

# Utility Based Scheduling and Call Admission Control for Long Term Evolution Networks

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## ABSTRACT

In this study, we propose to design a call admission control algorithm which schedules the channels for Real time and non-real time users. In Long Term Evolution (LTE) 3GPP Networks, several works were done on call admission control but these works rarely considers scheduling of resources to the real time and non-real time users. When the system meets traffic oriented performance degradation, maximum resources are utilized for load balancing and to maintain the consistent quality. In order to avoid the channel degradation and improve the Quality of Service (QoS), the call requests are classified into New Call (NC) request and Handoff Call (HC) request and the type of services are classified as VoIP and video. Then based upon the Received Signal Strength (RSS) value, the channel is estimated as good channel or bad channel. Resource allocation is made for VoIP users based on traffic density. Then non-VoIP users and the non-real time users are allocated resource blocks using the channel condition based marginal utility function. When there are no sufficient resources to allocate, it allocates the resources of bad channel users there by degrading their service. We have designed the network topology with G (n) and B (n) for representing the available good and bad channels. We investigate the performance degradation when the real time, Non real Time, video and VOIP environments based on RSS threshold value. Comparison is made with the VOS in terms of the parameters like throughput, bandwidth, delay, fairness and rate. Our proposed method provides good performance and quality. From our simulation results we show that this admission control algorithm provides channel quality and prioritizes the handover calls over new calls which allocates resources to all kinds of users.

**Key words:** CBR, CAC, Long Term Evolution (LTE), Quality of Service (QoS)

## 1. INTRODUCTION

### 1.1. Introduction-Long Term Evolution (LTE) Networks

LTE is homogeneous to 3GPP. When compared to the current radio access technologies, the newly evolved radio access technology (LTE or super 3G) is capable of providing equivalent service without exceeding the current fixed line accesses in an economic way. The data rate and the spectral efficiency of LTE can be increased effectively due to the involvement of Orthogonal

Frequency Division Multiple Accesses (OFDMA) and Multiple-Input Multiple-Output (MIMO) technologies (Bae *et al.*, 2009).

### 1.2. Design Goals of LTE

New and advanced mobile broadband services can be provided to LTE since they are capable of providing higher data throughput to mobile terminals. Below are few objectives of LTE

- When compared to the 3G evolution like HSDPA and enhanced uplink, LTE provides significantly

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higher data rates up to 100 Mb/s and 50 Mb/s for the downlink and uplink respectively

- Related to the 3GPP Release 6 (Rel-6) systems, three to four times higher average throughput need to be provided by LTE. Also, two to three times higher cell-edge throughput should be provided by LTE compared to the HSDPA or enhanced uplink.
- The spectrum efficiency of LTE needs to be three times more efficient compared to current standards
- LTE needs to provide a significant reduced control and user plane latency such that user plane RAN round-trip time is lesser than 10 ms and channel setup delay is lesser than 100 ms
- The cost of operator and end user should be reduced efficiently (Atayero *et al.*, 2011)
- Relocation into other frequency bands can be smoothly provided by using spectrum flexibility and by facilitating deployment in different spectrum allocations. The cellular technologies such as GSM and IS-95 are the second Generation (2G) systems operating in a different frequency

### 1.3. Call Admission Control in LTE Networks

The eNodeB in LTE provides basis for the admission control algorithm and is capable of operating separately on a per cell basis (Spaey *et al.*, 2010). Congestion avoidance is the main aim of CAC scheme which limits the number of ongoing connections in the system or denies new connection request so that QoS can be maintained and delivered to different connections at the target level. The below two conditions need to be satisfied in the CAC algorithm in order to admit the user to the network (Antonopoulos and Verikoukis, 2010).

### 1.4. Good Signal Strength

Since eNB provides maximum signal, the mobile selects this node and shortage in coverage can be caused when signal goes below a certain threshold. The mobile may get blocked in this situation.

### 1.5. Resource Availability in the Selected eNB

Huge amount of physical resources between a minimum and maximum threshold are provided by the mobile. Available resources are checked by the eNB once the initial condition is checked. Call gets blocked once the eNB goes below a minimum resource threshold.

