Delay and Loss Due to Uplink Packet Scheduling in LTE Network

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Abstract. In this paper we describe the packet scheduling process and investigate the reasons limiting the medium access control (MAC) layer capacity of the 3rd Generation Partnership Project Long Term Evolution (3GPP LTE) network. We show that although a scheduling process allows to assign dedicated channels to the users based on their quality of service (QoS) requirements, it introduces the additional delay in the uplink channel. We also show that the scheduling delay may increase significantly if some certain parameters of the system are not set appropriately, and suggest alternative approaches to reduce the scheduling delay in LTE network. Obtained analytical expressions of the packet scheduling delay and loss have been verified using simulation model developed in OPNET platform. Results of this work can be used for resource allocation, packet scheduling and network planning to establish the upper bounds on delay and loss for the users with strict QoS requirements.

Keywords: 3GPP LTE, packet scheduling, performance evaluation.

1 Introduction

Today a 3rd Generation Partnership Project Long Term Evolution (3GPP LTE) is considered to be the main standard for deployment in future wireless networks. In the downlink the LTE system uses Orthogonal Frequency-Division Multiple Access (OFDMA) due to its high spectral efficiency and robustness against interference. In the uplink a Single Carrier Frequency Division Multiple Access (SC-FDMA) is utilized because of its lower Peak-to-Average Power Ratio (PAPR) compared to traditional OFDM [1]. In LTE, available transmission resources are distributed among the users by the medium access control (MAC) schedulers in enhanced NodeBs (eNBs). LTE standardizes control signaling and a general framework on physical and MAC layer. An exact algorithm for resource allocation is not specified: depending on the implementation, it can be based on the queuing delay, instantaneous channel conditions, fairness, etc. Thus, the scheduling process allows assigning dedicated channels to the users based on their quality of service (QoS) requirements [2]. However, it introduces the additional delay in the uplink channel. In some cases (for instance, for real-time applications) such impact on the end-to-end service performance of the

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network cannot be neglected. Therefore, it is very important to analyze the delay due to packet scheduling in LTE system.

The scheduling performance for different types of users has been studied in many papers. Various multiuser scheduling strategies in the context of OFDMA downlink have been described in [3-7]. The uplink capacity of LTE system has been investigated in [8] and [9]. Although these works provide a closer look on the capacity and coverage of LTE network depending on channel conditions, resulting end-to-end performance of wireless communication systems is evaluated only by means of simulations, and no analytical verification of obtained results is conducted. The average values of various delay components including delay due to packet scheduling have been given in [10]. However, no proper mathematical analysis confirming the delay values have been presented.

In this paper we describe the packet scheduling process and investigate the reasons limiting the MAC layer capacity of LTE network. Based on LTE standard specifications, we provide a complete analysis of the delay and loss due to packet scheduling in LTE system. Obtained analytical expressions of the packet scheduling delay and loss are applicable to any resource allocation algorithm, and can be used in the analysis of the end-to-end packet delay for the users of LTE network. The rest of the paper is organized as follows. In Section 2 we provide some background information on the design issues of LTE system related to the packet scheduling process, and derive the expressions for packet scheduling delay and loss estimation. In Section 3 we present the simulation framework conducted in OPNET environment [15] to validate the analytical expressions of the scheduling delay and loss, and discuss the possible ways to reduce the scheduling delay and loss in the network. The conclusions are drawn in Section 4.

2 Packet Scheduling Process

2.1 The General Uplink Packet Scheduling Procedure

In LTE, the downlink transmission scheme is based on conventional OFDM. In an OFDM system the available spectrum is divided into multiple subcarriers, which are modulated independently by a low rate date stream. The main advantages of OFDM are its robustness against multipath fading and efficient receiver architecture. Besides, OFDMA-based channel access supports multiple users on the available bandwidth, because within one transmission time interval (TTI) subcarriers can be allocated to different users. The uplink transmission scheme is based on SC-FDMA, which has better PAPR properties then OFDMA-based signals [2, 11]. The basic radio resource unit in LTE is called a resource block (RB). In frequency domain one RB consists of 12 subcarries with a constant subcarrier spacing Δf = 15 kHz. In time domain it has length equal 1TTI with duration T_s = 1 ms. The number of RBs, denoted N_{RB} depends on the channel bandwidth B. The capacity of one RB depends on the Modulation and Coding Scheme (MCS) which determines the bit rate [12, 13].

