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HYBRID LTE-802.11AC NETWORK: QOS OPTIMALITY EVALUATION OF THE VOIP CODECS TECHNIQUES

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Abstract

In response to the growing demand for higher quality Voice over IP (VoIP) communication, there are multiple high-speed access links, including Wi-Fi and Long Term Evolution (LTE) for the mobile end-users. The combination of the access links provides a hybrid network environment in which the end-users can switch from one to another, whichever provides a higher level of VoIP quality of service (QoS). Apart from the type of the access link, the VoIP codecs are also another key factor that directly affects the overall QoS of the voice communication. Due to inherent characteristics, different networks have distinct limitations and requirements. Considering these differences, the visualizing and analysing the performance and behaviour of each codec on its underlying network will lead to a proper VoIP codec selection, which in turn will result in optimal voice QoS for the mobile end-users. This study proposes a method to quantify and analyse the performance of different VoIP codecs in a hybrid LTE-802.11ac network in which the mobile end-users have two corresponding network interface cards. The aim is to find the codecs that suit the most for LTE and 802.11ac networks and thereby optimize the QoS of the VoIP communication. The NS3 tool is used to develop and implement a variety of distinct scenarios within which different QoS performance metrics are precisely measured. The obtained results signify the extensive impact of the codecs on the QoS of the voice communication for both LTE and 802.11ac users and also the importance of the VoIP codec selection procedure for each network.

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

1. Introduction

As the number of mobile subscribers and also the speed of network links are growing, the demand for high-quality multimedia applications is increasing as well. The recent advancements of the long-term evolution (LTE) and IEEE 802.11ac networks aim to meet the subscribers demands by providing a broad range of high-performance network services including Voice over Internet Protocol (VoIP).

The VoIP systems include techniques to provide telephony services and to send voice over the packet data networks (PDNs) using Internet connections. Many key benefits presented by the VoIP systems, such as easy installation and configuration, a wide range of call features, and also cost benefits, make these systems extremely attractive to be used in a variety of data networks either wired or wireless.

Due to bandwidth constraints in the data networks, the VoIP systems use compression and decompression (codec) techniques to compress the voice signals before transmission in form of VoIP packets. Based on how the codecs perform their functions, they can be divided into different classes each with its own distinct features such as the voice quality, bandwidth, and computational requirements. The existing codecs can be categorized into three classes: waveform, hybrid, and source coding [1] while the first two classes are the main focus of this work.

The waveform codecs take the input audio signal and convert it into a digital signal, which subsequently is packetized 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 they provide good quality voices, they require lower bit rates in compared to the waveform codecs. This work investigates the most common waveform codecs including G.711 (PCM) and G.726 (ADPCM) along with the most common hybrid codecs including G.723 (ACELP), G.728 (LD-CELP), and G.729 (CS-ACELP).

As different networks have different requirements, the type of codec used in the VoIP communication is considered as a key factor that can directly affect the overall VoIP quality of service (QoS) as either optimizing or degrading it. Thus, the main question here is that which type of the VoIP codecs suits more in LTE or 802.11ac networks to optimize the QoS of the VoIP communication. To address the question, this work has three main contributions. First, using the NS3 network simulator tool, a method for evaluating different types of waveform and hybrid VoIP codecs for two different networks including LTE and 802.11ac in a hybrid environment is proposed. Furthermore, by measuring the QoS achieved by the mobile end-users in terms of different performance metrics, the functionality of the two networks are compared. In this stage, the performance of the end-to-end codecs is taken into account based on the user's experience upon the packets reception regardless of the amount of processor utilization. Eventually, the results are examined for optimal VoIP codec selection for either LTE or 802.11ac networks to optimize their performance in term of voice quality. This step is critical while considering the essential requirements of LTE and 802.11ac networks and their fundamental differences.

The paper structure is depicted as follow. Section 2 presents the related works and Section 3 describes the detailed design of the simulation setup along with the conducted scenarios. Section 4 provides the simulation results and Section 5 concludes the work.

