SIMULATION ANALYSIS OF PACKET SCHEDULING ALGORITHM FOR VOICE, WWW AND VIDEO STREAMING SERVICES IN UMTS DOWNLINK FDD MODE

Marko Porjazoski, Borislav Popovski
Department of Telecommunications, Faculty of Electrical Engineering, University St.s Cyril and Methodius

Keywords: UMTS, WCDMA, Radio Resource Management (RRM), Packet Scheduling, DCH, DSCH

Abstract: UMTS provides a new and important feature allowing negotiation of the property of the radio bearer. We have focused on data transmission in WCDMA systems using packet scheduling for DCH and DSCH, in case when voice, video streaming and WWW services are engaged. Network exploiting the DCH and DSCH as a data transport channels can provide higher throughput than a network without DSCH, if a good combination of resource sharing between the DCH and DSCH is select.

1 INTRODUCTION

The third-generation system known as the Universal Mobile Telecommunication System (UMTS) introduces very variable data rates on the air interface, as well as the independence of the radio access infrastructure and the service platform. UMTS allows user application to negotiate bearer characteristics that are most appropriate for carrying information. For users this makes available a wide spectrum of services through the Wideband Code Division Multiple Access (WCDMA). The variable bit rate and variety of traffic on the air interface have presented new possibilities for radio resource utilization (Holma, Toskala, 2001)(Laiho et al., 2001).

The expression Radio Resource Utilization (RRU) covers all functionality for handling the air interface resources of a radio access network. These functions together are responsible for supplying optimum coverage, offering the maximum planned capacity, guaranteeing the required quality of service (QoS) and ensuring efficient use of physical and transport resources (3GPP TS 25.211). The Radio Resource Management (RRM) consists of Power Control (PC). Handover control (HC), congestion control (typically subdivided into admission control (AC), load control (LC) and packet data scheduling) and the resource manager (RM).

In this paper we are focused on simulation analysis of performances of packet scheduling algorithm proposed in (Sallent et al., 2001). In the simulations 7 omnidirectional

BS's (Base Stations) are assumed with variable number of users uniformly distributed around the scenario, using voice, video and WWW services.

The paper is organized as fallows: at the beginning (section 2), we describe downlink transport channels (Downlink Dedicated Channel-DCH and Downlink Shared Channel-DSCH) used for voice and data transmission in our simulations; in third section some aspects of interference and code management are described with the formulas for interference and load factor estimation; section four is description of packet scheduling algorithm to be simulated; section five presents simulation scenario, in sections six simulation results are discussed and section seven concludes the paper.

2 TRANSPORT CHANNELS

UMTS have a reach set of dedicated channels and shared channels. These channels can support multimedia applications ranging from voice to best-effort data.

Two transport channels caring a downlink transmission are (3GPP TS 25.211):

a) Dedicated Channel (DCH) – is available exclusively to a specific user. Dedicated Channel is devoted to services with stringent transfer delay requirements (voice and video). The transmission rate of a DCH can be changed every 10 ms.

Porjazoski M. and Popovski B. (2004).
SIMULATION ANALYSIS OF PACKET SCHEDULING ALGORITHM FOR VOICE, WWW AND VIDEO STREAMING SERVICES IN UMTS DOWNLINK FDD MODE.
In *Proceedings of the First International Conference on E-Business and Telecommunication Networks*, pages 67-73
DOI: 10.5220/0001396700670073

b) Downlink Shared Channel (DSCH) – is defined to support flexible multiplexing of bursty data traffic (WWW) in WCDMA. DSCH is usually used to support best-effort data services, as it cannot provide guarantees for the service quality. Its Transmission capacity is divided up among several users. The number of users multiplexed on DSCH varies with time. Since the base station may transmit to many users at one time, co-channel interference exists.

Depending on the type of service to be provided, transport channels should be managed and allocated appropriately. In this paper we are focused on conversational (voice), streaming (video) and interactive (www) services, which quality requirements may be expressed trough achieved bit rate, percentage of lost packets, delay and the delay jitter.

