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ImprovingTCP Performance over 3G Links by Optimizing RLC Parameters

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***Abstract*— With rapid advances in wireless communications and internetworking, Universal Mobile Telecommunications Systems (UMTS) plays a main role in emerging 3G wireless networks, that have aimed to provision high speed data integrated with voice services. TCP, the dominant transport protocol for Internet applications, suffers severe performance degradation due to packet loss in the communication networks. Therefore, the 3G specification entity, 3GPP, has defined a reliable link layer protocol, RLC, to support packet switched services over UMTS. Since TCP will treat every packet loss as a sign of congestion, the schemes carried out at RLC layer reduces packet loss caused due to buffer overflow and the chances of mistaken invocation of the TCP Congestion Control Mechanism. Therefore, it is necessary to optimize the RLC protocol to upgrade TCP's performance. In this project two methods are carried out to improve the performance of TCP layer: ACK Rate Control Mechanism and ECN (Explicit Congestion Notification) to manage the buffer occupancy and avoid mistaken invocation of the TCP congestion control mechanism. Simulations on wireless networks have shown improved Packet Delivery Ratio (PDR) and minimized average end-to-end delay by using RLC parameters: number of nodes, packet length, speed of nodes, interval between the packets and rate at which the packets are sent over the network**

***Keywords*—emergency panic system, emergency call, war field secured information system, back up call system.**

I. Introduction

Internet applications are gaining momentum in the 3G mobile communication systems. Many important applications, such as file transfer, the Web, and Email rely on the dominant transport protocol of the Internet i.e., TCP (Transmission Control Protocol). TCP is a reliable, connection oriented, end-to-end protocol that controls the rate at which a source sends packets over the network, aiming to avoid congestion in the network. TCP was primarily designed for wired net-works, where packet loss is mainly caused due to packets being discarded on the congested network routers. Therefore, TCP congestion control mechanisms respond to packet loss by retransmissions and reductions in the rate of packet transmission at the source. However, packets are also lost due to reasons other than congestion like buffer overflow, high bit error rates (as high as 10 3) and handoff procedures that might result in degraded TCP performance.

The most efficient method to improve TCP performance is to use a re-liable link layer protocol: RLC (Radio Link Control) protocol; the option adopted by the 3rd Generation Partnership Project (3GPP) in its UMTS specifications.

TCP performance can be improved by configuring various parameters in the RLC protocol. Optimizing TCP performance by means of a suitable configuration of RLC parameters has the advantage of not requiring any modification at the end hosts and also it does not violate the TCP end-to-end semantics [14]. The main aim of this project is to achieve efficient recovery from packet loss and to obtain an optimized set of parameters in the RLC protocol like SDU size, PDU payload size, TTI (Transmission Time Interval), number of PDUs per TTI, data rate, block error rate, ACK delay control, transmission window size, receiving window size, buffer size, SDU delay of in-sequence delivery, maximum number of retransmissions, SDU discard function, poll window, poll timer etc.

The work carried out in this project is the simulation of the TCL code for a network with various parameters like number of nodes, speed of nodes, packet length, interval and rate to examine the number of packets generated, number of packets received in order to improvise Packet De-livery Ratio (PDR) and minimize average end-to-end delay using AODV routing protocol and generate graphical representation of the simulated results. The simulations are carried out by using network simulator NS-2.35 on the LINUX operating system.

Problem Statement

Buffer overflow in the RLC has been identified as one of the main threats to transport layer performance.

Inefficient buffer size degrades the TCP performance, since the packets that arrive at the RLC when buffer is full will be dropped. Leading to connections being timed out.

Due to packet loss in wireless link there is a possibility of mistaken invocation of the TCP congestion control mechanism, which again degrades the TCP performance.

