

Research Article

A Simulation Tool for Downlink Long Term Evolution-advanced

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Abstract: Long Term Evolution-Advanced (LTE-A) is an emerging mobile cellular system envisaged to provide better quality of multimedia applications. Packet scheduling becomes paramount as the LTE-A delivers multimedia applications using packet switching technology. Given that LTE-A is a new technology, its ability to satisfy the Quality of Service (QoS) requirements of multimedia applications demands further performance study. At present, a number of LTE-A simulators are available. However, these simulators in general are too specific in nature or their source codes are not publicly accessible for the research communities. As such, this study presents a novel simulation tool to assist the research communities to study the downlink LTE-A and further optimize packet scheduling performance. This simulation tool accurately models the downlink LTE-A taking user mobility, carrier aggregation, packet scheduling and other aspects that are relevant to the research communities into consideration. The efficacy of the simulation tool is validated through performance study of a number of well-known packet scheduling algorithms.

Keywords: Carrier aggregation, component carrier, orthogonal frequency division multiple access, packet scheduling, quality of service

INTRODUCTION

Long Term Evolution-Advanced (LTE-A) is a mobile cellular system proposed by the Third Generation Partnership Project (3GPP) to meet the requirements of International Mobile Telecommunications-Advanced (IMT-Advanced) for the Fourth Generation (4G) technology. LTE-A, also known as Release 10 standard, is a backward compatible enhancement of the legacy Long Term Evolution (LTE) Release 8 standard. The LTE-A is envisaged to provide a better quality of multimedia applications by providing higher peak data rates with reduced latency and increased capacity and coverage. The multimedia applications, which consist of Guaranteed Bit Rate (GBR) and Non-Guaranteed Bit Rate (Non-GBR) applications, have competing QoS requirements. For instance, most of the GBR applications are more sensitive to packet delay, whereas most of the Non-GBR applications are more sensitive to loss packets.

The main feature that differentiates LTE-A from the LTE Release 8 standard (referred to as LTE hereafter) is that the LTE-A uses Carrier Aggregation (CA). CA allows the system to support much wider transmission bandwidth by aggregating a number of Component

Carriers (CCs) of the same or different frequency bands (Fan *et al.*, 2011). Note that each CC contains a number of radio resources available to be shared among the users. This improves the peak data rate in LTE-A system (Pedersen *et al.*, 2011). LTE-A delivers multimedia applications using packet switching technology. Therefore, packet scheduling becomes paramount in the LTE-A. Packet scheduling is a process that efficiently selects a user's packets for (re) transmission at a given time using an available radio resource so as to provide a satisfactory Quality of Service (QoS), guarantee fairness and optimize system performance.

LTE-A uses Orthogonal Frequency Division Multiple Access (OFDMA) for downlink transmission. OFDMA is an access technology that divides the available wide bandwidth into multiple equally-spaced and mutually-orthogonal sub-carriers (Daoud and Alani, 2009). The minimum downlink LTE-A transmission unit that can be allocated to a user is referred to as a Resource Block (RB). A number of RBs are available for usage among the downlink LTE-A users at 1 ms Transmission Time Interval (TTI), but each RB can only be assigned to a single user in each TTI. The LTE-A Evolved Universal Terrestrial Radio Access Network (E-UTRAN) architecture consists of base stations only called enhanced Node Bs (eNBs). The eNB connects

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users to the core network and performs all radio resource management functions.

Given that LTE-A is a new mobile cellular system, the ability of LTE-A to satisfy the QoS requirements of multimedia applications demands further performance study. Theoretical analysis, test bed and computer simulation are a number of methods used for evaluating mobile cellular performance. Theoretical analysis method is too complex, cannot be performed in real time and requires a number of assumptions to simplify the analysis. Modeling the mobile cellular systems using test bed method is expensive (the method requires hardware and labor resources) and the results are heavily influenced by the testing environment (Kefeng *et al.*, 2010). On the other hand, computer simulation is a well-established and less expensive method that makes modeling and investigation of a large scale and complex mobile cellular system feasible (Bononi *et al.*, 2004). It gives a full control for the research communities to study the traffic flow behavior compared to the theoretical analysis and test bed. Moreover, this method allows the research communities to design and modify the mobile cellular scenarios easily.

At present, a number of LTE-A simulators are available. However, these simulators in general are too specific in nature (Bouras *et al.*, 2012) or their source codes are not publicly accessible for the research communities (Ikuno *et al.*, 2010; Piro *et al.*, 2011). To bridge this gap, this study presents a C++ computer simulation tool that dynamically models the large and complex downlink LTE-A. This tool aims to assist the research communities to investigate the characteristics and behavior of the downlink LTE-A via simulation. Additionally, it can be used to optimize carrier aggregation and packet scheduling performance in this system.