In order to differentiate quality levels for streaming services, we assume two layered video application that is characterized by two different flows: a basic layer, with the minimum requirements for a proper visualization, and an enhancement layer, that contains additional information to improve the quality of the received images. Data traffic from basic layer will be transmitted through DCH channel, along with conversational (voice) traffic, while the enhancement layer will be transmitted only if there is capacity in the DSCH channels. It is assumed that the possible retransmissions of the basic layer can be carried out in the DSCH channel together with the enhancement layer, and having a higher precedence than the latter.

In case of interactive (WWW) service users, data is transmitted trough DSCH.

3 INTERFERENCE AND CODE MANAGEMENT

Radio Resource Management (RRM) covers all functionalities for handling the air interface resources of a radio access network. These functions are responsible for supplying optimum coverage, offering the maximum planned capacity, guaranteeing the required QoS and ensuring efficient use of physical and transport resources (Laiho et al., 2001)(3GPP TS 23.107). Good resource allocation schemes will aim to assign many as possible links with adequate SIR to mobile users. Resource assignment is restricted by the interference caused by the BS and terminals when they start using assigned resources. RRM well not assign resources to a terminal if this assignment would cause excessive interference to other users. Decision about who should transmit and its transmission parameters (transport format and power level) are the responsibility of the packet scheduler.

For the n users transmitting simultaneously at a given cell, the following inequality for i-th user must be satisfied (Sallent et al., 2001):

$$\frac{\dfrac{P_{Ti}}{L_p(d_i)} \times SF_i}{P_N + I_{i-oth} + \rho \times \left[\dfrac{P_T - P_{Ti}}{L_p(d_i)}\right]} \geq \left(\frac{E_b}{N_o}\right)_i \cdot r \qquad (1)$$

$$P_T = \sum_{i=1}^{n} P_{Ti} \qquad (2)$$

PT – base station transmitted power; PTi – power devoted to i-th user; Ii-oth – intercell interference observed by i-th user; Lp(di) – i-th user path loss, r – channel coding rate; PN – background noise; SFi – spreading factor, relating i-th user data bit rate and chip rate; ρ - orthogonality between codes used in downlink direction.

If we rearrange equation (1), we will obtain condition that PTi have to satisfy:

$$P_{Ti} \geq L_p(d_i) \frac{P_N + I_{i-oth} + \rho \times \dfrac{P_T}{L_p(d_i)}}{\dfrac{SF_i}{\left(\dfrac{E_b}{N_o}\right)_i \cdot r} + \rho} \qquad (3)$$

Adding all n (users) inequalities, the total power transmitted by the base station can be expressed as:

$$P_{T\max} \geq P_T = \frac{P_N}{(1-\eta_{DL})} \sum_{i=1}^{n} \frac{L_p(d_i)}{\dfrac{SF_i}{\left(\dfrac{E_b}{N_o}\right)_i \cdot r} + \rho} \qquad (4)$$

where ηDL is load factor defined as (Holma, Toskala, 2001):

$$\eta_{DL} = \sum_{i=1}^{n} \frac{\left[\rho + \dfrac{I_{i-oth} \times L_p(d_i)}{P_T}\right]}{\dfrac{SF_i}{\left(\dfrac{E_b}{N_o}\right)_i} + \rho} < 1 \quad . \qquad (5)$$

Another common restriction is the number of available codewords that BSs can use. Beside interference control, packet scheduler should manage dynamical allocation of OVSF codes. The system has SFmax orthogonal codes with maximum spreading factor SFmax = 512. According to the properties of these codes, there availability is guaranteed when the Kraft's inequality is satisfied (Minn, Siu, 2000):

$$\sum_{i=1}^{n} \frac{R_{b,i}}{R_b} \le SF_{\max} \qquad (6)$$

where Rb,i is number of bits in transport block (TB) for i-th user and Rb is minimal number of bits in TB (corresponding to spreading factor SFmax = 512).

