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A Hybrid Scheduling Algorithm for Super 3G Enhanced (S3G-E) Network

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***Abstract*– A 3G LTE (long term evolution) has been defined by 3GPP, known as super 3G (S3G) network. This evolution enables the operators to transfer IP-based mobile broadband services at much higher speed. This project introduces some enhancements in S3G technology known as (S3G-E) and then its scheduling algorithms are proposed and investigated in multiuser OFDM environment. This work is intended to design a S3G-E scheduler with service differentiation as a frame of reference. A two layered scheduler is proposed. The first layer sorts the different FC-IND based on their policy profiles (committed rate, peak rate), the second layer makes a decision to choose which user to be served in a given slot. This work is practically investigated under traffic models involving file transfer, video streaming, web browsing and Voice over IP (VoIP) traffic models.**

***Index Terms*– Super 3G Enhanced Network (S3G-E), Service Differentiation (DiffServ), Flow Class Indicators (FC-IND), Quality of Service (QoS), Queuing Delay, Fairness and Throughput**

I. INTRODUCTION

3G wireless systems have now been installed on a prominently wide scale all over the world. The first step towards the development of wideband CDMA (WCDMA) was taken by 3GPP (Third Generation Partnership project). They introduced the concept of Enhanced Uplink (E-UL) [1] and High Speed Downlink Packet Access (HSDPA) [2]. These technologies highly contribute 3GPP towards the evolution of an efficient radio access network. However, with the evolving expectations and requirements of users and operators, new technologies are continuously emerging in order to fulfill these requirements. Thus it is important for 3GPP to consider the next step towards the evolution of 3G to compete with these evolving technologies and yet provide better results. Consequently 3GPP has started the project the study item of UTRAN and evolved UTRA which intended to study the sophisticated ways to attain substantial leaps in terms of cost reduction and efficient service provision. Overall aim is to arrive at such an evolved radio access technology that prevails current fixed line technologies at considerably low cost compared to that of current line technologies [3]. Smooth introduction of the new technology is another requirement thus it is required that new evolving wireless technology must be able to synchronize with the present 3G radio access architecture and technology. So in order to achieve these requirements, 3GPP requirement is to develop new wireless transmission technologies and some enhancements must be made to the existing architecture of the wireless transmission network. A characterization of super 3G network is the use of shared channels. Fair division of the resources among users is a critical process. This process of dividing the resources is scheduling. In this research we study the downlink scheduling algorithms. In wireless communication, these scheduling algorithms allows the efficient utilization of wireless channels and provides the user with the desired QoS depending upon the service thus providing efficiency and flexibility. One of the main issues faced in wireless scheduling algorithms is that the channel quality does not remain the same over time and frequency due to the user mobility and radio interference. On the other hand channel fading is also a cause of bad channel condition. A fair scheduling algorithm only serves the user with the good channel quality. Another issue is to serve the users with the desired QoS and stable data transmission service regardless of their channel condition.

In this work, we considered these issues and proposed a super 3G enhanced S3G-E network architecture by making enhancements in super 3G network architecture To reduce the cost of investment and latency, it is essential to consider an architecture of the system having reduced network nodes, resulting in prominent reduction of protocol-associated processing, which in turn reduces the cost of testing of interoperability. The reduction in number of nodes also makes it possible to reduce call setup time. QoS must also be improved with the induced enhancements and for this purpose a two layered hybrid scheduling algorithm is proposed. An approach based on DiffServ context has been used for the differentiation between the services requested by the user which considers QoS policy profile. The proposed algorithm is tested under different scenarios of flows to investigate its response to certain performance measures of QoS.

The remaining paper is organized in such a way that Section II gives the architecture of Super 3G enhanced S3G-E network, Section III presents a S3G-E hybrid scheduling algorithm. The simulation results are shown in Section IV followed by Section V in which conclusions are drawn.

