Evaluation of UMTS RLC Parameters for MPEG-4 Video Streaming

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ABSTRACT

Universal Mobile Telecommunication Systems (UMTS) known as Third Generation (3G) mobile phone systems using Wideband Code Division Multiple Access (WCDMA) standard aim to provide a high bit rate of services to enable a high quality of multimedia communications. The 3G specification entity, 3GPP, has defined a reliable link layer protocol, Radio Link Control (RLC), for hiding transmission errors from upper layers. Due to the complexity of the protocol and the a number of parameter configurations available, there are many ways that can degrade a system performance, such as incorrect parameters configuration, buffer missmanagement, protocol stalling, etc. MPEG-4 is widely used for video stream services due to its good video quality at low bit rate. In this paper, we have studied on the impacts of RLC parameters, especially RLC timer and polling mechanism, for MPEG-4 video stream transmission by using a computer simulation. Our simulation results provide some suggestions value to optimize such parameters, e.g. RLC time and polling.

Keywords: UMTS, 3G, WCDMA, MPEG-4, 3GPP, RLC

1. INTRODUCTION

The third generation (3G) wireless system based on WCDMA technology is designed for multimedia communications that can be enhanced with high quality voice, images, video, and multimedia communications because 3G systems provide a high bit rate of services, e.g. 2 Mbps [1]. UMTS is among the first 3G mobile systems to offer wireless wideband multimedia communications over the Internet protocol [2]. The mobile internet users can access a variety of multimedia contents on the internet at data rates between 384 kbps and 2 Mbps in a wide coverage area with high user mobility capable. The Radio Link Control (RLC) [3] is one of the major radio interface protocols of 3G systems consisting of flow control and error recovery. RLC is a sub-layer of layer 2 in 3G protocol reference model. It is on the top of the Medium Access Control (MAC) layer. RLC consists of three operation modes: Transparent Mode (TM), Unacknowledged Mode (UM), and Acknowledged Mode (AM). The latest mode offers a reliable data delivery, it can recover frame losses in the radio access network by using a selective repeat automatic repeat request (SR-ARQ) algorithm while TM and UM do not guarantee data delivery. Our study, in our computer simulation, is based on AM.

Video multimedia, e.g. video communication, video on demand, is expected to be widely used for 3G applications. MPEG-4 [4] is an open standard developed by the Moving Picture Expert Group (MPEG) [5] for low bit rate digital media applications. It produces good quality video at affordable bit rate for mobile and wireless services. Due to RLC protocol has a number of features and options for protocol tuning up, we may not get the best performance without a good parameter configuration. A suitable configuration for RLC parameters can be made without requiring any modifications at the end host. In this paper, we have studied some impacts of parameter values on the performance of MPEG-4 video stream services over 3G phone systems. We have suggested some optimal values of such parameters.

The next section of this paper provides an overview of the RLC protocol: description, operations, and the main RLC parameters. In section 3, some background information of MPEG-4 video codec is presented. The computer simulation models; tools, implementation, and parameters setting are described in Section 4. Section 5 presents the simulation results and analysis. Finally, conclusions and future works are addressed in section 6.

2. RLC PROTOCOL IN UMTS

The UMTS network architecture consists of three components: the Core Network (CN), the UMTS Terrestrial Radio Access Network (UTRAN), and the User Equipment (UE), as shown in Fig.1. Between UTRAN and UE is the radio interface called Uu where RLC is a sub-layer of layer 2 in this interface [6]. RLC sits above the MAC sub-layer which handles the scheduling of radio bearers with difference QoS requirements and mapping the logical channels into transport channels. On the top of RRC, there is Radio Resource Control (RRC) layer, which is re-

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sponsible for setting up, modifying, and releasing all the lower layer protocol entities.

