Performance Analysis of Real-Time Services On 5G Millimeter Wave Network

Doan Perdana, *Member, IAENG*, Nadira Mumtaz, and Yoseph Gustommy Bisono

***Abstract*—In this study, Millimeter Wave simulation has been carried out on 5G technology using Network Simulator 3.27. We use real-time performance measurements for Voice over Internet Protocol (VoIP) and video services. For VoIP, we use standard G.711, G.729 and G.732.1 and for video codecs, we use standard H.264 and H.265. From extensive simulations, we get the result that the G.711 codec for VoIP and H.265 codec for videos obtain the lowest latency and jitter. We also found that the best Mean Opinion Score (MOS) score on the G.711 and H.265 codecs for VoIP and video, respectively. Meanwhile, for Codec G.732.1 produces the greatest bandwidth efficiency for VoIP services. Savings of bandwidth consumption on the G.711 codec up to 5 times and G.729 codecs is 43% bandwidth consumption. For video service, codec H.265 obtained 51% higher bandwidth efficiency compared to the H.264 codec.**

***Index Terms*—5G, Real-time Services, Network Simulator 3, Quality of Service.**

I. Introduction

MILLIMETER wave works on high frequency band which is 3-300 GHz for mobile broadband communications [1]. The wavelength of millimeter wave is shorter than the radio wave- length which is about 1-10 millimeter and it effects this wave could only reach a few kilometers. A very short millimeter wave wavelength allows real shrimpy antennas to centre signals with enough gain to solve propagation losses. Short wavelengths also make it possible to build multi-elements, dynamic beamforming antennas that are quite small.

There are some researchers evaluated millimeter wave performance to support 5G despite the characteristics of millimeter wave is susceptible to interference in the propagation process, because it works on high band frequency. [2] has been studied about the motivation, methodology of 5G using millimeter-wave. Based on that research, it shows that the frequencies of 28 and 38 GHz can be used to implement the millimeter wave network when employing steerable directional antennas at base stations and mobile devices. Marco Mezavilla et al [3] evaluated all the wireless layer such as PHY, RLC and PDCP layer of millimeter wave to support 5G [4]. They also evaluated the transport layer for 5G millimeter wave [5].

However, in this case, the user is allowed to specify the length of the subframe in several OFDM symbols. For 1 GHz bandwidth it is divided into 72 sub-bands with a width of 13.89 MHz, each of which consists of 48 sub-bands. This makes it possible to allocate users to each sub-band

Manuscript received May 13, 2019; revised December 18, 2020.

Doan Perdana is a Director of Advanced Creative Networks Research Center in Telkom University, Indonesia (e-mail: doanperdana@telkomuniversity.ac.id).

Nadira Mumtaz is a core engineer in Huawei (e-mail: dira812@gmail.com).

Yoseph Gustommy Bisono is a Lecturer in Telkom University (e-mail: tommy.bisono@gmail.com)

This research contributed to evaluate the performance of VoIP and video services with G.711, G.729, G.723.1 as voice codec and H.264, H.265 as video codec in 5G millimeter wave which is has never been done before. The main objective for this research was would the millimeter wave band could achieve the target performance for 5G in the future. Another objective for this research were to what is the codec that can perform the best quality and also can perform the most efficient to save network bandwidth capacity.

Real time services on cellular network is very critical, because real time services have very tight latency requirements. Some of the real time services are VoIP and Video Streaming. Voice Over Internet Protocol (VoIP) is is a way of voice communication through the Internet Protocol (IP) network. Video Streaming is technology that is deliver the video data through Internet broadband. Real time services use codecs to deliver the packet data. Codec is used for converted the signal such as analog to a bit-stream digital and also converted vice versa. Codecs services can provide the ability to save network bandwidth in real time [6]. Several codecs which have been made by ITU-T for voice are G.711, G.729 and G.723.1 also for video are H.264 and H.265.

In video services, H.264 codec is a format that is often used for recording, compression and video content because it can support up to 4K resolution (UHD 4096 x 2304). Bitrate for this codec is 64 - 384 Kbps [7]. H.265 codec is a codec that is able to compress higher quality videos and with a lower bitrate compared to H.264. H.265 has 50% lower bandwidth compared to the H.264 with the same quality [8]. Real time services require end-to-end Quality of Service (QoS) value to ensure an optimal service quality. In this research an analysis of the performance of each VoIP codec and video codec on 5G millimeter wave network, performance analysis is also carried out based on the influence of node density and node velocity, network performance is measured using Mean Opinion Score (MOS) value from QoS results includes latency, throughput and jitter that is generated by Network Simulator 3.

This research is divided into several parts. The second part, about the scenario that has been done with 5G parameters. The third part, we have analyzed the extensive simulation results by considering the throughput, jitter, latency and MOS parameters. Finally, we make a comprehensive conclusion about the scenario that has been done.

