Performance Enhancing Proxy for Interactive 3G Network Gaming

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ABSTRACT

Unlike non-time-critical applications like email and file transfer, network games demand timely data delivery to maintain the seemingly interactive presence of players in the virtual game world. Yet the inherently large transmission delay mean and variance of 3G cellular links make on-time game data delivery difficult. Further complicating the timely game data delivery problem is the frequent packet drops at these links due to inter-symbol interference, fading and shadowing at the physical layer. In this paper, we propose a proxy architecture that enhances the timeliness and reliability of data delivery of interactive games over 3G wireless networks. In particular, a performance enhancing proxy is designed to optimize a new time-critical data type — variable-deadline data, where the utility of a datum is inversely proportional to the time required to deliver it. We show how a carefully designed and configured proxy can noticeably improve the delivery of network game data.

Categories and Subject Descriptors

C.2 [**Computer-Communication Networks**]: Network Architecture and Design—*Wireless communication*

General Terms

Algorithms, Performance, Design

Keywords

Wireless Networks, Network Gaming

1. INTRODUCTION

While network gaming has long been projected to be an application of massive economic growth, as seen in the recent explosive development on the wired Internet in South Korea and Japan, deployment of similar network games on 3G

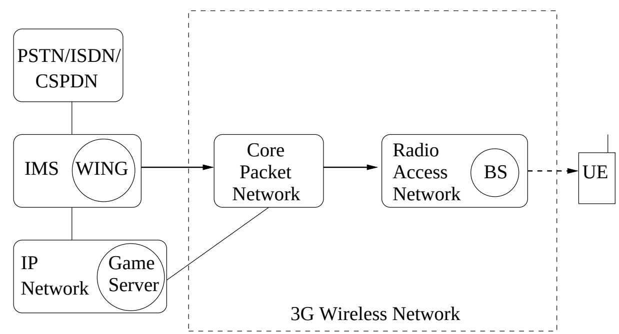


Figure 1: WING in 3G Wireless Network

wireless networks continues to be slow and difficult. One reason is that unlike their wired counterparts, wireless links are notoriously prone to errors due to channel fading, shadowing and inter-symbol interference. While 3G wireless networks, such as High Speed Downlink Packet Access (HSDPA) of 3rd Generation Partnership Project (3GPP) Release 5 (R5) [1] and CDMA 1x EvDO of 3GPP2 [5], combat wireless link failures at the MAC and physical layer with an elaborate system of channel coding, retransmission, modulation and spreading, with resulting packet loss rate being reduced to negligible 1 to 2%, the detrimental side-effect to network gaming is the large and often unpredictable transmission delay mean and variance [15]. Such large and variable delays greatly reduce the necessary interactivity of network game players and deteriorate the overall gaming experience.

In a separate development, a new 3G network element called IP Multimedia Subsystem (IMS) [3] has been introduced in 3GPP specifications R5 and later, as shown in Figure 1. The Session Initiation Protocol (SIP)-based IMS provides a multitude of multimedia services: from establishing connections from the legacy telephone networks to the new IP core network using Voice over IP (VoIP), to delivering streaming services such as video as a value-added service to mobile users (UE). Strategically located as a pseudo-gateway to the private and heavily provisioned 3G networks, it is foreseeable that IMS will continue to enlarge and enrich its set of multimedia services in future wireless networks.

In this paper, we propose a performance enhancing proxy (PEP) called (W)ireless (I)nteractive (N)etwork (G)aming Proxy (WING) to improve the timely delivery of network game data in 3G wireless networks. WING is located inside IMS as an application service on top of the myriad of services that IMS already provides. In a nutshell, WING improves the delivery of game data from the game server to 3G wireless game players[1] using the following three techniques. First, by virtue of locating at the intersection of the private wireless network and the open Internet, connection from the game server to the wireless game player can be strategically split; for the server-WING connection, only the statistically stable and fast round-trip time (RTT) and low wired-network-only packet loss rate (PLR) are used for congestion control, resulting in a steady yet TCP-friendly server-WING connection. Second, by configuring parameters in the radio link layer (RLC) specifically for gaming during session setup, excessive RLC retransmissions are avoided, and timeliness of game data is improved at the controlled expense of increased packet losses. Finally, by constructing small but error-resilient packets that contain location data, packets can be transmitted in fewer MAC-layer protocol data units (PDU) and hence further reduces delay.

