Transport Protocols Behaviour Study in Evolving Mobile Networks

Rui Li*†, Mehrdad Shariat*, Maziar Nekovee*
* Samsung Electronics R&D Institute UK, Staines-upon-Thames, Surrey, United Kingdom
† School of Informatics, The University of Edinburgh, Edinburgh, United Kingdom
Email: {rui7.li, m.shariat, m.nekovee}@samsung.com

Abstract---**In this paper, we present the performance evaluation of two widely used transport protocols, i.e. TCP and UDP, operating on top of the LTE network structure. We investigate key metrics that influence directly the user experience, such as the end-to-end throughput, under various channel conditions and protocol settings. We identify a number of performance issues when the current LTE protocol stack is exposed to inferior channel quality. Specifically, when the user is located at the cell edge, the interference from neighbouring cell becomes intenser while the signal power reduces due to distance. The SINR will drop, and thus the throughput and delay degrade significantly for both UDP and TCP traffic. Although traffic running on top of UDP obtains marginally better throughput, it observes very high packet loss. Further, we discover that the transport protocols investigated are sensitive to control plane errors. Enabling RLC acknowledged mode can mitigate partially the PDU loss, and hence it improves the throughput of TCP remarkably at the cell edge. However, AM introduces additional overhead and therefore may slightly cost the throughput and delay in good link conditions. Moreover, when reaching the maximum retransmission window, AM will rely on upper layers to recover the loss. Finally, we conclude that in the existence of high control error rate, robust modulation and coding scheme is needed. Alternatively, RLC acknowledged mode can be utilised to combat the packet loss, when TCP is used as transport protocol.**

I. Introduction

Developed by the 3rd Generation Partnership Project (3GPP), LTE employs novel technologies, e.g. downlink orthogonal frequency division multiple access and uplink single carrier - frequency division multiple access, which enable enhanced data rates and improved quality of service, compared to previous 3G networks. However, the emerging user scenarios e.g. virtual and augmented reality, personal portable gaming devices, and applications such as media on demand and cloud services, continuously drive the ever increasing data traffic demand. The research community is therefore constantly pursuing mobile network technology improvement to address the performance issues in divergent user cases. While moving towards 5G, edge-less experience is one of the key performance requirements, along with faster data rates, lower latency and better coverage.

The Transmission Control Protocol (TCP) provides reliable data transmissions by introducing hand-shaking, error checking, ordering correction and congestion avoidance mechanisms. It is widely utilised in today's internet applications. As reported in [1], over 95% of internet data traffic are based on TCP. However, when deployed in reality, TCP's level of performance may vary, depending on how the congestion control function react to the unpredictable radio link environment, how the protocol overhead impact, and to what level the retransmission scheme recovers packets. Comparing to TCP, User Datagram Protocol (UDP) eliminates transmission overhead, but it provides no guarantee in delivery.

It is therefore critical to understand the behaviour of the transport protocols in the current LTE systems. Driven by such motivation, we inspect several performance metrics, i.e. end-to-end throughput, number of packet loss, Round-Trip Time (RTT) and Congestion Window Sizes (CWND), for traffics running under TCP or UDP, under divergent protocol configurations and different qualities of channel environments. We are particularly interested in the users located at the cell edge, where the received SINR by the UE is under satisfaction.

Moreover, in the control plane of the current LTE system, correct decodification of the Data Control Indicators (DCIs) depends on the correct interpretation of both Physical Control Format Indicator Channel (PCFICH) and Physical Downlink Control Channel (PDCCH) [2]. However, PCFICH and PDCCH are not protected by any Automatic Repeat Request (ARQ) scheme. When error occurs in these symbols, the data carried in the subframe will no longer be decoded. With frequent control channel errors, data packet loss rate will increase. Our results confirm that erroneous control symbols cause further decrease in the overall user experience. Further, our simulation results suggest by enabling RLC layer acknowledged mode (AM), packet loss can be recovered to some extent, and the performance issue can be partially addressed.

