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A New Radio Resource Management Algorithm for Mixed Traffic Transmission in Mobile System

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ABSTRACT

Mobile telecommunication new services are based on data networks specially Internet. These services include http, telnet, ftp, SMTP, etc. In addition we recognize that a mobile network is a multi-user network. Layer 3 of UMTS system provides call admission control and radio resource management. QoS can be guaranteed based on these modules. In order to have a suitable distribution of resources in a multi-user data network we need to use a queue. Queue control in the system has an important role in quality enhancement of the network. 3GPP has proposed an algorithm to control this queue. In this paper we have added a new property to this algorithm that makes blocking rate very low, however it increases the delay by a negligible amount. This property is based on the retransmission of the blocked packets in predefined times.

I. INTRODUCTION

Wireless radio channels are affected by shadowing and fading. These, in turn, are affected by environment and mobility of subscribers. On the other side, developing data networks and their connections to the mobile network have introduced new attractive services [1]. These services are in addition to ordinary voice services.

Data services include file, video and voice transfer in the mobile or over the Internet networks. Indeed Internet traffic is a combination of the above traffic services. We know that the data traffic is sensitive to packet loss and the voice traffic is sensitive to delay [2].

Call rejection (outage) may occur for two reasons:

First because the queue of call admission control module may be full and secondly the packets may reside for a too long delay (more than a threshold) in the queue before receiving the service [3]. A full queue results in blocking and too much delay in the queue results in dropping the packets.

Services provided by radio resource control module include Session Management (SM), Connection Management (CM), Mobility Management (MM). These modules add properties such as admission control and handover (HO) to the mobile system [2,4].

QoS guarantee for a WCDMA system can be provided by a combination of admission and other radio resource management services. The admission control restricts the number of users in the system such that QoS requirements for all sessions can be met. So, the admission control allows a particular user to be connected, if, after admission, QoS specifications for all users are met.

In second section of the paper we pay attention to traffic models and to the most common traffic model which is Web browsing. The third section gives the proposed algorithm which is based on the retransmission of packets. In the section four we explain simulation which is done in three cases: data and voice users are applied distinctly without and with retransmission mechanism and mixed data and voice traffic are applied with retransmission mechanism. The last section is conclusion.

II. TRAFFIC MODELS AND QUEUE

A number of voice and data models are represented as ON/OFF source models [5,6]. When a source is ON, it generates packets with a constant inter arrival time. When the source is OFF, it does not generate any packets (Fig. 1).

Regarding to voice source modeling, the process of a voice call which transits between ON and OFF states can be modeled as a two-state Markov chain. The state transition diagram shown in Fig. 1 depicts how the state transition occurs in such a way that the amount of time spent in each state is exponentially distributed and, given the present state of source traffic, the future is independent of the past (Markov chain). λ is OFF to ON rate and μ is ON to OFF rate and the average lengths of the ON and OFF periods are $1/\mu$ and $1/\lambda$ respectively.

For all real time services, calls should be generated according to a Poisson process assuming a mean call duration of 120 seconds for speech and circuit switched data services. As mentioned above the traffic model of speech should be an on-off model with activity and silent periods being generated by an exponential distribution with mean values for active and silence periods both equal to 3 seconds and independent on the up and downlink [7].

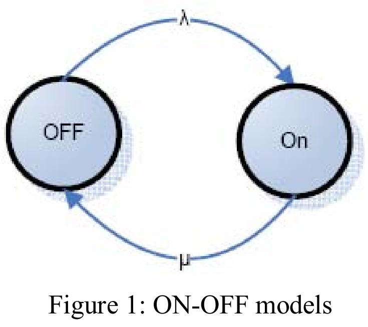

Figure 1: ON-OFF models

In addition to voice traffic there are a few different traffic models for each of the http, telnet, ftp and SMTP services of which we choose models offered in [8,9]. These models are compatible with the offered models in 3GPP [7]. Session arrival of the above services obey the poisson distribution(Table 1).