In LTE, resources are allocated to user equipments (UEs) for uplink and downlink data transmission in terms of RBs. Thus, one UE can be allocated only the integer number of RBs in frequency domain, and these RBs do not have to be adjacent to each other. Resource allocation (scheduling) is carried by the MAC layer packet scheduler in the eNB both for uplink and downlink transmissions. Resource allocation (scheduling) is usually performed periodically within a fixed time interval, called scheduling period T_{sc}. Depending on its implementation, the scheduler can allocate resources based on the quality of service (QoS) requirements, instantaneous channel conditions, fairness, etc. Besides, the scheduler has to ensure that Hybrid Automatic Repeat Request (HARQ) retransmissions are performed on a timely basis (in LTE system a packet retransmission should be send in exactly 8 ms after receiving a Negative Acknowledgement message) [2, 12, 13].

After resource allocation, the user data are carried by the Physical Uplink Shared Channel (PUSCH) in uplink direction and the Physical Downlink Shared Channel (PDSCH) in downlink direction. The scheduling decisions are carried by the Physical Uplink Control Channel (PUCCH) and Physical Downlink Control Channel (PDCCH) in uplink and downlink directions, respectively [14]. The general scheduling procedure shown on Figure 1 consists of the following steps [10, 13]:

1. First SR-SG exchange step:
   (a) UE generates initial scheduling request (SR) without any scheduling information, and starts the scheduling timer with a timeout T_{sc}.
   (b) If the eNB receives the SR, eNB transmits first scheduling grant (SG) to notify the UE that it is waiting for the buffer status report (BSR) data with scheduling information.
   (c) If the UE does not receive the SG before the scheduling timer expires, it repeats step 1(a).
2. Second SR-SG exchange step:
   (a) UE sends secondary SR with BSR MAC Control Element, and starts the BSR retransmission timer with a timeout T_{BSR}. The BSR MAC Control Element carries the information about the amount of uplink data waiting to be transmitted by UE.
   (b) If the eNB receives the second SR, it allocates the resources to UE and reports about allocation by generating secondary SG. The amount of resources allocated to UE depends on the received BSR information.
   (c) If the UE does not receive the SG before the BSR retransmission timer expires, it repeats step 2(a).
3. Third step:
   (d) After the secondary SG is received, a UE transmits the uplink data.

According to LTE standard, a UE is allowed to transmit at most N_{max} SRs per packet, after which the packet is dropped.

