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Hybrid Estimation of VoIP Codec Techniques in Long Term Evolution and 802.11ac Networks

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Abstract:
There are numerous high-speed access connections for mobile end-users, including Wi-Fi and Long Term Evolution (LTE), in reaction to the increasing demand for greater quality Voice over IP (VoIP) communication. The mixture of access connections offers a hybrid network environment in which end-users can switch from one to the other, whichever offers a greater level of VoIP service quality (QoS). In addition to the sort of access link, VoIP codecs are another important factor that directly impacts speech communication's of general QoS. Several networks have separate constraints and requirements due to intrinsic features. Taking these distinctions into account, the visualization and analysis of each codec's performance and behavior on its underlying network will result in a correct VoIP codec choice, resulting in ideal speech QoS for mobile end-users. This research proposes a technique for quantifying and analyzing the efficiency of various VoIP codecs in a hybrid LTE-802.11ac network where two respective network interface cards are available to mobile end-users. The goal is to discover the codecs that best fit LTE and 802.11ac networks and optimize the VoIP communication. The NS3 tool is used to create and execute a range of separate scenarios that accurately measure different QoS performance metrics. The findings collected indicate the comprehensive effect of codecs on the speech communication QoS for both LTE and 802.11ac users as well as the significance of the VoIP codec selection process for each network.

Keywords: *802.11ac, Hybrid LTE-802.11ac, LTE, QoS, VoIP codecs.*

I. INTRODUCTION

As the number of mobile subscribers and also the velocity of network connections are increasing, there is also increasing demand for high-quality multimedia apps. The latest advances in long-term development (LTE) and IEEE 802.11ac networks are aimed at meeting the needs of users by offering a wide variety of high-performance network facilities including Voice over Internet Protocol (VoIP). VoIP systems include methods for providing telephony facilities and sending voice via Internet links via packet information networks (PDNs). Many of the main advantages of VoIP systems, such as simple installation and setup, a broad range of calling features and cost advantages, make these systems highly appealing for use in a variety of wired or wired data networks. The VoIP systems use compression and decompression (codec) methods to compress speech signals before transmission in the form of VoIP packets due to bandwidth constraints in the data networks. They can be split into separate classes based on how the codecs perform their tasks, each with its separate characteristics such as speech quality, bandwidth, and computational requirements. The current codecs can be classified into three groups: waveform, hybrid, and source code[1] while this research focuses on the first two classes. The codecs of the waveform take the audio signal from the input and transform it into a digital signal that is then packaged after sampling. This technique provides high-quality voices but at the cost of a higher bitrate which can be a challenge for bandwidth-limited networks. This challenge is taken into consideration by the hybrid codecs. While providing excellent quality voices, reduced bit rates are required compared to codecs in the waveform. This research explores the most popular waveform codecs, including G.711 (PCM) and G.726 (ADPCM), together with the most popular hybrid codecs, including G.723 (ACELP), G.728 (LD-CELP), and G.729 (CS-ACELP). Since separate networks have distinct demands, the sort of codec used in VoIP communication is regarded to be the main variable that can directly influence the general quality of service VoIP (QoS) as it is either optimized or degraded. Thus, the primary issue here is what sort of VoIP codecs is more suitable for optimizing the VoIP communication QoS in LTE or 802.11ac network. This work has three primary contributions to solve the issue. First, a technique for assessing distinct waveform kinds and hybrid VoIP codecs for two distinct networks, including LTE and 802.11ac, is suggested using the NS3 network simulator tool in a hybrid setting. Besides, the functional measurement of the QoS obtained by mobile end-users in terms of distinct performance metrics. At this stage, the performance of end-to-end codecs is taken into account, irrespective of the amount of processor utilization, based on the experience of the user when receiving the packets. Ultimately, the findings will be examined for an appropriate choice of VoIP codecs for either LTE or 802.11ac networks to optimize their speech quality output. This step is critical while considering the essential requirements of LTE and 802.11ac networks and their fundamental differences. The structure of the document is shown as follows. Section 2 provides the associated works and Section 3 defines the comprehensive simulation setup design as well as the scenarios performed. Section 4 offers the outcomes of the simulation and the research is completed in Section 5.

