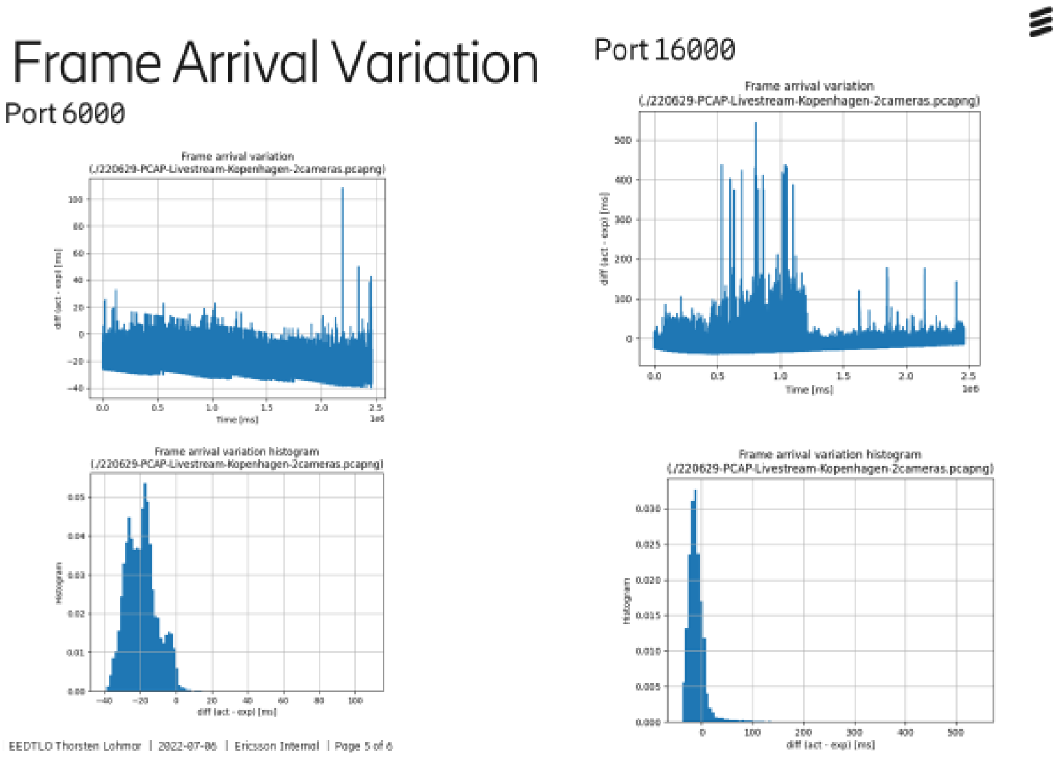


Figure 108: UC2 Aachen tests frame arrival variation.

The reception duration KPI (non-GDR) was also measured (Figure 109).

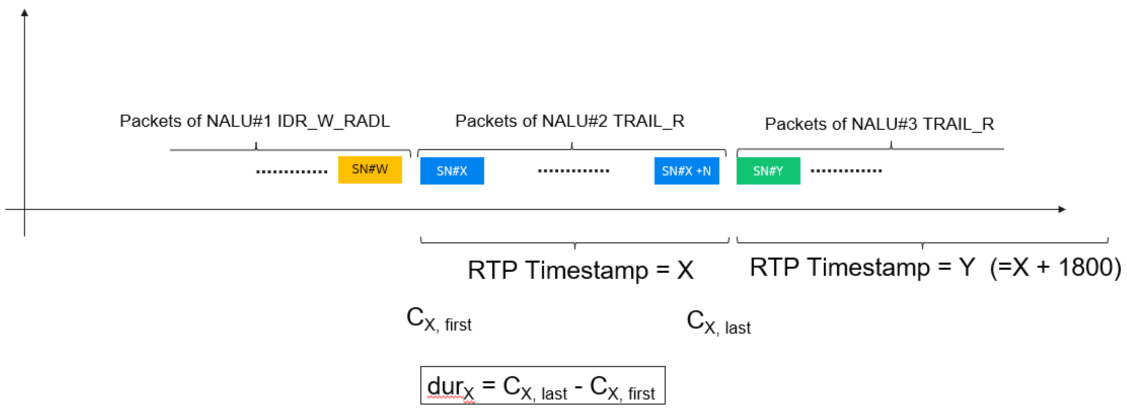


Figure 109: Reception duration KPI.

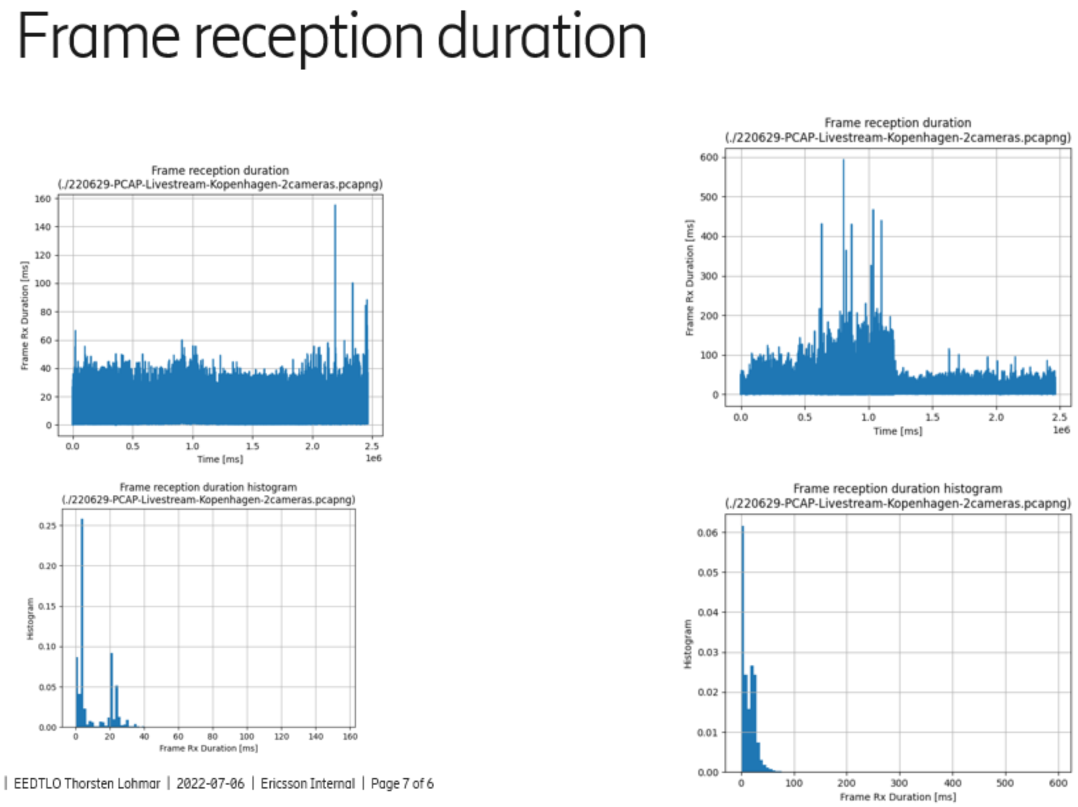


Figure 110: UC2 Aachen tests frame reception duration.

3.2.10.2 Remote production scenario

The transmission for the remote production trial in Tivoli was run for several hours each of the 3 day-event. Measurements here were also collected by the LU800Pro.

Expected network behaviors:

- Higher UL resources (with no load) enable more stable transmission also on single modem
- In real commercial network, outdoors, each link is unstable
- During the "antenna alignment" stage, performance of the mmWave link was poor with many disconnections and "drops". This is due to the directivity of the antennas with narrow beam (supposedly ~10deg) on both ends of the link. Once the alignment was done properly, this disappeared.
- Under same conditions, bonding (by the LU800Pro-LU2000) stabilizes the total UL BW while the network behavior is erratic

Unexpected network behaviors:

- Even when directly pointing at the 5G mmWave antenna at a reasonable distance, the 5G mmWave CPE link suffered from occasional drops

  The latencies over the 5G mmWave were both relatively very high and unstable/jittery. Whereas the firs of these issues might be caused by some unknown network configurations and hops, the instability of the latency is difficult to explain. This is more so as the 5G mmWave link/network were dedicated and as the backbone network was also not loaded. In simple speedtest from a computer the link+network provided around 380 Mbps uplink, so there is no issue of congestion or even partial load.

3.2.10.3 Results for 2 modems bonding, 5G mmWave and SNPN – Bonding mmWave CPE (0) and SNPN (1); configured priority to mmWave @30 Mbps

Results are shown in Figure 111. Occasional drops in the mmWave result in staying on the lower priority modem (1) until it drops.

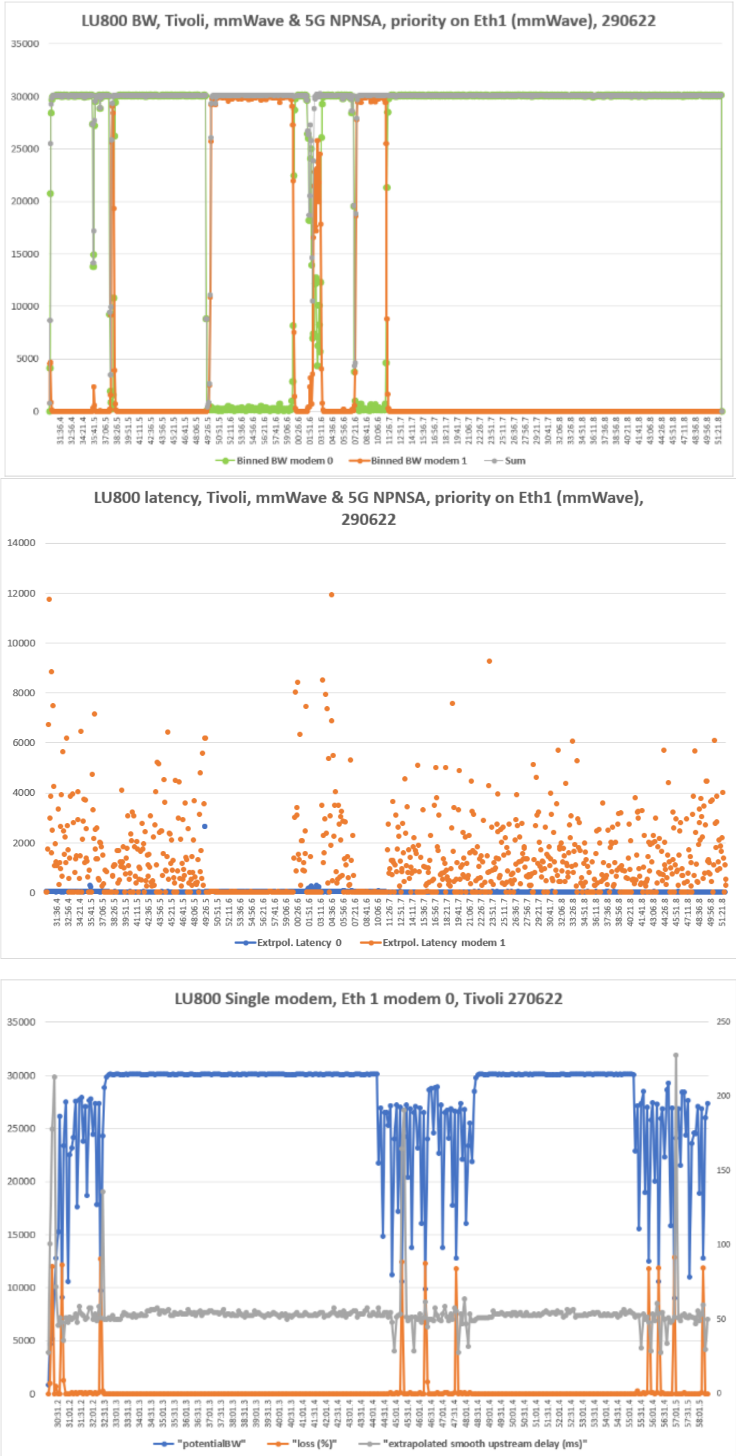


Figure 111: Results for 2 modems bonding, 5G mmWave and SNPN for remote production trial in Tivoli.

From the initial antenna pointing period, disconnections, see Figure 112:

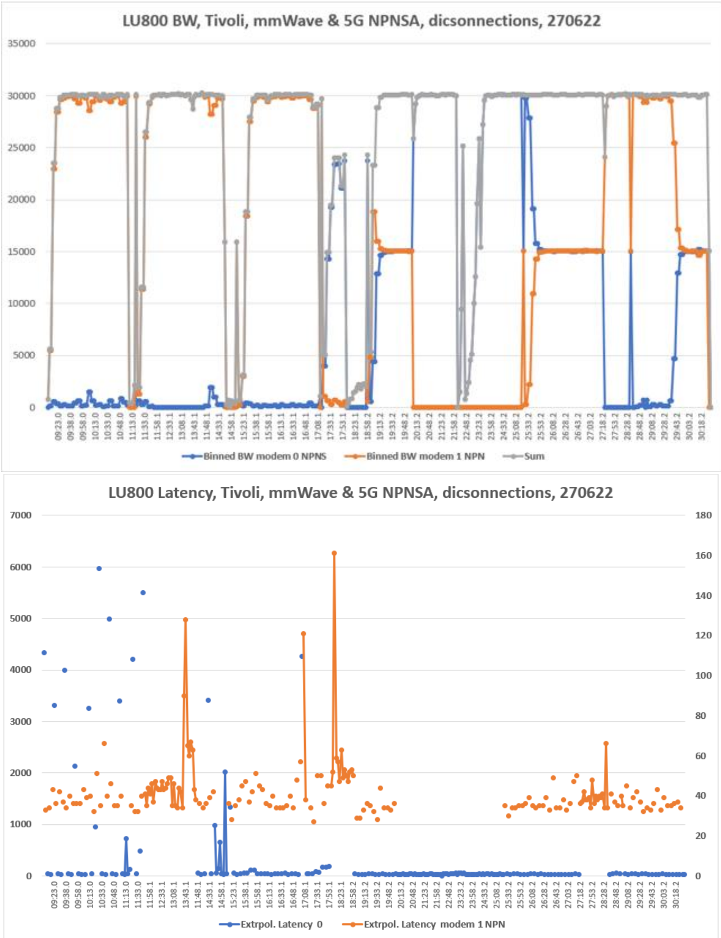


Figure 112: Disconnections for Tivoli remote production trial.

When directed, see Figure 113 and Figure 114:

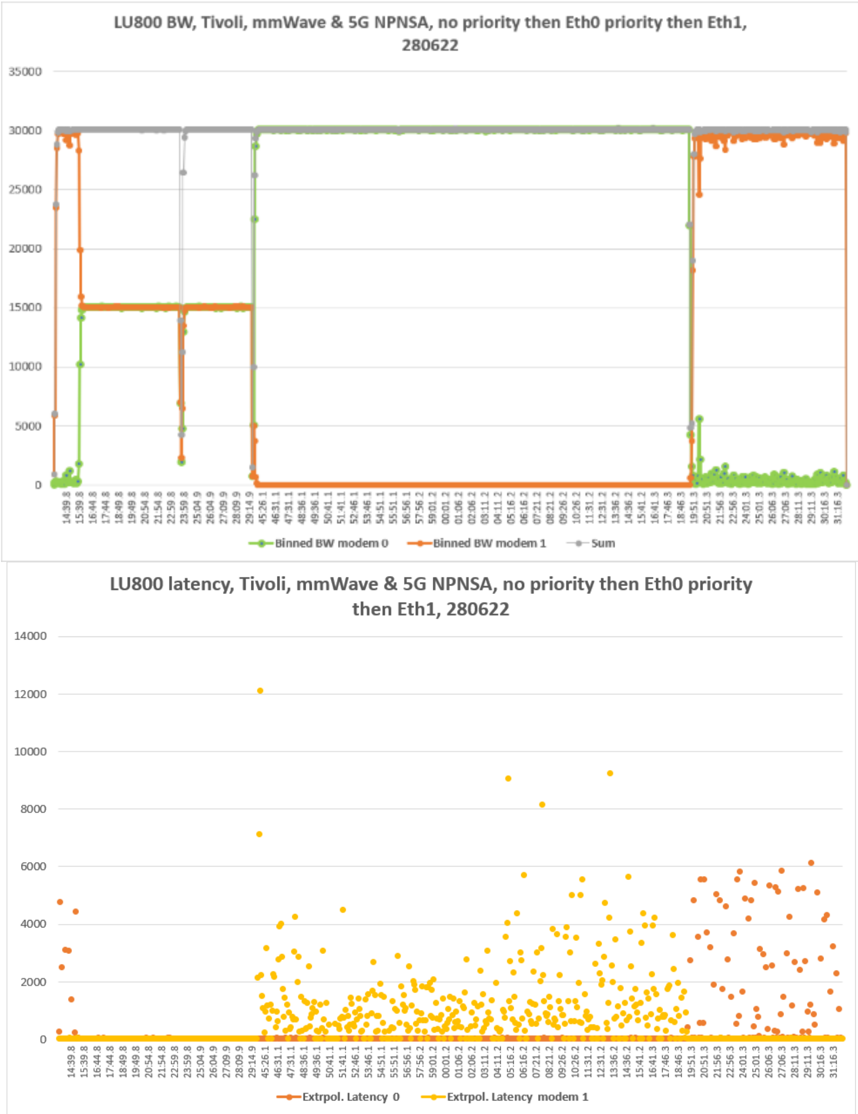


Figure 113. Bitrate and latency when no priority is used.

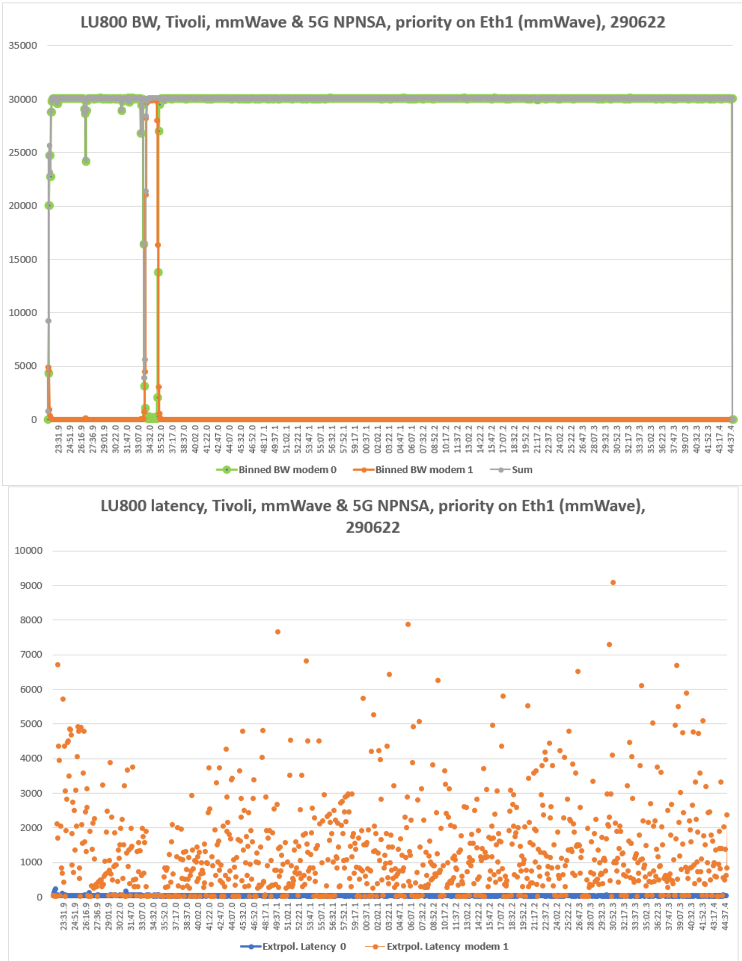


Figure 114 Bitrate and latency when priority is used.

