Simulation of a Tele-Surgery process through a Live Video Streaming service, using Simu5G and Wowza

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***Abstract*— Telematic services that require low latency for real-time applications and that make use of wireless mobile networks are increasingly popular. In the case of Tele-Surgery services that employ Live Video Streaming (LVS), latency times of the order of 1ms are needed. Given the difficulty of implementing real 5G test scenarios that enable this type of service to be characterized, this paper presents an emulation scenario that uses Simu5G to simulate the network, Wowza as a real video server, OBS Studio for transmission and VLC media player for content playback. This emulation scenario makes it possible to modify such parameters as bitrate, bandwidth, frequency and numerology index in order to evaluate different network configurations. By varying these parameters in a controlled way, packet losses are obtained for different bitrate values. The best quality video was obtained with a bitrate of 3000 Kbps.**

***Index Terms*— Bitrate; 5G; OBS; LVS, Simu5G; Telesurgery; VLC; Wowza.**

***Resumen*— Los servicios telemáticos que requieren baja latencia para aplicaciones en tiempo real y que utilizan las redes móviles inalámbricas son cada vez más populares. El caso del servicio de Tele-Cirugía que emplea la técnica de Live Video Streaming –LVS requiere tiempos de latencia del orden de 1ms. Ante la dificultad de implementar escenarios de prueba reales de 5G que permitan caracterizar este tipo de servicios, se presenta en este trabajo un escenario de emulación que emplea Simu5G para simular la red, Wowza como servidor real de video, OBS Studio para la transmisión y VLC media player para la reproducción del contenido. Este escenario de emulación permite modificar parámetros como bitrate, ancho de banda, frecuencia e índice de numerología; con el objetivo de evaluar diferentes configuraciones de red. Variando de forma controlada los parámetros mencionados, se obtienen las pérdidas de paquetes para diferentes valores de bitrate. De acuerdo a los resultados, y para el escenario de prueba particular, el vídeo con una mejor calidad se obtuvo con un bitrate de 3000 Kbps.**

***Palabras Clave*—5G; LVS; OBS; Telecirugia; Tasa de bit; Simu5G; VLC; Wowza.**

I. INTRODUCTION

AROUND five billion people have no access to surgical care when they need it, due to the unavailability of qualified surgeons [1]. Technological progress has made the gradual implementation of remote surgery possible. The current methods of transmission (3G, 4G and LTE Advanced) have suffered however from significant limitations, with the quality and quantity of data transmitted far from the latency times required (1-2 ms). Such limitations can be life-threatening for patients, especially when there is limited time to make decisions [2].

Tele-Surgery connects patients and doctors (Specialists) by means of a wireless/wired network and a robotic system. The robot translates each movement of the surgeon into a movement of the surgical instruments as the surgical operation is displayed on a video screen [3].

Today a number of state-of-the-art Tele-Surgery systems (robots) are available featuring different degrees of freedom and for use in a range of types of surgery. Surgical robots generally consist of three robotic arms - two to manipulate surgical instruments and one to control the laparoscope [4].

But a report from the University of Illinois indicates that in the 15 years from 2004 to 2019 at least 144 deaths and 1,391 injuries occurred in robotic surgery in the United States [5]. The deaths and injuries were attributed to system errors and network latency, producing no small degree of uncertainty as regards the reliability of remote procedures [6].

Solving some of these problems is possible through the use of 5G networks, given that traditional wireless networks such as 3G, 4G and LTE Advanced provide latencies that are not appropriate for long distance Tele-Surgery. 5G networks using URLLC (Ultra-reliable low-latency communication) provide more stable transmissions of data up to 100 times faster than their predecessors (10 GB/s), reducing latency up to 1 ms and enabling safer surgical procedures and improving the degree of satisfaction of the surgical team [7].

Accordingly, this article will analyze the critical hypothetical case of Tele-Surgery in which simulators are used to provide a 5G network that allows connection with tools that

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work as a server and client. An acceptable video quality can thus be displayed on reception, according to the analysis of the packet loss obtained.

The rest of the paper is organized as follows. Section 2 presents the related concepts and previous studies. The design and implementation of the scenario is described in Section 3. Section 4 presents the parameters to be modified and the modification to the Simu5G configuration files. The analysis and evidence are presented in Section 5. Finally, Section 6 presents the conclusions and future work.