The paper is organized as follows. Related work is presented in Section 2. We overview the 3G wireless system in focus, HSDPA of 3GPP R5, in Section 3. Note that because similar link and MAC layer transport optimizations that chiefly affect delay mean and variance are also employed in other 3G networks, our proposed WING can conceivably be applied to other wireless networks such as CDMA 1x EvDO of 3GPP2. We discuss the design of WING in details in Section 4. Finally, experimental results and conclusion are provided in Section 5 and 6, respectively.

2. RELATED WORK

We divide the discussion on the large volume of related work into two section. Section 2.1 discusses related research on wireless transport optimization. Section 2.2 discusses related research in transport of network game data.

2.1 Wireless Transport Optimization

We note that proxy-based transport optimization for last-hop wireless networks has a long history, with the majority of the research [4, 15] focusing on optimization of TCP over last-hop wireless networks. In particular, [15] showed that while 3G network packet losses can indeed be successfully overcome by using ample link layer retransmissions, the resulting large RTT mean and variance may severely affect the performance of a TCP-like congestion avoidance rate control that is based on end-to-end observable statistics of RTT and PLR. The limiting rate constraint and undesirable fluctuations can be alleviated using a proxy with split-connection — a theme we develop in Section 4.2.

Recently, efforts on proxy design have shifted to delay-sensitive multimedia transport [18, 13, 8, 9], though all of them focused exclusively on streaming media, while we focus on network gaming. Note that due to cited complexity reason, a competing end-to-end approach for rate control that does not rely on an intermediate proxy is popular as well [17, 6]. However, we chose the proxy-based approach and will juxtapose its advantages in Section 4.2.

[1]While peer-to-peer model for interactive network games is also possible, we assume the more common server-client model where the game server maintains and disseminates all game states in this paper.

2.2 Transport of Network Game Data

In [3], a general gaming platform for IMS that provides network services needed for network gaming such as session setup and registration is proposed to ease deployment over 3G networks. Our work is orthogonal to [3] since we focus only on the efficient transport of game data.

An early work on gaming protocol is [10], which defined a Game Transport Protocol (GTP) for massive multi-player on-line games (MMPOGs) over the Internet. Our proposed gaming proxy WING differs in the following respects: i) we design WING specifically for lossy, bandwidth-limited networks, hence focusing on design of network-optimized differential coding to produce small but loss-resilient packets; and, ii) we tailor WING for HSDPA of 3G wireless networks, optimizing performance by intelligently configuring parameters of the RLC layer.

The most similar related work is [12], which proposed an end-to-end adaptive FEC and dynamic packetization algorithm to combat packet losses due to wireless link failures and reduce packet sizes. Unlike [12], our approach is proxy-based, and we tailor our gaming optimization exclusively for 3G networks.

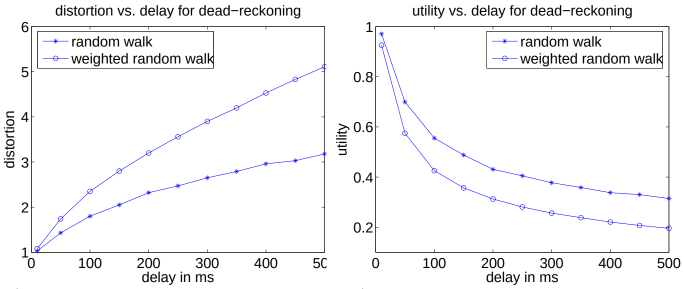
3. OVERVIEW OF UMTS RELEASE 5

HSDPA of UMTS Release 5, also known as 3.5G, improves upon Release 4 with numerous lower-layer optimizations. First, a shared channel is periodically scheduled to users in the cell with good observable network conditions to take advantage of user diversity during fading without sacrificing fairness. Second, an elaborate MAC-layer scheme chooses an appropriate combination of FEC, hybrid ARQ, modulation and spreading based on client observable network state. In this section, we instead focus on the RLC layer, where the user has limited control over behavior using configuration of parameters during session setup.

