Wireless Pers Commun (2015) 83:841–854
DOI 10.1007/s11277-015-2429-8

Performance Analysis of Service Based Scheduler in LTE OFDMA System

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Published online: 19 February 2015
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Abstract Long Term Evolution (LTE) is the wireless broadband technology developed by Third Generation Partnership Project (3GPP) to support higher data rate. Resource scheduling in LTE plays an important role in deciding the overall network performance. In this paper, a new scheduling approach called Service Based Scheduler (SBS) is proposed that is suitable for real time services. In this method, two tag values called maximum tag value (T_{max}) and minimum tag value (T_{min}) are estimated for the packets based on the type of bearers. T_{max} is calculated for Guaranteed Bit Rate (GBR) bearers and T_{min} is calculated for non GBR bearers. Then, separate ascending sorted lists are formed for T_{max} and T_{min}. The packets from the T_{max} list are allocated first in spite of users' channel condition. After the allocation of packets from T_{max} list, if the current Transmit Time Interval (TTI) has more resources, then packets from T_{min} list are allocated strictly based on users' channel conditions. The proposed scheduling approach provides better performance for real time services compared to other schemes like Proportional Fairness (PF), Modified Largest Weighted Delay First (MLWDF), Exponential rule (EXP rule) and Logarithmic rule (LOG rule).

Keywords LTE · Resource allocation · Scheduling · Tag values

1 Introduction

Over the past decade, there is considerable evolution in the mobile networks. Earlier, the mobile networks offer single voice service. With the introduction of highly developed mobile devices, the networks offer more advanced multimedia application services that

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result in increasing data traffic. Long Term Evolution (LTE) was introduced by Third Generation Partnership Project (3GPP) to meet the above requirements [1]. LTE network aims in enhancing the data rate to allocate resources to all active users with satisfied level of Quality of Service (QoS) guarantees. To achieve desired data rate, LTE system exploits Orthogonal Frequency Division Multiplexing (OFDMA) in Downlink (DL) and Single Carrier Frequency Division Multiplexing (SC-FDMA) in Uplink (UL). The peak data rate, which is a vital parameter by which different technologies are compared, depends on the usage of spectrum. LTE supports 300 Mbps and 75 Mbps peak data rate in DL and UL at 20 MHz bandwidth, respectively. Since LTE is fully Internet Protocol (IP) based architecture, QoS mechanisms become important as soon as real time services are deployed over the network [2]. The voice traffic is transmitted as Voice over IP (VoIP) and the QoS of voice traffic is decided by its delay and packet loss tolerance. Different applications have different QoS requirements. These requirements put challenge in resource allocation of real time services.

In OFDMA, the bandwidth is divided into number of subcarriers that can be allocated to the user, based on their QoS requirements. So designing an efficient scheduler is significant for effective use of existing radio resources to meet the system performance targets. In this paper, the main focus is on design of scheduling algorithm satisfying the requirements of real time applications. Though many algorithms are proposed for wireless networks, they cannot be directly applied to LTE network. In the proposed method, the packets are scheduled based on the tag values. Each packet in the queue is associated with either maximum tag value (T_{max}) or minimum tag value (T_{min}) value that is estimated according to the type of bearers. T_{max} is calculated for Guaranteed Bit Rate (GBR) bearers and T_{min} is calculated for non GBR bearers. This value is sorted in ascending order in two separate lists. The packet from the T_{max} list are scheduled first and then the packets from the T_{min} list are scheduled. Channel Quality indicator (CQI) value of each user is used to choose the appropriate Modulation and Coding Scheme (MCS).

The paper is organized as follows: Downlink system model in LTE network is described in Sect. 2. In Sect. 3, the proposed scheduler is explained in detail. Section 4 gives the simulation results and conclusion is drawn in Sect. 5.

