A proposed algorithm for scheduling and resource allocation For the downlink of LTE networks

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ملخص

Long-Term-Evaluation (LTE) هي احدى شبكات الجيل الثالث في نظام الاتصالات المحموله. و هي قادره على دعم تطبيقات الوسائط المتعدده اللحضيه و العديد من التقنيات الأخرى. من أجل ارضاء الزبون هذه التقنيات يجب ان تبث بمستوى عالي من الجوده. هذه الجوده يمكن تحقيقها من خلال أنجازمجموعه من المتطلبات مثل تقليل معدل الخسائر و التاخير في انتقال الحزمه .أما من و جهة نظر مقدم الخدمه (Provider) فان تحسين اداء النظام من خلال التوزيع العادل و الفعال للطيف تعتبر عمليه معقده. و كما هو معلوم فان هذه المتطلبات ترتبط بشكل مباشر بالمجدول (scheduler) الموجود ضمن المرسل. في هذا البحث تم اقتراح و تطبيق لوغارتميه جدوله جديده من اجل التطبيقات اللحضيه في شبكات (LTE). اللوغارتميه المقترحه تقوم بتقليل عدد الحزم اللتي اسقطت بسبب الزياده الكبيره في عددها (الاحتقان) و كذلك تقوم بتقليل عدد الجزم التي ضاعت بسبب سوء جوده القناة وبالتالي تحسين كفاءة النظام .ايضا الوغارتيميه الجديه تقوم باعطاء الأولويه للمستخدم الذي يمتلك اعلى معدل خساره في الحزم . يتم ذلك عن طريق اخذ معدل الخساره في الحزم لجميع المستخدمين بنظر الأعتبار و كذلك المحافظه على مؤشر العداله ضمن الحدوده المقبوله و خاصه عند حصول زياده في عدد المستخدمين . ايضا من خلال الاستفاده من نظام Adaptive Modulation and Coding Scheme (AMC) تم زياده كل من سعة النظام ،المخرجات و معدل الترميز. كذلك فأن اللوغارتيميه الجديده اظهرت فعاليه افضل من مثيلاتها من ناحية اداء النظام.

Abstract

Long Term Evaluation (LTE) networks can be categorized as a member of the third generation of mobile communication system, and it is capable of supporting real time multimedia and a variety of broadband technologies. In order to satisfy customers, these technologies must be transmitted with a good level of Quality of Service (QoS). This can be achieved by fulfilling a number of requirements such as lowest possible packet loss rate and delay. While from a provider's perspective, improving the systems performance by efficient and fair spectrum allocation is challenging issue. As known, these requirements are mainly managed by the scheduler at the transmitter side. In this research, a novel algorithm of scheduling is proposed and implemented for real time applications in the downlink of LTE networks. The proposed algorithm reduces the number of packets that has been dropped due to congestion as well as the number of lost packets due to bad channel quality, and thus enhancing the system performance. It also prioritizes the user with the highest packet loss rate by taking into account the packet loss rate of each user and maintaining the metric of fairness within normal levels as the number of users increases. Also the system capacity, throughput and modulation order were increased by taking advantage of the Adaptive Modulation and Coding Scheme (AMC). The proposed algorithm has proven to be more efficient in terms of system performance than similar algorithms

Keywords: LTE networks; Proportional Fair; Scheduling; QoS; Resource Block

Abbreviations

AMC: Adaptive Modulation and Coding, **AMR:** Adaptive Multi-Rate, **BER:** Bit Error Rate, **eNodeB:** Evolved Node Base Station, **HOL:** Head Of Line, **IP:** Internet Protocol, **LTE:** Long Term Evaluation, **MAC:** Medium Access Control, **OFDMA:** Orthogonal Frequency Division Multiple Access, **PDR:** Packet Drop Rate, **PER:** Packet Error Rate, **PLR:** Packet Loss Rate, **QoS:** Quality of Service, **QAM:** Quadrature Amplitude Modulation, **QPSK:** Quadrature Phase Shift Keying, **RAN:** Radio Access Networks, **RB:** Resource Block, **RE:** Resource Element, **3GPP:** Third Generation Partnership Project, **TTI:** Transmission Time Interval, **UE:** User Equipment, **UL:** Up Link, **UDP:** User Datagram Protocol, **VoIP:** Voice Over IP

1 Introduction

LTE networks have been recently proposed, and they support a wide range of applications including Internet Protocol (IP) based applications and internet access faculties for users [Dahlman *et.al*., 2011]. According to (Piro *et.al* .,2011) LTE networks are generally characterized by their improved spectral efficiency, operational simplicity, and scalable bandwidth.

It is well known that the scheduling algorithm plays a major role in communication systems as it controls the distribution of availableresources among users at any given period of time. In LTE networks the scheduler has even more importance since a better system performance can be obtained by simply distributing the amount of frequency channels assigned to different users in an appropriate way at any given period of time.

In this project, a scheduling algorithm for the downlink of LTE networks will be designed and simulated and its performance will be evaluated. The algorithm optimizes the rate of packet loss for real time applications by taking into account the delay budget, it also allocate resources fairly among users in the Medium Access Control (MAC) sub-layers. Besides, it increases system capacity by improving the link utilization. This is mainly done by taking into account parameters such as Channel Quality Indicator (CQI) which is used to locate a proper coding system and modulation to the users. The algorithm is practically simulated by MATLAB

2 Proposed approach

2.1 Parameters description of proposed algorithm

LTE networks are capable of providing different specific sub-sets of RBs for different ranges of bandwidth that they support. And These RBs are allocated for the active users. For example, each sub-frame (Transmission Time Interval,TTI) contains 25 RBs (Resource Blocks) for bandwidth of 5 MHz, therefore, there are only limited number of users which can be scheduled in each (TTI) (Holma andToskala ,2009, Li *et.al* .,2016)]. In this proposed algorithm every user possesses a queue with a finite length. Depending on the traffic model suggested within the next paragraphs, a packet (a voice) will be generated by each user in the normal time interval with a constant duration and variable data rate. Voice over IP(VoIP) is considered as normal class service, and the delay for any packet must be kept within acceptable limits. If the network core delay is assumed to be below 100 ms, then the highest allowable air to mouth delay for voice must be kept below 250 ms. The interface delay forhigh speed packet access (HSPA) should be kept under 80 ms, which is also the allowable MAC/buffering and scheduling delay if both users developed an Evolved Universal Terrestrial Radio Access (E-UTRAN) [Furht andAhson, 2009]. For LTE networks and in order to improve the quality of voice further, the allowable air interface delay was minimized to 50 ms. Therefore the MAC/ buffering and scheduling delay (target delay) T_{td} is set in this project to be 40 ms, and the packet waiting duration in the queue is defined as${T_w}^{ui}$, the packet will timed out and dropped if its waiting time exceeds the target delay, otherwise it will remain in the queue waiting for scheduling. Also T_{th} is defined as the delay threshold. If the packet waiting time exceeds the T_{th} it will be given priorty to avoid exceeding the target delay

$${T_w}^{ui} \leq T_{td} \tag{1}$$

According to Puttonen et al (2008) the threshold delayfor VoIP is defined in (2).