3.2.10.4 Results for the remote RAI TV production

Trial shows that the proposed scheme is feasible and usable for specific TV production. It's suitable mostly for scenarios that do not require a strict requirement in terms of E2E delay; less suitable for live production, for example.

A copy of the RAI PGM was recorded locally using a professional recorder in ProRes 422 (Proxy) format with a bitrate of, roughly, 90 Mbps. Some configuration issue was experienced with audio, resulting in the presence of an inaudible audio track.

The recorded stream was examined by the expert view of a RAI R&D employee working in the codecs and compression department. The verdict of this examination was a good overall picture quality with a lot of details but with relevant blocking artifacts, possibly related to the complexity of the scenes (i.e.: lots of very fine leaves moving with random patterns, presence of shaded and over lit areas, etc.).

Although the resulting footage could be suitable for some applications (e.g.: on field reporters), it is still not enough for more demanding applications (e.g.: sports), where picture quality is of utmost importance.

3.2.10.5 Latency measurement between Tivoli and Turin

During the Tivoli trial, RAI measured the E2E latency (2-way) between Tivoli and Turin. Figure 115 shows an E2E latency value of 3 seconds.

Figure 115: Tivoli - Turin latency measurement for remote production trial at Tivoli.

This measurement was achieved by superimposing a timecode on the SRT output coming from the GV AMPP instance. As it can be seen from Figure 115, on the way the video underwent the following video processing steps:

- GV AMPP generates the H264 SRT stream and superimposes the timecode
- reception of the SRT signal in Turin and subsequent decoding in SDI *(Step A)*
- signal processing inside the video mixer *(Step B)*
- coding of the mixer output video in SRT H264 and sending back to GV AMPP instance in Tivoli.
- The two incoming and outgoing signals are taken from AMPP in SDI and sent to a monitor for measurement. *(Step C)*

The major contribution to the overall latency is given by the double conversion from SDI to SRT H264 and vice versa. The latency contributions made by the other video processing steps are considered negligible.

3.3 KPI analysis

3.3.1 Multicam live production in Tivoli

During the Tivoli trial, the measured glass-to-glass latency was around 180ms that could be further improved using more performant low latency encoders. From the PCAP analysis, the 5G network did not introduce any packet losses. A more in-depth analysis is needed to understand better the jitter introduces by the 5G network and ensure a smooth traffic spike-free. Beam forming, sector forming with proper planning could increase the available bandwidth. During the live production, the cameras were in free-running (not genlocked) so no synchronicity tests were performed.

3.3.2 UL throughput, latency, packet loss rate (remote contribution)

As explained in the results above, these KPIs were met in full. UL bandwidth for single and multi-cam transmissions reached 30Mbps and 60Mbps respectively. UL latency was below 1 sec, tested at ~600 ms end-to-end, from image capture, through A/V encoding, to transmission over the 5G networks, through the public internet from one country to another (in the EU), to studio, to decoding, to output. Loss rate over this full end-to-end path was achieved by the network – at 0% or close to it (measured by the application level).

The KPIs were met when no congestion or load was applied. When load was applied, depending on the load, slices uses, UL: DL TDD patterns – sometime the latency and or BW got erratic, jittery, sporadically deviating etc. Bonding proved to be an efficient tool to overcome these phenomena. Details are provided in sections above.

3.3.3 Multi-cam via the LU800Pro (remote contribution)

Multi-cam was demonstrated successfully via the LU800P – up to 4 streams. BW, latency and loss rate of the full path (end-to-end, measured by the LiveU application level), were achieved and maintained when the network was not too loaded – per the above descriptions. Bonding again proved potent to mitigate and overcome network problems. QoE of the video was tested (in RAI) and passed successfully, even when issues were occurring with the network – if bonding was used.

3.3.4 Remote audio communication (remote contribution)

Remote audio communication was tested and passed. Audio communications between remote producer/director and camera operator/reporter, in parallel to the UL video streams including when 4 streams were uplinked, was working ok from RAI studio all the way to the Aachen or TV2 labs over the labs 5G networks. Quality was ok, operation worked fine.

3.3.5 Cameras remote control (remote contribution)

Camera control was tested and passed. Camera control from RAI labs all the way over the full path to the Aachen labs and TV2 labs over their local 5G network was done using the LiveU IP-PIPE link (with its LiveU IP PIPE server and configuring all IP addresses properly in the various subnets) via the Cyanview control boxes (and proxy server – as this is how the Cyanview architecture works). Shading of the camera was done and successfully. Latency from sending the command from the RAI lab to receiving the visual feedback after the camera iris actually electro-mechanically responded and moved and the video captured, encoded, transmitted over the 5G UL, the public internet, back into the RAI and decoded there, was about 800milisec. The camera lens moved at about 50-70 msec from command sending. The latency of the DL traffic over the LiveU IP PIPE itself (laptop to LU servers, public internet, 5G network, LiveU LU800Pro output, laptop), including full path, was 54 msec on average.

3.4 Technology validation outcome

3.4.1 Modem, media gateway and MCR

The 5G SA modem (Fivecomm), the media gateway (BISECT) and the MCR (EBU) have been validated in different test sessions, during the live trial in Copenhagen, the IBC event and the Ericsson Innovation Day (see more about the Ericsson Innovation Day in D6.4). The results are described above. Note that, although some problems were faced during the trials in Copenhagen, such as lack of connectivity at the beginning or hardware problems in the video interface unit, the three components were successfully validated in this trial. No major integration issues were encountered in the following trials.

3.4.2 LU800Pro & LU2000SMPTE

This pair of field unit and receiving server were validated in the tests and trials. The LU800Pro contained embedded Sierra Wireless 5G SA module which worked fine.

External 5G SA modems were also tested and worked fine with the unit (5G router, Fivecomm 5G modem, 5G mmWave CPE).

The LU2000SMPTE passed the SMPTE tests and trials, except for a very few non-complaint items that were declared upfront and that do not impact the video handling.

The UL transmission of the live video between the devices in this UC under various conditions, network configurations and 5G capabilities was tested (including SA/NSA, slices, TDD UL:DL patterns, channel BW [MHz], 5G mmWave, congestions) for both single video feed and 4 video feeds transmitted simultaneously by the same LU800Pro, and for single modem and 2-modems bonded transmission.

Conclusions about network behavior and especially slices were drawn and described above.

4 Use case 3: Live Immersive Media Production

This chapter describes the outcomes of the final stage of trials in the context of use case 3. The latest updates of the testbed architecture will be presented alongside new measurements results and a detailed KPI analysis. Also, the technology validation of this use case will be addressed.

4.1 Deployed testbed architecture

The architecture that has been deployed is shown in the following figure.

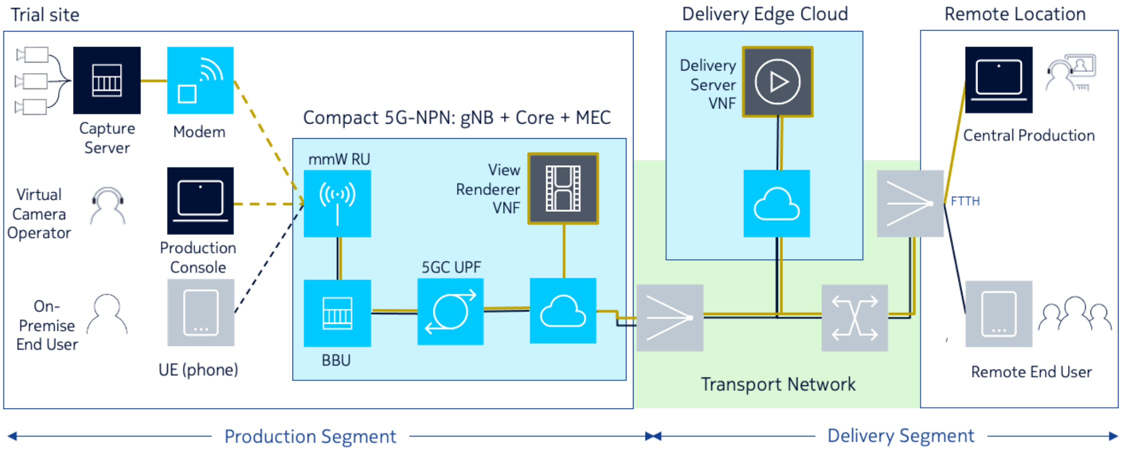


Figure 116: UC3 testbed architecture.

4.1.1.1 Free-Viewpoint-Video System

The final architecture for the FVV Live system is divided into 4 main modules: *(i)* capture, formed by the cameras and the capture servers, *(ii)* stream selector, *(iii)* the rendering module, consisting of multiple production consoles and renderers, and *(iv)* the replay module, that allows the playback of pre-recorded FVV content.

The capture module takes care of capturing the scene and computing the geometrical information in the shape of depth images, then compressing and transmitting the video and depth streams, together with an audio stream.

These streams are received by the stream selector, which redistributes each one of them to every renderer as needed, allowing for multiple views to be rendered at the same time.

Each renderer has a production console assigned, which controls the movement of the virtual camera. Each rendered view is compressed into a video stream and sent to both the console production, for the producer to get feedback, and to the media delivery to re-transmit it to the end users. The feedback stream can be also sent to the media delivery, allowing for remote production consoles to be used.

Additionally, pre-recorded content can be sent to the stream selector using the FVV replay module. This way, new virtual camera paths can be produced over the stored media.

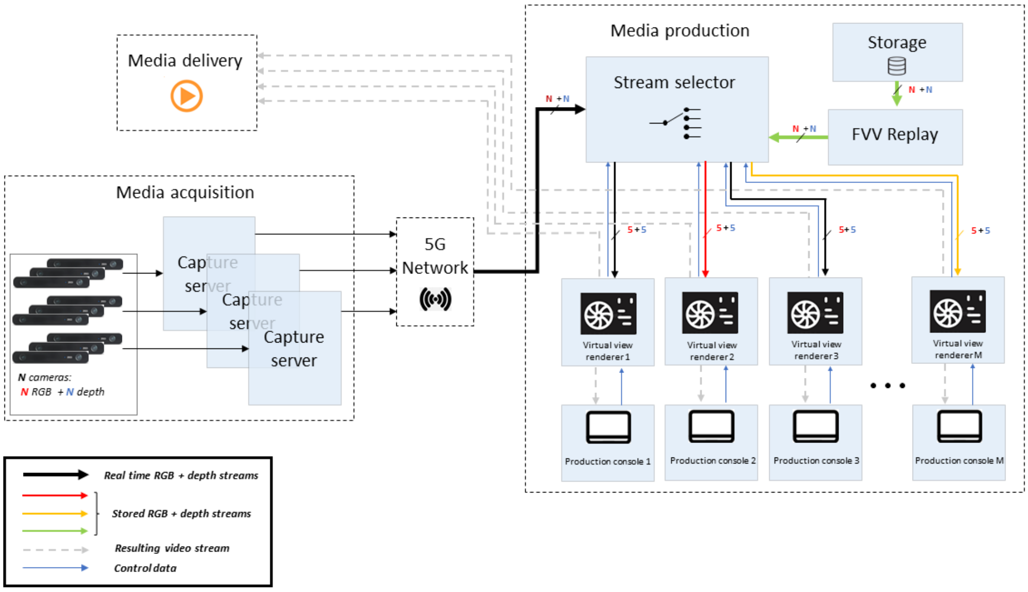


Figure 117: Free-Viewpoint-Video System.

4.1.1.2 Compact 5G+MEC Deployment

The architecture of the testbed deployed in the trial site is depicted in the figure below.

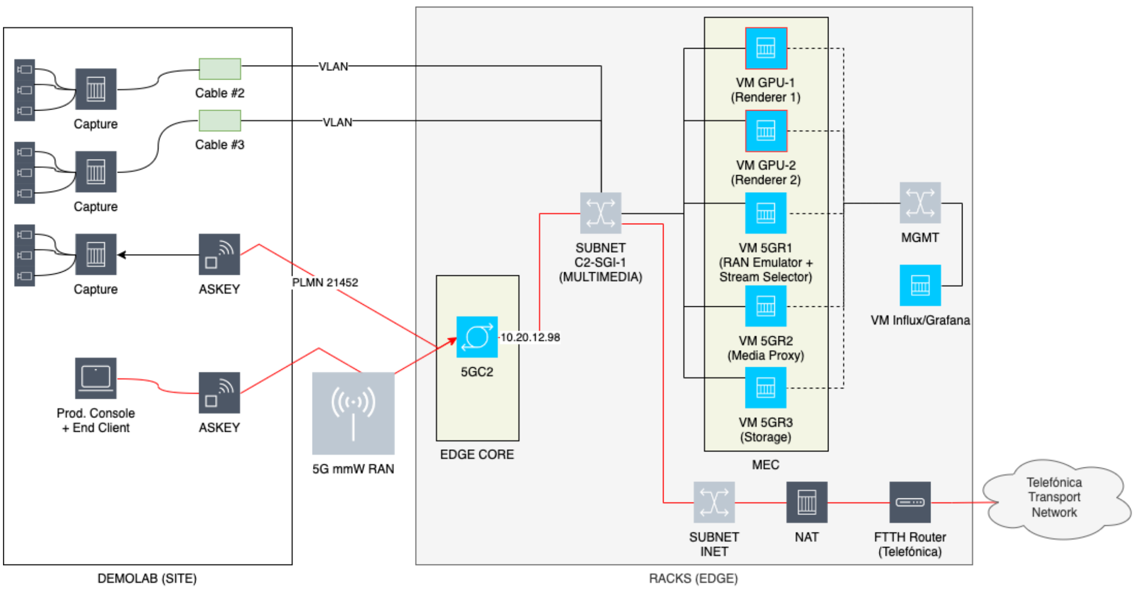


Figure 118: architecture of the testbed deployed in the trial site.

For the RAN, a 5G NSA 3.x cell was used in FR2 (millimetre wave). The cell aggregated a total of 8 carriers, with 100 MHz each, in TDD configuration with DL/UL rate 4/1. Two of the carriers were enabled for uplink (2CC-UL), while the 8 of them were aggregated for downlink (8CC-UL). The radio cell is configured using MOCN (Multi-Operator Core Networks), so that several private networks can be defined and get service simultaneously over the same physical cell. This allows for different projects to be supported in the same physical cell. ASKEY modems were used as UE. Two of the capture servers were directly connected to the core network and used network emulation where appropriate.

During the trials, only the 5G-RECORDS network was active. A private PLMN code was used, resulting in a S-NPN configuration. The network core (4G/5G) was fully installed in the near-edge platform in Nokia premises. A specific instance (5GC2) is used only for 5G-RECORDS project. The N6/SGi interface to the data network was connected to a local virtual LAN (VLAN), represented as C2-SGI-1 in the figure. This VLAN is used to connect the 5G devices with the MEC. Besides, northbound connectivity with the Internet is established from the service VLAN C2-SG1-1 using a FTTH router and a commercial FTTH line, provided by Telefónica. This enables the connectivity of the deployment with Telefónica's Edge Cloud and SDN, to support the upstream of the contribution video and the downstream of the end-user video.

The MEC was installed in a Nokia AirFrame Open Edge server with two Tesla T4 GPUs. The virtualization and orchestration platform used to run the virtual servers was MicroStack, a compact version of OpenStack developed by Canonical. Five VNF instances were deployed to accommodate for the content production functionalities of the use case: two instances of the View Renderer, with GPU access, one instance of the Stream Selector, one of the Media Proxy, and one of the Storage + Video Replay. The Stream Selector instance also contained a RAN emulator (Nokia FIkoRE), to be able to emulate the characteristics of the RAN link for the systems which were not connected to the physical network (the two remaining capture servers).

Two 5G modems were deployed in the trial site: one to connect one of the capture servers, another one to connect the production console. As there was not enough uplink capacity available to accommodate more capture servers in the wireless network, the rest of the capture servers were connected directly to the C2-SGI-1 network using cables. 10 GEth interfaces were used in all the links, to avoid having any potential bottleneck in the local network.

A monitoring server was also deployed in the trial site, collecting KPIs from the different systems using influxdb. A Grafana dashboard was setup to monitor the KPIs in real time.

4.1.1.3 Edge Cloud and SDN

The SDN is the same as previous revision, which is defined based on the following building blocks:

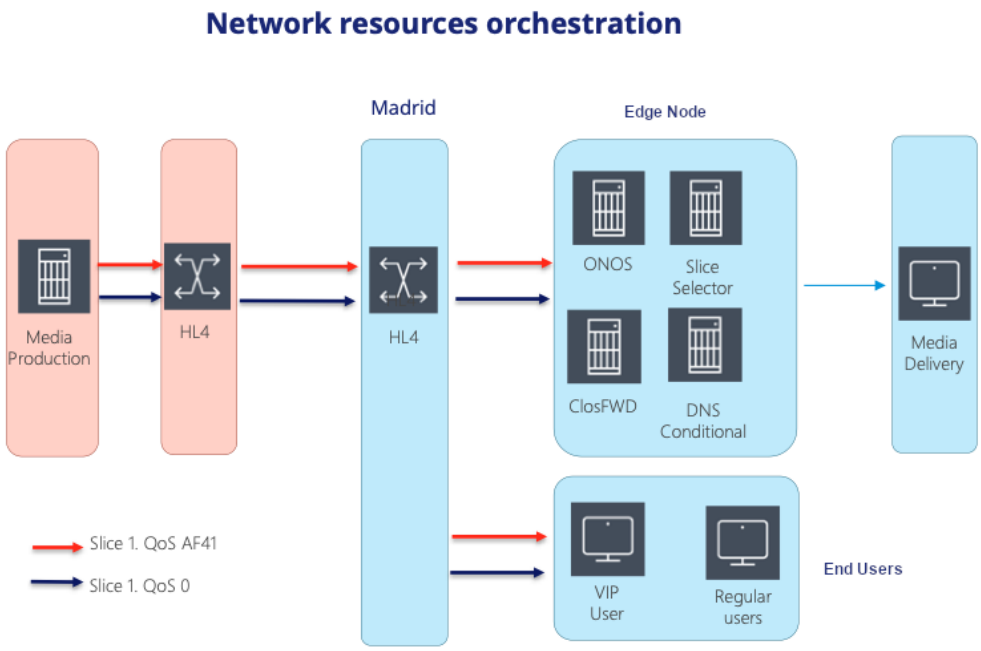


Figure 119: Network resources orchestration.

- **ONOS**: used as the SDN controller, in charge of managing the switch fabric, as described in section “Network connectivity subsystem”.
- **ClosFwd**: application of the ONOS environment is responsible for managing the CLOS fabric of Edge Cloud switching.
- **Slice Selector**: software component based on NGINX servers acting as reverse proxy with capability to redirect traffic to the correct slice based on the request IP and the URL. Combining server instances listening on different public interfaces and different VHOST to segment the traffic along the correct slice. It is connected to both slices gold and best effort internal and external, and to the internal video delivery.
- **DNS Conditional**: implemented using opensource software bind and several views configuration in order to response correctly. That means that responses for VIP users will be different than for regular users.

