Experimental Assessment of Voice Over IP in LTE Systems Under Different Cell Conditions

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Abstract This chapter presents a measurement-based performance assessment of voice over IP in long term evolution (LTE) networks for suburban and rural applications. The behavior of the embedded eNodeB quality of service (QoS) aware scheduler is evaluated when different LTE configurations of QoS and cell conditions are taken into consideration. Our results suggest that the aforementioned scheduler maintains VoIP quality using guaranteed bit rate (GBR) with concurrent traffic over default bearer or even over non-GBR dedicated bearer. The QoS aware algorithm works in such a fashion that the mean opinion score remains constant even in adverse conditions, for instance, under concurrent traffic delay increase conditions, or in the presence of signal reception degradation due to changes in the terminal position within the cell.

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A. Paradisi et al. (eds.), *Long Term Evolution*, Telecommunications and Information Technology, DOI 10.1007/978-3-319-23823-4_8

1 Introduction

The long term evolution (LTE) technology has been a hot research topic in wireless communications due to its central role in the standardization of next-generation cellular systems carried out by third generation partnership project (3GPP). In densely populated areas, LTE systems have mostly been deployed in standard 3GPP frequency bands, such as the 700 MHz and 2.6 GHz bands. In order to benefit from the superior propagation characteristics of lower frequencies in a new LTE profile suitable for sparsely populated areas, 3GPP has recently standardized the 450–470 MHz range as 3GPP Band 31 [1].

Since its Release 8, the LTE standard defines different quality of service (QoS) levels for data, video and voice over internet protocol (VoIP) services [2]. Scheduling mechanisms for VoIP calls in LTE systems have been studied via simulation. In [3], for instance, the capacity derived when dynamic schedulers are employed in multi-cell scenarios (low mobility, 5 MHz operation bandwidth) is shown to depend on the control plane resource allocation scheme. It turns out from this study that up to 50–300 simultaneous calls per cell can be supported by dynamic schedulers depending on the number of symbols available to the physical downlink control channel (PDCCH). In contrast, the VoIP traffic that can be accommodated using semi-persistent mechanisms is shown to be fixed and around 175 calls per cell. All in all, the message conveyed by Puttonen et al. [3] is that one has to cope with some capacity constraints when dynamic scheduling packets are used with persistent scheduling mechanisms.

The aforementioned multi-cell scenario is extended in [4] for different operation bandwidths and inter-cell distances. The results shown present the absolute VoIP capacity numbers of the LTE downlink. It is also shown that link adaptation together with packet bundling provides a clear gain in capacity, as more VoIP packets can be scheduled in each transmission time interval (TTI). Limitations in the control channel can also be effectively compensated by packet bundling. End-to-end delays and throughput in the presence of VoIP traffic are assessed in [5] for several scenarios considering stationary and mobile terminals. In the absence of mobility, it is shown via simulation work that the delays are slightly higher for networks congested with VoIP only. In other cases, better performance is derived due to the presence of moving terminals.

Notwithstanding the value of the related work toward a better understanding of VoIP using LTE infrastructure, their results primarily rely on simulation work and exploit various parameters in a multiple cell environment, leading to experiments almost impossible to be performed with real equipment. One alternative to complement such purely theoretical modeling and analysis is practical implementation, as it can be employed to validate the design of algorithms, protocols, software, and hardware under a genuine radio frequency (RF) environment. Motivated by the several ways in which measurement-based experimentation may be rewarding, the performance of VoIP calls on a real LTE system was evaluated in a previous work of the authors [6]. Among the conclusions drawn therein, it was found that the scheduler provides good quality of VoIP service when the QoS class identifier (QCI) type guaranteed bit rate (GBR) is used. In this chapter, we conduct a measured-based performance assessment of VoIP in LTE systems. Our analysis considers different positions of the terminal inside the cell, i.e., different signal reception conditions for a fixed number of VoIP calls. Concurrent traffic with congestion is also taken into account. These settings differ from those used in [6] in that there the number of calls is allowed to vary but terminal positions are kept fixed, and the concurrent traffic considered is now with and without congestion.