II. RELATED WORKS

The characteristics are investigated by several VoIP codecs, which can greatly affect the general speech quality in the underlying network. In [2], the writers provide a comparative codec assessment including GSM-FR, G.711, G.723, and G.729. The objective of the job is to use the OPNET Modeler simulation tool to discover the right codec for wired, UMTS and WiMAX networks. The G.723 was the best codec to be used in all WiMAX, UMTS, and wired networks based on their outcomes. However, to determine the ideal VoIP codecs, the work does not take into account LTE and 802.11ac networks. The G.711, G.729A, G.723.1, and GSM.AMR codecs wired networks are also examined using NS2 in [3]. The speech codecs' impact over LTE (VoLTE) on end-to-end voice is illustrated in[4]. The OPNET Modeler is used to assess the GSM-EFR, AMR12.2 K, IS641, G.711, and G.729A codecs in terms of metrics including MOS, packet delay, sent-received voice traffic, and speech packet delay variation. The research concludes that the G.711 and GSM EFR codecs can achieve better output. The writers do not examine the hybrid topology as a mixture of both LTE and 802.11ac and other codec kinds as well. The tool OPNET Modeler is also used to explore LTE network efficiency in the presence of G.729 codec in [5], G.711 in [6], and G.711 and G.723 in [7]. In the presence of the G.711, G.723.1, G.729A, G.728, G.726 and GSM-AMR codecs, the WLAN performance using OPNET Modeler is explored in[8]. The findings of OPNET throughput and delay indicate that GSM-AMR codec will provide the best quality of service over WLAN for the

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VoIP. The work does not, however, clarify the type of WLAN being simulated while LTE is not being examined either. In a testbed setting, the WLAN is also examined with the G.711, G.723, and G.729 codecs in[9] to assess the codecs ' power usage. In the presence of GSM-FR, G.711, G.723, and G.729 codecs, the MOS parameter in LTE are examined in[10] using OPNET Modeler. The findings indicate that the greater MOS values are given by the G.711 codec. The achievement of AODV, DSR, DYMO and OLSR routing protocols with G.711, G.723, and G.729 codecs is also given in [11] using the EXata / Cyber simulator tool. Based on the existing research, the preceding works primarily examine codec performance inhomogeneous networks while there is also no present work on the performance of VoIP codecs in 802.11ac networks. On the other side, owing to the presence of various kinds of network access connections such as Wi-Fi or LTE, consumers can move to whatever access connections provide better speech QoS. However, in a hybrid heterogeneous setting, there is no present work on comparing LTE and 802.11ac and this region is not yet known. Considering the constraints of prior research, this work is a preliminary effort to provide over a hybrid LTE-802.11ac network a thorough comparative assessment of popular VoIP codecs. Unlike the existing works, the main motive here is to determine precisely the type of VoIP codecs that can efficiently deliver higher voice quality in LTE and 802.11ac networks. Providing a thorough and precise results assessment will assist network developers and suppliers in designing and developing effective VoIP applications and services.

III. SIMULATION SETTINGS DESCRIPTION

A hybrid network is made up of LTE and 802.11ac is simulated in which a variety of scenarios are developed and the corresponding results are measured to achieve the three main contributions of this work. This chapter explains the information of the topology for hybrid simulation using the NS3 tool as well as implementing the scenarios created.

Hybrid simulation setup

Twelve hybrid mobile user devices (UE) are intended in this job. Each UE is fitted with two distinct network interface cards (NICs), one to support LTE and the other to support 802.11ac networks to be able to operate as a hybrid UE. The WIFI PHY STANDARD 80211ac and StaWifiMac are added as their PHY and MAC layer to the UEs for the 802.11ac NIC. This leads to the UE working in VHT mode. Also, the InstallUeDevice is used to create the LTE NICs, making the UEs work in LTE mode. After this method, either LTE or 802.11ac networks can be connected by the UEs, whichever has a greater signal-to-noise ratio (SNR). To do this, a Boolean switch parameter is made called a wifilteswitch with a true value for turning to the LTE network and a fake value for changing to the 802.11ac network based on the SNR obtained. The LTE core is triggered when moving to the LTE network (wifilteswitch is true) and each hybrid UE is linked to the Packet Data Network (PDN) via the eNodeB which is attached to the PDN portal (PGW). The 802.11ac access point is triggered when switching to the 802.11ac network (wifilteswitch is incorrect), and each hybrid UE is linked to the PDN via an 802.11ac access point. There is also a VoIP service that can manage voice calls between UEs with distinct kinds of codecs. You can find the constructed hybrid topology in Fig. 1.