The rest of this paper is organised as the following. In Sec. II, we review relevant works, and then we describe the simulation setup in Sec. III. The result obtained will be discussed in Sec. IV. Finally, we conclude in Sec. V.

II. Related Works

Originally designed for wired networks, TCP is able to check and correct errors as well as out-of-order deliveries, to provide reliable transmissions. However, existing study on TCP behaviour in LTE-EPC network suggest that sudden load increase in a cell will lead to significant bandwidth reduction and max delay increase [3]. Due to the uncertainty exposed in the wireless networks, especially the changing link quality, TCP retransmission lead to larger overhead and network inefficiency [1]. Zhang *et al.* argue that a small handover offset leads to better throughput performance in spite of the increasing probability of ping-pong handover [4]. Challenges of optimising cell-edge SINR are presented in [5], and it is suggested that inter-cell interference coordination schemes should be employed in PDCCH. To the best of our knowledge, our work is the first study on transport protocols' performance in LTE-EPC network with specific interests on cell edge users, and covers the issue introduced by errors in the control plane.

III. LTE-EPC Simulation Setup

In order to perform practical modelling of the interaction between the transport protocols and the lower layers, as well as end-to-end QoE evaluations, we utilise the build-in LTE module of NS-3, namely LENA [6], as the simulation tool. In this section, we describe the simulation settings and review some of the design aspects in LENA.

We examine the performance of UDP and TCP downlink data traffic from a remote server to a single UE. During each 50 s simulation, the UE is assigned a 20 m square box as an area of activity, and it moves towards random direction at 3 kmph velocity within this box. The simulation runs 14 times for each configuration, with the UE's area of activity placed at different distances to the eNodeB it is attached to. The longest and the shortest distance between the centre of UE's box to the eNodeB is 300 m and 40 m respectively, and the distance between the centres of two adjacent boxes for two simulation runs is 20 m. 4 groups of simulations are performed under the following difference types of settings: RLC operates in UM with and without the existence of control frame errors, and RLC AM with control frame error model switched on and off. For each simulation scenario, both TCP and UDP are examined. Default simulation seed is used for all simulations, hence when horizontally compare the simulation run, at the same location of UE but different simulation settings, the channel environment e.g. SINR, are the same at each simulation run time.

The topology of LTE-EPC network is shown in Fig. 1. Specifically, the UE is connected to a single eNodeB, which has wired connections with the Service Gateway (SGW) and other eNodeBs. The SGW links to the remote server based on a high-speed point-to-point (P2P) connection of 10 Gbps and this link introduces a delay of 10 ms. In the LTE network, the eNodeBs are grouped in a three-sector sites lay-ed out on a hexagonal grid, as depicted in the Radio Environment Map (REM) in Fig. 2. In order to evaluate realistic interference scenarios, the cell site of interest is surrounded by 2 layers of three-sector sites which generates interference on both data and control channel. All nodes are assumed to be placed outdoor. Throughput and packet loss measurements are collected by the NS-3 Flow Monitor module at the IP layer. As for TCP congestion control protocol, New Reno is employed throughout the simulation, and we record the RTT and CWND trace for all simulation runs with TCP traffic. TABLE I lists the general configuration of the simulation.

Regarding the propagation model, ITU-R P1411 path loss model [7] is used in the experiment scenario, and a log-normal shadowing model provides shadow fading values. LENA takes offline calculated fading trace generated from MATLAB. In our simulation, the multi-path fading conditions follow the Extended Pedestrian A profile specified in Annex B.2 of 3GPP standard TS 36.104 [8]. The fading amplitude is calculated as random process based on the commonly used Rayleigh model, which is a function of both time and frequency.