Our results suggest that, among the schedulers available at the eNodeB, only the one with QoS aware weighted priority (WP) and GBR maintains call quality under congestion operation conditions. It is worth mentioning at this point that one distinguishing aspect between VoIP and voice over LTE (VoLTE) is that the latter employs an IP multimedia subsystem (IMS) for call control [7]. VoLTE also calls for eNodeB support for semi-persistent scheduling, TTI bundling, header compression, and the terminal needs to support the AMR-WB codec [8]. The tests presented herein focus on the LTE QoS behavior using the G711u codec encapsulated within a real-time transport protocol (RTP) data packet under data traffic congestion [9]. The data packets are generated using a tool that simulates the G711u RTP payload and simply creates VoIP traffic between two personal computer (PC) endpoints.

The remainder of the chapter is organized as follows. Section 2 overviews the basics of QoS provisioning in LTE networks. Scheduler algorithm and the figures of merit used to evaluate voice quality in our experiments are also described there. We carry on in Sect. 3 with a description of the experimental setup based upon which three test scenarios are considered. Results for these scenarios are then discussed in Sect. 4. Section 5 wraps up the chapter with some remarks and future works.

2 Fundamentals of QoS in LTE Networks

This section describes briefly the fundamentals of QoS in LTE network. Section 2.1 explains about the service bearers used to establish a QoS connection between the devices that communicates in a LTE network. Section 2.2 introduces the QoS parameters and their properties used to define the bearer characteristics for the VoIP data packets' transport and Sect. 2.3 explains a resumed overview of the scheduler algorithm used to optimize the QoS within the network to allow the best VoIP data flow.

The LTE standard has been designed bearing in mind the need to provide appropriate capabilities for different service categories, such as voice, video, messaging, and exchange of data files. This is possible thanks to an implementation of QoS control supported by architectures defined by 3GPP [2, 10]. In evolved packet systems (EPS), QoS control takes place at the level of the service flows, referred to as bearer services in the context of LTE. The traffic mapped onto a specific bearer service is treated equally with respect to packet forwarding, including the application of policies for scheduling and queue management. Different service flows are available to make the implementation of different packet forwarding treatments possible, namely default bearers and dedicated bearers. Default and dedicated bearers are briefly described in the sequel along with QoS parameters, the scheduling algorithm implemented at the eNodeB, and the metrics used for evaluating voice quality later on in the performance characterization of our experiments.

2.1 Service Bearers

The default service bearer is established whenever the user equipment (UE) connects to the packet data network (PDN). This bearer possesses the basic QoS capacities. For each PDN that the UE connects to, a default bearer is established. Any additional bearer established between the UE and the same PDN connection resulting by demanded services with specific QoS requirements is called a dedicated bearer. In general, a bearer is characterized in terms of the bit rate guarantees it provides. A bearer is then said with GBR when the network resources related to the associated GBR value are permanently allocated in the constitution or the modification of the service flow. Likewise, a bearer characterized as without GBR, i.e., non-GBR, provides no guarantee that a given bit rate is supported for that service flow. In either case, a bearer is associated with a set of IP packet filters that control the user traffic carrying the user service for a specific bearer. Filtering mechanism in the uplink traffic flow template (UL-TFT) and downlink traffic flow template (DL-TFT) is performed according to the pictorial description given in Fig. 1.

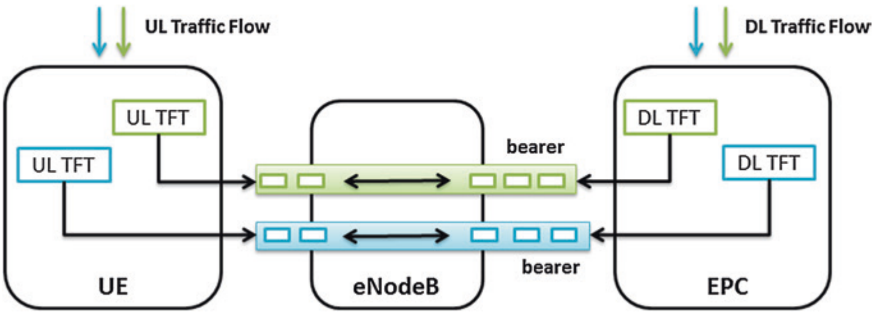


Fig. 1 EPS and TFTs bearer architecture

Table 1 QCI standard characteristics

QCI	Resource type	Priority level	PDB (ms)	PELR	Service samples
1	GBR	2	100	10^{-2}	Conversational voice
2	GBR	4	150	10^{-3}	Live streaming video
3	GBR	3	50	10^{-3}	Real-time games
4	GBR	5	300	10^{-6}	Buffered streaming video
5	non-GBR	1	100	10^{-6}	IMS signaling
6	non-GBR	6	300	10^{-6}	TCP-based services
7	non-GBR	7	100	10^{-3}	Video, voice, and games
8	non-GBR	8	300	10^{-6}	Buffered streaming video
9	non-GBR	9	300	10^{-6}	TCP-based services