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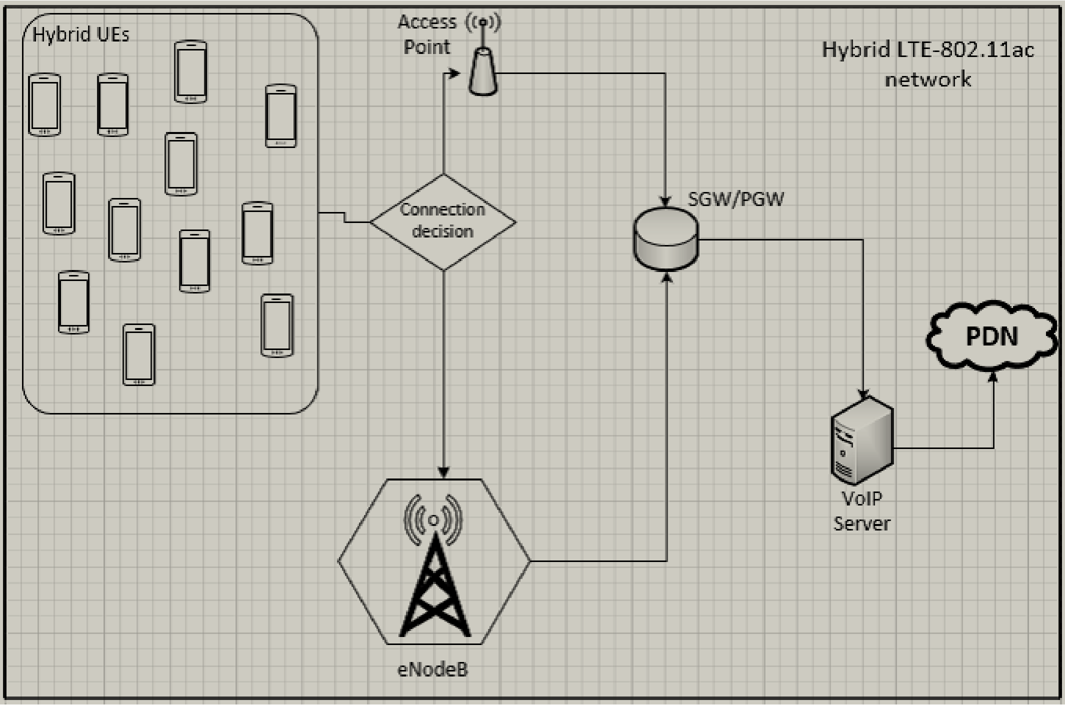

Figure 1.Simulation topology of the hybrid LTE-802.11ac network.

Simulation scenarios and design parameters

In this job, 48 separate scenarios are set up to examine different elements of LTE and 802.11ac concerning VoIP codecs. For both networks, the modulation algorithm, coding frequency, and channel width are regarded the same as 64QAM[12], 5/6, and 20 MHz, to provide accurate and fair circumstances for comparison of LTE and 802.11ac. The number of hybrid users in both networks is 12 and the simulation time is set as 8 seconds. The G.711 (PCM) and G.726 (ADPCM) categories of waveform codecs are analyzed and the G.723 (ACELP), G.728 (LD-CELP) and G.729 (CS-ACELP) categories are analyzed. The respective codec data rate and packet size are simulated to execute each codec, while each speech packet is pre-transmitted with 12 bytes of RTP header size. During the VoIP call between the customers, it is presumed that each discussion continues without intermittent silence. Four QoS performance metrics, including the percentage of throughput, delay, jitter, and packet loss, are evaluated to assess the efficiency of each VoIP codec. The per-flow measurements are carried out. Therefore, having 12 VoIP end-users offers 12 individual flows per second, which will be 96 individual flows per full simulation moment of 8 seconds. Some of the LTE-specific simulation parameters and 802.11ac networks in the hybrid topology are provided in Table 1.

Table 1. Simulation parameters

LTE parameters (wifilteswitch=true)	
Channel Bandwidth	100 RB (20 MHz)
eNodeB TxPower	14.0 dBm
Radio link control mode	RLC unacknowledged mode
Number of PGW	1
802.11ac parameters (wifilteswitch=false)	
Modulation coding scheme	VhtMcs7
Physical channel width	20 MHz
Number of 802.11ac AP	1
Wi-Fi type	SpectrumWifiPhy
Common parameters	
Number of users	12
Number of VoIP server	1
Types of VoIP codecs	G.711 (PCM), G.726 (ADPCM), G.723 (ACELP), G.728 (LD-CELP), G.729 (CS-ACELP).
Modulation algorithm	64QAM
Coding rate	5/6
Simulation time	8s

IV. PERFORMANCE EVALUATION

This chapter introduces the outcomes acquired in the constructed hybrid topology from the 48 tests. The associated comparative performance analysis

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is also given to highlight the speech quality in each specific situation.

