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Dynamic Buffer Management for Multimedia Services in 3.5G Wireless Networks

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Abstract— **This paper presents and investigates a dynamic buffer management scheme for QoS control of multimedia services in a 3.5G wireless system i.e. the High Speed Downlink Packet Access (HSDPA). HSDPA was introduced to enhance UMTS for high-speed packet switched services. With HSDPA, packet scheduling and HARQ mechanisms in the base station require data buffering at the air interface thus introducing a potential bottleneck to end-to-end communication. Hence, for multimedia services with multiplexed parallel diverse flows such as video and data in the same end-user session, buffer management schemes in the base station are essential to support end-to-end QoS provision. We propose a dynamic buffer management scheme for HSDPA multimedia sessions with aggregated real-time and non real-time flows in the paper. The end-to-end performance impact of the scheme is evaluated with an example multimedia session comprising a real-time streaming flow concurrent with TCP-based non real-time flow via extensive HSDPA simulations. Results demonstrate that the scheme can guarantee the end-to-end QoS of the real-time streaming flow, whilst simultaneously protecting non real-time flow from starvation resulting in improved end-to-end throughput performance.**

***Index Terms*— HSDPA, UMTS, QoS, buffer management, real-time streaming, multimedia traffic.**

I. Introduction

HSDPA is a 3.5G wireless system standardized as a set of technological advancements to UMTS in order to improve network capacity and increases the peak data rates up to 14.4 Mbps for downlink packet traffic [1] -[4]. HSDPA utilizes a common downlink shared channel known as high speed downlink shared channel (HS-DSCH), and employs fast link adaptation for downlink data transfer to mobiles, based on adaptive modulation and coding (AMC), hybrid automatic repeat request (HARQ) and a shorter minimum allocation time (transmission time interval, TTI) of 2ms. In addition to these physical layer features, the packet scheduling functionality is moved from the centralized radio network controller (RNC) to the base station (Node B), where it is embedded in a new MAC entity known as MAC-hs.

With HSDPA, the ability to support higher data rates will allow application developers to create content rich 'multimedia' applications and services, 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 [5]. Additionally, support for multimedia services/applications with different classes of media/flows is a key requirement of the UMTS-HSDPA system [1], [4],[6]. Furthermore, packet scheduling and HARQ mechanisms in the Node B, necessitate buffering at the edge of the air interface therefore posing a potential bottleneck to end-to-end multimedia traffic. In [4], the necessity for Node B buffer management in general, to improve traffic performance has been emphasized but without proffering solutions. The study in this paper is motivated by the aforementioned; hence, a proposed buffer management scheme for QoS control of end-user HSDPA multimedia traffic with concurrent real-time and non real-time flows such as streaming video and data (in a single user session), is presented and evaluated.

The scheme, termed the *dynamic time-space priority* (D-TSP) buffer management incorporates time priorities and space priories as well as dynamic transmission priority switching between the aggregated flows to suit changing QoS requirements. This is unlike most existing buffer management schemes which are either *time* or *space* priority based. In D-TSP, *time priority* allows the delay-sensitive and loss-tolerant real-time (RT) flow to fulfill delay and jitter requirements, whilst *space priority* given to the loss-sensitive but delay tolerant non real-time (NRT) flow enables loss minimization. The dynamic (time) priority switching allows any residual delay tolerance of the real-time flow to be exploited in order to prevent potential (bandwidth) starvation of the parallel non real-time stream. This is particularly crucial at the bottleneck air interface. Hence, D-TSP concept is important not only for downlink multimedia traffic QoS control in HSDPA, but also for other wireless systems with queuing at the air interface.

D-TSP is evaluated using HSDPA system simulations, and the impact of the scheme on RT streaming flow and concurrent NRT TCP-based data flow in an end-user multimedia session is studied under various HSDPA channel loads. The rest of the paper is organized as follows. The following section discuses buffer management in HSDPA MAC-hs layer. Next, D-TSP buffer management is described, followed by the performance evaluation and numerical results. Finally, concluding remarks are given in section V.