2 Downlink System Model

LTE network adopts OFDMA based air interface for DL transmission. Here, the whole bandwidth is divided into number of orthogonal subcarriers. A Resource Block (RB) is the smallest unit of allocation in LTE which can be modulated independently [3, 4]. Each RB consists of 12 consecutive subcarriers in frequency domain and 0.5 ms in time domain. The scheduling period in LTE involves two RBs and is termed as one Transmit Time Interval (TTI).

The packet scheduler determines the User Equipment (UE) and the number of RBs to be allocated by means of two factors: the channel condition and the QoS requirement. The base station of LTE called Evolved Node B (eNB) determines the channel condition derived from the CQI reports from the UE. Each UE calculates its own Signal-to-Noise Ratio (SNR) based on its experienced channel condition. SNR value is mapped into CQI value and is reported to eNB through Physical Uplink Control Channel (PUCCH). SNR to CQI mapping is presented in [5]. According to the CQI report corresponding MCS is selected by eNB. Many CQI methods are proposed in [6]. Another function of the eNB with respect to scheduling is to ensure that the necessary QoS should be met for a bearer.

Each bearer has an associated QoS Class Identifier (QCI) and each QoS class is characterized by priority, packet delay and tolerable packet loss [7]. Generally bearers can be classified into two categories based on the nature of the QoS they provide: GBR bearers, which are real time bearers and non GBR bearers, which are non real time bearers.

Packet scheduling for each UE is done based on comparison of per-RB metric. The detailed key factors in designing an algorithm are given in [8]. Many scheduling methods are studied broadly in the literature [9–22]. Some algorithms focus on providing fairness [9, 12, 13, 17], while some focus on increasing spectral efficiency [10, 19, 22].

In this paper four existing algorithms are investigated for comparison. Algorithms include Proportional Fairness (PF), Modified Largest Weighted Delay First (M-LWDF), Exponential rule (EXP rule), and Logarithmic rule (LOG rule) algorithms. PF algorithm aims in maximizing fairness of each user. M-LWDF algorithm is designed for satisfying real time services. EXP rule and LOG rule algorithms are designed to schedule efficiently in overload states. Simulation results show that the proposed scheduling method can achieve better performance than other schemes.

2.1 Proportional Fairness Algorithm

PF algorithm balances spectral efficiency and fairness among UEs. The priority metric used for PF algorithm is

$$p_i^k = \arg\max_i \left(\frac{d_k^i(t)}{\overline{R_i(t-1)}} \right) \tag{1}$$

where $d_k^i(t)$ is the expected data rate for *ith* user at time t on *kth* RB and $\overline{R_i(t-1)}$ is the past average throughput up to time slots $t-1$

2.2 Modified Largest Weighted Delay First Algorithm

M-LWDF algorithm uses Head-of-Line (HOL) delay value and delay based weighting parameter for scheduling.

$$p_i^k = \arg\max_i \left(\alpha_i D_{HOL,i} \cdot \frac{d_k^i(t)}{\overline{R_i}(t-1)} \right) \tag{2}$$

where α_i is a weighting parameter, $D_{HOL,i}$ is the head-of-line delay, i.e., delay of the first packet to be transmitted by *ith* user, $d_k^i(t)$ is the expected data rate for *ith* user at time t on *kth* RB and $\overline{R_i(t-1)}$ is the past average throughput up to time slots $t-1$.

2.3 Exponential Rule Algorithm

EXP rule considers normalized experienced delay of all UEs as a scheduling parameter.

$$p_i^k = \arg\max_i \left(b_i \exp\left(\frac{a_i D_{HOL,i}}{c + \sqrt{(1/N_{rt}) \sum_j a_i D_{HOL,j}}} \right) \cdot \Gamma_k^i \right) \tag{3}$$

where a_i, b_i and c are optimal parameter set for the system requirements, and N_{rt} is the number of active real time DL flows, $D_{HOL,i}$ is the HoL delay, Γ_k^i is the spectral efficiency of ith user on kth RB.