### 1.6. Scheduling in LTE Networks

In LTE systems, there are three main groups in scheduling algorithms which includes persistent

scheduling algorithm group, the dynamic and the semi-persistent one. In the persistent scheduling algorithm the persistent scheduler assignments to the UEs are predefined. So during DL period and when the eNB assigned resources to them, the UEs need to listen to a group of predefined resources. There is no necessity to specify UEs for every DL period in this kind of scheduling which seems to be a major advantage. Here the UEs can find the assigned RBs and the kind of modulation and coding scheme used is identified. But in the dynamic scheduling scheme, the scheduling decisions are taken for every DL period. The CQI feedbacks from the UEs are considered for scheduling decisions and there are chances for DL to change to other period (Parruca and Abt, 2011).

Types of scheduling algorithms in LTE includes:

- Max-Prod Scheduling Algorithm
- Max\_Sum Scheduling Algorithm
- Round Robin-Max\_Sum Scheduling Algorithm
- Multi Groups- Max\_Sum Scheduling Algorithm

The users can be scheduled in two dimensions namely time and frequency which is considered as the key feature of packet scheduling in LTE networks. For resource management, the aggregate bandwidth available is divided in subcarriers of 15kHz. A sub channel with a bandwidth of 180kHz can be formed by grouping twelve consecutive subcarriers (Dimitrova *et al.*, 2010).

### 1.7. Problem Identification and Proposed Solution

In our previous work (Franklin and Paramasivam, 2012) we have proposed call admission control algorithm for LTE networks. The call requests are classified into Handoff Call (HC), New Call (NC), VoIP call and Video type and prioritized. After the classification of the call requests, the channel estimation technique is based on the Received Signal Strength (RSS) value. When a call request arrives to the network, it is checked for HC or NC. If it is a HC, then it is handled first by the scheduler. After classifying the call as HC or NC, the scheduler checks for its class. If it is a VoIP call, then its bandwidth requirement is checked. If it is less than total available bandwidth, the bandwidth can be reserved based on the traffic density of the base station. For video calls, if the requested bandwidth meets the remaining available bandwidth, it can be admitted. If there are multiple video call requests, then the Tolerance of Latency (TOL) of each call is checked. The call with low TOL can be admitted first.

**1.8. Related Work**

Dimitrova *et al.* (2011) have discussed the mutual interference scheduling and inter-cell interference has on each other. It has been discussed that the particular service policy used by the scheduler is the basis for inter cell interference pattern. The impact of inter-cell interference on user performance at flow level is examined in this study.

Dimitrova *et al.* (2011) have presented a performance comparison of two distinct scheduling schemes for LTE uplink (fair fixed assignment and fair work-conserving) taking into account both packet level characteristics and flow level dynamics due to the random user behavior.

Piro *et al.* (2010) have proposed a novel two-level scheduling algorithm. At the upper level, discrete time linear control theory is the basis for this novel approach. A proportional fair scheduler is customized at the lower level. The performance and the complexity of the proposed scheme have been evaluated both theoretically and by using simulations.

Makara and Ventura (2011) have proposed a scheme that is optimized for offering improved quality of service for a diverse mix of traffic including real-time VBR traffic in the downlink of LTE networks. The application quality of service requirements can be satisfied and the overall system throughput can be improved in this algorithm using multiuser diversity. The average delay experienced by the real time packets in network needs to be minimized and the users in the sub channels which experience the best link quality are scheduled in order to imply higher data rates.

Yaacoub and Dawy (2011) have proposed a pricing-based power control scheme was in the presence of BS cooperation. The interference mitigation schemes were implemented in conjunction with a low complexity scheduling algorithm.

**2. MATERIALS AND METHODS**

**2.1. Resource Block Allocation for Non VoIP Users**

We consider B resource blocks for each TTI for K mobile users which are serviced by an eNodeB. Among these users, W users run an application with active connection to a server. This server is connected to the packet data network which is connected to eNodeB. This system includes the following parameters:

- di: The average data rate achieved by the ith user at a time t when a scheduling decision is to be made.
- Di: The minimum required data rate for the ith user at t.
- ti: The playback delay threshold for the ith user; maximum allowable time before the user’s head of line packet in its queue can be delivered.
- brb(i): Number of resource blocks allocated to the time variable bit rate user in one TTI.
- bn<sub>rb</sub>(i): Number of resource blocks allocated to a non real-time user.
- η<sub>i</sub>: The effective data rate of the ith user computed from the utility function of all the subcarriers (as if the user was allocated the whole of the system’s available band).

