Scheduling of Real-Time Traffic in Mobile Packet Switched Core Networks

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Summary
Interworking between UMTS (Universal Mobile Telecommunications System) and external networks is an important concept for fixed-mobile convergence as well as for next generation network. In order to achieve end-to-end QoS (Quality of Service) for broadband multimedia real-time services, DiffServ (Differentiated Services) mechanism should be used as QoS model and mapping of DiffServ Code Points (DSCP) into UMTS traffic classes should be applied to get efficient PHB (Per Hop Behaviour) configurations. According to the 3GPP specifications, standard QoS mapping between IP QoS classes and UMTS services authorizes both voice and video telephony to be mapped to the same QoS class. Since network resources are limited, if any segment of the network is congested, video traffic can cause significant delay of voice packets when aggregating both together to the same PHB group. In the context of QoS assurance is analyzed preposition of using LLQ (Low Latency Queueing) scheduler with the main idea of mapping voice and video telephony to different PHB groups. Within LLQ, PQ (Priority Queuing) is used for scheduling of both voice and video telephony. To proof the idea, simulation study was performed using NS-2 (Network Simulator Version 2) where changes were done within standard LLQ implementation in order to support scheduling of two queues with highest priority. New concept was compared with other schedulers like PQ (Priority Queuing), WFQ (Weighted Fair Queuing) and WRR (Weighted Round Robin). The obtained results were statistically processed using SPSS (Statistical Package for the Social Sciences) version 17.0. Summarizing research results, avenues of future research are identified.

Key words:
Diffserv, QoS, Scheduling, Mapping, LLQ

1. Introduction

The growing popularity and availability of Internet and IP (Internet Protocol) based transport networks, as well as introducing IMS [1] (IP Multimedia Subsystem) within core network of UMTS (Universal Mobile Telecommunication System), provides support for multimedia services like VoIP (Voice over IP) or Video Streaming. These and other innovative broadband multimedia services have strict requirements for QoS (Quality of Service) network parameters like delay, jitter and packet loss. Even if using IP transport reduces costs significantly, it provides only best effort services, without any guarantee in terms of QoS. In order to transport UMTS services through IP networks without losing end-to-end QoS provisioning, it is important to define QoS mapping scheme between UMTS services and IP QoS classes, as well as to use appropriate QoS mechanism on network nodes.

This paper will focus on the analysis of ensuring QoS for UMTS real-time traffic (Conversational and Streaming traffic classes) in interworking network scenario between UMTS and an external IP network based on DiffServ (Differentiated Services) QoS mechanism. DiffServ architecture [2] is based on a simple model where traffic entering a network is classified and conditioned at the boundaries of the network according to the DSCP (Differentatied Services Code Point) [3] field in IP header, and assigned to different behavior aggregates. Within the core of the network, packets are forwarded according to the Per-Hop Behaviour (PHB) group associated with the DSCP. PHB are implemented using some buffer management and packet scheduling mechanisms. The choice of a traffic scheduling algorithm is important for the implementation of behavior aggregates in a DiffServ based network. PHB definitions do not specify any particular implementation mechanism and therefore the problem of PHB implementation has recently gained significant attention. According to standard 3GPP specifications [4], mapping between UMTS traffic classes and PHB groups can be done at gateway GPRS support node (GGSN). Standard QoS mapping authorizes both voice and video telephony, which belong to Conversational UMTS traffic class, to be mapped to the same QoS class. Bursty video traffic has larger packet sizes than voice traffic and can cause significant delay of voice packets when aggregating both together to the same QoS class in case of high network overload.

In existing related work, authors of paper [5] have analyzed refined mapping between voice and video telephony but do not take other UMTS traffic classes into account. They found out that refined mapping using WFQ (Weighted Fair Queueing) scheduler performs better than

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FIFO (First In First Out) and PQ (Priority Queuing) schedulers. In a case study giving the best expected performances, authors found out that WFQ must be configured with a video weight three times higher than voice weight in order to have optimal delay for multimedia telephony traffic.