2.4 Logarithmic Rule Algorithm

LOG rule considers logarithmic value of the delay parameter.

$$p_i^k = \arg\max_i \left(b_i \log(c + a_i D_{HOL,i}) \cdot \Gamma_k^i\right) \tag{4}$$

3 Proposed Service Based Scheduler

Nowadays multimedia services like VoIP and video are becoming most important applications in telecommunication technology. Scheduling methods specifically designed for VoIP are studied in [19, 23–25] and for video are studied in [19, 26]. The main aim of the Service Based Scheduler (SBS) is to satisfy the QoS requirements of real time services. The basic mechanism of SBS is shown in Fig. 1.

3.1 Tag Value Calculation

The proposed SBS algorithm is a modification work of Weighted Fair Queuing (WFQ) algorithm [27]. In WFQ scheme, the scheduler schedules the packet according to the finish time of the packet. The finish time depends on the entry time of the packet, its size and its service rate. By using the similar approach, T_{max} and T_{min} values are calculated. But instead of entry time of the packet its theoretical departure time (T_d) value is used. This value is the time difference between the packet's delay budget parameter and its head-of-line delay value for a particular service. It is the time when the grant request can be served without violating the service contract.

Since packets are placed in a queue according to the type of service, resource allocation should be done in such a way that they can meet their QoS requirements. The role of the packet classifier is to classify packets according to the type of service and put separately in a queue. The SBS algorithm estimates T_{max} and T_{min} and put it in the separate lists. The lists are sorted ascending by means of any sorting algorithm and allocation is done first from T_{max} list and then from T_{min} list. T_{max} and T_{min} lists are with respect to the type of bearers.

T_{max} value is decided by T_d, its packet size and minimum guaranteed bit rate for GBR bearers and T_{min} value is related to T_d, its packet size, maximum bit rate and channel condition for non-GBR bearer. The values of T_{max} and T_{min} in the lists are considered as

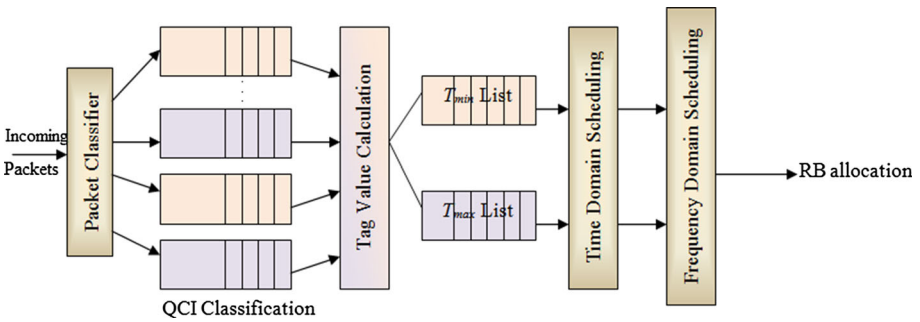


Fig. 1 Basic mechanism of service based scheduler

Table 1 Considered QCI classes and their characteristics

QCI	Resource type	Packet delay budget (ms)	Packet size	Example services
1	GBR	100	small	Conversational voice
2	GBR	150	Medium–large	Conversational video
8	Non-GBR	300	Large	www, email

integer value instead of time. In 3GPP specifications four QCI classes are dedicated to GBR bearers and five QCI classes are dedicated to non-GBR bearers [7]. In this paper, two GBR bearers and one non-GBR bearer are considered for simulation. This can be extended to all QCI classes. The QCI classes considered and their characteristics are listed in Table 1.