4 ALGORITHM DESCRIPTION

The scheduling strategy used in our simulations is presented in (Sallent et al., 2001). Algorithm is divided in two parts:

4.1 Prioritization

All users intended to transmit information must be classified according to a certain prioritization criteria. Firs scheduler will order requests depending on service class they belong to, from highest to lowest priority level. Conversational service class (voice) has highest priority. Second, in this prioritization scale, is streaming video service. WWW (interactive service) has lowest priority. Second prioritization rule is based on number of basic layer TB in user's buffer waiting for retransmission (This rule can be applied only for DSCH allocation, because there is no TB to be retransmitted in DCH). When two or more users have the same number of TB to be (re)transmitted, a third prioritization level based on the Service Credit (SCr) is considered.

The SCr is associated with an active link or user and it computes the difference between the bit rate requested by the user and the bit rate that system provides to him. So it calculates the amount of service that system owes to the user. The SCr value of each active connection must be updated every TTI (Transport Time Interval), following the expression:

$$SCr(k) = SCr(k-1) + \frac{R_G}{TB} - NumTx(k-1) \qquad (7)$$

where SCr(k) is the Service Credit for TTI=k, SCr(k-1) is the Service Credit in the previous TTI, RG is the guaranteed bit rate measured in bits/TTI, TB is the number of bits in Transport Block for the considered RAB (radio Access Bearer) and NumTx(k-1) is the number of successfully transmitted Transport Blocks in the previous TTI.

4.2 Resource Allocation and Availability Check

Once requests are ordered, the next step will be to decide whether or not they are accepted for transmission and which is the accepted TF. The limitations dealing with interference and code availability are taken into account in this phase. For this purpose we have to estimate the expected load factor and transmitted power level once all the requests are accepted. So, the expected load factor in system with n active transmissions in frame t can be estimated using equation (8).

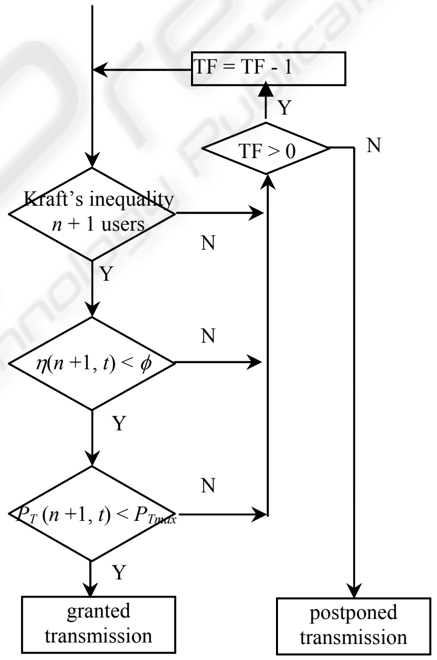

Figure 1 : Resource allocation process

$$\tilde{\eta}(n,t) = \sum_{i=1}^{n} \frac{\left[\rho + \dfrac{I_{i-oth}(t-1) \times L_p(d_i)}{P_T}\right]}{\dfrac{SF_i}{\left(\dfrac{E_b}{N_o}\right)_i} + \rho}. \qquad (8)$$

The expected power is given by:

$$\widetilde{P}_T(n,t) = \frac{P_N}{(1-\widetilde{\eta}(n,t))}\sum_{i=1}^{n}\frac{L_p(d_i)}{\dfrac{SF_i}{\left(\dfrac{E_b}{N_o}\right)_i \cdot r}+\rho} \tag{9}$$

Here we have to mention that some differences between real and estimated load factor value may occur, as a consequence of inaccuracies in the measurement of the other-to-own-cell interference and path loss.

Algorithm execution follows the flow shown on Figure 1. Assuming n already granted transmissions and initially selected TF for n+1 request, the Kraft's inequality is evaluated, the expected load factor is compared with a threshold ϕ and expected transmission power should be below the maximum transmitted power. If all tree conditions are satisfied, transmission is granted for this request during one TTI, otherwise, the transport format is reduced by one (i.e. transmission bit rate is reduced). If this is not possible, the request should wait for the next frame.

