Performance Modelling of a Queue Management Scheme with Rate Control for HSDPA

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***Abstract*- High Speed Downlink Packet Access (HSDPA) is being increasingly deployed to enhance UMTS Radio Access Networks. Scheduling of Node B (base station) buffered user data for transmission over a shared radio channel is a key HSDPA functionality which enables queue management techniques to be utilized to improve QoS provisioning for mixed 'multimedia' services. Thus, we have previously proposed and studied a Time-Space Priority (TSP) buffer management scheme for 'multimedia' QoS control in HSDPA Node B. In this paper the scheme is extended to incorporate a threshold-based rate control mechanism which provides flow control between the RNC and Node B entities over the Iub interface to improve the QoS performance of non-real-time (NRT) streams in the multimedia flow. Mathematical and simulation models are developed for comparative analysis with the previously studied TSP scheme lacking rate control. The results demonstrate the performance improvement achievable with the joint implementation of TSP queue management scheme and Iub flow control mechanism(s).**

***Index Terms*- HSDPA, Multimedia QoS, Queuing analysis, Rate control, Time-Space Priority.**

I. INTRODUCTION

Third generation (3G) Universal Mobile Telecommunication System (UMTS) was introduced to support higher data rate applications unavailable in previous cellular generations. HSDPA is a technology specified by the 3G Partnership Project (3GPP) to enhance UMTS Radio Access Networks (UTRAN) capacity to support broadband services like multimedia conferencing, VoIP, or high-speed internet access. The ability to support high data rates will enable application developers to create content rich 'multimedia' applications, typically consisting of a number of classes of media or data- with different Quality of Service (QoS) requirements- being concurrently downloaded to a single user [1].

HSDPA significantly reduces downlink transmission latency, enabling peak data rates of up to 14.4 Mbps in addition to a three-fold capacity increase in UMTS networks [2], [3]. A shared downlink channel is utilized, which adapts transmission capacity to changing radio propagation conditions (fast link adaptation). Fast link adaptation employs adaptive modulation and coding (AMC) whereby different modulation and coding schemes are selected for transmission of traffic to the User Equipments (UE) within a serving HSDPA cell. AMC scheme selection is based on the experienced radio channel quality of the UE. Other features of HSDPA include HARQ for error control, and channel-dependent Fast Scheduling.

Figure 1 shows the entities in a HSDPA Radio Access Network. The base station or Node B is responsible for scheduling packets to the UEs within a cell, unlike in basic UMTS where it is handled by the Radio Network Controller (RNC). The inclusion of packet scheduling in the Node B presents opportunity to implement buffer management in the Node B buffers to improve QoS guarantees for streams of traffic comprising diverse flows or 'multimedia' traffic.

Buffer management for QoS control in HSDPA Node Bs have been studied in our previous work [4]. In particular, a combined Time-Space Priority (TSP) buffer management strategy for 'multimedia' traffic QoS control over HSDPA downlink in Node B buffers was investigated. This paper explores the extension of TSP scheme to incorporate threshold-based rate control mechanism applied between the RNC and Node B i.e. over the Iub interface that connects the two entities. The performance of loss sensitive non-real-time flow within the multimedia traffic is shown to dramatically improve as a result. Thus significant QoS performance improvement is achievable for the downlink heterogeneous multimedia flow towards a UE in a HSDPA cell, using the combined TSP buffer management scheme with Iub rate/flow control.

A mathematical model is formulated and implemented using the numerical modelling tool MOSEL-2 [5]. This is used for the initial investigation. Further study is undertaken with a more detailed simulation model of a HSDPA cell with a subset of network entities and level of functionality sufficient for the performance evaluation. The organization of the rest of the paper is as follows. The following section reviews related work, while the TSP queue management with rate control mechanism is described next. The models for performance analysis are then presented, followed by results and discussions. Lastly, concluding remarks are given in the final section.