II. BASIC CONCEPTS AND RELATED WORKS

A. Related Work

An analysis was made in [8] of the communication requirements in a mission critical application, in which the system was separated into two sites - the virtual reality site and the robot site. The connection was created through an OpenVPN server running on a cloud server in Helsinki, Finland. The network connection setup used User Datagram Protocol (UDP) to achieve fast delivery of packets. During the broadcast, video quality was inadequate. Under optimal network conditions and without experiencing any attacks, the system ran smoothly with almost no delays. With 5G networks, there is a lack of research and testing for high-definition video feedback and other more demanding data streams that are necessary for such surgeries.

Elsewhere, in [9], a phantom pituitary tumor removal experiment was performed twice, once locally and once remotely via a robotic system to explore the feasibility of remotely controlling the surgical tools in long-distance procedures for endonasal skull base surgery. Tool movements and endoscopic video were transmitted over the Internet using free Skype software. Extremely low latencies were found to be possible over standard Internet connections. The main focus of the authors was exploring the feasibility of controlling surgical tools remotely over the internet using Skype, but no research has been carried out on its feasibility with such networks as 5G.

In [10], meanwhile, the authors investigated communication bandwidth (CB) limitation for robotic remote surgery (RRS) using Hinotori, a novel surgical robot made in Japan. The operating rooms of the Hokkaido University Hospital and the Kyushu University Hospital were connected through the Scientific Information Network (SINET). Ten surgeons were evaluated in a task (intracorporeal suture) at different levels of video compression. Packet losses were found to be between 3 and 7%. Therefore, a CB greater than 150 Mbps using Hinotori is feasible for an RRS.

Then, in [11], a 5G-powered Tele-Surgery study used a surgical robot controlled by a surgeon at a Qingdao tertiary hospital to remotely perform robot-assisted laparoscopic radical nephrectomy (RN) on 29 patients in eight primary hospitals. The total delay between the remote location and the operating rooms where the surgeries were performed was only 200 ms. The results demonstrated that the combination of 5G technology and surgical robots is a potential telemedicine-based option for kidney tumor therapy.

B. Basic concepts

The general concepts used in this research are presented below. The topics featured are video streaming technology, 5G networks, and network simulators.

1) Video streaming: Streaming is a technology of transmission through the network in which there is no download of the information on a local disk, but rather the information sent to the client is reproduced in real time on receiving it. [12]. For this to take place, streaming breaks the file data into small packets that are sent in a constant, continuous stream to the playback buffer. [13].

Two systems can be used to transmit video data: VoD (Video on Demand) and LVS (Live Video Streaming) [14]. VoD is a media distribution system that allows users to access videos without a traditional playback device, enabling users to access multiple content at the exact moment they want [15]. LVS meanwhile allows users to share audiovisual content in real time with viewers around the world. It faces a greater challenge than VoD, given that the service must be kept continuous in real time and with as few errors as possible [16]. Thus, regardless of when a client connects to the server, they all see exactly the same point of transmission at a given moment (except for the logical variations in network delays that cause some clients to receive data earlier than others) [12].

2) *5G networks*: 5G builds on the fundamentals implemented in 4G LTE networks, looking to meet the needs of future wireless applications such as autonomous vehicles, and ultra HD (UHD) 3D video transmission [17]. It has advantages such as: a data speed between 1 to 10 Gbps, which means almost 10 times the data speed in LTE, which theoretically is in the order of 100 Mb/s [18]; 1 ms latency for a bidirectional round of communication, just one tenth of the latency in 4G; high bandwidth to handle several different devices that are connected to each other in a given area; and the ability to handle the connectivity of a large number of devices in terms of IoT [19].

3) *Network simulators*: a large number of simulation tools are currently used to recreate the operation of the network in as real a way as possible, each with different characteristics and capabilities. Among the best known are OMNeT++, OPNET, NS3, NetSim, GNS3, EstiNet, Qualnet, and J-sim. For the purpose of this work, OMNeT++ was chosen. A simulation platform with a modular architecture, OMNeT++ is flexible when making designs and models of networks, protocols, multiprocessors or hardware architectures, which makes it viable for systems where large networks must be modulated. Compatible with Linux and Microsoft Windows operating systems, this tool is made up of modules written in C ++ and Network Description, its own language for defining the topology of the modules [20].