MATERIALS AND METHODS

System overview: There are three types of CA being specified for LTE-A namely Intra-band Contiguous CA, Intra-band Non-Contiguous CA and Inter-band Non-Contiguous CA (Iwamura *et al.*, 2010). The Intra-band Contiguous CA aggregates a number of adjacent CCs within the same frequency band whereas Intra-band Non-Contiguous CA aggregates a number of CCs within the same frequency band in a non-contiguous manner. A CA type that aggregates a number of CCs of different frequency bands is called Inter-band Non-Contiguous CA (Hua *et al.*, 2011). Intra-band Contiguous CA is easier to implement as it requires minimum changes to the radio frequency design of the legacy LTE system. However, given that the current mobile cellular spectrums are highly fragmented with large frequency separation, the Inter-band Non-Contiguous CA is more practical for use by the cellular operators at the initial stage of LTE-A.

As previously stated in the Introduction, the LTE-A should maintain backward compatibility with the legacy

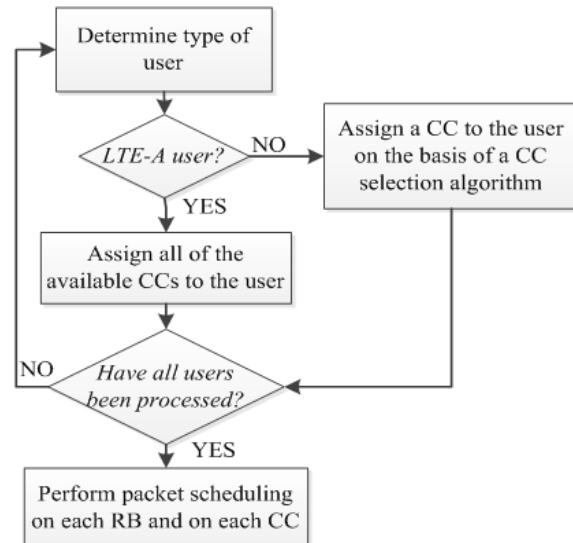


Fig. 1: A generalized model of the downlink LTE-A with CA

LTE system. This means that the LTE users should be able to co-exist with LTE-A users (ElBamby and Elsayed, 2012). With CA, the LTE-A users, which have high performance transceivers, are capable to simultaneously support transmission on multiple CCs whereas the LTE users are restricted to transmit on only a single CC due to the limited capability of their transceivers. Figure 1 illustrates a generalized model of the downlink LTE-A with CA consisting of LTE-A and LTE users. Given that there is more than one CC available, CC selection algorithm becomes necessary in the LTE-A. Though a large number of CC selection algorithms have been developed for the LTE-A, this paper studies and models the well-known Random CC selection algorithm (Chunyan *et al.*, 2012) into the simulation tool. This algorithm aims to balance the load from long term point of view by randomly assigns a CC to each LTE user. It should be noted that modeling and performance study of other CC selection algorithm will be a part of future studies.

Packet scheduling in LTE-A takes place after users have been assigned to a CC (LTE user) or a number of CCs (LTE-A user) (Fig. 1). Packet scheduling algorithm is used in the downlink LTE-A to determine the user whose packets which will be transmitted on each RB. This algorithm takes into consideration one or more scheduling information (i.e., Channel Quality Information (CQI), average throughput, packet delay information, buffer status, etc.) so as to provide satisfactory QoS, guarantee fairness and optimize system performance (Nan *et al.*, 2010). Conceptually, there have been numerous packet scheduling algorithms developed for the mobile cellular systems. This study focuses on four well-known algorithms and these algorithms are described next.

Maximum Rate (Max-Rate): The Max-Rate Tsybakov (2002) algorithm always selects for transmission the packets of a user with the best channel quality on a radio resource and is less likely to give any transmission opportunity to a user with a poor channel quality. It is a good candidate for throughput maximization. However, the Max-Rate is not capable of guaranteeing fairness among the users.

Round Robin (RR): Given that fairness has been an issue in the Max-Rate algorithm, the RR algorithm (Dahlman *et al.*, 2007) was developed to tackle this situation. This algorithm gives equal opportunity to each user to receive its packets in a cyclic fashion. The RR algorithm considerably improves fairness performance. However, the throughput degradation in this algorithm is significant as scheduling decisions in the RR algorithm do not take the channel quality of each user into consideration.

