

AN EFFICIENT PACKET SCHEDULING ALGORITHM FOR 4G IP-BASED MOBILE NETWORKS

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ABSTRACT: Next generation mobile networks are expected to provide seamless personal mobile communication and quality of service (QoS). Lossless handoff is a key issue for providing the QoS. This paper presents 4G node B Architecture, a two-layer downlink queuing model and proposes a scheduling mechanism for providing lossless handoff and QoS in mobile networks, which exploit IP as a transport technology for transferring datagrams between base stations and the high-speed downlink packet access (HSDPA) at the radio layer. In order to reduce handoff packet dropping rate at the radio layer and packet forwarding rate at the IP layer and to provide high system performance, new scheduling algorithms are performed at both IP and radio layer, which exploit handoff priority scheduling principles and take into account buffer occupancy and channel conditions. Performance results obtained by computer simulation show that, by exploiting the downlink queuing model and scheduling algorithms, the system is able to provide low handoff packet dropping rate, low packet forwarding rate, and high downlink throughput.

KEY WORDS: *Packet Scheduling, Soft-handoff, 4G Mobile Networks, Downlink Queue and Quality of Service.*

1. INTRODUCTION

In the current third generation (3G) mobile system architecture, services are provided to mobile users by using circuit switched and packet-switched transport technologies separately [1, 2]. This architecture causes many difficulties in terms of seamless interworking with other networks and QoS provisioning. Thus, more flexible all-IP network has been proposed for next generation mobile systems in recent studies [3]. The next generation, that is so-called the fourth generation (4G), of mobile communication systems, has to resolve current limitations of 3G and provide global multimedia personal mobile communications [4, 5]. 4G mobile networks are expected to provide higher data rate, QoS guaranteed, seamless, and multimedia communications.

When a user switches from one access point (base station) to another, in-queue packets in the previous base station might be dropped. In order to provide lossless handoff and QoS, the system has to provide as few lost packets as possible. The packet forwarding process has been proposed for reducing the number of lost packets caused by handoff.

Currently MLDROP is the most efficient packet scheduling algorithm for 4G networks [4]. But when most of the user are near the cell boundary then the packet drop rate and packet forwarding rate is higher for this algorithm. From this motivation, we propose a scheduling algorithm for next generation IP-based mobile access networks, which exploit IP processing at the network layer and adaptive modulation schemes for the physical layer.

Scheduling algorithms are performed at IP and radio layers with different properties aiming at providing QoS, lossless handoff, and high system performance. At the IP layer, a handoff-state-based priority scheduling algorithm is proposed which exploits service differentiation in order to provide QoS of different service types, while, at the same time, it takes handoff opportunities into account by assigning different priority levels for handoff flows. At the radio layer, a fast scheduling algorithm is proposed for the downlink shared channel which takes into account the handoff state-based priority, buffer occupancy of flows, and channel conditions.

This paper has been organized as follows: A 4G mobile network has been presented in section 2. Node Architecture and a downlink Queuing Model has been presented in section 3 and 4 respectively. Then in section 5 a new packet scheduling algorithm has been proposed. Simulation model is presented in section 6. Simulation results are explained in section 7. The overall gain of the new algorithm has been discussed in section 8. Finally, the conclusion remarks are given at the last section.

2. FOURTH GENERATION (4G) MOBILE SYSTEMS

4G mobile communication networks are widely considered to exploit Internet Protocol (IP) as a transport technology for both core and access networks shown in Fig. 1 [7, 10]. The IP based core network connects the 4G mobile RAN with other access networks such as wireless local area networks (WLAN), PSTN, and the Internet. In the 4G RAN, the current hierarchical tree topology might be replaced by a mix of ring, tree, and mesh topologies. Base stations (BS) are distinguished as core-BS and leaf-BS. Core-BSs performing control and management functionalities require high computation load, e.g., routing update. Public land mobile networks (PLMN), public switched telephone system (PSTN), and several BSs are grouped into a cluster for a better localization of handoff processing within a cluster. In this case, core-BSs can be cluster heads. The leaf-BSs are connected with each other and with core-BSs by high-speed optical or radio links. In this paper, we consider 4G mobile network architecture with a mixed network topology as shown in Fig. 1, where each BS operates as a radio access router performing IP-based functions in terms of forwarding, routing, and traffic and mobility management.

