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Optimal Packet Scheduling Scheme for LTE Systems Based on Scheme Utility Function

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Abstract: In the early 1980's evolution of wireless communication has witnessed remarkable growth in increasing data rates ever since its evolution. Long Term Evolution (LTE) is shaping as the future of next generation technology to satisfy the demand for growing data rates. An enhanced scheduling scheme is very essential in achieving an effective Quality of Service (QoS) Provisioning in LTE systems because it plays a significant role in distributing the wireless resources of the system among mobile users effectively. It also managing and guarantying the QoS requirements of the network responsibly. This research offers more flexibility in developing the network to design an appropriate scheduling algorithm. This study proposed a new dimension of packet scheduling scheme to made scheduling decisions using the channel quality conditions without ignoring the fairness criteria. Unlike the traditional schemes this optimum scheduling scheme for LTE system also supports multiple users with different QoS requirements within same class. The same QoS requirements for different users within each class are assumed in this scheme. The scheme also ensures maintaining a high QoS with a good throughput-fairness trade-off ratio.

Key words: LTE, QoS, bandwidth, throughput, India

INTRODUCTION

LTE is a new wireless technology which uses data communication evolution with the GSM/UMTS standards which improves the capacity and speed of wireless data networks by introducing a new techniques in digital signal processing and modulations techniques (Baker *et al*., 2011; Dahlman *et al*., 2003). A new scheme on novel packet-scheduling for LTE is introduced in this study.

The main aim of LTE system is to support and handle the maximum number of Mobile Stations (MS) in the network. More number of channels are divided from the total bandwidth or spectrum and they are shared for the MS. The shared channels support for higher number of MS, unlike circuit switching and thus improves the performance and utility of the network and QoS provisioning can be done. At three levels in the LTE systems (Ang *et al*., 2015; Zou *et al*., 2013; Al-Manthari *et al*., 2007), called as admission level, class level and packet level as shown in Fig. 1. Admission-level QoS provisioning is experienced by introducing a proper Call Admission Control (CAC) procedure. The CAC algorithm is very important as they are responsible for accepting or rejecting a new call request by the MS. The CAC always aims to maximise the number of admitted users, by satisfying the QoS requirements of the existing MS simultaneously. The new MS can be used only if there is enough amount of bandwidth in the network to tackle the QoS requirements for the new calls. Class level QoS provisioning means the procedure for scheduling in the system. Packet scheduling is the procedure of resolving contention for shared resources in a network. This process determines the transmission order and also allocates the bandwidth among users. Scheduling algorithms is selected for the system based on QoS system and the type of users in the network. It determines the frames with maximum time to be allotted for every every traffic class in order to solve the QoS requirements of each user. Packet level QoS provisioning is used to process the packets that are allotted for transmission in a frame with particular time:

- To provide the effective QoS provisioning in the network and also to improve the throughput of overall system
- In order to provide a better distribution of spectrum in the wireless resources
- As every class has its own QoS requirement it is used for supporting the traffic for different classes
- Congestion preventing
- Considering the functions of the network operators which are used

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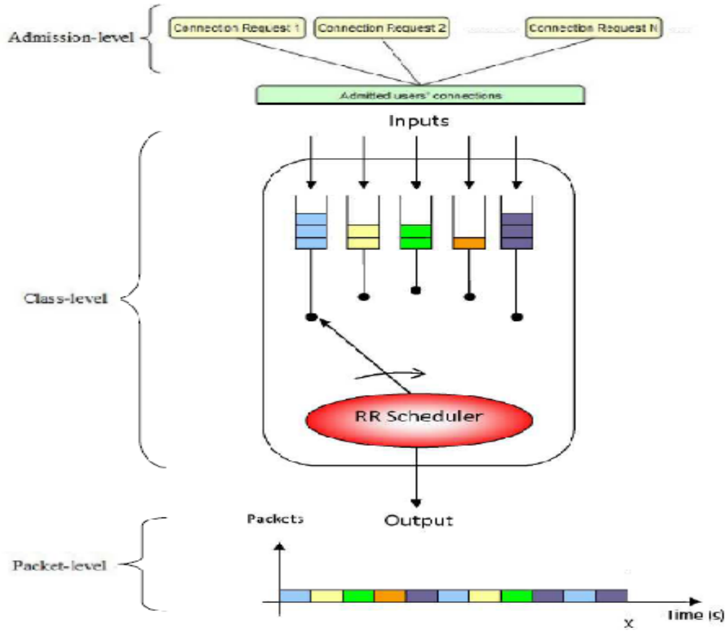