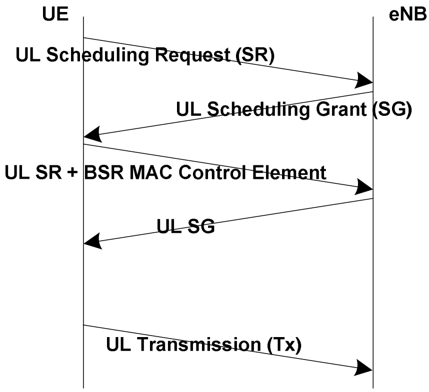

Fig. 1. The general uplink scheduling procedure

2.2 PUCCH and PDCCH Configuration

According to LTE standard, PUCCH carries the following uplink control information: HARQ Acknowledgement (ACK) and Negative Acknowledgement (NACK) information related to data packets received in downlink; scheduling requests (SRs) for packet scheduling; Channel Quality Indicator (CQI) reports for Adaptive Modulation and Coding (AMC); pre-coding matrix information (PCI) and rank indication for Multiple-Input-Multiple-Output (MIMO) [13]. Depending on the type of information, the LTE specifies different PUCCH formats. In particular, SRs are carried by the PUCCH format 1/1a/1b messages. According to the standard, a determined number of RBs N_{RB_1} are reserved for PUCCH format 1/1a/1b transmissions, and each reserved RB is further divided into a number of resource indexes (RIs). One RI is allocated for one PUCCH message. The number of RIs in a RB depends on message format. For format 1/1a/1b messages the direct mapping between PUCCH cyclic shifts and RIs cannot be used because of the block-spreading operation. For this purpose, PUCCH channelization is used to provide a number of parallel sub-channels with adjustable orthogonal properties, which are configured by means of the special system parameter δ (allowed values are δ = 1, 2 or 3). It is periodically broadcasted in the network to maintain the orthogonality of the sub-channels. The number of parallel orthogonal sub-channels (or RIs) per a RB allocated for PUCCH format 1/1a/1b messages N_{RI_1} can be calculated from [10, 11 - 14]:

$$N_{RI\_1} = \frac{12 N_{RS}}{\delta} \tag{1}$$

where N_{RS} is the number of reference signals on PUCCH format 1/1a/1b (N_{RS} = 3 for normal CP, N_{RS} = 2 for extended CP) [19]. Then, the total number of RIs allocated for a format 1/1a/1b messages N_{PUCCH_1} is equal [10]:

$$N_{PUCCH\_1} = N_{RB\_1} N_{RI\_1} = \frac{12 N_{RB\_1} N_{RS}}{\delta} \tag{2}$$

For instance, for normal CP with N_{RB_1} = 1 RB the allowed numbers of allocated channels are N_{PUCCH_1} = 12, 18 or 36 channels.

PDCCH is primarily used to carry the Downlink Control Information (DCI), such as number of OFDM symbols reserved within each time slot for PDCCH signals; scheduling information (downlink assignments and uplink SGs); HARQ and MCS. According to LTE specifications, PDCCH occupies the first 1, 2, or 3 OFDM symbols in a time slot extending over the entire system bandwidth. PDCCH is constructed from Control Channel Elements (CCEs). The number of CCEs N_{CCE} indicates the capacity of PDCCH: each UE generating a DCI message within a considered time slot is assigned a CCE (N_{CCE} for 5, 10 and 20 MHz bandwidth is given in Table 1) [14].

Table 1. The Number of CCEs for 5, 10, 20 MHz Bandwidth [14]

The number of CCEs per time slot, N_{CCE}	**Bandwidth, *B* (MHz)**		
	5	**10**	**20**
1 PDCCH symbol per time slot	3	8	17
2 PDCCH symbols per time slot	12	25	50
3 PDCCH symbols per time slot	20	41	84

2.3 Delay and Loss Due to Uplink Packet Scheduling

Delay and loss associated with the transmission of an SR or an SG depends on how fast the UE and eNB are able to get through the first and second steps. In particular, the delay associated with the transmission of one SR consists of the transmission, buffering, propagation and processing delays in the uplink direction. Similarly, the delay associated with the transmission of one SG comprises the transmission, buffering, propagation and processing delays in the downlink direction. For further analysis we assume, that the processing time for an SR/SG is equal to one TTI with duration T_s = 1 ms. The transmission, buffering and propagation delays for an SR/SG are in order of 1 μs, which is very small compared to T_s. Thus, the delay associated with the transmission of an SR/SG can be accurately estimated by T_s.

Now we are ready to state the main results of this paper, that is, to estimate the mean packet delay and loss due to uplink packet scheduling. In our analysis we consider the basic LTE network comprising one eNB and n active UEs sending SRs to eNB independently in a random fashion. The time interval between two consecutive SRs generated by a particular UE is fixed and equal T_{sc}. The eNB responds to each successfully received SR by sending an SG to UE.

Theorem 1: The probability of success for an SR-SG exchange attempt between the UE and the eNB in such network can be estimated by

$$p_s = \begin{cases} 1, \text{ if } n \le \frac{T_{SR}}{T_s} \times \min\{N_{PUCCH\_1}, N_{CCE}\} \\ \frac{T_{SR}}{T_s} \times \frac{\min\{N_{PUCCH\_1}, N_{CCE}\}}{n}, \text{ otherwise} \end{cases} \tag{3}$$

Proof: See Appendix A.