There are three operation modes in RLC; Transparent Mode (TM), Unacknowledged Mode (UM), and Acknowledged Mode (AM). However, in this paper, we will describe only the AM mode which is used in our simulation. This mode provides the segmentation, reassembly, sequence number checking, error recovery, and flow control services that hide transmission errors from upper layers as well as reduce the chances of mistaken invocations of the upper layers congestion control mechanisms by mean of automatic repeat request algorithm. There are mainly two kinds of RLC PDU:

• Acknowledged Mode Data PDUs (AMD PDU) is used to transfer user data, piggyback status information, sequence number, length indicator and polling bit to poll the receiver. • Status information control PDU (STATUS PDU) is used to exchange status information between receiver and sender (missing/ erroneous PDU, SDU discard notification, window size, etc).

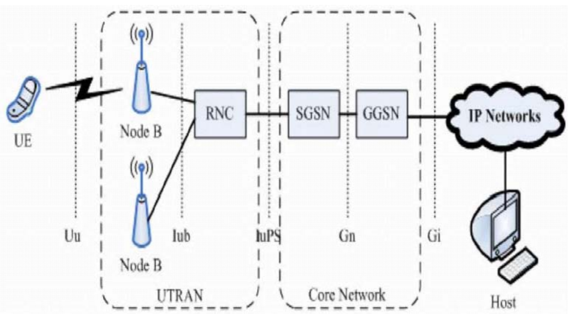

Fig.1: *UMTS Architecture*

When AMD PDUs are transmitted to a receiver, the sender will know the status of the reception from STATUS PDUs that come from the receiver. If the erroneous or missing AMD PDUs are detected by receiver, the NACK (negative acknowledgement) will send to the sender to request retransmission and that missing PDUs can be selectively retransmitted with high priority whereas the sender can advance its transmission window if one or more in-sequence frames are acknowledged, so that new AMD PDUs can be sent. The mechanisms in AM RLC that have the effects with performance of RLC and needed to make flow control and error control more efficiently are polling, status transmission and SDU discard mechanism.

2.1 Polling mechanism

For the polling mechanism, the sender sends the polling request to the receiver and the receiver sends one or more status report back to the sender indicating which of the RLC PDU have been successfully received and which have not. Retransmission is triggered if a negative acknowledgement is indicated in a status report. There are eight triggers in the transmitting side and used to set the polling bit in the PDUs header when this mechanism is initiated: poll timer, last PDU in buffer, last PDU in retransmission buffer, every Poll-PDU PDU, every Poll-SDU SDU, poll prohibit timer, window based and timer based. Which of the triggers will be used is/are decided by upper layer.

Last PDU in buffer and last PDU in retransmission buffer can avoid deadlock of the RLC entities [7]. For other triggers to control the frequency of polling the receiver and frequency of sending the status reports of the receiver including to control the acknowledgment feedback delay too. So the delay performance will be improved with fast polling request. It means that the more frequently status report is generated. But the status reports cause overhead, if there are too much status reports, the throughput and delay will be degraded.

2.2 Status transmission mechanism

When an incoming PDU of the receiver contains a poll request, it always generates and sends back a status report which is carried by one or more status PDUs. Another way of sending status PDUs more aggressively is driven by the receiver instead of the sender. Such mechanisms are the detection of missing PDUs trigger can make the receiver send status PDUs to indicate which PDUs are detect as missing or erroneous and request for retransmission. This trigger affects to make the RLC SDU delays shorten. The status period timer trigger controls frequency of sending status PDUs, these status PDUs are sent to sender periodically. The Estimated PDU Counter (EPC) mechanism and the status prohibit timer prevent excessive exchange of status PDUs between the sender and the receiver. The status packet consists of smaller parts called super fields (SUFIs) indicate which PDUs have been received and which are detected as missing in order to the sender will transmit next PDUs or retransmit the missing PDUs.