$$HOL > 0.8 * Target\ Delay \tag{2}$$

HOL is defined as the period from the moment of packet creation to the current time in the queue. Depending on the target delay proposed in this work, the threshold will be set to 32 ms. Afterwards, the next metric (PLR) is defined by the summation of the rate of dropped packets due to congestion of the queue (PDR) and the rate of packet loss because of bad channel quality (PER) up to time t for the i-th user. The packet with the maximum PLR will be prioritized by the proposed algorithm by exploiting the metric as shown in expression (3) which is given by [Tantawy et al (2011)].

$$PLR^{ui}(t) = PER^{ui}(t) * \left(1 - PDR^{ui}(t)\right) + PDR^{ui}(t) \tag{3}$$

The mathematical expression for PLR for user i is given in (4)[Tantawy et al (2011)].

$$PER^{ui}(t) = \frac{\sum_{t=1}^{T} lost_t^{ui}}{\sum_{t=1}^{T} txd\ pkt_t^{ui}} \tag{4}$$

Where $\sum txd\ pkt$is defined as the total number of sent packets to the i-th user (the lost and successfully delivered). While $\sum lost$ is defined as the total number of lost packets in the i-th user queue up to time t due to bad channel quality. The mathematical expression for PDR for the i user is given in (5).

$$PDR^{ui}(t) = \frac{\sum_{t=0}^{T} drop_t^{ui}}{\sum_{t=0}^{T} pkt_t^{ui}} \tag{5}$$

Where $\sum pkt$is defined as the total number of active packets in the queue of the i-th user. This parameter will be updated at the end of each TTI by the scheduler for the active user in the system. While $\sum drop$ is defined as the total number of dropped packets of the i-th user queue up to time t due to the lack of adequate radio resources for scheduling the packet before reaching the targeted delay.

The instantaneous SNR will be reported by the user to the eNodeB at the end of each TTI. This determines the data rate in number of bites for the RBS given to the particular user in each TTI. The data rate $R_{i,j}$ of the user i-th (bites/TTI) for j-th RB at a time t is expressed by equation (6) [Ali andZeeshan, 2011].

$$R_{i,j}(t) = \left(\frac{NumOfBits}{symbol} * \frac{NumOfSymbols}{slot} * \frac{NumOfSlots}{TTI} * \frac{NumOfSub-carriers}{RB}\right) \tag{6}$$

Except for the first section, all other sections of expression (6) are easily determined according to the system terminology, while the first part will be explained within the next paragraphs. Expression (7) defines the estimated PER measured for each RB.

$$P_p = 1 - (1 - P_b)^N \tag{7}$$

Where N is defined as the number of bites per packet and P_b is defined as bites error rate (BER), while P_p is defined as packet error rate (the likelihood that a packet is lost). The probability will be determined for all allocated RBs to the one particular packet. Based on the above value a decision will be made on whether the whole packet will be transmitted or lost.

2.2 Performance analysis by matrices

the proposed algorithm performance is evaluated depending on the following features: the total system PLR, fairness, system throughput and delay.

According to Ali andZeeshan (2011) the system throughput is generally known as the summation of the packets that are transmitted in time (seconds) from all users to the eNodeBand vice versa. While in this project only the downlink is being considered, also the aggregate throughput will be considered, and it is defined in [Ali andZeeshan, 2011] and given by expression (8).

$$System\ Throughput = \frac{1}{T}\sum_{i=1}^{K}\sum_{t=1}^{T} PSize_i(t) \tag{8}$$

Where K is defined as the total number of users that receive packets from eNodeB, and T is defined as the total time of simulation, while $PSize$ is defined as the packet size in number of bites that has been transmitted from eNodeB to a specific user i which is aggregated over the time period.

The average delay is defined as the period of time that a packet has to wait in the queue till being scheduled. And it is expressed in equation (9).

$$avgdelay = \frac{\sum_{p=1}^{n} T_w(p)}{\sum_{i=1}^{S} pkt} \tag{9}$$

Where $\sum pkt$ is defined as the total number of packets in the queue at the end of simulation time, while $\sum T_w$is considered as the total delay duration for all the packets in the queue (i.e. both timed-out and scheduled).

In the proposed algorithm, fairness works in a way that in the second level of prioritization the user with the highest PLR will be selected by the proposed scheduling mechanism. This method however cannot be used to calculate overall system fairness due to the fact that at higher levels and to avoid packets being dropped the selection mechanism selects the packet with maximum waiting time in the queue, therefore the effect of second level fairness function is measured using the intra-class fairness and compute the Jain's fairness index which is defined in [Ali andZeeshan, 2011].

$$Fairness = \frac{(\sum_{i=1}^{N} x_i)^2}{N(\sum_{i=1}^{N} (x_i)^2)} \quad (10)$$

Where N is the number of competing users that have packets in the queue, and x_i is the good-put achieved by user i.

The total packet loss rate can be defined as the total number of lost packets both :lost due to bad channel quality and dropped because of congestion of the queue over the total number of arrived packets to the eNodeB, which is expressed as:

$$PLR_{total} = PDR_{total} + PER_{total}(1 - PDR_{total}) = \sum_{ui=1}^{N} PDR^{ui} + \left(\sum_{ui=1}^{N} PER^{ui}\right) * \left(1 - \sum_{ui=1}^{N} PDR^{ui}\right) \quad (11)$$

In this paper the QoS will be improved by reducing the PLR_{total} . Theproposed algorithm efficient use of RBs is also evaluated using expression (12) which gives the average percentage of consumed RBs in each TTI

$$AvgUesdRBsRate = \frac{(\sum_{i=1}^{T} \left(\frac{NumOfUsedRB}{TotalNumOfRB}\right))}{T} \quad (12)$$

Where T is defined as the total simulation time

2.3 Proposed algorithm procedure

In this algorithm the packet loss rate will be considering both the PER at the physical layer (PHY) and the PDR at the MAC layer. As a consequence, both systems performance and average throughput will be maximized. Also the target error rate is considered 0.01 as given by [Korowajczuk, 2011]. Thus the algorithm will tend to improve the QoS by reducing the PER and PDR.