On the other side, in paper [6] are discussed QoS aspects both for real-time and non real-time traffic types in UMTS simulation environment, but only in case of PQ and WRR (Weighted Round Robin) schedulers. It has been shown that WRR is able to keep within the ITU recommendations for a network overload of 43%, whereas the PQ only meets the ITU recommendations for an overload of 13%.

Dekeris et al., [7] combine WFQ and LLQ (Low Latency Queueing) schedulers, using LLQ for video traffic and WFQ with highest priority for voice traffic. The main drawback of this idea is the property that delay of high priority class (Video conferencing) could be reduced, but at the same time Voice traffic got the highest delay time.

Our former research has shown that LLQ scheduler is able to meet ITU-T QoS requirements for real-time traffic and at the same time provides fair service for other traffic classes even in case of highly congested network link [8]. In our previous work [9], we introduced changed version of LLQ in Network simulator version 2 (NS-2) where PQ was used for scheduling of both voice and video telephony with respect to other traffic classes. This paper extends our research findings and presents novel approach of mapping voice and video telephony to different EF PHB values with implementation of LLQ traffic scheduler on network nodes, where voice and video telephony traffic flows are scheduled with highest priority. Performances of novel mapping approach with LLQ scheduler are compared with other most available traffic scheduling algorithms such as WFQ, WRR and PQ.

This paper is organized as follows. Section II makes a brief presentation of end-to-end QoS conceptual model and our suggestion of mapping between UMTS and DiffServ domain with changed version of LLQ scheduling mechanism. Section III presents NS-2 simulation model and simulation results, together with their discussion and analysis to show the conclusions that are warranted. Section IV concludes this paper and describes direction for the future work.

2. Novel approach to UMTS-to-IP QoS mapping and traffic scheduling

3GPP standard [10] proposes a layered architecture for the support of end-to-end QoS. To realize a certain network QoS, a Bearer Service (BS) with clearly defined functionalities has to be set up from the source to the destination of a service and includes all aspects to enable the provision of a contracted QoS. UMTS BS attributes form a QoS profile and define the grade of service provided by the UMTS network to the user of the UMTS bearer service. UMTS specification [11] defines four traffic classes and they are: Conversational, Streaming, Interactive and Background. The main difference between these classes is how delay sensitive the traffic is. Applications of Conversational and Streaming classes are the most delay sensitive and intended for real-time traffic, while applications of Interactive and Background classes require higher reliability. Examples of applications are voice and video telephony for Conversational class and Video Streaming for the Streaming class. Interactive class is used by interactive applications like interactive web browsing, while Background class can be used for background download of e-mails.

Since the UMTS packet switched core network is based on an IP, DiffServ can be used for QoS provisioning. Figure 1 shows the example of how end-to-end QoS may be accomplished for a significant number of scenarios. In this paper, first scenario from 3GPP specification [10] was chosen, where the GGSN supports DiffServ Edge function and the IP network is DiffServ enabled. The application layer identifies QoS requirements, which are mapped into (PDP) Packet Data Protocol context parameters in UE (User Equipment). Local mechanism in the UE uses the PDP context for QoS over the UMTS access network, and the IP backbone network uses DiffServ to provide QoS guarantees. According to [10] IP BS manager is located in GGSN and uses standard IP mechanisms to manage IP bearer service. Provision of IP BS manager is optional in UE and mandatory in the GGSN. Where the resources for the IP Bearer Service to be managed are not owned by the UMTS network, the resource management of those resources would be performed through an interaction between the UMTS network and that external network. In addition, where the UMTS network is also using external IP network resources as part of the UMTS bearer service (e.g. for the backbone bearer service), it may also be necessary to interwork with that network.