II. HSDPA RAN Buffer Management

In HSDPA, the packet scheduling functionality is performed in the Node B with a transmission time interval (TTI) of 2ms. The basic function of the Node B packet scheduler is to determine which user will receive transmission in the next

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TTI. The size of the data block transmitted in the TTI frame is determined by the AMC scheme selected, which in turn is based on the channel quality reported via the feedback uplink control channel. In addition to the user equipment (UE) channel quality information, ACK/NACK feedback is also carried on the return uplink channel to enable retransmissions during the HARQ operation.

In the 3GPP HSDPA standards [7], packet scheduling is specified as a MAC-hs functionality as shown in Figure 1. Additionally, *priority handling* and *priority queue distribution* functionalities are defined to cater for multiple queues associated with a single user that maintains several flows in the same HSDPA session. Most existing HSDPA packet scheduling algorithms are designed for inter user transmission scheduling and do not address inter-class/inter-flow prioritization for end-users with multiple flows; i.e. the packet scheduling algorithms assume a single flow per user. Where multiple flows or media with different QoS requirements, such as RT voice/video and NRT data exists for a given user, an efficient buffer management scheme can enable inter-class/inter-flow prioritization for each 'multimedia' user, whilst the packet scheduling algorithm provides the per TTI scheduling between the users.

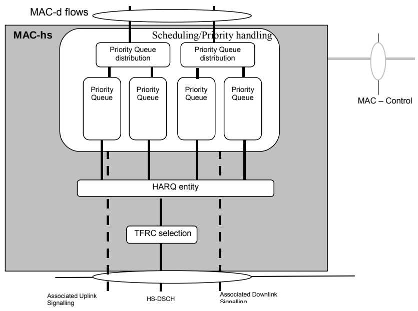

Figure 1 : HSDPA Node B MAC-hs architecture [7].

Thus, D-TSP is designed to fulfill not only the priority handling and queue distribution in the Node B MAC-hs for each user with aggregated RT and NRT flows, but also to enable QoS management for improved end-to-end traffic performance.

A. The D-TSP Buffer Management Scheme

The basic concept of dynamic time-space priority (D-TSP) is to simultaneously provide transmission (time) priority for the RT flow, and space priority to the NRT data flow of the same end-user. D-TSP utilizes a trade-off mechanism to switch transmission priority to the NRT flow at the expense of slight degradation of RT flow QoS (delay and loss) within the allowable RT QoS constraints. The idea behind D-TSP is to prevent potential NRT flow starvation at the bottleneck i.e. radio interface, without violating the RT flow QoS requirements. The scheme is illustrated in Figure 3. D-TSP is based on a novel Time-Space Priority queuing concept [8], [9], where RT flow and NRT flow destined for the same user are queued using a hybrid priority queuing mechanism. The RT flow packets, say from a conversational class voice, or real-time streaming video/audio are queued ahead of the NRT flow packets of the same user, for priority scheduling/transmission on the shared channel (i.e. time priority). At the same time, the NRT flow packets, say from background class like (TCP-based) FTP traffic, get space priority in the user's buffer queue because of their loss sensitivity, and lower transmission priority due to their delay tolerance. TSP queuing uses a threshold R (see Fig. 2), to restrict the maximum number of queued RT packets, whilst the NRT flow has unrestricted access to the entire buffer space (i.e. space priority). Threshold R allows loss tolerance of RT flow to be exploited to minimize NRT packet loss.

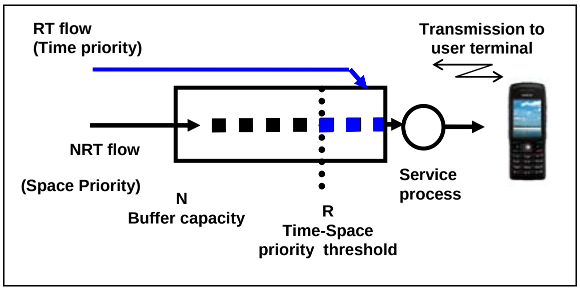

Figure 2: Basic TSP queuing concept for single user multimedia RT and NRT traffic.

Since RT flows typically do not employ retransmission protocols, RT packet losses within QoS bounds does not affect higher layer protocol performance. On the other hand, NRT packet losses being typically recovered with higher layer RLC and TCP retransmission protocols adversely affects their performance, resulting in end-to-end throughput degradation.