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Table 1: Traffic models of some Internet services

Traffic / Parameters	Telnet	www	ftp	E-mail
No. of packet calls/session	Geometric (mean=114)	Geometric (mean=5)	1	Geometric (mean=3)
Readin time between packet calls	Geometric (mean=1)	Geometric (mean=120)	----	Pareto (mean=90)
No. of packets in each packet call	1	Pareto (mean=25)	Pareto (mean=62)	Weibull (mean=480)
Packet size(byte)	Pareto (mean=90)	480	480	480
Packet interarrival time	Pareto (mean=1s)	Pareto (mean=0.067s)	Pareto (mean=0.067s)	Pareto (mean=0.067s)

Internet traffic of web browsing is of the most common traffic and currently contributes over than 80 percent of carried traffic over the Internet. So we use the web browsing traffic model presented in Table 1 and the Fig. 2 to simulate the proposed algorithm. As we see in Fig. 2 Internet traffic has a complex three layer model, however it is based on an On/Off model which On and Off distributions are Weibull and Pareto respectively [7].

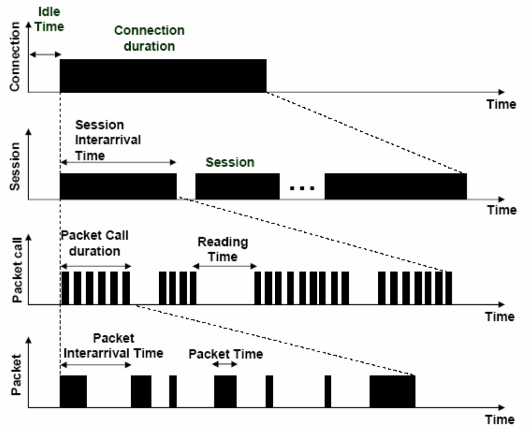

Figure 2: Multi-layer model of a web browsing packet service

In addition to that we see that a connection includes one or more sessions. A session includes one or more packet calls. A packet call may consist of one or more bursty packets.

III. SELECTED ALGORITM

The proposed algorithm designation is based on packet no–admission due to a full queue indication.

In this simulation we constitute two queues; a transmit (main) queue and a retransmit queue. First a packet is transferred to the main queue if the link is occupied and the queue has enough length otherwise it will be entered the retransmission queue.

We generate the desired voice and data traffic base on the distributions mentioned above. Then users packets are serviced (transmit on the output link) according to their arrival times to the system. A packet enters the queue if another packet is being serviced. The queues used in this program are FIFO queues. So the packets will be added to the end of queue and will be serviced according to the sequence of their arrival times. If a packet encounters a full queue it will be blocked. It means that admission control section prohibits it from coming to the system. This will cause a loss in the output link transmission. In the worst case if the blocked packet is the first input packet of a user that user will be blocked completely.

In this paper when we say a queue is full we mean that queue length has reached its maximum allowable size. In the simulation program the queue length begins from zero and can extend to the ultimate length where no user packet will be blocked.

This algorithm includes an internal and an external loop. The main algorithm which is in the internal loop is depicted in Fig. 3. The external loop is responsible for increasing the queue length and gathering the outputs. The internal loop is responsible for retransmitting the blocked packets without missing the following packets. In addition to that, this section adds the packets to the end of queue if its length is less than the threshold, otherwise will transfer them to retransmit buffer and retransmits them repeatedly by the specified mechanism.

Data traffic is more sensitive to error rate and voice traffic is more sensitive to delay. So all packets in an active duration of a voice call will be transmitted together but packet in a packet call of a web browsing session may be transmitted in different time segments. The mechanism applied in this algorithm is using a retransmission queue when the transmission queue is full. In this case the packet that encounters a full queue and has been blocked will be sent to the retransmission buffer and after a certain period of time will be transferred to the main queue if it has enough length. This procedure can be repeated as many times as we wish until the desired packet is transmitted. However this can increase the system delay too much.