II. Scenario and Design System

Fig. 1 describes the topology of simulation, that the server is used and we employee the remote host. Packet Gateway (PGW) is connected to route the data for the external area. Serving Gateway (S-GW) is used to transmit data for the E-NodeB that connected by PGW. Then E-NodeB would

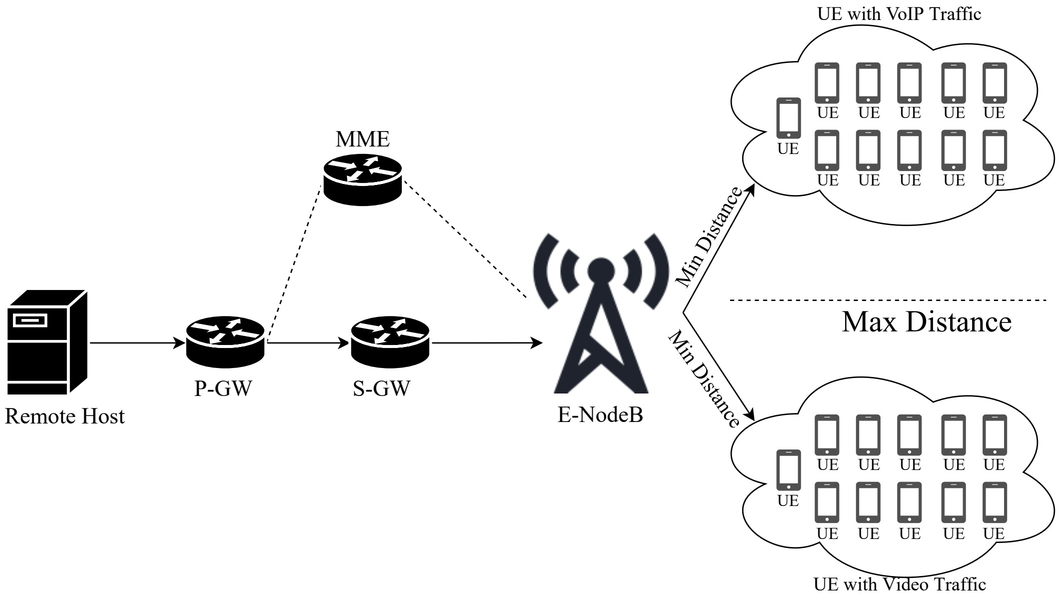

Fig. 1. Simulation topology.

TABLE I
SIMULATION PARAMETERS.

Parameter	Value
Frequency Carrier	28 GHz
Bandwidth	1 GHz
Number of UE	30, 50, 100, 150
User Speed	5 km/hours, 30 km/hours, 60 km/hours
Bit-rate Traffic	6.3 Kbps, 8 Kbps, 64 Kbps, 192 Kbps, 384 Kbps
Payload Traffic	20 bytes, 24 bytes, 160 bytes, 200 bytes
Mobility	Random Walk 2D Mobility

transmit radio transmission downlink and uplink to the user equipment and also connected with Mobility Management Entity (MME) which would control signaling session and security.This research using distance from 10 meters until 200 meters.

Simulation of real-time services performance simulated in Network Simulator software by modified the number of bit rate and payload on UDP traffic according to the characteristics of each codecs that is contained in the application structure on millimeter wave module. The traffic generated by this simulation is UDP traffic according to the characteristics of Real-time services. The simulation has been done for downlink traffic which is the remote host generated the traffic and then deliver it through the architecture until it reaches a user equipment.

Table I express the simulation parameter based on millimeter wave module specification. There is the converter to number of UE value to performs the network quality based on the capacity [9]. The change of user speed value is to performs the network quality based on the mobility. The value of traffic data rate and data payload are implemented for the characteristics of the codecs. The change of users speed represented as 5 km/hours is the average speed of the walking people, 30 km/hours is represented as the average speed of the motorcycle and 60 km/hours is represented as the average speed of the car.

The calculation of MOS based on latency results from the simulation have been computed for examine that codec obtain lowest latency and highest quality. The scenarios of this research are the influence of node density and the influence of node velocity. Users moved based on maximum and minimum distance that has been specified.

The flowchart for the research methods is presented in Fig. 2. First we have to design the script for 5G millimeter wave in Network Simulator 3 and the script also adjusted by the simulation parameters in Table I. Then, we carried out some scenarios which is node density and node velocity scenarios. After running the scenario, the output data will be collected. The output data include jitter, throughput and latency. The output data would be analyzed from the output file which generated by the simulator after we had run the simulation script. Based on the latency that we have got, we calculated the Mean Opinion Score. After that, we conducted the analysis and the conclusions.

The performance metrics are throughput, jitter and latency have been evaluated in this research. Throughput is defined as the total number of bytes successfully received in a certain of interval time [10], measured in Mbps. Latency is time of delivering the data package from the sender to the receiver and vice versa [11], measured in milliseconds. This research performance the end to end latency. Jitter is defined as the different the arrival delay of respectively data with measured in millisecond [12].

$$Total\ Packet\ Size = a + b + c, \tag{1}$$

where a is layer 2 header, b is IP/UDP/RTP Header, and c is payload size.