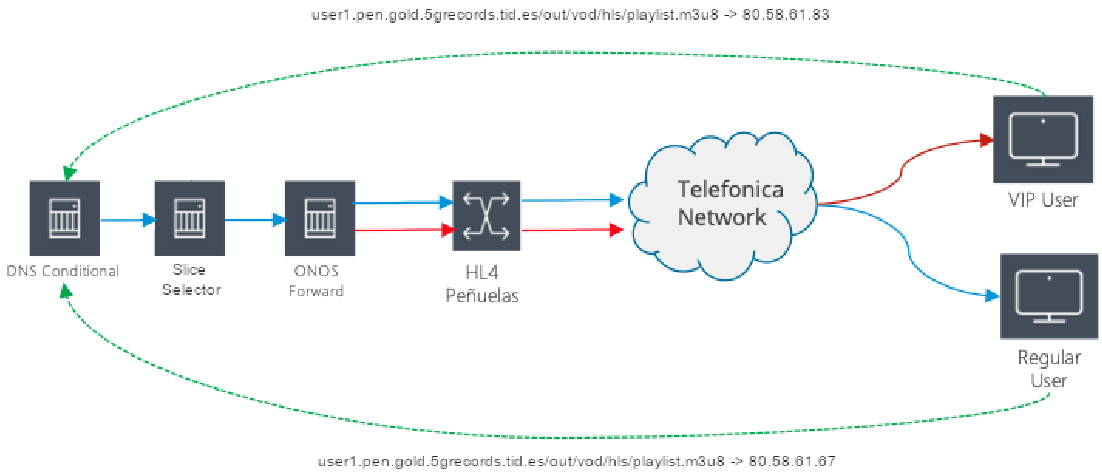


Figure 120: DNS conditional.

4.1.2 Updates on measurements planning

The following measurement points have been selected for the final field trial:

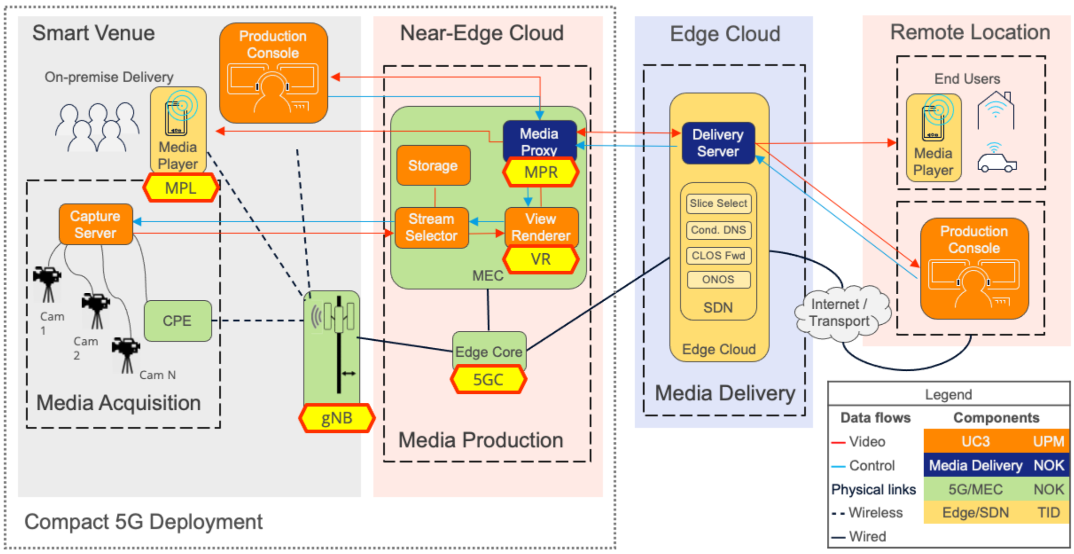


Figure 121: UC3 measurement points.

In all of them, measures are taken in real time during the field trial and sent to the influxDB + Grafana system for real-time monitoring.

- **gNB**. It reports the status of the 4G and 5G cells used for the trial. The following measures are reported:
  - UC3.gNB.1 Radio status
- **5GC**. The 5G Core is monitored. It monitors the traffic in the 5G network. It also includes some probes to monitor RTT with the UEs during the operation. The following measures are reported:
  - UC3.5GC.1 RTT (ping) UE-MEC
  - UC3.5GC.2 Uplink / Downlink throughput
- **VR**. The View Renderer monitors the input traffic (bitrate and losses), as well as the output performance (rendering time). The following measures are reported:
  - UC3.VR.1 Camera Bitrate
  - UC3.VR.2 Capture Server Bitrate
  - UC3.VR.3 Camera Loss Rate
  - UC3.VR.4 Rendering Time
- **MPR**. The Media Proxy monitors the rendered virtual view in production quality, before sending it to the Media Delivery. The following measures are reported:
  - UC3.MPR.1 Rendered View Bitrate
  - UC3.MPR.2 Rendered View Loss Rate
- **MPL**. The experimental Media Player contains a specifically developed automated platform to be able to take measures with different transport slices, as well as to generate background traffic noise. Different combinations of RAN QoS (QCI6/9) and transport QoS (Best Effort / Multimedia) were tested:
  - UC3.MPL.1 QCI9 – Best Effort
  - UC3.MPL.2 QCI9 – Gold
  - UC3.MPL.3 QCI6 – Best Effort
  - UC3.MPL.4 QCI6 – Gold

4.1.3 Uncertainties and risk assessment

Due to the complexity of the end-to-end trial, it was decided to divide it into two sessions:

- A live event (July 2022), validating the end-to-end chain and measuring KPIs at capture and production level.
- A specific trial of the delivery KPIs (September 2022)

4.2 Measurements results

This section reports the live measurements done in the final field trial.

4.2.1.1 End-to-end field trial (July 7th 2022)

This trial was performed to demonstrate the viability of a full end-to-end FVV live deployment to stream and record an event over a 5G network. The event consisted of a live music performance by professional artists which was produced as a FVV service in real time and streamed to the final user. In addition, the FVV content was also recorded to demonstrate the FVV playback functionality of the system.

The trial was performed in Nokia premises in Madrid, Spain. The location of the performance (trial site) was covered by the 5G NSA cell described in the previous section. The full trial was 30 minutes in duration, and it consisted of the end-to-end chain of the use case (capture, 5G uplink, rendering, production console, media proxy, media delivery, and end client) working together simultaneously, while covering the live event.

The performance was captured by the FVV Live's volumetric capture setup, involving nine stereo cameras and three capture servers which were connected to the MEC. A 5G link was used for one of the connections. The cameras were disposed forming an arc around the scene, separated by 40 to 50 cm, and were set to record in 720p at 15 fps.

This sequence had many objects in it and the singer was constantly walking around and dancing, so it is considered to be very complex. This is why more conservative parameters were selected (frame rate, resolution), compared to the maximum possible rate of 1080f30 supported by the system.

All the FVV streams (nine video streams, nine depth streams and one audio stream) were sent to the 5G MEC. As described in the previous section, each capture server was connected to a UE and handled three video streams and three depth streams. One of the UEs was connected to the 5G network and sent the traffic using the 5G Uplink. The other two were directly connected to the C2-SGI1 network due to the limitation of the total number of available cells in the deployment, since a single capture server requires the uplink capacity of a whole mmWave cell in the current configuration.

Once in the MEC, the media was managed by the Stream Selector, which re-distributed it to two instances of the view renderer, capable of generating two different but simultaneous virtual views of the live scene. One of the virtual views was controlled by a production console which was also connected to the view renderer using the 5G network (local). The other one was controlled by another production console that was placed at the UPM lab 14 km far from the stage (remote). This remote production console receives the rendered view from the Media Delivery in Telefónica edge cloud and sends its remote control commands back to it. The Media Delivery is connected to the renderer through the Media Proxy. The connection between Media Delivery and Media Proxy uses the Multimedia Gold transport slice at Telefónica transport network.

Finally, the two produced video streams (the output of each virtual renderer) were sent to the Media Proxy, which sent them to the Media Delivery. At the Media Delivery, both streams were segmented and distributed using HTTP Live Streaming (HLS) to end clients globally. During the live trial, each of the streams was delivered to local clients at the trial site, using the Multimedia Gold slice at the transport network. To avoid having an extremely complex logistic during the trial, only two clients were running during the end-to-end trial. A more complex trial of the delivery part was planned for later in the project, to fully test delivery capabilities.

During the operation of the system, its key metrics were tracked in real-time generating logs that could be monitored also in real-time. Grafana dashboards were shown and monitored during the whole session.

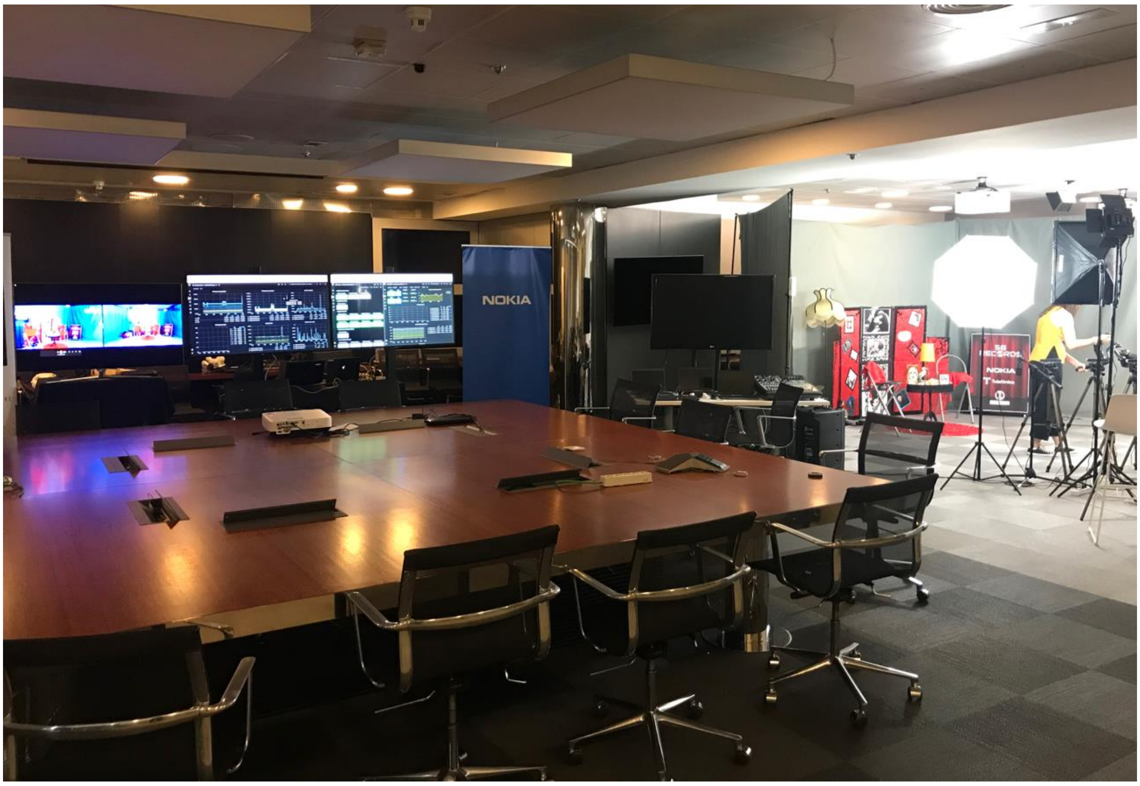


Figure 122: UC3 trial location.

Figure 123: UC3 trial music show.

4.2.1.2 Delivery trial (September 2022)

The figure shows the network configuration for the delivery trial with end-to-end slicing (QoS management).

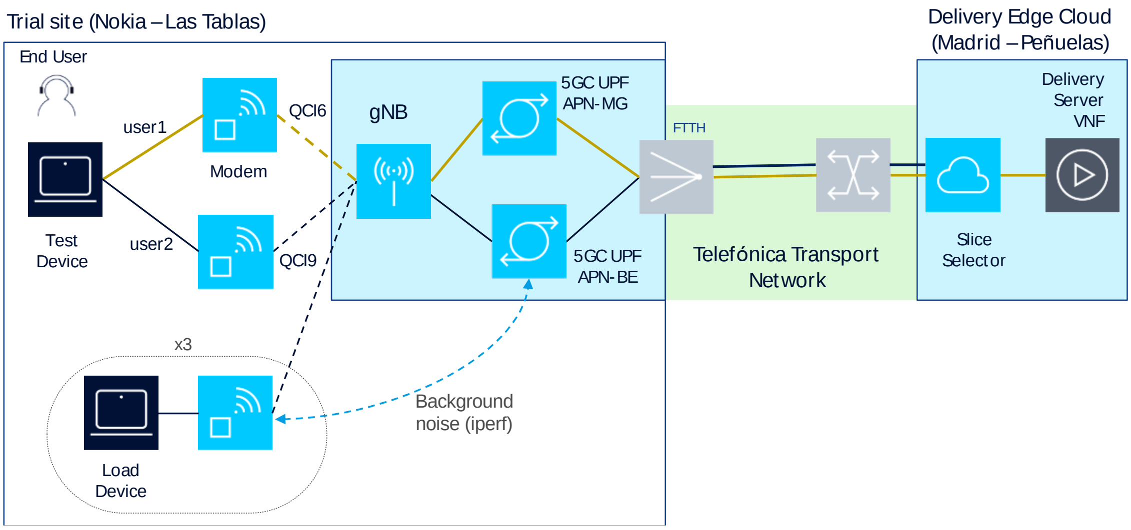


Figure 124: UC3 Network configuration for the delivery trial with E2E slicing.

Two traffic slices were configured (Multimedia Gold and Best Effort) in two different network segments (5G RAN and Transport Network), covering delivery of the produced video from the Media Delivery VNF in the Delivery Edge Cloud to the End User in the trial site. Since only 5G NSA is available for mmWave frequencies, QoS slicing in the RAN is implemented by giving different QoS parameters (QCIs) to different users. The slices are defined as follows:

- Multimedia Gold: QCI6 in RAN and DSCP AF41 QoS level in transport.
- Best Effort: QCI9 in RAN and CS0 (no DSCP marking) in transport.

The trials were executed using the automatic test framework developed by Telefónica and described in D4.2 [6]. They use an online web player which allows to play remote streams. It has been developed specifically for this project.

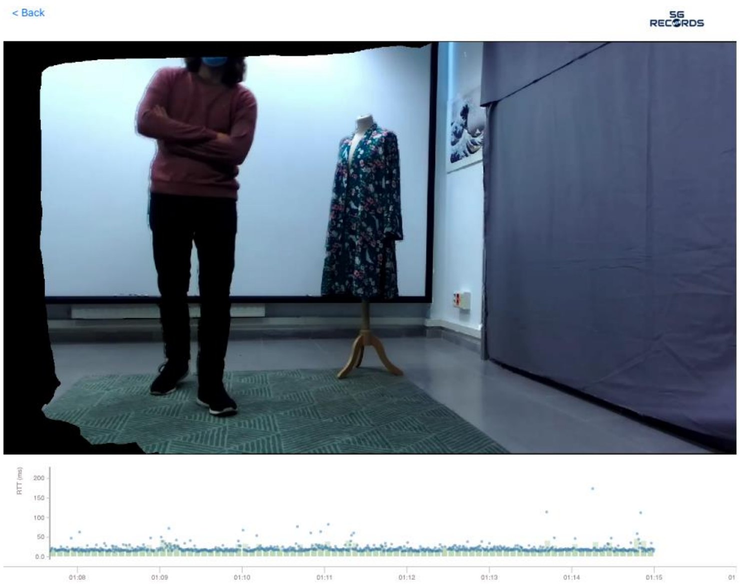


Figure 125: UC3 React APP.

This is a React APP that contains a web page for video Testing. It is based on open source HTML5 player videojs [7].

The most interesting part is the plugin (extended from [8]) that reports user QoE in term of quality metrics during the test execution (example can be found in Figure 126). Specifically, it adds following metrics:

- selectedQuality: the name of the selected video quality in a multi-quality video stream.
- qualityIndex: the index of the selected video quality.
- qualityId: the id associated with the selected video quality.
- width: the width of the selected video quality.
- height: the height of the selected video quality.
- bitrate: the bitrate specified in the selected video quality.

It also provides the capability to report periodically the KPIs by mean of a callback.

Variable	Value
bufferCount	4
bufferDuration	4.887 Seconds
initialLoadTime	0.792 Seconds
pauseCount	1
seekCount	0
totalDuration	373 Seconds
watchedDuration	107 Seconds
selectedQuality	2160
qualityIndex	4
qualityId	4-var24000000/playlist.m3u8
width	3840
height	2160
bitrate	24000000

Figure 126: UC3 automatic test framework.

Using the Web Player, various tests have been run playing content from a media player under different conditions (radio network & transport network), resulting in 4 scenarios:

1. User with QCI9 in a mobile network accessing a media server through a best-effort fixed access
2. User with QCI9 in a mobile network accessing a media server through a Gold fixed access
3. User with QCI6 in a mobile network accessing a media server through a best-effort fixed access

4. User with QCI6 in a mobile network accessing a media server through a Gold fixed access

QCI (QoS Class Identifier) is a mechanism used in 3GPP Long Term Evolution (LTE) networks to ensure bearer traffic is allocated with appropriate Quality of Service (#QoS).

Table 9: QCI is QoS Class Identifier (3GPP TS 23.501).

QCI	Resource Type	Priority	Packet Delay Budget	Packet Error Loss Rate	Example Services
1	GBR	2	100ms	10-2	Conversational voice
2		4	150ms	10-3	Conversational video (live streaming)
3		3	50ms	10-3	Real-time gaming
4		5	300ms	10-5	Non-conversation video (buffered streaming)
5	Non-GBR	1	100ms	10-3	IMS signaling
6		6	300ms	10-5	Video (buffered streaming) TCP-based (e.g., www, email, chat, FTP P2P file sharing, progressive video, etc.)
7		7	100ms	10-5	Voice, video (live streaming), interactive gaming
8		8	300ms	10-3	Video (buffered streaming) TCP-based (e.g., www, email, chat, FTP P2P file sharing, progressive video, etc.)
9		9	300ms	10-5	

In all the scenarios, content has been played in different resolutions, introducing noise (traffic) in the radio network. Traffic has been generated using iperf3 to generate downlink traffic to 3 additional UEs, all of them in QCI9.

Figure 127. UE parallel TCP streams.

X axis: Traffic added into radio section

Each UE receives 10 parallel TCP streams in the following configuration (as depicted in Figure 128):

- None: No additional traffic
- 5M: 5 Mbps per stream, i.e. 50 Mbps per UE and 150 Mbps in total
- 10M: 10MB Mbps per stream, i.e. 100 Mbps per UE and 300 Mbps in total
- 15M: 15MB Mbps per stream, i.e.1 50 Mbps per UE and 450 Mbps in total

Using 10 TCP parallel streams was decided to have a more stable throughput compared to a single TCP stream.

4.2.2 UC3.gNB.1 Radio status

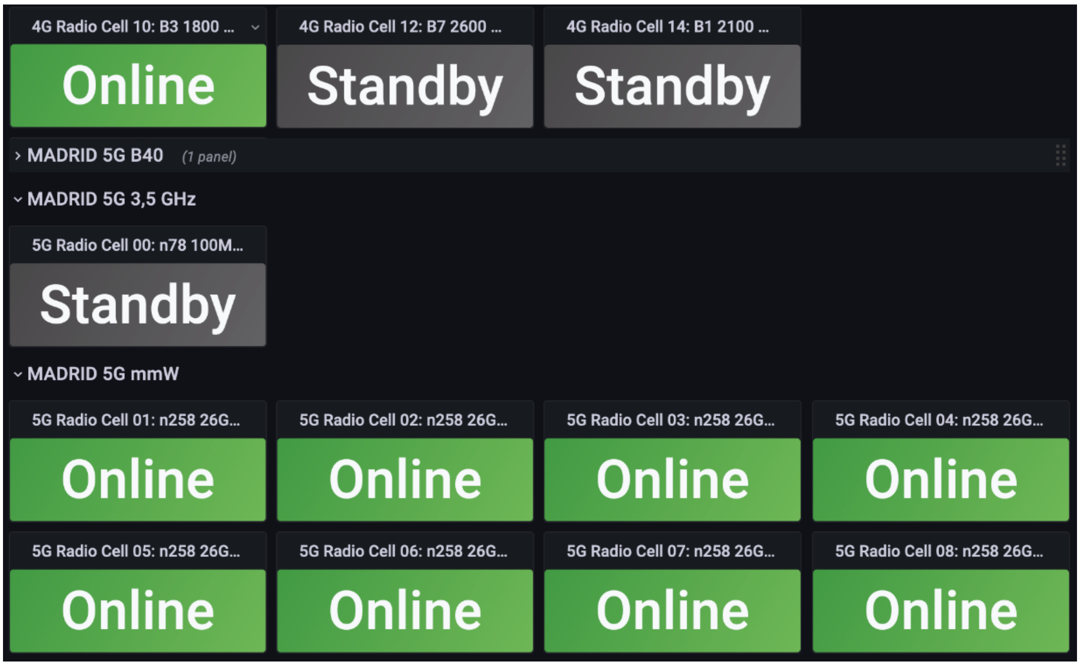


Figure 128: Online radio cells at Nokia AirScale during UC3 trial.