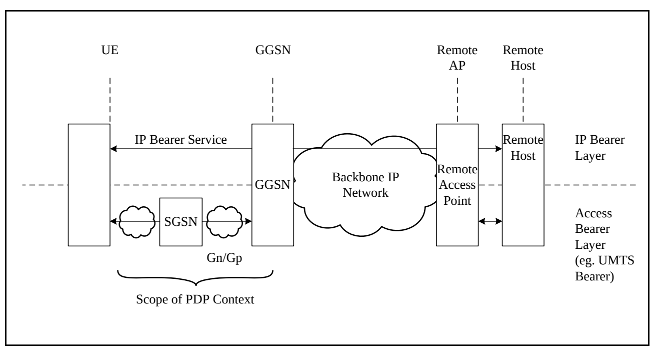

Fig. 1 Network architecture for QoS conceptual model [10]

Translation/Mapping function in GGSN provides interworking between the mechanisms and parameters used within the UMTS bearer service and those used within the IP bearer service. Mapping between UMTS QoS classes and DiffServ PHB groups is controlled by individual operator. PHB simply characterizes the externally observable forwarding behavior of a DiffServ router to the corresponding traffic stream. The DiffServ working group of IETF has defined different PHB groups for different applications. To instantiate a particular PHB, network administrator activates and tunes an appropriate combination of specific packet-scheduling algorithms and AQM (Active Queue Management) mechanisms supported by the DiffServ router. Performances of traffic scheduling mechanism have the highest impact on the level of service a packet receives. When multiple queues are sharing common transmission media, there must be a scheduler to decide how to pick up packets from each queue to send out and is responsible for enforcing resource allocation to individual flows. If there is no congestion on the interface, packets are transmitted as they arrive. If the interface is experiencing congestion, scheduling algorithms are engaged.

3GPP recognizes that it is operator's choice to define the mechanisms for the provisioning of resources among the different DiffServ PHB classes, as well as the mapping from the UMTS QoS classes, to the PHB groups. While 3GPP TS specifies four UMTS traffic classes, ITU-T recommendation Y.1541 [12] specifies six IP network QoS classes. These two sets of QoS classes have similarities, but there is no unique mapping between them. Contribution [13] considers one possible means of achieving QoS interoperability between 3GPP based wireless networks and ITU-T based wireline IP networks, by mapping between the Y.1541 QoS classes and a corresponding set of values for 3GPP-defined "bearer service attributes". According to this contribution, Y.1541 classes 0 and 1 correspond generally with the 3GPP Conversational and Streaming traffic classes, respectively. Y.1541 classes 2-4 correspond generally with the 3GPP Interactive class, while Y.1541 class 5 corresponds closely with 3GPP Background class. According to Y.1541 and 3GPP recommendations, the following target values should be guaranteed for Conversational traffic class:

- IP Transfer Delay, IPTD $\leq$ 100 ms,
- IP Delay Variation, IPDV $\leq$ 50 ms,
- IP Loss Rate, IPLR $\leq 1\times10^{-3}$,

while for Streaming traffic class target values that should be guaranteed are as follows:

- IP Transfer Delay, IPTD $\leq$ 400 ms,
- IP Delay Variation, IPDV $\leq$ 50 ms,
- IP Loss Rate, IPLR $\leq 1\times10^{-3}$.

Mapping between UMTS traffic classes and PHB groups suggested in this paper is depicted in Table 1. Traditionally, all traffic in Conversational class, i.e. voice and video telephony, should be mapped to the same EF PHB class which is intended for critical voice traffic. The EF PHB [14] is intended to support low-loss, low-delay and low-jitter services. The EF guarantees that traffic is serviced at a rate that is at least equal to a configurable minimum service rate (regardless of the offered load and non-EF traffic) at both long and short intervals. To preserve the low-latency queuing behavior, out-of-contract Conversational class is defined to be dropped. Voice packets have short and constant packet size while video packets have large and variable packet size. When injecting voice telephony traffic together with bursty video telephony traffic, video traffic can cause degradation as well as delay of voice service. Therefore, in our novel approach we suggest mapping of voice and video telephony into two different DiffServ virtual queues and DiffServ EF PHB groups. RFC 5865 [15] requests one DSCP value from IANA (Internet Assigned Numbers Authority) authority for a class of real-time traffic. This class conforms to the EF PHB group. It is also admitted using a CAC (Call Admission Control) procedure involving authentication, authorization, and capacity admission. This differs from a real-time traffic class that conforms to the Expedited Forwarding PHB but is not subject to capacity admission. The requested DSCP applies to the Telephony Service Class described in RFC 4594 [16]. IANA assigned a DSCP value to a second EF traffic class which is referred to VOICE-ADMIT. The value is parallel with the existing EF code point 101110 (46 decimal), as IANA assigned the code point 101100 (44 decimal), with first binary values the same in both. In this paper we have suggested to map voice telephony into decimal EF 46 value, and video telephony into decimal EF 44 value.