Fig. 1: Levels of QoS provisioning

The major disadvantage of the traditional LTE scheduling schemes is that the same QoS requirement are assumed to be used by different users within every class. Hence, the proposed scheme for optimal scheduling in LTE system supports multiple users, unlike the traditional scheme, it uses different QoS requirements within the same class. This is practical based approach as each and every traffic class in LTE systems can include various services based on QoS requirements. Another problem is that the decisions for scheduling are made by using the channel quality conditions without omitting the improved technique. An another important feature that has to be considered in this discussed design is that the scheduling algorithm is the adoption of the quality conditions inoder to maximize the capacity of the users channel of LTE systems (Zhang *et al*., 2011; Jiang *et al*., 2006; Kawan *et al*., 2009; Piro *et al*., 2011a, b; Gozalvez, 2011). In order to solve the problem in a right step, the Mobile Stations (MS) uses higher data rate with better channel quality conditions without ignoring the improved technique. As here the MS about to move away from Base Station (BS), therefore some changes coding scheme and in modulation towards higher modulation format and lower coding rate takes place. This occurs due to the channel quality that tends to degrade when the MS moves away from BS. Therefore the quality of the channel will be affected by distance and other physical parameters. Finally, the penality is undergone as the MS is located away from BS with lower channel quality conditions (Jiang *et al*., 2006; Zhang *et al*., 2011; Jiang and Liu, 2006; Piro *et al*., 2011; Gozalvez, 2011; Pedersen *et al*., 2011; Alasti *et al*., 2010). Thus the scheduling scheme proposed is designed to re-allocate the resources in a dynamic and with better manner without considering the location of the MS and quality of the channel.

MATERIALS AND METHODS

Scheduling scheme for scheme utility function: In this study, we present the newpacket scheduling scheme. The scheduling scheme which is proposed is developed to handle three different types of traffic from users, they are traffic which occur with minimum delay, traffic which have minimum data rate and best effort traffic. It also checks the upper and lower bounds on QoS that are provided to the users. This scheduling scheme is mainly developed for the downlink resource management. For the sake of clarity in the concept, we first introduce the definitions and parameters that are used here. This scheme is developed using network calculus (Alasti *et al*., 2010; Boudec and Thiran, 2004). We consider the ‘S’ classes of traffic where class i has higher priority traffic than class i+1. Different methods are used for the Users within the same class but with different QoS requirements. We Assume that a BS has ‘N’ user connections and if Nk is the number of user that are supported with ‘i’ class of traffic, then the expression will be as follows:

$$N = \sum_{K=1}^{n} N_K \quad (1)$$

After the arrival of call, the LTE system receives the traffic from upper layers in the form of IP packets and they segments it into fixed size Protocol Data Units (PDUs). These PDUs are connected correspondingly and are stored in the transmission queue. Every queue can be defined by two variables. They are:

Quantum value: This is the average amount of bytes served for each round.

Deficit counter: This is a counter which is used to view the number of bytes that are consumed in a queue during transmission for every round. And it is initialized to the quantum value.

There are 'n' number of queues and every queue has a deficit counter which is initialized with a quantum value. Like round robin method, every non-empty queue is given when the value of deficit counter is greater than zero. A queue can be given only when the value of deficit counter is positive; else it is not possible to give a queue when the value of counter is zero or negative. For every round, the value of deficit counter is incremented by a quantum value. At least one packet should be served for each round with a condition that the size of quantum should not be smaller than Maximum Transmission Unit (MTU). If this condition is satisfied the scheme utility function is invoked. The user requesting for a service is given by the scheme function utility:

$$D_{i,\,j} = \sup_{N}\left[\frac{\alpha_i^z(t)+\alpha_j^z(t)}{M}\text{-}t\right] \tag{2}$$

where, $\alpha_i^z = \{\alpha_i^z(t), \alpha_i^2, ..., \alpha_i^{z\text{-}1}(t)\}$ are the various QoS measurement matrics like average delay,throughput,data rate, etc,…, $\alpha_i^z(t)$ is the fairness measurement function. is the set of users.