III. Design and implementation of scenario

The success of the video streaming service focuses on the user being able to display the content on their device with a minimum of failures and delays. To ensure this, a number of management tasks must be performed on the network, such as monitoring and control of bandwidth [21].

A. Design

To carry out the study of the LVS service in a Tele-Surgery process on 5G networks, an experimentation scenario was built using a simulation made up of the elements of video server and client, which exchange information through a simulated 5G network.

The scenario to be simulated will be a hypothetical case of a surgical room that will allow Tele-Surgery practices to be carried out using systems (robots). Different remote end users will be able to connect to the service. These will receive the live video transmission of the surgery. Possible users are the surgeon and the supporting doctor.

For this study, three different elements will be needed; the first in charge of emulating the 5G network through which the clients and the server will connect; the second element is the server that will send the video transmission to the clients; and the last element will be the client that will be used for the reproduction and monitoring of the information received.

Fig. 1 shows the infrastructure of the scenario to be simulated. In it the main components to provide video streaming service are identified: video encoder, streaming service, network core and user network. Once the video starts streaming, the content is encrypted and made available to a streaming server. The content is divided into several packets and transmitted through the network core. Finally users are able to access this content on their devices through the access node.

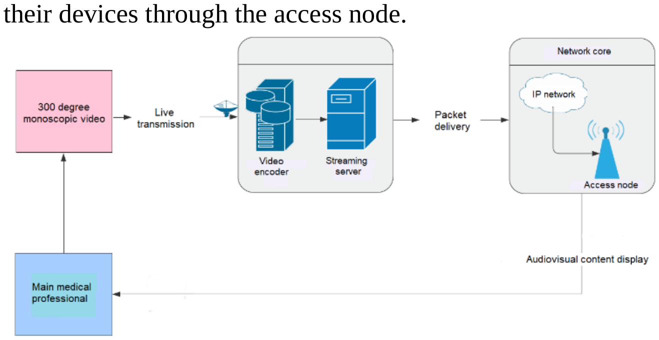


Fig. 1. Diagram of context

B. Implementation

For the simulation, Simu5G, a simulation library for 5G New Radio networks based on OMNeT++ was used. It includes a collection of models with well-defined interfaces, which can be instantiated and connected to build arbitrarily complex simulation scenarios, and is fully compatible with the INET library [22]. In addition, it allows the simulation of 5G communications in Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD) modes, with heterogeneous gNBs (macro, micro, pico, etc.), possibly communicating through the X2 interface to support inter-cell interference handover and coordination [23]. In particular, Simu5G models the data plane of the 5G RAN (rel. 16) and the core network [22].

Simu5G installation was achieved by downloading the Simu5G and Inet folders from the virtual machine (plug and play) found on the page "http://simu5g.org/simu5g-pnp.html", since in this way the simulation could be executed through the Ubuntu terminal and data packets sent through the 5G network. The Simu5G installation guide on Ubuntu 20.04 can be found in Annex A of the undergraduate work [24].

TABLE I
CHARACTERISTICS OF THE PC USED

PC	Characteristics	Operating system	Software
PC-1	Ryzen 5600X processor, 16G of RAM, Nvidia RTX 3060 graphic card	Linux Ubuntu 20.04 LTS	Simu5G, Wowza Streaming, OBS Studio, VLC

For the server setup process, it was decided to work with Wowza, a robust and customizable media server software that powers reliable streaming of high-quality video and audio to any device, anywhere. It supports any video format, transcodes it once, and reliably delivers it in multiple formats and to the best possible quality [25]. Wowza has many tools to generate video applications such as Video on Demand (VoD) and Live Video Streaming from any WEB or IP camera, and uses various streaming techniques such as DASH, RTSP, RTMP, Apple HLS, Adobe HDS and MS Smooth. [26].

OBS (Open Broadcaster Software) Studio, an open source application that allows both recording and transmission of audio and video in real time (streaming) was also selected for use [27]. It can be used to record presentations, screen capture sessions, eSports games, etc.. It comes with presets for streaming on YouTube, Twitch and Facebook, but can be used for any streaming platform that uses customized RTMP (Real Time Messaging Protocol) [28].

For the client, a free and open source multimedia player VLC was employed,. Configuration was carried out with Real Time Streaming Protocol, RTSP, taking into account that the transmissions of the data packets are made through the User Datagram Protocol, UDP. This protocol is used in transmission of live videos for its lack of retransmission delays, which makes it suitable for applications of these types and ideal for use with the RTSP transmission protocol [29].