In [9], we have shown TSP to be an effective queuing mechanism for joint RT and NRT QoS control compared to conventional priority queuing schemes. However, according to 3GPP standards [10],[11], the MAC-hs can incorporate flow control algorithms (Iub flow control) to regulate RNC to Node B data transfer to prevent MAC-hs buffer overflow. Hence, in addition to TSP queuing, D-TSP incorporates a credit-based flow control mechanism, necessitating the additional thresholds, L and H besides the TSP threshold R. Thus the overall D-TSP (logical) queue incorporates three thresholds as shown in Figure 3.

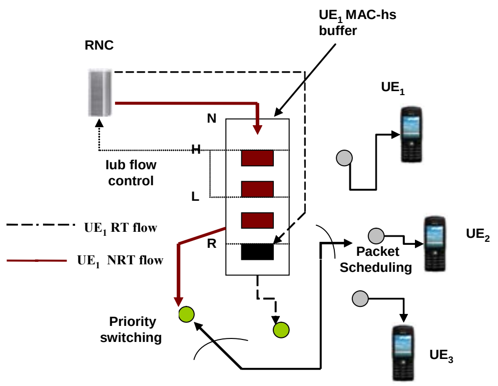

Figure 3: D-TSP buffer management scheme shown for UE_1 RT and NRT flow queuing, priority handling and QoS control in HSDPA Node B MAC-hs.

The D-TSP flow control mechanism issues credits which gives the number of data units of each flow to be transmitted from the RNC. Furthermore, the D-TSP flow control mechanism is designed to react to variation of the UE channel conditions, as well as buffer occupancy in order to mitigate buffer overflow and ensure efficient radio resource utilization. In addition, the D-TSP scheme incorporates time or transmission priority switching between the RT and NRT flows. Basically, for a given transmission opportunity assigned to the UE by the packet scheduler, when the head-of the-line (HOL) RT packet delay is below a given delay budget, transmission priority is switched to the NRT flow, otherwise it remains with the RT flow. Referring to Figure 3, The D-TSP algorithm is described with the following assumptions and notations:

- Assuming a total buffer allocation of N Protocol data units (PDUs) for a given UE with a multimedia connection/session in the Node B MAC-hs in the HSDPA cell. Let R denote the total number of allowed RT PDUs in the UE's MAC-hs buffer.
- Let r(t) be the number of the UE's RT PDUs in the buffer at time t, while we denote the number of the UE's NRT PDUs at time t as n(t). From TSP principle, $0 < r(t) < R$ and $0 < n(t) < N$ at any given time t.
- Denote the lower D-TSP Iub flow control threshold as L, where $R < L$. Likewise the higher flow control threshold is given by H, where $L < H < N$.
- Let the nth user's buffer occupancy at time t be given by $q(t) = r(t) + n(t)$. The average buffer occupancy is estimated using a moving average filter with ith sample given by:

$$q_i = w \cdot q_{(i-1)} + (1-w) \cdot q(t) \tag{1}$$

- Denote λ_{rt} as the Guaranteed Bit Rate (GBR) of the RT flow (obtainable from bearer negotiation parameters).
- Let λ'_{nrt} express the estimated average NRT flow data rate at the radio interface determined from:

$$\lambda'_{nrt\,(i)} = \alpha \cdot \lambda'_{nrt\,(i-1)} + (1-\alpha) \cdot \lambda_{nrt}(t) \tag{2}$$

where i is the ith TTI in which the user's NRT flow was transmitted during the UE's scheduling opportunity and $\lambda_{nrt}(t)$ is the amount of NRT data transmitted during the ith TTI. $\lambda_{nrt}(t) = 0$ if no NRT PDUs were transmitted in the ith TTI for the given UE.

- Let k denote a parameter for buffer overflow control, while T denotes the inter-frame period for RNC-Node B frame transfer (10ms), and PDU_size, the MAC-d PDU size in bits.
- Assuming a given delay budget, DB for the UE RT PDU MAC-hs queuing. RT PDU inter-arrival time, i at the Node B MAC-hs can be estimated from the already known GBR using:

$$i = \text{PDU\_size (bits)} / \lambda_{rt} \text{ (bits/sec)} \tag{3}$$

- Thus, we can define a time priority switching control parameter δ given by: $\delta = DB/\,i$ (4)
- Assuming a discard timer (DT) is used to discard MAC-d PDUs of the UE's RT flow when MAC-hs queuing delay exceeding a given maximum delay budget DB_{max}. If Y is the maximum allowable downlink delay then:

$$DB_{max} = Y - (\text{external network delays} + \text{Core Network delays} + \text{Iub transfer delay}) \tag{5}$$