G.711 (PCM) codec

This scenario is designed to evaluate the G.711 codec's operation and functionality to determine its limitations or advantages in LTE and 802.11ac networks. This experiment's simulation outcomes in terms of performance, loss ratio, delay, and jitter are shown in Fig. 2. Based on the outcomes acquired, the G.711 codec achieves nearly the same average throughput for both LTE and 802.11ac. While there is no significant distinction in the quantity of average throughput, at the start of the simulation run, the two networks demonstrate distinct behavior. The LTE begins with elevated throughput as the throughput gradually decreases and then stays continuous to the end. 802.11ac, on the other hand, begins with reduced throughput while gradually increasing until it stays continuous. LTE demonstrates much better efficiency in terms of loss proportion compared to 802.11ac. The average loss ratio in LTE is 0.0088 compared to 0.0102 in 802.11ac, which is much greater, resulting in degradation of the VoIP call. Also, LTE and 802.11ac demonstrate interesting distinctions in latency-related outcomes. The delay in LTE varies as time goes by while the values follow the same pattern. Meanwhile, with no significant differences, the delay in 802.11ac is greater and continuous (0.03s) throughout the simulation period. As a result, this offers zero jitters for 802.11ac unlike LTE with greater jitter, which can influence the quality of the VoIP packets obtained and leads to bad or scrambled speech communication.

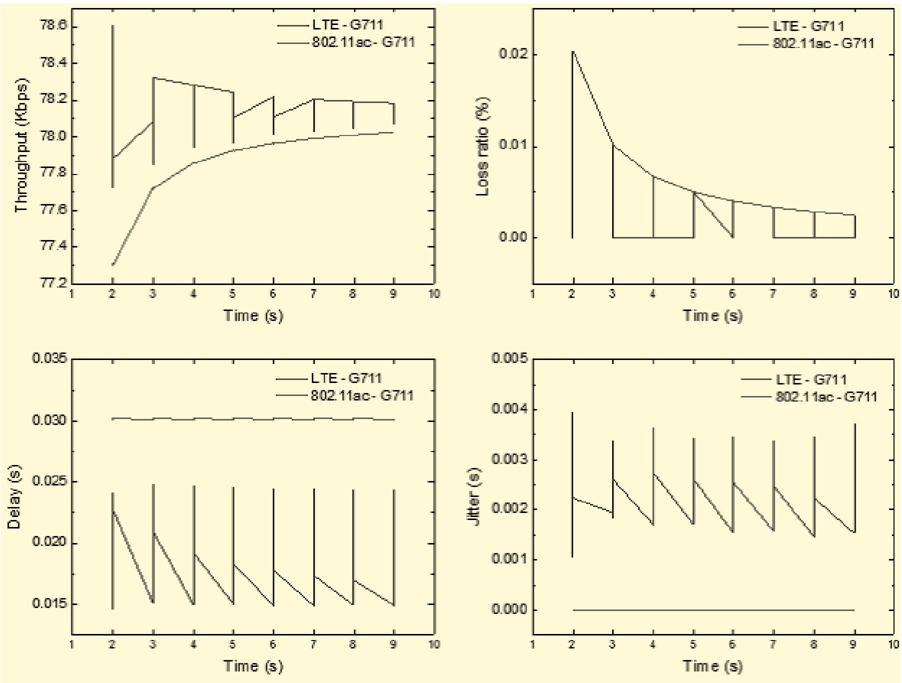

Figure 2. G.711 VoIP codec performance.

G.723 (ACELP) codec

To determine the level of QoS achieved by end-users applying the G723 codec in LTE and 802.11ac, the previous experiment is replicated under conditions where the G.723 codec is used in all VoIP communications. The main goal is to evaluate the features available in the G.723 codec that affect voice performance and quantify the relevant impacts. The findings associated with QoS are shown in Fig. 3. These results show that in LTE the G.723 codec works better than in 802.11ac. At the beginning of the time, the LTE throughput has its highest value while it declines gradually and remains steady until the simulation

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time has ended. On the other hand, over the entire time for 802.11ac, there is a constant throughput (15.55Kbps) with the values close to when LTE remains in its stable state. Furthermore, the findings suggest that the loss ratio in LTE reaches zero at some points while the ratio is equivalent to the values in 802.11ac for the remainder of the moment. Additionally, the 802.11ac delay in this experiment does not show any differences as compared to the previous experiment (G.711). In this case, the delay is constant with the same value as before (0.03s), while the LTE delay is higher with variable values over the whole simulation time. Because the continuous delay produces zero jitters, there are zero jitters in 802.11ac compared to elevated jitter in LTE. In the past experiment using the G.711 codec, the average jitter was 0.0024s, while in this experiment using the G.723, the jitter runs up to approximately 0.0031s, which impacts the speech packets ' QoS more negatively than before, causing noticeable speech degradation.