The characteristics of the PC used for the simulation are those shown in Table I.

IV. Adaptation of the scenario

A. *Parameters*

1) Bitrate and Encoding: Bitrate refers to the rate at which data is processed or transferred. It is usually measured in seconds, from bps for smaller values to Kbps and Mbps. In digital networks and telecommunications, bitrate refers to the measurement per second of data passing through a communications network. In this context, bitrate is synonymous with data transfer rate. For multimedia encoding, bitrate refers to the number of bits used per unit of playback time, such as video or audio after compression (encoding). Media size and output quality often depend on the bitrate used during encoding [30]. As such, the term bitrate will be used focused on multimedia encoding, specifically in video encoding.

2) Frequency: Although the physical and upper layers are designed as frequency independent, two independent radio performance requirements are specified for two frequency ranges (FR), namely FR1 and FR2. FR1 is below the 7 GHz range (450 - 7125 MHz) and FR2 is the millimeter wave range (24250 - 52600 MHz) [31].

3) Numerology: NR uses a flexible framework structure, with different Subcarrier Spacings (SCS). SCS is the distance between the centers of two consecutive subcarriers, and possible SCS values are (in kHz): 15; 30; 60; 120 and 240. This is known as "multiple numerologies." The time domain, meanwhile, is divided into 10 ms radio frames, each of which consists of 10 subframes of 1 ms each, as shown in Fig. 2 [31].

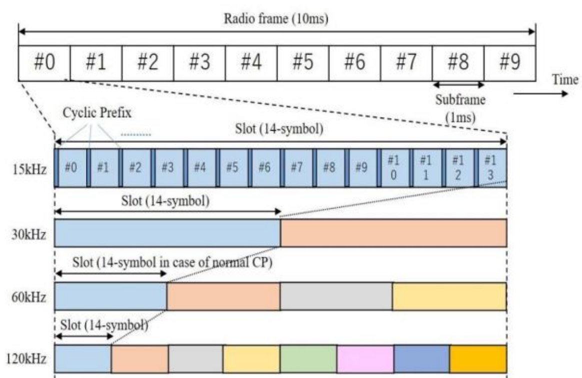


Fig. 2. Structure of frame in NR[32]

4) Bandwidth: In NR, the maximum bandwidth of an NR carrier is 100 MHz for FR1 and 400 MHz for FR2. For higher bandwidth, carrier aggregation (CA) of up to 16 NR carriers is also supported. Both CA within a frequency band (intra-band CA) and CA between frequency bands (inter-band CA) are supported. In the case of interband CA, CA with different numerologies is also supported, for example, CA between NR carrier in FR1 and NR carrier in FR2 [32].

The maximum transmission bandwidth NRB setting for each UE channel bandwidth and subcarrier spacing is specified in Table II.

TABLE II

MAXIMUM BANDWIDTH SETTING OF NRB TRANSMISSION FOR FR2

SCS (KHz)	50 MHz NRB	100 MHz NRB	200 MHz NRB	400 MHz NRB
60	66	132	264	N.A
120	32	66	132	264

Adaptive Coding and Modulation (ACM): A central feature in today's cellular networks, ACM technology can automatically change the modulation and error correction of link forwarding to compensate for changes in link conditions. These changes are usually due to weather, for example rain fade, but can also come from other sources, such as interference [33].

ACM optimizes the performance in a wireless data link, adapting the modulation order and the error correction code rate according to the noise conditions in the link [33]. In the 5G mobile communication system, the base station performs adaptive coding and modulation based on channel state information provided by the user and improves spectrum efficiency by selecting different combinations of modulation types and code rates [34].

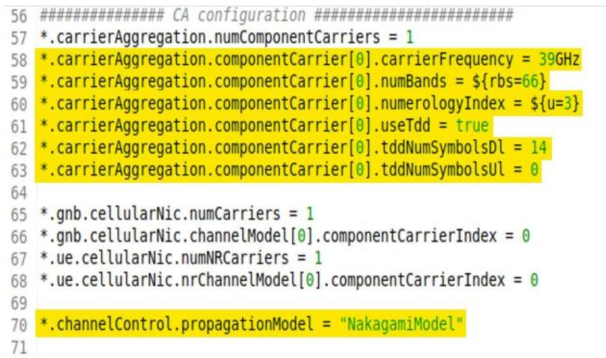
B. *Modification to the Simu5G configuration file*

In accordance with the parameters mentioned in Section 4.1, the modifications were made in the Simu5G configuration files found in "simu5G/emulation/extclientserver", from line 58 to line 70 regarding numerology, frequency, bandwidth and duplex type, to adjust the network conditions to the needs of the scenario in question and to be able to find the threshold points.