2.2 QoS Parameters

The QoS bearer profile includes QCI, GBR, allocation and retention priority (ARP), and maximum bit rate (MBR). For non-GBR bearers, QCI and ARP are the sole parameters specified. For GBR bearers, QCI, ARP, GBR, and MBR parameters are all specified. In this chapter, we consider the QoS parameters that influence the performance analysis of voice calls in 450 MHz LTE network only. The QCI parameter defines the QoS class associated with a bearer, and is used as a reference to specific parameters that control packet forwarding treatment in the bearer, namely packet delay budget (PDB) and packet error loss rate (PELR). The nine QCI values mandated by 3GPP are summarized in Table 1. Voice service flow, video, messaging, and file transfer have different characteristics and are associated with different QCIs. The GBR parameter identifies the bit rate that should be guaranteed for a given GBR bearer, while the MBR parameter sets the MBR allowed for that bearer.

2.3 Scheduler Algorithm

As with the identifiers assigned to dedicated bearers, which ensure the necessary traffic for voice packets in terms of resource type, priority level, PDB, and PELR, the scheduler role is also of utmost importance to ensure the correct scheduling of multiple packets traversing the eNodeB. Here, the QoS aware approach implemented in the eNodeB is based on the WP algorithm [11]. This proprietary algorithm basically computes the WP of each bearer created in accordance with modulation aspects, class of service, delay, and traffic prioritization. Once priorities are assigned to the bearers, all traffic flows are served with resources allocated according to the corresponding bearers, with the priority order set up to the limit of exhausted resources.

One alternative for measuring the resulting voice quality of a scheduler is mean opinion score (MOS), a subjective method defined by the ITU-T P.800 standard

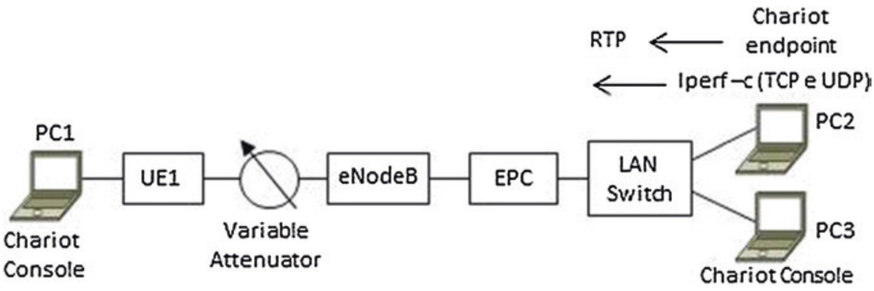


Fig. 2 Setup used in our experiments

Table 2 MOS scores obtained using the E-model algorithm

Inferior limit	Degree of user satisfaction
4,34	Very satisfied
4,03	Satisfied
3,60	Some users unsatisfied
3,10	Many users unsatisfied
2,58	Almost all users unsatisfied

[12]. The transmitted and received voice quality are evaluated according to an algorithm that gives scores on a scale of 1–5, where 1 and 5 correspond to "bad" and "excellent", respectively. The tool used in the VoIP service performance test between PC1 and PC2 (shown in Fig. 2, which will be explained in the next section) uses the E-Model algorithm, defined by the ITU-T G.107 standard [13] as a means to estimate the MOS quality for every pair of endpoints. Table 2 lists the MOS values obtained from the execution of this algorithm. In order to verify the behavior of the scheduler algorithm another metric that can be used in addition to MOS is PDB, as it is among the QCI characteristics in Table 1. In our experiments, we compute delay variations relative to the delay measured at the center of the cell using

$$\Delta_{\sigma}(\%) = \frac{\sigma_{\mathrm{pos}} - \sigma_{\mathrm{ref}}}{\sigma_{\mathrm{ref}}}, \tag{1}$$

where the subscript pos is the delay at the position of interest (measured in the middle or on the edge of the cell) and subscript ref is the reference delay (measured in the center of the cell).

3 Experiment Setup and Test Scenarios

This section presents the general experiment setup and test scenarios. Section 3.1 describes the real-world experiment setup considering the LTE system equipment and test tools as well as the configuration of these device parameters. Moreover, Sect. 3.2 describes the test scenarios in order to verify the behavior of VoIP calls considered in this chapter.