5 SIMULATION SCENARIO

The system model includes 7 omnidirectional base stations. Distance between two neighboring base stations is 1 km. Maximum transmitted power by the base station is 43 dBm. Mobile users are uniformly distributed in the scenario moving with speed of 50 km/h. At each position update (every TTI) we assume that mobile user will change his direction to left or to right in 45° with probability 0.1 for each side, and probability 0.8 that he will stay on previous course. Path loss model used in this simulation is adopted from (3GPP TS 25 942) and path loss calculations are made according to (10).

$$L = 128.1 + 37.6\ \mathrm{Log10}(R) \tag{10}$$

R is a distance from BS to ME. Path loss calculated buy (10) shall in no circumstances be less than free space path loss – FSPL = $20\log(4\pi R/\lambda)$. If during calculations L become smaller than FSPL, then FSPL should be considered instead L as path loss.

After L (FSPL) is calculated, log-normally distributed shadowing (LogF) with standard deviation of 10 dB should be added, so that the resulting path loss is the following:

$$\mathrm{Pathloss} = L + \mathrm{LogF} \tag{11}$$

Number of voice, video and www users is taken to be on of the parameters which will be changing during simulations.

Traffic generation model proposed in (Perez-Romero, 2002)(3FPP TR 101 112-UMTS 30.03) is used, including following parameters for:

- Conversational Service (Voice traffic): On-Off model with 0.3 activity factor. In the active period voice users generates 160 bits in 10 ms (16 kbps).
- Interactive Service (WWW traffic): Session arrival process – Poisson process; Number of packet call requests per session – geometrically distributed with mean 5 calls per session; Reading time between packet calls – geometrically distributed with a mean 33 [s]; Number of packets within a packet call – geometrically distributed with a mean value 25; Inter arrival time between packets – geometrically distributed with a mean 0.0625 [s] (for 64 kbps); Packet size – Pareto Distribution (with cut-off), max packet size 66666 bytes with parameters α=1.1, k=81.5.
- Video Streaming Basic and Enhancement Service: CBR model; bit rate 32 kbps (each 40 ms packet with 1280 bits is generated);

In our simulations we have adopted: 40 ms for TTI (Transport Time interval), 320 bits Transport Block Size for WWW and Video streaming service and 160 bits Transport Block Size for conversational service. DCH used for transmission of voice and video streaming basic layer will have two transport formats (TFs): TF0 – no transmission and TF1 transmission of 4 Transport Blocks (TBs) in TTI. Transport formats for DSCH, used for transport of WWW and video streaming enhancement layer, are listed in Table 1.

Table 1: Transport Formats for Downlink Shared Channel (DSCH)

TB sizes, bits		320 bits (payload) + 16 bits (MAC/RLC header)
TFS	TF0, bits	0×320
	TF1, bits	1×320 (8 Kb/s)
	TF2, bits	2×320 (16 Kb/s)
	TF3, bits	4×320 (32 Kb/s)
	TF4, bits	8×320 (64 Kb/s)
	TF5, bits	16×320 (128 Kb/s)
TTI, ms		40

Duration of this simulation corresponds to 3 min in real time. Life time of voice and video services packets was set to 1s and for www service to 10s. If the packet remains in the buffer longer then his life time it will be discarded.

6 RESULTS

Results from the simulation of packet scheduling algorithm described in this paper are shown in the figures below. We have considered average user bit rate, packet loss, delay and delay jitter as measures that will help as to estimate system behavior and performances. Figures provide a clear view of system behavior in regard of number of users in the scenario and the load factor threshold. Load factor and power estimations, as it's shown in equation (8) and (9), are based on interference measured in previous TTI.

As a result of prioritization of different service classes, it is obvious that best performances system will provide to voice users (as service class with highest priority), then to video users and at the finally to www users.

As you can see from the results, average bit rate, packet loss and delay for the voice users are nearly constant irrespectively to number of video and www users and load factor (unless a small degradation of performances for load factor values around 1).

On the other hand, system performances experienced by video streaming service users (second level in prioritization scale), appears to be more depend on number of voice users and load factor. It can be a little confusing, the packet delay decrease for load factor in the order of 1. The explanation for this behavior is that number of lost packets for video users tremendously increases, and in calculation of average packet delay we don take this packets into account. Real system behavior for the video streaming service can be noticed from average bit rate, packet loss and packet delay jitter.