Multimedia applications like VoIP and video are transmitted through GBR bearer. The services considered for simulation of proposed algorithm are VoIP, video and Best Effort (BE) services. VoIP traffic has small sized packets (e.g., 40–500 bytes) transmitted at constant rate (e.g., 8–15 kbps). It has strict requirements on packet delay and specified as high priority traffic in 3GPP specifications. Video traffic typically has medium to large sized packets (e.g., 500–1500 bytes) transmitted at variable rate (e.g., 128–768 kbps) with requirements of low delay and less packet loss. Packets of BE traffic are bursty in nature (e.g., 1600 bytes) and are transmitted at variable bit rate [28]. For GBR bearer T_{max} value for *kth* packet is calculated as follows:

$$T^k_{\max} = T_d + \frac{L^k_i}{r_{i(\min)}} \tag{5}$$

where T_d is the theoretical departure time that is calculated according to the service requirements, L^k_i is the size of the *kth* packet in queue i and $r_{i(\min)}$ is the minimum guaranteed bit rate for that service. For non-GBR bearer T_{min} value for *kth* packet is calculated as follows:

$$T^k_{\min} = T_d + \frac{L^k_i}{r_{i(\max)} \times N_b} \tag{6}$$

where T_d is the theoretical departure time that is calculated according to the service requirements, L^k_i is the size of the *kth* packet in queue i, $r_{i(\max)}$is the maximum bit rate for that service and N_b is the number of bits per symbol that is decided based on the channel condition of that user. Since T_{min} packets are allocated depending on the resource available in current TTI, the packets with good channel conditions is strictly chosen for allocation.

3.2 Resource Allocation

The main objective of SBS algorithm is to provide resources first to GBR bearers because of their highest service requirements. The values of T_{max} and T_{min} are considered as weighting factor to each packet by which they are prioritized. The packets from T_{max} list are chosen for allocation first. Based on the CQI reports, RBs are assigned for the packets in T_{max} list. If the current TTI has more resources, the packets from T_{min} list are selected, but the packets with good channel conditions are prioritized i.e., while allocating packets

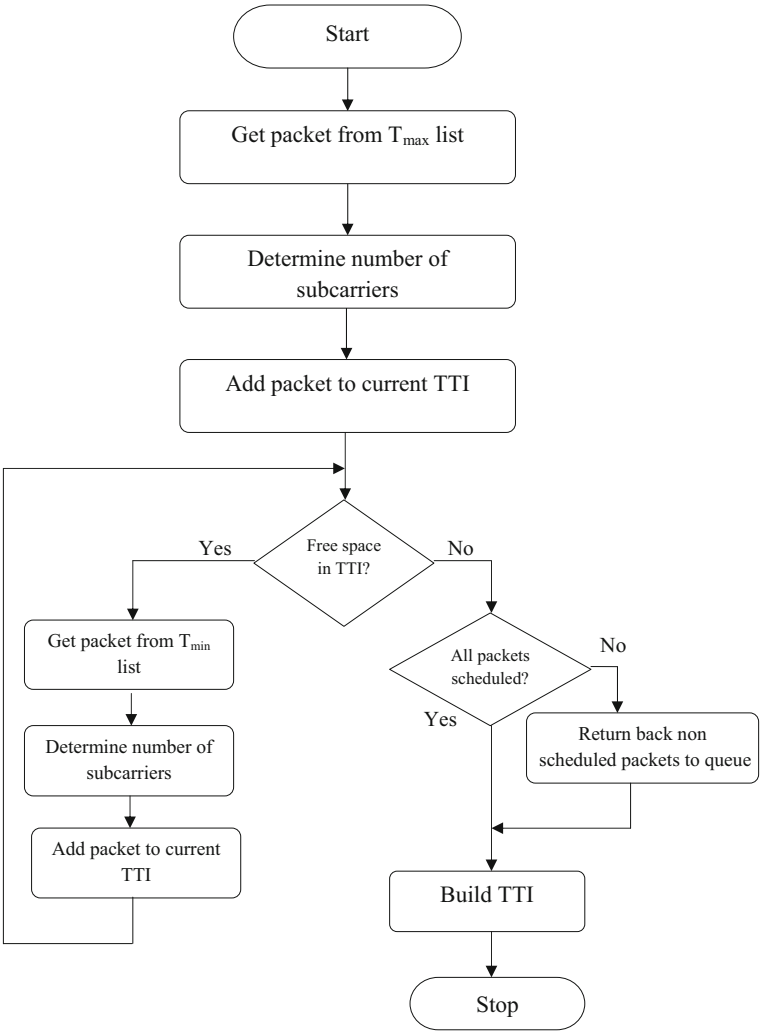


Fig. 2 Flowchart of SBS algorithm

from T_{min} list, packets of non real time service having good channel condition will be allocated first. The flowchart for SBS algorithm is shown in Fig. 2.