Fig. 3 shows a snippet of code as Simu5G was initially configured. Initially, the carrierAggregation module was configured with a CC by setting the numComponentCarriers parameter to 1, as seen in line 57. In line 58, the componentCarrier vector was set to index 0 (that is, it is going to select the first component from the vector initialized in line 57) in which numerologyIndex is set equal to 0, carrierFrequency equal to 2 GHz, and numBands equal to 10, as shown in the following lines 58, 59, and 60, which refer to the numerology index, carrier frequency and the number of resource blocks respectively. The number of CCs used by gNB and UE was configured by setting the numCarriers parameter to 1, as shown in lines 62 and 64. And the channel-Model module is associated with CCs with index 0, by setting the componentCarrierIndex parameter.

```
56 ############## CA configuration ######################
57 *.carrierAggregation.numComponentCarriers = 1
58 *.carrierAggregation.componentCarrier[0].numerologyIndex = 0
59 *.carrierAggregation.componentCarrier[0].carrierFrequency = 2GHz
60 *.carrierAggregation.componentCarrier[0].numBands = 10
61
62 *.gnb.cellularNic.numCarriers = 1
63 *.gnb.cellularNic.channelModel[0].componentCarrierIndex = 0
64 *.ue.cellularNic.numNRCarriers = 1
65 *.ue.cellularNic.nrChannelModel[0].componentCarrierIndex = 0
66
```

Fig. 3. Initial configuration file omnetpp.ini in the extclientserver folder



```
56 ############### CA configuration ########################
57 *.carrierAggregation.numComponentCarriers = 1
58 *.carrierAggregation.componentCarrier[0].carrierFrequency = 39GHz
59 *.carrierAggregation.componentCarrier[0].numBands = ${rbs=66}
60 *.carrierAggregation.componentCarrier[0].numerologyIndex = ${u=3}
61 *.carrierAggregation.componentCarrier[0].useTdd = true
62 *.carrierAggregation.componentCarrier[0].tddNumSymbolsDl = 14
63 *.carrierAggregation.componentCarrier[0].tddNumSymbolsUl = 0
64
65 *.gnb.cellularNic.numCarriers = 1
66 *.gnb.cellularNic.channelModel[0].componentCarrierIndex = 0
67 *.ue.cellularNic.numNRCarriers = 1
68 *.ue.cellularNic.nrChannelModel[0].componentCarrierIndex = 0
69
70 *.channelControl.propagationModel = "NakagamiModel"
71
```

Fig. 4. Modified configuration file omnetpp.ini from the extclientserver folder

Fig. 4 shows the changes that were made in the Simu5G code for the purposes of the research work. Variations of carrierFrequency, numBands, and numerologyIndex have been made, as evidenced in lines 58, 59, and 60. New parameters were added, such as TDD duplexing equal to true in line 61, which means that duplexing will be implemented by time division with tddNumSymbolsDl equal to 14 and tddNumSymbolsUl equal to 0, as shown in lines 62 and 63, allowing DL traffic only. In addition, a new channelControl module was added, in line 70, setting the propagationModel with the Nakagami model. This module by default was configured with the free space model. This change was made, since the Nakagami model is widely used in the literature, which demonstrates its relevance in estimating propagation conditions in indoor and outdoor scenarios in the presence of fade [26].

V. TESTING AND ANALYSIS

The main configuration parameters exhibited in Table III show that the direction of transmission is downlink, since only the client will consume the LVS service; the Duplex mode is TDD, since a single frequency band will be used in transmission; and the propagation model Nakagami, due to that shown in Section 4.2. The modulation, the transmission power and the positions of gNB and UE are already found by default in Simu5G. The chosen frequencies were in the mid-band range (below 7 GHz), which is the optimal point for 5G deployments, since it has higher bandwidth and capacity compared to low-band, and frequencies in the range of the millimeter wave or high band (above 24 GHz). It thus offers unprecedented maximum speeds and low latency. Numerology 2 and 3 were selected, taking into account that large SCSs are ideal for reducing latency [35] and bandwidths of 50 MHz and 100 MHz, since they allow working for FR1 and FR2 frequencies.