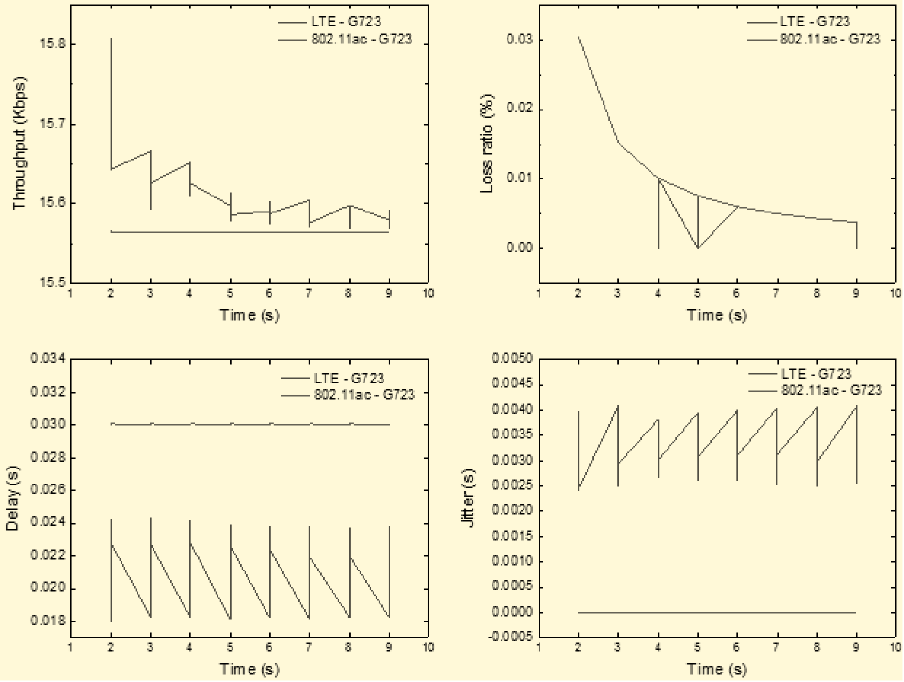

Figure 3. G.723 VoIP codec performance.

G.726 (ADPCM) codec

This scenario models the practical distinctions of the G.726 to be contrasted with the outcomes of the above studies to define the end-to-end performance of the G.726 codec. The findings will be shown in Fig. 4. The results of the G.726 codec application show that the average throughput is the same for both LTE and 802.11ac despite having distinct per-flow throughput patterns during the simulation moment. The findings indicate a greater frequency in 802.11ac compared to LTE concerning the loss ratio. This means that 802.11ac is more susceptible than LTE to missing the target audio data. Also, as compared to past experiments, we do not observe any noticeable difference in delay. As before, when using the G.726 codec, the delay in 802.11ac is continuous and also greater than LTE, resulting in a slightly greater delay in LTE when compared to G.711 or G.723 VoIP. Because of the continuous delay, jitter is zero in 802.11ac while the average value in LTE is about 0.0021s smaller than jitter in either the G.711 or G.723 VoIP codecs.

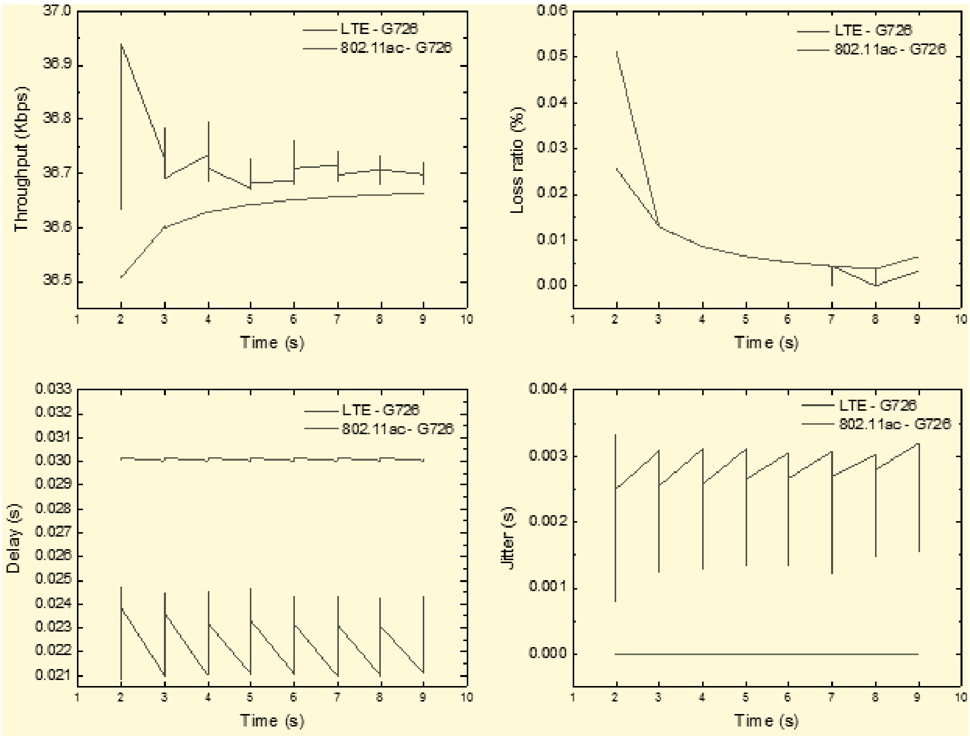

Figure 4. G.726 VoIP codec performance.