The number of resource blocks allocated to real time VBR users is then is determined as:

$$B_{rb}(i) = \left( \frac{\mu}{\mu + \lambda} \right) \left| \frac{\frac{1}{T_i} \cdot \left( \frac{d_i}{D_i} \right)}{\sum_{n=1}^W \left( \frac{1}{T_n} \right) \cdot \sum_{n=1}^W \left( \frac{d_i}{D_i} \right)} \right| B \tag{1}$$

In Eq. 1 the resource blocks for real time user are assigned with the ratios of di and Di with respect to time of ith user. The network’s operator assigns the parameters μ and λ. These parameters are selected such that the ratio signifies the amount of real time users flowing through the network and amount of non-real time users flowing through the network. For non real time traffic, the following rule is applied in the determination of resource blocks allocated:

$$B(n)_{rb}(i) = \left( \frac{\lambda}{\mu + \lambda} \right) \left| \frac{\eta_j \cdot \left( \frac{d_j}{D_j} \right)}{\sum_{n=1}^W (\eta_n) \cdot \sum_{m=1}^W \left( \frac{d_j}{D_j} \right)} \right| B \tag{2}$$

In Eq. 2 derives the reservation block B(n) based on the availability of the resources in the eNodeB. In this the utility factor for the subcarriers are determined to formulate the bandwidth reservation. Few resource blocks may be unallocated since both the rules have components that are rounded down. In order to improve the system’s overall throughput users with highest utility function in each block is used for allocating the remaining blocks (Makara and Ventura, 2011).

### 2.2. Selection of Utility Function

For ensuring channel quality, here we consider the utility function with RSS Hence, the utility function used in resource assignment for real time and non-real time users is given by:

$$Y_N(E_N(\rho_N, \delta_{sc,N})) = \frac{E_N(\rho_N, \delta_{sc,N})}{RSS_N} \tag{3}$$

Where:

- $Y_N$  = Utility function of user N
- $E_N$  = Rate
- $\rho_N$  = Transmit power on the subcarrier
- $\delta_{sc,N}$  = Set of subcarriers

In Eq. 3 shows the  $RSS_N$  is the received signal strength achieved by user N over the last T TTIs. The utility function is calculated based upon the set of subcarriers, transmit power and the Received signal strength.

Marginal utility calculation:

$$M_{N,c} = Y_N(E_N(\rho_N, \delta_{rb,N} \cup \{c\})) - Y_N(E_N(\rho_N, \delta_{rb,N})) \tag{4}$$

In Eq. 4 derives the marginal utility function for the sub carriers. In this  $\delta_{rb,N}$  = set of resource blocks and the marginal utility  $M_{N,c}$ , represents the gain in the utility function  $Y_N$  when RB c is allocated to user N, compared to the utility of user N before the allocation of c (Yaacoub and Dawy, 2011).

## 3. RESULTS

### Algorithm:

Consider the n user requests  $\{R_1, R_2, \dots, R_n\}$ .

Let us consider the user requests with good channels as  $G = \{G_1, G_2, \dots, G_k\}$  and bad channels as  $B = \{B_1, B_2, \dots, B_r\}$ , where  $k, r < n$ .

Among G, handover calls are represented as  $H = \{H_1, H_2, \dots, H_m\}$  and new calls as  $N = \{N_1, N_2, \dots, N_p\}$ , where  $m, p < k$ .

Among H, the VoIP calls and the video calls are represented as  $H_{v0} = \{V_1, V_2, \dots, V_q\}$  and  $H_i = \{I_1, I_2, \dots, I_t\}$  respectively, where  $q, t < m$ .

Among H, the real time users and the Non real time users are represented as

$H_R = \{K_1, K_2, \dots, K_t\}$  and  $H_N = \{S_1, S_2, \dots, S_t\}$  respectively.

**Table 1.** Simulation parameters

No. of Servers	1
No of aGw	1
No of eNB	1
No. of UEs	5
Traffic Types	CBR, Video and VoIP
Traffic Rate	10 to 50 kb
VoIP Codec	GSM.AMR
No. of VoIP frames per packet	2

Let necessary RSS condition for satisfying handover be  $RSS_v$  and the RSS threshold value be  $RSS_L$ .

Let  $\eta_A$  be the total available bandwidth,  $\eta_{vot}$ ,  $\eta_{it}$ ,  $\eta_B$ , be the reserved bandwidth for VoIP, video and bad channel classes, respectively.