Proportional Fair (PF): The PF algorithm (Jalali *et al.*, 2000) provides a better trade-off between throughput maximisation and fairness guarantee. In each scheduling interval, the PF algorithm schedules packets of a user that maximizes $\mu_i(t)$ in Eq. (1):

$$\mu_i(t) = \frac{r_i(t)}{R_i(t)} \quad (1)$$

$$R_i(t) = \left(1 - \frac{1}{t_c}\right) R_i(t-1) + I_i(t) * \frac{1}{t_c} * r_i(t) \quad (2)$$

where,

- $\mu_i(t)$: The priority of user i at scheduling interval t
- $r_i(t)$: The instantaneous data rate (across the whole bandwidth) of user i at scheduling interval t
- $R_i(t)$: The average throughput of user i at scheduling interval t
- $I_i(t)$: The indicator function of the event that packets of user i are selected for transmission at scheduling interval t
- t_c : A time constant

Modified-Largest Weighted Delay First (M-LWDF): The GBR applications are constrained by packet delay. The Max-Rate, RR and PF algorithms are inefficient to satisfy the QoS requirements of the GBR applications for not taking packet delay into consideration. As such, M-LWDF algorithm (Andrews *et al.*, 2001) was developed to address this situation. This algorithm selects a user that maximizes $\mu_i(t)$ to receive its packet in each scheduling interval:

$$\mu_i(t) = a_i * W_i(t) * \frac{r_i(t)}{R_i(t)} \quad (3)$$

$$a_i = - \frac{(\log \delta_i)}{T_i} \quad (4)$$

where, $\mu_i(t)$ is the priority of user i at scheduling interval t , a_i is the QoS requirement of user i , $W_i(t)$ is the delay of the Head-of-Line (HOL) packet of user i at scheduling interval t , $r_i(t)$ is the instantaneous data rate (across the whole bandwidth) of user i at scheduling interval t , $R_i(t)$ is the average throughput of user i at scheduling interval t , Eq. (2), δ_i is the application-dependent Packet Loss Ratio (PLR) threshold of user i and T_i is the application-dependent buffer delay threshold of user i . Note that the HOL packet of a user is the packet that has resided the longest in its buffer at the eNB while the buffer delay threshold is defined as the maximum allowable waiting time of a packet at the eNB buffer. This threshold is dependent upon the type of a multimedia application.

Simulation model: This simulation tool models the downlink LTE-A to contain a single cell with an eNB located at the centre of the cell. Users are uniformly located within the cell. The eNB uses a total of 43.01 dBm transmit power and only the frequency division duplex mode is used. As access to a large amount of contiguous CCs may not always be possible in practice due to highly fragmented spectrums with large frequency separation CA (as discussed in System Overview), Inter-band Non-Contiguous is considered. In this case, the downlink LTE-A contains 2 CCs of 900 MHz and 2 GHz carrier frequency. The number of CCs and bandwidth to be used in this simulation tool can be increased with minimum software changes.

Each CC is of 3 MHz bandwidth and contains 15 RBs. Note that, the RB is of 1 msec duration in the time domain and contains 14 OFDMA symbols with the usage of a normal Cyclic Prefix (CP) (Holma and Toskala, 2009). In the frequency domain, the RB contains 12 sub-carriers of 180 kHz total bandwidth (15 kHz bandwidth/sub-carrier). Each RB contains a total of 168 Resource Elements (REs). Majority of the REs are used to carry downlink data while the remaining REs are used for control and signaling purposes.

Mobility modeling: Each user moves within the cell at a constant speed in a constant direction. Each user is assigned a random direction at the beginning of its data session. A user is wrapped-around whenever it reaches the cell boundary so as to ensure that the user always remains within the simulation area throughout its data session (Orozco Lugo *et al.*, 2001).

Radio propagation modeling: Radio propagation refers to how radio signals are propagated/transmitted from the transmitter to receiver. It affects received signal strength experienced at a receiver. Channel gain is expressed as the ratio of the received signal strength to the transmitted signal. The channel gain is dependent upon path loss, shadow fading and multi-path fading gains. In this simulation tool, Hata model for urban environment (Holma and Toskala, 2007) is used to compute the path loss. The Hata model is based on experimental measurements and is considered as one of

the most accurate path loss model in mobile cellular systems. As there is more than one CC in this system, the path loss of a user on a CC is computed as follows:

$$pl_{i,k}(t) = 46.3 + 33.9 * \log_{10}(f_{i,k}(t)) - 13.82 * \log_{10}(h_b) - a(h_m) + (44.9 - 6.55 * \log_{10}(h_b)) * \log_{10}(|dis_i(t)|) \quad (5)$$

$$a(h_m) = (1.1 * \log_{10}(f) - 0.7) * h_m - (1.56 * \log_{10}(f_{i,k}(t)) - 0.8) \quad (6)$$

$$dis_i(t) = loc_i(t) - loc_i(t-1) \quad (7)$$

where,

$G(t-1)$: A Gaussian random variable of user i at time $t-1$

$\rho_i(t-1)$: The shadow fading autocorrelation function

$v_i(t-1)$: The speed of user i at time $t-1$

σ : The shadow fading standard deviation

d_0 : The shadow fading correlation distance

Gaussian lognormal distribution is a widely used probability distribution in signal processing. Therefore, the shadow fading gain is computed in this simulation tool using a Gaussian lognormal distribution with 0 mean and 8 dB standard deviation (Gudmundson, 1991). The equations below are used to determine the shadow fading gain for user i at time t ($\xi_i(t)$):