Aligned with GSM Association PRD IR.34 [17], Streaming traffic class is marked as Assured Forwarding (AF4x). AF PHB [18] allows the operator to provide assurance of delivery as long as the traffic does not exceed some subscribed rate. Traffic that exceeds the subscription rate faces a higher probability of being dropped if congestion occurs. AF PHB defines four independent PHB classes, each with three dropping precedence level.

Table 1: QoS mapping table

Traffic class	PHB group	DSCP
Conversational voice	EF	46
Conversational video	EF VOICE-ADMIT	44
Streaming video	AF41/AF42	34/36
Interactive	AF31/AF32	26/28
Background	BE	0

Each corresponding PHB is known as AF_{ij}, where i represents AF class, while j is the drop precedence. Within an AF class, packets of drop precedence p experience a level of loss lower (or equal to) than the level of loss experienced by packets of drop precedence q if $p<q$. Each AF class is configured with separate buffer and bandwidth.

Video streaming traffic because of its usually limited burst behavior and large packet sizes is more problematic to manage than Conversational voice traffic. It is recommended to use TSW2CM/TSW3CM (Time Sliding Window Two/Three Color Marker) [19] classification for Streaming traffic which is able to meter in-contract as well as out-of-contract traffic. In this paper we suggested using TSW2CM marker. If data rate is below CIR (traffic is in-contract), Streaming traffic is marked as AF41 PHB (34 decimal), while if data rate exceeds CIR value (traffic is out-of-contract), Streaming traffic is marked as AF42 PHB (36 decimal).

Interactive traffic class can use any of the remaining Assured Forwarding classifications. According to GSM association PRD IR.34 [17], the difference between AF3x, AF2x and AF1x traffic classes is the priority of the queuing behavior and the maximum buffer size, which is defined to accommodate the high delay possibilities for the lower-priority traffic. Suggestion made in this paper is to map Interactive traffic class into AF3x group since it doesn't have any special QoS requirements apart from reliability. Same as for Streaming traffic class TSW2CM was used as marker. If data rate is below CIR (traffic is in-contract), Interactive traffic is marked as AF31 PHB (26 decimal), while if data rate exceeds CIR value (traffic is out-of-contract), Interactive traffic is marked as AF32 PHB (28 decimal). Aggregate of Background class is mapped to default BE PHB which has the best effort forwarding characteristics.

Our novel approach for end-to-end QoS assurance proposes using LLQ scheduler on network elements with the main idea of mapping voice and video telephony to different PHB groups. LLQ [20] is a combination of PQ and Class-Based Weighted-Fair Queuing (CBWFQ) schedulers. CBWFQ extends the standard WFQ functionality to provide support for user-defined traffic classes. LLQ, like PQ checks low-latency queue first and takes a packet from that queue. If there are no packets in the low-latency queue, the normal scheduler logic applies to the other non-low-latency queues, giving them their guaranteed bandwidth. LLQ allows delay-sensitive applications to be given preferential treatment over all other traffic classes.

Standard LLQ implementation schedules packets only from one queue with highest priority. In order to provide support for two priority queues (for voice and video telephony), standard implementation of LLQ within NS-2 simulator was changed. Changes were done in LLQ scheduler part within dsScheduler.cc script of NS-2 code. The main objective of novel approach in this paper is to provide IP Transfer Delay (IPTD), IP Delay Variation (IPDV) and IP Loss Rate (IPLR) for Conversational and Streaming traffic classes to be below standard defined values, even in case of high network overload, but not to completely exhaust bandwidth for other traffic classes.