II. ARCHITECTURE EVOLUTION OF SUPER 3G ENHANCED S3G-E NETWORK

For the requirement of the reduced cost and latency, it is necessary to introduce a system architecture that has comparatively reduced number of network nodes. This would not only help in the reduction of net protocol-associated procedure but also reduces sum of interfaces, as a result of which the cost of interoperability testing will also be reduce. As the numbers of nodes are reduced, there will be fewer numbers of nodes involved in the call setup procedure; this in turn reduces the overall call setup time. This reduction also makes it possible to merge control panel protocols. A possible reduction in the number of nodes in current 3G architecture is shown in the Fig. 1.

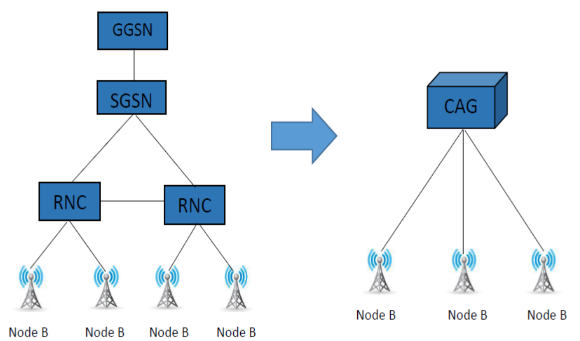

Fig. 1. Evolved architecture of Super 3G Enhanced (S3G-E) network

In the current 3G architecture, the lower layers of wireless access are handled by Node B. Radio network controller (RNC) is responsible for call control, mobility management and optimization of transport network. All of the radio protocols ends at this point. The purpose of gateway General Packet Radio Service (GPRS) support node (GGSN) is to provide a gateway to the other packet data networks, the serving GPRS support node (SGSN) functions as the foundation node in visiting network, it also performs the tasks of session management and mobility management [4].

In the proposed (S3G-E) architecture, the three nodes GGSN, SGSN and RNC are merged together to make a single node, this node is named as Core Access Gateway (CAG). CAG will jointly perform the functioning of GGSN, SGSN and RNC. Now the control plane protocol will be the same as RRC protocol. CAG will also manage the functions like ciphering, compression, automatic repeat request (ARQ) and integrity protection.

The architecture proposed for S3G-E network has the following advantages:

- As there are less number of nodes and less protocols packing/unpacking, the overall user-plane latency is reduced.
- Call setup time is decreased as lesser numbers of nodes participate in the setup process.
- Fewer interfaces lead to the reduction of complexity. The demand of testing of interoperability will also be minimized.
- Security solutions can be achieved by performing ciphering and integrity protection algorithms.
- Robustness against lower layers losses is achieved by placing an ARQ protocol.

III. HYBRID SCHEDULING ALGORITHM FOR S3G-E ALGORITHM

A) Service Differentiation

In this section a traffic engineering model based on service differentiation mechanism has been proposed for the Super 3G enhanced (S3G-E) network that guarantees a QoS to all types of traffic with a constraint of available resources. For this purpose some mechanisms have already being proposed includes Integrated Services (IntServ), Resource ReSerVation Protocol (RSVP), Differentiated Services (DiffServ), and Multi Level Switch Protocol (MPLS). Among all these mechanisms DiffServ is considered the most scalable QoS mechanism. In the context of DiffServ, a S3G-E packet flow is assumed to be divided into following three classes Signaling flows (SIG), Guaranteed Bit Rate flow (GBR) and Best Effort flows (BE). SIG flows contains the packets of signaling traffic. A fixed capacity is allocated to this flow and it should not be altered no matter what the channel condition is. A Guaranteed bit rate is provided to GBR flows. These flows are delay sensitive. All the real-time traffic lies under GBR flow like video calling, VoIP and audio calling. Whereas, the remaining capacity is allocated to the Best Effort (BE) traffic. In the DiffServ paradigm, each packet is marked before entering a network, so that `they can be associated with a forwarding policy. For this purpose we define a term traffic class indicator (TC-IND). The DiffServ mechanism can be described in three processes.

Marking: All the incoming packets are market by different values of TC-IND for different types of flows. E.g. RT traffic like the voice traffic is marked by the GBR (Guaranteed Bit Rate) TC-IND value.