The scheduler role is divided into two steps

2.3.1 Packet prioritization

At the eNodeB each user will be assigned with a queue. When a packet reaches the serving eNodeB, it will be stamped and queued in First In First Out order (FIFO) By the system. Then a list of packets that can be scheduled will be created in the current sub frame by the scheduler every TTI. Next by subtracting the packet arrival time from the current time, the HOL packet delay for each user's packet will be calculated. Then the highest priority will be given to the packets with the highest waiting time (HOL) of greater than T_{th}, this will reduce the number of dropped packets in the queue.

In the case of packets with HOL less than T_{th} the decision will be taken based on their PLR_t^{ui} parameter, where the packet with the highest PLR_t^{ui} will be selected. That is mainly because high PLR means either high packet drop rate or high bite error rate or both, thus fairness will be fulfilled between different users with different channel quality.

Also the algorithm will select the user with the highest PLR, if more than one packet for different users exceeded the T_{th}. Furthermore, if more than one packet has the same PLR and a waiting time less than T_{th} the algorithm will select the packet with the best channel quality in order to use the radio resources more effectively.

2.3.2 Allocation of resources

After scheduling a user from the first step and in order to assign a transmission resource for that user, the algorithm will determine the resource assignment size dynamically depending on the supportable user payload and his channel quality. In order to reach the targeted packet error rate, the necessary amount of transmission resources will be allocated to each user using dynamic approach. When making scheduling decisions, if the instantaneous downlink channel conditions for a particular packet error rate at the physical layer were exploited by the scheduler, then the radio resources can be fully utilized, this is mainly done using the MAC technique. The proposed scheduler will select the coding scheme depending on the targeted bit error rate BER and SNR value, it also chooses a modulation order ranging from QPSK, 16-QAM and 64-QAM. In order to reduce the number of lost packets because of low SNR (when the user experiences a fading channel the PER will increase significantly) the algorithm will determine the standard deviation and the mean value of the SNR for the available RBs. if the mean value is greater than the standard deviation, this means that the fluctuations of the SNR are close to each other. Therefore, the RB with the closest SNR to the mean value will be chosen by the algorithm. Otherwise, the RB with the minimum SNR will be chosen and consequently applies appropriate modulation order. Now both the coding rate and the modulation order have been determined. As itknow, each RB is combined of 7Orthogonal Frequency-Division Multiple Access (OFDMA) symbols and a group of 12 sub-carriers in every 0.5 m slot. And there are two slots in each sub-frame. So, there are (12*7) 84 resource element in each resource block. **Table 1** illustrates the calculations required to determine the total transmittable bits per each RB

Table 1: Transmittable bits per RB

64-QAM (6 bits per symbol)	84*6=504
16-QAM (4 bits per symbol)	84*4=336
QPSK (2 bits per symbol)	84*2=128

Now the number of bites that can be transmitted by one RB will be calculated by the scheduler, it also will calculate if more than one RB is required, and if so, the next RB with the maximum SNR will be chosen by the scheduler. Sometimes there are not enough resources for the scheduler to transmit the packet at the current TTI. In this case the packet will have to wait for the next TTI decision. Please refer for **figure 1** which shows the flow chart of the scheduling algorithm for more details.

3 System simulation

3.1 Modeling of the system

3.1.1 Assumptions

In order to minimize the system simulation complexity, a number of assumptions are made: at any scheduling time the same RB cannot be assigned to more than one user, thus avoiding overlapping in the allocation of RBs. Also the serving eNodeB is assumed to be at the center of the cell, thus only single cell scenario is considered. The available RBS are controlled by the serving eNodeBs' MAC schedulerby allocating them to active flows computing for resources. A uniform distribution of the users is also assumed and there is no inter-symbol or inter-subcarrier overlapping. The length of RB (one time slot with duration of 0.5 ms) is constant in the time and frequency domain (180 KHz).The channel is considered variable from one RB to another, but constant during each time slot.