2.1 Handoff Procedure and Addressing

Enhanced mobile IP version 6 (MIPv6) is proposed for packet addressing [7], routing, and for handling mobility management functions. Each mobile host (MH) has a permanent IP address (home address), which is managed by a home agent (HA). When an MH moves in a mobile network, the MH obtains two different care-of addresses (CoA) used for routing packets: region-CoA (RCoA) and link-CoA (LCoA). RCoA is assigned by an ingress gateway, which can be located at an edge router and used to inform the home agent and other correspondent hosts (CH) of the MH about which foreign network the MH is

attaching with. LCoA is assigned by a radio router (base station), which the MH is connected. Packets destined to the MH are transferred to the ingress gateway router and then delivered to the MH's LCoA. Whenever the MH switches from a radio router to another in the same region, a new LCoA will be assigned by the new router and then updated to the ingress gateway, which might be located at an edge router (ER). Updating CoA is termed as binding, which can be carried out locally or globally depending on handoff scenarios. Handoff within a 4G RAN occurs according to two following scenarios:

- Intracluster handoff: An MH is moving between BSs belonging to the same cluster, e.g., ER1. In this case, the MH does not change its RCoA. It is only assigned a new LCoA when it takes a handoff from one BS to another. The MH needs to send only binding updates to the ER1. Its CH and home agent still observe that the MH is served within the cluster of ER1.
- Intercluster handoff: An MH takes a handoff from the cluster of ER1 to the cluster of ER2. The MH is assigned a new RCoA by the ER2 and a new LCoA by the new BS. These new addresses are then updated to the CHs and the HA of this MH.

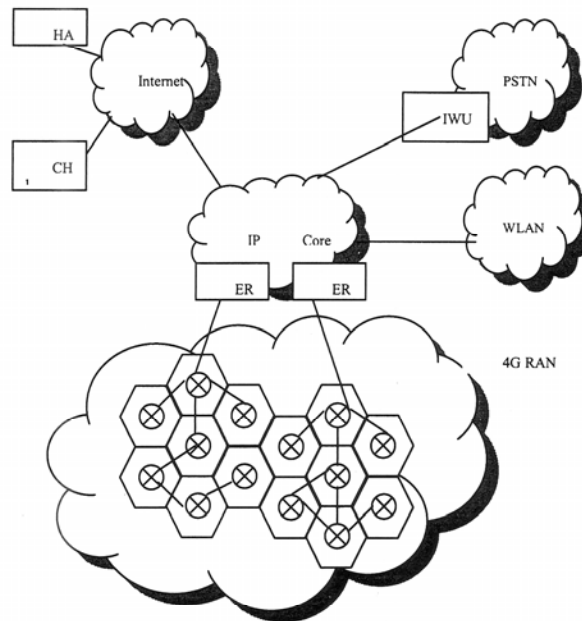


Fig. 1: Fourth generation mobile network architecture.

A handoff procedure for mobile networks based on mobile IP is shown in Fig. 2. When a MH moves to the overlap area of BS1 and BS2, a handoff-warning message is sent from the MH to BS1 if the MH notices that the current physical link quality (BS1) is decreased

while that of BS2 increases. If it finds that a handoff is needed, a handoff start message is sent to BS1 to inform the current BS that the MH is initiating the handoff procedure. A handoff request is sent to the BS2 to ask for the handoff processing. If the BS2 accepts the request, a handoff reply message is sent to the MH to indicate that the MH can now handoff to BS2. The handoff reply can contain a new CoA of the MH. The MH will access to the new channel and then release the current channel. After getting the access to BS2, a binding update message is sent to the ER (HA, CHs) to update the MH's new CoA. The MH sends a forwarding request to the current BS1 to ask for the forwarding process. After the handoff procedure is completed, some of MH's packets might still be in its queue at the BS1. If the in-queue packets are not sent to the MH, the handoff packet loss probability is increased. There are several solutions for reducing the number of handoff lost packets.

- Soft Handoff: The MH can keep the connection with both BSs for a particular interval.
- Packet Forwarding: In-queue packets are sent back from the output downlink queue to the input queue and then the current router (BS1) will forward them to the new router (BS2).
- Handoff-Based Scheduling: The current router (BS1) tries to send packets to the MH as much as possible before the MH releases its current channel.

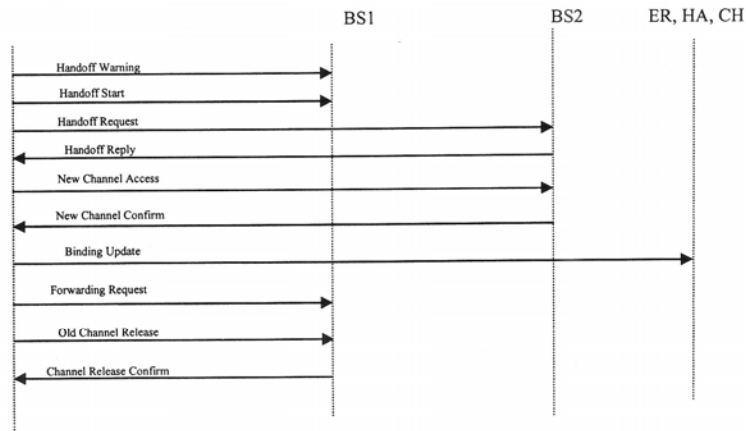


Fig. 2: Handoff Procedure.