2.3 SDUs discard mechanism

This mechanism allows an RLC sender to discharge RLC PDUs associated with a SDU from the transmission and retransmission buffer in case the buffers are full in TM and otherwise, the transmission of those PDUs is unsuccessful within a period of time or for a number of transmission attempts. This mechanism can avoid buffer overflow in the RLC layer and reduce the maximum transmission delay. The RLC receiver shall be notified of the discard, if this is configured by upper layer. There are three SDU discard functions that can be configured according to the QoS requirement of the Radio Access Bearer: Timer based discard with/without explicit signaling and SDU discard after MaxDat number of transmission. The differences between these functions are:

• "Timer based discard" function discards the SDU after its corresponding timer, timer-discard, expires. The SDU discard is upon this timer so, the variations of the channel data rate and error rate does not have an effect to the delay performance. The SDU loss rate increases as RLC SDUs are discarded.

• "SDU discard after MaxDat number of transmission" tries to keep the SDU loss rate constant but the delay performance is variable and dependent on the channel condition because the discard SDUs occurs when value VT(DAT)1 of PDU reaches the value MaxDat.

Besides the mechanisms and parameters are described above there are other that can affect to the performance of streaming MPEG-4 video but they are out of the scope. For our simulation investigates only the impacts of a poll timer, poll PDU and status prohibit timer.

3. OVERVIEW OF MPEG-4

MPEG-4 [4] is a video codec standardized by the Moving Picture Expert Group (MPEG) for low bit rate digital media applications. There are several parts in MPEG-4 such as the system, video coding, audio coding, file format, etc, in this paper, we used only the part of video coding. There are three types of MPEG-4 encoded frames; intra frame (I-frame) contains information from encoding a still image, predicted frame (P-Frame) are encoded from previous I-frame or P-frame and bidirectional frame (B-frame) are encoded bidirectionally from the preceding and the following Iframe and P-frame, the relation is shown in Fig.2. The dependency between each type of frames is shown in Iframes have the lowest compression and the error will be recovered only when an I-frame is received because the Iframe does not depend on other frame.

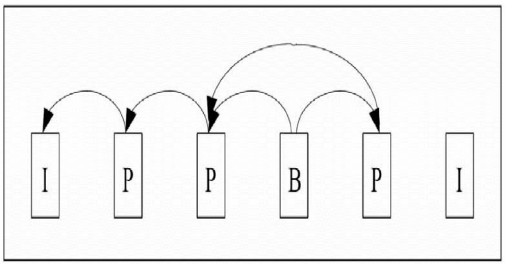

Fig.2: *MPEG-4 frame type*

4. SIMULATION DESCRIPTION

The simulation model is used to analyze the UMTS RLC parameters setting and their impact on the performance of MPEG-4 video streaming over UMTS created on ns-2 (Network Simulator version 2) [8]. Several extensions were made to this simulator for modeling UMTS. With the extensions, we chose an EURANE (Enhanced UMTS Radio Access Network Extensions) [9] developed within the European Commission 5th framework project (SEACORN) which instances of UMTS nodes (UE, Node B and RNC) can be instantiated.

And the MPEG-4 video streaming is conversed to trace file which contains video frame number, frame length in byte, frame type (I, P or B frame), frame quantize, Y-PSNR, U-PSNR and V-PSNR by using Java programming application. The trace file is then fed to the streaming server in the ns2 environment to generate video streaming over UMTS.

Transport protocols are used in the simulation to transfer video streaming are User Datagram Protocol (UDP), Real-time Transport Protocol (RTP) and Realtime Control Protocol (RTCP) that are formed to a new ns-2 agent, the same as used in [10] which can read the video trace file and send it via the transport protocols above. The network model used for simulation analysis is illustrated in Fig.3.

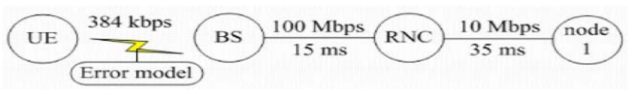

Fig.3: *Network simulation model*

A UE plays the role of a streaming client and a fix host is the streaming server in the internet, so this situation only tests the performance in a downlink direction.