2. Related Works

VoIP codecs, which can highly influence the overall voice quality in the underlying network, there are a number of VoIP codecs-related studies investigating the features. The authors in [2] provide a comparative analysis of the codecs including GSM-FR, G.711, G.723, and G.729. The aim of the work is to find the proper codec for wired, UMTS, and WiMAX networks using OPNET Modeler simulation tool. Based on their obtained results, the G.723 was the best codec to be used in all WiMAX, UMTS, and wired networks. However, the work does not take into consideration LTE and 802.11ac networks to determine the optimal VoIP codecs. The wired networks in presence of the G.711, G.729A, G.723.1, and GSM.AMR codecs are also examined in [3] using NS2.

The influence of the voice codecs on end-to-end voice over LTE (VoLTE) is highlighted in [4]. The OPNET Modeler is used in their work to evaluate the GSM-EFR, AMR12.2K, IS641, G.711, and G.729A codecs in terms of the metrics including MOS, packet delay, voice traffic sent-received, and voice packet delay variation. The work concludes that a better performance can be reached by the G.711 and GSM EFR codecs. The hybrid topology as a combination of both LTE and 802.11ac and also other types of codecs are not examined by the authors. The OPNET Modeler tool is also used to investigate the performance of LTE networks in presence of the G.729 codec in [5], G.711 in [6], and G.711 and G.723 in [7].

The WLAN performance using OPNET Modeler in presence of the G.711, G.723.1, G.729A, G.728, G.726, and GSM-AMR codecs is studied in [8]. The OPNET throughput and delay results show that GSM-AMR codec will give the best-effort quality of service for the VoIP over WLAN. However, the work does not clarify the type of WLAN being simulated while LTE also is not examined. The WLAN is also examined in [9] with the G.711, G.723, and G.729 codecs in a testbed environment to evaluate the energy consumption of the codecs. The MOS parameter in LTE is examined in [10] using OPNET Modeler in presence of the GSM-FR, G.711, G.723, and G.729 codecs. The results show that the G.711 codec gives the higher MOS values. Using the EXata/Cyber simulator tool, the performance of AODV, DSR, DYMO and OLSR routing protocols in WLAN with G.711, G.723, and G.729 codecs is also provided in [11].

Based on the existing studies, the previous works mainly examine the performance of the codecs in homogenous networks while there is also no current work regarding the VoIP codecs performance in the 802.11ac networks. On the other hand, due to the existence of multiple types of network access links such as Wi-Fi or LTE, the users have the option to switch to whichever access links that provides better voice QoS. However, there is no current work on comparing LTE and 802.11ac in a hybrid heterogeneous environment and this area is not yet known. Considering the limitations of the previous studies, this work is a preliminary attempt to provide a comprehensive comparative analysis of the common VoIP codecs over a hybrid LTE-802.11ac network. In contrast with the existing works, the main motive here is to precisely determine the type of VoIP codecs that can efficiently provide higher voice quality in LTE and 802.11ac networks. Providing a deep and accurate analysis of the results will aid the network developers and providers in their design and development of the efficient VoIP applications and services.

3. Simulation Settings Description

In order to achieve the three main contributions of this work, a hybrid network consists of LTE and 802.11ac is simulated in which a variety of scenarios are developed and the corresponding results are measured. This section describes the details of the hybrid simulation topology using the NS3 tool along with implementation of the developed scenarios.

3.1. Hybrid simulation setup

In this work, twelve hybrid mobile user equipment (UE) are designed. In order to be able to work as a hybrid UE, each UE is equipped with two different network interface cards (NICs), one for supporting LTE and the other one for supporting 802.11ac networks. For the 802.11ac NIC, the WIFI_PHY_STANDARD_80211ac and StaWifiMac are added to the UEs as their PHY and MAC layer. This results in the UE to work in VHT mode. Furthermore, to create the LTE NICs, the InstallUeDevice is used which makes the UEs to work in LTE mode. After this process, the UEs are capable of connecting to either LTE or 802.11ac networks, whichever has a higher signal to noise ratio (SNR). To do this, a Boolean switch parameter is created which is called wifilteswitch with true value for switching to LTE network and false value for switching to the 802.11ac network based on the received SNR. In case of switching to LTE network (wifilteswitch is true), the LTE core is activated and each hybrid UE is connected to the packet data network (PDN) through the eNodeB which in turn is connected to the PDN gateway (PGW). In case of switching to the 802.11ac network (wifilteswitch is false), the 802.11ac access point is activated and each hybrid UE is connected to the PDN through an 802.11ac access point. There also exists a VoIP server that is capable of managing the voice calls with different types of codecs between the UEs. The designed hybrid topology can be found in Fig. 1.