The status of the radio cells is monitored at Nokia AirScale system and reported in the Grafana dashboard. As seen in the figure, the following cells were online during the trial:

- An LTE cell in 1800 MHz band, transmitting with minimum power, used only for anchoring purposes of the 5G-NSA configuration.
- Eight 5G cells (carriers) in millimetre-wave band n258, each one with 100 MHz of bandwidth.

4.2.3 UC3.5GC.1 RTT (ping) UE-MEC

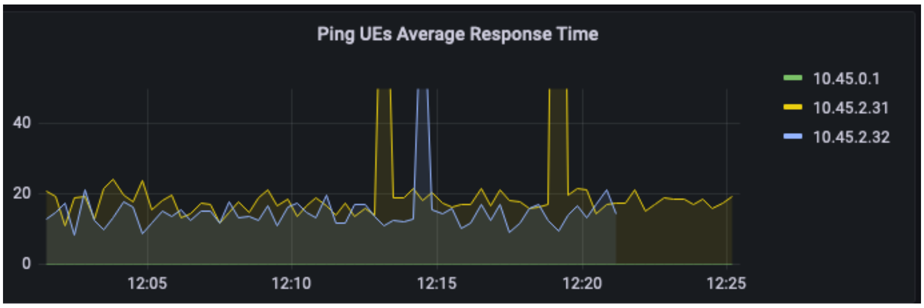


Figure 129: UC3 Ping UEs average response time.

During the whole trial, a periodic ICMP ping was sent from the MEC platform to the 5G core (10.45.0.1), the Capture Server UE (10.45.2.31) and the Production Console UE (10.45.2.32). The RTT with the core is negligible. The RTT to the UEs is consistently below 20 milliseconds, except in 3 individual measurement points, where it increased significantly (up to 300 ms).

4.2.4 UC3.5GC.2 Uplink / Downlink Throughput

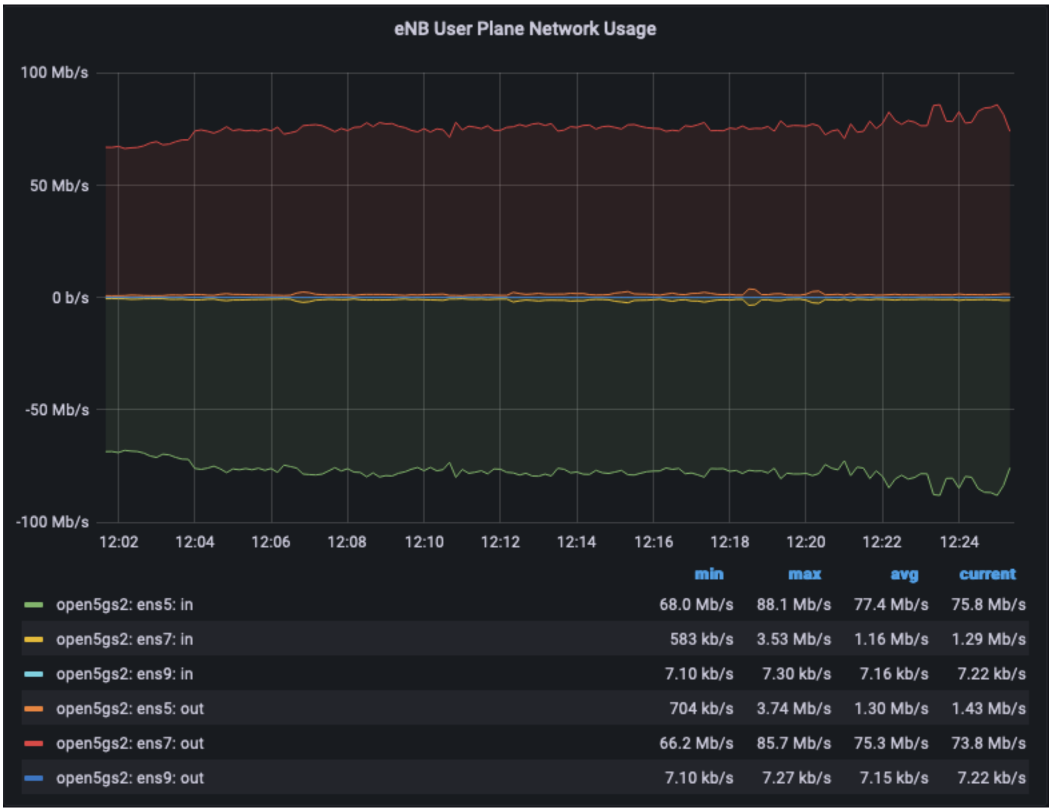


Figure 130: gNB User Plane network usage

The incoming and outcoming traffic was monitored in all the network interfaces of the 5G core. Due to the characteristics of the trial, uplink traffic from the 5G-enabled capture server was the dominant contributor to the total traffic, both at input (coming from the 5G gNB) and at output (going to the MEC). The traffic was very stable during the whole trial session.

Traffic incoming to the user plane function of the core (N3/S1-U) uses the interface ens5. Its average is 77.4 Mbps, and its maximum is 88.1 Mbps. Measures are taken each 10 seconds, so short-term traffic peaks are softened. It can be seen that the supported traffic during the whole session was consistently below 100 Mbps. The output of the core to the MEC (N6/SGi) is served through ens7 interface. It shows slightly lower rates than the input traffic, due to the overhead caused by GPRS Tunnelling Protocol (GTP) encapsulation at N3/S1-U.

4.2.5 UC3.VR.1 Camera Bitrate

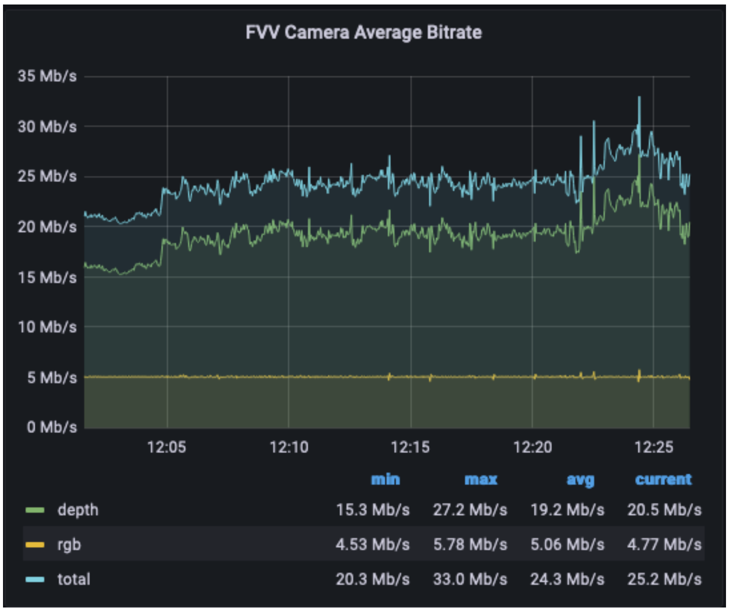


Figure 131: UC3 FVV camera average bitrate.

The average of all 9 cameras used is shown. The video streams were configured to output a bitrate of 5 Mb/s after compression, that is why their graph is mostly constant. For the depth streams, lossless compression is used, so their bitrates are not bounded and strongly depend on the contents of the scene and the foreground segmentation.

4.2.6 UC3.VR.2 Capture Server Bitrate

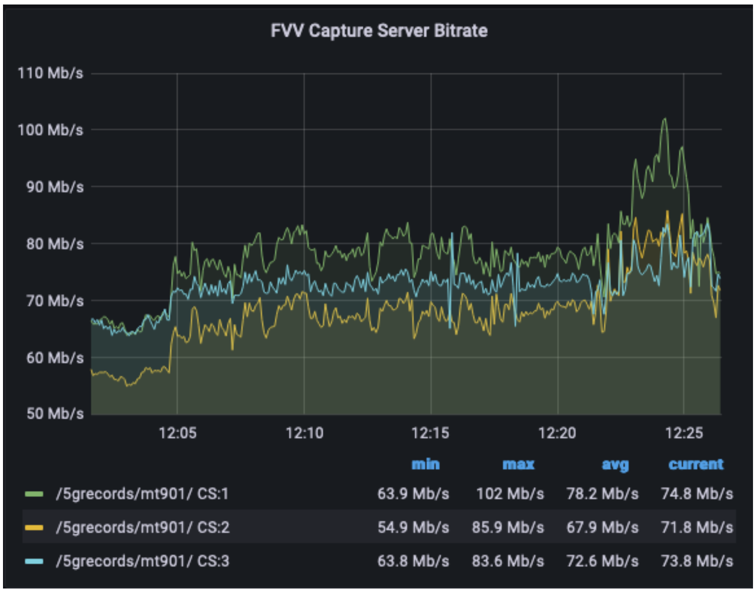


Figure 132: UC3 FVV capture server bitrate.

Each capture node had its own connection to the stream selector (cable or 5G link), so the data shown in this graph corresponds to their total output. Again, since the depth encoding scheme is lossless, we can observe that the output bitrate varies along time. All the plots look very similar since all the cameras are capturing the same scene just with slight variations on their point of view.

4.2.7 UC3.VR.3 Camera Loss Rate

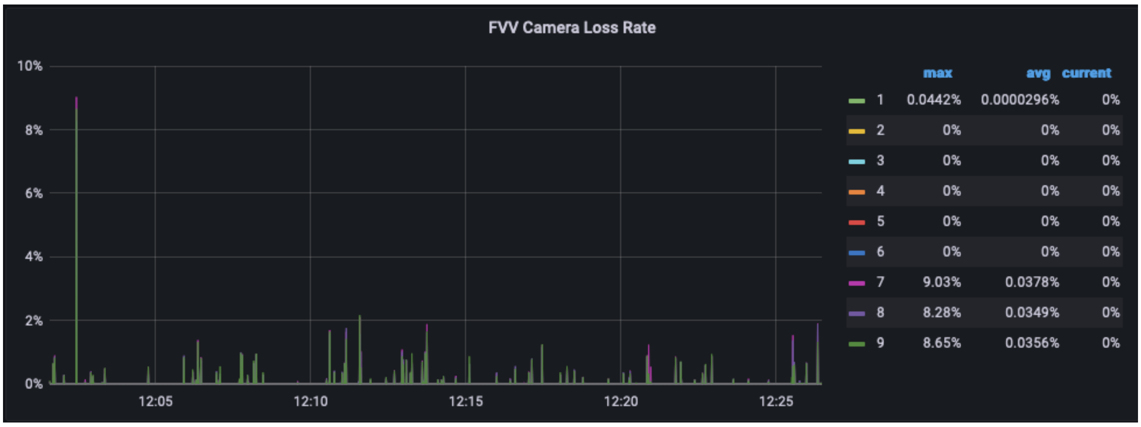


Figure 133: UC3 FVV camera loss rate.

This graph represents the missing packets of each camera streams (depth and texture). Both capture servers 1 and 2 were connected by cable, so their losses are negligible. Capture server 3 used a 5G link, and their cameras suffered from some losses due to the radio access, although the average value remained within a reasonably range.

4.2.8 UC3.VR.4 Rendering Time

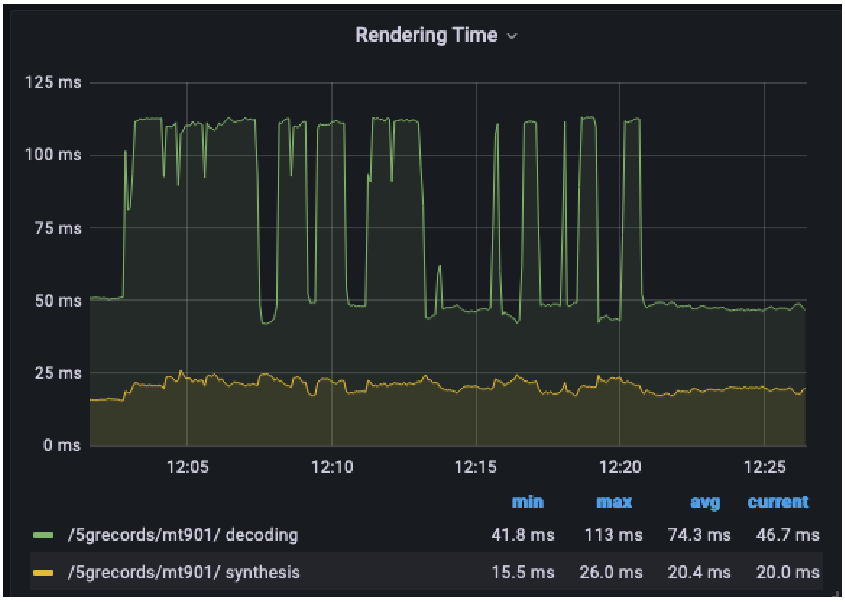


Figure 134: UC3 Rendering time.

In the graph, “synthesis” time represents the time taken by the rendering algorithm to render a view. The values obtained show that it can operate in real-time.

“Decoding” time represents the time the renderer had to wait for the original frames coming from the stream selector. The framerate for the trial was set to 15, so ideally, the sum of both numbers should be under 66 ms (1/15 s). This is not the case in some situations where some original frame is misaligned in time, because the algorithm will discard it and then wait for the next frame. This takes 66 milliseconds as explained, so the total time grows closer to 132ms, effectively skipping a frame.

4.2.9 UC3.MPR.1 Rendered View Bitrate

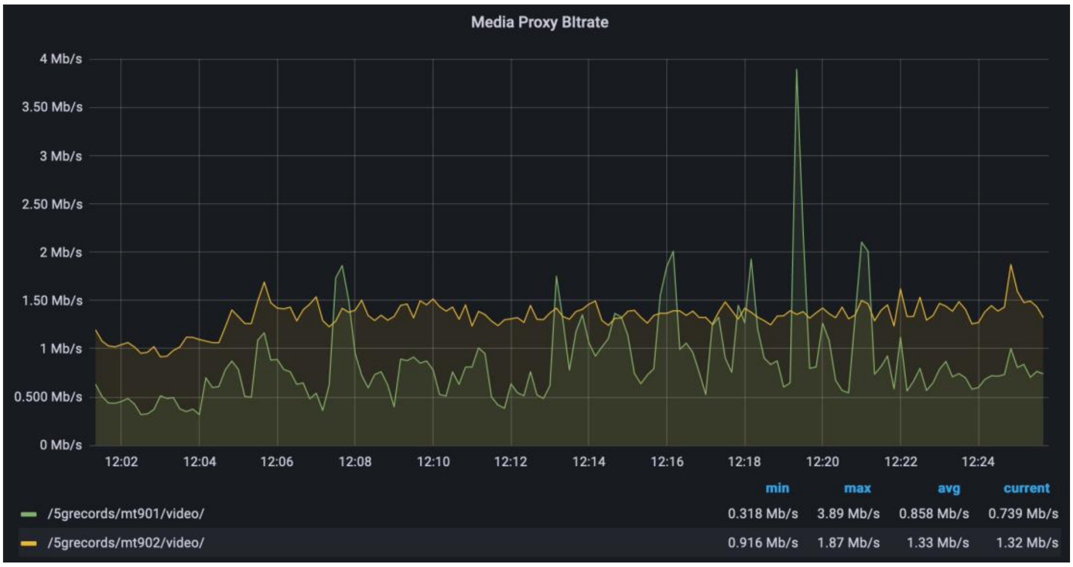


Figure 135: UC3 Media Proxy bitrate.

The graph shows the bitrate of the rendered stream, as received into (and sent from) the Media Proxy. Two video streams are monitored. Video labelled as “mt901” is controlled by the local console and manually operated by the researchers present in the trial site: sometimes the virtual camera is fixed in a set position, sometimes it is moving, according to the decision of the camera controller. The one labelled as “mt902” is the one controlled by the remote console, which was configured to use a predefined path for the virtual view, with constant virtual camera motion. For such reason, the bitrate pattern for the former presents higher temporal variation than the latter.

4.2.10 UC3.MPR.2 Rendered View Loss Rate

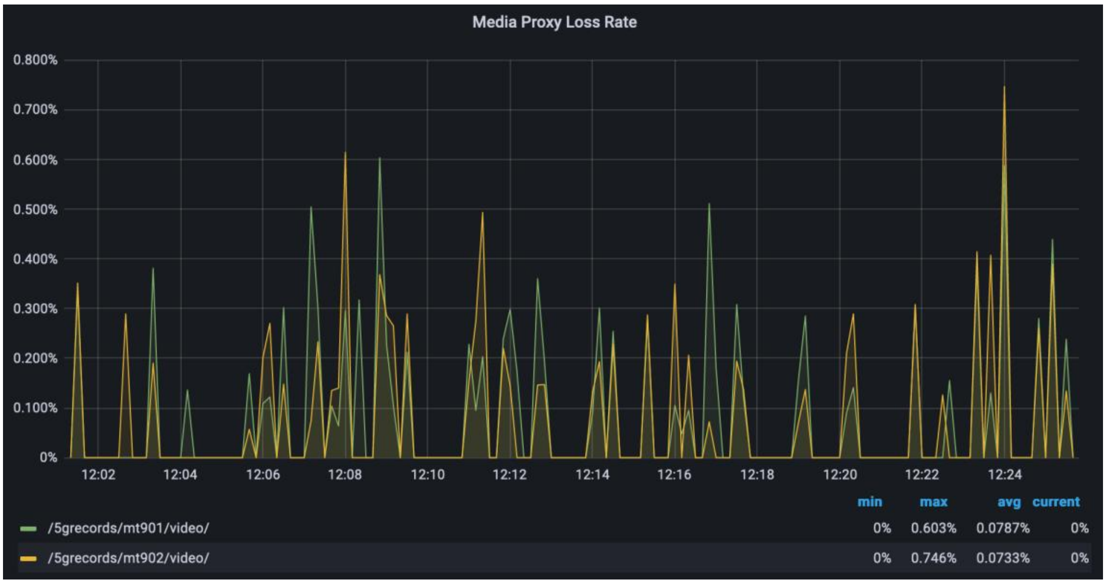


Figure 136: UC3 Media Proxy loss rate.

This graph shows the packet loss rate of the rendered streams, as received at the Media Proxy. As in the previous graph, two streams are monitored: video labelled as “mt901” is controlled by the local console and the one labelled as “mt902” is the one controlled by the remote console. The packet loss level is not null, but it is negligible and it does not perceptible affect the video quality.

4.2.11 UC3.MPL.1 QCI9 – Best Effort

Next plot shows Round Trip Time during the test. As we can see, RTT between UE and Media Delivery has experienced no significant variations during the measurement for the different video qualities tested.