3. Simulation model and results

Simulation is performed using network simulator NS-2, which is an event-driven simulator targeted at networking research [21] and independently developed *DiffServ4NS* module for scheduling algorithms used in this paper [22]. The aim of simulation was to evaluate performances of our proposed idea in a mixed network environment, composed of the UMTS core network and external IP network in terms of QoS parameters and perform comparison with other schedulers such as WFQ, WRR and PQ where voice and video telephony are mapped to the same virtual queue and traffic class.

3.1 Simulation Methodology

Simulation model is presented in Figure 2. and consists of five routers named SGSN, GGSN, CORE1, CORE2 and EDGE, five source nodes and five destination nodes. UMTS infrastructure is not fully simulated (radio interface), only the core network between SGSN (Serving GPRS Support Node) and GGSN nodes. This is not inconsistent with the concept of UMTS architecture which specifies that access and core networks are independent [23]. We only simulate the packet-based core network as an IP based network that implements DiffServ QoS mechanism, which is only core network bearer service between SGSN and GGSN. SGSN and GGSN are simulated through two regular routers that implement traffic conditioning, packet scheduling and the appropriate queue management techniques. Link between SGSN and GGSN nodes is not congested. The intention of simulation was to present the impact on the performance of the UMTS traffic when passing through the external IP backbone network in congested situations.

Two experiments are conducted, which differ from each other according to localization of congestion. In experiment 1 congestion occurs due to only one bottleneck link overload between nodes CORE1 and CORE2 in external IP backbone network, while in experiment 2 congestion occurs due to whole external IP backbone network overload. Each experiment includes four different simulation scenarios which differ from each other according to the scheduling algorithm implemented on network nodes. The capacities of links in external IP backbone network are dimensioned in a way to implement network configuration whose load equals 20%. Starting from this configuration, we decreased the capacity of appropriate links gradually to 200% according to the amount of traffic passing through external IP backbone, which is 6Mbps and is constant during all simulation.

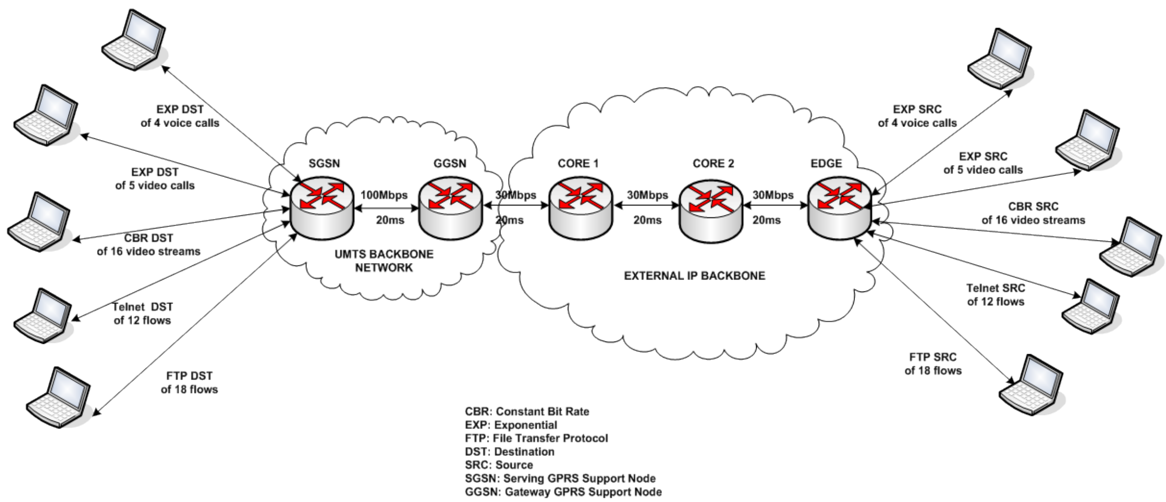

Fig. 2 Simulation model

GGSN and EDGE nodes implement DiffServ edge function. Therefore links between GGSN and CORE1 nodes as well as between EDGE and CORE2 nodes are implemented as a simplex links with dsRED/edge functionality of NS-2. On the other side, link between CORE1 and CORE2 is duplex with dsRED/core functionality of NS-2. GGSN and EDGE nodes perform mapping between UMTS QoS traffic classes and PHB groups.