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II. METHODOLOGY

A. Methodology

This section describes the proposed mechanisms that improve performance of the TCP protocol over 3G links.

i. ACK rate control algorithm

This algorithm is implemented at the RLC layer of the network side and consists of adjusting the inter-departure time of the acknowledgements (ACK) in the uplink direction, according to the congestion state of the downlink buffer. The underlying idea is to delay ACKs travelling through a node where its forward connection is congested. The ACK arrival rate determines the sending rate of new packets in a TCP source and the ACK departure rate is adjusted according to the downlink queue length. Since the throughput of a TCP source is basically determined by the arrival rate of ACKs, it is possible to reduce packet drops and avoid buffer overflow in the RLC layer if this rate is conveniently adapted to the link situation. The advantage of this mechanism is reduced computational cost and it makes more feasible for its implementation at the RLC level [6, 14].

Figure 1 illustrates the concept of this mechanism.

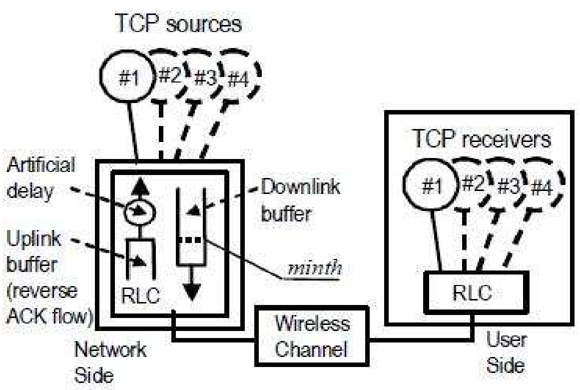

Figure 1: ACK rate control mechanism

ii. Explicit Congestion Notification (ECN)

This mechanism is another way to deal with congestion. In this mechanism, TCP sender can be informed in advance about a possible congestion event and take necessary steps to avoid the congestion and packet loss. ECN uses the two least significant (right-most) bits in the IP header to encode four different code points:

00: Non ECN-Capable Transport Non-ECT 10: ECN Capable Transport ECT(0)

01: ECN Capable Transport ECT(1) 11: Congestion Encountered CE

When both endpoints support ECN they mark their packets with ECT(0) or ECT(1). If the packet experiences congestion and the corresponding router support ECN, then it may change the code point to CE instead of dropping the packet. This act is referred to as marking and its purpose is to inform the receiving endpoint of impending congestion. At the receiving endpoint, this congestion indication is handled by the upper layer protocol (transport layer protocol) and needs to be echoed back to the transmitting node in order to signal it to reduce its transmission rate.

TCP supports ECN using two flags in the TCP header. Those two bits are used to echo back the congestion indication (i.e. inform the sender to re-duce the amount of data it sends) and to acknowledge that the congestion-indication echoing was received. These are the ECN-Echo (ECE) and Congestion Window Reduced (CWR) bits. When ECN has been negotiated on a TCP connection, the sender indicates that IP packets that carry TCP segments of that connection are carrying traffic from an ECN Capable Transport by marking them with an ECT code point. This allows intermediate routers that support ECN to mark those IP packets with the CE code point instead of dropping them in order to signal impending congestion.

Upon receiving an IP packet with the Congestion Experienced code point, the TCP receiver echoes back this congestion indication using the ECE flag in the TCP header. When an endpoint receives a TCP segment with the ECE bit it reduces its congestion window as for a packet drop. It then acknowledges the congestion indication by sending a segment with the CWR bit set. A node keeps transmitting TCP segments with the ECE bit set until it receives a segment with the CWR bit set [15].

III. IMPLEMENTATION DETAILS

Implementation is realization of the application or execution of the plan as per the design. It is a phase in the software development life cycle where the design is converted into code as a working model.

The computer system on which the simulations are carried out has the following hardware and software configurations:

Hardware Specifications:

Processor: Intel Core i5 CPU, 2.3 GHz Installed Memory (RAM): 3.0 GB

Software Specifications:

Operating system: Ubuntu 12.04 Network Simulator: NS 2.35 Programming Language: TCL (Tool Command Language)

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IV. SOFTWARE IMPLEMENTATION

A. Network Simulator

Features of NS-2 are:

Protocols: TCP, UDP, HTTP, Routing algorithms etc Traffic Models: CBR, VBR, FTP etc.
Error Models Uniform Periodic etc Radio propagation, Mobility models Topology Generation tools visualization tools Extensibility.