II. RELATED WORK

Threshold-based queue management techniques have been extensively studied in the literature. In [6] an analytical queuing model for a discrete-time finite queue incorporating two thresholds used to control the sending rate of packets from the traffic source is derived. The system is proposed as a model for congestion control but considers only one traffic class. In [7], an adaptive buffer management scheme for buffers aggregating traffic from various sources with prioritization based on a set of adaptive thresholds is presented. The analysis indicates the

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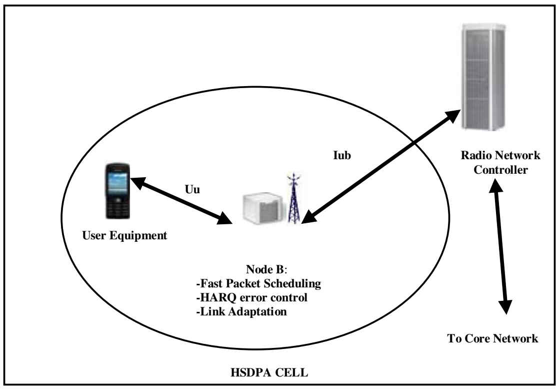

Fig. 1. HSDPA Radio Access Network with additional Node B Functionalities enhancing UMTS RAN

scheme's capability to provide QoS to higher priority traffic classes, but no flow or rate control is applied.

Active Queue Management (AQM) is a threshold-based queue management technique which is used to control the number of packets in a queue. It operates by dropping packets when necessary, to manage the length of a queue. Random Early Drop (RED) [8] is a well-known AQM algorithm recommended by Internet Engineering Task force (IETF) for Internet routers as a congestion control mechanism and to replace the traditional tail drop queuing management. A novel analytical model for a finite queuing system with AQM under two heterogeneous classes of traffic is reported in [9]. The model employs two thresholds which control the dropping rate of queued packets the queue to ease congestion. Similar to the work in [9], the study in [10] describes a simulation study of Extended RED mechanism for a finite queue of two classes of traffic, but with two sets of thresholds considered for each class of traffic. In [11], a threshold-based queue management is applied for QoS-aware rate control and uplink bandwidth allocation for polling services in IEEE 802.16 wireless networks.

The study in this paper differs from the aforementioned in the following respects. Queue management is considered in the context of heterogeneous multimedia QoS control in HSDPA downlink Node B buffers. Furthermore, priority service scheduling for real-time traffic class is proposed in a Time-Space Priority (TSP) scheme. Threshold-based rate control with two thresholds which provide feedback to control the arrival rate of non-real time traffic class to the buffer queue is incorporated to the TSP scheme. Blocking of arriving packets is employed when thresholds are exceeded rather than dropping of packets from the buffer. Finally, rate control is not applied to delay-sensitive, priority real-time traffic to prevent adverse effect on its delay and jitter performance.

III. Model Description

A. Time-Space Priority Queuing

The Time-Space Priority (TSP) queue management scheme combines time priority and space priority schemes with fixed or variable thresholds to control the QoS parameters (loss, delay, and jitter) of diverse flows within a 'multimedia' stream. Thus, real time (RT) flows, such as Video or Voice packets, are given service priority because of their stringent delay requirements; while non real time (NRT) flows, such as email or File downloads, have buffer space priority to minimize loss. This concept is illustrated in Fig. 2. When applied to a user data buffer in the Node B, arriving RT packets will be queued in front of the NRT packets to receive non-pre-emptive priority scheduling for transmission on the shared channel. NRT packets will only be transmitted when no RT packets are present in the buffer. This way, the RT QoS delay and jitter requirements would not be compromised. In order to fulfil the QoS of the loss sensitive NRT flow, the number of admitted RT packets is restricted to devote more space to the NRT flow.