Equation (1) is employed to compute amount of packets size for transmitted. It involved a number of header which is 20 bytes for IP header, 8 bytes for UDP header and 12 bytes for RTP header [12]. Also, it includes the length of payload, this payload is adjusted by the characteristics for each codec.

$$PPS = \frac{Codec\ Bit\ Rate}{Payload\ Size} \tag{2}$$

Equation (2) is employed to compute amount of packets per second that will be transmitted. It calculated by divided

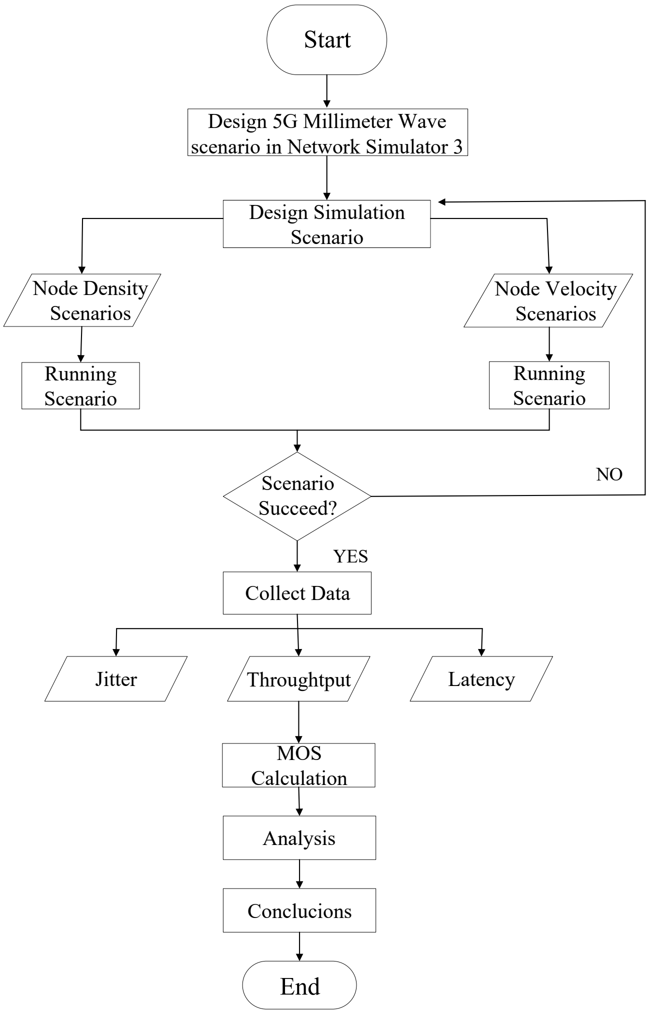

Fig. 2. Research flowchart.

TABLE II
PERFORMANCE REQUIREMENT BASED ON ITU-T.

Service	Typical Data Rate	Latency	Jitter
Audio	4-64 Kbps	<150 ms	<1 ms
Video	16-384 Kbps	<150 ms	-

the codec bit rate and payload size [13]. Those value is adjusted by the characteristics for each codec.

$$Channel\ Bandwidth = Total\ Packet\ Size \times PPS \quad (3)$$

Equation (3) is employed to compute a mount channel resource [13]. The equation is employed in the simple network. Total channel bandwidth is calculated to know how much the bandwidth that will be used for transmitting the data.

Table II shows the target value for acceptable latency and jitter based on ITU-T G.1010 in End-user multimedia QoS categories. Target latency for each service is < 150 ms and target jitter for voice service is < 1 ms but there is no target jitter value for video service [14]. 5G network in the future has the target for end-to-end delay (latency) is about 1 ms [15].

MOS is a method used to measure voice quality on IP networks [16], it has correlated with R Factor in E-Model ITU-T (5). E-model is a technique to evaluated the quality of services, because of the damage due to low bit rate coding, latency, echo and loss [17]. We used R-Factor to calculate MOS and is expressed in

$$R = 94.2 - (I_t - I_d) \quad (4)$$

$$I_l = 0.024 \cdot t + 0.11 \cdot (t - 177.3)\ H(t - 177.3) \quad (5)$$

$$I_t = 7 + 30\ \ln(1 + 15 \cdot l) \quad (6)$$

$$\text{MOS} = 1 + (R \cdot 0.035) + \left(\left(7 \times 10^{-6}\right) R\,(R - 60\,(100 - R))\right) \quad (7)$$

where t is latency (ms), l is packet loss, H for condition, that $H(x) = 0$ when $x \geq 0$ and $H(x) = 1$, when $x \leq 0$

Table III express MOS value and relationship between satisfactions of user level based on QoS value of VoIP services using WiFi-UMTS Technology. This study is also considered to obtain the value either fulfilled or fulfilled level [19].

TABLE III
THE SUBJECTIVE MEAN OPINION SCORE [18].

Quality Scale	Score	Listening Effort Scale
Excellent	5	effort required is none
Good	4	Appreciable effort required is none
Fair	3	Moderate effort should be required
Poor	2	Considerable effort required
Bad	1	No meaning understood with effort

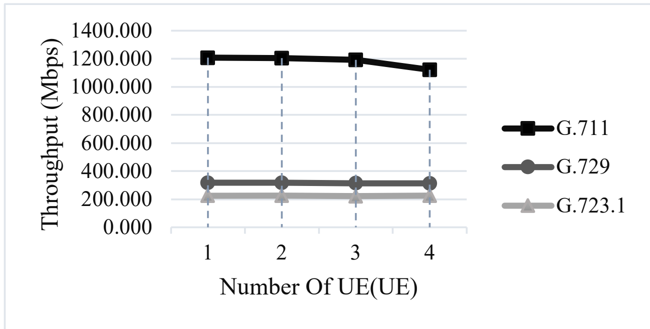

Fig. 3. Throughput Results for VoIP Based on Node Density.