B. Need of two languages

NS-2 uses two languages, because it has two things to do. On one hand, detailed simulations of protocols requires a system programming language which can efficiently manipulate bytes, packet headers, and implement algorithms that run over large data sets. For these tasks, run-time speed is important. C++ is slow to change, but its speed makes it suitable for protocol implementation.

C. Running wired Simulations in NS

In this section, a Tcl script for ns is developed which simulates a simple topology. It describes how to set up nodes and links, how to send data from one node to another, how to monitor a queue and how to start nam from your simulation script to visualize your simulation. An example of a TCL code is given below, for a very simple topology with two nodes that are connected by a link.

D. Creating output files for X graph

This section describes how to create output files for x graph. One part of the ns all in one package is ’x graph’, a plotting program which can be used to create graphical representation of simulated results. A simple way is described below to create output files in Tcl scripts which can be used as data sets for x graph, and also see how traffic generators are used.

E. Running Wireless Simulations in NS

This section describes how to use the mobile wireless network simulation model in NS. Let us start to create and simulate a very simple 2-node wireless scenario. The topology consists of two mobile nodes, node(0) and node(1). The mobile nodes move about within an area whose boundary is defined in this example as 500mX500m. The nodes start to move initially at two opposite ends of the boundary. Then they move towards each other in the first half of the simulation and again move away for the second half. A TCP connection is setup between the two mobile nodes. Packets are exchanged between the nodes as they come within hearing range of one another. As they move away, packets start getting dropped. Just as with any other NS simulation, we begin by creating a tcl script for the wireless simulation.

V. SIMULATION RESULT

Assumptions are made to carry out the simulation of different wireless network scenarios: the interval between the packets is set to 5ms, the rate is set to 40 kbps, number of nodes are assumed to be increased in each case and the nodes in the network are assumed to be in motion.

The simulation results of a wireless network topology that makes use of AODV routing protocol and other parameters are tabulated below.

No. of Nodes	No. of Packets Generated	No. of Packets Received	Packet Delivery Ratio (%)	Average End-to-End Delay (ms)
5	824	819	99.39	76.59
10	477	466	97.69	71.47
25	903	895	99.11	71.21
50	907	899	99.11	51.43
100	1906	1887	99.00	48.09

Table 1: Simulation parameters for the network using ECN and TCPSink/DelSink

The tables below show the number of packets generated, number of packets received, Packet Delivery Ratio (PDR) and average end-to-end delay for the networks with parameters: number of nodes, speed, packet length, interval and rate.

In this case the number of nodes is varied to 5, 10, 25, 50 and 100, the packet length is set to 1000 bytes and the speed of node is assumed to be variable (0-9 m/s).

Parameters	Values
Simulator	NS-2(version 2.35)
Channel Type	Wireless Channel
Radio Propagation Model	Two ray ground wave
Network interface type	Phy/WirelessPhy
MAC type	Mac/802.11
Interface queue type	Queue/DropTail
Link layer type	LL
Antenna	Omni Antenna
Maximum packets in ifq	50
Area	1000X1000
Routing Protocol	AODV
ECN	true
Sink	TCPSink/DelAck

Table 2: Simulation result with Variable Speed (0-9 m/s), Packet Length of 1000 bytes, and Interval of 5ms and Rate of 40 kbps.

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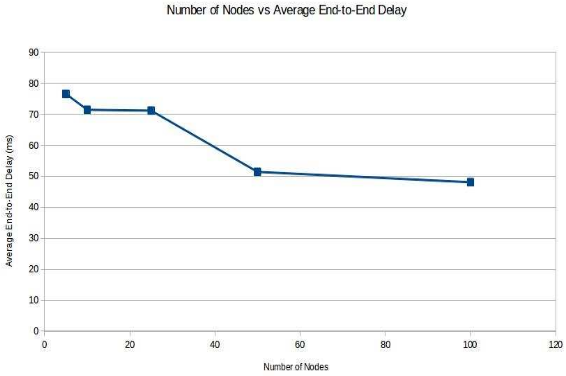

Figure 2: Graph of Number of Nodes in random motion vs. Average End-to-End Delay.