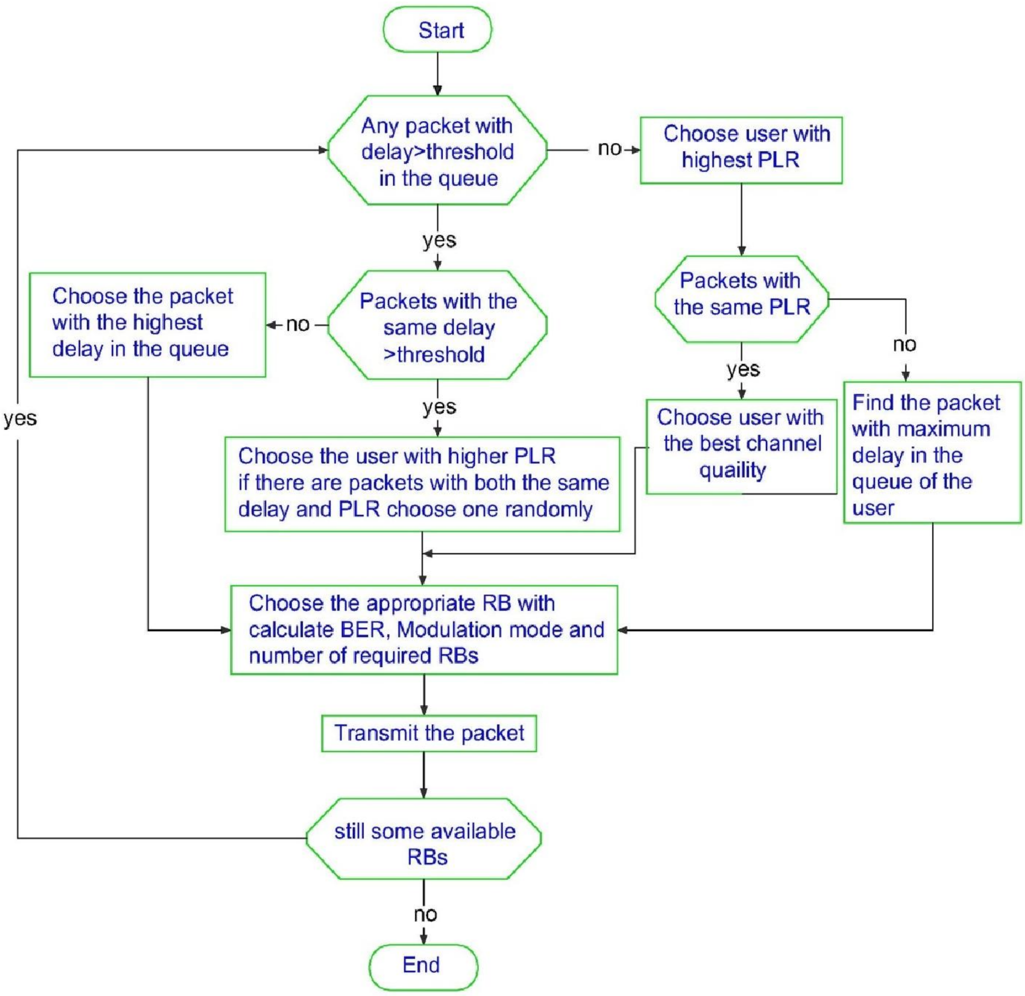

Figure 1: flow chart of the downlink QoS Aware Scheduling Algorithm

The same format is used to module all the RB symbols at the physical layer. Only one packet of a user can be transmitted in each TTI. In each TTI the packet is either transmitted entirely or ignored (it cannot be divided between TTIs). Also the whole packet will be considered lost even if only one part of it exceeded the targeted PER, which equals to 0.01 as discussed earlier. There are three different types of LTE networks. In this work, only type 1 will be considered. This is mainly because the RBs with the best channel condition will be selected by the proposed algorithm, thusthe authorization for allocating nonconsecutive RBs is required. The simulation scheme is illustrated in **figure 2**

3.1.2 Modelingof traffic

In this project, packets are modeled based on the AMR at the data link layer. **Figure 3** shows the relationship between time and data rate (using 7 different modes of the AMR).the data ranged from 12.5-4.75 kbps [Puttonen et al, 2008]. During this simulation the application layer will create the packets and pass them to the IP and user datagram protocol (UDP). The IP datagram is mapped using an IP based packet classifier to the radio bearers. Also some overhead bits might be carried out with the packet payload when it reaches the data link layer. The total overhead can be calculated using the following expression.

$$General\ packet\ size\ of\ the\ VoIP = Voice\ payload + IP\ overhead + Encapsulation\ overhead + Frame\ overhead$$

Encapsulation overhead is ignored in this simulation.IP overhead referred as Cisco as overhead occurring at layer 3 and above so this meansreal-time transport protocol RTP (12 Bytes), UDP (8 Bytes) and IP header 20 Bytes (IPv4) [Piro et al , 2011]. Due to IP overheadthis is the total of 40 Bytes. the frame overhead would be 18 Bytes, If the packet is going across LAN,for Etherent II. However, the frame structure of AMRcan bedeterminedusing the following expression:

$$Frame\ Structure\ of\ AMR = \big(Frame\ Quality\ Indicator\ (1\ bits) + \ Frame\ Type\ (4\ bits)\big) + AMR\ header + AMR\ Auxiliary\ information\ (14\ bits) + AMR\ core\ frame\ =\ 19\ bits$$

Then the total overhead is 58 Bytes (18 + 40 = 58 Bytes = 464 bits + 19 bits = 483 bits). **Table 3**illustrates the simulation parameters.

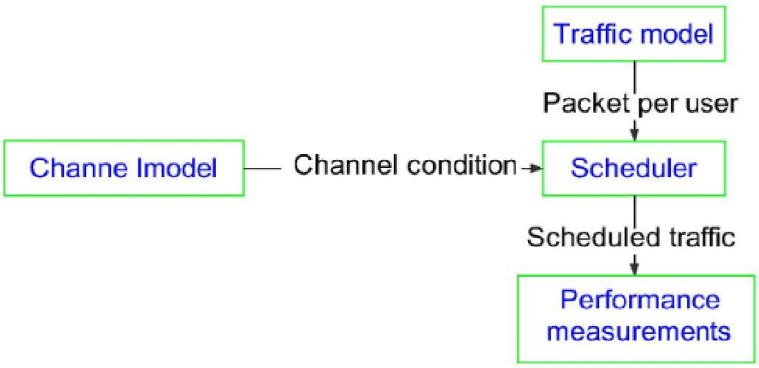

Figure 2: general scheme of simulation procedure

3.1.3 Modeling of the channel

The channel will be approximated as Additive White Gaussian Noise(AWGN)with flat fading and flat Rayleigh distributed channel attenuation coefficient. In the case of fading channels, a single parameter will capture the channel quality, the parameter mainly is the instantaneous SNR γ. And during a frame it will remain invariant. While γ is described statically by adapting the general Nakagami-m model [Wang *et.al.*, 2005].Thus, it can be considered that each receivedSNR γ per frame is a random variable and is distributed exponentially with the gamma probability density function (PDF) [Liu *et.al.*, 2005].

$$p_{\gamma}(\gamma) = \frac{m^m \gamma^{m-1}}{\overline{\gamma}^m I(m)} \exp\left(-\frac{m\gamma}{\overline{\gamma}}\right) \tag{13}$$

$$p_{\gamma_{ui}}(\gamma_{ui}) = \frac{1}{\overline{\gamma}_{ui}} \exp\left(-\frac{\gamma_{ui}}{\overline{\gamma}_{ui}}\right) \tag{14}$$

Where $\overline{\gamma}_i = E(\gamma_i)$ is defined as the average received SNR and γ_i is defined as the SNRinstantaneous value per RB. When m=1 it includes the Rayleigh channel.