For UMTS RLC parameters setting, 3GPP specifications define several RLC operation options and configurable parameters [3, 11, 12], moreover there are many previous works [13-16] have been studied about the suitable value for setting RLC parameters but it is still no general guideline. And based on previous researches and our own findings, a default parameters setting are shown in Table 1 after that we varied the value of poll timer, poll PDU and status prohibit timer to find the suitable value setting that give the best performance in many MPEG-4 video streaming environments.

Moreover, the video file which we used in this simulation imitates from the real MPEG-4 video file that it has I- and P-frame type and I-frame is bigger than Pframe. That mean, this video file which fed into UMTS stack in ns-2 is not converted from the real MPEG-4 file because it is easier for result analysis. The video characteristics are shown in Table 2.

5. SIMULATION RESULTS AND DISCUSSIONS

The effect of the poll timer on the UE received frame delay when using difference values of poll timer with the same error rate and the difference error rates with the same poll timer values. In the first case

Table 1: *Simulation parameters*

App	MPEG-4				
RTP	RTP Header size (bytes)	12			
UDP	Maximum Segment Size (bytes)	1460			
	UDP Header size (bytes)	8			
IP	IP Header size (bytes)	20			
RLC	RLC Mode	AM			
	Payload size (bytes)	40			
	RLC header (bytes)	2			
	Transmission window (PDUs)	1024			
MAC	MAC Header (bytes)	0			
	MAC Multiplexing	Not required for DCH			
PHY	Physical Channel Type	DCH			
		Uplink		Downlink	
		Bit rate (kbps)	TTI (ms)	Bit rate (kbps)	TTI (ms)
		384	10	384	10
	Transport BLER	0 - 25%			
	Error Model	Exponential Distribution			

Table 2: *MPEG-4 Video parameters*

Number of frame long (frames)	1200
Key frame interval (frames)	5
Length of I-frame (bytes)	9000
Length of P-frame (bytes)	2000
Sending rate (fps)	10
Sending time (sec)	120

is illustrated in Fig.4, we have found that a smaller poll timer value gives better frame delay even though the error rate in air interface is increased but not for all cases as the result shown in Fig.5 when the error rate is up to 20timer values do not have any effect to the received frame delay that mean it has the limitation of error rate that poll timer can improve the performance. This frame delay is very importance for playing back at the receiver because if the delay is more than the play back buffer time, the receiver will think that frame is lost even if it doesnt that means the video which end user see is not smooth at that time but in this simulation, the play back buffer time is not set for comparing the results

And in the second case, the difference error rates with the same poll timer values, we can see the maximum of error rate that the poll frequent can has the effect to frame delay. As shown in Fig.6 and Fig.7, when we fixed the poll timer value and then tried to increase error rate in each time. We have found that poll timer can give low received frame delay only

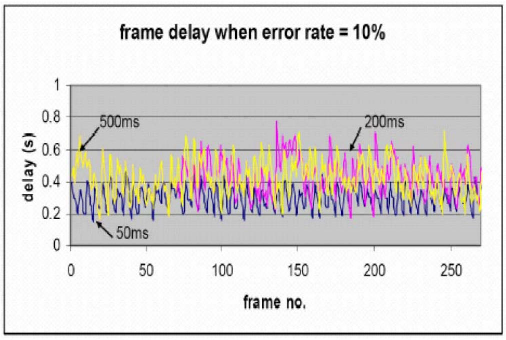

Fig.4: *The comparison of frame delay when using difference poll time with 10% error rate*