In the table 5.2 and the figure 5.1 shown above, it is observed that as the number of nodes is increased, the Average End-to-End Delay is de-creased. In second case, the speed of nodes in the network is set to 3m/s (10.8km/hr), the number of nodes is varied and in each scenario the packet length is varied to 200, 400, 600, and 800 bytes.

No. of nodes	Packet length	No. of packet generated	No. of packet received	Packet delivery ratio (%)	Avg. end-end delay (ms)
5	200	2966	2931	98.82	631.66
	400	1696	1693	99.82	347.31
	600	1303	1300	99.79	219.20
	800	993	990	99.69	169.81
10	200	2043	2029	99.26	162.49
	400	1029	1016	98.73	94.43
	600	782	770	98.46	77.80
	800	592	580	97.97	72.68
25	200	3451	3264	94.58	401.40
	400	1699	1694	99.70	46.58
	600	1506	1502	99.73	40.07
	800	1127	1125	99.82	36.32
50	200	3634	3511	96.61	386.83
	400	1709	1698	99.35	57.19
	600	1508	1504	99.73	40.86
	800	1127	1120	99.37	38.21
100	200	5785	5549	95.92	351.46
	400	3840	3674	95.67	126.58
	600	3094	3057	98.80	113.96
	800	2388	2358	98.74	61.38

Table 3: Simulation result with a Speed of 3.0 m/s (10.8 km/hr), Interval of 5ms and Rate of 40 kbps.

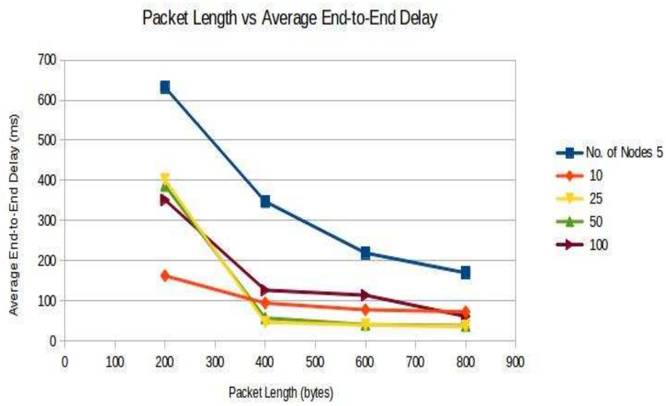

Figure 3: Graph of Packet Length vs. Average End-to-End Delay

In third case, the number of nodes and packet length are varied (similar to second case) however, the speed of nodes in the network is set to 7m/s (25.2km/hr).

No. of Nodes	Packet Length	No. of Packets Generated	No. of Packets Received	Packet Delivery Ratio (%)	Average End-to-End Delay (ms)
5	200	3445	3397	98.60	294.49
	400	1698	1693	99.70	133.13
	600	1357	1352	99.63	99.99
	800	1028	1023	99.51	82.94
10	200	2112	2096	99.24	157.53
	400	1062	1049	98.77	91.57
	600	812	800	98.52	75.54
	800	615	603	98.04	70.75
25	200	3458	3234	93.52	317.88
	400	1715	1704	99.35	50.10
	600	1510	1497	99.13	45.85
	800	1112	1099	98.83	43.22
50	200	3549	3454	97.32	375.86
	400	1708	1699	99.47	44.17
	600	1508	1504	99.73	38.05
	800	1131	1123	99.29	32.26
100	200	5037	4732	93.94	253.99
	400	3767	3563	94.58	158.21
	600	3015	2919	96.81	149.35
	800	2461	2356	98.12	116.51

Table 4: Simulation result with a Speed of 7.0 m/s (25.2 km/hr), Interval of 5ms and Rate of 40 kbps.