As described above, each UE receives 10 parallel TCP streams in the following configuration:

- None: No additional traffic
- 5M: 5 Mbps per stream, i.e. 50 Mbps per UE and 150 Mbps in total
- 10M: 10MB Mbps per stream, i.e. 100 Mbps per UE and 300 Mbps in total
- 15M: 15MB Mbps per stream, i.e.1 50 Mbps per UE and 450 Mbps in total

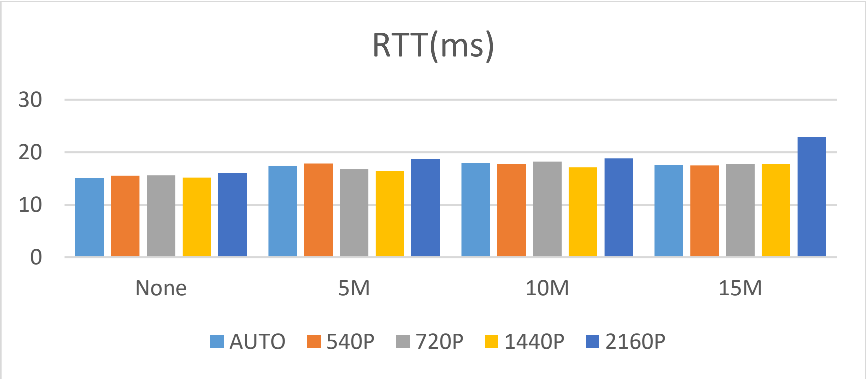


Figure 137: UC3.MPL.1 RTT results for QCI9 – Best effort.

Next plot shows Jitter during the test. As we can see, the jitter has experienced no significant variations during the measurement for the different video qualities tested.

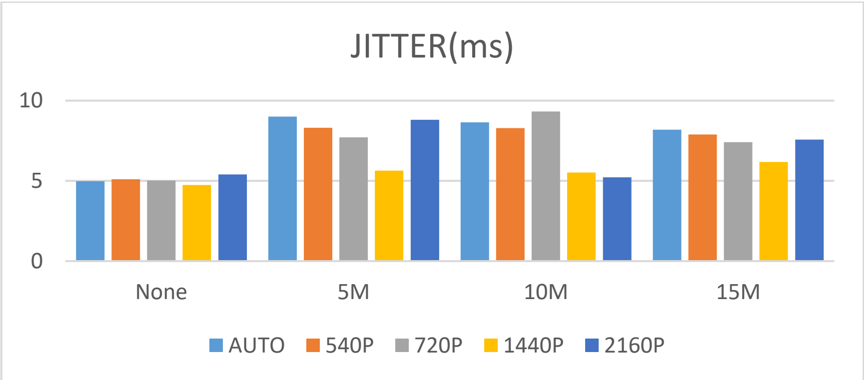


Figure 138: UC3.MPL.1 Jitter results for QCI9 – Best effort.

Next plot shows Initial load time during the test. As we can see, the initial load time has experienced no significant variations during the measurement for the different video qualities tested and is always under 1 second.

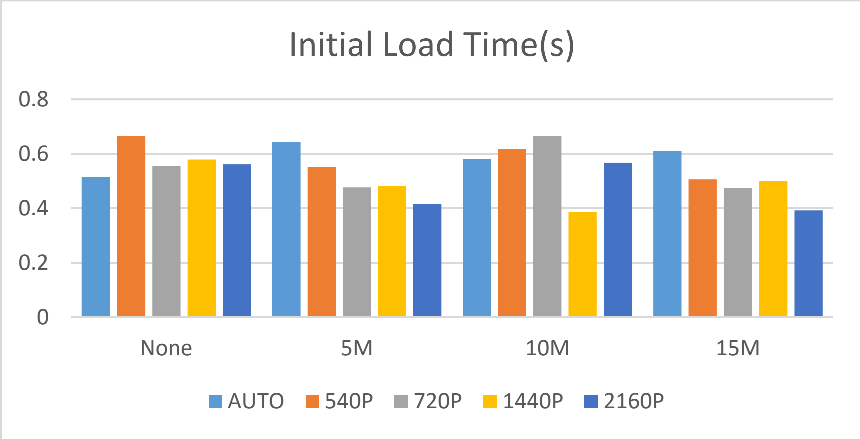


Figure 139: UC3.MPL.1 Initial Load Time results for QCI9 – Best effort.

4.2.12 UC3.MPL.2 QCI9 – Gold

Next plot shows Round Trip Time during the test. As we can see, RTT has experienced no significant variations during the measurement for the different video qualities tested.

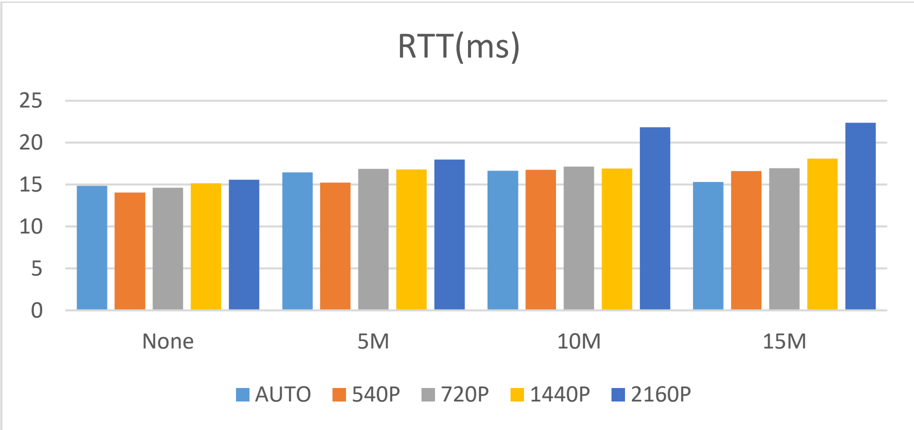


Figure 140: UC3.MPL.2 RTT results for QCI9 – Gold.

Next plot shows the jitter during the test. As we can see, the jitter has experienced no significant variations during the measurement for the different video qualities tested.

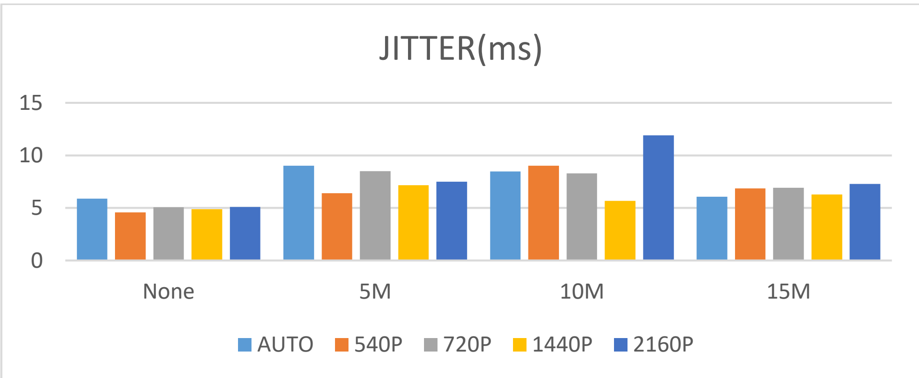


Figure 141: UC3.MPL.2 Jitter results for QCI9 – Gold.

Next plot shows Initial load time during the test. As we can see, the initial load time has experienced no significant variations during the measurement for the different video qualities tested and is always under 1 second.

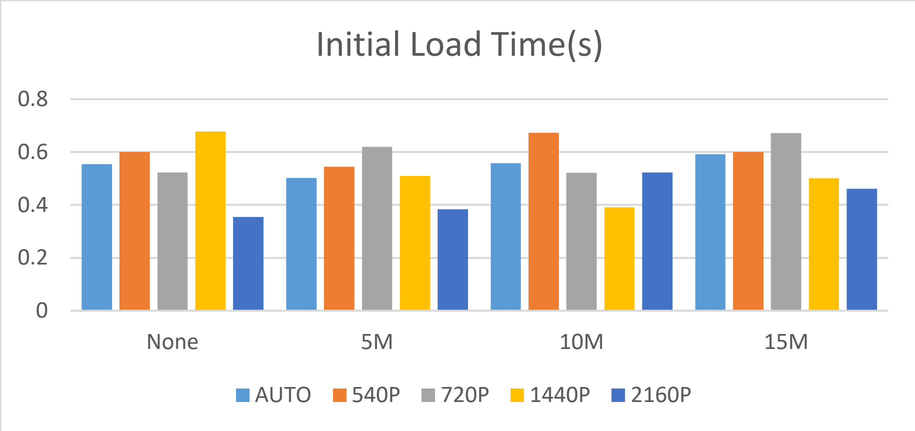


Figure 142: UC3.MPL.2 Initial Load Time results for QCI9 – Gold.

4.2.13 UC3.MPL.3 QCI6 – Best Effort

Next plot shows Round Trip Time during the test. As we can see, RTT has experienced no significant variations during the measurement for the different video qualities tested.

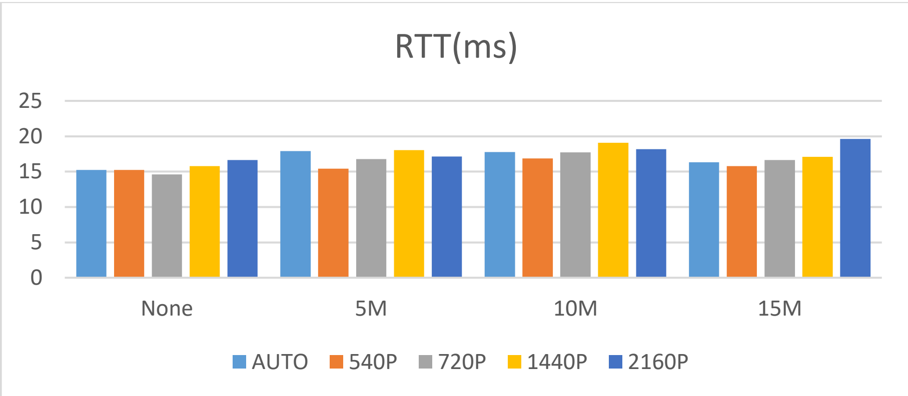


Figure 143: UC3.MPL.3 RTT results for QCI6 – Best Effort.

Next plot shows jitter during the test. As we can see, the jitter has experienced no significant variations during the measurement for the different video qualities tested.

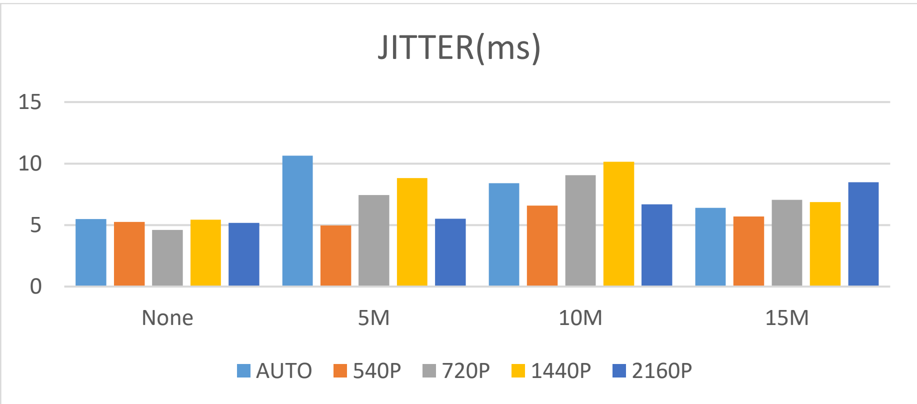


Figure 144: UC3.MPL.3 Jitter results for QCI6 – Best Effort.

Next plot shows initial load time during the test. As we can see, the initial load time has experienced no significant variations during the measurement for the different video qualities tested and is always under 1 second.

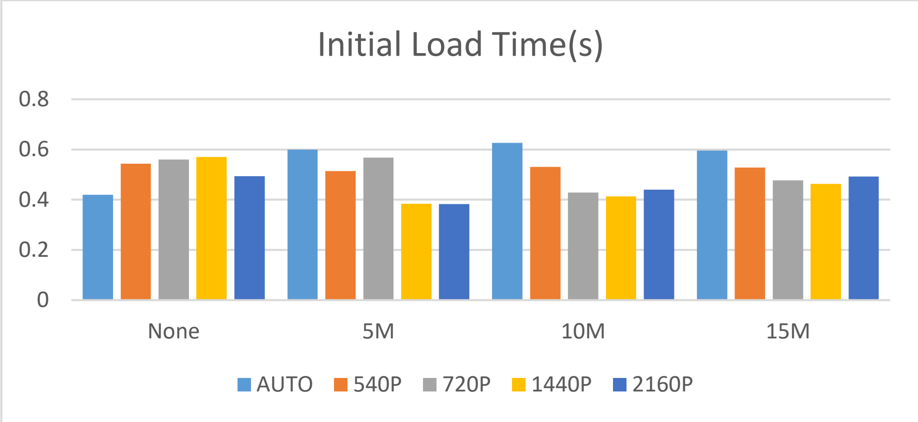


Figure 145: UC3.MPL.3 Initial Load Time results for QCI6 – Best Effort.

4.2.14 UC3.MPL.4 QCI6 – Gold

Next plot shows Round Trip Time during the test. As we can see, RTT has experienced no significant variations during the measurement for the different video qualities tested.

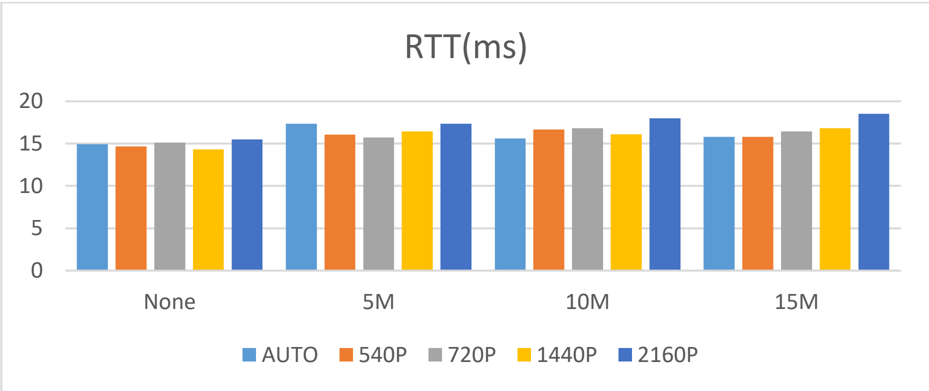


Figure 146: UC3.MPL.4 RTT results for QCI6 – Gold.

Next plot shows jitter during the test. As we can see, the jitter has experienced no significant variations during the measurement for the different video qualities tested.

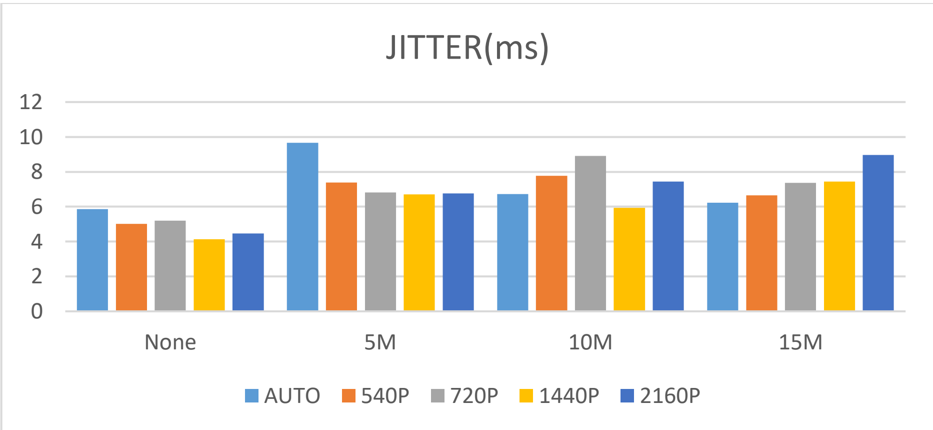


Figure 147: UC3.MPL.4 Jitter results for QCI6 – Gold.

Next plot shows initial load time during the test. As we can see, the initial load time has experienced no significant variations during the measurement for the different video qualities tested and is always under 1 second.

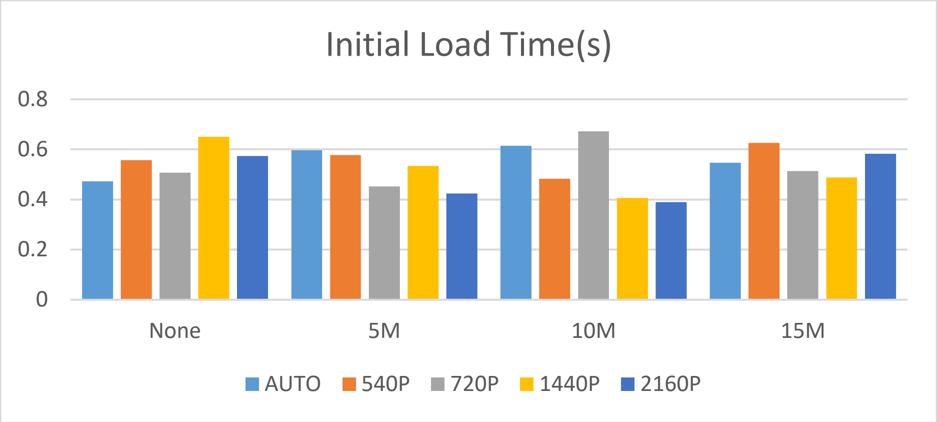


Figure 148: UC3.MPL.4 Initial Load Time results for QCI6 – Gold.

4.3 Subjective assessment tests

4.3.1 Subjective experiment

This section presents the study of the perceptual quality of different coding configurations of the Free-Viewpoint Video (FVV) system FVV Live through a subjective experiment. In addition, different pre-defined camera trajectories were considered to analyze their impact on the visual quality and their relationship with the trajectories of the observers when freely exploring the content. For this experiment, a novel test methodology was used based on the participation of few observers who repeat the test in different moments. The results provide useful insights on options to reduce, if necessary, the amount of data to deliver FVV providing the highest possible quality to the end users. Also, they can help define trajectories that can be appealing for the users if they do not have the possibility to freely navigate through the content or that can be useful to perform valid subjective tests with pre-defined trajectories. Finally, the FVV dataset that has been created and used for this experiment will be made publicly available for the research community once it is completed with more videos and results from future subjective tests.

4.3.1.1 Test Material

Five different source sequences were considered for the study. These sequences were captured by members of the *Grupo de Tratamiento de Imágenes* with the FVV Live system in *Universidad Politécnica de Madrid* (Spain) in collaboration with the theater student group *No es Culpa Nuestra* [9]. Figure 149 shows the setup used to record the sequences, while Figure 150 shows screenshots of them and their properties. All of them were acquired with the 720p resolution configuration.

Figure 149: Setup of the recordings of the source sequences.

Amarillo (SRC1)	Pelea (SRC2)	Brilla (SRC3)	Multitud (SRC4)	Fiesta (SRC5)
				
One person in the scene with reduced movement.	Few people in the scene with some fast and slow movements.	Few still people in the scene making fast movements.	A lot of people in the scene with slow movements.	A lot of people entering and leaving the scene with fast movements.

Figure 150: Screenshots and properties of the source sequences.

For the subjective tests addressed in this work, segments of ten seconds (without audio) were extracted from each source sequence. Then, in order to generate the Processed Video Sequences (PVSs, i.e. test sequences) [10] six quantization parameters (QPs: 25, 30, 35, 40, 45, and 50) were considered to encode the RGB views of the selected segments with H.264/AVC. In addition, to study the influence of the framerate on the visual quality, test sequences were generated with 15 and 30 fps. All this processing was done with Ffmpeg [11] using the Nvidia Encoder [12] for compatibility with the real-time FVV system. Finally, five different pre-defined trajectories, also known as Hypothetical Rendering Trajectories (HRTs) [13] were created using the view renderer module of the FVV Live system to be applied to the test videos:

- HRT1: Video taken from a real camera (static view without view transitions).
- HRT2: Video taken from a synthetic view located between two real cameras (static view without view transitions).
- HRT3: Horizontal sweep from the right to the left side the scene.
- HRT4: Horizontal sweep from the center of the scene, going from one side to the other and back to the center.
- HRT5: Custom trajectory generated for each particular source content by mixing horizontal sweeps, vertical sweeps and in-and-out movements.