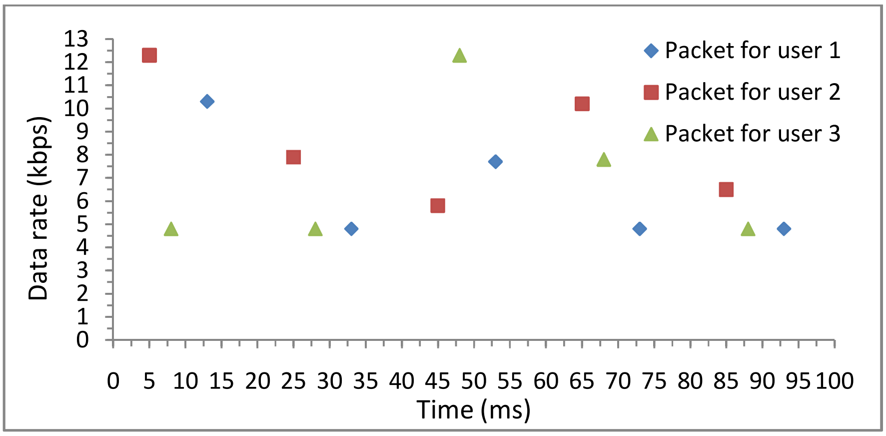

Figure 3: AMR-based traffic model

Figure 4 shows the Rayleigh cumulative distribution function (CDF). While **figure 5** illustrates the performance of BER for different coding schemes and modulations for PDSCH .the mapping of the modulation scheme and the boundaries of the SNR are performed based on **table 2** [Chaudhuri *et.al*., 2011]. Depending on Goldsmith (2005) the target BER for voice packets is considered to be 10^{-3}. The RB range for different bandwidths and the peak bit rate of the downlink are considered based on [Holma andToskala, 2009], while the bandwidth for this work is considered to be 5 MHz (15 RBs for every TTI).

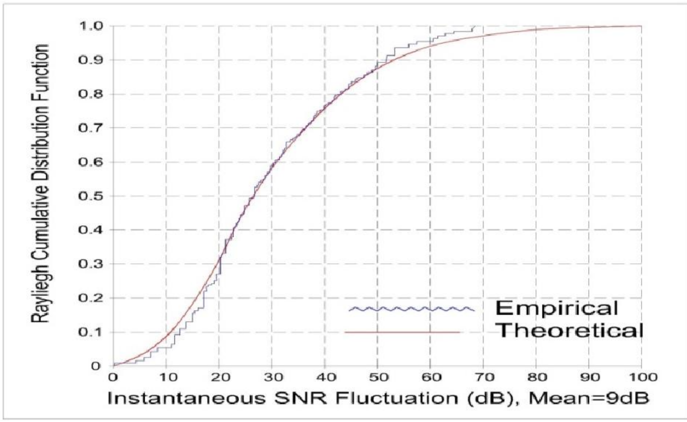

Figure 4: SNR fluctuation per RB in a Rayleigh distributed channel

In order to compare the result from the simulation of the proposed algorithm, the PF scheduling algorithm is also implemented using the same matrices used to evaluate the proposed algorithm. The proposed algorithm is shown in **figure 1**.

3.2 Simulation of the proportional fairness scheduling algorithm (PF algorithm)

The PF scheduling algorithm is considered as a good solution according to various researchers, this is mainly because it offers a compromise between user fairness and maximum average throughput. But it ignores other QoS performances such as PLR and packet delay. Before implementing the PF scheduling algorithm and because the PF algorithm has been developed for a single carrier wireless system, some modulation must be made so it can support scheduling of packets in the downlink of 3GPP LTE system

Table 2: modulation schemes and corresponding SNR intervals

Modulation order	SNR boundaries
QPSK 1/2	$for\ 0 \leq \text{SNR} \leq 8\text{db}$
16-QAM 1/2	for $8 \leq \text{SNR} \leq 14\text{db}$
64-QAM 3/4	for $14 \leq \text{SNR} \leq \infty$

Table 3: parameters of the simulation

Parameters	Value
Packet type	VoIP
Bit rate (Kbit/s)	4.75,5.15,5.90,6.70,7.40,7.95,10.2,12.2
Packet duration	20 ms
Time-space between packets	20 ms
Simulation duration	100 ms (100 TTI)
Target delay	40 ms
Delay threshold	32 ms
Target BER	10^{-3}
Subcarriers per RB	12
Channel bandwidth	5 MHz
Number of RBs	15
Modulation / coding rate (AMC)	QPSK: 1/2, 16-QAM:1/2, 64-QAM:3/4
Channel attention coefficient	Flat fading/ Rayleigh distributed

3.2.1 PF scheduler procedure

3.2.1.1 Packet prioritization

Packets for different users will arrive to their assigned queues, when reaching the eNodeB each packet will be stamped by the buffer management system and queued in First In First Out (FIFO) manner. At the eNodeB active users (who have packets within the queue) will report their channel state at each time slot. Then the ratio of the achievable rate to the average throughput for each user will be calculated by the eNodeB, which is considered as the key selection criteria. Then an exponential moving average will calculate the average throughput for each user. The user with the maximum preference metric will be selected for transmission at the next coming slot. The channel state is considered constant during each transmission time slot. The formal description of the procedure is as follows.

$r_{i,j}(t)$refers to the achievable transmission data rate on the j-th RB (time-slot) for the i-th user and it can be calculated by using AMC [Huang *et.al.*, 2011]. While $\overline{R_t(t)}$ refers to average throughput and it is updated according to Li et al (2010) as follows:

$$\overline{R_t(t+1)} = \begin{cases} \left(1-\frac{1}{t_c}\right)\overline{R_t(t)} + \frac{1}{t_c}R_t(t), & i = i* \quad if\ the\ user\ i\ is\ scheduled \\ \left(1-\frac{1}{t_c}\right)\overline{R_t(t)} & i \neq i* \quad if\ the\ user\ i\ is\ not\ scheduled \end{cases}$$

(15)