And finally, www service is treated as best effort service class. When there is available resources left by voice and video users, it will be allocated to www users. As a result of smaller granularity of transport formats, www user traffic is more adaptive to severe system conditions.

7 CONCLUSIONS

In this paper, the performance of packet scheduling algorithm, used the WCDMA network area with seven cells, exploiting the DCH and DSCH was studied by simulations. It was shown that as the traffic load is increasing, the network resource utilization increases until it reach the maximum. If load is further increased, the resource utilization saturates and starts decreasing again.

For smaller load factor threshold values (0.8-0.9), packet scheduler does not provide best performances. Load factor is the limiting factor that do not allow new transmissions to be granted. For increased load threshold values (values from 0.9 to 0.95) system provides best performances: maximum average user bit rate, minimal packet loss, delay and jitter.

For higher numbers of users and load factor values around 1, difference between estimated and real value of load factor is growing, system is becoming unstable, base station transmitted power reach its maximum value faster and becomes main obstacle in provisioning better performances.

The results suggest that there is an optimal set of parameters (load factor values form 0.9 to 0.95 and number of users around 150, 75 voice, 50 video and 25 www users), by which the network resources utilization reaches the maximum.

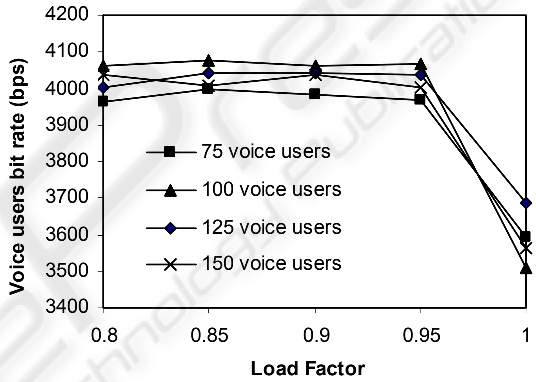

Figure 2: Average bit rate for voice service users in scenario with constant number of video streaming users (50) and constant number of www users (25)

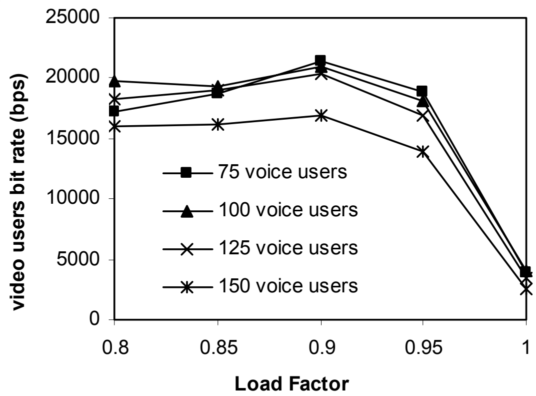

Figure 3: Average bit rate for 50 video streaming service users (basic and enhancement layer) in scenario with variable number of voice users and fixed number of www users (25)

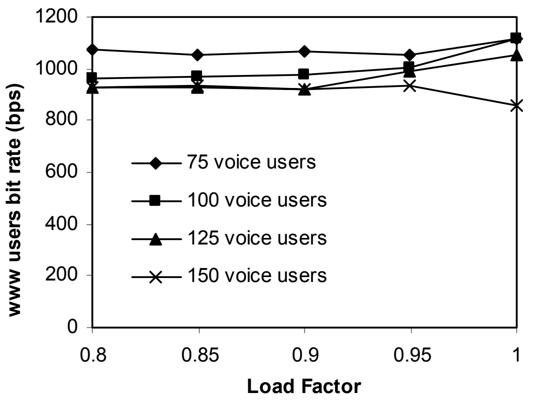

Figure 4: Average bit rate for 25 www service users in scenario with variable number of voice users and fixed number of video streaming service users (50)

