Performance Evaluation of VoIP Traffic in 5G Millimeter Wave Network

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Abstract **- Fifth Generation of Cellular Network (5G) will provide large bandwidth capacity to support a variety of services. One of the commonly used is Voice over Internet Protocol (VoIP). VoIP service is very critical due to its real-time communication characteristics. There are several standards for VoIP codec, which are G.711, G.723.1 and G.729. Millimeter wave on 5G is expected to produce a more reliable Quality of Service as it works in high band frequency. This research addresses the performance of voice codecs on 5G millimeter wave network to be performed based on node density and node velocity. This research is carried out by running a millimeter wave algorithm with several voice traffic based on the codec in Network Simulator-3.27. The results are analyzed and compared for Quality of Service (QoS) which includes: latency, throughput and Mean Opinion Score (MOS) of each voice codec. The results show that G.711 codec provides the best voice quality on 5G millimeter wave, giving a throughput of 1.15 Mbps on average, the latency obtained was 1.19ms on average with a MOS value of 4.26370. While G.723.1 with 5.3 Kbps bitrate is the most efficient codec to save network bandwidth capacity as it obtained the lowest throughput of 0.21 Mbps among the other codecs. It can: i) save 5x network bandwidth capacity compared with G.711, ii) save 53% compared with G.729 and iii) save 7% compared with G.723.1 bitrate 6.3 Kbps. The latency value to achieve the expected characteristics for 5G network in the future is about 1 ms.**

Keywords - 5G, Millimeter Wave, VoIP, QoS, Network-Simulator 3

I. INTRODUCTION

Increasing of cellular data usage has grown very rapidly, it has created a challenge for cellular service providers to overcome bandwidth shortages. Cellular service providers are also challenged to provide high quality, low latency and reliable for data, video and multimedia services. Cellular service providers are begun to switch to next generation cellular technology (5G) which has a multi-Gigabit per Second (Gbps) data rate which is expected to answer the challenge of providing high-quality, low latency and reliable services. 5G also has wide bandwidth about 1 GHz to be able to serve many variety of services. For now, there is no standard for 5G technology, therefore researchers are currently conducting research to produce supporting technology for support development in 5G, one of the popular ones is millimeter wave technology is expected to achieve the target of 5G which is high quality, low latency and reliable services because this wave works on high frequency bands. However, there are several problems that must be studied so that millimeter wave technology is ready to be patented to support fifth generation cellular technology (5G).

Several researches have been conducted about millimeter wave and VoIP on cellular network. In [3] Marco Mezzavilla et al have been conducted some research about the implementation of 5G Millimeter Wave, they overviewed the function of 5G millimeter wave network module which they built on Network Simulator 3 [4]. They evaluated their MAC Layer about the performance of schedulers, RLC Layer about the multi connectivity with LTE network. They also have been conducted research about transport layer in 5G millimeter wave. They compared UDP and TCP traffic in 5G millimeter wave [5]. Theodore S. Rappaport et al have been conducted the research about millimeter wave mobile communications for 5G, they presented the motivation methodology, and hardware for new mm-wave cellular systems [6]. There are some researches about performance analysis of VoIP on LTE network [7] [8] [9]. But, the research about performance analysis of voice codec on 5G network have never been done before.

As a result of the rapid development of the internet, people prefer to use VoIP for voice communication rather than using PSTN because it is cheaper and more efficient. Voice over IP is a real-time service with very tight latency requirements, so that it requires a strong network to serve optimally. Voice over Internet Protocol (VoIP) is a technology that uses Internet broadband to make voice calls, not like a traditional (analog) telephone lines. VoIP uses codecs to encapsulate audio into data packets, send packets over IP networks and send packets into audio at the end of the connection. Codecs (coder / decoder) are used during voice communication. At the end of the transmission, the codec converts the analog signal to compress a digital bit stream, another codec converts the digital bit stream back into an analog signal. For package Real Time Protocol (RTP), the codec uses a type of payload or encoding method in VoIP technology. Generally, codecs provide the ability to save network bandwidth [1].