Soft handoff is not suitable for high-speed channels because it requires more codes and causes additional interference, thus, hard handoff is used for HSDPA. The packet forwarding method can reduce the number of handoff lost packets, but it causes longer delay and consumes network resources for forwarding in-queue packets from the current

BS to the new BS. The handoff based scheduling solution reduces the number of forwarding packets. Thus, less network resource is consumed for forwarding packets. The combination of packet forwarding and handoff based scheduling seems to be the most efficient method for minimizing handoff packet dropping probability.

3. NODE ARCHITECTURE

A node (base station-BS) is modeled as a queuing system including three queuing components as shown in Fig. 3 - Input Queue, Downlink Queue, and Inter-BS Queue. The numbers of Input Queues and Inter-BS Queues are equal to the number of input and output links, respectively. For simplicity, assume that there is only one Downlink Queue for the radio downlink. Packets arriving at BS are classified and stored either in Downlink Queue if the packets are for users connecting with the node or in appropriate buffers of an Input Queue if the packets are for delivering to other nodes. The input scheduler (SInput) selects a packet in Input Queue, reads its destination address, looks up the routing table, and then forwards the packet to an Inter-BS Queue of an appropriate output link. Packets stored in downlink queue are selected for transmission to destination mobile users by the downlink scheduler (SDown). The downlink queuing model is described in detail in the next section. The handoff packet forwarding process operates as follows: Assume a mobile user A is connected with a base station BS1. Packets for user arrive and are stored in the Downlink Queue before being transmitted to the user. When the mobile user handoffs from BS1 to another base station, there might be some packets left in the Downlink queue. The Downlink Queue sends the in-queue packets back to the forwarding buffer and then the SInput forwards these packets to the new BS.

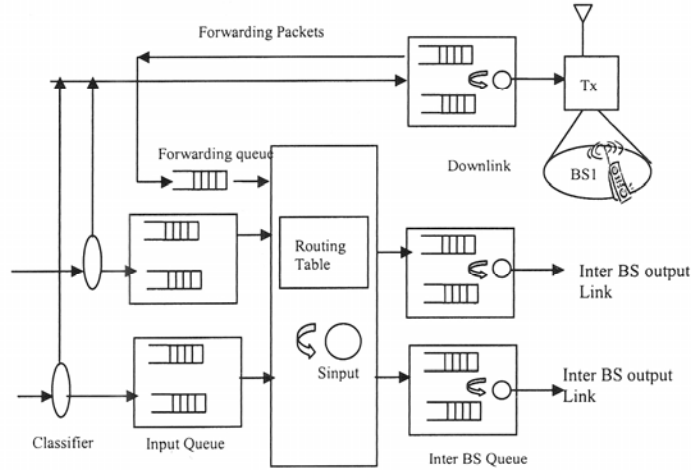


Fig. 3: Node Architecture.

4. DOWNLINK QUEUING MODEL

A two-layer queuing model is shown in Fig. 4. Where interlayer information exchange between IP and radio layers is exploited. Different from all existing queuing models, our model takes into account the impacts of handoff and the forwarding process by monitoring the flow. The IP scheduler SOutput allocates downlink resources to individual IP flows and provides their required QoS, e.g., delay at IP layer. When a MH takes a handoff to a new base station, there may be a number of packets still being in the downlink queue. In this case, the Frow processor will send the packets back to the forward buffer in input queues as described previously. The Handoff Table is used to store the handoff states of existing flows. The Fragmentation/Mapping component at the radio layer performs fragmentation of variable-length higher layer (here, IP) packets into/from smaller radio layer control (RLC) payload units (PU). One RLC protocol data unit (PDU) carries one PU. The RLC PDU sizes are set according to the smallest possible bit rate of the service using the RLC entity. The Radio Scheduler, which is modeled as a multiinput/multioutput queuing system, allocates downlink bandwidth and power to MHs. A dedicated channel, which is modeled as a single queue/single server system, is allocated to a RT connection. A shared channel, which is modeled as multi queues/single server system, is used for transmitting data of NRT connections. A Handoff Table is associated with a shared channel and used for storing handoff states of NRT users, which are sharing the shared channel. As described earlier, HSDPA is exploited for the shared channel.

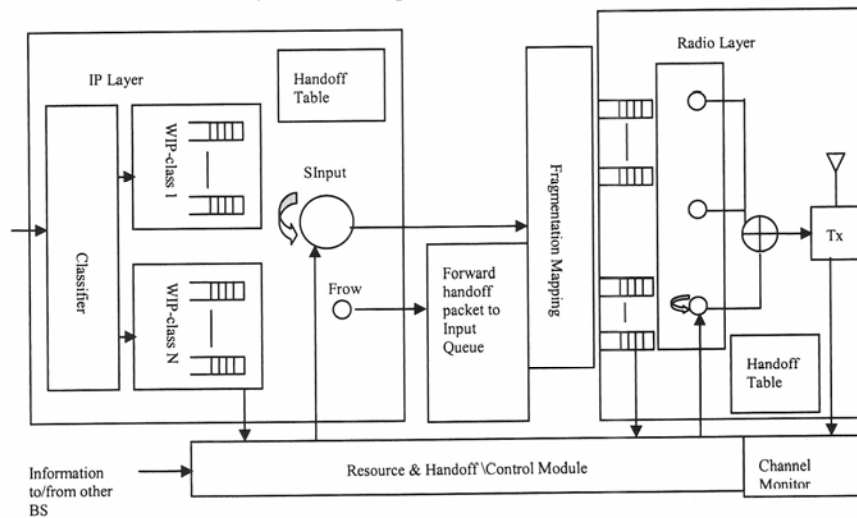


Fig. 4: Downlink queuing model.

