Voice Quality Testing Using VoIP Applications over 4G Mobile Networks

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Abstract. Nowadays, many mobile applications in the telecommunication market provide Voice over IP (VoIP), video and data services. This paper presents the performance evaluation of a 4G mobile network in the city of Pristina. This was accomplished by testing two of the most popular VoIP applications, Skype and Viber. Stationary and dynamic scenarios have been considered to evaluate the voice quality and the Quality of Experience (QoE) of both applications under the same mobile network radio conditions, throughput capacity and coverage requirements. The Mean Opinion Score (MOS) method was used to evaluate the recorded speech files based on the feedback perceptions of different users. These results were found to be very useful for the continuously increasing demand on VoIP services and applications.

1 Introduction

Over the past decade subsequent expansions of the Global System for Mobile Communications (GSM) system were created. Data rates of 14 Mbps were achieved by using High Speed Packet Access (HSPA), increasing to a potential data rate of 42 Mbps for evolved HSPA+ standards if advanced Quadrature Amplitude Modulation (QAM) schemes are combined with Multiple Input Multiple Output (MIMO) techniques. Long Term Evolution (LTE) network was initiated as a project in 2004 and describes the standardization work by the Third Generation Partnership Project (3GPP) to define a new high-speed radio access method for mobile communication systems. The main radio access design parameters of this new system include Orthogonal Frequency Division Multiplexing (OFDM) waveforms to avoid the inter symbol interference that typically limits the performance of high-speed systems, and MIMO techniques to boost the data rates. Data rates are about to feature a range of 100 Mbps for high mobility scenarios up to 1 Gbps for low mobility scenarios using advanced LTE (LTE-A) networks [1]. In general, the LTE system has already become a world-wide standard. In western Balkans there are more than 10 million active users, introducing an economic growth and investments for the developing regions. Furthermore, it has created an adequate environment for mobile application services along with internet services such as VoIP, web browsing, and video streaming, with constraints on delays and bandwidth requirements [2]. An

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efficient listen before talk (LBT) technique is analyzed in [3], to reduce interference and improve spectral efficiency of LTE network users.

To evaluate 4G mobile networks, it is generally necessary to investigate on different network radio conditions the Quality of Service (QoS) for voice, video and data traffic, as expected to be received by the users. VoIP applications present an interesting option as they combine voice and data communications and can be used to evaluate the performance of the mobile network in terms of QoS [4]. The mobile technology is evolving towards the 5G standard, which is expected to start its service in 2020. In LTE-A, the network delay budget for VoIP is about 100 ms [5] and the key challenge of 5G technology will be to reduce the delay less than 1 ms [6].

In this work, two of the most popular social mobile applications, such as Skype and Viber, have been considered and tested over the LTE mobile network operating within the city of Pristina. Skype is a telecommunication application introduced for the first time in 2003, mainly designed for personal computers (PC) as a tool for business. Skype was considered the best PC application on providing voice and video call services however it was limited due to mobility. To fit the market requirements, different Skype mobile device applications were developed under the trademark of Microsoft Corporation. Viber application was first lunched in 2010, initially for iPhone and later for Android platforms, as an alternative to Skype. It provides voice, SMS and video call services for mobile devices and recently available for PC, tablets and other electronic devices. Both applications are commonly used in the developing Balkan regions, where users are able to call their friends account without being charged or conveniently charged for landline or mobile number calls.

Stationary and dynamic experimental investigations over LTE mobile network have been considered using VoIP applications and MOS method was used to evaluate and compare the voice quality of both Skype and Viber applications. The MOS test has been used for decades in telecommunication networks to present the human's perceptions for the voice quality [7, 8]. The results of this experimental work were found to be interesting for the network operators in order to improve the LTE mobile network performance and for the mobile application users, always demanding for better voice quality with convenient costs.

2 Analysis of the Experimental System and Testing Scenarios

LTE technology has experienced some major changes of the network topology compared with the former 3G networks, as shown in Fig. 1. Node-B of 3G system was replaced by evolved Node-B (eNB), which is a combination of Node-B and radio network controller (RNC). The eNB communicates with the User Equipment (UE) and can serve one or several cells at one time. The serving gateway (SGW), as a part of the evolved packet core (EPC), is responsible for routing and forwarding packets between UE and packet data network (PDN) and charging. The mobility management entity (MME) manages UE access and mobility and establishes the bearer path for UE. Packet data network gateway (PGW) is a gateway to the PDN, and policy and charging rules function (PCRF) manages policy and charging rules [9].