ITU-T (International Telecommunication Union - Telecommunication Sector) makes several standards for the implementation of VoIP. Some commonly known standards are G.711, G.723.1, and G.729.

In VoIP services, G.711 codec applied the Pulse-Code Modulation (PCM) method for voice frequency signals which are sampled 8000 samples / second which is 8 binary digits per sample. G.711 encoder has a 64 Kbps bit rate. G.729 Codec is one of the good quality codecs because it can convert voice to 8 Kbps which is less than G.711, this causes the G.729 codec to have more effective and efficient bandwidth utilization. G.723.1 codecs has two types of bit rates. First is the 6.3 Kbps bit rate is better quality to optimize for high-quality voice. The second bit rate is the smallest bit rate of 5.3 Kbps [6].

II. RESEARCH METHODS

Simulation of voice codecs performance simulated in Network Simulator 3 by changing the amount of data rate and payload that is contained in the application structure on millimeter wave module. Before carrying out the simulation, the calculation of total payload voice and channel bandwidth of each codec for conventional network will be carried out, the results of calculations performed will be compared with the throughput results of simulation in Network Simulator 3.

TABLE I. SIMULATION PARAMETERS

Parameter	Value
Frequency	28 GHz
Bandwidth	1 GHz
Number of EnodeB	1
Number of UE	10, 30, 50, 100, 150
Scheduler	Earliest Deadline First
PGW to Remote Server Latency	0.1 ms
Traffic Bitrate	64 Kbps, 8 Kbps, 6.3 Kbps
Traffic Payload	160 bytes, 20 bytes, 24 bytes
User's Velocity	50 Km/Hours, 100 Km/Hours, 150 Km/Hours, 200 Km/Hours
Velocity	Random Walk 2D
Core Network	EPC

Table I shows the simulations parameters that carried out into the simulation based on millimeter wave module characteristics. There is a change for number of UE and user's velocity, it for tested the capacity for each network. The traffic bitrate and traffic payload are adapted to the characteristics of the codecs.