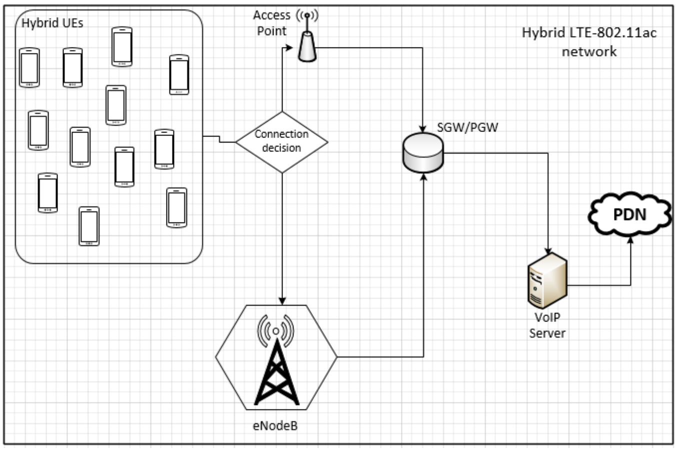

Fig. 1. Simulation topology of the hybrid LTE-802.11ac network.

3.2. Simulation scenarios and design parameters

In this work, 48 distinct scenarios are set up in order to examine various aspects of LTE and 802.11ac in regard to the VoIP codecs. In order to provide reliable and fair conditions for comparison of LTE and 802.11ac, the modulation algorithm, coding rate, and channel width are considered the same as 64QAM [12], 5/6, and 20 MHz respectively for both networks. The number of hybrid users in both networks is 12 and the simulation time is set as 8 seconds.

From the category of the waveform codecs, the G.711 (PCM) and G.726 (ADPCM) are analysed and from the category of the hybrid codecs, the G.723 (ACELP), G.728 (LD-CELP), and G.729 (CS-ACELP) are analysed. To implement each codec, the corresponding codec data rate and packet size are modelled while 12 bytes RTP header size is also added to each voice packet before transmission.

During VoIP call between the users, it is assumed that each conversation goes on continuously with no silence in between. In order to evaluate the performance of each VoIP codec, four QoS performance metrics are measured including the throughput, delay, jitter, and packet loss ratio. The measurements are done in per-flow basis. Thus, having 12 VoIP end-users provides 12 individual flows per second, which will be 96 individual flows per entire 8 seconds simulation time. Some of the simulation parameters specific to LTE and 802.11ac networks in the hybrid topology along with the common simulation parameters are presented 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

4. Performance Evaluation

This section presents the results obtained from implementation of the 48 experiments in the designed hybrid topology. The corresponding performance comparative analysis to highlight the voice quality in each certain scenario is also provided.

4.1. G.711 (PCM) codec

This scenario aims at evaluating the operation and functionality of the G.711 codec to determine its constraints or benefits in LTE and 802.11ac networks. The simulation results of this experiment in terms of throughput, loss ratio, delay, and jitter are provided in Fig. 2.

Based on the obtained results, almost the same average throughput is achieved for both LTE and 802.11ac using the G.711 codec. While there is no remarkable difference in the amount of average throughput, the two networks show completely different behaviour at the beginning of the simulation run. The LTE starts with high throughput while gradually the throughput drops and then remains constant to the end. In contrast, 802.11ac starts with lower throughput while gradually increases until it remains constant. In term of loss ratio, LTE shows much better performance as compared to 802.11ac. The average loss ratio in LTE is 0.0088 compared to 0.0102 in 802.11ac, which is much higher and causes the VoIP call gets degraded.