Let  $M_{N,c}$  be the marginal utility function.  $\delta_{avai}$ ,  $N^{(c)}$  be the set of available users,  $\delta_{rb}$ ,  $N^{(c)}$  be the set of resource blocks. c is the user and c-1 is previous user. we simulate the proposed Utility Based Scheduling and Call Admission Control (UBSCAC) scheme using Network Simulator (NS2) which is a general-purpose simulation tool that provides discrete event simulation of user defined networks. We have used the LTE/SAE implementation model for NS2 (Qiu *et al.*, 2009). The simulation parameters are given in **Table 1**.

In the simulation settings, we have one server to provide HTTP, FTP and signaling services, one aGw to provide HTTP cache and flow control, one eNB to provide flow control information and five UEs. In this model, ULAirQueue is used for uplink flows in the link between UE and eNB. For the downlink flow, (ie) in the link between eNB and UE, DLAirQueue is used. For both the links, the link bandwidth is set as 500kb and link delay as 2ms.

For the link between eNB and aGw, ULS1Queue is used and for the downlink between aGw and eNB, DLS1Queue is used. For both the links, the link bandwidth is set as 5 Mb and link delay as 2ms. For the link between the server and aGw, a simple DropTail queue is used with link bandwidth as 50Mb and link delay as 2 ms. We compare the UBSCAC scheme with the VBR-Optimised Scheduler (VOS) scheme (Makara and Ventura, 2011).

## 4. DISCUSSION

### 4.1. Case-1 (CBR)

Based on rate: In this experiment, we vary the data sending rate from 10-50 kb to measure the received bandwidth, fairness, throughput and delay for the CBR non-real time traffic.