Classifying: The marked packets are now classified into following three classes depending upon their TC-IND values. All packets market with SIG TC-IND value goes to SIG class. Real-time traffic like voice and video packets marked with GBR TC-IND goes to GBR class. The remaining non-real time traffic belongs to BE class

Scheduling: In the process of scheduling, logical queues are assigned to each of the traffic class and the total bandwidth is divided among the three classes with some scheduling mechanism depending upon the QoS guarantees to be provided to a particular class.

B) Upper Class Scheduler

As there is a strict priority between the three types of traffic flows, so the algorithm 1 can be represented as below:

Algorithm#1:

- *Begin*
  - Schedule the packets that belong to *SIG flow*.
  - Schedule the packets that belong to *GBR flow.*
  - Schedule the packets that belong to *BE flow*.
- *End*

Fig. 2 shows the upper class scheduler which illustrates policy profile designed for individual traffic class. For SIG and GBR, a strict priority based policy is used. In this work, we will only concentrate on the GBR and BE flow.

Two main decisions should be made by a scheduler at the same time i.e., to choose which user from which TC-IND. GBR and BE schedulers now contain two level layers the Inter TC-IND scheduler and the Intra TC-IND scheduler. The first layer sorts the different TC-IND based on their policy profiles (committed rate, peak rate) is done by Inter TC-IND scheduler, the second layer makes a decision to choose which user to be served in a given slot and this is done by Intra TC-IND Scheduler.

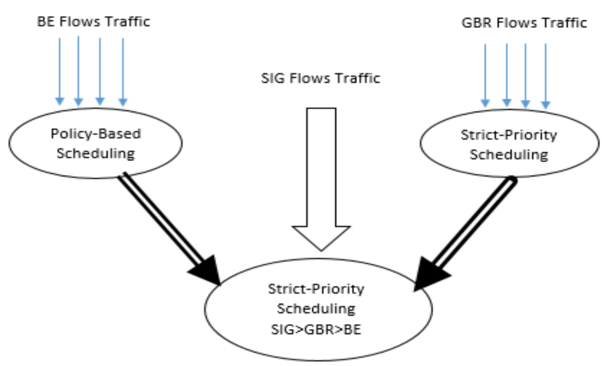

Fig. 2: An overview of the High level Scheduler

C) Inter TC-IND Scheduling Algorithm

In this section the first layer scheduler Inter schedulers are outlined. Beginning with the algorithm prepared for the Best Effort flow Inter TC-IND scheduler and then Guaranteed Bit Rate traffic Inter TC-IND scheduler is described.

i) Best Effort Flow: The two layers Scheduler for the Best Effort Inter TC-IND is shown in the Fig. 3.

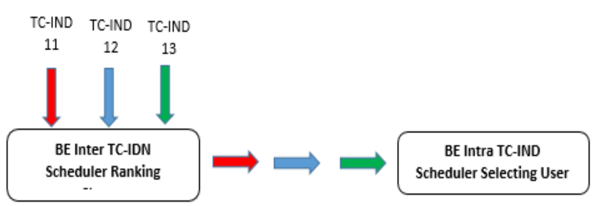

Fig. 3: An overview of the two layer scheduler of BE class traffic

Let suppose $\propto_j^i$ is the data rate perceived for the user i of j^{th} TC-IND. In that sense the throughput aggregated for the j^{th} TC-IND will be defined as $\propto^j = \sum_i \propto_j^i$.As the Best Effort (BE) policy profile is based on Peak Rate, Committed Rate and priority. So our algorithm takes its scheduling decisions based on the values of these parameters. An autoregressive process is introduced which measures the amount of information bits assigned to each TC-IND at each transmission time interval (tti). There are 1200 slots in the process. Let $\{\propto^1 \dots\dots\dots \propto^n\}$ be the throughputs aggregated for each TC-IND. We can compute now the average number of bits that are allocated to each class by using the following equation.

$$\propto^i (t+1) = (1-\lambda) \propto^i (t)\lambda \propto^i \quad (1)$$

$$i = 1 \dots\dots.n, \ \lambda = 1/1200$$

Scheduled Bits Ratio (SBR) is defined as:

$$SBR^i = \frac{\propto^i}{\sum_{j=1}^{n} \propto^j}, \qquad i = 1 \dots..n \quad (2)$$

Below is given the algorithm designed for the BE Inter-TC-IND scheduler.