Furthermore, the latency-related results show interesting differences between LTE and 802.11ac. In LTE, the delay varies as the time passes whereas the values follow the same pattern. Meanwhile, the delay in 802.11ac is higher and constant (0.03s) during the entire simulation time with no considerable variations. This consequently provides zero jitter for 802.11ac unlike LTE with higher jitter, which can affect the quality of the received VoIP packets and results in poor or scrambled voice communication.

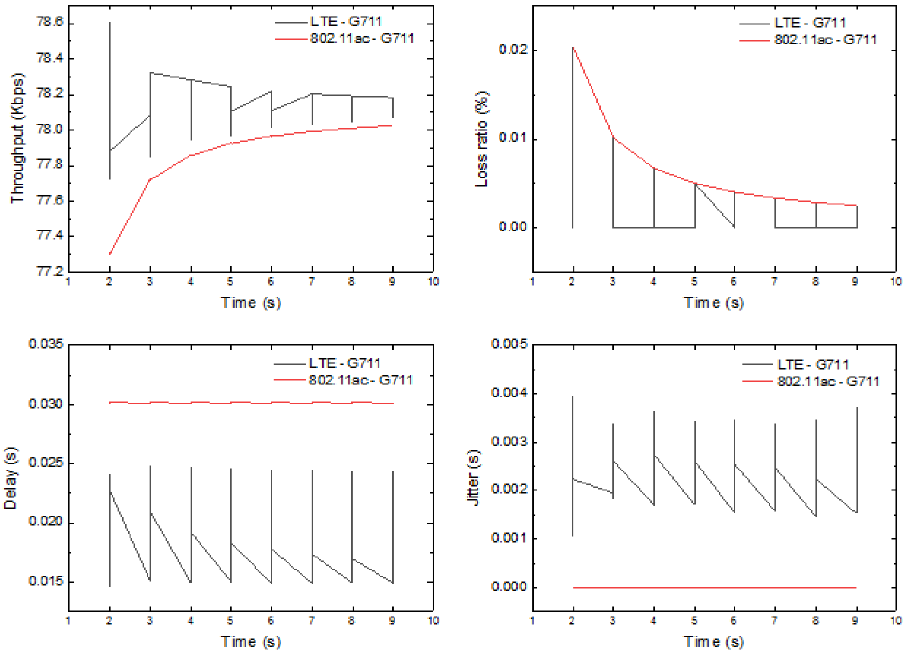

Fig. 2. G.711 VoIP codec performance.

4.2. G.723 (ACELP) codec

To determine the level of QoS achieved by the end-users applying the G723 codec in LTE and 802.11ac, the previous experiment is replicated under conditions in which the G.723 codec is utilized in the entire VoIP communications. The main objective is to assess the features available in the G.723 codec affecting the voice performance and quantify the corresponding impacts. The QoS-related results are provided in Fig. 3.

These outcomes indicate that the G.723 codec performs better in LTE compared to 802.11ac. The LTE throughput has its highest value at the beginning of the time while it gradually declines and remains steady until the end of simulation time. On the other hand, there is a constant throughput (15.55Kbps) during the entire time for 802.11ac with the values near to when LTE remains in its steady state. The results further imply that in some points the loss ratio in LTE reaches zero while for the rest of the time the ratio is equal to the values in 802.11ac. 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 exact same value as before (0.03s) while delay in LTE is higher with variable values during the entire simulation time. As the constant delay causes zero jitter, 802.11ac comes up with zero jitter compared to high jitter in LTE. In the previous experiment using the G.711 codec, the average jitter was 0.0024s while in this experiment using the G.723, the jitter goes higher to about 0.0031s, which negatively affects the QoS of the voice packets more than before and causes noticeable voice degradation.