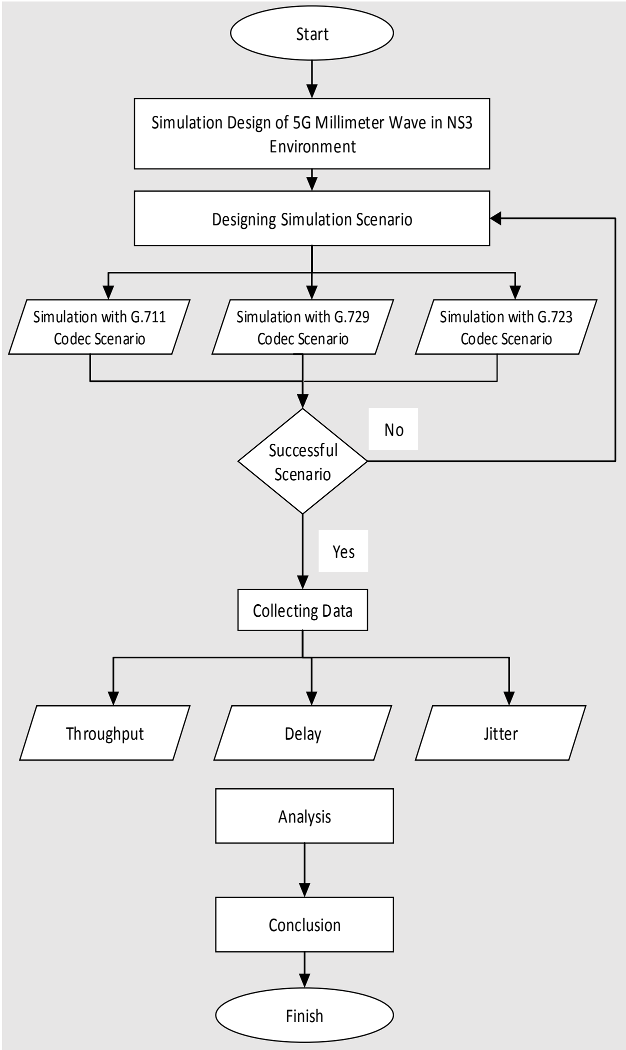

Figure 1. Flowchart System

The flowchart system for this research is presented in Figure 2. Based on the system, after designing the simulation of 5G millimeter wave in NS3 environment, designing simulation scenario based on Table 3 is implemented. If the scenario was succeed, throughput, latency and jitter will be collected.

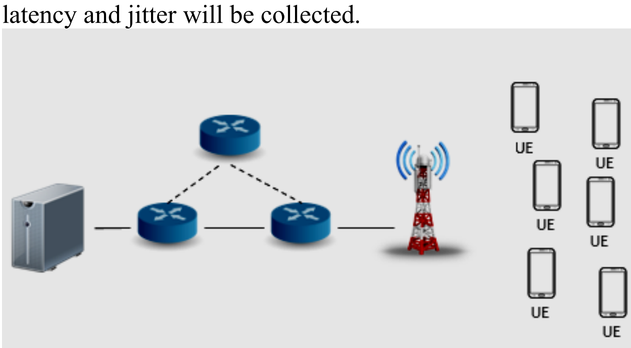

Figure. 2. Simulation Topology

Figure 2 shows the simulation topology, there is remote host as a server. It connected with the Packet Gateway which will route the traffic to an external network. PGW connected with S-GW for transfer traffic data to the EnodeB. EnodeB as an equipment to send radio transmission uplink and downlink to user equipment and also it is connected with MME to control signaling session. Min distance for the simulation is 10 meters and 200 meters for max distance.

$$\text{Total packet size} = a + b + c \tag{1}$$

Where:
a= L2 Header (bytes)
b= IP/UDP/RTP Header (bytes)
c= Voice Payload Size (bytes)

Equation (1) is used to calculate total packet size that will be transmitted. It includes the length of header and the length of payload.

$$PPS = \frac{\text{code bit rate}}{\text{voice payload size}} \tag{2}$$

Equation (2) is used to calculate total packet per second that will be transmitted. It includes the length of header and the length of payload. It influenced of the bitrate and the

$$\text{Bandwidth} = \text{total packet size} \times \text{PPS} \tag{3}$$

Equation (3) is used to calculate total bandwidth, but this bandwidth calculation used on conventional network.

The output from the simulation is QoS parameters which are as follows:

- Latency is the time needed for the package delivery process, which includes coding or decoding, 'packet-ization', processing, and network latency [10].

$$Delay = \frac{\text{T. Received Packet - T. Packet Sent}}{\text{Packet Received}} \tag{4}$$

- Throughput which is defined as the speed (rate) effective for transferring the data [11].

$$Throughput = \frac{\sum \text{Received packet size}}{\sum \text{Delivery Time}} \tag{5}$$

- Jitter is defined as the variation in the arrival time of consecutive packets. Jitter is calculated over an interval of time [12]. Target jitter for this research is < 1 ms based on ITU-T standardization.

Based on the results of QoS parameters, the Mean Opinion Score (MOS) is calculated. The Mean Opinion Score (MOS) is used to measure the voice quality in VoIP network In this research, MOS was calculated based on the results of E-model/R Factor calculations. E-Model/R Factor is a standard by ITU-T which is defined as a transmission quality factor caused by latency, echo, codec and compression and packet loss for calculate the quality of VoIP network. The equation of E-Model / R factor is presented in (6) (7) (8) [13].