The Resource and Handoff Control Module collects information of input traffic and queue occupations (both IP and radio layers), available radio resource, and channel quality received from the channel monitor. Output information is provided to the IP and radio

schedulers for optimizing resource allocation. The following problems of resource allocation and packet scheduling arise due to handoff.

- At the radio layer, when a user takes a handoff from a previous BS to another, there might be some or many packets of the user storing in the buffer at the radio layer of the previous BS. Forwarding radio packets is a complicated process. Therefore, the number of these packets should be as small as possible.
- At the IP layer, the IP scheduler should take into account the buffer information of the radio layer. The number of forwarding packets per handoff should be minimized.

5. PROPOSED SCHEDULING ALGORITHM

Depending on the strength of pilot signals measured at a mobile terminal, a connection has one of the following flow states: normal, handoff warning, and handoff processing. The normal state indicates that the MH receives the strongest pilot signal from the current BS; thus, it does not require any handoff. When an active MH moves to an overlapped area of the current BS and another BS, the MH measures and reorganizes a little difference of broadcasting pilot signals received from the current and neighbor base stations. A handoff warning message is sent to the current BS as shown in Fig. 5. The resource and handoff control module receives this message and changes the state of the MH's flow to handoff_warning. If the MH needs to perform the handoff procedure, it sends a handoff start message to the current BS and then the control module will change its flow state to handoff_processing.

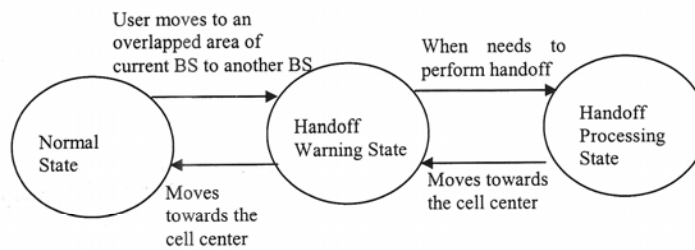


Fig. 5: Handoff States.

5.1 Parameters for the Proposed Algorithm

To propose the new scheduling algorithm several parameters were in use. The parameters are here presented in Table 1 along with the definition and the unit.

Table 1: Parameters for the Proposed Algorithm.

Symbol	Definition	Unit
i	Packet flow	-
r_i	Transmission rate requested by user	-
q_{ith}	Queue Threshold	-
q_i	Number of packets stored in the Queue	-
T_h	Time Duration to perform handoff	s
L	Propagation Loss	-
r	Distance between BS and MS	m
α	Path Loss Factor	-
ζ	Gaussian Random Variable	dB
c_o	Central Cell	-
N_p	Number of packets	-
D_p	Packet Interarrival Time	s

5.2 Scheduling Algorithm for the Downlink Shared Channel at the Radio Layer

At the radio layer, the scheduling algorithm carried out by the radio scheduler is to select packets of data NRT flows for transmission on the downlink shared channel. Our proposed fast scheduling algorithm is operating as follows:

5.2.1 Normal Mode:

For a flow i , several parameters are monitored while performing the scheduling algorithm: the transmission rate requested by the user (r_i) which depends on the user's Signal-to-Interference Ratio (SIR), the queue threshold (q_{ith}), and the amount of packets stored in the queue (q_i). At the time before a new frame starts, the algorithm is performed to select a user for packet transmission as follows:

Step1. Eliminate users whose queues are empty. Active users having packets for transmission are taken into the selection procedure. Based on SIR values updated by users, requested rates of users are calculated.

Step2. Check queue occupancies of the active users and then classify them into two groups: *over_threshold* and *non_over_threshold*. In the *over_threshold* group, the queue sizes of these users exceed their threshold.

Step 3: Select a user for transmission

- IF. There are several users belonging to the *over_threshold* group.

- THEN. Choose the user who has the maximum value of q_i .

-ELSE. Among users belonging to the *non_over_threshold* group, choose the user having maximum value of q_i/r_i .

5.2.2 Handoff Mode:

Handoff-state-based priority scheduling is applied in order to transmit packets of flows experiencing handoff before the flows are switched to the new access point. Flows having handoff processing states obtain the highest priority for transmission. Handoff warning flows have lower priority and flows having normal states have the lowest priority. If a flow has a handoff warning state and if the buffer size is larger than the threshold, the radio layer informs the IP scheduler not to select its IP packets.