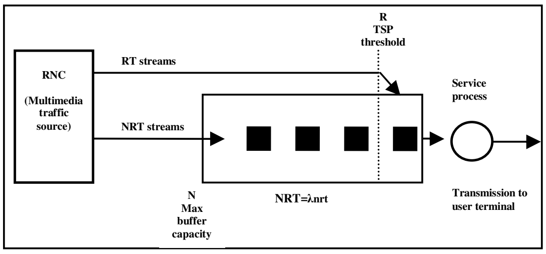

Fig. 2. TSP in Node B buffer without rate control

B. The Rate Control Mechanism

Rate control involves adapting the arrival rate of incoming traffic according to instantaneous buffer level to prevent overflow. Threshold-based rate control uses set thresholds which, when exceeded, triggers a reduction in the traffic source sending rate. Thus the system relies on an explicit feedback mechanism from the buffer to the traffic source. This technique is applied with the TSP buffer management scheme to yield an extended TSP scheme that includes rate control for non-time-sensitive traffic. The overall system is shown in fig. 3. It is proposed for QoS control of heterogeneous multimedia downlink traffic in HSDPA Node B buffers, where individual queues are maintained for each destination UE in the HSDPA cell. In addition to the TSP threshold, **R**, two additional rate control thresholds **L** and **H** are included to control the arrival rate of the non-real-time (NRT) traffic streams in the multimedia flow from the RNC. The Node B buffer state feedback is achieved through signalling between the Node B and RNC on the **Iub** interface.

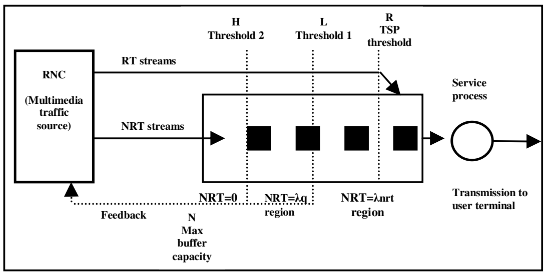

Fig. 3. TSP with threshold-based NRT rate control

Let us assume that λ_{nrt} and λ_q denote queue-state dependent NRT arrival rates. The rate control mechanism operates thus:

- If total number of packets in queue < **L** then set NRT arrival rate = λ_{nrt}
- If total number of packets in queue ≥ **L** and **< H** then set NRT arrival rate = λ_q
- If total number of packets in queue ≥ **H t**hen set NRT arrival rate = 0 (no NRT packets sent).

IV. MATHEMATICAL MODELLING OF THE SCHEME

To investigate the performance of the extended TSP scheme we develop an stochastic model from the conceptual model using the following assumptions:

- An M (2)/M (2)/1/R, N priority queuing model is assumed for the Node B buffer queue.
- All packets Inter-arrival times and service times are assumed to be exponentially distributed.
- RT mean arrival rate is denoted by λ_{rt}
- RT mean service rate is denoted by μ_{rt}
- NRT mean service rate is denoted by μ_{nrt}
- We assume **R + H = N** the total buffer capacity

Let the system state be described by the stochastic process S (t) = (R (t); N (t)), t ≥ 0; where R (t) is the number of RT packets and N (t) represents the number of NRT packets in the queue. As all random variables are assumed to be exponentially distributed, a two-dimensional continuous-time Markov chain (CTMC) with finite state space describes the underlying stochastic process. The steady-state probabilities, $P(i,j)$, of the system states are defined by:

$$P(i,j) = \lim_{t\to\infty} P(R(t)=i, N(t)=j)\,, \quad i=\overline{0,R},\ j=\overline{0,N}$$

If the steady-state probability vector of all the possible states $P(i,j)$ of the CTMC is denoted by **P,** then the steady-state probabilities can be obtained by solving :

$$\mathbf{PG=0} \quad \text{and} \quad \mathbf{Pe=1} \tag{1}$$

Where **G** is the transition rate matrix and **e** is a column vector of the appropriate dimension consisting of ones. If we denote by $P_{ij\to i'j'}$ the steady-state rate of transition from a given state S = (i, j) to another state S′ = (i', j'), the entries of the matrix **G** are given by the expressions in the first column of Table 1. The expressions are derived from the CTMC state transition diagram of the system shown in fig. 4.The system performance measures are calculated from the following set of equations.