$$R = 94.2 - Ie - Id \tag{6}$$

$$Ie = 0.024d + 0.11(d - 177.3)H(d - 177.3) \tag{7}$$

$$Id = 7 + 30\ln(1 + 15e) \tag{8}$$

Where:
d = one way latency (ms)
e = packet loss
H =

- H(x) = 0 , if x<0
- H(x)=1, if x>=0

The results of E-model / R Factor was correlated by the MOS. The relation between R factor and MOS has been presented as follows [14]:

$$MOS = 1 + (0.035R) + ((7 \times 10^{-6})R(R - 60)(100 - R)) \tag{9}$$

Where:
R= R factor

TABLE II. MEAN OPINION SCORE CATEGORIES

MOS	User Satisfaction
4.3 – 5.0	Very Satisfied
4.0 - 4.3	Satisfied
3.6 – 4.0	Some User Satisfied
3.1 – 3.6	Many User Dissatisfied
2.6 – 3.1	Nearly All User Dissatisfied
1.0 – 2.6	Not Recommended

Table II shows the value of MOS and its relationship with user satisfication level. This research is expected to reach the level either satisfied or very satisfied level.

III. RESULTS AND ANALYSIS

This section would perform the results that we have got after simulated voip and video traffic in 5G millimeter wave and used NS-3 as the simulation tool. The results divided into 4 parts which is throughput, latency, jitter and MOS result in Voip traffic.

A. *Throughput Performance Evaluation*

Based on equation (1) (2) and the value of bitrate and payload size for each codecs, the calculation of total packet size and total packet per second for each codec presented on Table III.

TABLE III. CHANNEL BANDWIDTH CALCULATION

	G.711	G.729	G.723.1	
Datarate (bps)	64000	8000	5300	6300
Payload	160	20	20	24
Header IP/UDP/RTP	40	40	40	40
VoIP Packet	200	60	60	64
VoIP Packet/s	50	50	33.125	32.8125
Channel Bandwidth (bps)	80000	24000	15900	16800

Table III shows the widest bandwidth channel obtained by G.711 codec, because the G.711 codec has the highest data rate and payload value among the other voice codecs.

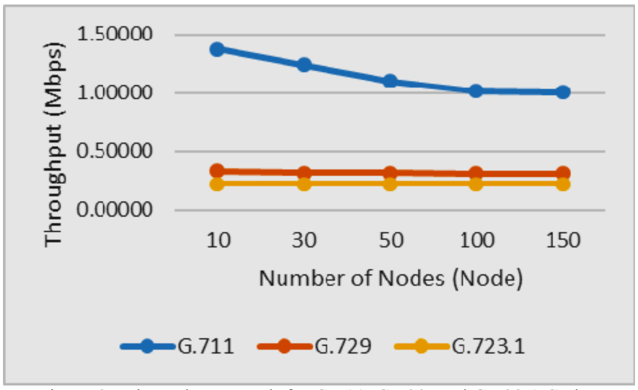

Figure 3. Throughput Result for G.711, G.729, and G.723.1 Codec

Figure 3 shows the throughput from the simulation results. It shows that G.711 codec gets the highest throughput among the other codecs. The G.711 codec gets an average throughput of 1.15 Mbps. The G.729 Codec gets an average throughput value of 0.32 Mbps. The G.723.1 codec with a 5.3 Kbps bitrate gets an average throughput of 0.21 Mbps while the G.723.1 codec with a 6.3 Kbps bitrate gets an average throughput of 0.22 Mbps.

The G.711 codec gets the highest throughput because it has 64 Kbps bitrate so it can send more packet at one time. Throughput is defined as the speed (rate) effective for transferring data, so bitrate affects the amount of packet data that will be delivered.