G.728 (LD-CELP) codec

To characterize the efficiency of the G.728 VoIP codec used in LTE and 802.11ac, this scenario is carried out. The findings will be shown in Fig. 5. For both LTE and 802.11ac, the evaluation of the output outcomes indicates low voice throughput. The interesting finding is the steady performance in 802.11ac while LTE follows the same trend as before, which is greater at the start and then reaches a steady state after a while. The loss percentage of the G.728 codec in 802.11ac is much greater than that of LTE. The measurement values indicate that the ratio in LTE is approximately 0.009 while in 802.11ac it is 0.012. We noted the same coherent pattern and near values based on the analysis of the delay measurements in the three past studies. Therefore, in this experiment, we were curious about the pattern of delay whether or not the pattern will be repeated. Measuring the delay also demonstrates the same values related to latency for the G.728 codec. The delay for 802.11ac is continuous (0.03s), while it differs in LTE with the same pattern up and down as time goes by. Like before, the constant delay causes zero jitters in 802.11ac but we observe higher jitter (0.003s) in LTE.

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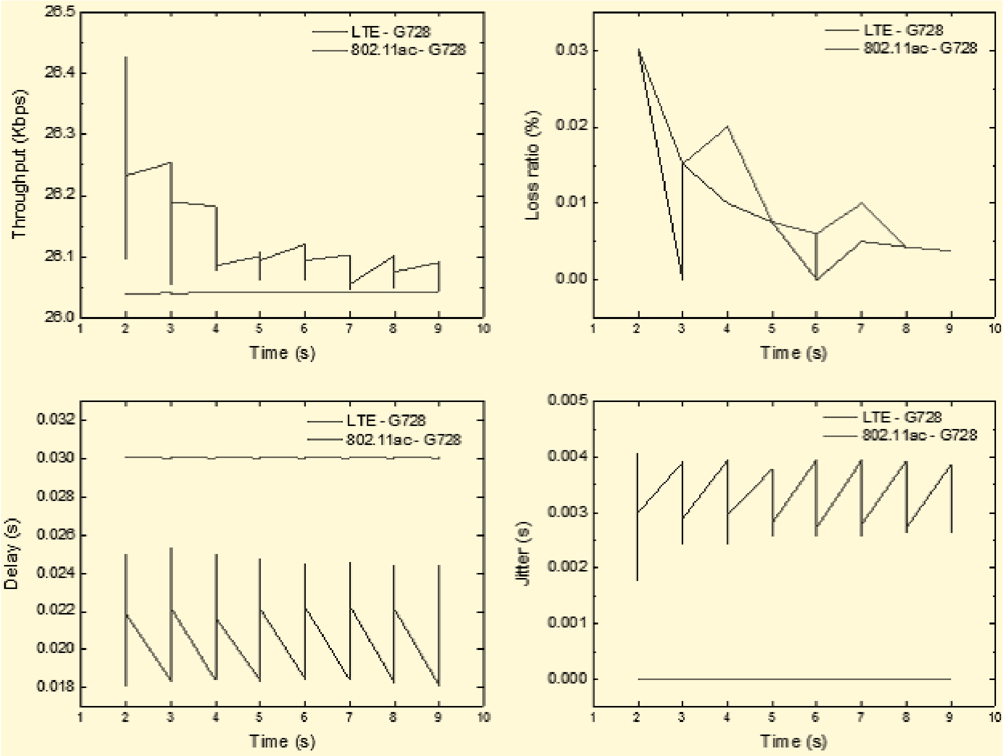

Figure 5. G.728 VoIP codec performance

G.729 (CS-ACELP) codec

In the presence of the G.729 VoIP codec, the design of this situation is based on evaluating LTE and 802.11ac operating efficiency. The findings will be shown in Fig. 6. Comparing the outcomes of this experiment with prior studies shows that the G.729 codec in both LTE and 802.11ac networks provides the greatest throughput accomplishment. The average LTE output is around 234.88Kbps compared to 233.58Kbps in 802.11ac, which in our studies has been the largest throughput so far. The loss ratio for LTE and 802.11ac is also about 0.005 and 0.01. These ratio values indicate that 802.11ac is much more likely to lose speech packets using the G.729 codec, resulting in reduced VoIP communication compared to LTE. In terms of delay, we observe the same pattern of latency as our prior studies, while LTE can attain the least quantity of delay (0.016s) between them in this experiment. The jitter as before in 802.11ac is zero while for the jitter this moment LTE provides nearly a steady value (0.0025s).

iLBC codec

In this experiment, the performance evaluation of the iLBC VoIP codec is presented to be compared in previous experiments with the other codecs. The results will be shown in Fig. 7.