Table 3 Reception conditions considered in all test scenarios

Terminal position	Reception intensity	SNR (dB)
In the center of the cell	High	22
In the middle of the cell	Medium	13
On the edge of the cell	Low	5

3.1 General Settings

In order to verify the behavior of VoIP calls in a LTE network, consisting of real-world equipment, we set up the experiment shown in Fig. 2. External interference is kept under control by connecting the eNodeB to the UE through an RF cable with a variable attenuator, which allows us to vary the signal-to-noise ratio (SNR) to mimic changes in the UE position within the cell. We consider the frequency range 450–470 MHz (3GPP Band 31), 5 MHz operation bandwidth, and different modulation schemes using 25 physical resource blocks. The test topology comprises two PC endpoints monitored by the testing tool IxChariot [14]. Such PC endpoints are responsible for sending and receiving dedicated RTP traffic [9]. Termination points PC1 and PC2 relate to UE and evolved packet core (EPC), respectively. The network that connects to the EPC includes PC3, with IxChariot tool controlling the two endpoints and generating reports for different traffic parameter configuration, and MOS and delay variation measurements.

The VoIP traffic performance tests were conducted using test scenarios including LTE network devices with QoS parameters appropriately configured, and with the IxChariot tool configured to generate, measure, and report both dedicated RTP traffic and concurrent traffic based on the transmission control protocol (TCP) or the user datagram protocol (UDP). In all tests the RTP traffic generated is always transported through a dedicated bearer, and the concurrent traffic (either TCP or UDP) flows are transported through a default bearer. Both types of concurrent traffic were generated using Iperf [15]. The QoS control behavior was verified for RTP traffic under the cell reception conditions listed in Table 3. These conditions reflect different signal intensities obtained by reception through RF attenuator adjustments. Finally, the system was evaluated under different eNodeB modulation configurations, such as adaptive modulation and fixed modulation, quadrature amplitude modulation (64-QAM), and quadrature phase shift keying (QPSK) used in the uplink and downlink. All tests were performed with 20 pairs of RTP conversation, corresponding to 20 concurrent voice calls with the G.711 codec configured and generated with the IxChariot tool for rate 1.280 kbps.

3.2 Test Scenarios

Table 4 summarizes the test scenarios considered in this chapter. In all scenarios and cases considered, we assume that default bearers are set with QCI = 9 (non-GBR) and adaptive modulation is in use at the eNodeB unless otherwise stated.

Table 4 Specific settings associated with each test scenario

<table>
<tr><th>Test scenario</th><th>QCI</th><th>Bearer type</th><th>Concurrent traffic</th><th>Modulation scheme</th><th>Figure</th></tr>
<tr><td rowspan="4">I</td><td>1</td><td>Dedicated</td><td rowspan="2">None</td><td rowspan="4">Adaptive</td><td rowspan="4">3a–d</td></tr>
<tr><td>7</td><td>Dedicated</td></tr>
<tr><td>1, 7</td><td>Dedicated</td><td rowspan="2">Without congestion</td></tr>
<tr><td>9</td><td>Default</td></tr>
<tr><td rowspan="4">II</td><td>1</td><td>Dedicated</td><td rowspan="2">Without congestion</td><td rowspan="4">Adaptive</td><td rowspan="4">4a, b</td></tr>
<tr><td>9</td><td>Default</td></tr>
<tr><td>1</td><td>Dedicated</td><td rowspan="2">With congestion</td></tr>
<tr><td>9</td><td>Default</td></tr>
<tr><td rowspan="6">III</td><td>1</td><td>Dedicated</td><td rowspan="6">Without congestion</td><td rowspan="2">Adaptive</td><td rowspan="6">5</td></tr>
<tr><td>9</td><td>Default</td></tr>
<tr><td>1</td><td>Dedicated</td><td rowspan="2">QPSK</td></tr>
<tr><td>9</td><td>Default</td></tr>
<tr><td>1</td><td>Dedicated</td><td rowspan="2">64-QAM</td></tr>
<tr><td>9</td><td>Default</td></tr>
</table>