Algorithm#2

- *Begin*
  - Calculate the *SBRs* for each BE classes: $\{SBR^1, \dots\dots\dots, SBR^n\}$
  - Let N_1 be the set of TC-INDs that are not reaching their Committed Rates (CR)

    $N_1 = \{i \,/\, SBR^i < CR_i\}$
  - N_2 is the set of TC-INDs that are reaching their Committed Rate but are below the Peak Rate (PR).
  - Sort N_1 and N_2 by class priority such that output = $\{N_1, N_2\}$
  - i =1,
  - While resources available,

    *Call the Intra-TC-IND Scheduler set N(i),

    *i=i+1,
  - End While,
- *End*

ii) Guaranteed Bit Rate: The algorithm for GBR scheduler is relatively insignificant because this algorithm simply makes the strict priority decisions while its the matter of choosing TC-IND to schedule. It always schedule that class first which has the highest priority after which the class having second highest priority is served and so on. The algorithm for GBR Inter TC-IND scheduler is given below:

Algorithm#3

- *Begin*
  - Suppose we have the Queues represented by $[Q]_{i \times j}$ in which Q_{ij} represents the i^{th} TC-IND queue having priority i for the j^{th} UE

- For $i = 1$ to n
  - * While the resources are available,
    - Call the Intra-TC-IND scheduling algorithm on UEs having i^{th} priority
  - *End While,
- End For,
- *End*

D) Intra TC-IND Scheduling Algorithm

Let us consider a base station (BS), all the packets that reach the base stations are classified into different classes based on TC-IND and then transmitted to different queues. They are then sent into Low-Level TC-IND Scheduler (LLS), the function of this scheduler is to choose the user to be served next and dispatch the head-of-line (HOL) of chosen UE flow to the MAC and Transmission Module (MTM). This module determines the quality of medium of each user and governs the suitable rate to communicate with the station. A regular feedback of the highest rate is updated to the scheduler for taking decisions. We here presume that the base station has the instant recognition of best rate of each user.

Scheduling Policy: In this algorithm each flow *i*, is given a weight w_i that shows the amount of bandwidth committed to it by the system. To keep a track of theoretical services received by each flow *i*, a virtual time v_i is maintained in this way we can compare the flows in terms of services received by them. A factor lag_i is used to maintain the lead/lag of the flows, this factor is helpful for organizing the compensation services. The flow *i* is said to be the lagging flow when $lag_i < 0$, called leading when $lag_i > 0$ and called satisfied when $lag_i = 0$. If the queue of the user *i* is nonempty then this flow is called *backlogged* on the other hand if the flow is empty then it is called *unbacklogged*, the flow will be called *satisfied* if it is a leading flow. Our scheduling algorithm will only select the active flows to be served. When a leading flow which is unbacklogged is chosen, its service will be used as compensation service for another flow that needs this service. When an unbacklogged flow *i* transits to a backlogged flow then its virtual time v_i is set to $max\ \{v_i\ , min_{j\in M}\ \{M\}\}$ where M is the set of all flows that are in operation.

The Fig. 4 shows the overall scheduling policy adopted in this scheduling algorithm. To begin with scheduling process, a scheduler always selects that user's flow having least virtual time v_i. If the opted flow *i* is backlogged then its HOL packet can be served provided that channel quality is also satisfactory. The lagging factor of the flow *i* is checked, if $lag_i>0$ (lagging) then this service is referred as *Normal Service (NS)*. The v_i is then updated as $\{v_i + l_p/w_i\}$ where l_p is the length of the packet. Now if the channel condition for flow *i* is bad or if its queue is empty then this flow has to give up its service, this type of service is termed as *Extra Service (ES)*. While if the flow is leading ($lag_i<0$) which means the flow *i* has been over served and a *Service Degradation Scheme* will be triggered to confirm the eligibility of service for the flow *i*. If flow *i* is not eligible then this flow has to give up its service for the compensation purpose. This scheme chooses flow *i* for scheduling, then update v_i and start its packet transmission. While if a flow *j* is chosen, then variables of v_i , lag_i , lag_j are changed as follows:

$$v_i=v_i+ l_{p'}/ w_i \tag{3}$$

$$lag_i=lag_i+l_{p'} \tag{4}$$

$$lag_j=lag_j- l_{p'} \tag{5}$$

(p' is the packet being sent)