III. SIMULATION ANALYSIS

This part performs the yield which have obtained after simulating video and VoIP services for 5G millimeter wave using Network Simulation version 3 as a simulator. The research results was divided into 4 section to analysis latency, throughput, MOS and jitter result for VoIP and video services.

A. Throughput Performance Evaluation

The results of throughput used as a reference for the actual bandwidth capacity that used to transmit data

Table IV shows the total channel bandwidth for each codecs in conventional network. It shows G.711 codec has the widest bandwidth for VoIP service and H.264 codec for video service. This channel bandwidth is used as a reference for a simple network architecture. Network architecture in this research used the 5G architecture. Based on the how it works to transmitting the data, it would be transmitted per Time Transmission Interval (TTI) which is on 5G millimeter wave it has 1 ms of TTI.

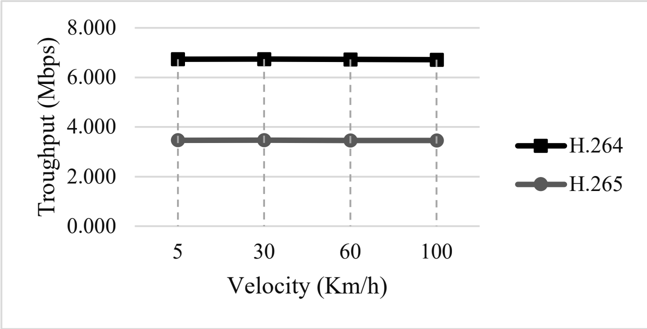

Fig. 4. Throughput Results for Video Based on Node Density.

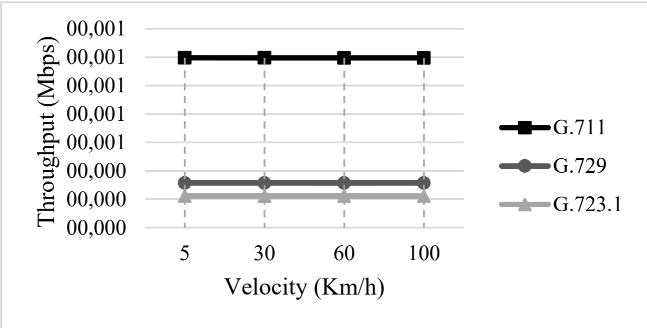

Fig. 5. Throughput Results for VoIP -Based on The Speed of User Mobility in G.711, G.729 and G.732.1.

Fig. 3 and Fig. 4 show the greater number of UE, throughput reduced from capacity of bandwidth due to shared to another subscriber. Based on Fig. 5, codec G.711 obtains the greatest throughput for VoIP service due to that has the highest bit-rate among the other VoIP codecs. Based on Table IV, it also has the highest amount of packet/s about 50 packet/s. It can be concluded that the bit-rate and the amount of packet/s affect the throughput value. Based on Figure 6,

H.264 codec obtain the greatest throughput for video service due to that has higher throughput than the H.265 codec. Codec G.723.1 was the highest efficiency codec to consume bandwidth for VoIP traffic. The result show that, the codec save 5 times of bandwidth consumption compared to the G.711 codec, and 43% compared to the G.729 codec. Codec H.265 is the highest efficient codec for Video service. The codec saves up to 51% of bandwidth consumption compared to the H.264 codec.

Fig. 5 and Fig. 6 show user speed was not significantly affecting the throughput. G.711 codec is still getting the highest throughput in VoIP service and H.264 still gets the highest throughput in video service. All the codecs in this result gets higher for throughput value compared with Table IV. Due to cellular network, for real time service such as video and VoIP streaming have own path which is Evolved Packet Core for send the traffic data, so it can produce higher throughput. It also because millimeter wave which works in high band frequency has high data rate (Gbps) to send the traffic over radio transmission.

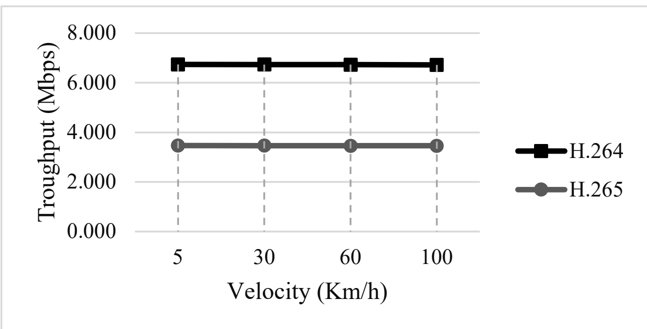

Fig. 6. Throughput Results for VoIP -Based on The Speed of User Mobility.

TABLE IV
CHANNEL BANDWIDTH CALCULATION.

	G.711	G.729	G.723.1	H.264	H.265
Datarate (Kbps)	64	8	6.3	384	192
Payload	160	20	24	200	200
Header (IP+UDP+ RTP)	40	40	40	40	40
Packet Length	200	60	64	240	240
Packet/s	50	50	33	240	120
Channel Bandwidth (Kbps)	80	24	16.8	460.8	230.4

TABLE V
MAXIMUM THROUGHPUT CALCULATION PER RESOURCE BLOCK.

Parameter	Value
Nsym	24
Nsubcarrier	48
Subframe length	100 μs
Nre	1152
Bit per Symbol	2
Bit Efficiency	0.1523
Total Modulation Bit	0.3046
Total bit per RE	350.8992
Throughput per RB	3508992

All the codecs in this result gets higher for throughput value compared with Table IV. Due to in cellular, for real time service such as video and VoIP streaming have own path which is Evolved Packet Core for send the traffic data, so it can produce higher throughput. It also because millimeter wave which works in high band frequency has high data rate (Gbps) to send the traffic over radio transmission.