Scenario I consists of passing a RTP traffic flow through a dedicated bearer with QCI = 1 (type GBR), and a second RTP traffic flow through a dedicated bearer with QCI = 7 (type non-GBR). The experiment is repeated for the self-explanatory cases “no concurrent traffic”, “TCP concurrent traffic without congestion”, and “UDP concurrent traffic without congestion”. In scenario II, the test consists of passing a RTP traffic flow through a dedicated bearer with QCI = 1 (type GBR), and a concurrent TCP traffic flow through a default bearer. The cases considered here are “no concurrent traffic”, “TCP concurrent traffic with congestion”, and “TCP concurrent traffic without congestion”. Scenario III augments scenario II by letting the modulation scheme used at the eNodeB vary among the cases “adaptive modulation”, “QPSK”, and “64-QAM”. For quick referencing, the last column of Table 4 links the test scenarios to the figures providing their corresponding results. These will be discussed later on in Sect. 4.

TCP traffic transmitted through the default bearer under concurrent condition corresponds to the maximum throughput supported by the LTE channel under different cell reception conditions and modulation schemes. This ascertains that the default bearer traffic competes with the dedicated bearer traffic in the event of a congestion condition, thus allowing us to check the system behavior using the QoS aware WP scheduler. In the absence of congestion, the traffic transmitted in the default bearer assumes a value below the maximum throughput supported by the LTE channel, enabling RTP traffic through the dedicated bearer below the congestion threshold.

4 Experimental Results

In this section, we present preliminary results aimed at evaluating the implementation of the WP QoS aware algorithm for maintaining quality of VoIP service as mandated by the 3GPP LTE standard. The experimentally driven assessment discussed in what follows considers the test scenarios in Tables 3 and 4.

4.1 Test Scenario I

MOS and delay variation results obtained for scenario I are shown in Fig. 3. According to Fig. 3a, the MOS decreases as the UE moves from the center to the middle of the cell to then increase as the UE further moves toward the cell

Fig. 3 Test scenario I: MOS and delay variation measured for different QCI settings using RTP traffic in the absence and in the presence of TCP/UDP concurrent traffic without congestion. **a** MOS without concurrent traffic. **b** Delay variation without concurrent traffic. **c** MOS with concurrent traffic (QCI = 1). **d** MOS with concurrent traffic (QCI = 7)

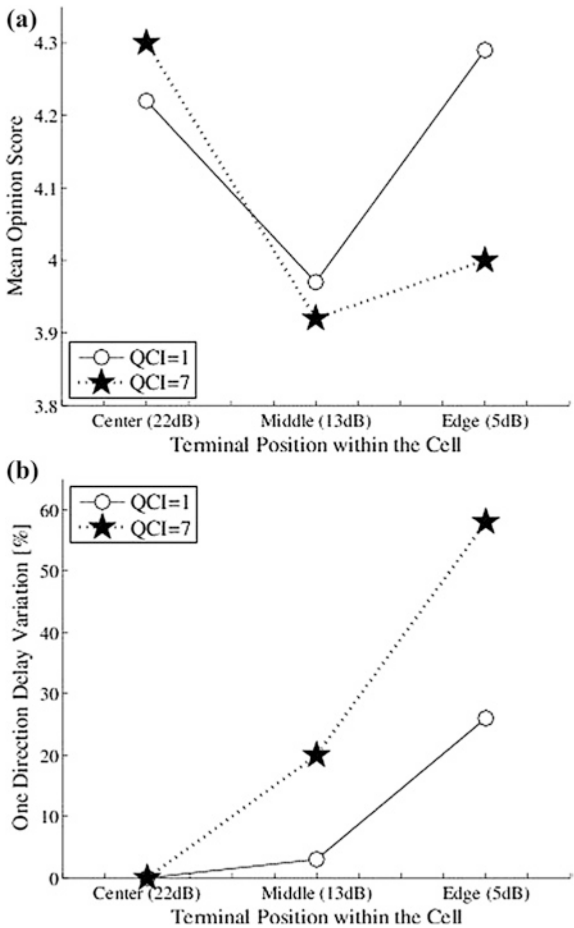
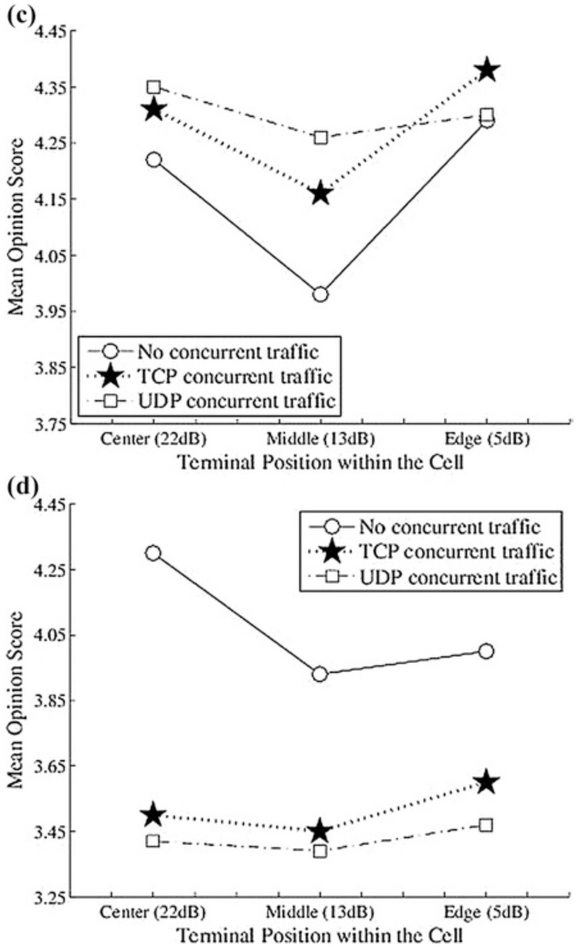