Whenever the HOL packet of the i^{th} flow is transmitted, its queue size is also taken into account, if the queue size of the respective flow is found to be empty then this leads to the activation of *Lag Scattering Scheme*. Below is the description of three schemes of the Intra FC-IND scheduling algorithm.

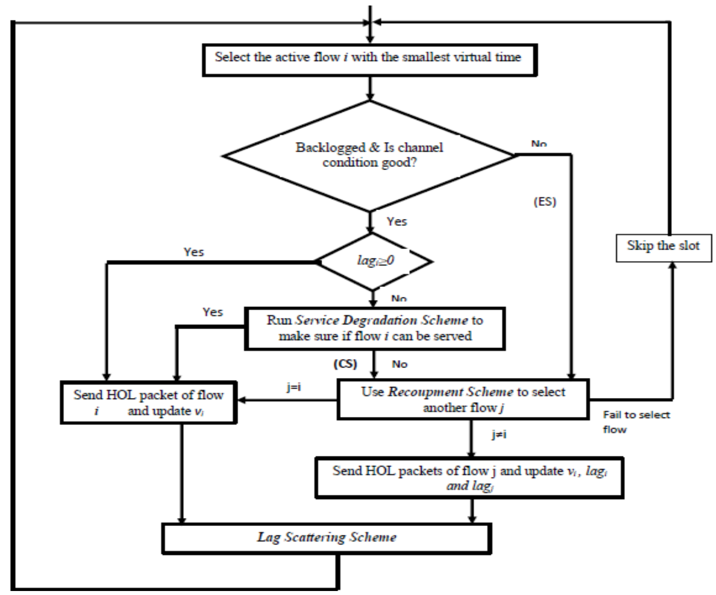

Fig. 4: Overview of Intra FC-IND Scheduler

i) Service Degradation Scheme: If a scheduler selects a leading flow to be served then Service Degradation Scheme will be triggered that checks the amount by which a flow is leading. In order to limit the amount of such services we follow the idea of CIF-Q algorithm. In this case the flow receives an extra service compared to its normal service. When a flow *i* switches from lagging/ satisfied to leading state then a variable is set up $u_i = \alpha.v_i$ where α ($0< \alpha <1$) is a constant defined by the system. The virtual time v_i of the flow *i* will rise, whenever it is selected for scheduling process. Let $v_{i'}$ be the recent virtual time of flow *i* when chosen by the scheduler. The flow *i* will be allowed to be served only when $u_i \leq \alpha v_{i'}$. We can now say that the flow will only be served for $\alpha(v_{i'} - v_i)$ this is known as service degradation.

ii) Recoupment Scheme: When the flow *i* that has been selected to be served encounters a poor channel quality or unable to satisfy the condition of service degradation, then Recoupment Scheme gets activate. In this scheme the lagging flow will be given a higher priority as compared to the non-lagging flows to gain the extra services. This scheme will deal with the two cases:

- ES / CS dispatches to lagging flow

- ES dispatches to leading flow

a) ES / CS Dispatches to lagging flow: The Recoupment Scheme gives the priority to the lagging flow and dispatches the services (ES/CS) to these flows first. The lagging flows are further divided into seriously lagging set L_s and moderately lagging set L_m. The services are dispatched according to the weights of the flows. A weight W_s is assigned to seriously lagging flow and W_m to a moderately lagging flow. A lagging flow *i* is considered to be seriously lagging flow L_s if $lag_i/w_i \geq \beta$ where β is a predefined value. Otherwise the flow *i* is assigned to L_m. Services are distributed in accordance with the weights of these flows i.e., W_s and W_m. A *recoupment virtual time* R_i^s and R_i^m is maintained for each flow *i* which will record the normalized amount of service provided to flow *i.* A scheduler will always select the error-free flow with minimum recoupment virtual time R_i^s from the set of seriously lagging flows. Similarly the scheduler will select the errorless flow *i* having smallest R_i^m from the set of moderately lagging flows. The Recoupment virtual time will be updated whenever the scheduler chooses a flow *i* to be served