On the PHY layer, frequency-division duplexing is implemented in LENA, and the Transmission Time Interval (TTI) is 1 ms. In the data frame, reference signal power received every TTI is used to calculate the SINR, and Channel Quality Information (CQI) feedback is generated using the SINR obtained. Interference is modelled by the Gaussian interference models, according to which the overall interference power is calculated by summing up all interfering signals power. The adopted error model for both control and data plane is based on link-

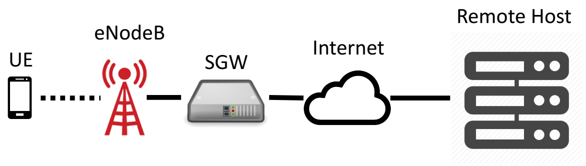

Fig. 1: LTE-EPC Topology.

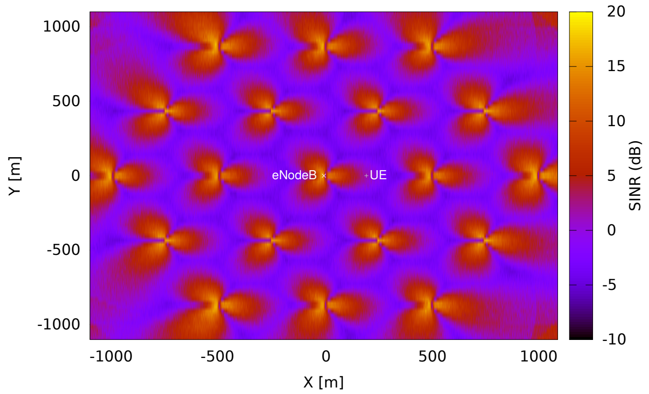

Fig. 2: REM for LTE RAN control channel.

Parameter Name	Value
Antenna type	Parabolic
Beamwidth	70°
Transmission power	46 dBm
Site height	30 m
Sector offset at each 3-sector site	0.5 m
Inter-site distance	500 m
UE height	1.5 m
Carrier Frequency for downlink	2.1GHz
Carrier Frequency for uplink	1.9GHz
Bandwidth	50 RBs (10 MHz)
Standard deviation of shadowing	$\sigma = 1$
Traffic Pattern	Backlogged
Packet size	1024 bytes
TCP EPS bearer	QNGBR_VIDEO_TCP_DEFAULT
RLC transmission buffer size	1024 Kbytes

TABLE I: General simulation settings.

to-system mapping. Furthermore, the Hybrid ARQ (HARQ) is utilised for data channel, and it employs soft combining hybrid incremental redundancy scheme with multiple stop-and-wait, which means that the retransmissions contain only new information respect to the previous transmissions. The HARQ model is integrated with the error model, and the retransmissions are arranged by the scheduler.

As per TS 36.211 standard [2], downlink control frame, i.e. PCFICH and PDCCH, starts at the beginning of each subframe and in total lasts no more than three symbols. The subframe structure in LENA is implemented accordingly [9], as shown in Fig. 3. PCFICH indicates the actual length of the control frame, and PDCCH mainly carries the DCI assigned by the MAC layer, including resource allocation for the UE. Error in the control channel thus results in the loss of corresponding TBs transmitted in the TTI.

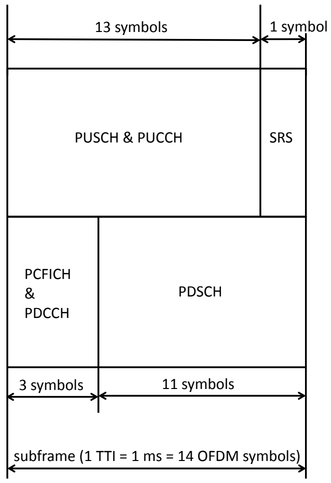

Fig. 3: Subframe structure of LTE. [9]

IV. Simulation Results

In this section we discuss the results obtained from the simulations. We start with analysing the throughput and error occurrence when control error model is turned off, for RLC running in both AM and UM and for both UDP and TCP traffic. Then we will delicately investigate the throughput performance of TCP when exposed to control frame signal failures, with the reference of UDP throughput. Further, we compare the throughput values of TCP operating with RLC AM and UM. Finally, we will take a look at the CWND and RTT traces for TCP traffics obtained from a number of representative scenarios.