3.3 Complexity Analysis

Here, the complexity of the algorithm is analyzed based on its allocation time rather than its computational time. PF algorithm calculates the ratio of current data rate to the mean data rate and allocates resources to the best metric user. Since the PF scheduler has to select among n users, its allocation complexity is $O\ (log\ n)$. M-LWDF algorithm includes a weighting parameter to the PF metric. Though its computational time is different to that of PF, it also has to select among n users. So it has the allocation complexity of $O\ (log\ n)$. In a similar manner, EXP rule and LOG rule algorithm computes exponential and logarithmic

value of delay parameter, respectively for calculating priority metric. So, both these algorithms have allocation complexity of $O\ (log\ n)$. In proposed SBS algorithm, resources are allocated according to the T_{max} and T_{min} value. The allocation time complexity of the proposed SBS algorithm depends on the sorting algorithm used. If simple insertion sorting algorithm is used then the complexity of whole scheduling algorithm is $O\ (n^2)$. If sorting algorithms like merge sort, quick sort are used then the complexity of the scheduling algorithm is $O\ (n\ log\ n)$. These sorting algorithms are more efficient if a number of elements to be sorted are higher [29].

4 Simulation Results

In order to deliver the expected Quality of Experience (QoE) to the end user, the important characteristic to be analyzed is its QoS. It is examined by means of many parameters such as goodput, packet delay, Packet Loss Rate (PLR), fairness index etc. In this section the proposed algorithm is analyzed by using an open source simulator, LTEsim [30]. The modules present in LTEsim are explained in [31]. To make comparison, the proposed method is simulated and compared with other well known scheduling schemes like PF, M-LWDF, EXP rule and LOG rule.

4.1 Simulation Scenario

A single cell scenario is considered with fixed eNB where the users are uniformly distributed among the cell. Users are moving with a speed of 30 kmph. When they reach cell boundary a new direction is taken towards eNB. It is assumed that at a particular time, 40 % of users receive VoIP flow, 40 % of users receive video flow and 20 % of users receive BE service. The buffer at eNB is considered to be infinite. Video and VoIP flows that are inbuilt in LTEsim are used for simulation. Video flows have source rate of 128 kbps, are produced by H.264 trace based coder. VoIP flows have source rate of 8 kbps, are generated by G.729 audio coder. Finally, BE traffic are generated by ideal source that

Table 2 Simulation parameters

Parameter	Value
Bandwidth	10 MHz
Carrier frequency	2 GHz
Frame structure	FDD
Channel model	6-Ray Typical Urban (TU)
Path loss	128.1 + 37.6 log r (dB)
Penetration loss	20 dB
Mobility model	Random
Speed of UE	30 kmph
UE application flow	40 % video, 40 % VoIP, 20 % BE
Maximum delay	0.1 s
Simulation duration	100 s
Traffic model	Video—trace based (H264)
	VoIP—G.729 voice flows
	BE—infinite buffer

always has packet to send. The delay bound is set to 0.1 s. The main simulation parameters used for analysis are shown in Table 2.

4.2 Results and Discussion

The main objective of the proposed SBS algorithm is to improve the performance of resource allocation for real time services. As a result of simulation, the performance analysis is carried out in terms of average goodput, delay, PLR and fairness index by increasing the number of users with the step of ten.