Based on Table III, for G.711 codec can send 50 voip packet per second. Moreover, the G.711 codec has more voip packet than the other codec. Voip packet value and voip packet per second value affect the amount of throughput. Codec G.723.1 is the most efficient codec to save network bandwidth in VoIP service. It can save 5x network bandwidth capacity compared with G.711, save 53% compared with G.729 and save 7% compared with G.723.1 bitrate 6.3 Kbps

Throughput result from simulation gets higher than the throughput result from calculation on Table 4. It is because in cellular network beyond 3G, for real-time services especially for VoIP, it has own path called IP Multimedia Subsystem (IMS). So VoIP traffic will not collide with the other traffic. Based on Figure 3 it can be concluded that the more number of users, the throughput will decrease due to the bandwidth capacity which will be shared with several users.

B. *Latency Performance Evaluation*

Latency is used to determine the duration time to deliver the packet. Target latency for 5G is 1 ms and target latency of ITU-T standardization is <150 ms.

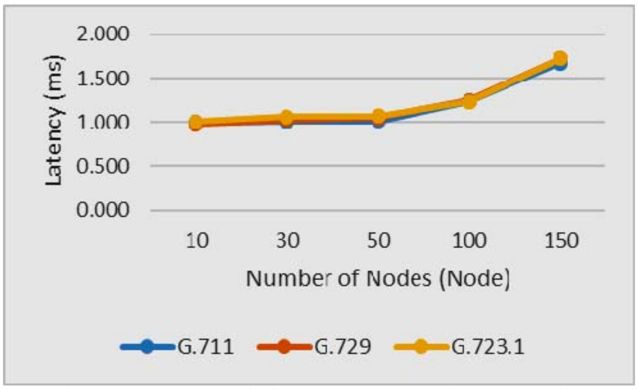

Figure 4. Latency Result for G.711, G.729, and G.723.1 Codec

Figure 4 shows the latency from the simulation results. The G.723.1 bitrate 5.3 Kbps codec obtained the highest average latency of 1.24 ms, G.723.1 codec with a 6.3 Kbps bitrate obtained an average latency of 1.22 ms. G.729 codec obtained an average latency of 1.21 ms and G.711 codec obtained the lowest average latency of 1.19 ms.

The latency of G.723.1 bit rate 5.3 Kbps codec gets the worst / highest latency, because it has lower bitrate than the other codecs. It only has bitrate about 5.3 Kbps.

The latency for this research has achieved the latency characteristics that set by ITU-T G.1010 which is around 0-150 ms for acceptable latency [15]. This latency also achieved the expected latency for 5G network which is about 1 ms [16]. Based on Figure 4 it can be concluded that the more number of users, the latency will increase due to the waiting time for each user to be served is getting longer.

C. Jitter Performance Evaluation

Jitter is used to calculate the variation in the arrival time of consecutive packets. The standard jitter based on ITU-T is < 1 ms [15].

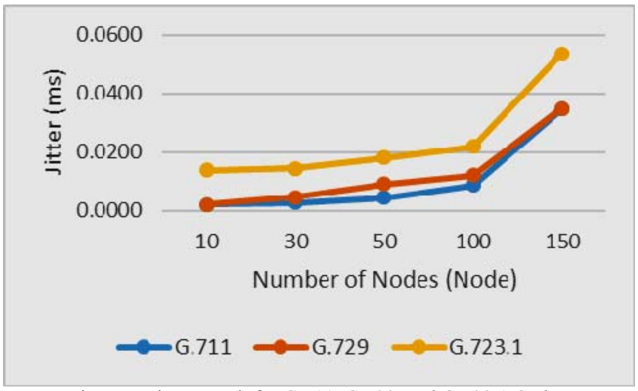

Figure 5. Jitter Result for G.711, G.729, and G.723.1 Codec

Figure 5 shows jitter from the simulation results. The G.723.1 with 5.3 Kbps bitrate codec obtained the highest jitter an average jitter of 0.0237 ms. The G.723.1 with 6.3 Kbps bitrate codec obtained an average jitter of 0.0244 ms. The G.729 codec obtained an average jitter of 0.0125 ms. and G.711 codec obtained the lowest latency of 0.0105 ms.