Lemma 1: If the probability of success for a particular SR-SG exchange attempt between the UE and the eNB is given by p_s, then the probability that a packet is lost in the scheduling process is

$$P_{sc} = (1-p_s)^{N_{\max}} + N_{\max} p_s (1-p_s)^{N_{\max}-1} \tag{4}$$

Proof: A packet is lost in the scheduling process if there is at most one success in N_{max} consecutive SR/SG scheduling transmission attempts, probability of which is given by (6) using the well-known results on repeated trials.

Theorem 2: The packets which are successfully scheduled experience an average uplink packet scheduling delay

$$\begin{gathered} T_{PS} = 4T_s + \frac{1}{A}(BT_{sc} + CT_{BSR}), \\ \text{where: } A = 1 - q_s^{N_{\max}-2}(1 - N_{\max} p_s), \quad B = q_s\left[1 - q_s^{N_{\max}-3}\left\{2N_{\max} - 3 + 2p_s + (N_{\max}-2)p_s^2\right\}\right], \\ C = \frac{2 - q_s^{N_{\max}-2}\left\{2 + 2(N_{\max}-2)p_s + (N_{\max}-2)(N_{\max}-3)p_s^2\right\}}{2p_s}, \quad q_s = 1 - p_s. \end{gathered} \tag{5}$$

Proof: See Appendix B.

3 Simulation Framework

In this section we provide the numerical validation of the derived analytical expressions for the packet scheduling delay and loss given by (3) - (5) by comparing the values of the mean packet scheduling delay and loss obtained analytically with the results gathered in simulations. The simulation model of the network (show on Figure 2) has been developed according to requirements of LTE standard [13, 14] using OPNET platform [15]. The users generate a mixed traffic comprising of voice, video and data applications in proportion 1:1:2, respectively. The user traffic is simulated in accordance with the requirement stated in [16]. Voice traffic is generated by using the G.723.1 (12.2 Kbps) codec with a voice payload size 40 bytes and a voice payload interval 30 ms. Each voice user might be either in active (talk-spurts period) or inactive (silent period) state. The durations of the talk-spurts and silent periods are exponentially distributed with 0.65s and 0.352s means, respectively. Video services are simulated using a high resolution video model with a constant frame size equal 6250 bytes and exponentially distributed frame inter-arrival intervals (with mean equal

0.5s). Data users in simulations are HTTP1.1 users generating pages or images with exponential page inter-arrival intervals (mean equal 60s). It is assumed that one page consists of one object, whereas one image consists of five objects. The object size is constant and equal 1000 bytes. Simulations have been carried for the networks with B = 5 MHz and B = 10 MHz bandwidth. The other parameters necessary to estimate the packet scheduling delay and loss are listed in Table 2.

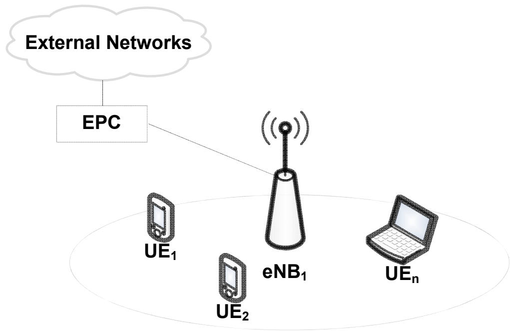

Fig. 2. The simulation model of the network

Table 2. Common Simulation Parameters

Parameter	Value
Bandwidth, B	5MHz, 10MHz
Cyclic Prefix Type	Normal
PDCCH symbols per subframe	3
PUCCH Reserved Size for format 1 messages, N_{RB_1}	1 RB
Cyclic shift, δ	3
TTI duration, T_s	1 ms
Scheduling Timer timeout, T_{sc}	1 ms
BSR Retransmission Timer timeout, T_{BSR}	256 ms
Maximal number of SRs per packet, N_{max}	10