Fig. 4 depicts the average throughput and the total number of packet loss at each location, for TCP and UDP in both RLC AM and UM scenarios, when the control channel, i.e. PCFICH and PDCCH are free of error. It can be seen that as the distance between the UE's active area and the eNodeB increases from 40 m to 300 m, both UDP and TCP will see a significant reduction in throughput in both RLC modes. Specifically, UDP achieves approximately 0.5 Mbps better throughput than TCP at most in UM, and around 1 Mbps in AM. However, TCP manages to eliminate all data segment losses by its retransmission scheme, whereas UDP suffers from severe data packet loss in RLC UM scenarios. With the help of RLC ARQ, packet loss in UDP is reduced approximately by half compared to the equivalent cases when RLC operates without ARQ.

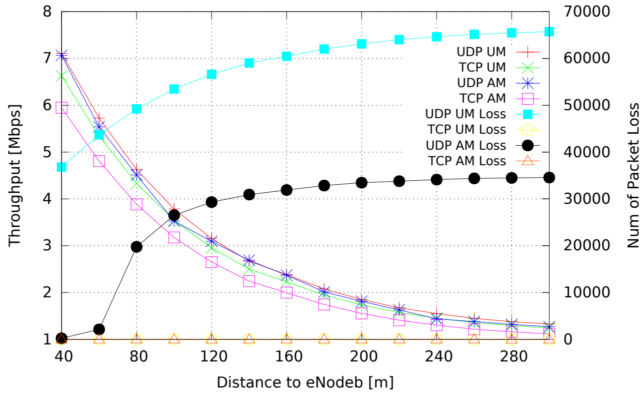

Fig. 4: The average throughput for TCP and UDP versus the distance to eNodeB, and the total number of packet loss during each simulation run, under RLC AM and UM. Control channel error model is disabled.

It is also worth noting that, as currently identified an open issue between RLC AM and MAC scheduling schemes, the scheduler takes into account only the length of data in RLC PDUs and discards RLC headers when making scheduling decisions. As a consequence, a RLC PDU may have been allocated a Transport Block (TB) according to the amount of data it carries, but after adding RLC and MAC headers it can no longer fit in the TB size. RLC will perform another segmentation, then the PDU will take an additional transmit opportunity. This unwanted segmentation results in slight reduction of UDP throughput in some scenarios, but it has a greater impact on TCP, due to the additional delay introduced by segmentation, which in turn leads to longer RTTs.

From this set of results, we conclude that in the absence of control channel errors, despite the inconsiderable loss of throughput when compared against UDP, TCP is able to recover all packet loss in all cases. In addition, UDP experiences significant packet loss, even though it shows a marginally higher throughput, the user experience can be anticipated as poor due to the noncontinuous packet streams. In real life, such on-and-off behaviour of a traffic can hardly satisfy any data-intensive applications such as video streaming. However, when RLC introduces ARQ to request retransmissions, the packet loss or out-of-order delivery can partially be recovered for UDP. Hence the user experience can potentially be improved to some extent without considerably compromising the throughput for UDP, by enabling AM in RLC.

Nevertheless, there exists a maximum retransmission attempt of RLC PDUs. As per TS 36.322 and TS 36.331 [10, 11], upon such event, RLC shall inform the upper layers, to trigger a Radio Link Failure (RLF). Note that RLF is not currently implemented in NS-3, and RLC will simply stop forwarding down any PDUs when the maximum retransmission threshold is reached. However, this means that in practice RLC may fail to recover all lost packets, and in extreme cases, it will rely on upper layers to recover packet loss. In such cases, UDP will take the packet loss for granted, whereas TCP will endeavour to recover as many lost packet as possible, while performing congestion control at the meantime.