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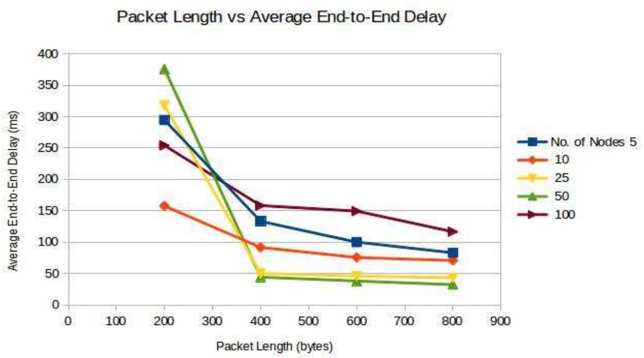

Figure 4: Graph of Packet Length vs. Average End-to-End Delay.

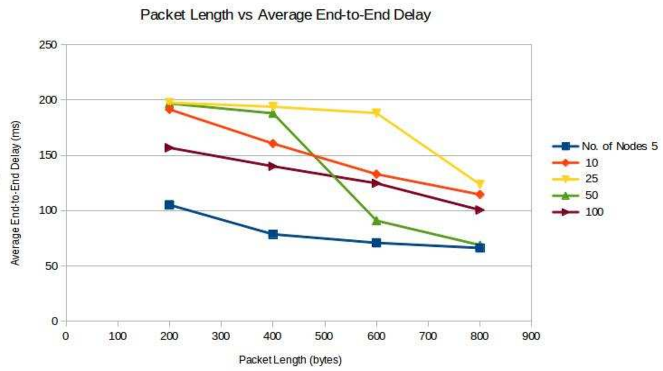

Figure 5: Graph of Packet Length vs. Average End-to-End Delay.

In fourth case, the number of nodes and packet length are varied and the speed of nodes in the network is set to 16.6m/s (60km/hr).

No. of Nodes	Packet Length	No. of Packets Generated	No. of Packets Received	Packet Delivery Ratio (%)	Average End-to-End Delay (ms)
5	200	3429	3406	99.32	105.24
	400	1706	1701	99.70	78.64
	600	1448	1444	99.72	70.89
	800	1094	1091	99.72	66.26
10	200	980	966	98.57	191.54
	400	497	486	97.78	160.65
	600	390	380	97.43	132.99
	800	297	287	96.63	114.58
25	200	2867	2655	92.60	197.61
	400	1821	1751	96.15	193.97
	600	1449	1415	97.65	188.21
	800	1115	1096	98.29	123.67
50	200	3362	3221	95.80	196.91
	400	1625	1562	96.12	188.02
	600	1481	1455	98.24	90.96
	800	1131	1112	98.32	68.85
100	200	3879	3601	92.83	156.85
	400	3665	3628	98.99	140.11
	600	2904	2770	95.38	124.74
	800	2286	2241	98.03	100.59

Table 5: Simulation result with a Speed of 16.66 m/s (60 km/hr), Interval of 5ms and Rate of 40 kbps.

In the tables and their corresponding graphs shown above, it is observed that as the number of nodes and packet length are increased, the average end-to-end delay in the network is decreased.

CONCLUSION

The objective of this project is achieved using network simulator NS 2.35 to analyze the performance of TCP layer in the wireless network by using the RLC parameters: number of nodes, speed of nodes in motion, packet size, interval and rate, AODV routing protocol, ECN mechanism and Delayed-ACK TCP sink.

The presented simulation result illustrates that if ECN mechanism and Delayed-ACK TCP sink is implemented in the wireless networks with optimized values of RLC parameters, the Packet Delivery Ratio (PDR) is improved and the average end-to-end delay is minimized as the number of nodes and packet length are increased in the network. Thereby improving the performance of the TCP layer.

The future work is to analyze the performance of different TCP types by using other parameters of RLC like: Block Error Rate (BLER), Maximum number of retransmissions, SDU delay of in-sequence delivery, SDU discard function, etc., along with routing protocols like Hybrid routing (both Proactive and Reactive), flow-oriented routing, Geographical routing protocols and so on.

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