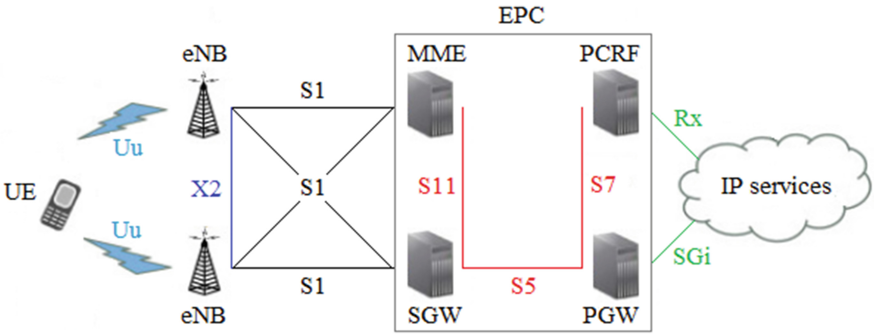

Fig. 1. LTE network topology.

Measurements of the required throughput for Skype and Viber applications were performed in the city of Pristina and the Test Mobile System (TEMS) discovery network equipment was used for data processing. The experiments are designed to conduct calls from a Samsung S5 as the calling side to the laptop's USB dongle 4G adapter in receiving mode, using Skype and Viber applications over a LTE network, as shown in Fig. 2.

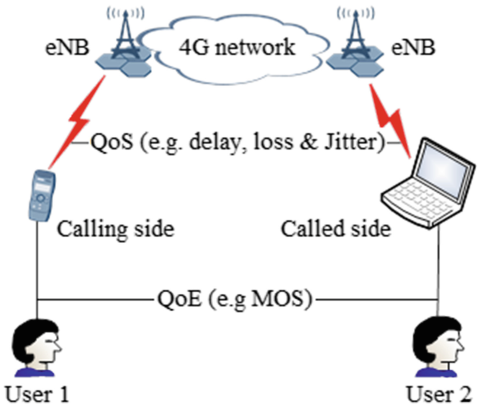

Fig. 2. QoS and QoE in LTE network.

Two different static test scenarios were considered during the measurement. The calling UE was first set at −105 dBm of signal strength, far from the antenna of the serving cell, and then it was set at −60 dBm of signal strength, near the antenna of the serving cell. Furthermore, high mobility calling tests were considered, with the calling UE moving with a constant speed of 30 km/h from a point having −120 dBm of signal strength to a point having −60 dBm of signal strength. The focus was to test the voice quality of the recorded calls under different network radio conditions by using the MOS method [7], which is based on a large number of users' perceptions. The better the QoS provided from the network operator, in terms of packet loss, jitter, delay and data rate, the better the QoE will be expected as a result, especially for real time applications such as VoIP.

Beside the required throughput for the VoIP applications, measurements of the Reference Signal Received Power (RSRP) parameter from the UE were reported to the core network to make decisions about cell selections or handovers. RSRP can be expressed in terms of the number of resource blocks (N), and the Received Signal Strength Indicator (RSSI), measured for a specific bandwidth as described in Eq. (1). RSRP typically levels in the range from −75 dBm, near the LTE's serving cell, to −120 dBm, far from the serving cell, and informs the system for the quality of signal in terms of Reference Signal Received Quality (RSRQ) [10]:

$$\mathrm{RSRP}\,(\mathrm{dBm}) = \mathrm{RSSI}\,(\mathrm{dBm}) + 10 * \log\,(12 * \mathrm{N}) \tag{1}$$

The signal quality is also indicated from the measured Signal to Interference and Noise Ratio (SINR), which can be expressed as the ratio of the measured power of the useful signal to the sum of the neighbor cell's interference and the background noise over the considered bandwidth.

3 Results and Discussion

Measurement results of the considered scenarios have been reported in this section. Figure 3 presents the probability distribution function (PDF) for the Physical Downlink Shared Channel (PDSCH) and the Physical Uplink Shared Channel (PUSCH) throughput, required by a static UE to conduct the VoIP calls in Skype, as in Fig. 3(a), (c) and Viber as in Fig. 3(b), (d). The results of throughput distribution in the case where the UE is distant from the serving antenna, at −105 dBm of signal strength, are reported