Taking this into account, Table 10 shows the QPs used for each source clip and HRT. It is worth noting that only three QPs were used for SRC1 due to problems with the synthesis process for high QPs in this case (which will be solved for future tests). Also, specific selections of QPs were done for each source content and HRT with the generation of a quality dataset in mind with a good distribution of quality ratings covering the whole range. In addition to the QPs shown in Table 10, two versions of each source clip were created with QP 25 and 15 and 30 fps, to analyze the influence of the framerate. Thus, this results in a total of 120 PVSs.

Table 10: QPs used to create the PVSs for each source content.

HRT/SRC	SRC1 (QP1 to QP3)	SRC2-SRC5 (QP1 to QP4)
HRT1	25, 30, 35	25, 35, 45, 50
HRT2	25, 30, 35	25, 35, 40, 45
HRT3	25, 30, 35	25, 35, 45, 50
HRT4	25, 30, 35	25, 35, 45, 50
HRT5	25, 30, 35	25, 35, 40, 45

4.3.1.2 Evaluation Methodology

The subjective tests were divided into two phases. In the first one, the set of video sequences with the encoding configurations and pre-defined trajectories described in Subsection 4.3.1.1 were evaluated by the participants. After each test sequence, a grey screen was displayed for five seconds, so the participants were able to rate them using the five-grade quality scale (from 5: excellent to 1: bad) [14] on a smartphone with an online form. The recently proposed methodology Few Observers With Repetitions (FOWR) [10] was used, which, basically, consist in using the Absolute Category Rating (ACR) method, but with a reduced number of participants that repeat the tests in distant sessions. In our case, four participants (all watched the videos at the same time) rated all the test sequences in four sessions that took place in four subsequent days. This way, according to the authors [10], it is possible to obtain comparable quality assessments to those obtained with fifteen subjects in a typical subjective test with 2D videos. In our study, we will explore if this method can also be valid for FVV sequences. It is worth noting that the presentation order of the test sequences was randomized for each session, with the conditions of not showing the same source content consecutively. Also, each session was divided into two parts of fifteen minutes approximately with a break of five minutes in between and a training session at the beginning showing some video samples with different qualities.

Once the four sessions of the first phase of the tests were completed, the second phase of the tests was performed individually by each participant. In this phase, the participants rated the quality (using the same scale and questionnaires as in the first phase) of fifteen test sequences selected from the total set (QPs 25, 30, and 35 were used for SRC1 and 25, 35, and 45 for the rest, see Table 10) after freely exploring them using the arrows of a keyboard to move horizontally and vertically (within a limited range along the surface of a sphere). This session lasted around twenty minutes, including a training phase at the beginning of it, so the participants could familiarize with the system.

Prior to the first phase of the tests, an introductory session was performed with the participants. In this session, they passed a visual screening and they received instructions and an explanation of the tests to clarify any doubts or questions. In addition, the participants filled a consent form and a background questionnaire.

4.3.1.3 Environment and Equipment

The subjective tests were carried out in a room with controlled ambient light to avoid disturbing reflections. For the first phase of the tests, a 55-inches curved screen Samsung HU8500 was used to display the videos for all the users. They were played in full screen using VLC media player[5] running on a computer connected to the screen by HDMI, which allowed a smooth playback of the sequences.

For the second phase of the tests, a custom software that controls the synthesis module of the system was used to allow the free navigation of the observers using the arrows of a keyboard. In this case, the video sequences were displayed on a 22-inches monitor Samsung T220HD.

4.3.1.4 Participants

Four participants (two males and two females) took part on the tests, with an average age of 24 years and all of them with normal or corrected-to-normal vision. All of them were naive in terms of participation in quality evaluation tests and in watching FVV contents. They were economically rewarded for their participation in the tests.

4.3.2 Results

4.3.2.1 Compression

The Mean Opinion Scores (MOSs) obtained for the different encoding configurations and for each of the considered HRTs are shown (with 95% confidence intervals) in Figure 151. Specifically, Figure 151(a) shows the results obtained aggregating all the SRCs, except SRC1 since the QPs used in this case were different due to synthesis problems. In general, as expected, the higher the QP (QP1 is 25) the lower the perceived quality with all HRTs, although the differences between QP1 and QP2 are imperceptible, as well as between QP2 and QP3 in certain cases as for HRT2. Also, HRT1 offers the highest quality, since it provides a fixed viewpoint of a real camera, so there are not synthesis artifacts. In fact, for the rest of the trajectories even the lowest QP do not offer a good quality (MOSs around 3), due to the annoyance of the synthesis artifacts. HRT2 also shows a fixed viewpoint but from a synthesized view, however, although synthesis artifacts appear, they may be less annoying than for HRT3-HRT5 since there are no camera movements, which is especially evidenced in low qualities. In addition, Figure 151(b)-(f) show the individual results for each source content. Firstly, it is possible to see the small differences among the results obtained for SRC1, since three high QPs were used due to the aforementioned problems. Then, although comparable results were obtained for the three dynamic HRTs for the other SRCs, it is worth noting that HRT5 provides higher MOSs for lower qualities, probably thanks to the effect of showing a more complex and exploratory trajectory that can be more appealing for the observers. Also, HRT5 seem to offer a better distribution of the MOSs with the QPs, with a more defined staircase shape in general. These results from the exploratory test show the expected tendencies and provide useful insights, however, in the future, they will be compared and validated with deeper statistical tests and further experiments with conventional test methodologies (e.g., ACR [15]).

[5] https://www.videolan.org/vlc/

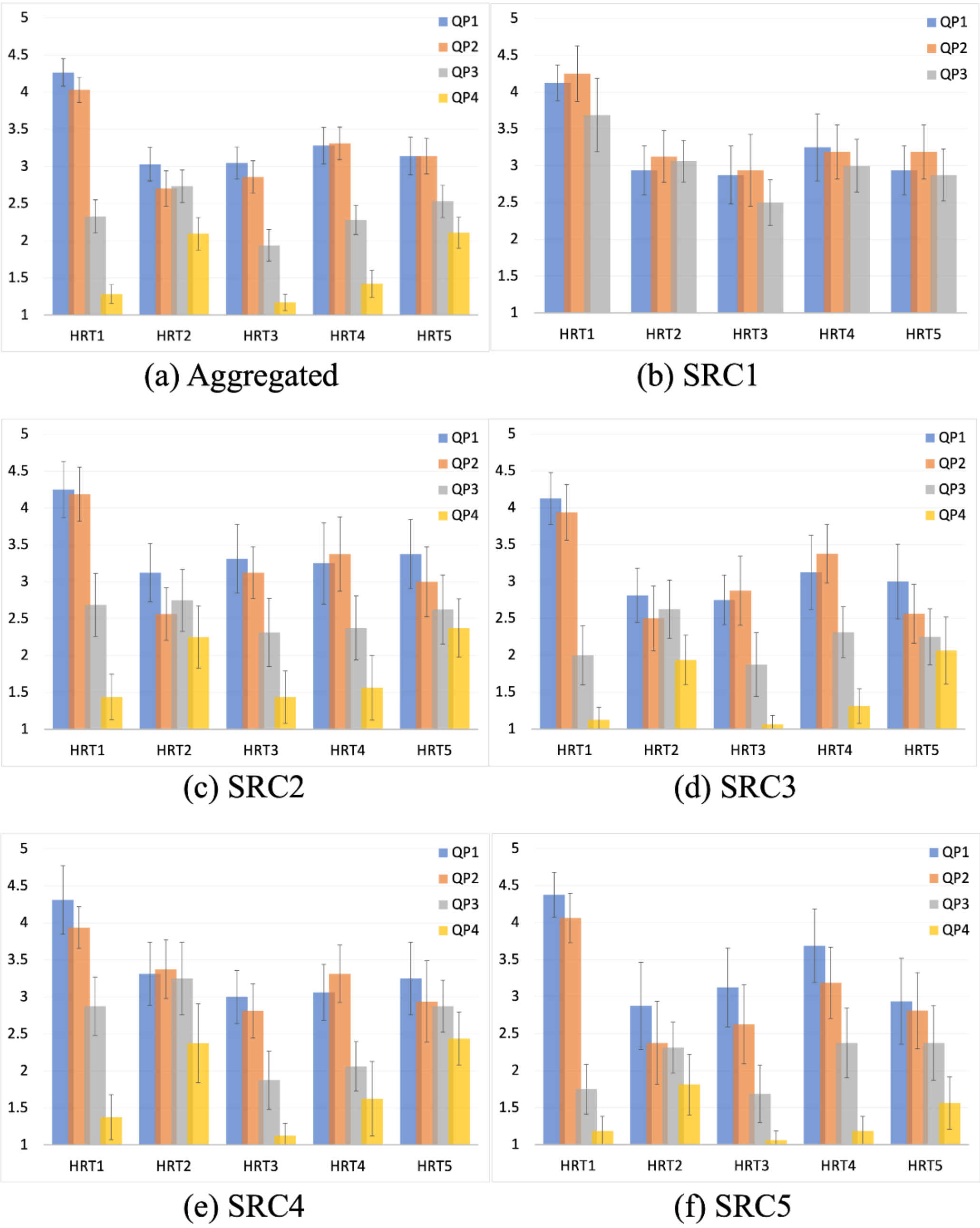


Figure 151: Results of the perceptual quality for the different coding configurations: a) Aggregating all SRCs except SRC1, b) - f) Results for each SRC content.

4.3.2.2 Framerate

To analyze the effects of the framerate in the quality perceived by the users, Figure 152 show the MOSs obtained for the test videos with QP 25 and 15 and 30 fps for each of the considered HRT. In particular, Figure 152(a) shows the aggregated results for all the SRC contents, which evidence of the perceptual quality reduction when the framerate is reduced from 30 to 15 fps. However, the results for each SRC content, shown in Figure 152(b)-(f), reflect that the effect of the framerate is highly dependent on the content. For example, it can be seen that for SRC1 and SRC4, which are contents with slow movements of the actors, the differences in the perceived quality with 30 fps and 15 fps are minimal. Thus, in certain cases, decreasing the framerate can be a good option to reduce the amount of data to store or transmit FVV contents.

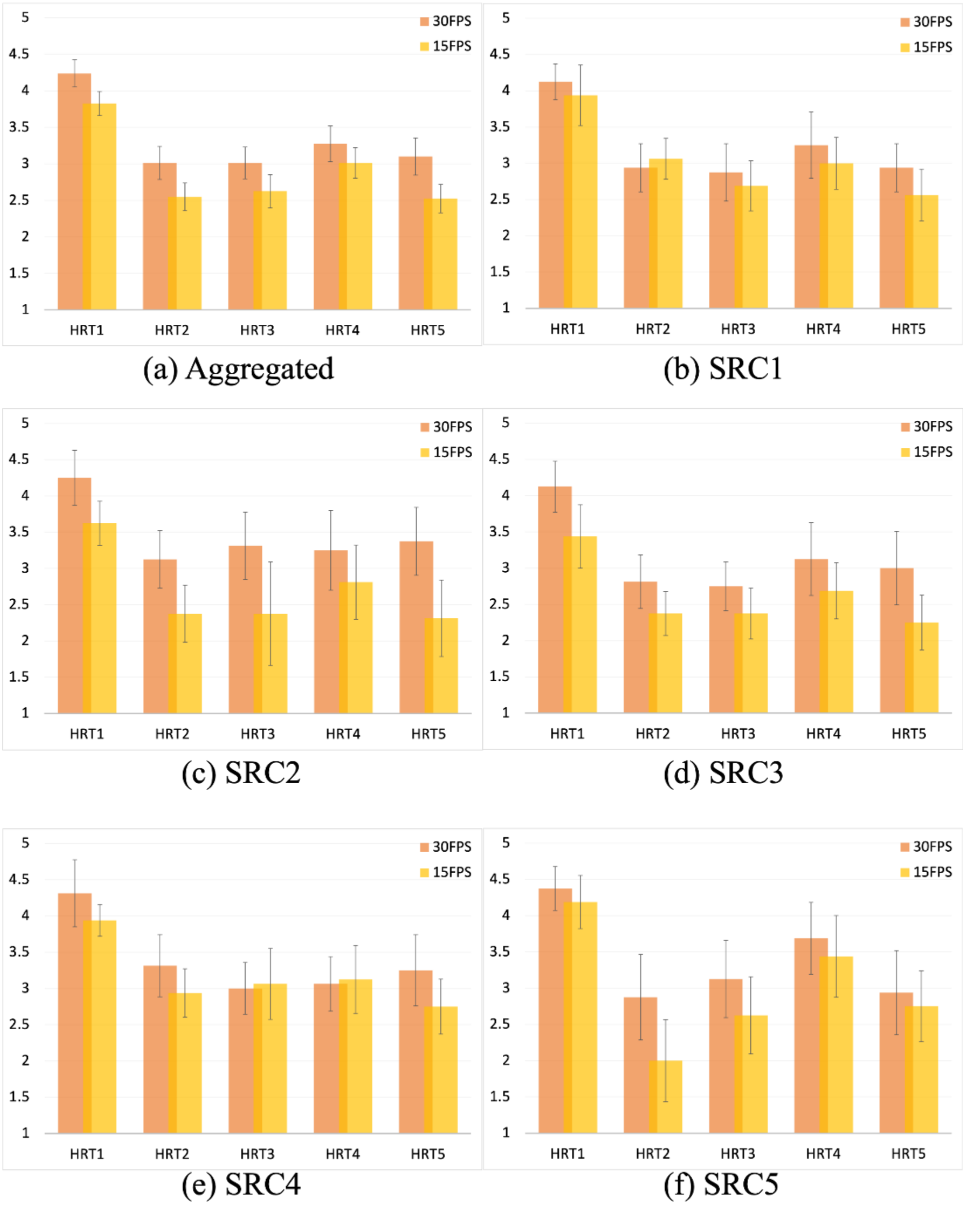


Figure 152: Results of the perceptual quality for the different considered framerates: a) Aggregating all SRCs, b) - f) Results for each SRC content.

4.3.2.3 Free navigation vs. pre-defined trajectories

In this subsection, the results obtained from the evaluations provided by the participants after freely navigating through the test sequences are compared with those obtained when showing predefined trajectories. Firstly, Table 11 shows the Pearson correlation coefficients between the scores obtained with free navigation and each of the considered HRTs. As it can be seen, in addition to HRT5, which was the pre-defined trajectory with more degrees of movement (horizontal vertical and in/out), the two HRTs based on static viewpoints (HRT1 from a real camera and HRT2 from a synthesized view) are those providing a higher correlation with the scores obtained from the free-navigation assessments. To further analyze the relation among the scores provided after free navigation and the pre-defined trajectories, Figure 153 shows the MOSs obtained for the three QPs considered in both phases of the tests. This figure shows that there are no big differences among the pre-defined trajectories except for HRT1, which offers the best quality since it shows a viewpoint from a real camera and, thus, without synthesis artifacts. The high scores obtained after free navigation (even comparable to those obtained with HRT1) show how offering this possibility improves the quality experienced by the observers, possibly masking or diminishing the effect of compression and synthesis artifacts.

Table 11: Pearson correlation between the scores obtained in the free navigation (FN) session and with the pre-defined trajectories (HRTs).

Trajectories	HRT1	HRT2	HRT3	HRT4	HRT5
FN	0.781	0.792	0.694	0.741	0.786

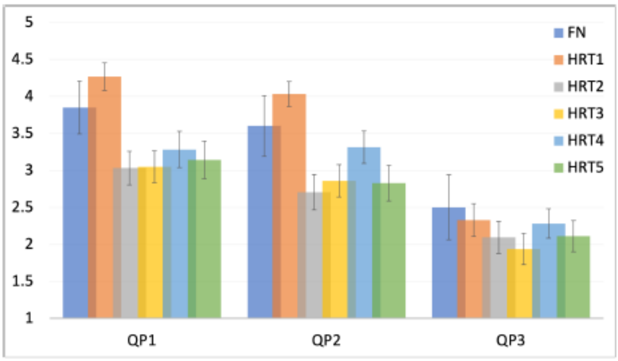


Figure 153: Perceptual quality results for the considered three QPs with free navigation and the pre-defined trajectories.

4.3.2.4 Evaluation methodology

Given the exploratory aspect of this subjective test in terms of using the FOWR [10] methodology with FVV content, this subsection presents the analysis of the scoring behaviour of the participants. On one side, it can be expected that the participants somehow learn how to assess the quality as they perform more sessions. In this sense, Table 12 shows the Pearson correlation coefficients between the scores provided by each participant in each session with the averaged global scores (MOSs). As it can be seen, there is a convergence of the scores in the last day with acceptable values of correlation with the MOSs, considering the values indicated in international recommendations in relation with conventional subjective tests [15].

Table 12: Pearson correlation between the scores provided by each participant in each session with the global MOSs.

Session / Participant	P1	P2	P3	P4
Session 1	0.681	0.739	0.758	0.770
Session 2	0.667	0.812	0.825	0.817
Session 3	0.705	0.841	0.677	0.751
Session 4	0.775	0.839	0.767	0.836

To better understand the behaviour of the participants when scoring the FVV contents following the FOWR methodology, the obtained distribution of scores is shown in Figure 154. Although the distribution follows the typical shape obtained in visual quality tests, it can be noticed how participant 4 seems to be more "negative", providing more scores of the lower part of the scale, while participant 1 is the one providing more higher scores.

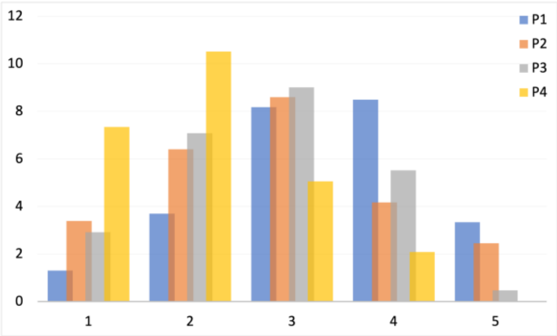


Figure 154: Distribution of the scores (in percentage) given by each participant (P).

Nevertheless, to validate this methodology to be used in tests with FVV content, the results should be compared with those obtained from a subjective test with the same test stimuli and conditions but following a conventional methodology (e.g., ACR with at least 15 participants [15]). This is one of the activities envisioned for future work.

4.4 KPI analysis

This section discusses the level of achievement for the selected KPIs, according to the results of the field trial, the integration tests (reported in D4.2 [6]), and the subjective assessment tests.

4.4.1 Motion-to-photon latency < 170 ms

Motion-to-photon (MTP) latency describes the delay experience by the virtual camera operator using the production console. A specific tool has been developed to measure it (see D4.1 [4]). It can be measured in the integration or trial environment, but not while live operation.

Formal measures were done in the field trial environment in both phases of the project:

- In the 5G network in Segovia (phase 1), achieving around 290 ± 80 ms (D5.2 [1]).
- In the compact 5G deployment in Madrid (phase 2), achieving 210 ± 20 ms (D4.2 [6]).

Additionally, it was measured in an emulated 5G link with higher uplink throughput (next generation of 5G mmWave networks), achieving 145 ± 30 ms (D4.2 [6]).

Consequently, it can be concluded that:

- The target motion-to-photon latency < 170 ms was **not fully achieved** with the existing FVV + 5G network configuration. However, **the obtained result (210 ± 20 ms) is close enough** (25% higher) so that the effective Quality of Experience is not affected significantly.
- The **next generation** of 5G networks, with higher uplink capacity, will be able to achieve the target of 170 ms.
- Current deployment has been validated to be usable in the field, even if MTP latency sometime increases up to 25 % above the original target.

4.4.2 Uplink bitrate

Uplink bitrate is the critical KPI for the deployment of FVV cameras in the field so that FVV production is possible. This KPI is a trade-off between the bitrate generated from each camera (the lower the bitrate, the higher the number of cameras that can be allocated), the uplink capacity of each UE, and the total aggregated capacity of the 5G deployment (which determines the maximum number of cameras that can be supported). Therefore, this requirement is split in three.