Maximum throughput can be calculated by calculate how many resource blocks carried out the traffic. Based on (10), (12) and (12) equations, resource block calculated by assumes the modulation is QPSK for worst case which is carry 2bit/symbol and has 0.1523 for bit efficiency [20].

$$Total\ Bit = 8 \times (Codec\ Sample + Total\ Header) \quad (8)$$

$$Total\ Bit/s = \frac{Total\ Bit\,(bit)}{Codec\,Sample\ Interval\ (s)} \quad (9)$$

$$Nre = Nsym \times\ Nsubcarrier \quad (10)$$

$$Modulation\ Bit = \frac{bit}{symbol} \times bit\ efficiency \quad (11)$$

$$\frac{Troughtput}{HE} = \frac{Nre \times Modulation\ Bit}{Subframe\ Length\,(s)} \quad (12)$$

Based on (8) and (9) functions, the calculation of codec throughput can be calculated as shown in Table V.

Table VI shows the maximum throughput contains in each resource block. Then, the total number of resource block has

TABLE VI
CODEC THROUGHPUT CALCULATION FOR VOIP.

	G.711	G.729	G.723.1
Codec Sample Interval (ms)	10	10	30
Codec Sample (byte)	80	10	24
RTP Header (byte)	12	12	12
IP Header (byte)	20	20	20
UDP Header (byte)	8	8	8
Total Header (byte)	40	40	40
Total Bit	960	400	512
Total Data rate	48000	20000	17066.67

TABLE VII
MAXIMUM THROUGHPUT CALCULATION.

	G.711	G.729	G.723.1
Total Bit per RE	3508992	3508992	3508992
Total Bit Codec	48000	20000	17066.67
Total RE	1	1	1
Throughput Maximum (Mbps)	3.508992	3.508992	3.508992

to be calculated by compare it with the codecs throughput on Table V as shown in (13) equation.

$$Total\ RE = \frac{Datarate\ codec}{\frac{Throughtput}{RE}} \quad (13)$$

Table VII shows the maximum throughput that would be carrying the traffic. It shows each codecs have same throughput with the total resource block which would be carry is 1 resource block.

Table VIII shows the throughput for video codec. It shows H.264 has higher throughput than H.265 codec. Then, the total number of resource block has to be calculated by compare it with the maximum throughput per resource block on Table V.

B. Jitter Analysis

Jitter is employed to examine the interval between time of duration to transmit the packet data. Jitter target value from ITU-T standardization is <1 ms.

Fig. 7 and Fig. 7 show amount of users affect the jitter results. The greater amount of users, than latency will increased because the holding time for UE to be served is more delay and the latency is getting larger too as latency is relates to the jitter. It is the total variance of latency. Fig. 8 shows jitter of G.723.1 codec gets the worst or highest jitter for VoIP service, Also H.264 codec in Figure 10 gets the worst/highest jitter for video service. This is proportional to the value of latency that has been obtained. Based on Table II, this research has achieved the jitter target by ITU-T which is $<$ 1 ms for VoIP service.

TABLE VIII
CODEC THROUGHPUT CALCULATION FOR VIDEO.

	H.264	H.265
Codec Interval (ms)	0,625	1,25
Codec Payload (byte)	240	240
RTP Header (byte)	12	12
IP Header (byte)	20	20
UDP Header (byte)	8	8
Total Header (byte)	40	40
Total Bit	2240	2240
Total Datarate	3584000	1792000

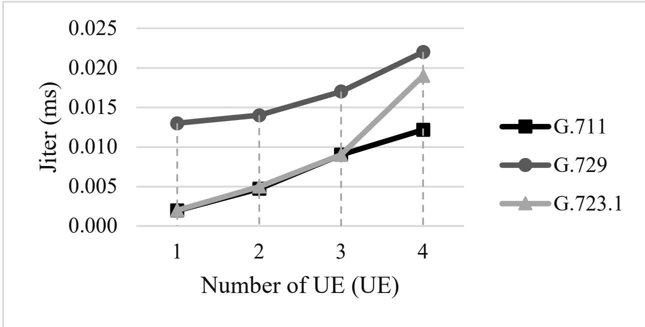

Fig. 7. Jitter Results for VoIP Based on Node Density in G.711, G.729 and G.732.1.

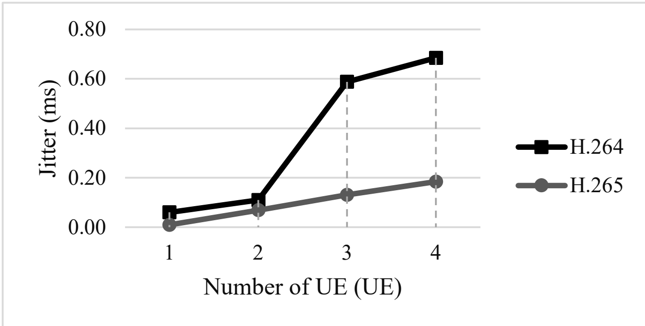

Fig. 8. Jitter Results for Video Based on Node Density.