$$R_i^s = R_i^s + l_p/w_i \qquad \text{if } lag_i/w_i \geq \beta \qquad (6)$$

$$R_i^m = R_i^m + l_p/w_i \qquad \text{otherwise} \qquad (7)$$

(b) ES dispatches to Non-lagging flows: If there is no lagging flow then the service will be returned back to its initial source. If the service arrives from CS then it will be sent back to the original flow that had been selected. While the ES service will be provided to the non-lagging flow. ES service will be dispatched in accordance with the weights of non-lagging flows. So a recoupment virtual time R_i^n will be maintained that record the normalized amount of ES service received by the flow *i.* A scheduler will select the flow *i* with the smallest R_i^n. When a flow is served its virtual time will be changed as follows

$$R_i^n = R_i^n + l_p/w_i \qquad (8)$$

iii)Lag Scattering Scheme for Unbacklogged Flows: If the queue is being served and its status does not change from lagging to satisfied while its queue state changes to unbacklogged then in this case the credits of this flow will be scattered to the other lagging flows that are in debt according to their weights. After this lag_i is set to zero.

IV. SIMULATION RESULTS

The simulation is performed using NS2 simulator. This simulator is widely used for networking research [5]. We have used its module DiffServ4NS developed for the scheduling algorithms. These simulations are carried out for the purpose of evaluation of the proposed scheduling algorithms for S3G-E network. This network consists of the UMTS core network and the external IP network. The QoS parameters (queuing delay, throughput and fairness) are measured, analyzed and compared with the other scheduling algorithms. We have considered three scheduling algorithms for comparison with the proposed S3G-E scheduler, Fair Throughput (FT) [6], Channel-Condition Independent Fair Queuing (CIF-Q) [7] and Exponential Rule [8].

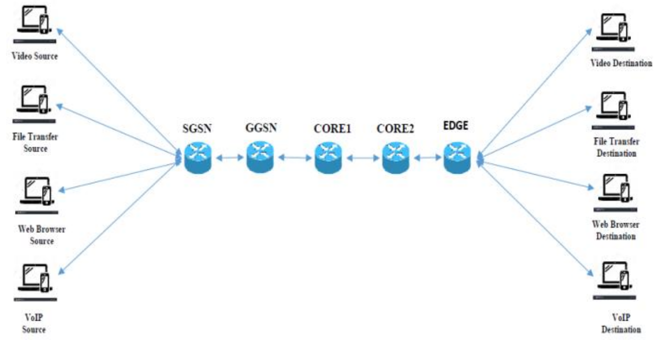

Fig. 5: Simulation overview diagram

The model of simulation is shown in the Fig. 5. This model consist UMTS network and an external IP backbone network. Four routers SGSN, GGSN, CORE1, CORE2, and EDGE, four source nodes and four destination nodes. The four source and destination nodes represent different type of traffic. Simulation parameters are defined in the Table I.

Table I: Simulation parameters

Simulation time	50s
Flow duration	45s
User mobility speed	3m/s
Area of simulation	3 cells
Size of the Cell	500m
Bandwidth	20MHz
Frame Period	6ms
Number of simulations	4

Table II: Comparison of scheduling algorithm based on different QoS parameters

QoS Parameters	Fair Throughput (FT)	Channel condition Independent Fair Queuing (CIF-Q)	Exponential Rule (ER)	Super 3G Enhanced (S3G-E)
Queuing Delay	0.1sec	0.09sec	0.15sec	0.05sec
Throughput	13 Mbps	23 Mbps	15 Mbps	25 Mbps
Fairness	0.7	0.5	0.45	0.6