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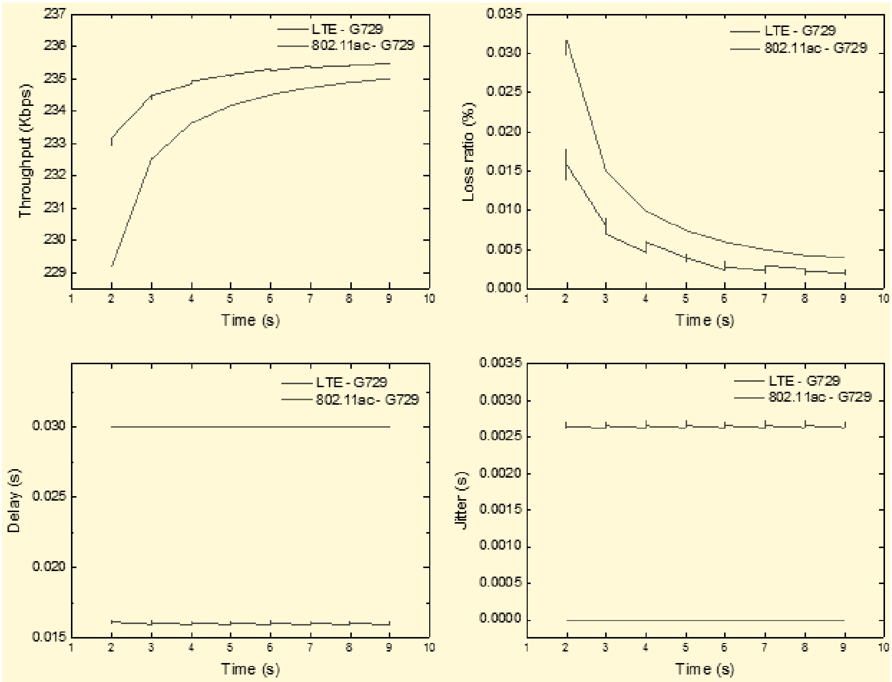

Figure 6. G.729 VoIP codec performance

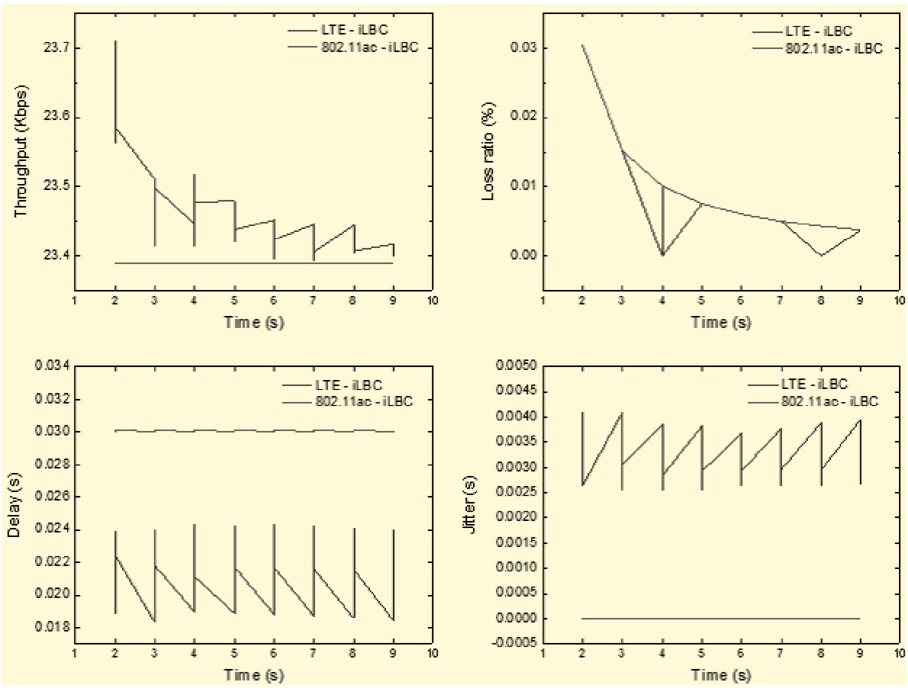

Figure 7. iLBC VoIP codec performance

The last codec to look at is the iLBC. The findings acquired to verify the consistency of the other studies. A summary of the measurement outcomes from the execution of all the tests is given in Fig to finish our job. 8, which means that the G.729 codec can be enhanced in a hybrid setting for both LTE and 802.11ac networks.