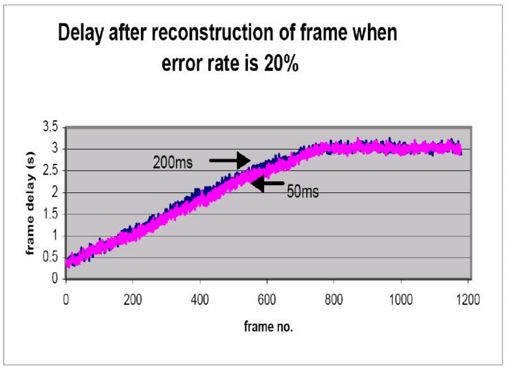

Fig.5: *The comparison of frame delay when difference poll time with link error rate is 20%*

when error rate value not more than 18% both 50ms and 200ms poll timer. The frame delay will be higher if the error rate is increased. But the high frequency of poll leads to many status reports will be sent back to the sender that means to a lot of bandwidth in the backward channel are used but in our simulations we do not monitor in that value, we attend the traffic in the forward channel only so the results do not show how the bandwidth of the backward channel be.

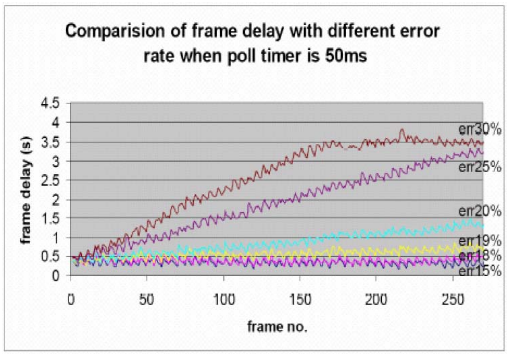

Fig.6: *The comparison of frame delay when using 50ms poll time at difference error rate*

From the results in Fig.4 Fig.7, we can summarize them into Table 3 for the average values. It shows that increasing poll frequency does not reduce the average time delay when error rate goes beyond

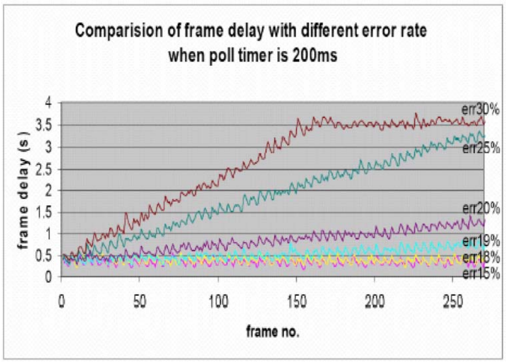

Fig.7: *The comparison of frame delay when using 200ms poll time at difference error rate*

some threshold, for example 19% as shwon in Fig.8. When the link has error rate goes higher than 19%, there are a lot of packets stay in RLC buffer waiting for retransmission. Each PDU must wait until all fragment packets are complete received without any error. When error rate increases, a number of retransmissions also increases. The background traffic plus retransmitted traffic may get higher than the link bandwidth.

Table 3: *The comparison of average frame delay*

Poll timer (ms)	Average frame delay (s)							
	Err5%	10%	15%	18%	19%	20%	25%	30%
50	0.257	0.299	0.345	0.408	1.282	2.214	2.959	3.292
200	0.280	0.343	0.412	0.504	1.440	1.440	3.065	3.302
500	0.291	0.460	0.485	0.613	1.516	1.516	3.102	3.331

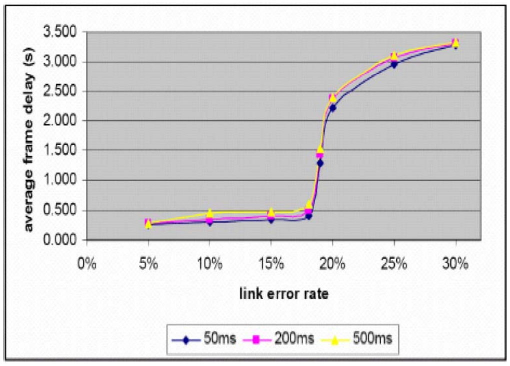

Fig.8: *Average frame delay vs link error rate when using difference poll timer*

For the other simulation results, the UE received frame rates are illustrated in Fig.9 to Fig.11. From Fig.9 and Fig.10, in case of no error occurred in the link, the UE received frame rate does not change event if the value of poll time is changed. For error 0 18% (in case no RLC discard mechanism only), the received frame rate values are not 10 frames per second all the times (sometime the received frame rate is 11 fps and the sometime is 9 fps) but the average received frame rate is 10 fps that is equal to the sending rate. As in the results, the poll timer value does not have the obviously effect that the smaller poll value can make a good received frame rate like in case of received frame delay. Because in our simulation model, the RLC module has not implemented the discard mechanism yet so the RLC tries to retransmit packet until it can transmit completely.