Figures 3, 4 show the mean packet scheduling delay and loss estimated analytically and obtained using simulations (experimentally) for the networks with B = 5 MHz and B = 10 MHz bandwidth, respectively. Results on Figure 3 show that in the network with B = 5 MHz the scheduling delay and loss start increasing when the number of users n > 20 users. In the netowork with B = 10 MHz the scheduling delay and loss stat growing when n > 35 users. Obtained results correspond to expressions (3) - (5). In particular, for $n \le T_{SR}/T_s \times \min\{N_{PUCCH\_1}, N_{CCE}\}$ UEs the probability of the success for SR-SG exchange $p_s = 1$ and the values of the scheduling delay and loss are equal $T_{PS} = 4T_s = 4$ ms and $P_{PS} = 0$, respectively. Starting from the point $n = T_{SR}/T_s \times \min\{N_{PUCCH\_1}, N_{CCE}\}$ UEs the probability of success for SR-SG exchange p_s start to decrease, and consequently the delay and loss due to packet scheduling T_{PS} and P_{PS} start growing.

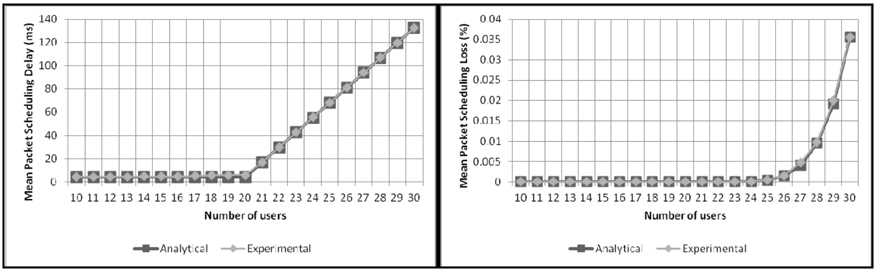

Fig. 3. Analytical and experimental values of the scheduling delay and loss for B = 5 MHz

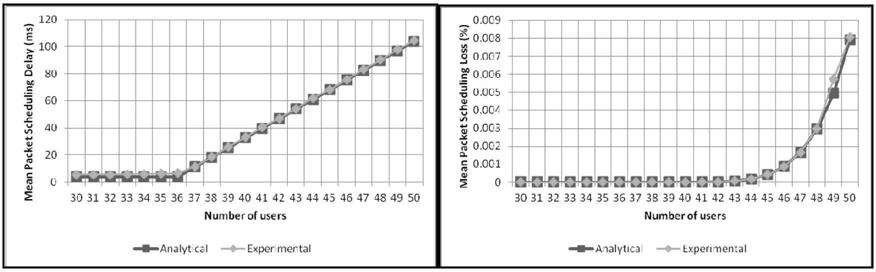

Fig. 4. Analytical and experimental values of the scheduling delay and loss for B = 10 MHz

To finalize this section, we note that the scheduling delay and loss cannot be neglected if the number of active users of the network exceeds the number of available PUCCH parallel subchannels N_{PUCCH_1} or the number of PDCCH control channel elements N_{CCE}. However, as it follows from expressions established by the Theorems 1 and 2, there exist a number of ways to reduce the delay due to packet scheduling. The easiest and most obvious way is to increase N_{PUCCH_1} (by increasing the number of RBs reserved for PUCCH) and N_{CCE} (by increasing the number of PDCCH symbols per subframe). In this case, however, the number of RBs reserved for data will decrease, which may eventually lead to overall QoS degradation.

An alternative strategy to reduce the expected delay and loss in the network would be to decrease the BSR retransmission timer timeout T_{BSR} (in LTE, T_{sc} is in orders of T_s, and therefore its impact on the scheduling delay is neglectably small). To validate this claim, a number of simulations have been conducted in the network with B = 5MHz bandwidth and different BSR timeouts T_{BSR} = 2520ms and T_{BSR} = 320ms. Figure 5 illustrate the packet end-to-end delay and loss for the network users. Obtained results confirm our expectations – delay and loss decreases in times, which can be explained by faster network reaction on resource re-allocation during retransmissions. Therefore, we recommend to set the BSR retransmission timer timeout T_{BSR} as small as possible to minimize the scheduling delay (which is especially important in case of delay-sensitive applications, such as voice or video). Note, that in order to allow the serving eNB to receive and send the BSR MAC Control Element, as well as to re-allocate the resources, the BSR retransmission timer should not be less than the maximal packet round trip time. Thus, the maximal packet round trip time will serve as the lower bound for BSR retransmission timer timeout T_{BSR} in LTE system.