The Radio Link control (RLC) layer [1] buffers upper layer service data units (SDU) on a per-session basis — IP packets in this case, and segments each SDU into smaller protocol data units (PDU) of size S_{PDU} and await transmission at lower layers. There are three transmission modes: transparent mode (TM), unacknowledged mode (UM) and acknowledged mode (AM). Only AM performs link-layer retransmissions if transmission in the lower layer fails. For error resiliency, we focus only on AM. In particular, we look at how SDUs are discarded in the RLC layer: using a method of retransmission-based discard (RBD), an SDU can be discarded before successful transmission. In a nutshell, an SDU is discarded if a predefined maximum number of retransmissions B has been reached before successful transmission of a PDU belonging to the SDU. We will investigate how the value B can be selected to trade off error resiliency with delay in Section 4.3.

4. WING FOR WIRELESS INTERACTIVE NETWORK GAMING

Before we discuss the three optimizations of our proposed gaming proxy WING in details in Section 4.2, 4.3 and 4.4, we first define a new type of transport data called *variable deadline data* in Section 4.1 — a consequence of a prediction procedure used at a network game client to predict locations of other game players in the virtual game world.



a) distortion vs. delay b) utility vs. delay

Figure 2: Examples of Dead-Reckoning

4.1 Variable Deadline Data Delivery

Unlike media streaming applications where a data unit containing media data is fully consumed if it is correctly delivered by a playback deadline and useless otherwise [9], the usefulness (utility) of a game datum is inversely proportional to the time it requires to deliver it. This relationship between utility and transmission delay is the behavioral result of a commonly used game view reconstruction procedure at a game client called *dead-reckoning* [2]. It works as follows. To maintain time-synchronized virtual world views among game players at time t_0, a player P_A predicts the location $\bar{\xi}_{t_0}$ of another player P_B and draws it in P_A's virtual world at time t_0, extrapolating from previously received location updates of P_B in the past, $\xi_\tau, \tau < t_0$. When location update ξ_{t_0} arrives at P_A from P_B at a later time t_1, P_A updates its record of P_B's locations with (t_0, ξ_{t_0}), in order to make an accurate prediction of $(t_1, \bar{\xi}_{t_1})$ for display in P_A's virtual world at time t_1. Regardless of what prediction method is used at the client, it is clear that a smaller transmission delay will in general induce a smaller prediction error. We term this type of data with inversely proportional relationship between quantifiable utility and delay *variable deadline data*. We next show examples of how such utility-delay curve $u(d)$ can be derived in practice given a player movement model and a prediction method.

4.1.1 Examples of Dead-Reckoning

We first consider two simple movement models that model a game player in two-dimensional space (x, y). The first is *random walk*, where for each time increment t, probability mass function (pmf) of random variable of x-coordinate x_t, $p(x_t)$, is defined as follows:

$$\begin{aligned} p(x_{t+1} = x_t + 1) &= 1/3 \\ p(x_{t+1} = x_t) &= 1/3 \\ p(x_{t+1} = x_t - 1) &= 1/3 \end{aligned} \tag{1}$$

Random variable of y-coordinate y_t is calculated similarly and is independent of x_t.

The second movement model is *weighted random walk*, whose pmf is defined as follows:

$$\begin{aligned} p\left(x_{t+1} = x_t + ((x_t - x_{t-1} + 1) \bmod 2)\right) &= 1/6 \\ p\left(x_{t+1} = x_t + (x_t - x_{t-1})\right) &= 2/3 \\ p\left(x_{t+1} = x_t + ((x_t - x_{t-1} - 1) \bmod 2)\right) &= 1/6 \end{aligned} \tag{2}$$

In words, the player continues the same movement as done in the previous instant with probability 2/3, and changes to one of two other movements each with probability 1/6. Random variable y-coordinate y_t is calculated similarly.

We defined a simple prediction method called 0*th-order prediction* as follows: each unknown x_t is simply set to the most recently updated x_τ. Using each of the two movement models in combination with the prediction method, we constructed distortion-delay curves experimentally as shown in Figure 2a. As seen, 0th-order prediction is a better match to random walk than weighted random walk, inducing a smaller distortion for all delay values. Utility $u(d)$ — shown in Figure 2b – is simply the reciprocal of distortion. Having derived $u(d)$ gives us a quantifiable metric on which we can objectively evaluate game data transport systems.