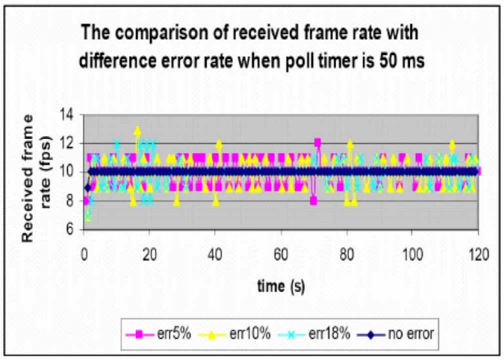

Fig.9: *The comparison of received frame rate when poll timer is 50 ms*

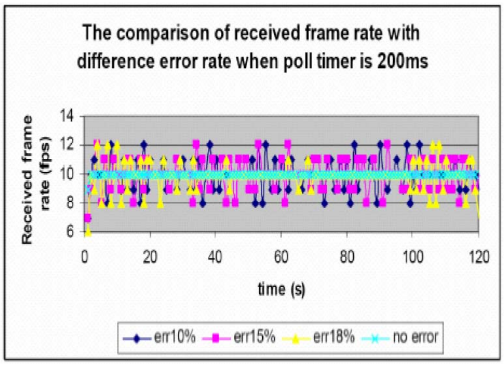

Fig.10: *The comparison of received frame rate when poll timer is 200 ms*

However when the error rate is higher than 18polling mechanism cannot recovery the packets efficiently, the average received frame rate less than 10 fps as shown in Fig.11. As a result it takes more time delay as shown in Fig.5. The error recovery in RLC, polling mechanism, is very useful for the applications that send data through the unreliable protocol such as UDP. UDP does not have the error recovery mechanism when we use UDP with RLC working in Acknowledged Mode (AM). RLC can recovery some data loss in data link layer by using polling mechanism. The values that we use for the poll timer based the frequency of the sending a request or information. For example, 50 ms poll timer means in every 50 ms, the receiver must send the status PDU to report the sender whether success or fail. Upon the received status, the sender will delete the packets that sent successful from the queue or retransmit them again in case of error. The speed for receiving status packet can help the sender to quickly decide what it should do, so it can reduce the number of drop packets (that stay in the buffer for a long time until their timer expire) and minimise buffer overflow also.

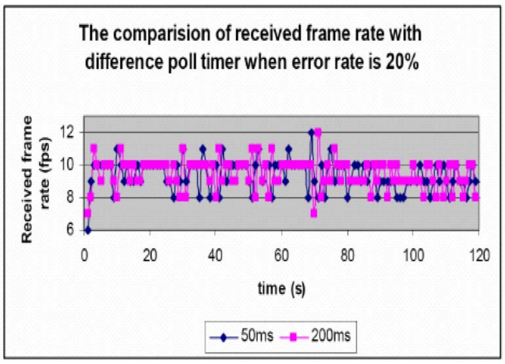

Fig.11: *The comparison of received frame rate when error rate is 20%*

More frequently polling means the receiver will send more status packets, so available bandwidth on the backward channel will be decreased. As in the simulation result, if we use 50 ms poll timer, the UE will send the PDU status with length of 40 bytes to the destination node every 50 ms. That is, in 1 second, the status PDUs needs 6.4 kbps of available bandwidth. However, for 150 ms poll timer, it needs 3 times lesser of the bandwidth. We need to take into account when available bandwidth is limited.