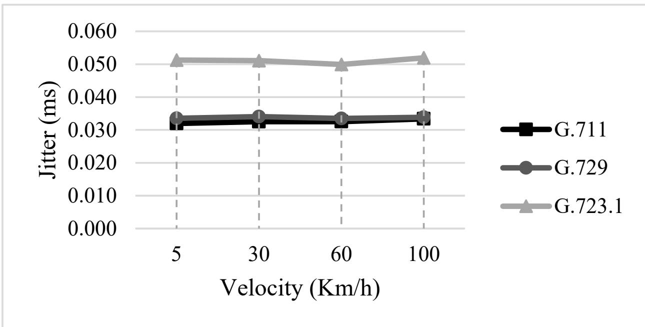

Fig. 9. Jitter Results for VoIP Based on The Speed of User Mobility in G.711, G.729 and G.732.1.

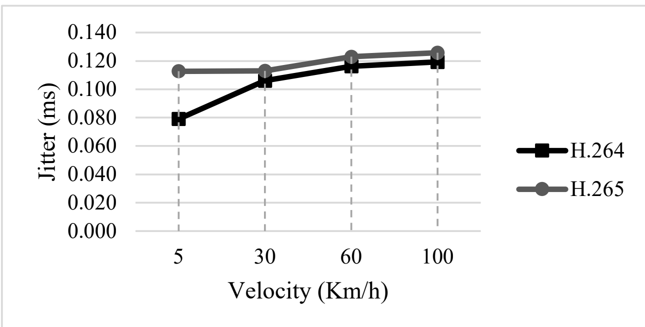

Fig. 10. Jitter Results for Video Based on The Speed of User Mobility.

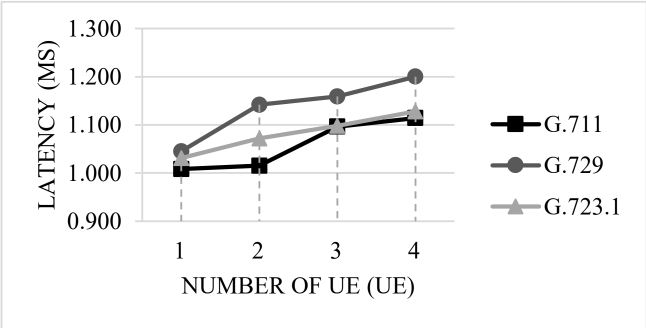

Fig. 11. Latency Results for VoIP Based on Node Density in G.711, G.729 and G.732.1.

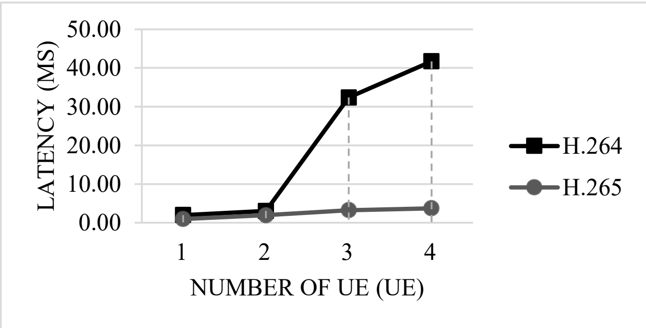

Fig. 12. Latency Results for Video Based on Node Density.

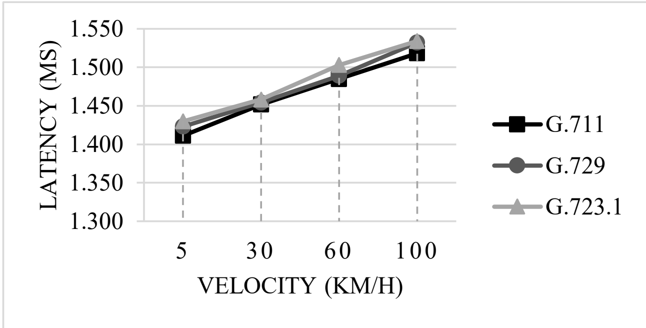

Fig. 13. Latency Results for VoIP Based on The Speed of User Mobility in G.711, G.729 and G.732.1.

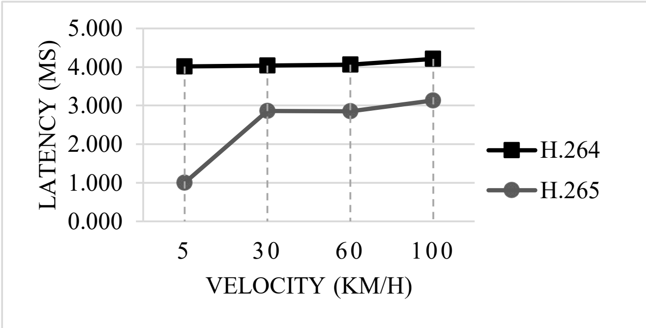

Fig. 14. Latency Results for Video Based on The Speed of User Mobility.

Fig. 9 shows user speed was not significantly affecting the jitter in VoIP service. While, Fig. 10 shows that user speed affect the jitter in video service. The higher user speed, jitter gets worst. G.723.1 codec still gets the highest jitter in VoIP service [21] and H.264 codec in video service still gets the highest jitter too [22].