Mean number of real-time packets is given by:

$$N_{rt} = \sum_{i=0}^{R}\sum_{j=0}^{H} i\,P(i,j) \tag{2}$$

Mean number of non real-time packets is given by:

$$N_{nrt} = \sum_{i=0}^{R}\sum_{j=0}^{H} j\,P(i,j) \tag{3}$$

Loss probability of real time packets is given by:

$$L_{rt} = \sum_{j=0}^{H} P(R,j) \tag{4}$$

Loss probability of non real time packets is given by:

$$L_{nrt} = \sum_{i=0}^{R}\sum_{j=0}^{H} P(i,j) = N \tag{5}$$

TABLE 1

ELEMENTS OF THE TRANSITION RATE MATRIX **G**

Transition Rate: $P_{ij\to i'j'}$	State transition	Condition(s)
λ_{nrt}	$i' = i,\ j' = j+1$	$i+j < \mathbf{L}$
λ_q	$i' = i,\ j' = j+1$	$\mathbf{L} \le i+j < \mathbf{H}$
λ_{rt}	$i' = i+1,\ j' = j$	$i < \mathbf{R}$
μ_{nrt}	$i' = i,\ j' = j-1,$	$j > 0$
μ_{rt}	$i' = i-1, j' = j,$	$i > 0$
$(\lambda_{rt} + \lambda_{nrt}) - (\mu_{nrt} + \mu_{rt})$	$i' = i,\ j' = j$	$i = 0,$ $j = 0$
$\lambda_{rt} - \mu_{rt}$	$i' = i,\ j' = j$	$i = 0,$ $0 < i+j < \mathbf{L}$
$(\lambda_{rt} + \lambda_q) - (\lambda_{nrt} + \mu_{rt})$	$i' = i,\ j' = j$	$i = 0,$ $i+j = \mathbf{L}$
$\lambda_{rt} - \mu_{rt}$	$i' = i,\ j' = j$	$i = 0,$ $\mathbf{L} < i+j < \mathbf{H}$
$(\lambda_{rt} + \mu_{nrt}) - (\lambda_q + \mu_{rt})$	$i' = i,\ j' = j$	$i = 0,\quad j = \mathbf{H}$
$\lambda_{nrt} - \mu_{nrt}$	$i' = i,\ j' = j$	$0 < i < \mathbf{R}$, j=0
$(\lambda_{nrt} + \mu_{rt}) - (\lambda_{rt} + \mu_{nrt})$	$i' = i,\ j' = j$	$i = \mathbf{R},\quad j = 0$
$\mu_{rt} - \lambda_{rt}$	$i' = i,\ j' = j$	$i = \mathbf{R}$, $0 < i+j < \mathbf{L}$
$(\lambda_q + \mu_{rt}) - (\lambda_{rt} + \lambda_{nrt})$	$i' = i,\ j' = j$	$i = \mathbf{R}, i+j = \mathbf{L}$
$\mu_{rt} - \lambda_{rt}$	$i' = i,\ j' = j$	$i = \mathbf{R},$ $\mathbf{L} < i+j < \mathbf{H}$
$\lambda_{rt} - (\lambda_{nrt} + \mu_{rt})$	$i' = i,\ j' = j$	$i = \mathbf{R},\ i+j = \mathbf{H}$
$\mu_{rt} - \lambda_{rt}$	$i' = i,\ j' = j$	$i = \mathbf{R},\quad \mathbf{N} > i+j >$ H
$(\mu_{rt} + \mu_{nrt}) - \lambda_{rt}$	$i' = i,\ j' = j$	$i = \mathbf{R},\ i+j = \mathbf{N}$
$-\mu_{nrt}$	$i' = i,\ j' = j$	$0 < i < \mathbf{R},$ $i+j < \mathbf{N}, j = \mathbf{H}$
λ_q	$i' = i,\ j' = j$	$0 < i < \mathbf{R},$ $i+j = \mathbf{H}$
$\lambda_q - \lambda_{nrt}$	$i' = i,\quad j' = j$	$0 < i < \mathbf{R},$ $i+j = \mathbf{L}$
0	$i' = i,\quad j' = j$	$0 < i < \mathbf{R},$ $0 < i+j < \mathbf{L}$
0	$i' = i,\quad j' = j$	$0 < i < \mathbf{R},$ $\mathbf{L} < i+j < \mathbf{H}$
0	$i' = i,\quad j' = j$	$0 < i < \mathbf{R},\ \mathbf{H} < i+j < \mathbf{N},\ j < \mathbf{H}$