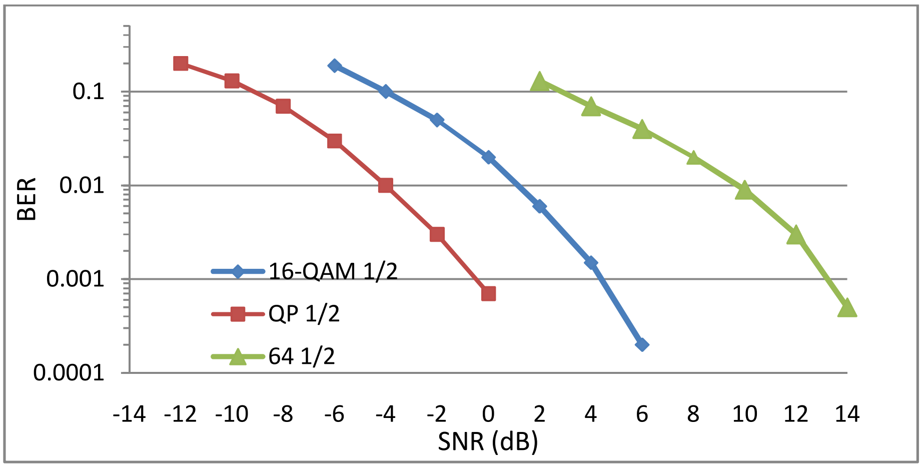

Figure 5: BER vs. SNR

3.2.1.2 Allocationof resources

To reduce the number of differences between the tow algorithms and to have a better compression, the same procedure as step 2 in (section 2.3) will be also applied for the PF scheduling algorithm

The evaluation of the PF algorithm is based on (Sadiq *et.al* .,2008),(Liu *et.al*.,2005),(Gidlund and Laneri 2008), (Trivedi& Patel 2014),(Radhakrishnan *et.al*.,2016)].

4 Results and discussion

4.1 Packet drop rate (PDR)

Figure 6illustrates the packet drop rate for both the PF algorithm and the proposed algorithm. From the figure, it can be observed that the proposed algorithm outperforms the PF algorithm by achieving a very low PDR (almost zero); this is mainly because the proposed algorithm avoids the packet being dropped by prioritizing the packets that exceed the defined threshold. But when the number of users increases, queue congestion occurs and some packets will be timed-out, due to the insufficient numbers of RBs to transmit them. That's why when reaching 70 users the proposed algorithm starts to reach the targeted PDR, which is approximately 1%.

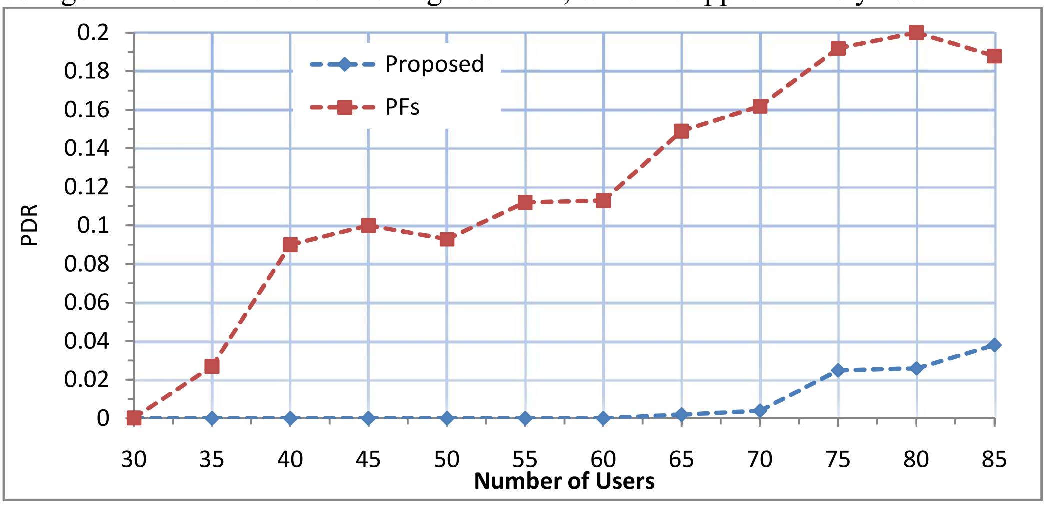

Figure 6: packet drop rate

4.2 Packet error rate (PER)

Figure 7 shows the PER (the total number of packets which have been lost due to bad channel quality) for both the PF algorithm and the proposed algorithm, as can be seen from the figure, both algorithms have almost the same level of performance. It also can be observed that when the number of users increases, the proposed algorithm shows unstable performance, this is mainly because during the simulation the average SNR for each user is produced randomly. But in general, it is possible to say that the proposed algorithm improves the system performance, because it kept the PER level below 0.005 in most cases, this is achieved by transmitting user packets based on their best possible link conditions.

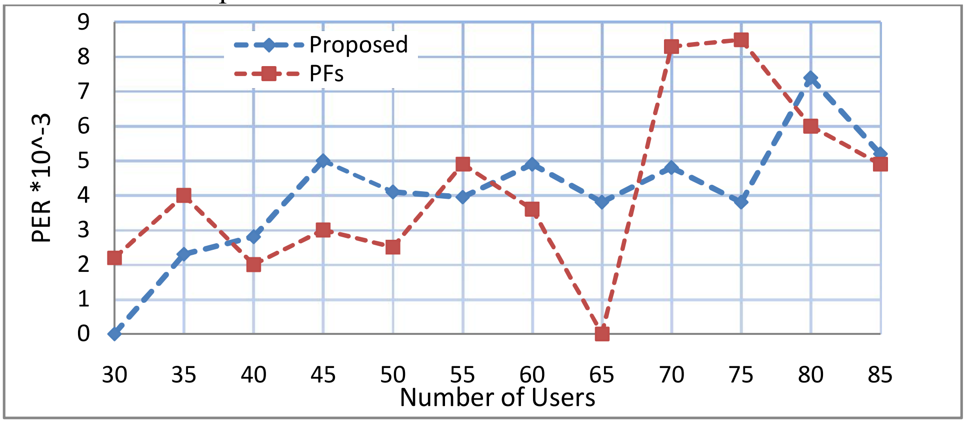

Figure 7: packet error rate

4.3Total packet loss rate (PLR)

Figure 8 illustrates the PLR for both the proposed and PF algorithms. The figure shows that the PLR performance for the proposed algorithm is significantly lower than the PF algorithm, this comes as a natural result since the PLR comes from the aggregation of the PDR and the PER, which both have low values for the proposed algorithm as shown earlier. Therefore, it is possibletoconclude that the proposed algorithm performs better than the PF algorithm by supporting higher number of users at the recommended PLR. The main reason of the PF algorithm bad performance is that it doesn’t consider neither the PLR nor the HOL in the scheduling decisions.