5.3 Scheduling Algorithm for the IP Scheduler

At the IP layer, the scheduling algorithm performed by the IP scheduler selects packets of both RT and NRT flows and operates as follows:

-IF (more than 50% of users are in the handoff mode)

-THEN Transmit the packets of that user first having high SIR.

-ELSE Execute as in 5.2.1 & 5.2.2.

5.3.1 Normal Mode:

Service priority scheduling is applied, i.e., RT services are always served before NRT services. The packets of NRT flows are transmitted if all RT flows are empty. To provide the required delay of RT flows, the early deadline first (EDF) scheduling is applied [13]. If a flow has a full buffer at the radio layer, its IP packets are not selected.

5.3.2 Handoff Mode:

Service priority scheduling is applied as described above, i.e., RT flows are always selected priority than NRT flows. Within a service class, in order to minimize the number of forwarding packets, handoff-state-based scheduling is applied. Within RT flows, handoff_processing flows have the highest priority for transmitting, whereas handoff_warning and normal_state flows have equal priorities. For NRT flows, handoff_processing IP flows obtain the highest priority and handoff_warning flows are assigned higher priority than normal flows.

6. SIMULATION MODEL

A discrete event simulation program written in JAVA has been used for simulation. For simplicity, we have used Table 1 to select the data rates for the user at different locations. The program is developed based on the cell layout in Fig. 6. Simulation models are as described below:

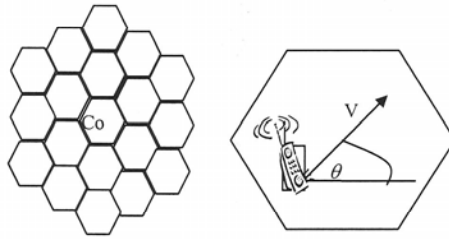


Fig. 6: Cell Layout for Simulation.

6.1 Buffer Threshold Model:

If the buffer at the radio layer of a handoff warning flow is over threshold, the IP scheduler does not select the flow's packets. The buffer threshold is an important parameter. If the buffer threshold is high, more packets are queued at the radio layer and packet drop rate increases as in Fig. 7.

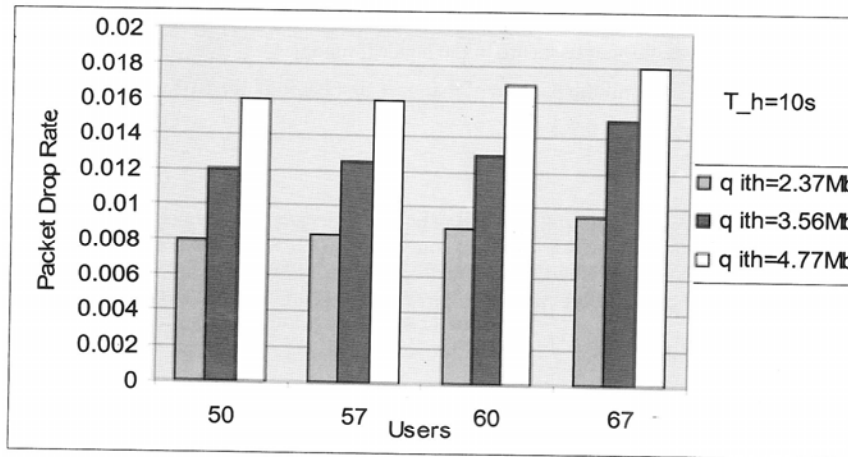


Fig. 7: Packet Drop Rate at various buffer thresholds.

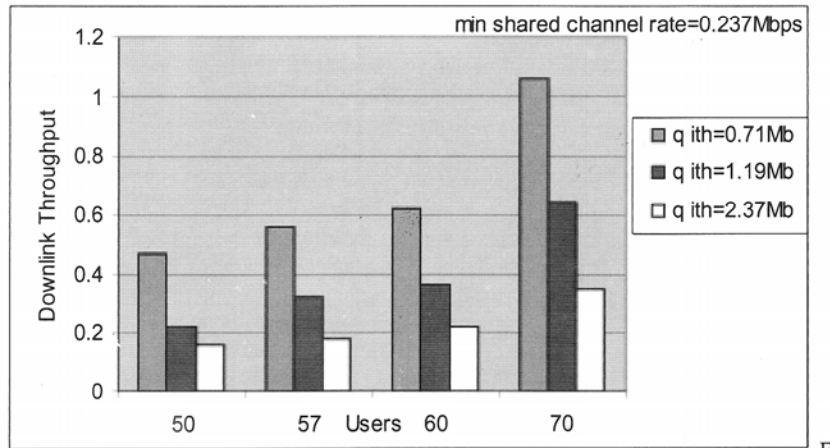


Fig. 8: Downlink Throughput at various buffer thresholds.