From the equations (2) – (5), the mean delay or waiting time can be derived using Little's theorem thus [12]:

Mean delay for real-time packets is given by:

$$D_{rt} = \frac{N_{rt}}{\lambda_{rt} \times (1 - L_{rt})} \tag{6}$$

Similarly, mean delay for non real-time packets is given by:

$$D_{nrt} = \frac{N_{nrt}}{\lambda_{nrt} \times (1 - L_{nrt})} \tag{7}$$

The Gauss-Seidel algorithm can be applied to solve the state transition matrix equation 1. But, due to the large size of the state space and hence the recursive linear equations required for the solution, we apply the modelling tool MOSEL-2 to generate the state space, numerically solve the Markov model and evaluate the performance measures from MOSEL-2 statements equivalent to equations 2 to 7.

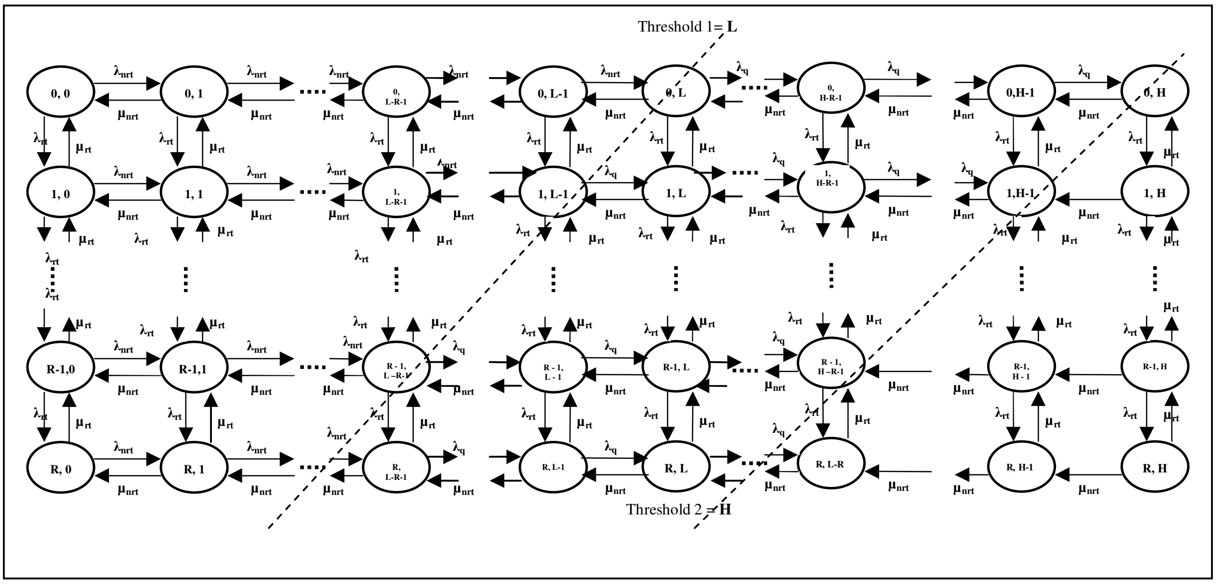

Fig. 4. State Transition diagram for the CTMC of the TSP queue with two rate control thresholds **H** and **L**

V. Simulation Modelling of the Scheme

After the mathematical analysis, a simulation model was developed in C++ to enable us evaluate the scheme with greater level of system detail. A single HSDPA cell under the control of an RNC, served by a Node B with heterogeneous downlink traffic towards an End User (EU) was modelled. The model includes MAC-hs functionality in the Node B and Adaptive Modulation and Coding (AMC) schemes which results in variable data rates towards the EU. Higher layer protocols and control plane protocols are not explicitly simulated and the EU is assumed to be stationary to simplify the model. The heterogeneous multimedia traffic is modelled as a VoIP session with a concurrent FTP file download. Table 2 summarizes the assumed simulation parameters.