4.4.2.1 Camera bitrate: 50 Mbps (ideal) – 100 Mbps (max)

Each camera stream (RGB+Depth) should be less than 100 Mbps, and ideally less than 50 Mbps. As seen before, this bitrate is based on 2 components:

- The RGB stream is compressed at a constant bit rate. This bitrate is selected based on the resolution, frame rate, and expected complexity of the scene, and it is always a fraction of the depth stream. The subjective tests have shown that, provided that the bitrate is above 1250 kbit/s (which is achieved, in general, for all the configurations in the project), compression in the RGB stream does not affect quality.
- The Depth stream is compressed following a lossless scheme which produces a variable bitrate, which depends on the resolution, frame rate, and complexity of the scene. It accounts for the most part of the bit rate.

As seen in the final field trial, it is possible to obtain functional FVV production over a moderately complex scene (two people with some motion, several objects) using a configuration of **720p15**. This results in **25 Mbps** of average bitrate (33 Mbps maximum), which is clearly within the ideal target bitrate.

In D4.2 [6], several scenes were tested at **1080p30**, which is the highest resolution and frame rate supported by the system. This resulted in streams between **15** Mbps (simple scene) and **72** Mbps (super-complex scene) per camera, meaning that the general target of less than 100 Mbps per camera is always achieved, and the ideal target of less than 50 Mbps per camera is achieved in most scenarios (all except the super-complex one).

4.4.2.2 Capture UE UL: 150 Mbps (min) – 300 Mpbs (ideal)

In our reference architecture, each capture server processes 3 cameras and is connected to a 5G UE. Therefore, each UE should support between 150 (for "ideal" cameras) and 300 Mbps (for any camera).

Two types of measures have been done to validate this aspect.

On the one hand, several iperf3 measures have been done to test the overall capacity of the uplink, reported in D4.2 [6]. Using TCP it is possible to obtain the highest supported throughput in a **stable** situation, without saturating the network, which is **170-180 Mbps**. This same bitrate was also supported in the outdoor field deployment in Segovia tested in phase 1 (D5.2 [1]), where we verified that such high bitrates can be kept at about **70 m** distance from the antenna, provided that there is line of sight. UDP tests were used to measure the **peak capacity** of the uplink (in saturation circumstances), which is **250 Mbps** (D4.2 [6]).

On the other hand, measures have been done with actual FVV streams, to ensure that the traffic characteristics of the FVV cameras are supported with low packet loss ratio (< 1% in all cases). In the live trial, the sustained throughput in the UE was about 80 Mbps, and the system was fully performant for 30 minutes. Shorter tests have been done both in Madrid environment (indoor link, D4.2 [6]) and Segovia environment (outdoor link, D5.2 [1]), supporting sustained bitrates up to **168 Mbps**.

4.4.2.3 Total # cameras: 3-5 (min) to 9-12 (ideal)

The total number of cameras deployable in the system depends on:

- The number of simultaneous cameras supported by the FVV system.
- The total uplink capacity supported by the 5G deployment.

Regarding the former, the system has been consistently tested with three capture servers which were able to support up to 9 cameras, which is part of the ideal scenario. This includes the live trial configuration.

Regarding the latter, each mmWave cell (with 2CC UL) can support up to 170-180 Mbps of sustained traffic. This means that:

- With one single cell, it is possible to support 3 to 5 cameras simultaneously over the RAN. Numerous tests have been performed of this, both in Madrid and Segovia environments (see e.g. D4.2 [6] and D5.2 [1]).
- It is possible to support 9 cameras by deploying 3 cells in current configuration, or one cell in the next-generation configuration (8CC UL). This has been tested only using simulation (since only one cell was physically available in the live trial).

4.4.3 RTT UE-MEC < 40 ms

Keeping a consistent low delay between UE and MEC is key to support interactivity and low motion-to-photon latency.

As seen in the previous deliverables (D4.2 [6], D5.2 [1]), the RTT between the UE and the MEC in our environment is about 12 ms in idle conditions, and 25 ms in load conditions (but without saturation). When the network is saturated, with uplink traffic loads higher than 150-180 Mbps, then the ping time increases above 100 ms.

As shown in the live trial reported in this deliverable, the FVV system must work in non-saturation conditions. In such conditions, the **target KPI of RTT < 40 ms has been kept during the whole field trial** (30 minutes), except in some individual moments where the network showed a saturation situation.

4.4.4 Virtual View Frame Rate: 15 fps (min) - 30 fps (ideal)

Due to the complexity of the scene, the system was configured to work at 15 fps. According to the results of the subjective tests performed, this reduction of temporal resolution does not greatly affect the quality of experience of the streaming.

In the realistic scenario of the trial, problems such as packet losses and latency differences between servers cause frame drops in the rendered video. Even though these problems only appear momentarily in very challenging situations, they reduce the average framerate.

4.4.5 Remote user throughput: up to 50 Mbps

With the different video resolutions that have been used in the tests and demos of this project, the 50Mbps barrier never has been exceeded.

Table 13: UC3 video resolutions and bitrates.

Quality ID	Resolution	Reference Bitrate (Mbps)
SD	960x540	1.5
HD	1280x720	3.0
Full-HD	1920x1080	6.0
3K	2560x1440	12.0
4K	3840x2160	24.0

4.4.6 Remote user QoE

These are the expected delivery KPIs as defined in Deliverable D2.1 [3].

Table 14: UC3 expected delivery KPIs (see D2.2).

User	Protocol	Initial Load Time	Pause Count
Premium remote End-users (GOLD)	HTTP	<=1s	= 0
Regular remote End-users (Best Effort)	HTTP	<=3s	<=1

During the final trial for UC3, measurements have been taken to validate these KPIs have been met.

4.4.6.1 Scenario 1: QCI9 – Best Effort

With the results obtained with this setup, it can be certified that the expected KPIs have been met. Initial Load time was maintained constantly under 1 second. There were no pauses during the medial delivery.

Selected Quality2	Iperf Bandwidth Megabytes/s	rtt	jitter	initialLoadTime	pauseCount
auto	None	15,096616	4,98875722	0,516	0
auto	5M	17,4192409	9,01100569	0,644	0
auto	10M	17,8932512	8,65419134	0,58	0
auto	15M	17,5968922	8,18354129	0,611	0
540p	None	15,5421715	5,10663104	0,665	0
540p	5M	17,8754212	8,31181966	0,551	0
540p	10M	17,7464442	8,29234511	0,617	0
540p	15M	17,493112	7,90048228	0,506	0
720p	None	15,6115295	5,01588401	0,555	0
720p	5M	16,7552159	7,70958613	0,477	0
720p	10M	18,2010622	9,33105607	0,666	0
720p	15M	17,7852395	7,41253936	0,475	0
1440p	None	15,1584737	4,74037926	0,579	0
1440p	5M	16,4480237	5,64907439	0,483	0
1440p	10M	17,1250109	5,5247643	0,386	0
1440p	15M	17,754854	6,18356604	0,5	0
2160p	None	15,9947632	5,3932082	0,562	0
2160p	5M	18,6903734	8,80576451	0,416	0
2160p	10M	18,8531799	5,22275506	0,567	0
2160p	15M	22,8818098	7,56705616	0,392	0

Figure 155: UC3 results for Scenario 1 (QCI9 – Best Effort).

4.4.6.2 Scenario 2: QCI9 – Gold

With the results obtained with this setup, it can be certified that the expected KPIs have been met. Initial Load time was maintained constantly under 1 second.s

There were no pauses during the medial delivery.

Selected Quality	erf Bandwidth Megabytes/	rtt	jitter	initialLoadTime	pauseCount
auto	None	14,84032404	5,890121969	0,554	0
auto	5M	16,46483011	9,004713661	0,502	0
auto	10M	16,64874413	8,459645923	0,558	0
auto	15M	15,31309873	6,064651734	0,592	0
540p	None	14,04951717	4,566274845	0,6	0
540p	5M	15,22041881	6,399197805	0,545	0
540p	10M	16,76673617	9,01057941	0,673	0
540p	15M	16,62764509	6,84796265	0,6	0
720p	None	14,63732158	5,076947253	0,523	0
720p	5M	16,8898478	8,505197036	0,62	0
720p	10M	17,1395524	8,286234866	0,522	0
720p	15M	16,97114316	6,907996991	0,672	0
1440p	None	15,16849013	4,87787154	0,678	0
1440p	5M	16,79629554	7,161754173	0,509	0
1440p	10M	16,91819006	5,662658214	0,391	0
1440p	15M	18,08809456	6,268696971	0,501	0
2160p	None	15,56968059	5,099203362	0,354	0
2160p	5M	17,997856	7,485188882	0,384	0
2160p	10M	21,83731963	11,89770318	0,523	0
2160p	15M	22,37585985	7,292905694	0,461	0

Figure 156:UC3 results for Scenario 2 (QCI9 – Gold).

4.4.6.3 Scenario 3: QCI6 – Best Effort

With the results obtained with this setup, it can be certified that the expected KPIs have been met. Initial Load time was maintained constantly under 1 second.

There have been no pauses during the medial delivery.

Selected Quality	Iperf Bandwidth Megabytes/s	rtt	jitter	initialLoadTime	pauseCount
auto	None	15,22773738	5,498141624	0,419	0
auto	5M	17,89839118	10,63767098	0,6	0
auto	10M	17,75167017	8,399730761	0,626	0
auto	15M	16,33985208	6,39204276	0,596	0
540p	None	15,23658228	5,262684735	0,544	0
540p	5M	15,3968747	4,965193612	0,514	0
540p	10M	16,87001447	6,576825276	0,53	0
540p	15M	15,79154799	5,698371925	0,528	0
720p	None	14,59328856	4,602384138	0,56	0
720p	5M	16,76701449	7,434042202	0,567	0
720p	10M	17,72863388	9,050781764	0,428	0
720p	15M	16,629499	7,053972915	0,477	0
1440p	None	15,76496431	5,444804633	0,57	0
1440p	5M	18,01550739	8,816447344	0,384	0
1440p	10M	19,05800734	10,15634149	0,413	0
1440p	15M	17,06643378	6,863677162	0,463	0
2160p	None	16,61614047	5,185052348	0,494	0
2160p	5M	17,11894479	5,51725616	0,383	0
2160p	10M	18,15999132	6,699667993	0,44	0
2160p	15M	19,61052305	8,473251741	0,492	0

Figure 157: UC3 results for Scenario 3 (QCI6 – Best Effort).

4.4.6.4 Scenario 4: QCI6 – Gold

With the results obtained with this setup, it can be certified that the expected KPIs have been met. Initial Load time was maintained constantly under 1 second.

There were no pauses during the medial delivery.

Selected Quality	Iperf Bandwidth Megabytes/s	rtt	jitter	initialLoadTime	pauseCount
auto	None	14,9408573	5,861499622	0,473	0
auto	5M	17,34189887	9,656286704	0,597	0
auto	10M	15,62995713	6,719114986	0,615	0
auto	15M	15,81836792	6,224208054	0,547	0
540p	None	14,64890863	5,00705174	0,557	0
540p	5M	16,06038063	7,382654214	0,578	0
540p	10M	16,6762004	7,767521662	0,483	0
540p	15M	15,81573409	6,652353352	0,626	0
720p	None	15,10604657	5,198436564	0,507	0
720p	5M	15,72632522	6,80999358	0,452	0
720p	10M	16,83342044	8,9134883	0,672	0
720p	15M	16,45239928	7,372104719	0,513	0
1440p	None	14,33578973	4,127313892	0,65	0
1440p	5M	16,44629459	6,705067961	0,534	0
1440p	10M	16,10526674	5,933325258	0,406	0
1440p	15M	16,83586317	7,445799317	0,488	0
2160p	None	15,4825955	4,464671727	0,573	0
2160p	5M	17,33287521	6,759824531	0,424	0
2160p	10M	18,00355031	7,440530294	0,39	0
2160p	15M	18,52911158	8,957785549	0,583	0

Figure 158: UC3 results for Scenario 4 (QCI6 – Gold).

4.5 Technology validation outcome

This section discusses the technology validation outcome of the different components of UC3.

4.5.1 5G-Ready FVV Live

FVV Live is a system capable of working in real time in a very controlled network with very high bandwidth cable links. This system has been modified and its parameters have been tweaked so it was able to operate on a much more challenging environment, a 5G network and a cloud environment. Some of these upgrades involve easily tracking network parameters, reordering received packets, smoothing the traffic and being able to deploy several renderers as docker containers.

The Capture module is designed to be as portable as possible, using light consumer grade cameras easily deployable in a free configuration without racks. It is able to capture both audio and video, and to compute the geometric information of the scene in the shape of depth images. It also handles the compression and transmission of the media.

The Stream Selector is capable of receiving all the media and then manages its re-transmission to multiple view renderers, which then render the virtual view in the desired position. This means the system can work in a cloud-like environment, where multiple renderers can be deployed as Docker containers at the same time, so different camera trajectories can be produced over the same media.

The Production Console is a very light software with little requirements that can run on almost any device. It connects to the View Renderer with minimal latency and has an intuitive user interface to easily control the virtual camera.

Lastly, recorded sequences can be stored and then replayed in a transparent way to the Stream Selector and the View Renderer, meaning no new software has to be deployed to generate new camera paths over the recorded media.

In terms of appealing camera paths that can be generated to provide a good quality of experience to the end users, the subjective results showed that, although horizontal camera paths can offer acceptable impressions of the FVV content to the users, using more customized trajectories to the specific content with more degrees of movement (horizontal vertical and in/out) can improve the QoE. Nevertheless, the subjective results also showed the potential of offering free navigation possibilities to the end users, which provides a more immersive experience and improves the QoE, possibly masking or diminishing the effect of compression and synthesis artifacts.

4.5.2 Compact 5G + MEC + Media Delivery deployment

The 5G deployment designed for this use case has been proven to be valid to support live immersive media production scenarios and, in particular, the production of FFV over 5G networks. The system has been validated in two different deployment scenarios:

- A pilot network on the public infrastructure of Telefónica in Segovia (Spain). A gNodeB was deployed on the trial site (about 10 km from Segovia downtown) and an edge core (UPF) and MEC platform was deployed in a Central Office in Segovia. The base station and the MEC were integrated within Telefónica transport network.
- A non-public network deployed in Nokia premises in Madrid (Spain). The deployment included a gNB, network core and MEC on a compact platform using Nokia AirFrame hardware. The whole network was connected to Telefónica transport network using a residential FTTH access.

The tested solution uses 5G RAN in FR2 (mmWave band), in a 8CC DL / 2CC UL configuration with TDD 4DL/1UL, aggregating a total of 8 carriers of 100 MHz each. The solution uses 5G in NSA 3.x configuration. During the life of the project, it has become clear that FR2 is going to be supported mostly in NSA configuration in the short future, due to the unavailability of chipsets supporting SA in such frequency range. Next generation configuration will be able to aggregate more carriers in uplink, getting to 4CC and 8CC configurations. In 5G Records, these configurations have also been tested using a software RAN emulator developed by Nokia.

Regarding the MEC platform, the validated solution uses Nokia AirFrame OpenEdge server with 2 Nvidia Tesla T4 GPUs onboard. Besides this configuration, tests have been done using a GeForce 3090 GPU to assess the performance of the system for the next generation of edge cloud GPUs.

As seen previously in the KPI validation section, as well as in previous deliverables, both deployments offered similar capacities in terms of achieved performance. It was shown that such performance is enough to provide a first generation of FVV services. Besides, by using network emulation and testing the performance of different GPU families, it has also been established that the next generation of the platform (5G network and MEC) will be able to support more advanced functionality of the FVV system (higher resolution and frame rate).

A complementary component to the 5G RAN and Edge infrastructure is the Media Delivery VNF. The Media Delivery VNF can support several media delivery functions, including traffic routing at RTP level, encapsulation in TCP to prevent packet losses in the backbone (transport) network, hardware and software-supported transcoding, real-time segmentation, adaptive streaming delivery (HLS), and KPI monitoring. This component is a complement to the FVV Live system, and it can be used to package and deliver the output of the production subsystem (the rendered views) to end users, as well as to support the connectivity of a remote production console to the compact network. Therefore, it has been proved to be a very flexible component to connect live immersive media production functions with content delivery networks.

4.5.3 Edge Cloud and SDN

The Edge Cloud and SDN infrastructure deployed in the Use Case 3 (Figure 159) has proved to be appropriate for media delivery required for the use case. Two different Quality of Service have been defined and used ("Best Effort" and "Gold") to transport the video content during the tests and trials. Additionally, traffic noise has been generated during the tests to stress the network and compare the behaviour of the two qualities defined.

As reported in D4.2 [6], the Gold QoS improves the video delivery in heavy traffic conditions, while in low traffic conditions, the results are similar, as expected.

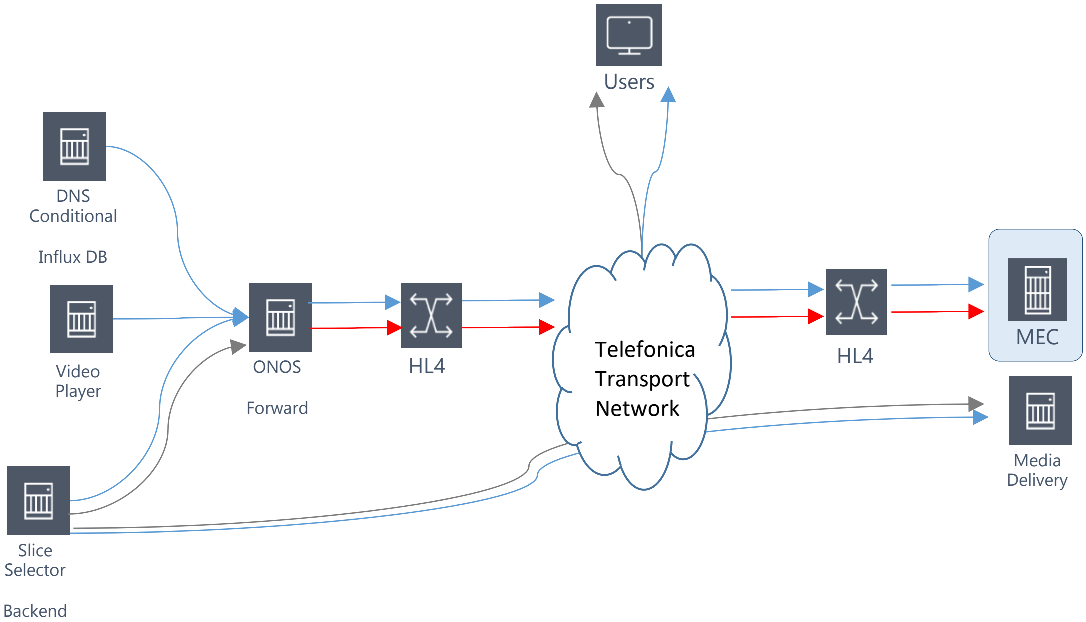


Figure 159: UC3 High level SDN & Edge setup.

During the final trial, there has been no noise traffic in the transport network segment, only in the Radio Access. The results depicted in sections 4.2.10 and 4.4.6 shows no significant differences between "Best Effort" and "Gold" as the Radio Access Network has not been stressed enough for QoS to produce these specific results.

As a summary, we can conclude for the Edge Cloud and SDN setup:

- The whole setup is working as expected with two QoS slices
- The QoS/slicing dynamic solution works as expected, changing from "Best Effort" to "Gold" quality under network conditions degradation.
- The defined QoS slice show significant differences under stressed network conditions.
- The Edge Cloud and SDN infrastructure has shown the expected flexibility to be used in three different setups: as a standalone testbed, connected to Nokia facility in Segovia and connected to Nokia facility in Madrid. In all cases, media has been delivered using the two QoS slices defined during the project.