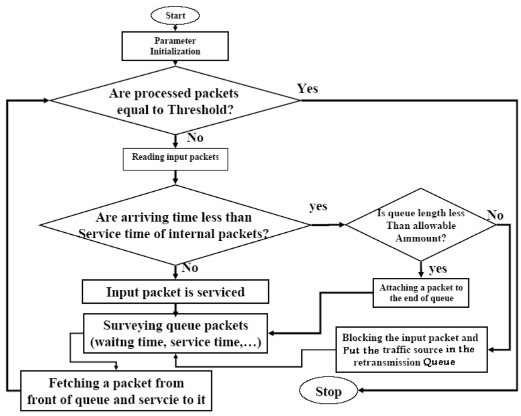

Figure 3: Flowchart of the simulated algorithm

IV. SIMULATION

We performed this simulation by MATLAB. Users' packets are serviced according to their time arrivals which is based on poisson distribution. In the simulation a matrix is responsible for finding which packet and user's turn is. We first assign a user matrix (UMAT) to a user. The first row consists of packet arrival times and the second row consist of the corresponding packet lengths. In each cycle of internal loop of algorithm we compare the arrival times of un-serviced packets of each user and transmit the packet corresponding to the smallest arrival time. For voice traffic all packets in the active time are transmitted together.

The block diagram of different parts of the simulated program is depicted in Fig. 4. As it is shown the program is constituted of different parts such as the traffic generator, timing unit, buffers (queues) unit, the admission control unit and the output link.

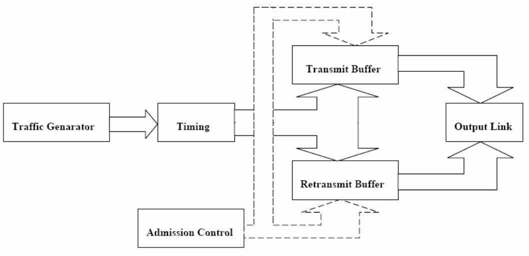

Figure 4: Different parts of simulation program

The simulation is implemented based on 60 users including 30 data user and 30 voice users. The Internet data rate is 64Kbps and the voice data rate is 12.2Kbps and the output link rate is 384Kbps. Voice traffic has exponential distribution with 3 second means in each of its On/Off times. Data traffic has been generated based on Table 1.

Simulation has been done in three distinct cases [10]:

Case1: Data users and voice users are applied distinctly without retransmission mechanism.

Case2: Data users and voice users are applied distinctly with retransmission mechanism (3 times retransmission).

Case3: Mixed data and voice traffic are applied with a special retransmission mechanism.

Simulation results are depicted in Figs. 5 to 8.

Fig. 5 shows average delay for web traffic without and with the retransmission mechanism (cases 1&2 respectively). Fig. 6 also shows percentage of blocking reduction in cases 1&2 for web packets. Fig. 8 shows the average delay of the voice packets in the cases 1&2. As it is shown the delay will be increased by a factor 10 in the case 2. Fig. 8 shows the blocking and delay parameters in the mixed traffic case. The figures show the better system performance in the case 3 compared to the two previous cases.

As it is shown the queue length is also reduced to 80 for lossless packet transmission.

V. CONCLUSION

we proposed here a radio resource control algorithm for transmission data and voice traffic simultaneously using a queue and a retransmission mechanism.

After simulation in three cases we found that packet blocking rate is reduced. But applying this algorithm to voice traffic increased the voice delay. So in the third case we applied this mechanism based on traffic recognition to only use the algorithm for data packets. Simulation showed improvement in delay and packet loss.