TABLE 2

HSDPA Simulation Parameters

Traffic model	
VoIP	Packet length=304bits, constant bit rate=15.2 kbps
FTP	ETSI WWW model: mean packet length= 480 bytes (Geometrically distributed)
Radio parameters	
TTI interval	2ms
Mobility model	Stationary EU
Node B buffer parameters	
TSP	TSP threshold R=4, Total buffer size N=20 (in packets)
TSP with Iub flow control	TSP threshold R=4, L=2R, H=4R and Total buffer size N=20

VI. Numerical Results

In this section, the results to compare the performance of the TSP scheme with dual threshold rate control and that of TSP lacking rate control is presented. The first set of results (Fig. 5-8) are obtained from the MOSEL-2 specification of the mathematical model in section IV. The performance metrics taken are the loss probabilities and mean delay of both RT and NRT traffic in a multimedia session comprising the two classes, when the NRT packet arrival rate is varied. The following set of parameters was assumed in the experiment: $\lambda_{rt} = 10$, $\mu_{nrt} = 10$, $\mu_{rt} = 20$, $\mathbf{R} = 4$, $\mathbf{L} = 2^* \mathbf{R}$, $\mathbf{H} = 4^*\mathbf{R}$, $\mathbf{N} = 20$, $\lambda_q = 0.5\lambda_{nrt}$, while λ_{nrt}= 2, 4, 6, and 8 respectively.

The same experiment was performed for the TSP without rate control thresholds using the MOSEL-2 model developed in [13] under the assumption of the following parameters: $\lambda_{rt} = 10$, $\mu_{nrt} = 10$, $\mu_{rt} = 20$, $\mathbf{R} = 4$, $\mathbf{N} = 20$ while λ_{nrt}= 2, 4, 6, and 8 respectively. In all experiments Iub latency is assumed to be zero.

A. *MOSEL-2 Results: NRT Traffic Performance*

Figure 5 illustrates NRT loss probability for both TSP with rate control and TSP without rate control. Comparing the two, it can be seen that at low NRT arrival rates their performance is identical i.e. very low NRT packet losses occur. But with higher NRT rates the TSP with rate control gives a much better loss performance with zero packet loss. Whereas for the TSP without rate control increased NRT packet loss is observed with higher NRT arrival rates. These results indicate that significant improvement in NRT loss performance could be achieved by incorporating threshold-based rate control into a TSP buffer management scheme.

Figure 6 shows mean NRT delay for the various NRT arrival rates considered, for both schemes. Again, much better delay performance is observed with the rate-controlled scheme compared to the non-rate controlled scheme. With the non-rate controlled scheme, the possibility of encountering a full queue is much higher, especially at high NRT arrival rates. The problem of full queue is further aggravated by the priority access to service accorded to arriving RT packets. This results in longer queuing delays and in turn larger mean NRT traffic delay. Whereas, with the rate-controlled scheme, the thresholds limit the possibility of full queue and thus minimizing queuing delay.