$$\xi_i(t) = \rho_i(t-1) * \xi_i(t-1) + \sigma * \left(\sqrt{1 - \rho_i(t-1)^2} \right) * G(t-1) \quad (8)$$

$$\rho_i(t-1) = \exp\left(\frac{-v_i(t-1)}{d_0}\right) \quad (9)$$

where,

$G(t-1)$: A Gaussian random variable of user i at time $t-1$

$\rho_i(t-1)$: The shadow fading autocorrelation function

$v_i(t-1)$: The speed of user i at time $t-1$

σ : The shadow fading standard deviation

d_0 : The shadow fading correlation distance

The multi-path fading gain is determined in this simulation tool based on a frequency flat Rayleigh fading (Patzold *et al.*, 1996). The Rayleigh fading is based on statistical model and is considered as a reasonable model for signal propagation. The frequency flat Rayleigh fading is approximated by a complex Gaussian random process and has the following equations:

$$\xi_i(t) = \rho_i(t-1) * \xi_i(t-1) + \sigma * \left(\sqrt{1 - \rho_i(t-1)^2} \right) * G(t-1) \quad (10)$$

$$\rho_i(t-1) = \exp\left(\frac{-v_i(t-1)}{d_0}\right) \quad (11)$$

$$f_{i,n} = f_{max} \sin\left(\frac{\pi}{2} \mu_n\right) \quad (12)$$

where,

$\mu_{ap_i}(t)$: The approximated uncorrelated filtered white Gaussian noise with zero mean of process i at time t

$c_{i,n}$: The Doppler coefficient (which represents a real weighting factor) of process i of the n^{th} sinusoid

$f_{i,n}$: The discrete Doppler frequency of process i of the n^{th} sinusoid

$\theta_{i,n}$: The Doppler phase of process i of the n^{th} sinusoid

N_i : The number of sinusoids of process i

μ_n : Uncorrelated filtered white Gaussian noise with zero mean of the n^{th} sinusoid

$\sigma_{\mu 0}$: The variance (mean power)

f_{max} : The maximum Doppler frequency

Signal-to-Interference-plus-Noise-Ratio (SINR): As each sub-carrier in an RB has a 15 kHz spacing, it is assumed that there are minimum variations of multi-path fading among the sub-carriers of an RB of a CC. Consequently; in this simulation tool, the instantaneous SINR on an RB and on a CC is computed on a sub-carrier located at the centre frequency of the RB (Ramli *et al.*, 2009a). The instantaneous SINR ($\gamma_{i,j,k}(t)$) experienced by user i on RB j on CC k at time t is computed as follows (Kim *et al.*, 2007):

$$\gamma_{i,j,k}(t) = \frac{P_{total} * gain_{i,j,k}(t)}{RB_{max}(I + N_o)} \quad (13)$$

$$gain_{i,j,k}(t) = 10^{\left(\frac{pl_{i,k}(t)}{10}\right)} * 10^{\left(\frac{\xi_i(t)}{10}\right)} * 10^{\left(\frac{mpath_{i,j,k}(t)}{10}\right)} \quad (14)$$

where, $\gamma_{i,j,k}(t)$ is the instantaneous SINR (in dB) of user i on RB j on CC k at time t , $mpath_{i,j,k}(t)$ is the multi-path fading gain (in dB) of user i on RB j on CC k at time t , $pl_{i,k}(t)$ is the path loss (in dB) of user i on CC k at time t , $\xi_i(t)$ is the shadow fading gain (in dB) of user i at time t , P_{total} is the total eNB transmit power (in dBm), RB_{max} is the maximum available number of RBs, N_o is the thermal noise (in watts) and I is the inter-cell interference (in watts). It is assumed that inter-cell interference is constant as only one hexagonal cell considered.

Channel Quality Information (CQI): The instantaneous SINR computed at a user is mapped into a CQI value (referred to as SINR-to-CQI mapping) and the user reports the CQI value to the eNB through the uplink feedback channel. Each CQI value corresponds to the Modulation and Coding Scheme (MCS) for which a Block Error Rate (BLER) shall not exceed 10% threshold (Melfuhrer *et al.*, 2009). The SINR-to-CQI mapping based on the 10% BLER threshold is illustrated in Fig. 2.