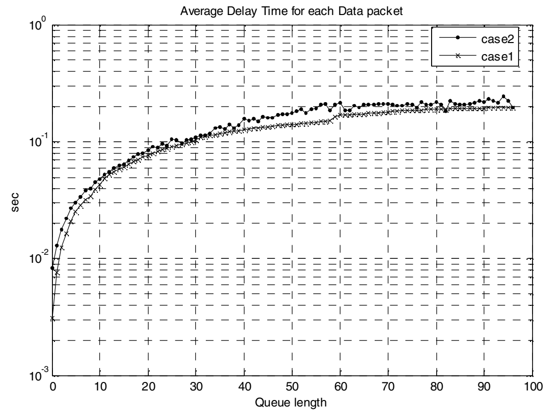

Figure 5: Web packets delays of 30 users for cases of 1&2

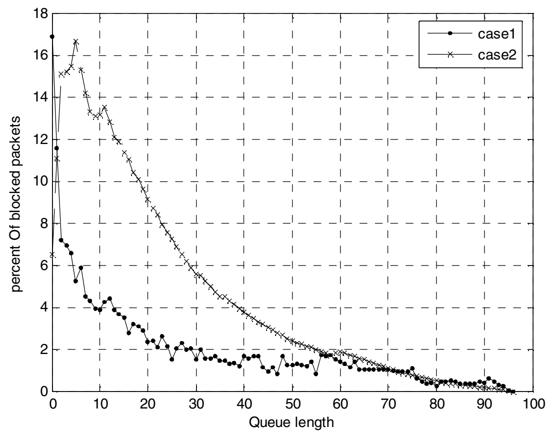

Figure 6: Web packet blocking percentage for cases 1&2.

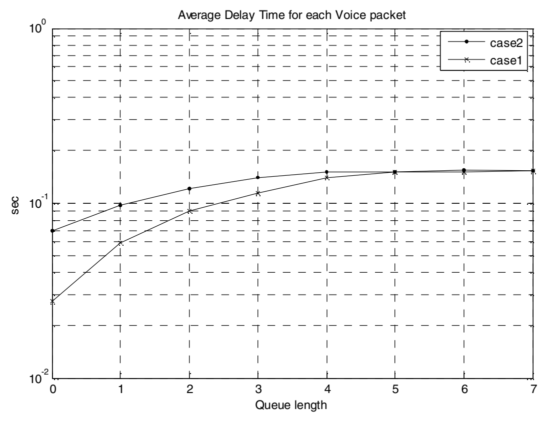

Figure 7: Voice packet delays of 30 users for cases 1&2.

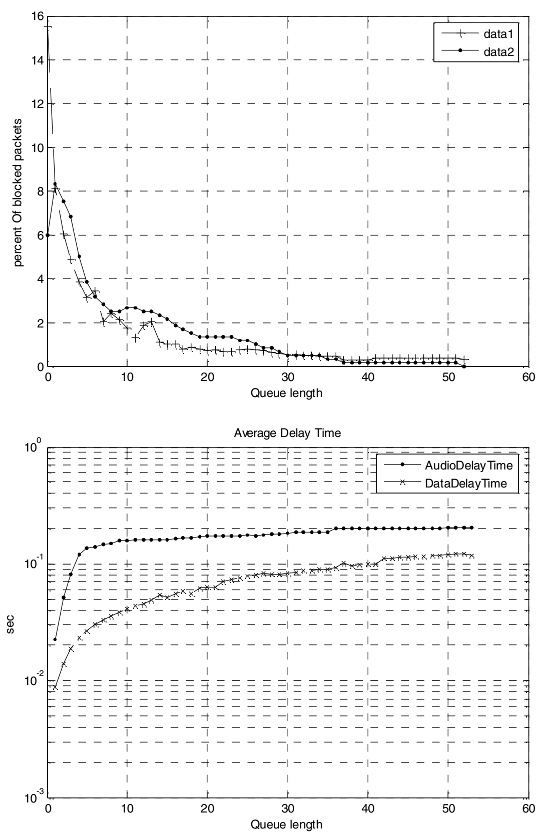

Figure 8: Mixed traffic of 30 voice and 30 data users for case 3

VI. ACKNOWLEDGMENT

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