The main aim of the scheduling scheme is to increase the number of users in the system and to achieve the LTE system that satisfies the following optimization problem:

$$\max_{(i,\,j)\in N},\sum_{i\,=\,1}^{K}\sum_{j\,=\,1}^{N}\sup_{(i,\,j)\in N},\left[\frac{\alpha_i^z(t)+\alpha_j^z(t)}{M}\text{-}t\right] \tag{3}$$

where, j represents the number of users in every class. The scheduling scheme utility function is expected to satisfy the following conditions:

- The scheme utility function should be wide sense increasing function of $\alpha_i^z(t)$, i.e., it should satisfy

$$\sup D_{i,\,j} = \inf S_{Di,\,j} \tag{4}$$

- If the measurement of QoS for all the users are either maximum or minimum then the scheme utility function will also be either maximum or minimum and that is given by Eq. 5 and 6:

$$\sup_{N}\left[\frac{\alpha_i^z(t)+\alpha_j^z(t)}{M}\text{-}t\right] = D_{i,\,j\,min}$$

If, $D_{i,j}(t) = \alpha_i^{z,\,min}(t)+\alpha_j^{z,\,min}(t)$:

$$\sup_{N}\left[\frac{\alpha_i^z(t)+\alpha_j^z(t)}{M}\text{-}t\right] \le D_{i,\,j\,min} \tag{5}$$

$$\sup_{N}\left[\frac{\alpha_i^z(t)+\alpha_j^z(t)}{M}\text{-}t\right] = D_{i,\,j\,max}$$

If, $D_{i,j}(t) = \alpha_i^{z,\,max}(t)+\alpha_j^{z,\,max}(t)$:

$$\sup_{N}\left[\frac{\alpha_i^z(t)+\alpha_j^z(t)}{M}\text{-}t\right] \ge D_{i,\,j\,max} \tag{6}$$

Here the packets are classified into various classes at the packet schedular. The scheme utility function is expected to satisfy minimum delay constraint for traffic with minimum delay type that is given by Eq. 7:

$$D_{i,j}(t) \ge \left(\frac{R_{ij}}{C}\text{-}S_{ij}\right) \qquad 0 \le S_{ij} \le 1 \tag{7}$$

Where:

S_{ij} = Average data rate
R_{ij} = Maximum through put
C = The normalized system capacity

The scheme utility function is expected to satisfy the condition for traffic with minimum data rate:

$$D_{ij} = \int_0^t \frac{S_{i,j}}{\sum_{j\in C}} 1_{(i\in c)}\,ds \tag{8}$$

In Eq. 8, $1_{(i\in c)}$ is equal to 1, if the condition is satisfied else 0. This value 0 indicates that the call is blocked for need of sufficient resources. For the scheme utility function to satisfy the BE effort traffic is given by:

$$D_{ij} = \sup_{S_{ij}:\, S_{ij} < C} \{R_{ij} - R_{ij}(t)\} \quad (9)$$

If once the scheme utility function is satisfied, by verifying the default queue, the algorithm starts the initialization process. For a non-empty queue the scheduling process will be invoked. The technique called Type of Service (ToS) based scheduling is introduced in this algorithm. If the scheduling scheme is invoked, initially it functions like a normal Modified Deficit Round Robin (MDRR). In the first round, the pointer moves from queue 1 to n by verifying the queues having quantum values that is less than the deficit counter. In the network, the capacity of queue will vary with time and that is given by Eq. 10 and 11 where 'h' denotes the packets in queue at time t, 'f' is the packets arrival at time t and 'g' is defined in LTE which is the percentage ratio of Multiple Transmit Multiple Receive (MTMR) and Total System Capacity and is 'h' the number of packets being processed:

$$(f+h)(t) = f(t)+h(t) \quad (10)$$

$$(f \wedge h)(t) = f(t) \wedge h(t) \quad (11)$$

At the end of the first round, the new quantum value Carrier to Interference plus Noise (CINR) value is obtained from the channel by calculation. The value of CINR is substituted in Eq. 12 to find new quantum value where $w(x, g) = g + \Pi(x)o$ is found after extensive simulation, A is Maximum Transmission Unit and $\Pi(x)$ is the CINR assigned to every round that varies with a function of distance x. The quantum value is calculated from Eq. 12:

$$q_i = A \wedge 512 * w(x, g) \quad (12)$$

If once the quantum value is found, the MDRR algorithm is executed with the modified quantum value and that is given by Eq. 13 where d_i is the deficit counter value:

$$q_i = q_i \wedge d_i \quad (13)$$

If d_i>0. Do serve packet:

$$d_i = d_i - C \quad (14)$$

where, C is the packet size for different service flows which is the variable. Further, for a particular region, the CINR value is compared with the pre-assigned threshold λ. Change of modulation takes place, when the CINR value is below the threshold value.