When the buffer threshold is small, there are more users in the over_threshold group and as the radio scheduler doesn't select these packets for transmission, the downlink throughput reduces as in Fig. 8.

The fact is, each MH needs time duration (T_h) since it moves from handoff warning state until the time it performs handoff. Users in handoff warning states are normally located in the edges of cells and their channel quality might be only good enough for transmission at the minimum data rate of the shared channel. Therefore, we estimate the buffer threshold as following:

$$q_{ith} = T_h * \text{min shared channel rate} \tag{1}$$

6.2 Propagation Model:

The relative propagation loss is as follows:

$$L = 10^{\zeta/10} r^{-\alpha} \tag{2}$$

where r is the distance (in km) between a mobile user and a base station, α is the path loss factor, and ζ in dB is a Gaussian random variable with zero mean and a standard deviation represented for shadowing effects. For performance comparison, assume that packets are transmitted without errors. The channel loss by multipath fading is also neglected. Eq. (2) analyzes that the user farthest from the center of the cell will receive lowest SIR and therefore lowest data rate [5, 6]. As the cell radius $r=500m$, we assume and divide the area into following five categories according to Table I depending upon the distance from the center of the cell for simplicity.

6.3 Mobility Model:

As shown in Fig. 6, a user moves randomly with a speed V and a direction angle θ . Every 5 seconds, a mobile user changes its speed, which is following a uniform distribution of the range from 0 km/h to 36 km/h. The direction angle changes with the angle difference $\Delta\theta$, which is uniformly distributed within $0^\circ - 360^\circ$.

Table 2: Selected MCS and their Parameters.

MCS	Data rate per code	Minimum SIR	Distance
QPSK 1/2	0.237 Mbps	-20 dB	401-500 m
QPSK 1/4	0.356 Mbps	-16 dB	301-400 m
16QAM 1/2	0.477 Mbps	-9 dB	201-300 m
16QAM 1/4	0.716 Mbps	-4 dB	101-200 m
64QAM 3/4	1.076 Mbps	6 dB	0-100 m

Simulation parameters for radio propagation and high-speed shared channels taken from [1] are shown in Table 2. HSDPA uses 10 codes with the spreading factor of 16. The minimum transmission per code is 0.237 Mbps for QPSK with coding rate 1/2. The maximum transmission per code is 1.076 Mbps in the case of using 64QAM with coding rate 3/4. The central cell C_o can allocate up to 80 percent of maximum power.

6.4 Traffic Model:

After a mobile user performs a handoff from cell C_o to another, for the comparison purpose, assume there is a new active mobile user coming to the cell C_o . It means that, at any given time, there is the fixed number of active users in the center cell. Depending on services, packets are generated as following:

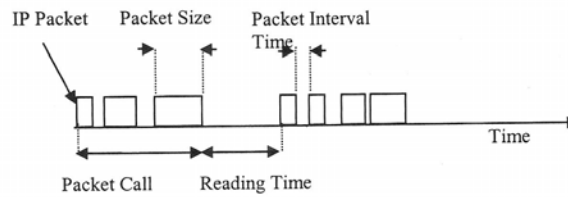


Fig. 9: Traffic Model for Non Real Time (NRT) Services

6.4.1 Real-time services:

An on-off model is exploited for real-time sources. The active (on) and silent (off) periods are negative exponentially distributed random variables. During an active period, packets are generated deterministically with a fixed packet size, this is fitted to a radio time slot.

6.4.2 Non real-time services:

A traffic model for NRT services based on the UMTS proposal for WWW traffic model is shown in Fig. 5 an NRT source generates a number of packets during a packet call's duration and then waits for an interval so-called reading time. The IP packet size is a normal Pareto distribution random variable (P) with cut-off, i.e., PacketSize = min (P, m), where m is the maximum allowed packet size. The normal Pareto distribution with cut-off has the probability density function:

$$fp(x) = \begin{cases} \frac{\alpha k^\alpha}{x^{\alpha+1}} & k \leq x \leq m \\ \beta & \beta = \left(\frac{k}{m}\right)^\alpha \end{cases} \quad (3)$$

Where α is the shape of the distribution, k is the minimum value, and m is the maximum value. The mean value is calculated as follows:

$$\mu = \frac{\alpha k - m \left(\frac{k}{m}\right)^\alpha}{\alpha - 1} \quad (4)$$

The number of packets (N_p) in a packet call is geometrically distributed with a mean μN_p . The packet interarrival time D_p is geometrically distributed with a mean μD_p . The reading time (DPC), i.e., the interarrival time of packet calls is geometrically distributed with a mean D_{pc} . Parameters of traffic generators are shown in Table 3.

Table 3: Parameters for traffic generator.