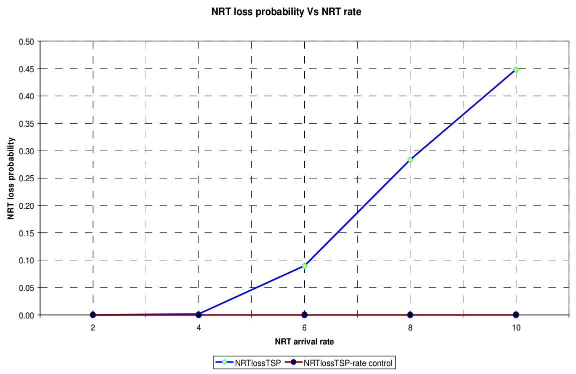

Fig. 5. Comparing NRT Loss Performance of Rate-Controlled TSP and Non-Rate-Controlled TSP Schemes

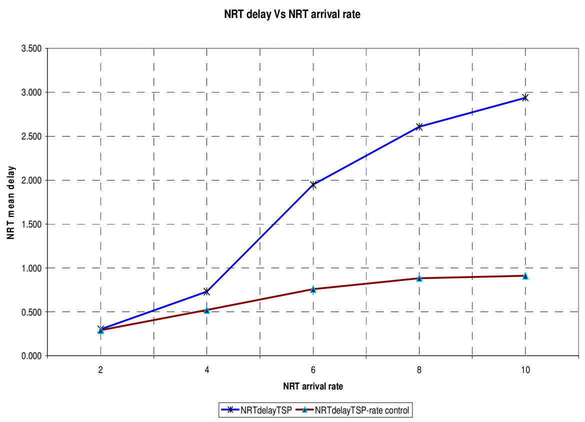

Fig. 6. Comparing NRT delay Performance of Rate-Controlled TSP and Non-Rate-Controlled TSP Schemes

B. MOSEL-2 Results: RT Traffic Performance

In figs. 7 and 8, RT performance is seen to be unaffected by the incorporation of the rate control. Instead, a slight improvement in the RT performance is noticed. The service priority mechanism for RT traffic is the reason behind the limited impact the rate control applied to the NRT streams has on the RT traffic QoS performance.

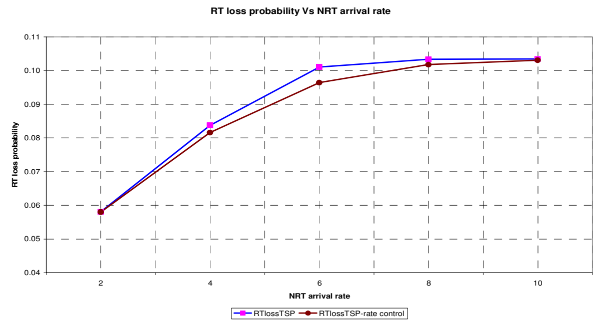

Fig. 7. Comparing RT Loss Performance of Rate-Controlled TSP and Non-Rate-Controlled TSP Schemes

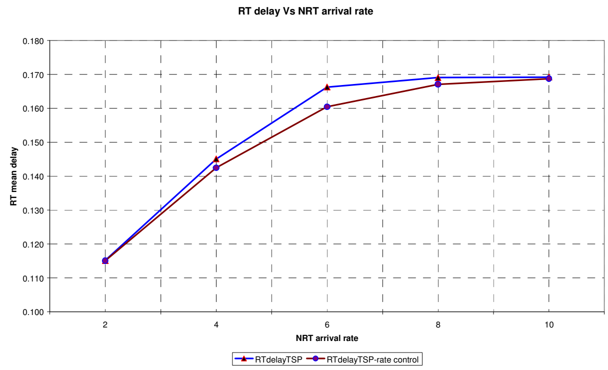

Fig. 8. Comparing RT delay Performance of Rate-Controlled TSP and Non-Rate-Controlled TSP Schemes

C. HSDPA Simulation Results: NRT Performance

The remaining results are from the simulation model of section V. Loss and delay performance metrics of the real-time (VoIP) and non real-time (FTP) flows are obtained by varying the data arrival rate of the FTP flows in the mixed multimedia traffic scenario. Performance measures are taken at FTP data rates of 8, 16, 32, 56 and 64 kbps respectively. Figs. 9 and 10 show similar behaviour to 5 and 6 therefore confirming that TSP with flow control achieves much better performance for NRT traffic even under the modelled HSDPA scenario.