If $\Pi(x)<\lambda$ dulation technique may result in change in the data rate either increase or decrease based on the movement of mobile station either moving toward or away from the BS. The changes in CINR may be attributed temporarily due to various factors and before switching to the other modulation schemes, the continuous monitoring value of CINR over a period of time is needed. During this waiting time u, the packets may be lost and therefore reduction in the weight of the packets will deliver properly even data rate is less. Packet delivery is given by Eq. 15:

$$(f \varnothing h)(t) = \inf_{u \geq 0}\{f(t+u) - g(u)\} \quad (15)$$

If the queue is null or empty $(f\varnothing h)(t)=0$, t<0, it is not allowed in the next round. In the next round, only the non-empty queues are allowed to service and that is given by Eq. 16:

$$(f \otimes g)(t) = \sup_{0 \leq s \leq t}\{f(t-u) + g(u)\} \quad (16)$$

Before change in modulation, the waiting period happens for five rounds of servicing might also makes the service station complex. For the above specified problem a unified modulation technique can be used which produce best result.

Based on the quantum value, the algorithm has to process the packets and assure the proper delivery of packets and that is defined by Eq. 17:

$$(h \bullet h')(t,u) = \inf_{u \in s}\{h(t, u) + h'(t, u)\} \quad (17)$$

The mobile station in the outer region or which is farthest away from the base station will have the lowest CINR value. In order to obtain the best CINR value is given by Eq. 18:

$$\Pi(x)(t) \inf\{h_i(t, s) + x_i(s)\}\} s \in t1 \leq i \leq J \quad (18)$$

These mobile station requires a robust modulation scheme that might decrease the data rate. Hence, the far off stations with low coding rates will be penalized. Change in modulation scheme is given by Eq. 19:

$$\overline{(\Pi \wedge h_n)} = \overline{(h_n \bullet \Pi)} h_n \quad (19)$$

Simulation process: It is necessary to study about the real life scenarios in order to analyze QoS in a network. The simulation set up is more important inoder to obtain the actual deployment of the LTE system. There are multiple MS in the range of a base station. The base station is located at the center of the cell and connected to the core of the network.

There is a four servers which acts as backbone for this scheme. Four types of servers includes an application server, database server, voice server and Video server. During simulation the base station is set to operating mode and the performance is checked on taking any MS. The MS transmission power is adjusted to 32 mW and the BS transmission power is adjusted to 10 W. The retain time of the resource during handover is 200 m sec. The BS duration for scanning is 5 frames and interleaving interval is 240 frames. OFDMA-1024 is configured in the simulation. It is called as the OFDMA 1024 FFT mode which says that there are 1024 subcarriers that are available for a single channel use. Here, the voice traffic is PCM. H.264 format is used for video conferencing.

The path-loss and multipath model are set to 60 kmph speed for a vehicular move. Here the signal propagates before reaching the receiver when each path loss model is appropriate for a certain kind of environment. As a path loss model, the suburban fixed vehicular environment is chosen. The hilly terrain with moderate-to-heavy tree densities model is adjusted for the terrain type of the suburban fixed path loss model. Shadow fading is one of the additive correction (in dB) which is experienced by the received signal to the path-loss. The term shadow fading is introduced in order to correct site-specific departure from the generic path-loss model, due to obstructions in the signal path. Poisson process is modeled for the user connection arrival. The simulation takes step time as one time frame which is 2 m sec in LTE and the simulation time taken is 600 s.

RESULTS AND DISCUSSION

In this study, using OPNET simulator, the performance of the proposed scheduling algorithm in mobile LTE system is evaluated. The ability of the proposed scheduling scheme is demonstrated in order to support different classes with users having different requirements in QoS. There are two different classes namely, VoIP (class i) and video streaming (class i+1) that are considered.

Observation on voice scenario: In order to demonstrate the improvement of our scheme, the throughput of the LTE system is considered. Assume that the class i have higher priority than class i+1. Observations are made on the performance of projected algorithm over predictable algorithm over the voice packets. The throughput of a particular service is measured by average traffic received. Figure 2 and 3 shows the performance of the scheme in terms of average traffic received.

Here the voice packets that is received at the receiver shows that the algorithm which is proposed achieves >100% efficiency than the predictable algorithm. Here the adaption of a dynamic weight assigning strategy is taken in the proposed scheduling algorithm. It is used to ensure the call that admitte only when there is sufficient resources. In the proposed scheduling algorithm the streams of voice data can be sent over high quality channels but in the predictable algorithm any channel can be allotted to the user.