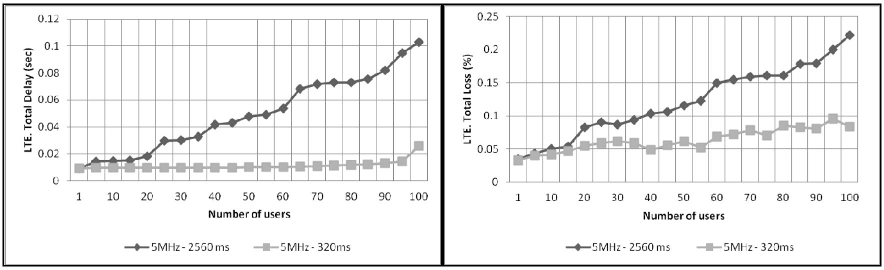

Fig. 5. Packet end-to-end delay and loss in the network with $B = 5$ MHz

4 Conclusions

This paper presents the analytical method to estimate the delay and loss due to packet scheduling in LTE network. The expressions derived in the paper can be used for resource allocation, packet scheduling and network planning. Simulation results have shown that the theoretically obtained values of the scheduling delay and loss closely follow the actual values. It has also been found, that the values of the scheduling delay and loss grow rapidly when the number of users in the network exceeds some certain bounds. To prevent this multiplicative growth of delay, a couple of alternative strategies have been proposed and discussed in the paper.

Appendix A: Proof of Theorem 1

In LTE standard, SRs are carried via PUCCH in format 1/1a/1b messages, SGs are carried via PDCCH in DCI messages (see description of PUCCH and PDCCH provided in Section II.D). This means that each TTI there are exactly N_{PUCCH_1} parallel sub-channels available for all SRs generated by UEs to eNB in uplink direction, and NCCE control channel elements reserved for all SGs generated by eNB to UEs in downlink direction. A particular SR-SG exchange between the UE and the eNB is successful if both SR and SG are received successfully by the eNB and the UE, respectively. Let p_s^{SR} be the probability of a successful reception of an SR from UE by eNB, and p_s^{SG} be the probability of a successful reception of an SG from eNB by UE. Then, the probability of a successful SR-SG exchange between the UE and the eNB ps is equal to the product of both of these probabilities

$$p_s = p_s^{SR} p_s^{SG} \tag{6}$$

If N_{SR} is the number of SRs generated within one TTI, then it is readily verified that

$$p_s^{SR} = \begin{cases} 1, \text{if } N_{SR} \le N_{PUCCH\_1} \\ \dfrac{N_{PUCCH\_1}}{N_{SR}}, \text{otherwise} \end{cases} \tag{7}$$

or:

$$p_s^{SR} = \frac{\min\{N_{SR}, N_{PUCCH\_1}\}}{N_{SR}} \tag{8}$$

Let N_{SG} be the number of SGs generated within one TTI. The eNB can receive at most N_{PUCCH_1} SRs from the UEs, and can respond to at most N_{CCE} users. This means that

$$N_{SG} = \min\{N_{SR}, N_{PUCCH\_1}\} \tag{9}$$

$$p_s^{SG} = \begin{cases} 1, \text{if } N_{SG} \le N_{CCE} \\ \dfrac{N_{CCE}}{N_{SG}}, \text{otherwise} \end{cases} \tag{10}$$

or:

$$p_s^{SG} = \frac{\min\{N_{SG}, N_{CCE}\}}{N_{SG}} \tag{11}$$

Combining (8), (9) and (11) we get:

$$p_s = \frac{\min\{N_{SR}, N_{PUCCH\_1}, N_{CCE}\}}{N_{SR}} \tag{12}$$

or:

$$p_s = \begin{cases} 1, \text{if } N_{SR} \le \min\{N_{PUCCH\_1}, N_{CCE}\} \\ \dfrac{\min\{N_{PUCCH\_1}, N_{CCE}\}}{N_{SR}}, \text{otherwise} \end{cases} \tag{13}$$