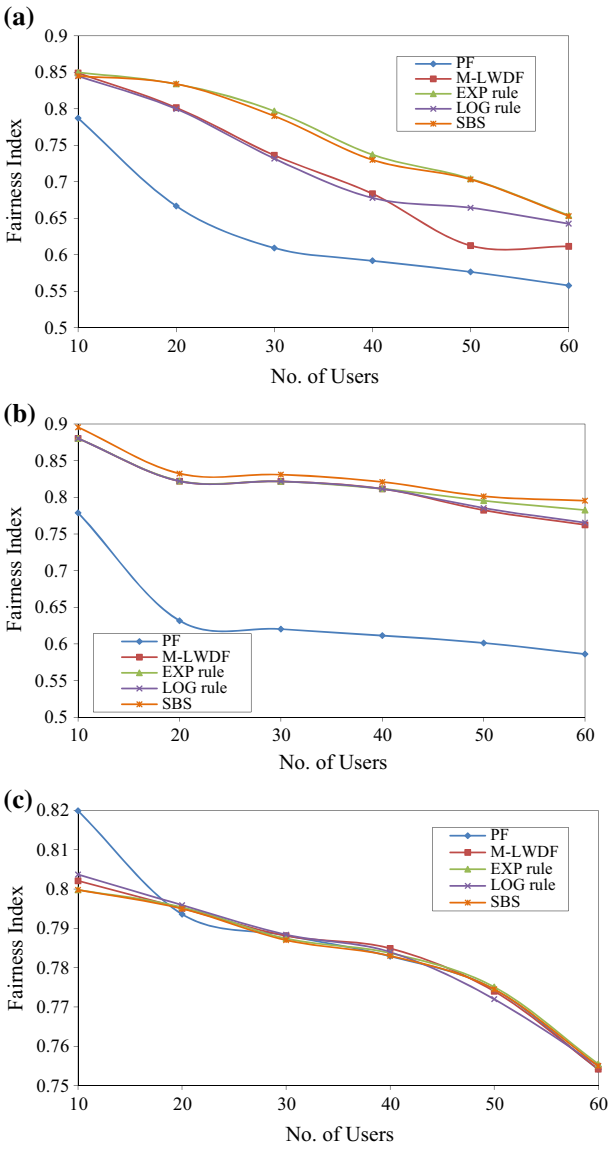


Fig. 3 Fairness index of **a** video flow, **b** VoIP flow and **c** BE flow

Fairness index for video, VoIP and BE services are shown in Fig. 3. Jain's fairness index method is used for evaluation of fairness among users [32]. It is noted that all the schedulers except PF show comparable level of fairness to real time flows closer to 0.85, and then decreases with increasing number of users. Since PF algorithm tries to balance between throughput and fairness, it shows lower fairness index for real time services.

The proposed SBS algorithm provides good fairness level for real time services even with the increasing number of users. High quality real time services allow lesser delay and tolerable packet loss. These parameters are very important for providing QoS to real time services. Delay experienced by video and VoIP services are presented in Fig. 4.

It is noted that the delay experienced by video services lies between 7 and 15 ms and by VoIP services lies between 1.7 and 2.1 ms for all the schedulers taken for comparison except PF algorithm, for maximum number of users (60 users) used in this simulation. Among existing algorithms, the performance of EXP rule algorithm is comparative with that of proposed SBS algorithm. It can be concluded that the PF algorithm cannot afford sufficient video and voice quality to the users. Since VoIP users have small packet size, it has lower T_{max} value and gets prior resource allocation by the SBS algorithm. The proposed SBS algorithm provides lesser delay for real time services when compared with other existing algorithms. It is shown that there is 14 % decrease in delay of video service and 5 % decrease in delay of VoIP service when compared with the existing EXP rule algorithm. Another important parameter for providing QoS to real time services is its PLR. At a particular time, it is the rate of packets discarded that are not meeting the delay