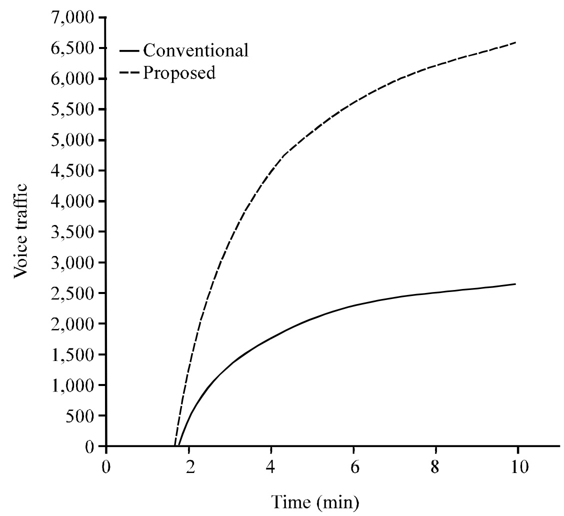

Fig. 2: Average traffic received; time-average (in voice. taffic received (bytes/sec))

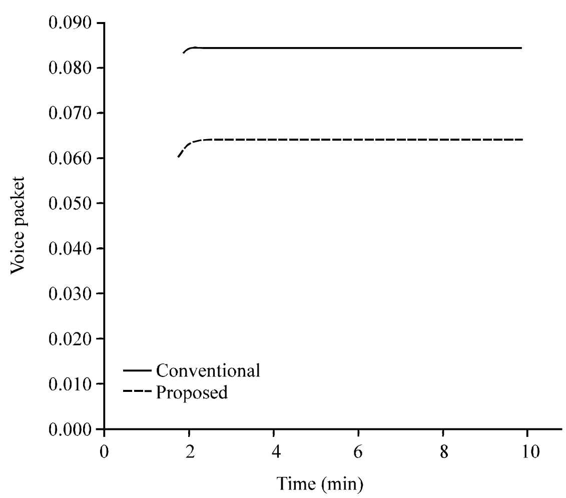

Fig. 3: Packet end-to-end delay; time-average (in voice. packet end-to-end delay (sec))

Packet end to end delay: In end-to-end delay a packets are travelled from source to destination across the network. Figure 4 shows the packet end-to-end delay algorithms. From the plot we can obtain that the packet end-to-end delay is much lower for the projected algorithm than the predictable algorithm.

Jitter: In a time interval t1 and t2, the two consecutive packets leaves the source node and reaches the destination node at time t3 and t4, then; jitter = (t4-t3)-(t2 -t1). If the time difference between the packets at the destination node is less than the source node, then it is indicated by negative jitter.

MOS: MOS means the Mean Opinion Score which means the measurement of the quality of the reconstructed voice signal. MOS is used to give the quality score for the reconstructed signal that ranges from 1 (worst) to 5 (best).

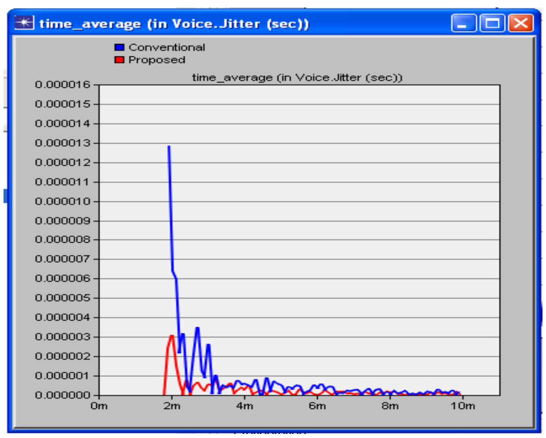

Fig. 4: Voice jitter in seconds

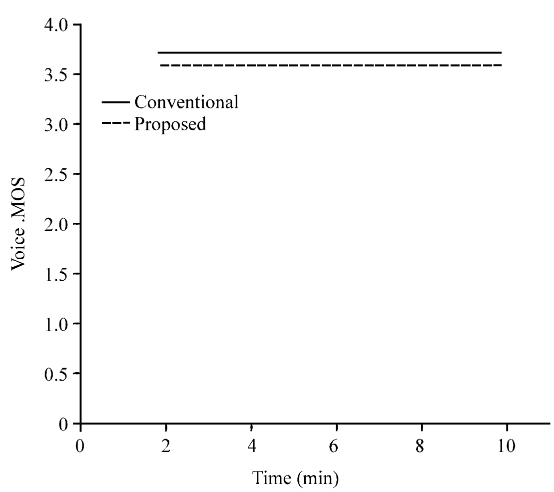

Fig. 5: MOS value for voice environment; time-average (in voice.MOS value)

When there is any loss in packet, the MOS score on a VoIP network will be reduced. The score will be further reduced for excessive delays. Figure 4 and 5 shows the action of jitter and MOS by voice for both algorithms. But here the increase in throughput is not achieved for the voice packets which is due to the parameters like Jitter and MOS shown in Fig. 5 and 6.