TABLE III
CONFIGURATION PARAMETERS

Parameter	Value
Direction of transmission	Downlink
Duplex mode	TDD
Scenario	Indoor
Frequencies	TDD-related
Bandwidth	50 MHz and 100 MHz
Numerology índices	2 for FR1 and 3 for FR2
Propagation model	Nakagami
Modulation	ACM
Position gNB (x,y)	450 m, 300 m
Position UE (x,y)	450 m, 350 m

The transmissions were made with a 3-minute video, where variations were initially made regarding frequencies, numerologies (2 and 3) and bandwidth with a bitrate of 3000 kbps HD quality (see Fig. 5 and 6). Then, the same procedure is repeated for a bitrate of 5000 kbps HD 1080 quality (see Fig. 7 and 8) and 9000 kbps 4K quality (see Fig. 9 and 10).

The results of the data obtained for each bitrate are graphically presented below.

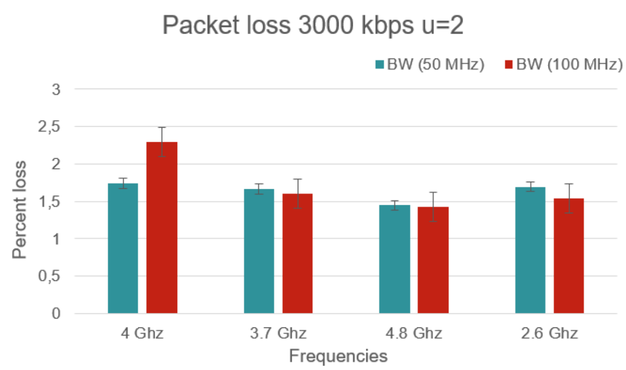


Fig. 5. Result of packet loss with bitrate 3000 kbps and u = 2
Fig. 6. Result of packet loss with bitrate 3000 kbps and u = 3