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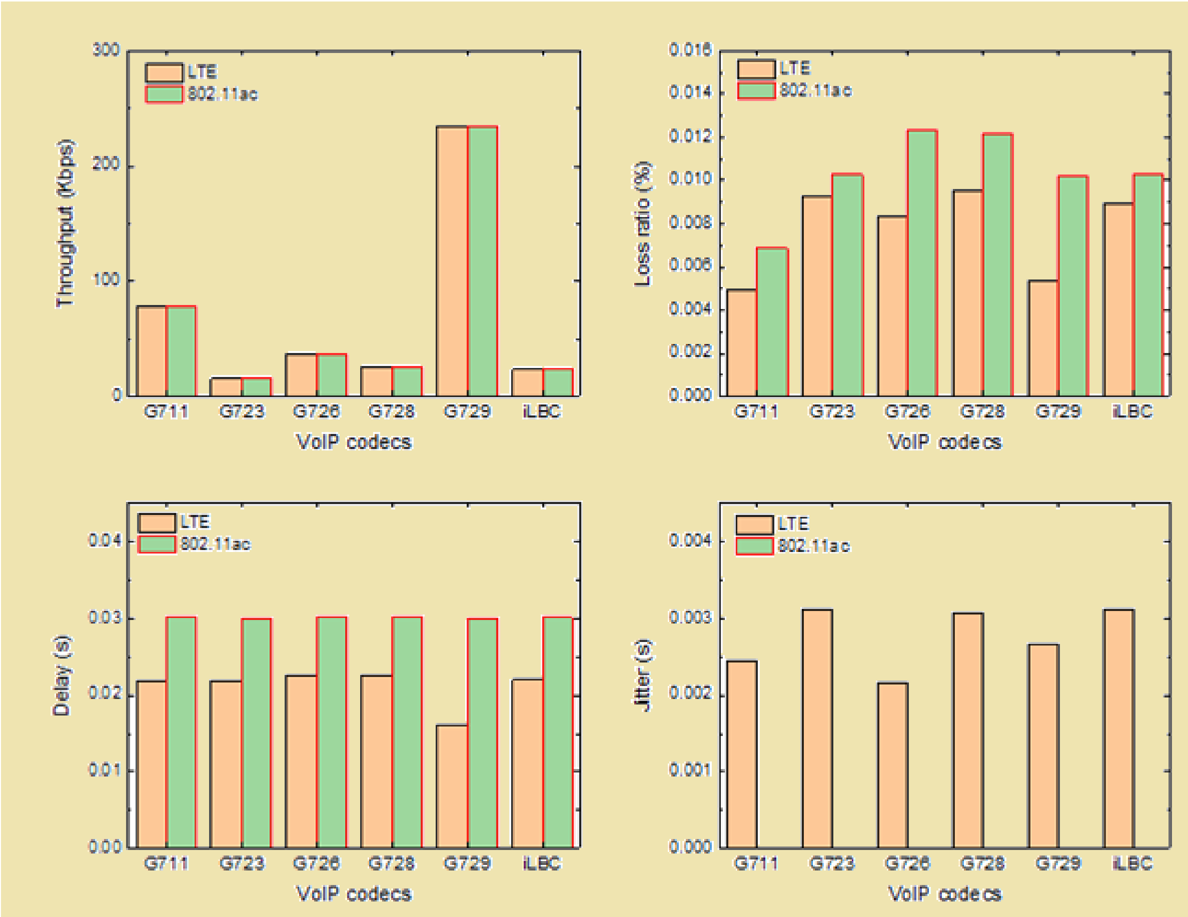

Figure 8. Performance comparison of all the VoIP codecs

V. CONCLUSIONS

This work has created a technique of application in a hybrid network LTE-802.11ac. Different scenarios were intended and implemented based on the most prevalent VoIP codecs, and four performance metrics related to QoS were analyzed. The experimental results outlined two findings related to the measurements of the throughput. First, there are no significant variations in the throughput between LTE and 802.11ac as the values are tightly linked to the same codecs. Second, regardless of whether you are an LTE user or 802.11ac user, the highest performance can be obtained with the G.729 VoIP codec while the lowest performance was assessed when the G.723 VoIP codec was introduced to the network. Also, the experimental findings indicated that for 802.11ac end-users the frequency of loss packets was always greater than for LTE end-users. The thorough assessment of the outcomes of the delay does not confirm important distinctions in the 802.11ac network between distinct codecs. Regardless of the codec type, end users in the 802.11ac network always experience the same amount of delay during the voice conversation, which is also always higher than the delay for end-users of the LTE network. Furthermore, the smallest value can be achieved when LTE end-users use the G.729 codec. In comparison to the greater jitter experienced by LTE end-users, the continuous delay given by all codecs for 802.11ac users outcomes in zero jitters.

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