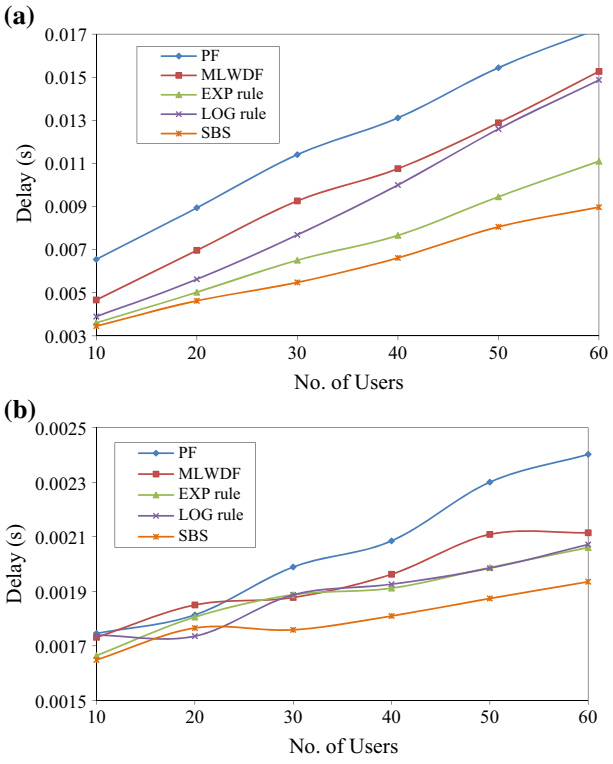


Fig. 4 Delay of **a** video flow and **b** VoIP flow

requirements to the sum of packets arrived to the eNB buffer. The PLR experienced by video and VoIP flows are shown in Fig. 5.

VoIP packets experience smaller PLR than video because of their smaller packet size. Since the proposed algorithm allocates resources based on the value of T_{max}, GBR bearers gets resources in spite of their channel conditions. The proposed SBS algorithm provides maximum of 0.2 % packet loss for video flows and 0.05 % packet loss for VoIP flows, for maximum number of users (60 users) used in this simulation. It is noted from the figure that the proposed algorithm provides adequate quality for real time services. When compared with the existing EXP rule algorithm, proposed SBS algorithm shows 51 % decrease in PLR for video flows and 36 % decrease in PLR for VoIP flows. The average goodput for video, VoIP and BE flows are shown in Fig. 6.

Average goodput is nothing but the rate of useful bits successfully transmitted during the entire duration of the simulation. It is observed that the goodput decreases with increasing number of users. For real time flows, the proposed SBS algorithm shows better performance compared to other algorithms because SBS algorithm gives priority to GBR bearers. It is shown that, when compared with the existing EXP rule algorithm there is 3 % improvement in average goodput of video flows and 15 % improvement in the average goodput of VoIP flows using proposed SBS algorithm. All algorithms including proposed algorithm provide similar performance for BE flows. For non real time services, the proposed SBS algorithm allocates resources with the best channel conditions.