4.2 Proxy-based Congestion Control

We argue that by locating WING between the open wired Internet and the provisioned wireless networks to conduct split-connection data transfer, stable TCP-friendly congestion control can be maintained on top of UDP in the wired server-WING connection. Traditional congestion control algorithms like TCP-friendly Rate Control (TFRC) [11] space outgoing packets with interval T_{cc} as a function of estimated packet loss rate (PLR) ϵ_{cc}, RTT mean m_{cc} and RTT variance σ^2_{cc} due to wired network congestion:

$$T_{cc} = m_{cc}\sqrt{2\epsilon_{cc}/3} + 3(m_{cc} + 4\sigma^2_{cc})\epsilon_{cc}\left(1 + 32\epsilon^2_{cc}\right)\sqrt{3\epsilon_{cc}/8} \tag{3}$$

Past end-to-end efforts [17, 6] have focused on methodologies to distinguish wired network congestion losses from wireless link losses, in order to avoid unnecessary rate reduction due to erroneous perception of wireless losses as network congestion. Split connection offers the same effect regarding PLR by completely shielding sender from packet losses due to wireless link failures. Moreover, by performing TFRC (3) in the server-WING connection using only stable wired network statistics, split connection shields the server-WING connection from large rate fluctuations due to large RTT variance in the last-hop 3G link as shown in [15]. For this reason, [15] showed experimentally that indeed proxy-based split-connection congestion control performs better than end-to-end counterparts, even in negligible wireless loss environments.

Lastly, we note that split connection can benefit from a *rate-mismatch* environment [8, 9], where the available bandwidth R_1 in the server-WING connection is larger than the available bandwidth R_2 in the WING-client connection. In such case, the surplus bandwidth $R_1 - R_2$ can be used for redundancy packets like forward-error correction (FEC) or retransmission to lower PLR in the server-WING connection. We refer interested readers to [8, 9] for further details.

4.3 Optimizing RLC Configuration

Given utility-delay function $u(d)$ in Section 4.1, we optimize configuration of RLC to maximize utility. More precisely, we pick the value of maximum retransmission limit B — inducing expected SDU loss rate l^* and delay d^*, so that the expected utility $(1 - l^*)u(d^*)$ is maximized.

We assume a known average SDU size S_{SDU}, PDU loss rate ϵ_{PDU}, and probability density function (pdf) of PDU transmission delay $\Phi(\phi)$ with mean m_ϕ and variance σ^2_ϕ. First, the expected number of PDUs fragmented from an SDU is $N = \left\lceil \frac{S_{SDU}}{S_{PDU}} \right\rceil$. For a given B, the expected SDU loss rate l_{SDU} can be written simply:

$$P_{PDU} = \sum_{i=1}^{B} \epsilon_{PDU}^{i-1}(1 - \epsilon_{PDU}) \quad (4)$$

$$l_{SDU} = 1 - P_{PDU}^{N} \quad (5)$$

where P_{PDU} is the probability that a PDU is successfully delivered given B.

The delay d_{SDU} experienced by a successfully delivered SDU is the sum of queuing delay d_{SDU}^{q} and transmission delay d_{SDU}^{t}. Queuing delay d_{SDU}^{q} is the delay experienced by an SDU while waiting for head-of-queue SDUs to clear due to early termination or delivery success. d_{SDU}^{t} is the expected wireless medium transmission delay given the SDU is successfully delivered. d_{SDU}^{t} is easier and can be calculated as:

$$X_{PDU} = \frac{1}{P_{PDU}} \sum_{i=1}^{B} i\epsilon_{PDU}^{i-1}(1 - \epsilon_{PDU}) \quad (6)$$

$$d_{SDU}^{t} = N \; m_{\phi} \; X_{PDU} \quad (7)$$

where X_{PDU} is the expected number of PDU (re)transmissions given PDU delivery success. To calculate d_{SDU}^{q}, we assume a $M/G/1$ queue[2] with arrival rate λ_q, mean service time m_q, and variance of service time σ_q^2. Using Pollaczek-Khinchin mean value formula [14], d_{SDU}^{q} can be written as:

$$d_{SDU}^{q} = \frac{\lambda_q m_q \left(1 + \sigma_q^2/m_q^2\right)}{2\left(1 - \lambda_q m_q\right)} m_q \quad (8)$$