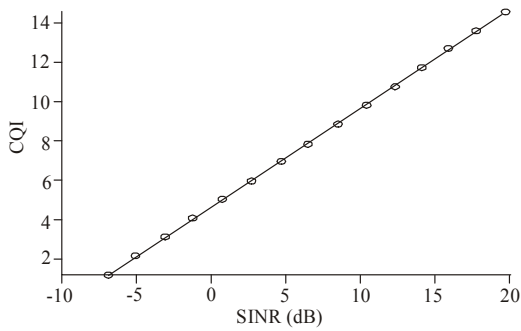


Fig. 2: SINR-to-CQI mapping for 10% BLER threshold

This simulation tool in general assumes that users with active data sessions report their CQI to the eNB on each RB and on each CC. Besides being used for making a scheduling decision, the CQI report is used to determine the data rate of a user for data transmission (i.e., efficiency of each RE). It is also assumed that, only 148 out of 168 REs within an RB are used for downlink data transmission. This is a practical assumption as some REs are used for control and signaling purposes (Holma and Toskala, 2009). In this simulation tool, the instantaneous data rate of user i on RB j on CC k at time t ($r_{i,j,k}(t)$) can be computed as follows:

$$r_{i,j,k}(t) = Efficiency_{i,j,k}(t) * \frac{RE_{data}}{TTI} \quad (15)$$

where,

$Efficiency_{i,j,k}(t)$: The efficiency (in bits/RE) of RB j on CC k of user i at time t

RE_{data} : The total number of REs specified for downlink data transmission

Packet scheduling: The data destined for each user arriving from the core network is stored into its associated buffer at the eNB. These data are segmented into smaller packets of fixed size, time-stamped and queued in the user buffer (at eNB) for transmission based on a first-in-first-out basis. The buffer capacity of each user at the eNB is assumed to be infinite. For each packet in the eNB buffer, its delay is computed. The packet delay is the total waiting time of a packet from the time it arrives at the eNB buffer until current time t . The packet delay is computed only for the packets that are residing within the eNB buffer or transmission buffer. The packets that have been discarded or correctly received at the users are not considered for packet delay computation. A packet is discarded if it has been residing within the eNB buffer for more than a buffer delay threshold.

This simulation tool assumes that the downlink LTE-A contains an equal number of LTE-A and the legacy LTE users (50:50). The LTE-A users can utilize all of the available CCs whereas each LTE user is assigned to a CC according to the Random CC selection

algorithm (as described in System Overview). Packet scheduling takes place after all users have been assigned to a CC/a number of CCs. In each TTI and on each CC, the packet scheduler selects a user with the highest priority to receive its packets on each RB. For each selected user, the packet scheduler always prioritizes retransmission packets ahead of the packets that are waiting for the first transmission.

A group of packets that are transmitted to a user in a TTI is called a Transport Block (TB). Each TB has a unique Transmission Sequence Number (TSN). This TSN is used by the user for in sequence delivery of packets towards the Application Layer (Dongmyoung *et al.*, 2008). Each TB is inserted with Cyclic Redundancy Check (CRC) bits for error-detection. The size of a TB which determines the data rate for packets transmission is dependent upon the MCS on each RB of a CC that is assigned to the user. In each TTI, a user can either receive a TB of first transmission or a retransmitted TB.

All packets are stored into a transmission buffer at the eNB upon transmission (Liu *et al.*, 2010; Shirani-Mehr *et al.*, 2010; Yao-Liang and Zsehong, 2010). The packets of a TB are removed from the transmission buffer when:

- A positive Acknowledgement (ACK) feedback associated with the TB is received
- They have exceeded maximum number of retransmissions or
- An RLC feedback indicating the expiry of re-sequencing timer is received

Note that a user sends an ACK feedback to indicate a successful reception of a TB and a Negative Acknowledgement (NACK) feedback in case of a failure in decoding the TB. In addition to that, all packets of a TB are removed from the transmission buffer if delays of some of these packets exceed the buffer delay threshold.

Hybrid Automatic repeat Request (HARQ): Each TB is encoded prior to (re) transmission (3GPP, 2011a, b). An encoded TB contains systematic bits (i.e., information bits and CRC bits) and parity bits. A user decodes the TB by checking the CRC bits upon reception. The user sends an ACK feedback to the eNB if the TB is correctly decoded. If the decoding fails (i.e., not all errors within the TB are correctable), a NACK feedback indicating that the TB needs to be retransmitted is sent to the eNB. The erroneous TB is stored in the user's buffer and later combined with subsequent retransmission (referred to as Type II HARQ).