C. Latency Analysis

To examine the holding time for transmit the data, we consider to use latency measurement. For 5G network is about 1 ms [18] and latency for ITU-T standardization is $<$ 150 ms for target measurement. The average of latency both in node density and node velocity scenarios was calculated for MOS [23].

Fig. 11 shows latency of G.723.1 codec obtains the largerst latency for VoIP service, due to has lowest datarate among the other.

Fig. 12 shows H.264 codec gets the highest latency for video service. It can be concluded the lower bitrate and higher packet/s, latency value will get worst. The greater number of UE, latency increased because the holding time to UE to be served is getting larger because an increase of traffic.

Fig. 13 shows user speed affects the latency in VoIP service. It shows G.711 codec obtained the lowest latency and G.723.1 obtained the highest latency in this results. It can be concluded the latency would gets higher due to the time needed to transfer data would be longer if users move quickly.

Fig. 14 shows user speed affects the latency in Video service. It shows H.264 has the highest latency and H.265 has the lowest latency in these results.

Based on Table II, this study has yield the latency target from ITU-T. In node density scenario, the latency score for VoIP has fulfilled the latency target for 5G technology, that is around 1 ms, but for video service, this latency was not achieved yet. In video service, only at 30 UE for H.264 and 30-50 UE for H.265 have achieved the latency target for 5G network because the core network for this simulation supported by LTE, which is uses Evolved Packet Core [20].

D. Mean Opinion Score Evaluation

For examine which codec result the highest quality of video and VoIP traffic, we consider to compute MOS [24]. Based on formula (4) - (7), the R Factor and MOS score are used in this research and has been seen in Table V.

From Table IX, G.711 gets the highest values for R Factor and MOS. It show that G.711 is the best codec for VoIP quality. H.265 codec gets the highest values for R Factor and MOS, it can be analyzed that this codec has highest quality video codec for the best video quality.

IV. CONCLUSIONS

Voice over IP and video streaming are the real-time service and need the low latency value. The best service quality is the codec that has the lowest latency. From the simulation, the result show that the highest quality of codec for the VoIP highest quality was G.711 codec and the greatest quality

TABLE IX
R-FACTOR AND MOS COMPARISON.

	G.711	**G.729**	**G.723.1**	**H.264**	**H.265**
R Factor	87.1746	87.174	87.1729	86.7255	87.1408
MOS	4.2639	4.2638	4.2637	4.2508	4.2628

of codec for the video is H.265 codec. In addition, G.711 codec has the lowest latency and the highest MOS than any VoIP codecs in this research and also H.265 codec for video services has the nethermost latency and the highest MOS than any Video codecs.

Codec is implemented to save network bandwidth capacity. G.711 codec is not enough saving for bandwidth consumption in VoIP service, it obtained the best throughput among the other VoIP codec. Therefore, G.723.1 codec is the highest quality codec in efficiency of bandwidth. It can reduce the use of network bandwidth capacity up to 5 times than G.711 codec and obtain 43% network bandwidth capacity saving than G.729 codec. In video services, H.265 codec has 51% network bandwidth savings than the H.264 codec. The amount of UE influence the score of latency, throughput, and jitter. We conclude that the throughput is reducing performance as increasing the number of UE. We also shows that jitter and latency have increased because the of total UE is rising. The speed of UE influence the score of latency, jitter and throughput. Latency and jitter get worst if the speed of user gets faster.

The latency value and jitter for all VoIP codecs and Video codecs have been achieved the target latency and jitter by ITU-T which is $<$ 150 ms [25] for latency and $<$ 1 ms for jitter. In VoIP, the latency has been obtained the target latency of 5G technology around 1 ms, but for video services has not achieved yet. It only achieved the latency target at 30-50 UE scenario.

REFERENCES

[1] S. I. Popoola, N. Faruk, A. A. Atayero, M. A. Oshin, O. W. Bello, and M. Adigun, "5g radio access network technologies: research advances," in *Proc. of World Congress on Engineering and Computer Science (San Francisco)*, vol. 1, 2017.

[2] T. S. Rappaport, S. Sun, R. Mayzus, H. Zhao, Y. Azar, K. Wang, G. N. Wong, J. K. Schulz, M. Samimi, and F. Gutierrez, "Millimeter wave mobile communications for 5g cellular: It will work!" *IEEE access*, vol. 1, pp. 335–349, 2013.

[3] M. Mezzavilla, M. Zhang, M. Polese, R. Ford, S. Dutta, S. Rangan, and M. Zorzi, "End-to-end simulation of 5g mmwave networks," *IEEE Communications Surveys & Tutorials*, vol. 20, no. 3, pp. 2237–2263, 2018.

[4] ——, "End-to-end simulation of 5g mmwave networks," *IEEE Communications Surveys & Tutorials*, vol. 20, no. 3, pp. 2237–2263, 2018.

[5] M. Zhang, M. Mezzavilla, R. Ford, S. Rangan, S. Panwar, E. Mellios, D. Kong, A. Nix, and M. Zorzi, "Transport layer performance in 5g mmwave cellular," in *2016 IEEE Conference on Computer Communications Workshops (INFOCOM WKSHPS)*. IEEE, 2016, pp. 730–735.

[6] A. H. Mohammed, K. H. Bilal, and M. A. Hassan, "Voice over ip over lte network: A review," *Research gate*, vol. 28, no. 01, 2015.

[7] K. Jack, *Video demystified: a handbook for the digital engineer*. Elsevier, 2011.