Fig. 3 (continued)



edge. It can also be seen from the figure that regardless of the QCI considered, the MOS assumes values ranging from "some users unsatisfied" to "very satisfied". However, thanks to the QoS aware WP scheduler, which prioritizes GBR flows over non-GBR flows, the MOS degradation perceived as the UE moves toward the cell edge is more significant for QCI = 7 than for QCI = 1. As for the delay variation, a substantial increase is observed in Fig. 3b as the UE departs from the center of the cell and approaches the cell edge. For QCI = 7, the delay variation on the cell edge is roughly three times higher than that measured for QCI = 1 at the same position. If we now move on to the results of MOS with concurrent traffic, shown in Fig. 3c, d for QCI = 1 and QCI = 7, respectively, we see a dramatic MOS degradation for QCI = 7 but not for QCI = 1. While the MOS remains in between "satisfied" and "very satisfied" for QCI = 1, it undergoes a big drop from "very satisfied" to "many users un satisfied". This is in good agreement with our

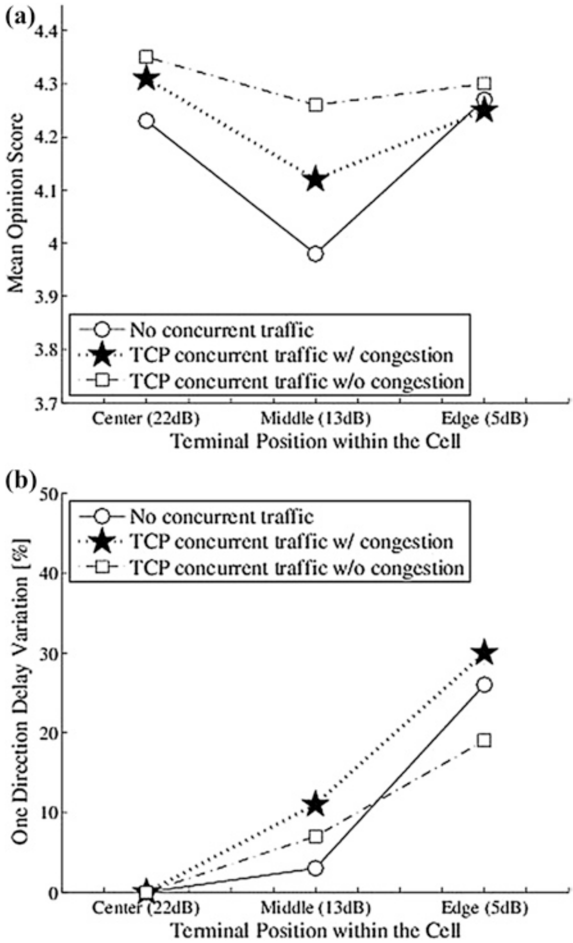


Fig. 4 Test scenario II: MOS and delay variation measurements obtained using RTP traffic under TCP concurrent traffic with and without congestion. **a** MOS (QCI = 1 for RTP, QCI = 9 for TCP). **b** Delay variation (QCI = 1 for RTP, QCI = 9 for TCP)

expectations, as the QoS aware WP scheduler again prioritizes QoS with QCI = 1 (even under unfavorable concurrent traffic conditions). When it comes to the type of concurrent traffic, the variations in MOS observed for TCP and UDP traffic are minimal for both QCIs.