This simulation tool considers Chase Combining (Chase, 1985), which is one of the well-known Type II HARQ. Chase Combining is a technique that retransmits an identical TB as in the first transmission. To allow the retransmitted TB to be identical to the first transmission,

Table 1: Video streaming parameters with 512 kbps average data rate

Information types	Distribution and parameters	PDF
Inter-arrival time between the beginning of successive frames (F_a)	Deterministic (based on 20 fps) 50 msec	-
Number of packets (slices) in a frame (N_p)	Deterministic 8 packets	-
Packet (slice) size (S_p)	Truncated Pareto Mean = 200 bytes Maximum = 350 bytes Minimum = 125 bytes	$f_x = \left[\frac{\alpha k^\alpha}{x^{\alpha+1}} \right]$ K = 110 bytes, $\alpha = 1.2$
Inter-arrival time between packets (slices) in a frame (P_a)	Truncated Pareto Mean = 3.65 msec Maximum = 6.25 msec	$f_x = \left[\frac{\alpha k^\alpha}{x^{\alpha+1}} \right]$ K = 2.5 msec, $\alpha = 1.2$

the retransmitted TB has to use the same MCS and the same number of RBs as in the first transmission. The retransmitted TB is later combined with any previously received TB with the same TSN at the user. The HARQ considered in this simulation tool uses a Stop-and-Wait (SAW) protocol. The SAW protocol takes 8 msec duration to complete a cycle. This time duration is used by a user to decode a received TB, perform a CRC, encode and send a HARQ feedback (ACK/NACK) and by the eNB to decode the HARQ feedback, construct and encode a TB (based on the HARQ feedback).

Traffic model: Video streaming is one of the delay-sensitive GBR applications. This application is becoming increasingly popular among the mobile cellular users and a good portion of the radio resources has to be provided by the LTE-A to support for video streaming. As such, this simulation tool assumes that a GBR user with an active data session runs video streaming application. Traffic model based on statistical model of real traffic is used to model the GBR application. Table 1 shows how to produce a video streaming session with an average data rate of 512 kbps. Besides 512 kbps average rate, this simulation tool supports video streaming of 128, 256, 1024 and 2048 kbps average data rates, respectively.

Simulation environment: The performances of the well-known packet scheduling algorithms are evaluated using the simulation tool within a pico cell with 300 m radius. The CQI delay is set to 0 msec, the probability that the CQI report is in error is fixed at 0% and the interval for CQI reporting is set to 1 msec interval. The HARQ feedback is modelled error-free with a 4 msec delay. All packets of an erroneous TB are discarded after they have been retransmitted three times.

The buffer delay threshold of the video streaming application is capped at 80 msec, which is within an appropriate range of the 3GPP recommendation (3GPP, 2009). It is assumed that the video streaming packets are played back while they are streamed over variable bit rate mobile cellular channels (Basukala *et al.*, 2010). In this performance evaluation, minimum user throughput is assumed to be maintained above 469 kbps. This is to allow a user to run 2 min video streaming session without its buffer running dry (if the size of de-jitter

buffer is assumed to be 10 sec when a user starts its session). The delay-sensitive GBR application require the number of discarded packets for delay violations (i.e., Packet Loss Ratio-PLR) to be minimized. As such, the PLR threshold of 10^{-3} (3GPP, 2009) is set for the GBR application.

The performances of the well-known algorithms are evaluated on the basis of PLR and mean user throughput metrics that are defined as follows:

$$PLR = \frac{\sum_{i=1}^N \sum_{t=1}^T p_{discard_i}(t)}{\sum_{i=1}^N \sum_{t=1}^T p_{size_i}(t)} \quad (16)$$

$$mean\ user\ throughput = \frac{1}{N} \frac{1}{T} \sum_{i=1}^N \sum_{t=1}^T prx_i(t) \quad (17)$$

where,

- $p_{discard_i}(t)$: The total size of discarded packets (in bits) of user i at time t
- $p_{size_i}(t)$: The total size of all packets (in bits) arrive into the eNB buffer of user i at time t
- $prx_i(t)$: The total size of correctly-received packets (in bits) of user i at time t
- N : The total number of users
- T : The total simulation time

The downlink LTE-A is an OFDMA based system that performs packet scheduling in time and frequency domains. In this system, there are a number of RBs available to be shared among the competing users. The well-known packet scheduling algorithms discussed in System Overview were developed for the legacy mobile cellular systems that allocate all of the available radio resources to a single user in each scheduling interval. As such, a number of modifications are made to adapt the well-known packet scheduling algorithms into the downlink LTE-A. In this case, in each TTI, on each CC and on each RB, the Max-Rate, PF and M-LWDF algorithms select a user for packet transmission that maximizes $\mu_{i,j,k}(t)$ in Eq. (18), (19) and (20), respectively:

$$\mu_{i,j,k}(t) = r_{i,j,k}(t) \quad (18)$$

$$\mu_{i,j,k}(t) = \frac{r_{i,j,k}(t)}{R_i(t)} \tag{19}$$

$$\mu_{i,j,k}(t) = a_i * W_i(t) * \frac{r_{i,j,k}(t)}{R_i(t)} \tag{20}$$

where,

- $\mu_{i,j,k}(t)$: The priority of user i on RB j on CC k at TTI t
- $r_{i,j,k}(t)$: The instantaneous data rate of user i on RB j on CC k at TTI t
- $R_i(t)$: The average throughput of user i at TTI t
- a_i : The QoS requirement of user i
- $W_i(t)$: The delay of the HOL packet of user i at TTI t

The RR algorithm is modified such that a user that is to receive its packets on each RB and on each CC is selected in a cyclic fashion.