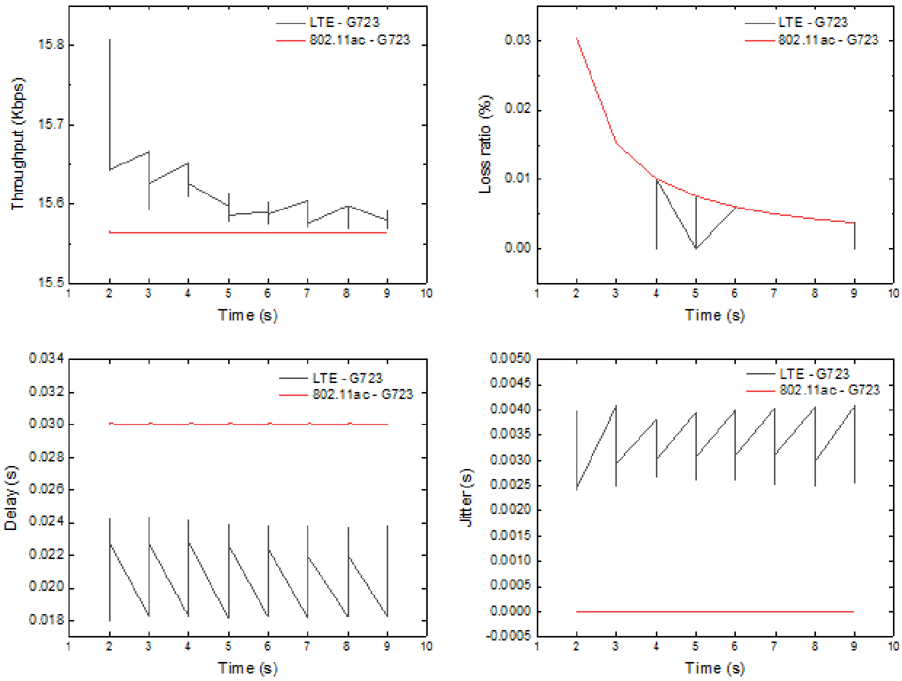

Fig. 3. G.723 VoIP codec performance.

4.3. G.726 (ADPCM) codec

In order to identify the end-to-end performance of the G.726 codec, this scenario models the practical differences of the G.726 to be compared with the results of the above experiments. The results are provided in Fig. 4.

The throughput results from implementation of the G.726 codec reveal that despite having different per-flow throughput pattern during the simulation time, the average throughput is the same for both LTE and 802.11ac. In regard to the loss ratio, the results show higher rate in 802.11ac as compared with LTE. This implies that 802.11ac is more vulnerable to missing the audio information at the destination than LTE. Moreover, we do not observe any noticeable delay difference as compared with the previous experiments. As before, the delay in 802.11ac is constant and also higher than LTE while using the G.726 codec results in a bit higher delay in LTE when comparing with the G.711 or G.723 VoIP codecs. Due to the constant delay, jitter in 802.11ac is zero while in LTE the average value reaches about 0.0021s which is lower than jitter in either the G.711 or G.723 VoIP codecs.