In the above studied cases, we note that the congestion control of TCP is hardly constraining the transmission behaviour, thanks to the HARQ that provides timely correction of byte-wise errors, and TCP itself that detects and corrects packet losses efficiently enough without triggering slow-start frequently. We further investigate the error-prone case of control plane, where the control channel is assumed to be exposed to full interference from the surrounding cells, and consequently, PCFICH and PDCCH symbols become erroneous. Each occurrence of such error in control plane, as mentioned previously, will result in the loss of all the TBs carried in the TTI. It can be therefore anticipated that when UE observe such high rates of packet loss, TCP may experience severe performance degradation, but switching to RLC AM can presumably recover the packet loss to some extent. This is confirmed by our simulation results shown in Fig. 5.

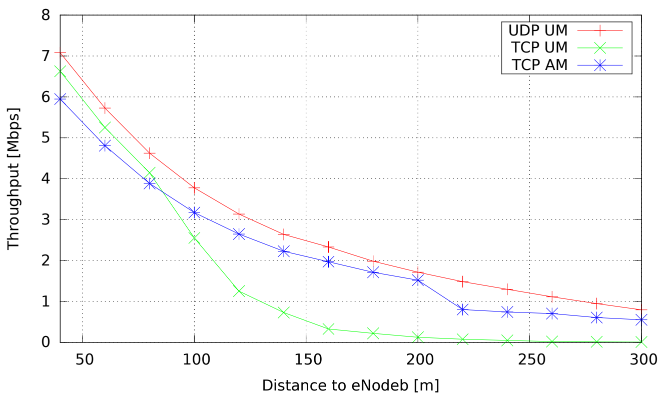

Fig. 5: The average TCP throughput in RLC AM and UM, with the reference of UDP UM, versus the distance to eNodeB. Control channel error model switched on.

Similar to the previous observation of throughput-distance behaviour in control error-free cases, the throughput of UE decreases dramatically when the UE moves away from the cell centre when control symbols encounter error, while Fig. 5 also suggests that the degradation is particularly significant for TCP. This is because that when link quality is poor, the TCP is aware of the numerous packet loss, so it reduces CWND and carries out retransmissions to recover the lost packets. Such retransmission will increase RTT, and further degrades the throughput in UM, particularly at the cell edge scenarios.

Further, when RLC AM is enabled, a number of packet losses caused by control symbol errors can be actively recovered between the eNodeB and the UE. Then fewer DupAcks will be received by the TCP at the transmitter's side, so congestion as seen by TCP is less serious, and the TCP therefore has larger CWND and allows more packet to be transmitted. Consequently, the throughput of TCP is therefore improved, up to a level that is just marginally lower than UDP's. When UE is located near the eNodeB, e.g. at the distance of 40 m to 80 m, both data and control plane observes less interference and therefore good SINR. Thus the impact of packet loss is greatly reduced, and RLC AM has less chance to recover packet loss but introduce more unwanted segmentation effect, as discussed previously. Eventually, we see a moderate throughput degradation in AM compared to UM, when the UE is located at the cell centre.

Packet loss in UDP is again ignored, as by nature, UDP provides no guarantees for delivery. The numerous packet losses especially at the cell edge can drastically degrade the user experience. Different from the scenario where the control channel is error-exempt, with a larger number of packet loss added due to erroneous control symbols, RLC will frequently jump into the maximum retransmission limit, and in the current protocol design of RLC, it will cease transmitting any more PDUs and rely on upper layers to recover the packet loss. Occasionally, TCP can recover this when uplink ACK is received, but for UDP, there is no current rescue method that can trigger a recovery, resulting in RLF in practice. In both TCP and UDP cases, we see severe throughput decrease in the simulation scenarios for distances above 200 m.