4.5.4 Lessons learned

Due to the required bandwidth, millimeter wave frequencies (FR2) were used, which provide significantly more bandwidth (800 MHz) compared to lower bands (FR1). However, radio features in FR2 are less mature and only non-standalone mode (NSA) is supported, which implies no native slicing in 5G. Moreover, only a few TDD configurations are available today, focused on downlink capacity, which was a challenge since we required uplink.

Besides, the COVID-19 pandemic had a significant impact on the project's roadmaps, and we had to keep the original "Phase 1" configuration involving NSA and uplink-oriented TDD for the entire project. To overcome this issue, we used emulation and additional hardware to design the blueprint for the next-generation RAN.

5 Summary of the work carried out for UC1, UC2 and UC3

5.1 UC1: Live Audio Production

5.1.1 Achievements:

- Implementation, integration, and optimization of disaggregated 5G components
- Optimization of 5G testbed for latency (latency reduction by factor of 10)
- Capturing the state-of-the-art of current 5G ecosystem and benchmarking towards use case requirements
- Identification of remaining gaps in 5G ecosystem
- Network integration of live audio production into different 5G testbeds
- Proof-of-concept remote live audio production
- Fully remote-controlled trials and measurements
- Proof-of-concept for shared access to spectrum for a private 5G network
- Proof-of-concept to transport audio and video over the same private 5G network in TV production scenarios

5.1.2 What couldn't be achieved:

- One-way latency requirement of 1 ms was not met (latency is at ~10 ms for a single audio UE (microphone or IEM), and at ~20 ms for up to three audio UEs)
- Design space exploration for valid operation points including the evaluation of the trade-off between all relevant use case KPIs (latency, reliability, and spectral efficiency)
- Time synchronization is not yet supported by available 5G ecosystems

5.2 UC2: Multi-camera wireless studio

5.2.1 Achievements:

- UC2 partners have managed to test end-to-end multi-camera live production and remote contribution.
- PTP performances have been analysed on 5G networks Release 15 and URLLC test bed Release 16.
- The partners have managed to perform tests in both stationary and mobility scenarios.
- A portable camera interface unit inclusive of a 5G standalone modem and encoder has been designed and integrated to overcome Image Matters, the partner responsible for the codec, leaving the consortium.
- The media gateway and the media operational control gateway have been developed from scratch to enable a seamless integration of devices in hybrid network made up of 5G network and IP (e.g., ST 2110) media production facilities.
- An experimental remote camera control software was developed and a demo of this software was shown during IBC.

- UC2 managed also to test and use the MCR instance provided by a third party (GV AMPP), work that initially was for Redbeemedia technology who left the consortium.
- Network slicing in the contribution scenario worked well. 5G network performance in various conditions and configurations was extensively tested.
- Transmission over mmWave 5G link was also achieved.
- UC2 and UC1 managed to work together during the live trial where not only the video (UC2), but also the audio (UC1) was transmitted over 5G. During the live trial, a small remote production was also simulated inside RAI labs in Turin.
- A portable 5G standalone setup was also provided and used during the IBC dissemination activity.
- A glass-to-glass latency of 180ms was achieved with no packet lost over the 5G network.

5.2.2 What couldn't be achieved:

- The genlock functionality for the cameras from the PTP over the 5G network was not achieved: Image Matters was responsible for this feature implementation, but the company left the consortium.
- 4:2:2 production wasn't achieved because of the limitation of the encoder board used to replace Image Matters boards.
- Further investigation is needed to experiment more with QoS management on real-world NPNs.
- Further work on authentication and authorization should provide a way to control access to a production's resources based on identity established in the 5G network, and rules provided by the production.

5.3 Use Case 3: live immersive media production

5.3.1 Achievements:

- Development of a FVV system which can work on live and offline contents and is adapted for 5G and cloud production.
- Design and validation of a 5G+MEC compact deployment which can be used in immersive media workloads.
- Deployment and testing of end-to-end transport slicing over a software-defined network, including automatic slice change.
- Integration of all the elements into two different test beds in Segovia and Madrid, and interconnection of both test beds using a commercial network with QoS guarantee.
- End-to-end live trial of a music live event, showing all the elements of the use case working together.
- Pioneer tests on immersive content production over millimetre-wave 5G RAN.
- Analysis of the performance of the system and its limitations, as well an analysis of how the system will perform over the next generation of infrastructure elements.

5.3.2 What couldn't be achieved:

- Due to the lack of devices supporting 5G Standalone in FR2 (mmWave), full 5G slicing capability could not be tested.
- Likewise, limitations on the total uplink capacity of the deployment made it impossible to operate in the field trial with the maximum possible resolution supported by the FVV system, resulting in sub-optimal quality.

6 Conclusions

During the second phase of the project several final tests and trials were successfully deployed to assess and validate the 5G-RECORDS components and E2E solutions in the context of the three project use cases: live audio production (UC1), multiple camera wireless studio (UC2) and live immersive media production (UC3). Those trials allowed project partners to study to which degree 5G fulfils the technical KPIs and requirements of the project use cases in the context of professional content production.

6.1 Live Audio Production Use Case

UC1 5G disaggregated infrastructure was used to *(i)* study the performance of the E2E system when using a single UE, *(ii)* measure deterministic audio streams through multiple 5G modems and *(iii)* conduct mobility tests to better understand the use-case KPIs in a more realistic environment, i.e., when musicians are moving around the stage.

Even if the UC1 network has evolved gradually during the project to reduce latency, UC1 requirements for local audio production applications, e.g., 1 ms one-way network latency, are still not met. After multiple iterations, UC1 partners were able to reduce the one-way network latency in the disaggregated 5G testbed to about 10 ms for a single audio UE and about 20 ms for up to three audio UEs. However, the availability and maturity of available 5G components remained a major issue until the end of the project. For instance, the COTS 5G modem used in UC1 had major influence on the support for specific features and achievable KPIs.

During the trials, UC1 partners identified that some components introduced significant latency jitter into processing and forwarding of audio IP packets in the 5G system. This is the case of the 5G UPF deployment and the parametrization of the CU. Also, while the latency in the UL is determined by 5GS timing and jitter, the latency in DL is determined by asynchronous processing and the USB connection. This means that, not only is the finally achieved latency dependent on the 5G radio timing grid (e.g., slot-length), but it is also significantly defined by implementation of interfaces and processing functions, and types of deployments.

Another important outcome of 5G-RECORDS was the collaboration between UC1 partners and components within the UC2 trial in Tivoli Garden in order to explore the possibility of using 5G in a local TV production to also deliver wireless audio. The goal was to demonstrate the delivery of audio and video over the same 5G network, as well as to conduct latency measurement as part of the evaluation of the state-of-the-art 5G components. Test results showed that packets were faster than 75 ms.

All in all, it can be stated that significant effort is still needed to finally achieve the full set of requirements for live audio production scenarios. Low latency must be considered end-to-end. All components and interfaces on all layers in the full signal path need to be designed with low latency paradigms. This remains especially challenging in complex wireless connectivity systems with many individual components and standardized interfaces. Since the latency requirement was not met yet, it remains an open question in what way the trade-off between latency, reliability and (spectral) efficiency can result in valid operation points in this use case context. Also, state-of-the-art 5G components do not yet support sufficient time synchronization or provide corresponding interfaces on application level.

Nonetheless, 5G-RECORDS has shown that it was possible to integrate live audio production on network layer into multiple 5G testbeds and that the latency in a state-of-the-art 5G system can be reduced significantly with extensive optimizations.

6.2 Multiple Camera Wireless Studio Use Case

Several tests were performed by UC2 towards the technology validation and execution of the final trial at Tivoli Garden. Despite several setbacks due to development and integration delays and partners leaving the consortium, it was possible to perform the desired tests and trials in the second phase of the project. Following, conclusions about the last tests and trials performed are presented.

The PTP performance tests were initially envisioned to decide if PTP over 5G networks could provide enough time synchronization precision for 2 purposes: first, so PTP could be used by the IM encoder board to extract genlock signals from the PTP synchronization and to use the synchronization for frame timestamping (on RTP streams). Genlock extraction was not possible, as IM, the partner responsible to implement it, left the consortium. The PTP tests were successful in demonstrating that PTP over 5G is sufficient for frame-level synchronization. It was also demonstrated that the basic PTP performance can be greatly enhanced by client tweaking and using advanced TSN features (from about 117 μs to 3,6 μs median offset).

Regarding the local and remote production, several tests have been performed to test the complete production chain to prepare it for the trial in Tivoli. In these last tests performed in Aachen, UC2 team was able to integrate the media gateway into the infrastructure. The tests allowed to study the traffic behavior and extract different KPIs (frame delay, interarrival, packet latency, etc.) In summary, the most important KPIs such as E2E latency and uplink throughput have been achieved with a glass-glass latency around 200ms and 50 Mbps per video stream.

For the remote production scenario, tests were also conducted to assess the performance of uplink video transmission using different combination of parameters, including the use of network slices in Aachen. The use of network slicing proved to be beneficial, as the devices using the guaranteed performance slice did not have their throughput affected by the best-effort devices. Additionally, remote production tests were performed at UPV campus, focusing on bonding, comparing 2 different modems and network configurations. Tests were successful, as they allowed to discover performance differences between modems and network configurations.

The final trial in Tivoli aimed to perform a real-world scenario with the components and architecture designed and developed within the project. Both local production and remote production scenarios, after a joint effort within the consortium to solve problems encountered in the integration, were tested successfully since a live event could be covered via 5G private network-enabled content production with low latency (200 ms) and sufficient quality. The integration of the encoder and the 5G modem in a single box allowed for seamless integration with the TV cameras, as this unit could be docked to the back of them, providing low latency encoding and 5G connectivity. Remote production scenario was made connecting 2 facilities: Tivoli and Turin via 5G network and also using the MCR (GV AMPP) that proved to be suitable for content production with 2 second G2G latency.

6.3 Live Immersive Media Production Use Case

During the final UC3 field trial, the viability of a full end-to-end FVV Live deployment to stream and record an event over a 5G network was demonstrated. The trial was chiefly intended to bring the use case into a real environment and validate each of the modules and components. This final trial was successful and provided relevant information as a result of all the work carried out during the project.

The event consisted in a live music performance by professional artists which was produced as a FVV service in real time and streamed to the final user. The event took place in Nokia premises in Madrid (Spain), and the FVV content was also recorded to demonstrate the FVV playback functionality of the system. Furthermore, Grafana dashboards were shown and monitored during the whole session.

Specifically, two traffic slices were configured (Multimedia Gold and Best Effort) in two different network segments (5G RAN and Transport Network), covering the delivery of the produced video from the Media Delivery VNF in the Delivery Edge Cloud to the end user in the trial site. The Gold QoS improves the video delivery in heavy traffic conditions, while in low traffic conditions, the results are similar. Since only 5G NSA is available for mmWave frequencies, QoS slicing in the RAN is implemented by giving different QoS parameters to different users. Various tests have been run using the Web Player and playing content from a media player under different conditions. In all the scenarios, content has been played in different resolutions, introducing noise (traffic) in the radio network which has been generated using iperf3 to produce downlink traffic to 3 additional UEs. The status of the radio cells was monitored at Nokia AirScale system and reported in the Grafana dashboard. The incoming and outcoming traffic was monitored in all the network interfaces of the 5G core. Due to the characteristics of the trial, uplink traffic from the 5G-enabled capture server was the dominant contributor to the total traffic, both at input (coming from the 5G gNB) and at output (going to the MEC). It is worth noting that the traffic was very stable during the whole trial session.

The results collected provide useful insights on options to reduce, if necessary, the amount of data to deliver FVV providing the highest possible quality to the end users. Also, they can help define trajectories that can be appealing for the users. Regarding metrics, both capture servers 1 and 2 were connected by cable, so their losses are negligible. Capture server 3 used a 5G link, and their cameras suffered from some losses due to the radio access, although the average value remained within a reasonably range. Also, RTT has experienced no significant variations during the measurement for the different video qualities tested as well as jitter and initial load time.

Additionally, selected KPIs were measured and validated. In general, as expected, the higher the QP (Quantization Parameters) the lower the perceived quality with all HRTs (Hypothetical Rendering Trajectories). Also, HRT1 offers the highest quality. Results for all the SRC contents, evidence the perceptual quality reduction when the framerate is reduced from 30 to 15 fps, besides, the effect of the framerate is highly dependent on the content. It is worth noting that the target motion-to-photon latency under 170 ms was not fully achieved with the existing FVV + 5G network configuration. However, the obtained result is close enough (25% higher) so that the effective QoE is not affected significantly. Regarding the critical uplink bitrate, some tests have been conducted to test the overall capacity of the uplink. Regarding the delivery network, with the results obtained during the final trial for UC3, we can certify that the expected KPIs have been met for four different scenarios and the whole setup is working as expected with two QoS slices.

The technology validation of Use Case 3 in 5G-Records aimed to enable advanced content production services in live events using 5G. However, the project faced significant challenges due to the required bandwidth, which necessitated the use of millimeter wave frequencies (FR2). FR2 has less mature radio features, and only non-standalone mode (NSA) is supported, which implied no native slicing in 5G, and only a few TDD configurations are available today, which are focused on downlink capacity. Moreover, the COVID-19 pandemic impacted the roadmaps of 5G chipset manufacturers, requiring the original "Phase 1" configuration, involving NSA and uplink-oriented TDD, to be kept for the entire project. To circumvent this issue, the project used emulation and additional hardware to design the blueprint for the next-generation RAN. Despite these challenges, the project successfully prepared for future immersive production use cases, which require uplink-heavy use cases, providing a blueprint for next-generation RAN and enabling advanced content production services in live events.

A Annex A

Use Case 1 - Final Shared Access Spectrum validation

Some details of the actual procedures validated are shown below.

- Registration procedure

Sep 12 15:49:43.796 erik-XPS user.debug|6:1948165888|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:77|encode|Encoding result = <{"registrationRequest":[{"airInterface":{"radioTechnology":"NR"},"cbsdCategory":"A","cbsdInfo":{"firmwareVersion":"N.A.","hardwareVersion":"N.A.","model":"1.0","softwareVersion":"3.1.5","vendor":"Accelleran"},"cbsdSerialNumber":"0002","fccId":"ACC-TEST-FCCID","installationParam":{"antennaAzimuth":0,"antennaBeamwidth":360,"antennaDowntilt":0,"antennaGain":6,"height":0,"heightType":"AGL","indoorDeployment":false,"latitude":43.614565,"longitude":7.071114},"measCapability":[""],"userId":"ACC-TEST-USERID"}]}>

Sep 12 15:49:43.962 erik-XPS user.debug|1:1895540480|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"registrationResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","response":{"responseCode":0,"responseMessage":"SUCCESS"}}]}>

- Spectrum Inquiry

Sep 12 15:49:43.963 erik-XPS user.debug|7:1948165888|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:83|encode|Encoding result = <{"spectrumInquiryRequest":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","inquiredSpectrum":[{"highFrequency":3700000000,"lowFrequency":3550000000}]}]}>

Sep 12 15:49:44.130 erik-XPS user.debug|1:1895540480|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"spectrumInquiryResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","availableChannel":[{"frequencyRange":{"lowFrequency":3550000000,"highFrequency":3560000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3560000000,"highFrequency":3570000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3570000000,"highFrequency":3580000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3580000000,"highFrequency":3590000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3590000000,"highFrequency":3600000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3600000000,"highFrequency":3610000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3610000000,"highFrequency":3620000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3620000000,"highFrequency":3630000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3630000000,"highFrequency":3640000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3640000000,"highFrequency":3650000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3650000000,"highFrequency":3660000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3660000000,"highFrequency":3670000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3670000000,"highFrequency":3680000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3680000000,"highFrequency":3690000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"},{"frequencyRange":{"lowFrequency":3690000000,"highFrequency":3700000000},"channelType":"GAA","ruleApplied":"FCC_PART_96"}],"response":{"responseCode":0,"responseMessage":"SUCCESS"}}]}>

- Grant procedure

Sep 12 15:49:44.131 erik-XPS user.debug|7:1948165888|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:89|encode|Encoding result = <{"grantRequest":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","operationParam":{"maxEirp":20.0,"operationFrequencyRange":{"highFrequency":3650000000,"lowFrequency":3550000000}}}]}>

Sep 12 15:49:45.138 erik-XPS user.debug|2:1895540480|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"grantResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"1eefaa0c-8012-49d0-b0ef-8401e1e55bbd","grantExpireTime":"2022-09-24T03:36:38Z","heartbeatInterval":10,"channelType":"GAA","response":{"responseCode":0,"responseMessage":"SUCCESS"}}]}>

- Heartbeat procedure with grant suspension and relinquishment

Sep 12 15:49:45.139 erik-XPS user.debug|7:1948165888|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:95|encode|Encoding result = <{"heartbeatRequest":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"1eefaa0c-8012-49d0-b0ef-8401e1e55bbd","operationState":"GRANTED"}]}>

Sep 12 15:49:45.312 erik-XPS user.debug|2:1895540480|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"heartbeatResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"1eefaa0c-8012-49d0-b0ef-8401e1e55bbd","response":{"responseCode":501,"responseMessage":"SUSPENDED_GRANT"},"heartbeatInterval":10,"grantExpireTime":"2022-09-24T03:36:38Z","transmitExpireTime":"2017-01-01T00:00:00Z"}]}>

Sep 12 15:50:21.826 erik-XPS user.debug|7:1948165888|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:95|encode|Encoding result = <{"heartbeatRequest":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"1eefaa0c-8012-49d0-b0ef-8401e1e55bbd","operationState":"GRANTED"}]}>

Sep 12 15:50:21.979 erik-XPS user.debug|7:1895540480|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"heartbeatResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"1eefaa0c-8012-49d0-b0ef-8401e1e55bbd","operationParam":{"maxEirp":-26.0,"operationFrequencyRange":{"lowFrequency":3550000000,"highFrequency":3650000000}},"response":{"responseCode":500,"responseMessage":"TERMINATED_GRANT"},"grantExpireTime":"2022-09-24T03:36:38Z","transmitExpireTime":"2017-01-01T00:00:00Z"}]}>

Sep 12 15:50:21.980 erik-XPS user.debug|0:1948165888|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:101|encode|Encoding result = <{"relinquishmentRequest":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"1eefaa0c-8012-49d0-b0ef-8401e1e55bbd"}]}>

- Heartbeat procedure with authorisation

Sep 12 15:50:22.798 erik-XPS user.debug|7:1895540480|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"grantResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"cf465c55-12f6-4eb3-aa34-377fd72520a0","grantExpireTime":"2022-09-24T03:37:16Z","heartbeatInterval":10,"channelType":"GAA","response":{"responseCode":0,"responseMessage":"SUCCESS"}}]}>

Sep 12 15:50:23.094 erik-XPS user.debug|1:1895540480|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"heartbeatResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"cf465c55-12f6-4eb3-aa34-377fd72520a0","response":{"responseCode":0,"responseMessage":"SUCCESS"},"heartbeatInterval":10,"grantExpireTime":"2022-09-24T03:37:16Z","transmitExpireTime":"2022-09-12T13:54:37Z"}]}>

Sep 12 15:50:32.095 erik-XPS user.debug|2:1948165888|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:95|encode|Encoding result = <{"heartbeatRequest":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","grantId":"cf465c55-12f6-4eb3-aa34-377fd72520a0","operationState":"AUTHORIZED"}]}>

- Deregistration procedure

Sep 12 16:36:26.948 erik-XPS user.debug|4:1435449088|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:107|encode|Encoding result = <{"deregistrationRequest":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b"}]}>

Sep 12 16:36:27.119 erik-XPS user.debug|1:1416394496|LOG_APPL_CBSD| |-|cbsd/codecs/module/codec_ifprod.cpp:71|parse_cbsd_pdu|Request to decode <{"deregistrationResponse":[{"cbsdId":"ACC-TEST-FCCID/c5e8754637504e5ebf868efc915ae09cb8ba1c3b","response":{"responseCode":0,"responseMessage":"SUCCESS"}}]}>

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