Observation on video scenario: Figure 7 shows the packet delay variations which takes place in algorithms It is obvious from the above graph that the packet end-to-end delay is much lower for the projected algorithm compared to the predictable algorithm.

Overall system throughput: Overall LTE system throughput from Fig. 8 shows that the effective channel

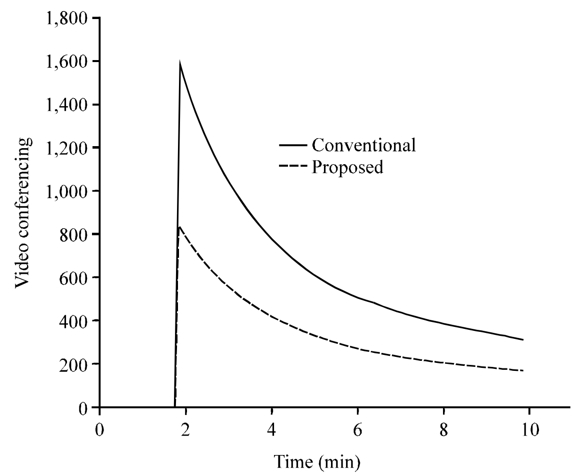

Fig. 6: Average video traffic received in bytes/sec; time-average (in video conferencing.trffic received

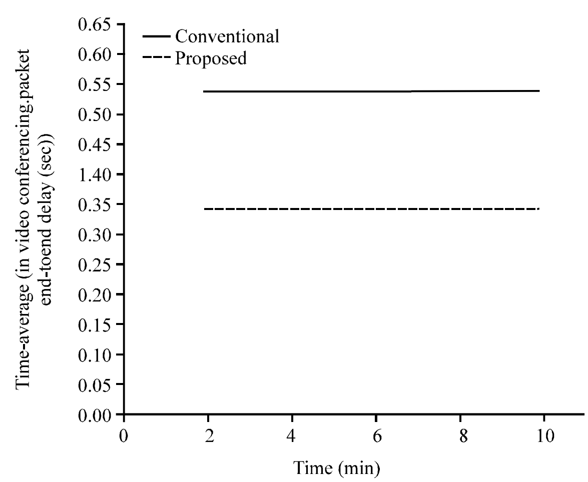

Fig. 7: Packet end-to-end delay

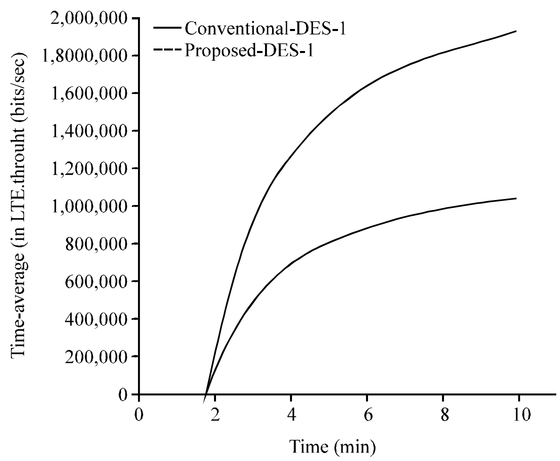

Fig. 8: Overall LTE system throughput

assignment strategy adopted in the projected algorithm increases the overall throughput by >50% when compared with the predictable algorithm in the network which proves that the projected algorithm is very efficient in running the resources in the network. Thus it is observed that the conditions of the channel quality increases the throughout f the network.

CONCLUSION

In a novel development scheme for LTE systems, this gives better improvement for all the users in a cell. The diverse QoS requirements for mobile users are satisfied simultaneously by the use of scheme utility functions which employs practical economic models. It also improves the performance of operators. Thus the performance of the projected algorithm is better than the predictable algorithm when compared with its performance metrics. OPNET is used to evaluate the performance of our projected algorithm and it is used to find the performance of our projected algorithm in a real-time LTE switch.

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