In our application, λ_q is the rate at which game data arrive at WING from the server, which we assume to be known. m_q is the mean service rate for *both* cases of SDU delivery success and failure and can be derived as follows:

$$Y_{SDU} = \frac{1}{l_{SDU}} \sum_{i=1}^{N} \left(B + (i-1)X_{PDU}\right) \epsilon_{PDU}^{B} P_{PDU}^{i-1} \quad (9)$$

$$m_q = \left(1 - l_{SDU}\right) d_{SDU}^{t} + l_{SDU} \; m_{\phi} \; Y_{SDU} \quad (10)$$

where Y_{SDU} is the expected total number of PDU (re)transmissions in an SDU given SDU delivery failure. Similar analysis will show that the variance of service rate σ_q^2 for our application is:

$$\sigma_q^2 = \left(1 - l_{SDU}\right) \; N^2 \; X_{PDU}^2 \; \sigma_{\phi}^2 + \; l_{SDU} \; Y_{SDU}^2 \; \sigma_{\phi}^2 \quad (11)$$

We can now evaluate expected queuing delay d_{SDU}^{q}, from which we evaluate expected delay d_{SDU}. Optimal B^* is one that maximizes $(1 - l_{SDU}^{*})\, u(d_{SDU}^{*})$.

4.4 Loss-optimized Differential Coding

If the location data — player position updates sent to improve dead-reckoning discussed in Section 4.1 — are in absolute values, then the size of the packet containing the data can be large, resulting in large delay due to many PDU fragmentation and spreading. The alternative is to describe the location in relative terms — the difference in the location from a previous time slot. Differential values are smaller, resulting in fewer encoded bits and smaller packets, and hence smaller transmission delay. This *differential coding* of location data is used today in networked games.

The obvious disadvantage of differential coding is that the created dependency chain is vulnerable to network loss; a single loss can result in error propagation until the next absolute location data (refresh).

To lessen the error propagation effect while maintaining the coding benefit of differential coding, one can reference a position in an earlier time slot. An example is shown in Figure 3, where we see position 3 (ξ_3) references ξ_1 instead of ξ_2. This way, loss of packet containing ξ_2 will not affect ξ_3, which depends only on ξ_1. The problem is then: for a new position ξ_t, how to select reference position ξ_{t-r} for differential coding such that the right tradeoff of error resilience and packet size can be selected? This selection must be done in an on-line manner as new position becomes available from the application to avoid additional delay.

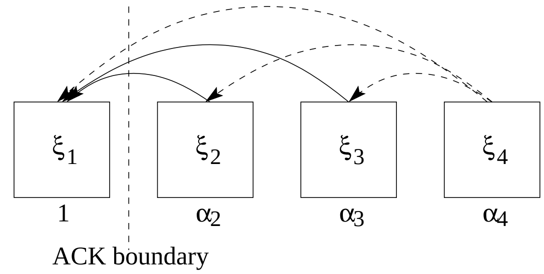


Figure 3: Example of Differential Coding

mode	mode marker	ref. size	coord. size	total
0	00	0	32	$2 + 64n$
1	01	0	16	$2 + 32n$
2	10	2	8	$4 + 16n$
3	11	4	4	$6 + 8n$

Table 1: Differential Coding Modes

4.4.1 *Specifying Coding Modes*

To implement loss-optimized differential coding, we first define a coding specification that dictates how the receiver should decode location packets. For simplicity, we propose only four coding modes, where each mode is specified by a designated bit sequence (mode marker) in the packet. Assuming the original absolute position ξ is specified by two 32-bit fixed point numbers, mode 0 encodes the unaltered absolute position in x-y order, resulting in data payload size of $2 + 64n$ bits for n game entities. Mode 1 uses the previous position as reference for differential encoding with 16 bits per coordinate, resulting in $2 + 32n$ bits for n entities. Mode 2 uses the first 2 bits to specify r in reference position $t - r$ for differential encoding. Each coordinate takes 8 bits, resulting in $4 + 16n$ total bits for n entities. Mode 3 is similar to mode 2 with the exception that each of the reference marker and the two coordinate takes only 4 bits to encode, resulting in $6 + 8n$ bits for n entities.