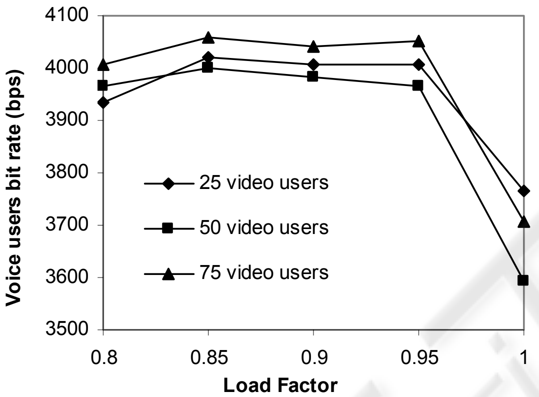

Figure 5: Average bit rate for 75 voice service users in scenario with variable number of video streaming users and fixed number of www service users (25)

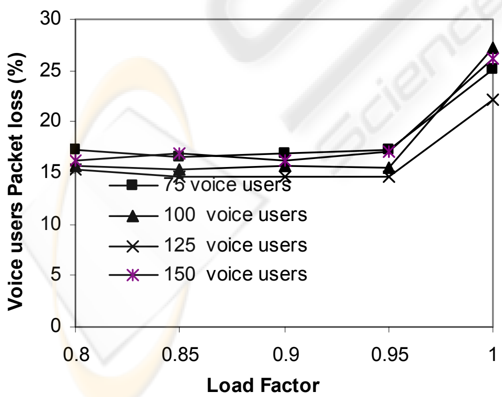

Figure 6: Packet loss for voice users in scenario with fixed number of video streaming users (50) and number of www service users (25)

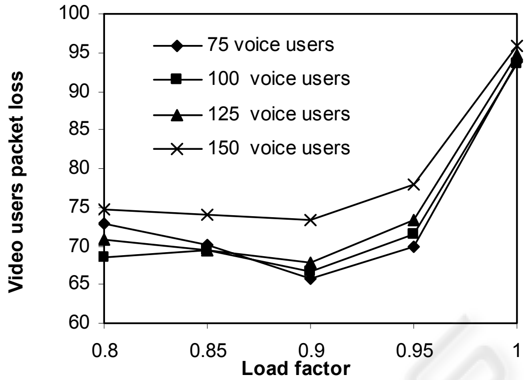

Figure 7: Packet loss for voice users in scenario with fixed number of video streaming users (50) and number of www service users (25)

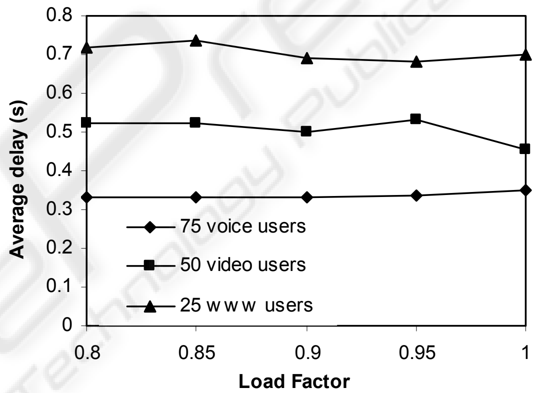

Figure 8: Average packet delay in scenario with 75 voice service users, 50 video streaming users and 25 www service users

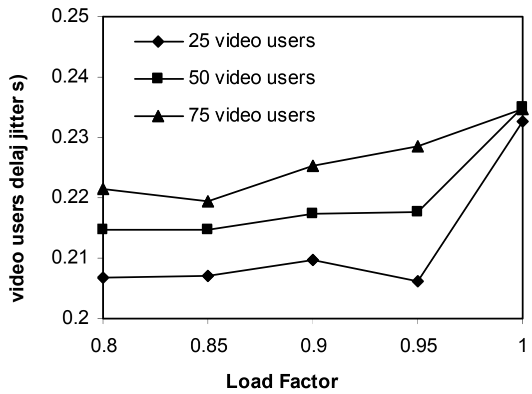

Figure 9: Packet delay jitter for video streaming users in scenario with variable number of voice service users, 50 video streaming users and 25 www service users

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