We record the CWND at each time it changes, and plot the median value of its distribution in Fig. 6. This graph shows that in the absence of control errors, the CWNDs are relatively higher. This is because without frequent and consecutive packet loss, the CWND will be increased every RTT. When moving away from the cell centre, the UE may observe increase in RTT, and therefore the raise of CWND is slowed down. Median value of CWND is hence reduced.

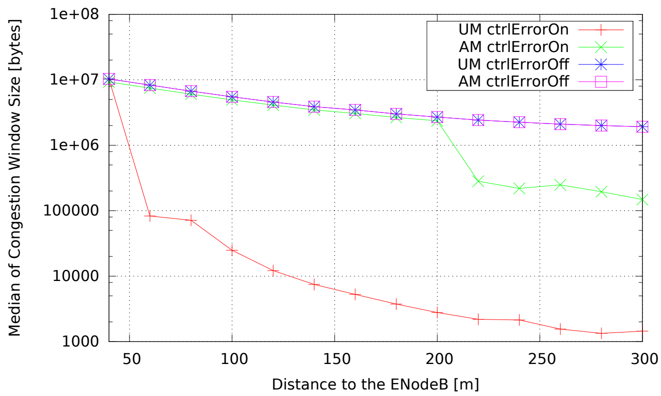

Fig. 6: Median of CWND when UE is placed at different locations, for all four simulation settings.

When control error exists but the distance between UE and eNodeB is short, implying mitigated packet loss, the CWND is maintained relatively high. This suggest that under good link quality, congestion control is hardly constraining the transmission of packets from the TCP. Whereas when the control symbols are erroneous, and RLC deactivates AM, the CWND is defectively low, which will result in very limited transmission allowed by TCP congestion control. This presumably contribute to the throughput degradation of TCP in UM, and explains why TCP throughput is much lower compared to UDP. With the presence of ARQ at RLC layer, the CWND is enlarged, and in other words, more TCP segments are allowed to be transmitted.

The RTT reports the time consumed for a complete TCP segment transmission, i.e. from the segment leaves the transmitter to the complete receiving of an ACK. As a reference we run a single simulation with interference model switched off and the UE is placed at 300 m to the cell centre. We note that, the UE in such case experience superior link quality, and the RTT is the shortest among all scenarios shown in Fig. 7. The RTT curves for UM mode indicate that the further the UE is located to the eNodeB, the longer the RTT tends to be. This suggests that with worse link quality, a successful transmission of TCP segment takes longer. Such behaviour is perhaps due to 1) more retransmissions are taking place, and hence a packet spans more TTIs; and 2) the queue builds up in lower layer buffers, so the PDUs wait for longer to be transmitted. On the other hand, the RTT in AM demonstrates the similar behaviour, so we only shows the CDF plot of RTT at a single location, i.e. 40 m, to compare with UM. Depicted in solid black curve, the RTT distribution in AM is slightly higher than that of the UM scenario. This can be explained by the issue of unwanted segmentation taking place at the RLC layer, as already mentioned in above paragraphs. Combining the throughput performance discussed previously, the cell edge users experience both increased delay and decreased throughput, which could potentially lead to degraded user experience.

Also, as more RTTs are sampled thanks to the better links observed, the cell centre scenarios (distance $\leq$ 180 m) have a smoother curve of CDF. Furthermore, within 180 m, the width of the RTT distribution grows as the distance increases. Note that due to the severe throughput reduction at the distant locations when control error model is enabled, the number of samples collected by the simulation is very limited and is therefore less representative. We therefore only investigate the case that is free of control errors for RTT.