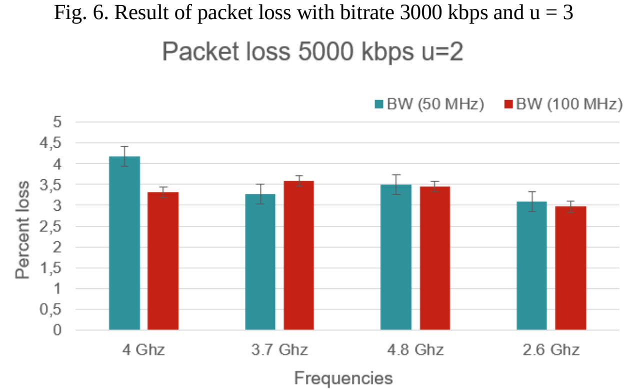


Fig. 7. Result of packet loss with bitrate 5000 kbps and u = 2

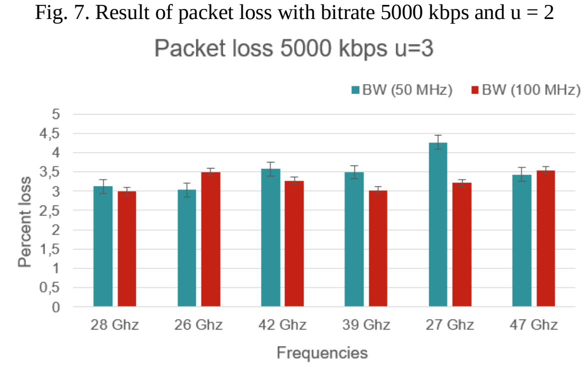


Fig. 8. Result of packet loss with bitrate 5000 kbps and u = 3

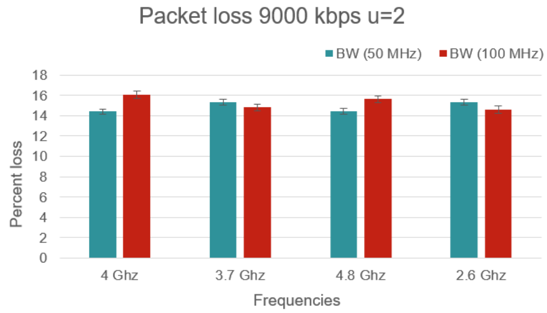


Fig. 9. Result of packet loss with bitrate 9000 kbps and u = 2

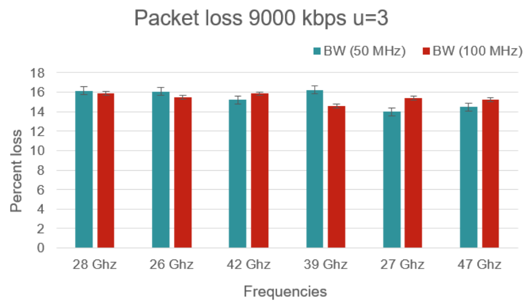


Fig. 10. Result of packet loss with bitrate 9000 kbps and u = 3

From the results obtained from the graphs, it could be seen that the higher the bitrate, the higher the packet losses for both a 50 MHz and 100 MHz bandwidth. For Figures 5 and 6, with a bitrate of 3000 Kbps, but with different numerology, approximate losses between 1.4% and 1.7% were obtained. Figures 7 and 8, which show a bitrate of 5000 Kbps, showed losses between 3% and 4%. Finally, Fig. 9 and 10, which show a bitrate of 9000 Kbps, showed losses between 14% and 16%.

However, no significant differences were found on modifying frequencies, bandwidths and numerology. So it is assumed that the network parameters are sufficient to carry this type of traffic to the destination.

Viewing of the videos was done with the RTSP protocol, but carrying out some tests such as transmitting the video without going through the network, using only Wowza, OBS Studio and VLC, and varying the bitrate, to high bitrates , approximately at 50,000 kbps, it was observed that there were no differences in video quality. In other words, even without going through the network, the video recovered in VLC has packet losses and pixelations similar to those observed in a transmission with low bitrates but going through the simulated network. Meanwhile, the same tests were done, but changing the protocol to RTMP and it was observed that the video is recovered correctly, without packet loss or pixelation, but when trying to recover the video with the RTMP protocol and passing through the network, the video is never displayed.

Since the transport protocol used in the simulation is UDP, RTMP is not capable of retrieving the video, since it is strictly a TCP-based protocol, while RTSP, using both protocols, depends on reliable transmission of TCP in the control and delivering best efforts of UDP to display audio and video to client-side applications, before the full file arrives, to provide a proper experience. Therefore, for this scenario using Simu5G and Wowza tools, the best option to retrieve the video is to use RTSP.

Taking into account what was observed, it is not possible to visualize the content of a video transmission with a high bitrate (greater than 9000 Kbps), see Figures 9 and 10 in which the losses are approximately 15%, while for a video with a low bitrate (approximately 3000 Kbps) it is indeed possible to reproduce its content, since losses are less, approximately 1.5%, see Figures 5 and 6. Furthermore, varying the bandwidth, frequency or numerology index does not lead to changes in the reception of the video. The most suitable bitrate to carry out transmissions in this scenario with the tools used however is with a bitrate of 3000 Kbps, since the losses found were low and in terms of video quality did not result in distortions in the received video.

VI. Conclusions

A simulated Tele-Surgery scenario was presented in which a series of videos were transmitted through a simulated 5G network using the OMNeT++ Simu5G library, Wowza as a server, OBS for transmission and VLC for reception. The results showed that varying the bandwidth, frequency and numerology index using the tools in question did not provide significant differences in the results obtained in the reception, as significant changes were found to occur on varying the bitrate. Moreover, it was highlighted that the video with the best quality displayed on reception was with a bitrate of 3000 Kbps using the exhibited tools.

VII. Future Work

In this research work, a simulation was carried out using Ubuntu 20.04, Wowza and Simu5G. Transmission of the LVS service was done using the RTSP protocol. However, during the tests carried out it was detected that at very high bitrates the RTMP protocol gives a better result than RTSP, but it was not possible to pass this type of traffic through Simu5G due to compatibility issues. Therefore, it is necessary to investigate other tools that allow this type of traffic. Also, with the Simu5G simulator, there were some limitations in performing network emulations. It is therefore necessary to investigate other simulators that allow emulating 5G networks between various devices connected to each other and that allow video transmissions with high bitrates.

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