In the considered scenario the eNB serves n active users, sending SRs to eNB periodically within the interval T_{sc}. Let p_{SR} be the probability that a single UE will generate SR within a particular TTI. It readily follows that

$$p_{SR} = \frac{T_s}{T_{sc}} \tag{14}$$

Taking into account that UEs generate the SR randomly and independently, the probability that k of the n UEs will generate the SRs within any particular TTI denoted via $P_{SR}(k)$ is binomially distributed, i.e.:

$$P_{SR}(k) = \binom{n}{k} p_{SR}^k (1 - p_{SR})^{n-k} \tag{15}$$

where:

$$\binom{n}{k} = \frac{n!}{k!(n-k)!} \tag{16}$$

Then, the mean number of SRs generated in the network is equal

$$N_{SR} = np_{SR} \tag{17}$$

Appendix B: Proof of Theorem 2

Suppose the first SG is received after n_1 SR attempts, the second SG is received after n_1 SR attempts. Note that n_1 and n_2 are random variables. It is straightforward to check that

$$\Pr\{n_1 = i, n_2 = j\} = p_s^2 q_s^{i-1} q_s^{j-1} \tag{18}$$

Let S denote that event that the scheduling is done successfully. Hence

$$\Pr\{S\} = \mathrm{prob}\{n_1 + n_2 \le N_{\max}\} = \sum_{i=1}^{N_{\max}-1} \sum_{j=1}^{N_{\max}-i-1} p_s^2 q_s^{i-1} q_s^{j-1} \tag{19}$$

Using Bayes' theorem

$$\Pr\{n_1 = i, n_2 = j \mid S\} = \frac{p_s^2 q_s^{i-1} q_s^{j-1}}{\sum_{i=1}^{N_{\max}-1} \sum_{j=1}^{N_{\max}-i-1} p_s^2 q_s^{i-1} q_s^{j-1}} \tag{20}$$

Let the time needed to obtain the first SG be t_1, and the time needed to obtain the second SG be t_2. Then

$$t_1 = 2T_s + (n_1 - 1)T_{sc} \tag{21}$$

because, the waiting time for n_1–1 unsuccessful attempts is $(n_1-1)T_{sc}$, while the total time needed for successful transmission of SR and SG is $2T_s$ (a T_s to transmit SR and another T_s to transmit SG). Similarly,

$$t_2 = 2T_s + (n_2 - 1)T_{BSR} \tag{22}$$

as the waiting time for n_2–1 unsuccessful attempts is $(n_2-1)T_{BSR}$, while the time needed for successful transmission is $2T_s$. Hence the total scheduling delay experienced by the packet is given by

$$t_1 + t_2 = 4T_s + (n_1 - 1)T_{sc} + (n_2 - 1)T_{BSR} \tag{23}$$

Since n_1 and n_2 are random variables, so are t_1 and t_2. Now

$$\begin{aligned} T_{PS} &= E\{t_1 + t_2 \mid S\} = \sum_{i=1}^{N_{\max}-1} \sum_{j=1}^{N_{\max}-i-1} \Pr\{n_1 = i, n_2 = j \mid S\}\left[4T_s + (i-1)T_{sc} + (j-1)T_{BSR}\right] \\ &= \frac{\sum_{i=1}^{N_{\max}-1} \sum_{j=1}^{N_{\max}-i-1} p_s^2 q_s^{i-1} q_s^{j-1}\left[4T_s + (i-1)T_{sc} + (j-1)T_{BSR}\right]}{\sum_{i=1}^{N_{\max}-1} \sum_{j=1}^{N_{\max}-i-1} p_s^2 q_s^{i-1} q_s^{j-1}} \end{aligned} \tag{24}$$

After some modifications to (24) we get the expression given by (5).

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