Voice telephony traffic sources are represented with Exponential (EXP) traffic generator, which generates 4 flows with packet size of 80 bytes and 100 kbps rate, while video telephony sources generate 5 flows with 1000 bytes and 100 kbps rate. Constant Bit Rate (CBR) generator is used in order to generate 16 video traffic flows with packet size of 1000 bytes and 300 kbps rate. Interactive traffic sources are simulated with Telnet application, and generate 12 flows with 500 bytes packets size with 10 kbps rate. Background data sources are configured with File Transfer Protocol (FTP) traffic generator with 1000 bytes packet length and 10 kbps rate from 18 flows. CBR and EXP traffic generators are attached to UDP agents, while FTP and Telnet traffic generators are attached to TCP agents. Conversational traffic (voice and video telephony together) produces 15%, Streaming traffic produces 80%, while Interactive and Background traffic produce 2% and 3% of overall generated traffic.

TSW2CM is used as a policer to determine how to mark and prioritize the packet according to user requirements. Weighted Random Early Detection (WRED) is used as Active Queue Management mechanism for Streaming and Interactive traffic, while for Conversational and Background traffic is used Drop Tail. The configured CIR value equals 2 Mbps for EXP flows, 6 Mbps for CBR flows and 1 Mbps for Interactive and Background traffic flows. In case that PQ scheduler is used, Conversational traffic has the highest priority, while Background traffic has the lowest priority. Weights of other schedulers are configured in such a way, that weight represents percentage of output port bandwidth: 15 for Conversational, 80 for Streaming, 2 for Interactive and 3 for Background traffic class. The queue lengths are constant and are defined with 30 packets for Conversational class and 50 packets for all other traffic classes.

3.2 Simulation Results and Discussion

Simulation results in this paper are depicted only for second micro flow generated from all traffic sources, which is chosen randomly, but could be for any of the generated flows. Table 2 depicts the average values of the link throughput between nodes CORE1 and CORE2 in Experiment 1, as well as the average values of the link throughput between EDGE and CORE 2 nodes in Experiment 2, in both cases for Interactive and Background traffic classes. The network/link load percentage value above which the critical performance metrics for Conversational and Streaming traffic classes exceed the target values is determined in both simulation experiments and shown in table 3.

Table 2: QoS mapping table

		WFQ	WRR	PQ	LLQ
Exp. 1	Interactive class throughput [kbps]	8.69	7.34	4.87	8.59
	Background class throughput [kbps]	36.51	29.52	30.48	33.31
Exp. 2	Interactive class throughput [kbps]	8.93	7.79	4.62	8.72
	Background class throughput [kbps]	37.19	33.05	27.39	35.41