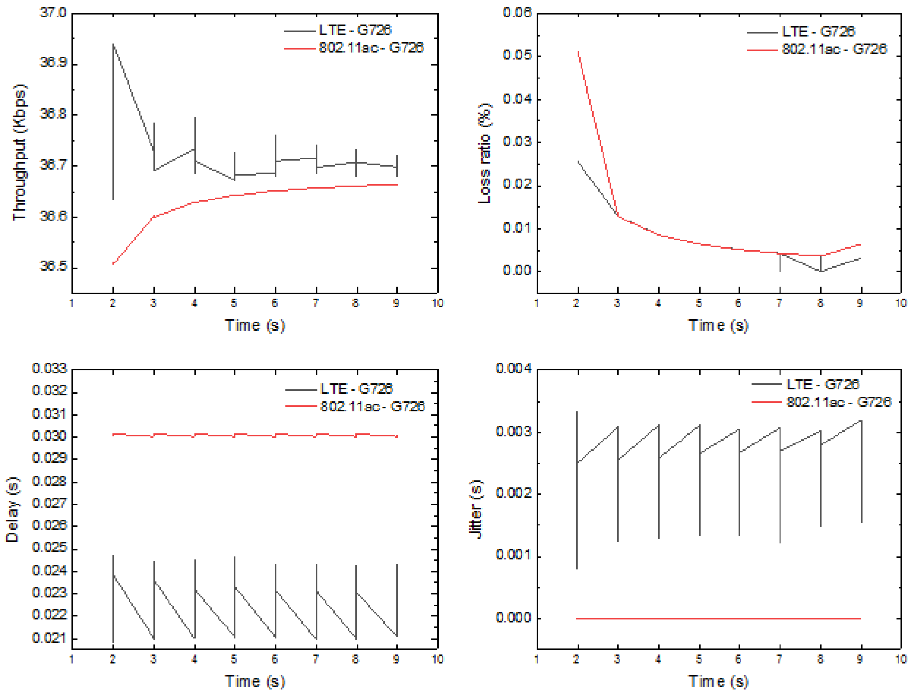

Fig. 4. G.726 VoIP codec performance.

4.4. G.728 (LD-CELP) codec

This scenario is carried out in an attempt to characterize the performance of the G.728 VoIP codec utilized in LTE and 802.11ac. The results are provided in Fig. 5.

Evaluating the throughput results shows low voice throughput for both LTE and 802.11ac. The interesting finding is the constant throughput in 802.11ac while LTE follows the same pattern as before, which is higher at the beginning and then reaches a steady state after a while. The loss ratio in 802.11ac involving the G.728 codec is much higher than LTE. The measurement values show the ratio is about 0.009 in LTE while it is 0.012 in 802.11ac. Based on analysing the delay measurements in the three previous experiments, we observed the same consistent pattern and close values. Therefore, we were curious about the delay pattern in this experiment whether the pattern will be repeated or not. Measuring the delay proves the same latency-related values for the G.728 codec as well. The delay is constant (0.03s) for 802.11ac while in LTE it varies with the same up and down pattern as the time passes. Like before, the constant delay causes zero jitter in 802.11ac but we observe higher jitter (0.003s) in LTE.

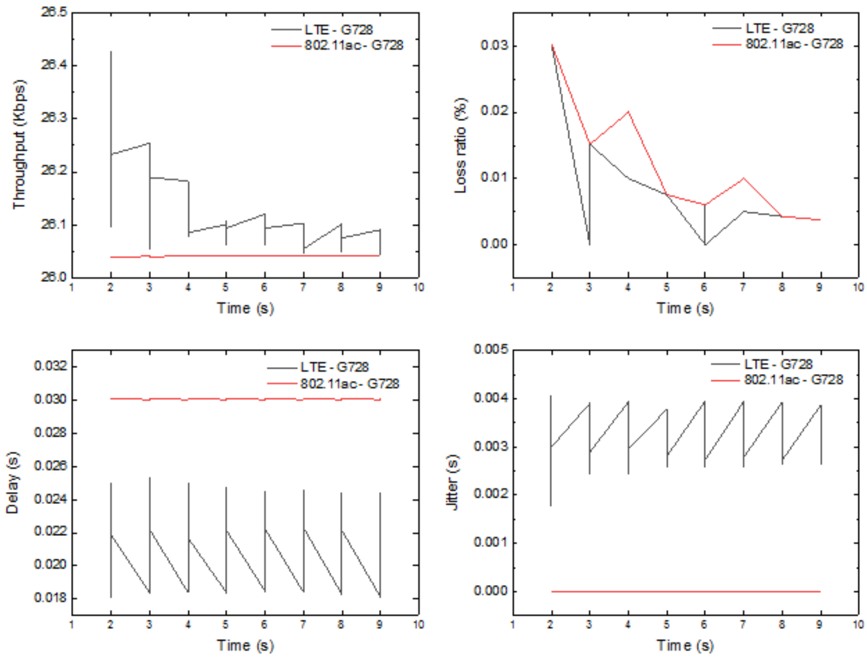

Fig. 5. G.728 VoIP codec performance.

4.5. G.729 (CS-ACELP) codec

The design of this scenario is based on measuring LTE and 802.11ac operation performance in presence of the G.729 VoIP codec. The results are provided in Fig. 6.