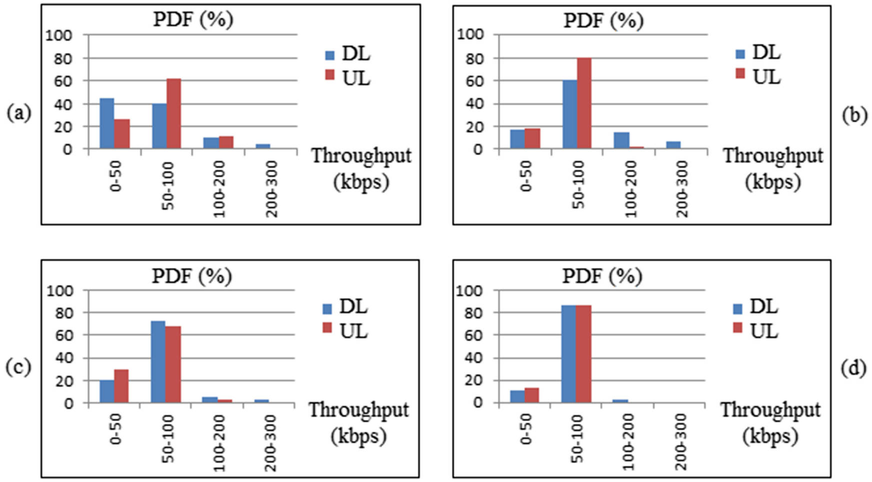

Fig. 3. Downlink/Uplink (DL/UL) throughput distribution for Skype and Viber VoIP applications.

in Fig. 3(a), (b) and the results for the case where the calling UE is at −60 dBm of signal strength are reported in Fig. 3(c), (d). In order to minimize the network traffic effect, measurements were conducted on the same time of the day, around the peak evening hours.

The modulation scheme used in both cases was the Phase Shift Keying (QPSK) and the required throughput for about 90% of the samples does not exceed the recommended value of 100 kbps. The better the radio conditions were the better data rates were achieved. The results show the minimum bandwidth that the VoIP applications need compared with the LTE cell capacity, especially when using Skype, which requires 30 kbps for about 40% of the samples.

The VoIP applications have also been tested under UE mobility conditions, as shown in Figs. 4 and 5, using Skype and Viber respectively. The measurements of the network parameters were conducted from a point of the cell having −120 dBm of signal strength, to a point of the cell having −60 dBm of signal strength.

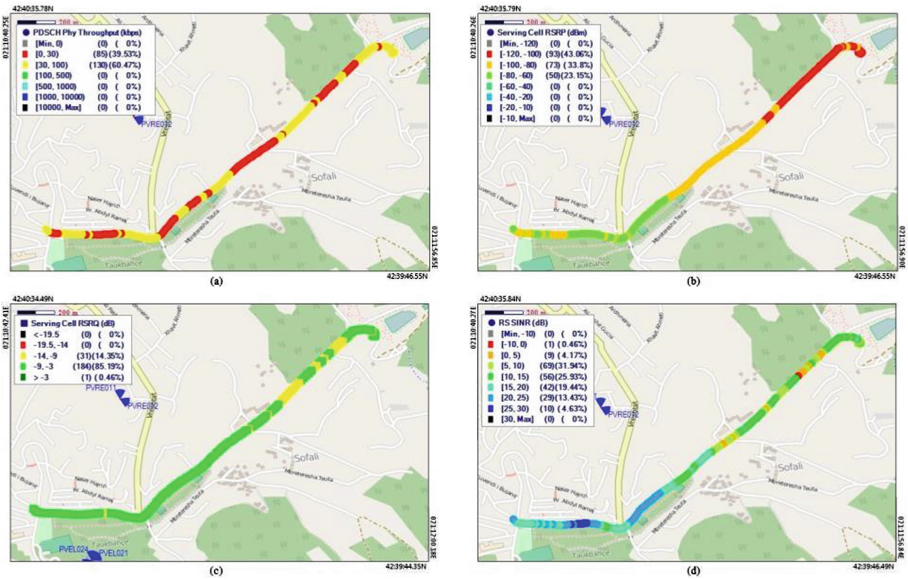

Fig. 4. Calling UE mobility test using Skype VoIP application.

Figure 4(a) shows that there was different throughput utilization during the considered scenario however it didn't exceed 100 kbps. The measured RSRP far from the serving cell, as reported in Fig. 4(b), commands the core network to make cell revelation or handovers. Based on the results of Fig. 4(c), the quality of signal has affected the quality of the recorded voice causing dropped calls and interference during the conversation on the reception side. It is obvious that the degradation of radio conditions directly affects the quality of VoIP calls in Skype. Figure 4(d) presents the SINR and shows that the cells are well optimized with acceptable noise and interference, except of a segment of the path where the SINR is between −10 dB and 0 dB.