For given position $\xi_t = (x_t, y_t)$ and reference $\xi_{t-r} = (x_{t-r}, y_{t-r})$, some modes may be infeasible due to the fixed coding bit budgets for reference and coordinate sizes. So limited to the set of feasible modes, we seek a reference position / mode pair that maximizes an objective function.

[2] Our system is actually more similar to a $D/G/1$ queue, since the arrivals of game data are more likely to be deterministic than Markovian. Instead, we use $M/G/1$ queue as a first-order approximation.

	PLR	RTT mean	RTT variance
Tokyo-Singapore (50)	0	94.125ms	178.46
Tokyo-Singapore (100)	0	95.131ms	385.30
Tokyo-Singapore (200)	0	96.921ms	445.63
HSDPA (50)	0	62.232ms	7956.8
HSDPA (100)	0	72.547ms	25084
HSDPA (200)	0	152.448ms	143390

Table 2: Comparison of Network Statistics

4.4.2 Finding Optimal Coding Modes

For an IP packet of size s_t containing position ξ_t that is sent at time t, we first define the probability that it is correctly *delivered* by time τ as $\alpha_t(\tau)$. $\alpha_t(\tau)$ depends on expected PLR $l(s_t)$ and delay $d(s_t)$, resulting from retransmission limit B chosen in Section 4.3:

$$N(s_t) = \left\lceil \frac{s_t}{S_{PDU}} \right\rceil \quad (12)$$

$$l(s_t) = 1 - (P_{PDU})^{N(s_t)} \quad (13)$$

$$d(s_t) = d^q_{SDU} + N(s_t)\; m_\phi \; X_{PDU} \quad (14)$$

where $N(s_t)$ is the number of PDUs fragmented from an SDU of size s_t. $l(s_t)$ is PLR in (5) generalized to SDU size s_t. $d(s_t)$ is the expected queuing delay in (8) plus the transmission delay in (7) generalized to SDU size s_t. We can now approximate $\alpha_t(\tau)$ as:

$$\alpha_t(\tau) \approx \begin{cases} 1 & \text{if ACKed by } \tau \\ (1 - l(s_t))\mathbf{1}\,(\tau - t - d(s_t)) & \text{o.w.} \end{cases} \quad (15)$$

where $\mathbf{1}(x) = 1$ if $x \geq 0$, and $= 0$ otherwise. If no acknowledgment packets (ACK) are sent from client to WING, then $\alpha_t(\tau)$ is simply the second case in (15).

We next define the probability that position ξ_t is correctly *decoded* by time τ as $P_t(\tau)$. Due to dependencies resulting from differential coding, $P_t(\tau)$ is written as follows:

$$P_t(\tau) = \alpha_t(\tau) \prod_{j \prec t} \alpha_j(\tau) \quad (16)$$

where $j \prec i$ is the set of positions j's that precedes t in the dependency graph due to differential coding.

Given utility function $u(d)$ in Section 4.1 and decode probability (16), the optimal reference position / mode pair is one that maximizes the following objective function:

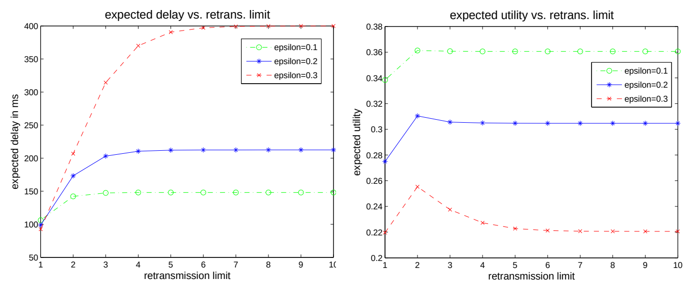
$$\max P_t(t + d(s_t))\; u(d(s_t)) \quad (17)$$