Packet Size	Pareto distribution with cut off $\alpha = 1.1$, $k = 81.5$ Bytes, $m = 666666$ Bytes.
D_p	Geometric Distribution
N_p	Geometric Distribution (mean 2.5 packets)
D_{pc}	Geometric Distribution (mean 10s)

7. SIMULATION RESULT

Impact of handoff based scheduling scheme to NRT services in terms of packet dropping rate caused by handoff at the radio layer

$$\text{Packet Drop Rate} = \frac{\text{No. of dropping packets}}{\text{arriving packets at radio layer}} \quad (5)$$

In Fig. 10 user traffic load varies with changing mean time for 50 NRT users. As the time increases, the packet dropping rate decreases due the decrease of traffic load to each user. The average packet dropping rate is equal or less than MLDROP scheme. Figure 8 shows packet dropping rate at 6 s whereas the number of NRT mobile user is varied. When the number of mobile user increases, network traffic load increases. So, the packet dropping rate also increases.

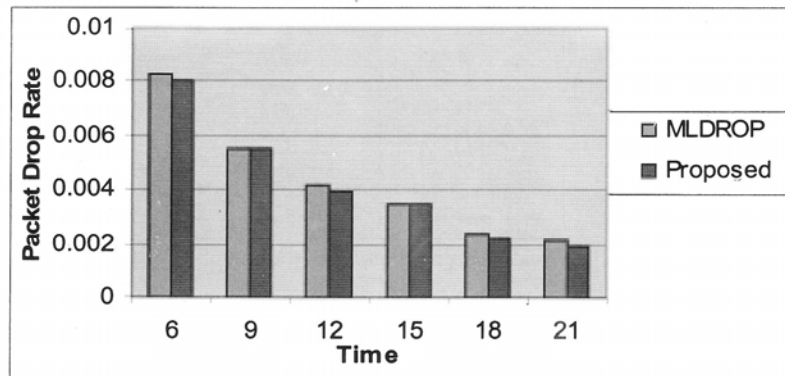


Fig. 10: Packet Drop Rate with Time.

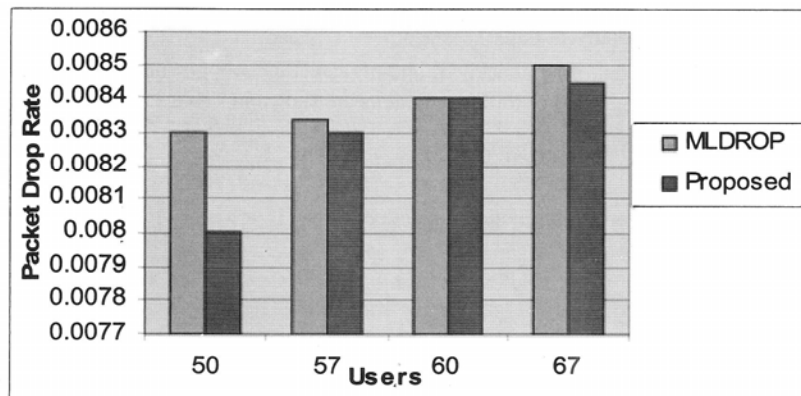


Fig. 11: Packet Drop Rate with Varying Use.

In both the scenarios above, we can see that the packet dropping rate is very low in our proposed algorithm than the MLDROP. The MLDROP algorithm is the most efficient packet scheduling algorithm. But however, the MLDROP algorithm sometimes provides

distorted results and shows high packet dropping rate when most of the user are at the cell boundary. We have added an extra condition in the algorithm to get better result than the MLDROP.

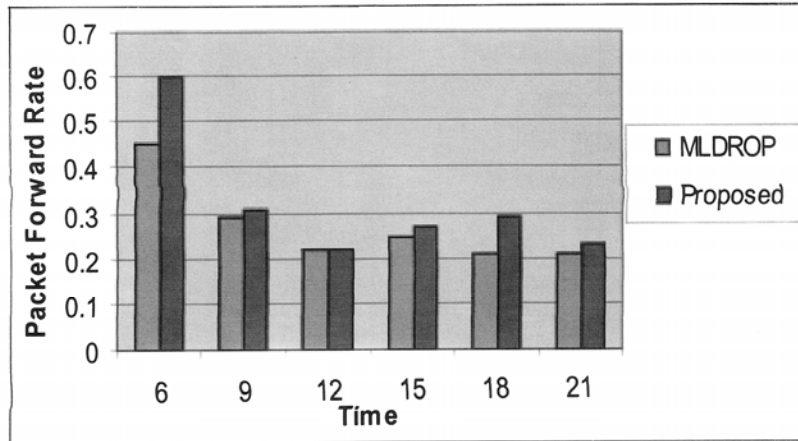


Fig. 12: Packet Forwarding Rate with Time.