With the above given assumptions and defined notations, the overall D-TSP scheme in MAC-hs operates as follows:

Part 1: Credit allocation for multimedia UE:

- Step 1: Compute per frame RT flow credit allocation

$$C_{rt} = (\lambda_{rt} / \text{PDU\_size}) \cdot T \tag{6}$$

- Step 2: Compute per frame maximum NRT credits

$$C_{NRTmax} = \begin{cases} (\lambda'_{nrt} / \text{PDU\_size}) \cdot T & \text{if } q_i < L \\ k \cdot (\lambda'_{nrt} / \text{PDU\_size}) \cdot T & \text{if } L \le q_i \le H \\ 0, & \text{if } q_i > H \end{cases} \tag{7}$$

- Step 3: Compute per frame NRT credit allocation

$C_{NRT} = \min\{C_{NRTmax}, UBS_{NRT}\}$ where UBS_{NRT} is the number of NRT PDUs present in the RNC for the UE. Hence, total per frame credit for the UE is $C_{rt} + C_{NRT}$.

Part 2: TSP queue management for multimedia UE:

- Step 1: For each arriving HS-DSCH data frame from RNC for the UE determine the flow class - RT or NRT.
- Step 2: If flow belongs to RT class, for each MAC-d PDU in the payload:

  If (r(t) < R) *queue PDU at RT queue tail*
  Else *drop MAC-d PDU and update RT loss*

  Else If flow belongs to the NRT class, for each MAC-d PDU in the payload:

  If r(t) + n(t) < N *queue PDU at buffer queue tail*
  Else *drop MAC-d PDU and update NRT loss*

Part 3: Transmission priority control (D-TSP only):

- For each scheduled UE transmission opportunity:

IF ($r(t) < \delta$ AND RT HOL delay < DB_{max} AND $n(t) > 0$)
  Time Priority = NRT flow
  Generate MAC-hs Transport Block from NRT PDUs
ELSE
  Time Priority = RT flow
  Generate MAC-hs Transport Block from RT PDUs

III. D-TSP PERFORMANCE EVALUATION

In order to evaluate D-TSP for streaming RT traffic and TCP-based NRT traffic in a concurrent HSDPA user's multimedia session, we used the static equivalent, (s-TSP), and complete buffer sharing (CBS) as baseline schemes for comparison. The static TSP (s-TSP) scheme consists of the TSP queuing and flow control thresholds and mechanisms described in the previous section for the D-TSP scheme but without the dynamic transmission priority switching aspect (i.e. part 3). This means that static TSP always prioritizes the UE's RT packets (PDUs) for transmission.

With complete buffer sharing, NRT flow is guaranteed some bandwidth allocation at the radio interface in the presence of the RT streaming flow of the same user, because CBS inherently possesses some degree of buffer and transmission bandwidth allocation fairness [12]. For this reason, CBS provides a comparative baseline scheme to evaluate the NRT flow starvation mitigation capabilities of D-TSP.

Recall that D-TSP uses dynamic time priority switching in order to prioritize NRT transmission while RT flow delay is within a given delay budget. RT streaming being a 'greedy source' traffic has the potential to cause NRT bandwidth starvation. So, the study is aimed at investigating the impact of RT streaming flow on the concurrent NRT TCP flow of the same HSDPA user, and the effectiveness of D-TSP in mitigating NRT flow starvation while still ensuring that RT streaming end-to-end QoS requirements are not violated.

For the study, a HSDPA system model was developed with detailed UTRAN mechanisms including, RNC MAC queues, RLC layer AM and UM modes with ARQ retransmission for AM mode. In the Node-B, MAC-hs queues (applying CBS, s-TSP and D-TSP), HARQ processes, AMC schemes, and Packet Scheduling on the HSDPA air interface are modeled. Effect of the core network (CN) is abstracted as an assumed fixed delay to arriving packets. In the receiver (UE), we included SINR calculation and CQI reporting, HARQ processes, RLC modes with ARQ for AM retransmission , packet reassembly queues, peer TCP entity, and playout buffer for the streaming RT flow.