6. CONCLUSIONS

In this paper, we have studied the performance of RLC with MPEG-4 video stream services. Our simulation results have shown the impacts of RLC parameters configuration, Poll timer, on the performance of streaming MPEG-4 video over UMTS in terms of the video frame delay (after frame reconstruction) and the received video frame rate. We have found that

The small values of poll timer which send poll bit to the receiver frequently give better received video frame delay more than the bigger values. But it has the limitation of error rate, from the simulation result around 18%, which this parameter can have the effect. If we want to keep low delay for high error rate, only one parameter does not enough, we must try to use another parameter together to get better performance and also the better received frame delay or better received frame rate come with more using bandwidth in backward channel, therefore we must decide what the best value to configure poll timer is and it could solve the problem tread of between received frame rate and bandwidth.

Another founding is when error rate goes higher than some threshold, increasing poll frequency does not help, e.g. the average time delay. The background traffic plus retransmitted traffic may get higher than the link bandwidth.

The future works; we will show the suggestion values of poll timer that suitable in each kind of data and design the algorithm for choosing and controlling the poll timer in automatically according to the sending data and traffic resource.

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References

[1] Harri Holma and Antti Toskala, WCDMA for UMTS Radio Access for Third Generation Mobile Communications, John Wiley & Sons, Ltd., 2001.

[2] W. Stevens, TCP/IP Illustrated, vol 1, Reading MA: Addison Wesley, 1994.

[3] 3GPP TS 25.322 v5.5.0, Radio Link Control (RLC) Protocol Specification, http://www.3gpp.org, 2004.

[4] "ISO/IEC 13818," Part 1 to 10, 1996 to 2000.

[5] www.mpeg.org

[6] 3GPP TS 25.301 v5.5.0, Radio interface protocol architecture, http://www.3gpp.org, 2004.

[7] Qinqing Zhang and Hsuan-Jung Su, "Performance of UMTS radio link control," Proceedings of IEEE International Conference on Communications, volume 5, pages 3346-3350, 2002.

[8] The Network Simulator, NS-2, http://www.isi.edu/ nsnam/ns/

[9] http://www.ti-wmc.nl/eurane/

[10] Santichai Chuaywong, Sinchai Kamolphiwong and Thossaporn Kamolphiwong, "Adaptive Quality Control for Real-time MPEG-4 Video Communications," International Symposium on Communications and Information Technologies 2005(ISCIT2005), Oct 2005.

[11] 3GPP TS 34.108 v5.0.0, Common test environments for User Equipment (UE) conformance testing, http://www.3gpp.org, 2004.

[12] 3GPP TS 34.123-1 v5.5.0, User Equipment (UE) conformance specification Part 1: Protocol conformance specification, http://www.3gpp.org, 2004.

[13] Qinqing Zhang and Hsuan-Jung Su, "Performance of UMTS radio link control," Proceedings of IEEE International Conference on Communications, volume 5, pages 3346-3350, 2002.

[14] Xiao Xu, Hua Xu, Yi-Chiun Chen, Eren Gonen and Peijuan Liu, "Simulation analysis of RLC timers in UMTS systems," Proceedings of the

2002 winter simulation conference, vol. 1, pp. 506-512, 2002.

[15] Michele Rossi, Lorenzo Scaranari and Michele Zorzi," On the UMTS RLC Parameters Setting and their Impact on Higher Layers Performance," IEEE Proceeding VTC 2003, vol. 3, Oct 2003, pp. 1827 - 1832.

[16] Juan J. Alcaraz, Fernando Cerdan, Joan Garcia-Haro and Polytechnic University of Cartagena, "Optimizing TCP and RLC Interaction in the UMTS Radio Access Network," IEEE Network, vol. 20, issue 2, March/ April 2006.

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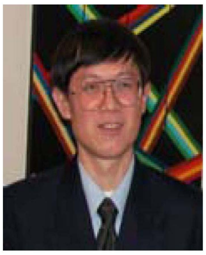

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