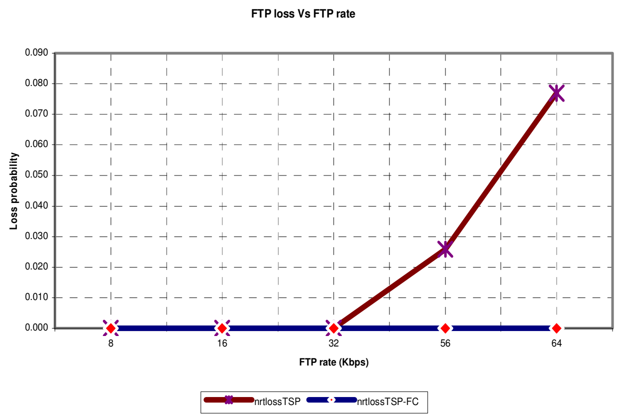

Fig. 9. FTP loss Performance of Rate-Controlled TSP and Non-Rate-Controlled TSP Schemes in a HSDPA Node B

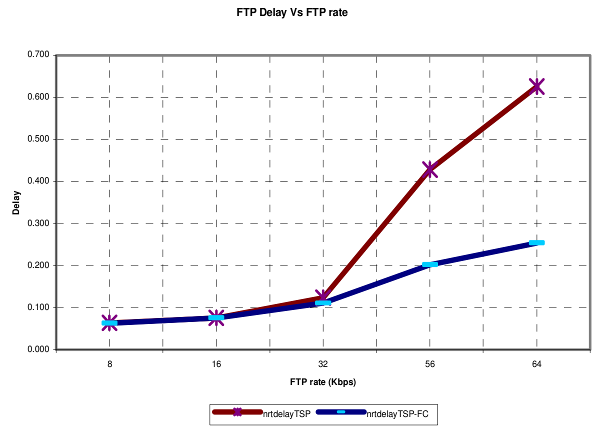

Fig. 10. FTP delay Performance of Rate-Controlled TSP and Non-Rate-Controlled TSP Schemes in a HSDPA Node B

D. HSDPA Simulation Results: VoIP performance

VoIP delay performance curves are shown in Fig. 11. Again, VoIP performance is unaffected by the incorporation of the rate control. Instead, we notice improvement in the performance as FTP rate is increased. The service priority mechanism for VoIP traffic accounts for this behaviour.

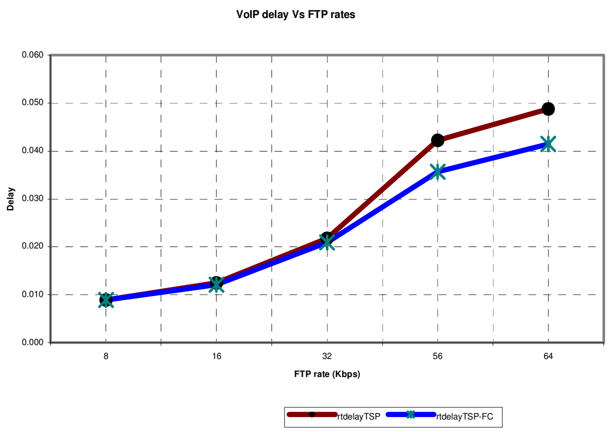

Fig. 11. VoIP delay Performance of Rate-Controlled TSP and Non-Rate-Controlled TSP Schemes in a HSDPA Node B

V. CONCLUDING REMARKS

This paper has compared the performance of a Time-Space Priority queue management scheme previously proposed for HSDPA Node B QoS-aware buffer management, with a new version which includes a dual threshold-based rate control mechanism. The rate control mechanism uses two thresholds to control the arrival rates of incoming NRT packets into the buffer queue in addition to the TSP threshold. Mathematical and simulation models were used for analysis, with the latter assuming a simplified HSDPA cell model. The results indicate that RNC-Node B Iub flow control mechanism(s) is/are a necessary requirement in buffer management schemes for mixed traffic if optimal QoS performance of the constituent streams are to be realized. Other possible flow control mechanisms under various HSDPA scenarios will be considered in future work.

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