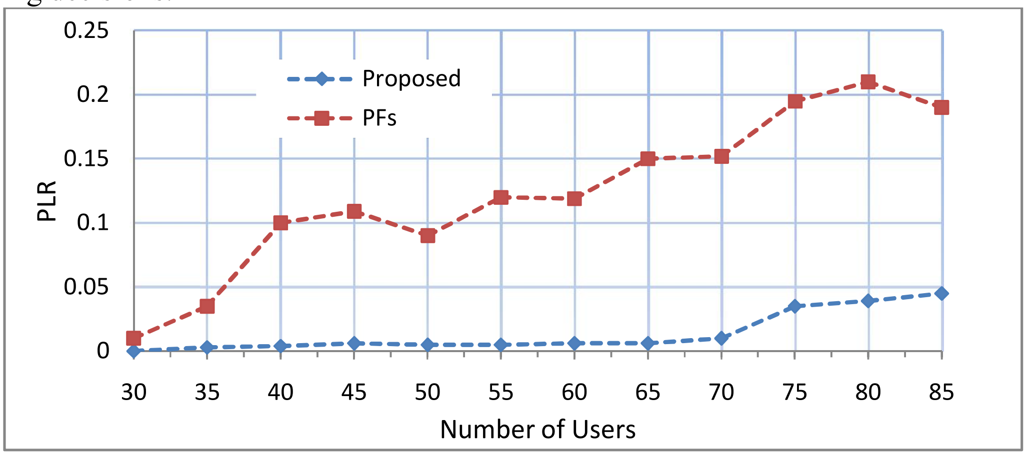

Figure 8: total packet loss rate

4.4Average system throughput

Figure 9shows the average system throughput which is the total number of bits that has been transmitted within the simulation time. From the figure, it can be seen that the proposed algorithm has improved the systems throughput. This is mainly because the number of dropped packets is near zero for the proposed algorithm, which means that most of the packets are being transmitted, thus better system throughput. Also the proposed algorithm prioritizes the users with the best channel conditions, thus transmitting with higher modulation order and number of bits.

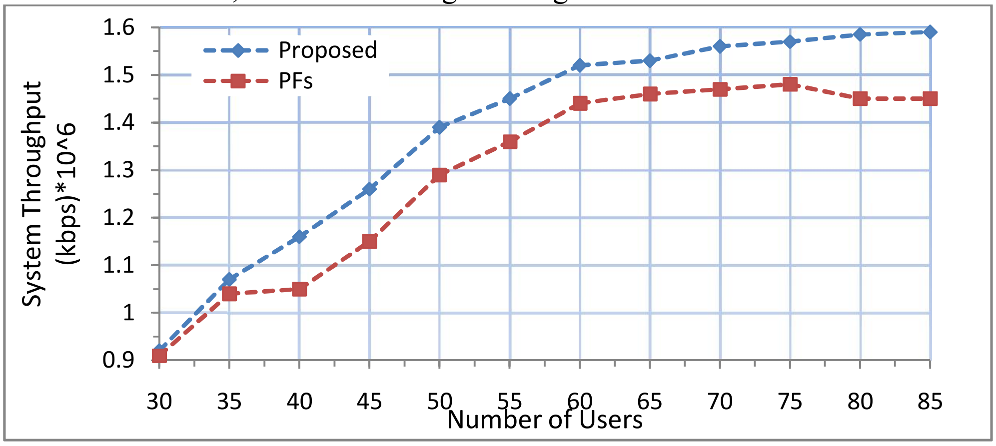

Figure 9: average system throughput

4.5 Average delay

Figure 10 shows the average delay per packet for both algorithms. The figure illustrates that at number of users below 50 the proposed algorithm performs better than the PF algorithm, and worse above 50. The main reason behind this is that the proposed algorithm gives the scheduling opportunity to the users packets with the highest HOL delay.As the number of users increases, more VoIP packets will be waiting for transmission at the serving eNodeB queue, thus the queue will become crowded, and due to the lack of sufficient RBs to transmit all packets, only the packets with the highest HOL will be transmitted by the proposed algorithm, hence increasing the HOL delay of the users packets who have not been transmitted. And thus the average delay will also increase as consequence. That has been said, the proposed algorithm has shown an ability to maintain reasonable packet delay when the number of users is below 80.

A second reason is the fairness performance of the proposed algorithm. In order to fulfill fairness, the proposed algorithm gives the priority to the user with the highest PLR, thus the average packet delay will increase with the increment in the number of users. It can be concluded that more delay in the system will be paid as an expense for reducing the number of dropped packets and increasing the system throughput. But the quality of packets (voice) should not be effected by this delay.

4.6 Fairness

Figure 11 shows fairness index for both algorithms. The figure illustrates that the fairness performance for the PF algorithm decreases considerably with the increment in number of users, while the proposed algorithm maintains almost a constant value. This is mainly because unlike the PF algorithm, the proposed algorithm avoids packet delay valuations by assigning more resources for users with packet delay. The packet delay is mainly due to either higher PER (bad channel quality) or higher PDR (HOL packet delay valuation). From this argument, it can be concluded that the proposed algorithm outperforms the PF algorithm in both system throughput and fairness.