V. Conclusions

In this preliminary study, we examined the impact of several parameters in the LTE networks that influence the performance of TCP and UDP. We conclude that as the distance between UE and eNodeB increases, both transport protocols' performance decrease significantly. The lack of ARQ scheme in the current implementation of LTE control plane brings increasing control channel error rate when exposed to very low SINR scenarios, which will further degrade the throughput. The recovery of data packet loss is currently relying on data plane ARQ schemes. Moreover, RLC AM mode overcomes

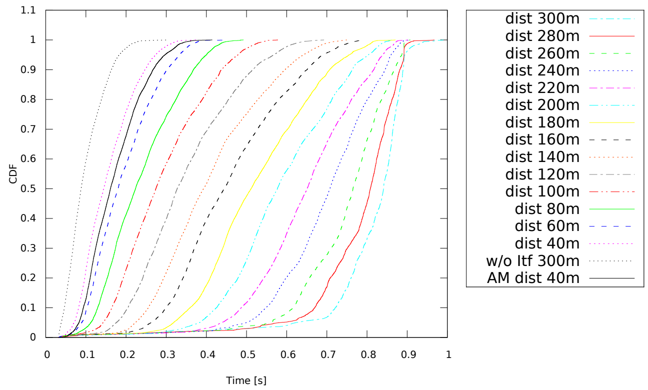

Fig. 7: RTT CDF without control error. First 14 curves listed in the legend are obtained in UM, with UE at different locations. The dashed black curve represents the reference scenario where the channel is free of interference and the UE is at 300 m. The solid black curve is obtained in AM, with UE located at 40 m.

the performance bottleneck introduced by control symbol errors up to a certain level. Therefore, robust modulation and coding schemes are needed to mitigate error rate in the control channels of LTE networks. Future work can be carried out to improve the transport protocol design, probably by enabling cross-layer cooperation to allow better user experience.

References

[1] J. Huang, F. Qian, Y. Guo, Y. Zhou, Q. Xu, Z. M. Mao, S. Sen, and O. Spatscheck, ''An in-depth study of LTE: effect of network protocol and application behavior on performance,'' in *ACM SIGCOMM Computer Communication Review*, vol. 43, pp. 363--374, ACM, 2013.

[2] ''Evolved Universal Terrestrial Radio Access (E-UTRA); Physical channels and modulation,'' Standard, 3rd Generation Partnership Project Technical Specification Group, July 2016.

[3] B. Nguyen, A. Banerjee, V. Gopalakrishnan, S. Kasera, S. Lee, A. Shaikh, and J. Van der Merwe, ''Towards understanding TCP performance on LTE/EPC mobile networks,'' in *Proceedings of the 4th workshop on All things cellular: operations, applications, & challenges*, pp. 41--46, ACM, 2014.

[4] L. Zhang, T. Okamawari, and T. Fujii, ''Performance evaluation of tcp and udp during lte handover,'' in *Wireless Communications and Networking Conference (WCNC), 2012 IEEE*, pp. 1993--1997, IEEE, 2012.

[5] ''Enhancing LTE cell-edge performance via PDCCH ICIC,'' Whitepaper, Fujitsu Network Communications Inc, 2011.

[6] ''LENA: LTE-EPC network simulator.'' http://networks.cttc.es/mobile-networks/software-tools/lena/. Accessed: 2016-07-1.

[7] ''Propagation data and prediction methods for the planning of short-range outdoor radiocommunication systems and radio local area networks in the frequency range 300 MHz to 100 GHz,'' Recommendation, International Telecommunication Union, July 2015.

[8] ''Radio Access Network Evolved Universal Terrestrial Radio Access (E-UTRA) Base Station (BS) radio transmission and reception,'' Standard, 3rd Generation Partnership Project Technical Specification Group, Mar. 2016.

[9] ''LENA Design Documentation.'' https://www.nsnam.org/docs/models/html/lte-design.html. Accessed: 2016-07-1.

[10] ''Evolved Universal Terrestrial Radio Access (E-UTRA); Radio link control (RLC); Protocol specification,'' Standard, 3rd Generation Partnership Project Technical Specification Group, July 2016.

[11] ''Evolved Universal Terrestrial Radio Access (E-UTRA);Radio resource control (RRC); Protocol specification,'' Standard, 3rd Generation Partnership Project Technical Specification Group, July 2016.