In the experiments, a test user equipment (UE_1) is connected to the UTRAN and configured to receive 'multimedia' traffic of simultaneous 64 kbps Constant Bit Rate (CBR) encoded real-time video stream and non-real-time TCP streams in a simulated 120s streaming and file download session. The overall set up models a single HSDPA cell with fair time (round robin) scheduling to *m* users. A summary of the simulation parameters are given in Table 1.

A. Buffer dimensioning

Assuming a downlink maximum transfer delay of 250ms, the maximum MAC-hs delay budget DB_{max} can be calculated from equation (5), given that assumed Core Network + external delay + Iub delay sum up to 90ms (see Table 1). It is reasonable to assume that RNC queuing contributes very little delay comparatively because the D-TSP flow control algorithm design ensures that RT PDUs are not held back in the RNC queues. Moreover, this was confirmed during the simulations. Thus, from equation (5) DB_{max} = 160ms, and, therefore RT discard timer DT is set to 160ms.

For the 64 kbps CBR RT stream, λ_{rt} = 64 kbps hence:

$R = (\lambda_{rt} * DB_{max})$ /PDU_size = 32 PDUs

Likewise, assuming a maximum bit rate of 256 kbps for NRT flow and maximum average MAC-hs delay budget of 200ms:

Buffer size = (256 000 * 0.2)/PDU_size = 160.

Hence, we take a total buffer size N = 32 +160 = 192 PDUs in the MAC-hs for the UE. For the flow control thresholds we take H= 0.75* N = 144, and L =0.5 * H = 72.

B. Performance metrics

In the experiments, we consider MAC-hs queuing delay budgets, DB, of 40, 80, 120 and 160 ms which from equation (4), correspond to δ = 8, 16, 24 and 32 respectively. Note that the discard timer, DT discards RT streaming packets whose MAC-hs queuing delay $\geq$ 160ms from the head of the D-TSP/s-TSP queue. The performance metrics observed are:

- *Average end-to-end NRT throughput*: The time average of the throughput of the TCP-based NRT flow measured in the UE TCP layer.
- *RT PDU discard ratio*: The ratio of late RT streaming PDUs discarded in the MAC-hs as a result of DT timeout.
- *RT inter-packet playout delay*: The playout delay between successive packets of the RT streaming flow queued in the UE playout buffer after the first initial buffering delay of DB_{max}.

Several scenarios with different HSDPA cell loads were considered i.e. m = 1, 5, 10, 20 and 30 users simultaneously active during the test UE's concurrent streaming and file download session of 120s duration.

Table 1: Simulation parameters for D-TSP investigation.

HSDPA Simulation Parameters	
HS-DSCH TTI	2ms
Path loss Model	148 + 40 log (R) dB
Transmit powers	Total Node-B power=15W, HS-DSCH power= 50%
Shadow fading	Log-normal: σ = 8 dB
AMC schemes	QPSK ¼, QPSK ½, QPSK ¾, 16QAM ¼, 16 QAM ½
HS-DSCH codes	5
CQI letency	3 TTIs (6ms)
HARQ processes	4
HARQ feedback latency	5ms
Packet Scheduling	Fair time
MAC PDU size	320 bits
Iub transmision delay	20ms
External + CN delays	70ms
TCP (Reno)	MSS =536 bytes, RWIND = 64
Flow control parameters	α = 0.7; *w*= 0.7; *k*= 0.5
D-TSP /s-TSP parameters	R = 32, L= 72, H = 144, N =192
CBS parameter	N = 192

IV. NUMERICAL RESULTS

A. End-to-End NRT throughput evaluation

Figure 4 plots the average end-to-end NRT flow throughput of an HSDPA end user (UE1) terminal running a session of simultaneous CBR encoded 64 Kbps RT video streaming and TCP-based file download. The average throughput is plotted against the number of users sharing the HSDPA channel in a single cell with fair time packet scheduling. The time average of the obtained throughput measured in the UE over a session period of 120s for δ = 8, 16, 24 and 32 delay budget settings are compared to that of s-TSP and CBS buffering. In all the scenarios UE1 is assumed to be located at 0.2 km from the base station and moving away at 3 km/h, while other users are placed at random positions in the cell. From Figure 4, we can see that in the single user scenario i.e. when UE1 occupies the channel alone, the D-TSP scheme in all DB settings give only slightly better throughput than s-TSP or CBS. This represents a lightly loaded HSDPA channel scenario where the user is being allocated all available channel codes in every TTI, the resulting high bandwidth allocation prevents NRT flow starvation despite the presence of the 'greedy source' RT streaming flow. For the same reason, increasing the D-TSP parameter does not yield any throughput improvement.