Jitter is related to latency, it is the value of the total variance of latency, it occurs due to the difference in arrival time between packets at the destination. Based on ITU-T G.1010 the value of acceptable jitter for real-time services. It can be concluded that all the codec in this research have an acceptable jitter. But, G.711 codec has the best/lowest jitter as it has the highest bitrate, so it only takes a short time to deliver all the packets.

The number of users affect the jitter. Based on Figure 5 , it can be concluded that the more number of users, the jitter will increase due to the waiting time for each users to be served is getting longer and the latency is getting higher too. As we know, jitter is related to latency.

D. Mean Opinion Score

Mean Opinion Score (MOS) has been calculated to determine which codec produces the best quality for VoIP and video service. Based on equation (4) and (5), the R Factor and MOS value for each services presented in Table V. Based on (7) we calculated the value of H(x) is 0, because the value of x is <0. So we can concluded the value of Ie for each codec is 0.024d. Based on (8) packet loss for each codec is 0, so the value if Id for each codec is 7.

a. G.711 CODEC

Based on simulation results, delay for G.711 codec is 1.19 ms. So the value of R factor is:

$$R = 94.2 - (0.024 \times 1.19) - 7$$
$$R = 87.1715$$

b. G.729 CODEC

Based on simulation results, delay for G.729 codec is 1.21 ms. So the value of R factor is:

$$R = 94.2 - (0.024 \times 1.21) - 7$$
$$R = 87.1710$$

c. G.723.1 (5.3 Kbps Bitrate) CODEC

Based on simulation results, delay for G.723.1 (5.3 Kbps) codec is 1.24 ms. So the value of R factor is:

$$R = 94.2 - (0.024 \times 1.24) - 7$$
$$R = 87.1703$$

d. G.723.1 (6.3 Kbps Bitrate) CODEC

Based on simulation results, delay for G.723.1 (6.3 Kbps) codec is 1.22 ms. So the value of R factor is:

$$R = 94.2 - (0.024 \times 1.22) - 7$$
$$R = 87.1707$$

TABLE IV. MOS AND R-FACTOR COMPARISON

	G.711	**G.729**	**G.723.1 (5.3 Kbps)**	**G.723.1 (6.3 Kbps)**
R Factor	87.1715	87.1710	87.1703	87.1707
MOS	4.26370	4.26369	4.26366	4.26368

Table IV shows the MOS for each codec is around 4.2 but codec G.711 has the highest value of MOS among the other codecs. It can be concluded that codec G.711 has the best quality for VoIP services based on this MOS value.

IV. CONCLUSIONS

In this research, we compared the throughput and latency for VoIP codec G.711, G.729 and G.723.1 based on simulation results in Network Simulator 3. Voice over IP is a real-time service with very tight latency requirements, so we concluded that the best codec for the best voice quality is G.711 codec. Based on the results, G.711 codec has the lowest latency than any codecs in this research. But codec G.711 is not efficient for bandwidth consumption to sending voice packages in one shipment, because this codec uses a bandwidth of 64 Kbps for sending voice packets. Therefore, G.723.1 with bitrate 5.3 Kbps is the best codec for bandwidth efficiency. It can save 5x network bandwidth capacity compared with G.711, save 53% compared with G.729 and save 7% compared with G.723.1 bitrate 6.3 Kbps. Because it is the most efficient codec, so it's the best codec to use in communication which need low bandwidth capacity, for the example is teleconference. The latency value for all VoIP codecs, has been achieved the expected characteristics for 5G network which is about 1 ms. The number of users affect the value of throughput, latency and jitter. For the throughput, the more number of users, the throughput will decreased. Meanwhile, the latency and jitter will increased due to the increasing of total user. In the future, other researcher could add some services such as IoT, game streaming, d2d, etc and could achieve the target latency of 5G network and also have a standalone architecture which is use 5G Core, not 4G Core (EPC).

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