Fig. 6(a) represents the queuing delay for the flow of video streaming as a function of time. The Fig. 6 clearly shows the distinction between four different scheduling algorithms. Amongst these algorithm the S3G-E scheduler suffers much less as compared to the Fair Throughput (FT), Channel-Condition Independent Fair Queuing (CIF-Q) and Exponential Rule (ER). Queuing delay increases slowly with time as number of user increases and the buffers reaches its limit. The Fig. 6(b) shows the throughput of file transfer flow In this scenario, the users enters the network, request a file of size 10MB, downloads the file and then the flow terminates. Fig. 6(c) is the simulation result showing the fairness of the algorithm for VoIP flow. In Table II a detail comparison of proposed scheduler for S3G-E network with other commonly used schedulers is discussed and analyzed. The comparison proves that the proposed scheduler is efficient in all aspects. An average is taken of all the performance measures.

V. CONCLUSION

In this work, a downlink hybrid Scheduling algorithm has been proposed for the evolved Super 3G Enhanced (S3G-E) network, which gives a high performance in all kind of traffic flows in a limited bandwidth. An approach based on DiffServ context has been used for the differentiation between the services requested by the users and a hybrid Schedulers has been designed which considers QoS policy profile. The scheduler consist of two layers of scheduling, first is Inter FC-IND scheduler is applied while in the second layer Intra FC-IND is The first layer sorts the different FC-IND based on their policy profiles (committed rate, peak rate), the second layer makes a decision to choose which user to be served in a given slot. The scheduler is tested by NS2 simulator by developing a traffic engineered framework. Different scenarios has been investigated for packet flows belonging to different traffic classes mainly the Guaranteed Bit Rate (GBR) and Best Effort (BE) class. The simulation results of the proposed scheduler and its comparison with other scheduler shows that this scheduler surpasses the other schedulers in terms of queuing delay, throughput and fairness.

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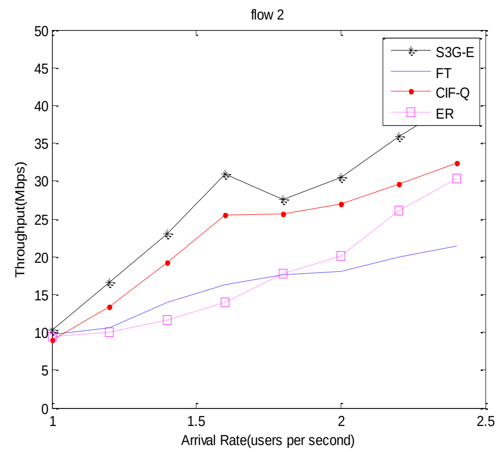
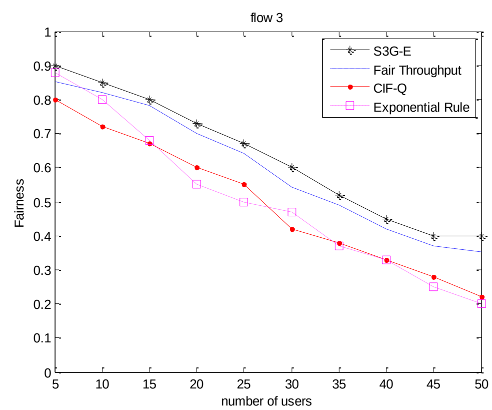


(a)



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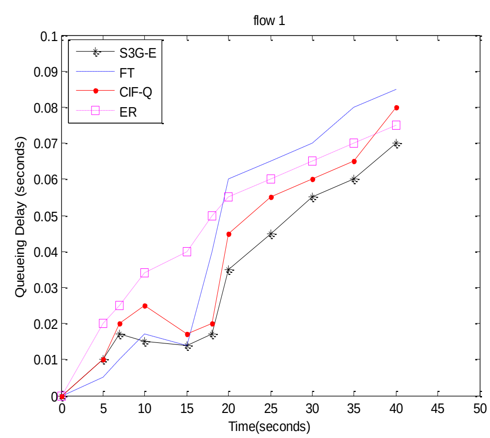

(c)

Fig. 6: Simulation result of a) queuing delay of flow 1, b) throughput of flow 2, and c) fairness of flow 3