RESULTS AND DISCUSSION

Figure 3 illustrates the PLR performances of the four well-known packet scheduling algorithms with increasing system capacity at 3 km/h user speed. With increasing system capacity, there will be more packets waiting for downlink transmission at the eNB buffers. As there are insufficient radio resources to transmit all the packets, some packets whose delays have reached the buffer delay thresholds are discarded. This situation leads to degradation in the PLR.

To satisfy the QoS requirement of the GBR application, the PLR should always be kept below 10-3 threshold (i.e., QoS constraint of the GBR application). Table 2 shows that the maximum system capacities where the QoS requirement of the GBR application is satisfied in the Max-Rate, PF, M-LWDF and RR are 24, 30, 32 and 23 users, respectively. It can be observed in the table that the maximum system capacity to satisfy the QoS requirement of the GBR application achieves in the M-LWDF algorithm is approximately 39.1% higher when compared with the RR algorithm. The M-LWDF has a significant system capacity improvement as the algorithm takes packet delay into consideration (besides the channel quality and average throughput). On the other hand, the Max-Rate for example, prioritizes users according to their channel quality only. This may lead to the users that are located at the cell edge being deprived from receiving their packets. After some duration, the packets of these users will be discarded for delay violations and hence degrading the PLR in the Max-Rate algorithm. The RR algorithm has the worst PLR performance for blindly allocating the available RBs to the users regardless of their channel quality.

The mean user throughput performances obtained by the four well-known packet scheduling algorithms at 3 km/h user speed are shown in Fig. 4. It can be observed in the figure that both M-LWDF and PF algorithms are capable to maintain the minimum throughput requirement of 469 kbps for a higher system

Table 2: Maximum system capacities to satisfy the QoS requirements of the GBR application at 3 km/h user speed

	Max. system capacity	Improvement over RR (%)
Max-Rate	24	4.3
PF	30	30.4
M-LWDF	32	39.1
RR	23	-

Max.: Maximum

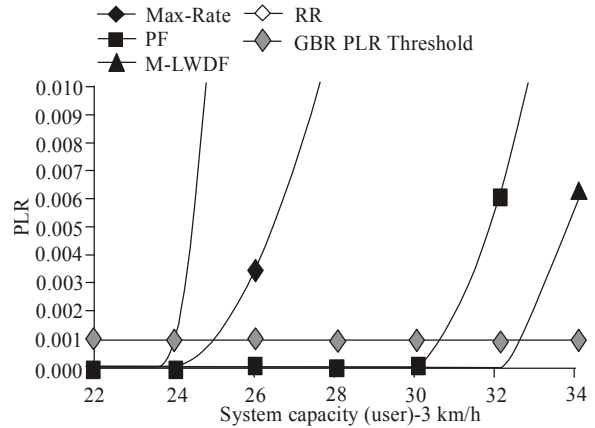


Fig. 3: PLR vs. system capacity at 3 km/h

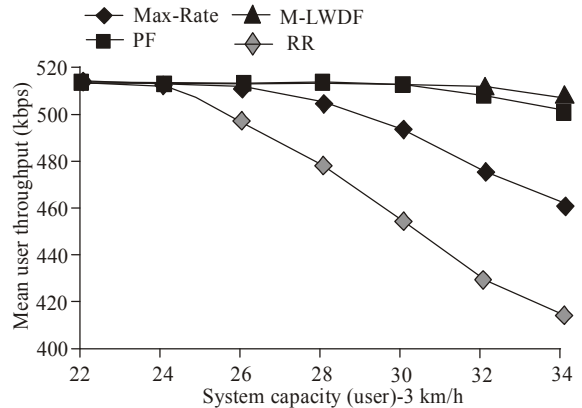


Fig. 4: Mean user throughput vs. system capacity at 3 km/h

capacity (i.e., more than 34 users). On the other hand, if the minimum throughput of 469 kbps is required to be maintained, then the maximum system capacities are limited to 30 and 26 users in the Max-Rate and RR algorithms, respectively.