5. EXPERIMENTATION

We first present network statistics for HSDPA and discuss the implications. We collected network statistics of 10,000 `ping` packets, of packet size 50, 100 and 200 bytes, spaced 200ms apart, between hosts in Tokyo and Singapore inside HP intranet. The results are shown in Table 2. We then conducted the same experiment over a network emulator called `WiNe2` [16] emulating the HSDPA link with 10 competing ftp users each with mobility model `Pedestrian A`. We make two observations in Table 2. One, though results from both experiments had similar RTT means, HSDPA's RTT variances were very large, substantiating our claim that using split-connection to shield the server-WING connection from HSDPA's RTT variance would drastically improve TFRC bandwidth (3) of server-WING connection.

number of entities n	4
frame rate	10 fps
IP + UDP header	20 + 8 bytes
RLC PDU size	40 bytes
RLC PDU loss rate	0.1 to 0.3
average packet size	61 bytes
shifted Gamma parameter α_g	2
shifted Gamma parameter λ_g	0.1
shifted Gamma parameter κ_g	10.0

Table 3: Simulation Parameters



a) delay vs. retrans. limit B b) utility vs. retrans. limit B

Figure 4: Delay and Utility vs. Retrans. Limit

Two, larger packets entailed larger RTT means for HSDPA. This means that the differential coding discussed in Section 4.4 indeed has substantial performance improvement potential.

We next used an internally developed network simulator called (mu)lti-path (n)etwork (s)imulator (`muns`) that was used in other simulations [7] to test RLC configurations and differential coding. For PDU transmission delay $\Phi(\phi)$, we used a shifted Gamma distribution:

$$\Phi(\phi) = \frac{\lambda_g^{\alpha_g}\,(\phi - \kappa_g)^{\alpha_g - 1}\, e^{-\lambda_g(\phi - \kappa_g)}}{\Gamma(\alpha_g)}, \qquad \kappa_g < \phi < \infty \quad (18)$$

where $\Gamma(\alpha)$ is the Gamma function [14]. The parameters used are shown in Table 3.

Figure 4 shows the expected delay and utility as a function of retransmission limit B for different PDU loss rates. As expected, when B increases, the expected delay increases. The expected utility, on the other hand, reaches a peak and decreases. For given PDU loss rate, we simply select B with the largest expected utility.

Next, we compare the results of our loss-optimized differential coding optimization `opt` in Section 4.4 with two schemes: `abs`, which always encodes in absolute values; and, `rel`, which uses only previous frame for differential coding and refreshes with absolute values every 10 updates. `abs`

ϵ_{PDU}	B^*	abs(1)	abs(B^*)	rel(1)	rel(B^*)	opt
0.10	2	1.181	1.134	1.806	1.154	1.070
0.15	3	1.222	1.166	2.288	1.108	1.073
0.20	2	1.290	1.192	2.619	1.380	1.086
0.25	2	1.356	1.232	3.035	1.568	1.090
0.30	2	1.449	1.268	3.506	1.750	1.110
0.35	2	1.509	1.300	3.556	2.054	1.121

Table 4: Distortion Comparison

represents the most error resilient coding method in differential coding, while `rel` represents a reasonably coding-efficient method with periodical resynchronization. Note, however, that neither `abs` nor `rel` adapts differential coding in real time using client feedbacks.

`abs` and `rel` were each tested twice. In the first trial, limit B was set to 1, and in the second, B was set to the optimal configured value as discussed in Section 4.3. 20000 data points were generated and averaged for each distortion value in Table 4. As we see in Table 4 for various PDU loss rate ϵ_{PDU}, the resulting distortions for `opt` were always lower than `abs`'s and `rel`'s, particularly for high PDU loss rates. `opt` performed better than `rel` because of `opt`'s error resiliency of loss-optimized differential coding, while `opt` performed better than `abs` because `opt`'s smaller packets induced a smaller queuing delay and a smaller transmission delay due to smaller number of RLC fragmentations. This demonstrates that it is important not only to find an optimal RLC configuration, but a suitable differential coding scheme to match the resulting loss rate and delay of the configuration.

6. CONCLUSION

We propose a performance enhancing proxy called WING to improve the delivery of game data from a game server to 3G game players using three techniques: i) split-connection TCP-friendly congestion control, ii) network game optimized RLC configuration, and, iii) packet compression using differential coding. For future, we will investigate how similar techniques can be applied for the 3G uplink from game player to game server.

7. ACKNOWLEDGMENTS

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