In the 5 user scenario (UE_1 sharing with 4 other UEs), it is interesting to note that at this point starvation of NRT flow starts to occur with s-TSP while CBS gives about 128 Kbps average throughput. Recall that s-TSP also incorporates an Iub flow control algorithm. In the experiments it was determined that the flow control algorithm effectively prevented buffer overflow, so no NRT PDUs were lost with s-TSP, indicating that starvation was the cause of end-to-end TCP

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throughput degradation. Since according to equation (7), the NRT flow credit allocation by the flow control algorithm depends on buffer occupancy, UE radio conditions and NRT flow throughput at the radio interface, the cause of NRT bandwidth starvation can only be attributed to the static prioritization of the greedy source RT streaming flow by s-TSP, which has no transmission switching mechanism. In the same 5 user scenario, it can be seen that D-TSP was effective in allowing NRT PDUs through by delaying the RT flow PDUs for up to the given delay budget. As D-TSP parameter increases (i.e. delay budget is relaxed more), a corresponding improvement in NRT throughput is noticeable. Also all the D-TSP configurations outperform CBS, indicating better fairness in bandwidth allocation to the NRT flow (a property which the is inherent in the latter).

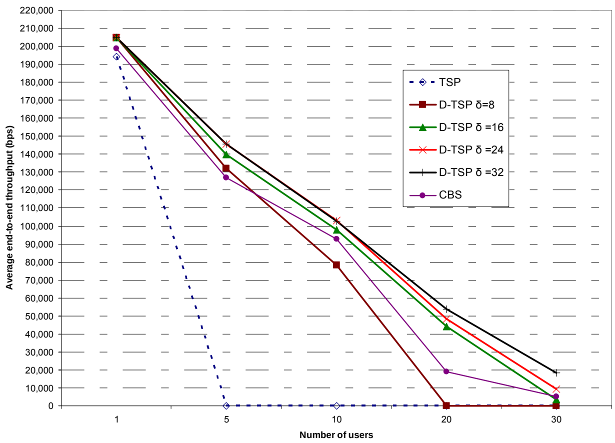

Figure 4: End-to-end NRT throughput of UE1 for s-TSP, D-TSP (δ = 8, 16, 24, and 32 respectively) with 1, 5, 10 , 20 and 30 active users in the HSDPA Cell.

In the 10 user scenario, i.e. with heavier channel load, a similar trend is observed. D-TSP with DB of 80, 120 and 160 ms (i.e. δ = 16, 24 and 32) performed better than CBS. Again, starvation by RT streaming is apparent with s-TSP. In the 20 user scenario, again with heavier load and consequent less frequent scheduling opportunities, NRT flow starvation occurs with s-TSP and D-TSP δ = 8 (40ms delay budget). Only by increasing δ to 16 and above does D-TSP become effective in preventing NRT flow starvation and also exceeding CBS in average end-to-end throughput. With 30 users, we also encounter NRT flow starvation which is prevented again by D-TSP of delay budget 80ms and above. From the results, we conclude that D-TSP provides an effective mechanism through transmission priority switching to prevent imminent NRT flow starvation by a concurrent RT stream in a HSDPA user's session comprising both flows. Next we consider the impact of this mechanism on the streaming RT flow to see whether the trade-off for end-to-end NRT flow improvement was worthwhile.

B. RT Streaming performance evaluation

Since a discard timer DT is used to discard HOL packets with delay exceeding DB_{max} (160ms), the RT streaming PDUs violating this bound will not be received at the UE_1. The D-TSP mechanism deliberately stalls RT PDUs to allow NRT PDUs transmission, thus increasing the probability of RT PDUs exceeding DB_{max} and being discarded. Therefore in order to determine whether D-TSP provides the NRT end-to-end improvement without violating the RT streaming flow QoS bound, we recorded the number of RT PDUs discarded in the MAC-hs as a result of DT timeout. Figure 5 plots the RT PDU discard ratio vs. number of users in the cell. The plots are shown for D-TSP for δ = 24 and 32 corresponding to 120ms and 160ms delay budget. For s-TSP, D-TSP δ = 8 and 16, there were no RT PDUs discarded by the discard timer. Likewise for D-TSP δ = 24 single user, 5 user and 10 user scenarios; and also for D-TSP δ = 32 single user and 5 user scenarios.