The PLR and mean user throughput performances of the well-known packet scheduling algorithms at 30 km/h user speed are given in Fig. 5 and 6. When compared with the case of 3 km/h user speed, the PLR and mean user throughput performances significantly degrade at 30 km/h. This can be explained as follows. The CQI experienced by a user when a TB is received and the CQI that was used to determine the MCS for the TB before it was transmitted will be different especially for a higher user speed due to rapid channel variation. This leads to significant degradation of PLR and mean

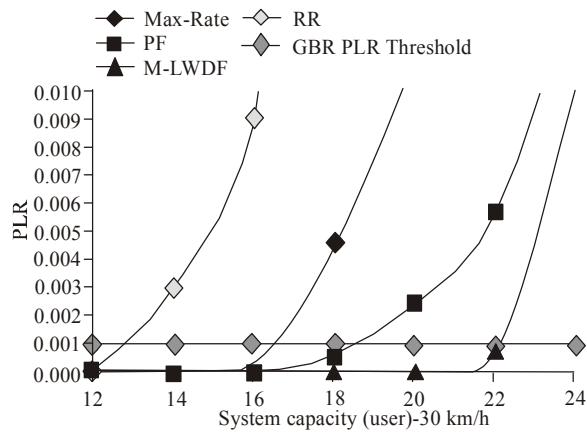


Fig. 5: PLR vs. system capacity at 30 km/h

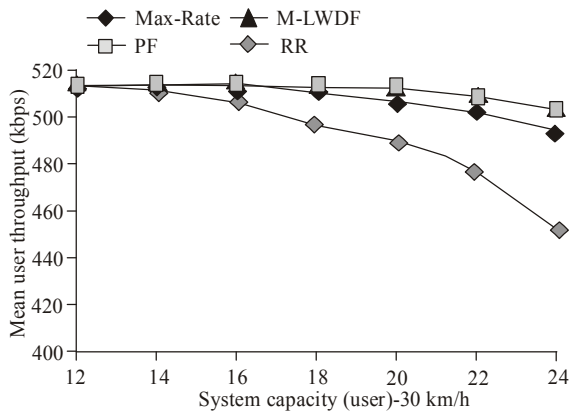


Fig. 6: Mean user throughput vs. system capacity at 30 km/h

Table 3: Maximum system capacities to satisfy the QoS requirements of the GBR application at 30 km/h user speed

	Max. system capacity	Improvement over RR (%)
Max-Rate	16	33.3
PF	18	50.0
M-LWDF	22	83.3
RR	12	-

Max.: Maximum

user throughput when compared to a slowly moving user (i.e., 3 km/h).

If the QoS requirement of the GBR application is to be satisfied at the 30 km/h speed, then the maximum system capacities can be as high as 22 users in the M-LWDF algorithm and limited to 12 users in the RR algorithm (Table 3). This is equivalent to approximately 83.3% improvement in the system capacity achieved in the M-LWDF algorithm over the RR algorithm. Moreover, it can be observed in Fig. 6 that the M-LWDF, PF and Max-Rate are capable of maintaining the minimum throughput requirement of 469 kbps for more users as compared to the RR algorithm which can only support approximately 22 users.

It can be concluded based on this performance study that the M-LWDF is efficient for usage in the

downlink LTE-A. It outperforms other well-known packet scheduling algorithms by optimizing the system capacity without compromising the QoS of delay-sensitive GBR application. The M-LWDF algorithm gives a higher priority for transmission for the packets of a user that has resided the longest in the eNB buffer (if the channel quality and the average throughput of each user is equal). This minimizes the number of discarded packets for delay violation. In addition to that, the results obtained from this performance study are in line with (Ramli *et al.*, 2009b) that showed a similar trend among the four algorithms. Therefore these results validated the simulation tool.

CONCLUSION

A novel C++ simulation tool is presented in this study. This simulation tool accurately models the CQI, packet scheduling, HARQ, carrier aggregation and traffic characteristics of the GBR application. Performance study of a number of well-known packet scheduling algorithms is conducted using the simulation tool. It is demonstrated that the M-LWDF is efficient for usage in the downlink LTE-A for maximizing the system capacity whilst providing satisfactory QoS of the GBR application. With the PLR kept below 10^{-3} practical threshold and the average data rate of all users fixed at 469 kbps, the M-LWDF has approximately 39.1 and 83.3% system capacity improvement over RR algorithm at 3 and 30 km/h user speeds, respectively. Moreover the results of the performance study verified validity of the simulation tool.

This simulation tool provides a platform for the research communities to further enhance the downlink LTE-A through performance evaluations and comparisons of new downlink LTE-A packet scheduling algorithms. This paper is limited to a single cell supporting multiple users with one active GBR application and it does not account the effect of inter-cell interference in the performance evaluation. Additionally, this simulation tool made a number of assumptions due to time limitations as well as to further reduce complexity of the simulation. These limitations will be addressed in the future studies. Moreover, future studies will extend the simulation tool to support packet scheduling with admission control, congestion control and handover such that more multimedia users can be supported and their QoS requirements can be simultaneously satisfied. The performance study of a number of well-known CC selection algorithms in the downlink LTE-A will also be a part of future studies.

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