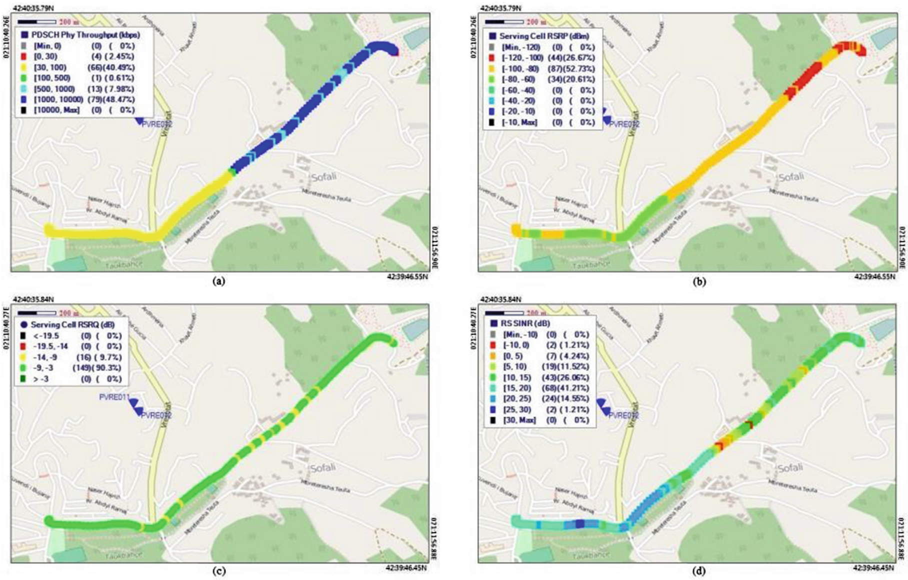

Fig. 5. Calling UE mobility test using Viber VoIP application.

Figure 5(a), (b), (c) relieves that Viber needs more than 1Mbps of throughput as the UE moves far from the serving cell, and the RSRP signs between −100 dBm to −80 dBm. From the measurement results we can observe that Skype is more rational than Viber in terms of the required bandwidth. This is mainly because Viber introduces more signaling and headers in order to keep the better signal quality referred to Skype application and as a consequence requires more bandwidth. The SINR measured for the VoIP Viber application, as shown in Fig. 5(d), is similar to the Skype tests, where an acceptable noise and interference was introduced.

So far, we have discussed the results in terms of QoS parameters, which have presented the ability of the network to provide good services based on the radio conditions quality. In order to compare the considered VoIP applications in terms of QoE the MOS method was used.

In Table 1 the results of the MOS test for each of the considered scenarios are presented. The voice quality of 15 min long recorded calls in Skype and Viber have been evaluated by 60 participants.

Table 1. MOS test results for the considered scenarios.

Scenario	Skype	Viber
Static (−105 dBm)	4.6	4.4
Static (−60 dBm)	3.4	4.6
UE in mobility	2.5	4.1

The MOS test results relive the better quality of signal for the VoIP Viber application. Usually a MOS result of 3.6 is mentioned as a limit for minimum acceptable quality of voice communication services. However, both platforms are excellent choices for the VoIP services.

4 Conclusions and Future Work

In this study the voice quality over a 4G LTE network in terms of QoS and QoE using Skype and Viber VoIP applications has been tested and analyzed. The results of this study were shown to be very useful for the users and operators of the considered 4G network in the city of Pristina. Based on the MOS method results the Viber voice quality seems to be better than Skype, especially for the mobility scenarios. However, in this case the throughput needed for the two applications was drastically different. The Viber VoIP application requires up to 10 Mbps of bandwidth, under the worst case of radio network conditions of −105 dBm of signal strength. That means that Skype is a more rational VoIP application in terms of bandwidth requirements than Viber, which didn't exceed more than 100kbps of throughput for all the considered scenarios. This difference between the considered VoIP applications is mainly because Viber introduces more signaling and headers in order to keep the better signal quality referred to Skype and as a consequence requires more bandwidth.

As a future work, we are looking to test the QoS and QoE of video call services and increase the number of participants in the MOS test, in order to improve the reliability of the voice and video quality perceptions. Comparing voice or video calls quality under different 4G network operators and considering 5G networks prospective would be of a great interest in the future works.

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