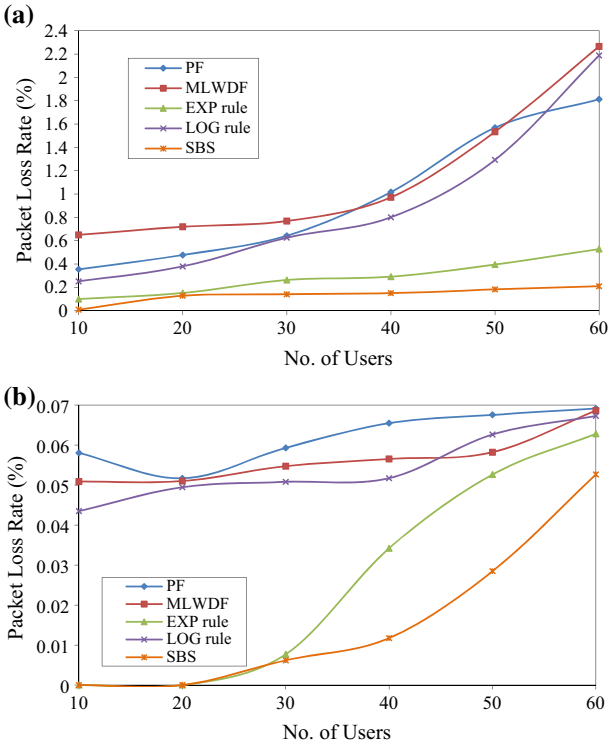


Fig. 5 PLR of **a** video flow and **b** VoIP flow

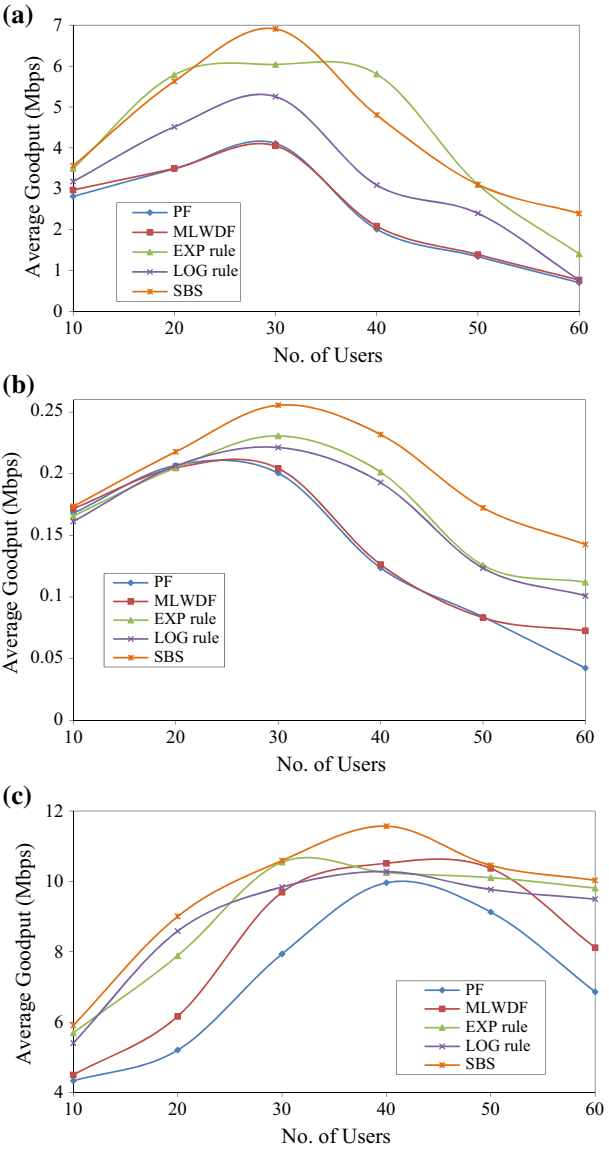


Fig. 6 Average goodput for **a** video flow **b** VoIP flow and **c** BE flow

5 Conclusion

In this paper a new scheduling scheme namely SBS algorithm is proposed that is suitable for real time services. In SBS, two lists called, T_{max} and T_{min} are created based on the service requirements. T_{max} list consists of packets from real time services and T_{min} list consists of packets from non real time services. The packets from the T_{max} list are chosen for allocation first in spite of their channel conditions. If the current TTI has more resources, then the packets from the T_{min} list are allocated. While allocating the packets from the T_{min} list, the users with the best channel conditions are prioritized. The proposed

scheduling approach is compared to other schemes like PF, M-LWDF, EXP rule, LOG rule. Among existing algorithms, the performance of EXP rule algorithm is comparative to that of proposed SBS algorithm. By using proposed SBS algorithm there is 14 % reduction in delay, 51 % reduction in PLR and 3 % improvement in average goodput for video flows. For VoIP flows, there is 5 % reduction in delay, 36 % reduction in PLR and 15 % improvement in average goodput. From the simulation results, it is confirmed that the proposed algorithm provides improvement in the system performance in terms of fairness, delay, PLR and average goodput.

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