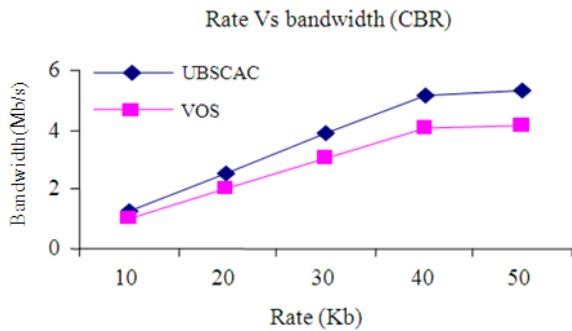


Fig. 1. Rate Vs bandwidth

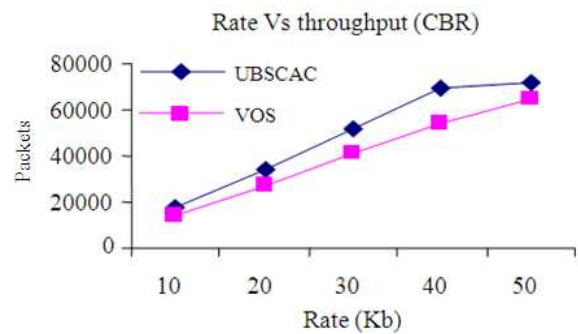


Fig. 4. Rate Vs throughput

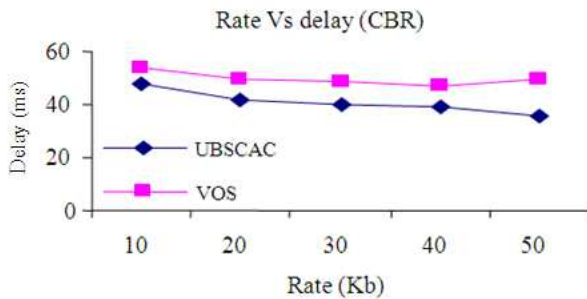


Fig. 2. Rate Vs delay

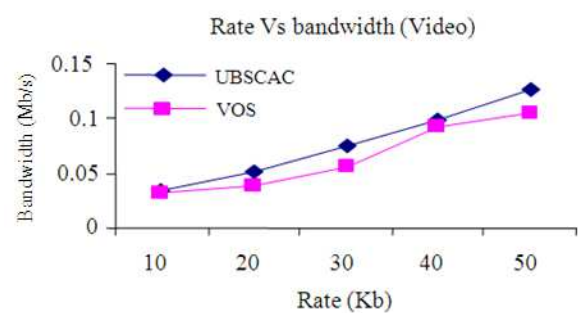


Fig. 5. Rate VS bandwidth

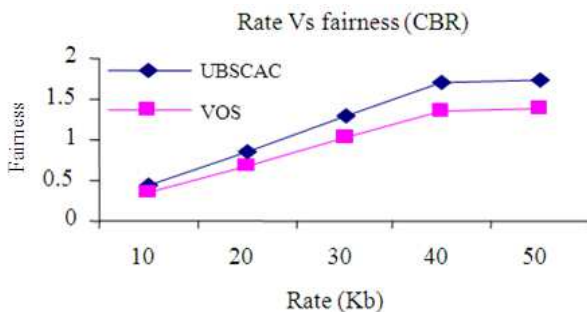


Fig. 3. Rate Vs fairness

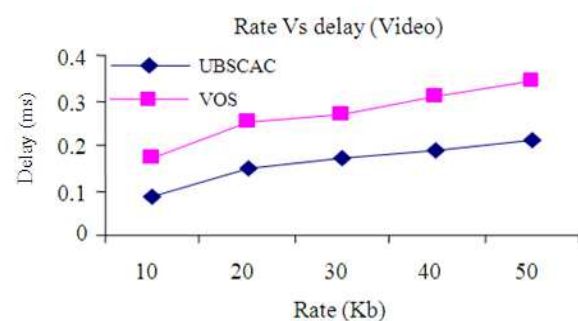


Fig. 6. Rate Vs delay

It can be seen from Fig. 1, the received bandwidth gradually increases when the rate is increased. We can see that the received bandwidth of the UBSCAC is higher than the existing VOS scheme. From Fig. 2, we can see that the delay of the proposed UBSCAC is less than the existing VOS scheme. Figure 3 shows that UBSCAC provides better performance over VOS when the environment is in CBR. Figure 4 shows that throughput of UBSCAC is higher than existing VOS scheme.

#### 4.2. Case-2 (Video)

Based on rate: In this experiment, we vary the data sending rate from 10-50 kb to measure the received bandwidth, fairness, throughput and delay for the Video exponential traffic.

From Fig. 5, we can see that the received bandwidth of the proposed UBSCAC is higher than the existing VOS scheme. From Fig. 6, we can see that the delay of the proposed UBSCAC is less than the existing VOS scheme.

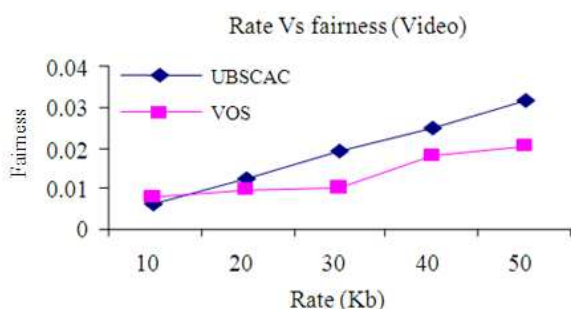


Fig. 7. Rate Vs fairness

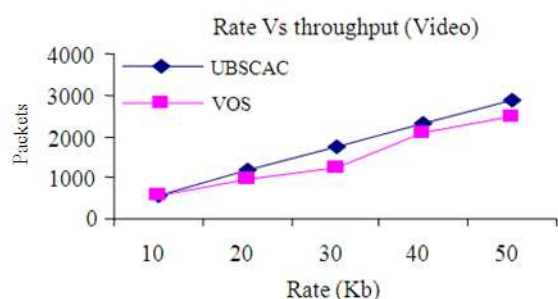


Fig. 8. Rate Vs throughput

Figure 7 and 8 show the fairness and throughput obtained, respectively for the UBSCAC and VOS schemes. From the statistics, it can be seen that, the throughput and fairness of both schemes are increased, when the rate is increased from 10-50 kb.

## 5. CONCLUSION

In this study, we have proposed to design a call admission control algorithm which schedules the channels for Real time and non-real time users. The call requests are classified into New Call (NC) request and Handoff Call (HC) request and the type of services are classified as VoIP and video. Then based upon the Received Signal Strength (RSS) value, the channel is estimated as good channel or bad channel. Resource allocation is made for VoIP calls based on traffic density and for video calls, it is done based on the tolerance of limit. For allocating resources to other users, utility function is calculated based on the channel condition. Then the real time users and the non-real time users are allocated resource blocks based upon the highest marginal utility function. When there are no sufficient resources to allocate, it allocates the resources of bad channel users there by degrading their service. Thus

from our simulation results we show that this admission control algorithm provides channel quality and prioritizes the handover calls over new calls which allocates resources to all kinds of users.

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