4.2 Test Scenario II

MOS and delay variation results obtained for scenario II are shown in Fig. 4. Under different traffic conditions, it is shown in Fig. 4a that the MOS once again decreases as the UE moves from the center to the middle of the cell to then increase when the UE further moves toward the cell edge. However, the careful reader will see that the overall MOS score without concurrent traffic is lower than with concurrent traffic. While this finding may sound somewhat counterintuitive, it is indeed related to the QoS aware WP scheduler (which prioritizes the GBR traffic flow so as to maximize the system performance under concurrent non-GBR traffic flows). MOS variations due to both type of concurrent traffic and changes in the UE position within the cell are not that significant, as a result of QoS aware WP scheduling working alongside with adaptive modulation.

Figure 4b shows that the increase in delay variation in one direction varies from 20 to 30 % as the UE moves from the cell of the center toward the cell edge. This suggests that the QoS aware WP scheduler is prioritizing the GBR traffic flow so as to maximize its performance in the presence of a concurrent non-GBR traffic flow.

4.3 Test Scenario III

MOS results obtained for scenario III are shown in Fig. 5. When 64-QAM is in use, the MOS decreases as the distance between the UE position and the center of the cell increases. In contrast, when adaptive modulation or QPSK is employed, the MOS decreases as the UE moves from the center to the middle of the cell to then increase as the UE further moves toward the cell edge. Regarding the adaptive modulation, the message conveyed by the MOS behavior observed here is that the modulation order does not change as the UE moves from the center to middle of the cell, but does change as the UE moves from the middle to the edge of the cell.

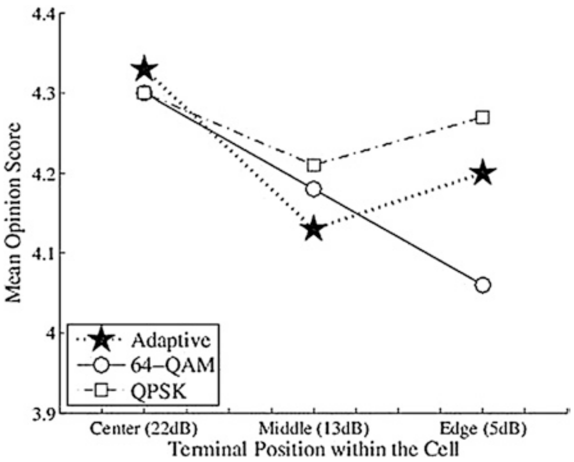


Fig. 5 Test scenario III: MOS measurements obtained for different modulations using RTP traffic under TCP concurrent traffic without congestion

5 Concluding Remarks

This chapter has presented a measurement-based performance assessment of VoIP in LTE networks for suburban and rural applications. The behavior of the embedded eNodeB QoS aware scheduler is evaluated when different LTE configurations of QoS and cell conditions are taken into consideration. Our results suggest that the aforementioned scheduler is capable of maintaining VoIP quality using QCI = 1 with concurrent traffic over default bearer or even over the non-GBR dedicated bearer.

The QoS aware algorithm works in such a way that the MOS parameter remains constant even in adverse conditions, for instance, under concurrent traffic delay increase conditions, or in the presence of signal reception degradation due to the UE position within the cell. Regarding adaptive modulation, our results suggest that the modulation order decreases as the UE moves from the middle to the edge of the cell. However, the same does not happen as the UE moves from the center to the middle of the cell. Despite the preliminary nature of our results, this is a clear indication that the adaptive modulation is not working properly on the eNodeB software version used in our test scenarios. Apart from that, the results obtained in our tests demonstrate accordance with the QoS standards mandated by 3GPP. As future work, new tests will be carried out to augment the present work with other important variables, such as, new codecs used by the LTE technology, other QCI comparisons, different LTE VoIP QoS control mechanisms, and different modulation schemes, with the aim of verifying the technology robustness in application scenarios with UE mobility and interference presence. We also plan to repeat the tests after having corrected the adaptive modulation software, and consider over-the-air test scenarios focusing on QoS tests for performance of VoIP calls over LTE.

Acknowledgment This material is based upon the work supported by FUNTTEL/FINEP 01.12.0481.00.

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