Comparing the results of this experiment with the previous experiments points out that the highest throughput achievement is provided by the G.729 codec in both LTE and 802.11ac networks. The average throughput for LTE is about 234.88Kbps compared to 233.58Kbps in 802.11ac, which is the highest throughput so far in our experiments. The loss ratio also reaches about 0.005 and 0.01 for LTE and 802.11ac respectively. These ratio values imply that using the G.729 codec, 802.11ac network is much more prone to loss voice packets, which results in lower VoIP communication than LTE. In term of delay, we observe the same latency pattern like our previous experiments while LTE in this experiment can achieve the least amount of delay (0.016s) among them. The jitter in 802.11ac as before is zero while this time LTE presents almost a constant value (0.0025s) for the jitter.

4.6. iLBC codec

The performance assessment of the iLBC VoIP codec is presented in this experiment to be compared with the other codecs in previous experiments. The results are provided in Fig. 7.

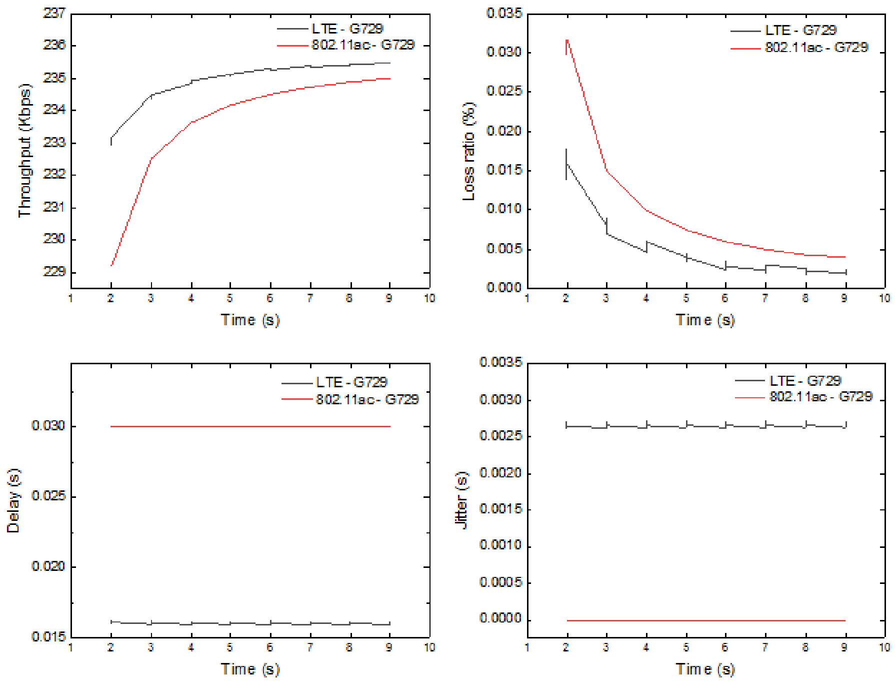

Fig. 6. G.729 VoIP codec performance.