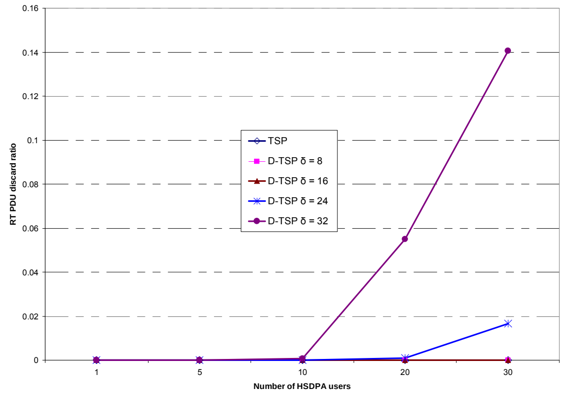

Figure 5: RT PDU discard ratio vs. number of users.

It is clear from the graph that with less than 30 users in the cell, D-TSP with 120ms delay budget can guarantee less than 2% discard with discard timer set to 160ms. However, when D-TSP is set with 160ms delay (which is the upper limit), with 20 users, about 5% of the PDUs are discarded by the DT, while with 30 users about 14 % of the RT PDUs are discarded by the DT. This implies that a 160ms delay budget setting for D-TSP is too high to be used in 30 user scenario without severely compromising the streaming RT flow QoS. However, it is worth noting also that the delay of RT PDUs is not due to D-TSP switching mechanism alone but also the high channel loading is a major contributing factor. Hence, from the results we conclude that considering both results (Figures 4 and 5 together) D-TSP is still effective in NRT throughput enhancement whilst keeping RT streaming losses to a minimum that will not violate its required QoS.

Lastly, we consider the UE_1 RT playout buffer to observe the effect of D-TSP on the end-to-end performance of the RT streaming flow. Since the RT streaming video is assumed to be 64 kbps CBR encoded, the arriving packets were buffered and played out at 64 kbps after an initial buffering delay equal to the maximum MAC-hs delay budget DB_{max} of 160ms. We measured the inter-packet playout delay i.e. the delay between each successive packet played out from the UE buffer. A constant delay is expected if the buffer always contains a packet for playout, otherwise if the buffer empties at certain times, delay spikes will occur. Figure 6 shows the observed delays between successive played out streaming packets for s-TSP over the 120s session with simultaneous NRT and RT streaming flows for all the channel load scenarios. The inter packet delay is observed to be constant at 0.005s (corresponding to 64 kbps playout rate of 320 bit long packets) indicating no playout jitter. Hence the de-jittering buffer was effective in eliminating any jitter in the arriving packets. The same result was obtained for D-TSP δ = 8 and 16 (not shown).

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Figure 7 shows the results for D-TSP 120ms (δ= 24) for 20 user and 30 user scenario, while Figure 8 shows that of D-TSP 160 ms (δ= 32) for 10, 20 and 30 user scenarios. All the omitted results for these D-TSP settings showed constant playout rate as in Figure 6. The 20 user scenario in Fig. 7 (top half) showed only a few instances where the delay spiked (i.e. playout gaps in successive packets). This delay spikes correspond to periods of empty buffer and being few, we can assume minimal impact on the RT stream quality. The same goes for the 10 user scenario of D-TSP 160ms in Fig. 8. Whereas, for the D-TSP 120ms 30 user scenario in Fig. 7 and D-TSP 160ms 20 and 30 user scenarios in Figure 8, the delay spikes are more frequent depicting high playout jitter which will severely compromise playout quality. On the other hand, we note that degradation in RT streaming QoS in the UE in these scenarios cannot be attributed to the effect of D-TSP alone, but also to channel congestion due to higher cell loading. Nevertheless, the results prove that with careful parameter setting, D-TSP can operate within end-to-end RT streaming QoS constraints.