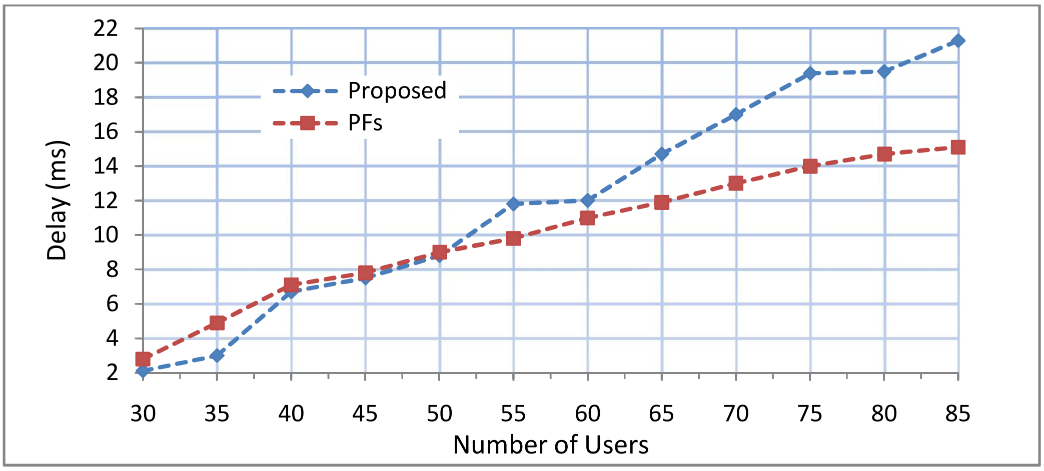

Figure 10: average delay per packet

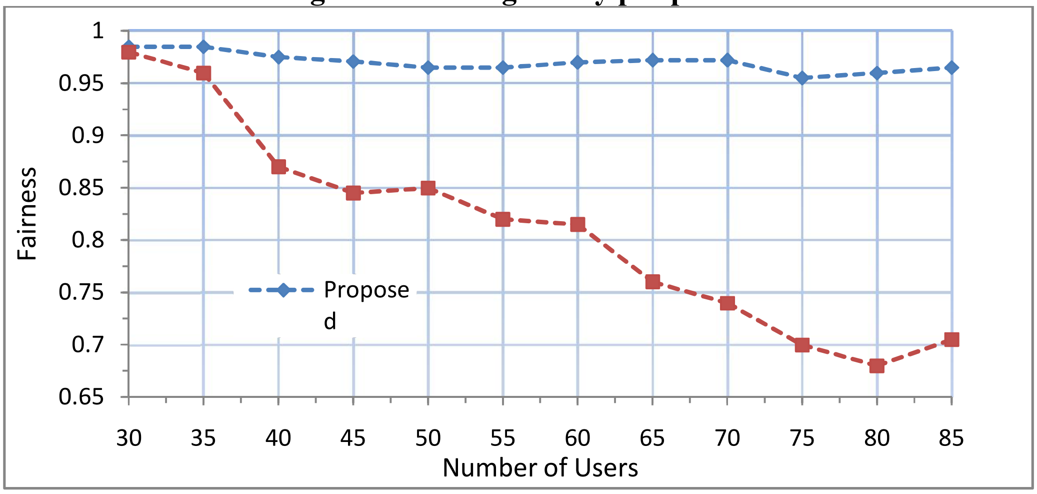

Figure 11: fairness index

4.7 Average resource utilization

Finally the average resource utilization is illustrated in **figure 12** for both algorithms. The average resource utilization can be defined as the average of consumed RBs in each TTI. From the figure it can be seen that the proposed algorithm showed better performance than the PF algorithm, the reason is that the proposed algorithm schedules more users during the whole simulation time, due to lower PDR. And thus it can be concluded that the proposed algorithm is more efficient than the PF algorithm in terms of radio resources utilization.

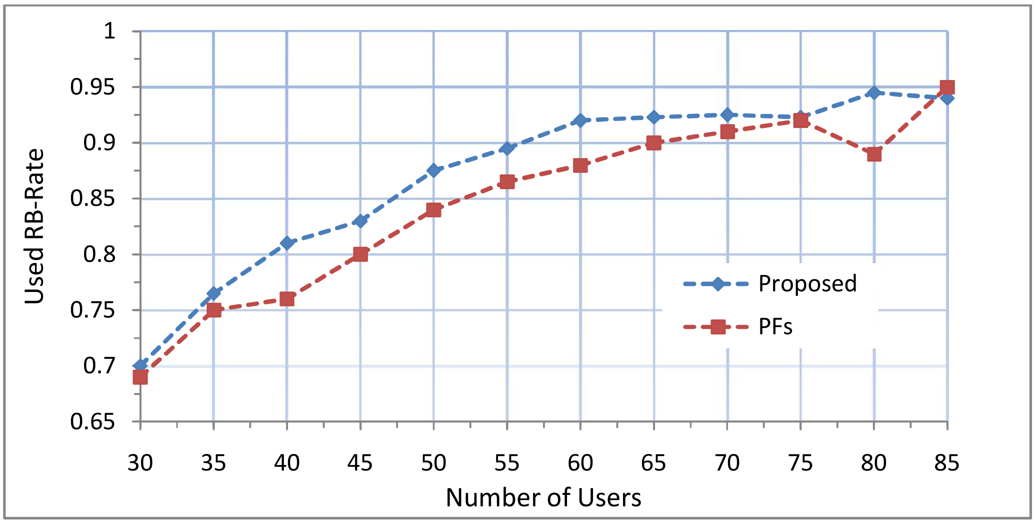

Figure 12: average resource utilization

Finally, it worth mentioning that many scheduling algorithm have proposed recently, yet most of those algorithms provide a compromise between fairness and throughput such as algorithms proposed by [Bahreyni and Naeini (2013),Aboul Hassan et al (2014), Bechir et al (2016)]. Unlike those algorithms, the proposed algorithm maintains almost a constant farness value of 0.95 regardless of the number of users, while improving the systems throughput considerably

5 :-Conclusions

From the results section the fooling conclusions can be obtained:

- The Packet Loss Rate (PLR) performance of the proposed algorithm is significantly lower than the PF algorithm, this comes as a natural result since the PLR comes from the aggregation of the Packet Drop Rate (PDR) and the Packet Error Rate (PER), which both showed low values for the proposed algorithm.
- When applying the proposed algorithm, the system throughput was improved considerably compared with the PF algorithm. This is mainly because the number of dropped packets has been reduced considerably and it almost reached zero when using the proposed algorithm, which means that most of the packets are being transmitted, thus better system throughput
- In terms of average delay, and when the number of users is below 50 the proposed algorithm performs better than the PF algorithm, yet worse when the number of users is above 50. This is mainly because the proposed algorithm transmits the users packets with the highest HOL delay,hence increasing the HOL delay of the users packets who have not been transmitted. And thus the average delay will also increase as consequence. Yet the proposed algorithm has shown an ability to maintain a reasonable packet delay when the number of users is below 80
- Unlike the PF algorithm, the proposed algorithm has shown the ability to maintain a constant fairness value when increasing the number of users. This is mainly because the proposed algorithm avoids packet delay valuations by assigning more resources for users with packet delay.
- Also the proposed algorithm showed better performance than the PF algorithm in terms of resource allocation, the reason is that the proposed algorithm schedules more users during the whole simulation time, due to lower PDR.

- Finally, it can be stated that the proposed algorithm maximized the system throughput while satisfying the QoS requirements.

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