[8] N. Barman and M. G. Martini, "H. 264/mpeg-avc, h. 265/mpeg-hevc and vp9 codec comparison for live gaming video streaming," in *2017 Ninth International Conference on Quality of Multimedia Experience (QoMEX)*. IEEE, 2017, pp. 1–6.

[9] B. B. Bista, S. Sasaki, K. Ogura, and T. Takata, "Reducing congestion in two-tier cellular network using multi-relay d2d communications," in *Proceedings of the International MultiConference of Engineers and Computer Scientists*, vol. 1, 2018.

[10] A. H. Wheeb, "Performance evaluation of udp, dccp, sctp and tfrc for different traffic flow in wired networks," *International Journal of Electrical and Computer Engineering*, vol. 7, no. 6, p. 3552, 2017.

[11] T. Wulandari, D. Perdana, and R. M. Negara, "Node density performance analysis on ieee 802.11 ah standard for voip service," *International Journal of Communication Networks and Information Security*, vol. 10, no. 1, pp. 79–84, 2018.

[12] M. H. Miraz, M. A. Ganie, S. A. Molvi, M. Ali, and A. H. Hussein, "Simulation and analysis of quality of service (qos) parameters of voice over ip (voip) traffic through heterogeneous networks," *International Journal of Advanced Computer Science and Applications (IJACSA), Online ISSN*, pp. 2156–5570, 2017.

[13] A. T. Liem, I.-S. Hwang, A. Nikoukar, and A. F. Pakpahan, "Sd-enabled mobile fronthaul dynamic bandwidth and wavelength allocation (dbwa) mechanism in converged twdm-epon architecture," in *2018 6th International Conference on Cyber and IT Service Management (CITSM)*. IEEE, 2018, pp. 1–6.

[14] I. Union, "Itu-t g. 1010: End-user multimedia qos categories," *G SERIES: Transmission Systems and Media, Digital System and Networks-Multimedia Quality of Service and Performance Generic and User-Related Aspects*, 2001.

[15] T. Janevski, "5g mobile phone concept," in *2009 6th IEEE consumer communications and networking conference*. IEEE, 2009, pp. 1–2.

[16] S. R. Arif, D. Perdana, T. Hasan, and I. Nashiruddin, "Analysis of connectivity model and encoding standards on ip interconnection implementation in indonesia (study case: Low data rate up to 72 mbps)," *Buletin Pos Dan Telekomunikasi*, vol. 16, no. 1, pp. 55–74, 2018.

[17] L. Derar and A. Mustafa, "Quality of service in umts network and improvement voip performance," *International Journal of Technology Enhancements and Emerging Engineering Research*, vol. 2, no. 10, 2014.

[18] M. H. Miraz, S. A. Molvi, M. Ali, M. A. Ganie, and A. H. Hussein, "Analysis of qos of voip traffic through wifi-umts networks," in *Proceedings of the World Congress on Engineering*, vol. 1, 2014.

[19] B. Pamukti, D. Perdana, and M. R. Kirom, "Thermal Effect Analysis of Arrayed Waveguide Grating in NG-PON2 Network." *Engineering Letters*, vol. 26, no. 2, pp. 216–223, 2018.

[20] H. Zarhouni, G. Orhanou, S. El Hajji, and R. Benaini, "Traffic engineering and optimization routing for voip traffic in wireless mesh networks," in *Proceedings of the World Congress on Engineering*, vol. 2. Citeseer, 2012.

[21] V. Parthasarathy, P. Anandakumar, and V. Rajamani, "Design, simulation and FPGA implementation of a novel router for bulk flow TCP in optical IP networks," *IAENG International Journal of Computer Science*, vol. 38, no. 4, pp. 343–349, 2011.

[22] E. Setijadi, I. Purnama, M. Hery *et al.*, "MPR Selection to the OLSR Quality of Service in MANET Using Preferred Group Broadcasting." *IAENG International Journal of Computer Science*, vol. 46, no. 2, pp. 192–198, 2019.

[23] H. Ifijeh, F. Idachaba, and I. Oluwafemi, "Performance evaluation of the quality of voip over wlan codecs," 2015.

[24] S. El Kafhali and M. Hanini, "Stochastic Modeling and Analysis of Feedback Control on the QoS VoIP Traffic in a single cell IEEE 802.16 e Networks," *IAENG International Journal of Computer Science*, vol. 44, no. 1, pp. 19–28, 2017.

[25] D. Perdana and R. F. Sari, "Performance comparison of ieee 1609.4/802.11 p and 802.11 e with edca implementation in mac sublayer," in *2013 International Conference on Information Technology and Electrical Engineering (ICITEE)*. IEEE, 2013, pp. 285–290.

Doan Perdana received his BSc and MSc degrees in Telecommunication Engineering, from the Institute of Technology Telkom, Indonesia in 2004 and 2012, respectively. He completed his PhD in Electrical Engineering Department, University of Indonesia. His interests include telecommunication systems and computer engineering.

Nadira Mumtaz received her BSc degrees in Telecommunication Engineering, from the Telkom University, Indonesia in 2019. Her interests include telecommunication systems and network engineering.

Yoseph Gustommy Bisono received his BSc and MSc degrees in Telecommunication Engineering, from the Institute of Technology Telkom, Indonesia in 2008 and 2012, respectively. His interests include network and telecommunications resource management.