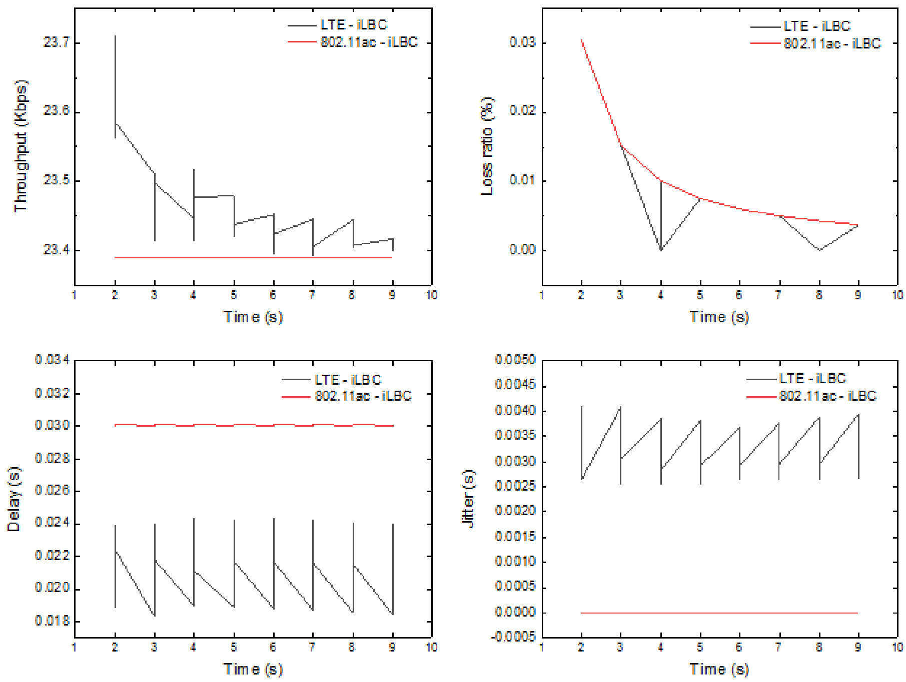

Fig. 7. iLBC VoIP codec performance.

The last codec to be examined is the iLBC. The obtained results confirm the consistency with the other experiments. In order to complete our work, a summary of the measurement results from implementation of all the experiments is provided in Fig. 8, which implies that the QoS can be improved by the G.729 codec for both LTE and 802.11ac networks in a hybrid environment.

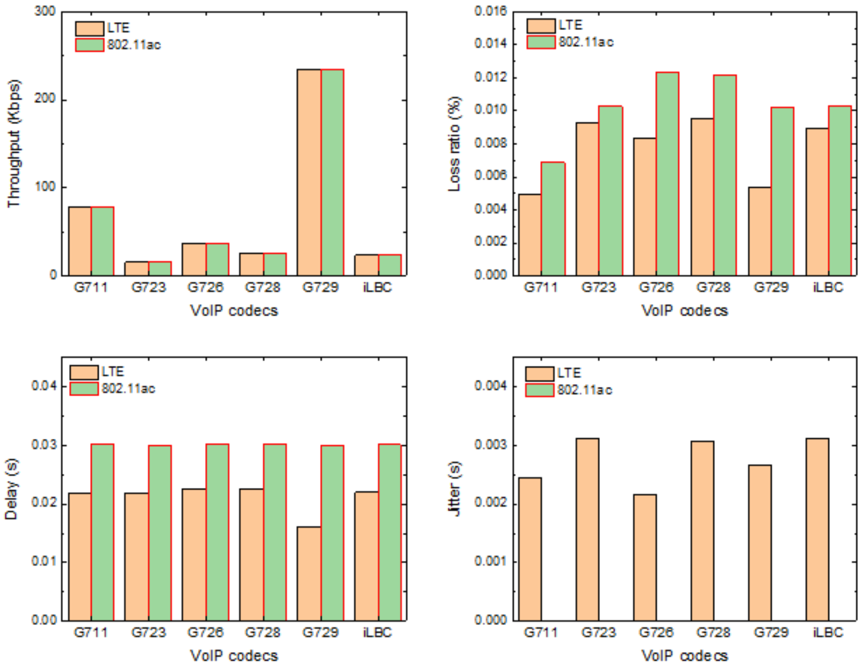

Fig. 8. Performance comparison of all the VoIP codecs.

5. Conclusions

This work developed a method to apply in a hybrid LTE-802.11ac network. Various scenarios based on the most common VoIP codecs were designed and implemented and four QoS-related performance metrics were measured. The results of the experiments highlighted two findings regarding the throughput measurements.

First, there is no substantial throughput differences between LTE and 802.11ac as the values for the same codecs are closely related. Second, regardless of being an LTE user or 802.11ac user, the best performance can be achieved using the G.729 VoIP codec while the weakest performance was measured when the G.723 VoIP codec was applied in the network.

Furthermore, the results of the experiments revealed that the rate of the loss packets was always higher for 802.11ac end-users than LTE end-users. The careful analysis of the delay results confirms no significant differences between different codecs in 802.11ac network. Regardless of the type of the codecs, the end-users in 802.11ac network always experience the same amount of delay during the voice conversation, which also is always higher than the delay for LTE end-users.

Additionally, the lowest value can be obtained when the G.729 codec is used by LTE end-users. The constant delay provided by all the codecs for the 802.11ac users results in zero jitter in contrast with the higher jitter experienced by LTE end-users.

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