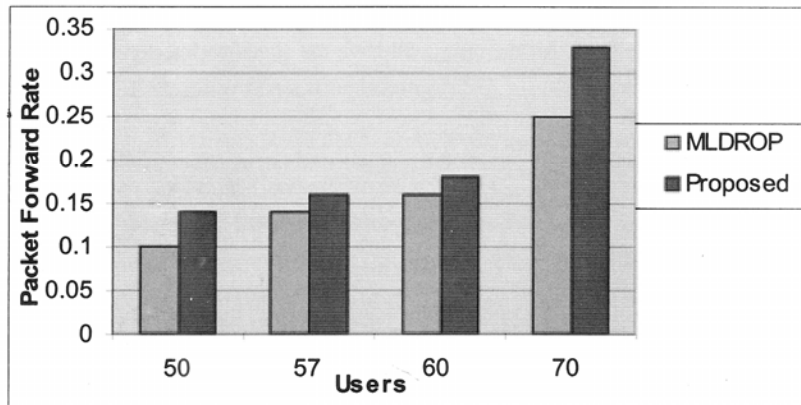


Fig. 13: Packet Forwarding Rate with varying User.

Figure 12 shows packet forwarding rate with the time for 50 NRT users. Figure 13 shows packet forwarding rate with varying users at 6s. In both the scenarios, the packet forwarding rate for our proposed algorithm is higher than that of the MLDROP algorithm. This is because that in our algorithm the user having highest SIR is transmitted first. That is the packet of those users is transmitted first who are near the center of the cell. By that time most of the user at the handoff_warning zone moves to the handoff_processing zone and the packets of the handoffed user are forwarded to the destination BS. So, the packet forwarding rate in our proposed algorithm is a bit higher than that of the MLDROP. But

nowadays the speed of the processor is immense. So, it will not consume much network resources to forward the packet to the destination BS.

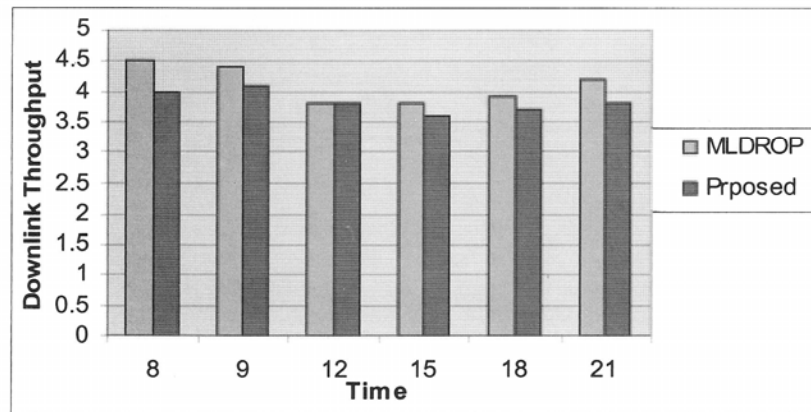


Fig. 14: Downlink throughput with time.

As packets are not forwarded from IP layer to radio layer when the radio layer buffer is full to provide low packet dropping rate, the downlink throughput of our proposed algorithm is a little bit lower than the MLDROP as shown in Fig. 14. But our proposed scheme utilizes radio resources better than MLDROP as Downlink Throughput is not the parameter to provide QoS. All the users will have the signal to Interference Ratio (SIR) data rate as in Table I.

8. DISCUSSION

In a nutshell the overall simulation result can be concluded in Table 4.

Table 4: Comparison between MLDROP & proposed algorithm

Packet drop rate	Decreased by 6.5% (approx.) than the MLDROP
Packet forward rate	Increased by 8.26% (approx.) than the MLDROP
Time complexity	$O(n)$ for both the algorithm, so both the algorithm will take almost similar time to execute
Downlink Throughput	Decreased by 4.7% than the MLDROP

From Table 3 it is clear that the packet drop rate of the proposed algorithm has been decreased by 6.5% (apprx.) while the packet forwarding rate and downlink throughput has been increased by 8.2% (apprx.) and 4.7% (apprx.) respectively. The time complexity for

both the proposed and MLDROP algorithm is $O(n)$. So, both the algorithm will take almost same time to execute.

The algorithm has successfully decreased the packet drop rate. To ensure the utilization radio resource utilization, minimizing the packet drop rate is a key concern. Although the proposed algorithm has lower packet drop rate than the MLDROP, but the packet forwarding rate is higher. The Downlink through put is lower than the MLDROP. High speed processors can minimize the higher packet drop and increase the downlink throughput, but this part needs to be analyzed further.

9. CONCLUSION

In this paper, a Node B architecture and a two layer downlink queuing model has been presented and a new efficient packet scheduling algorithm has been proposed for providing lossless handoff, QoS, and high system performance for the next generation IP-based mobile networks. System performance of the proposed scheduling algorithm has been evaluated and compared with the MLDROP scheme. MLDROP is the currently existing most efficient packet scheduling algorithm. It is shown that using proposed scheduling algorithm at radio and IP layers can provide low handoff packet dropping rate. The packet forwarding rate is also very competitive to MLDROP which means the network does not waste resources for the forwarding process. In summary using the two layer queuing model and handoff state-based scheduling priority, the system can provide lossless handoff and QoS differentiation and achieve high system performance.

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