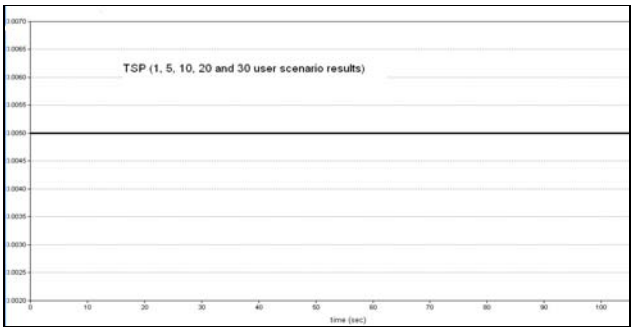

Figure 6: RT inter-packet playout delay for all s-TSP scenarios. The same constant playout rate was obtained for all user scenarios of D-TSP δ= 8 and D-TSP δ= 16.

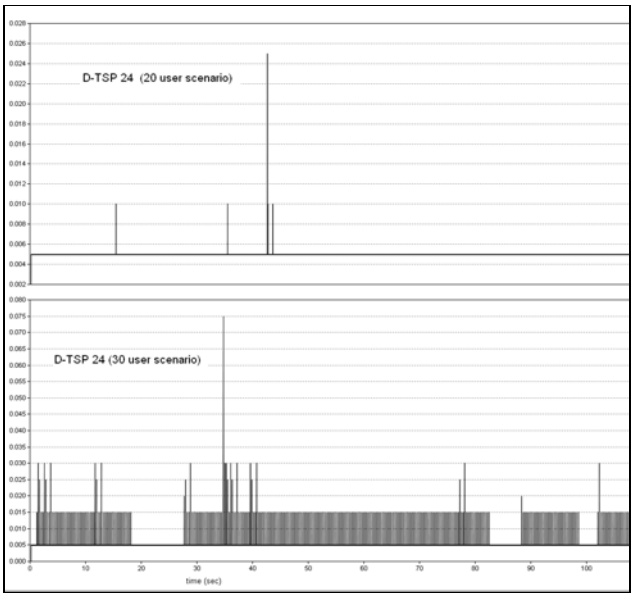

Figure 7: RT inter-packet playout delay for D-TSP δ= 24. 20 and 30 user scenarios are shown. The scenarios with less users gave constant inter-packet playout delay as in Figure 6.

V. CONCLUDING REMARKS

The dynamic buffer management scheme presented and evaluated in this paper allows effective QoS management of multiple flows in an end-user multimedia session with concurrent RT and NRT flows in 3.5G wireless networks and similar systems with buffering at the air interface. The novelty of the scheme lies in not only utilizing time and space priorities in a combined manner to suit the different QoS requirements of the RT and NRT flows, but also in employing transmission priority switching to further optimize QoS control. Based on evaluations in a simulated HSDPA system, the proposed buffer management scheme was efficient in mitigating end-to-end NRT bandwidth starvation whilst simultaneously maintaining acceptable RT streaming flow QoS for the UE multimedia session comprising both flows.

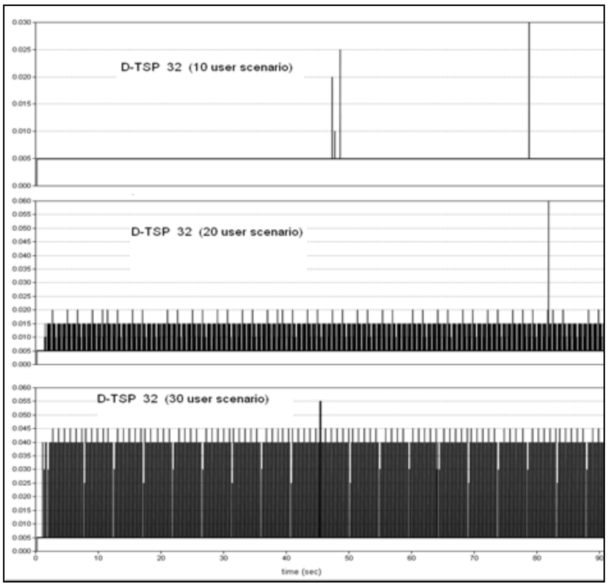

Figure 8: RT inter- packet playout delay for D-TSP δ= 32. 10, 20 and 30 user scenarios are shown. Scenarios with less users gave constant inter-packet playout delay as in Figure 6.

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