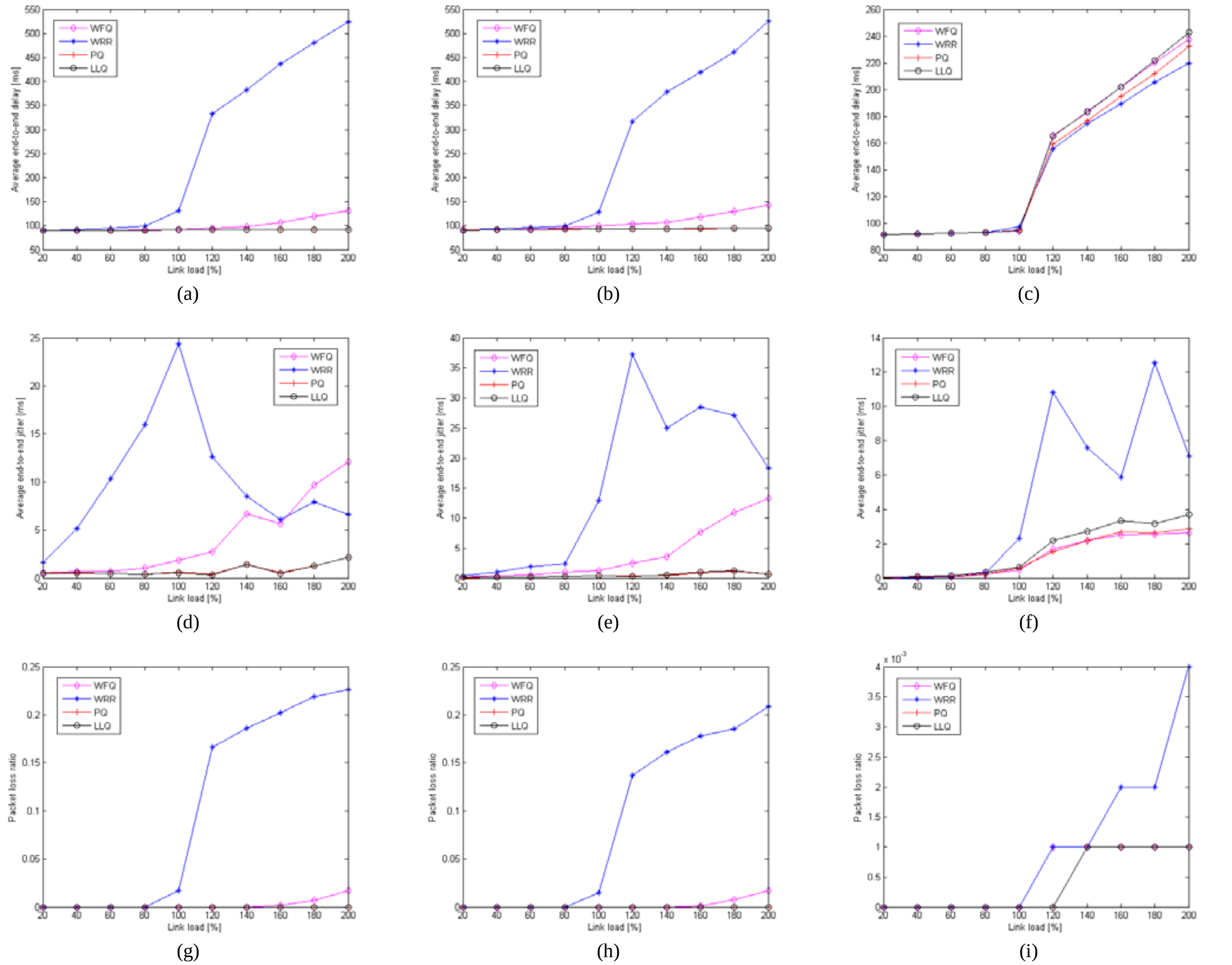

Fig. 3 Simulation results for Exp 1: (a) IPTD of voice telephony; (b) IPTD of video telephony; (c) IPTD of video streaming; (d) IPDV of voice telephony; (e) IPDV of video telephony; (f) IPDV of video streaming; (g) IPLR of voice telephony; (h) IPLR of video telephony; (i) IPLR of video streaming

Table 3: Network/link load percentage above which performance metrics exceed target QoS values

Experiment/Scenario		**Scenario 1 WFQ**			**Scenario 2 WRR**			**Scenario 3 PQ**			**Scenario 4 LLQ**		
		IPTD	IPDV	IPLR	IPTD	IPDV	IPLR	IPTD	IPDV	IPLR	IPTD	IPDV	IPLR
Exp. 1	Voice tel.	160	-	160	100	-	100	-	-	-	-	-	-
	Video tel.	120	-	180	100	-	100	-	-	-	-	-	-
	Video stream.	-	-	-	-	-	160	-	-	-	-	-	-
Exp. 2	Voice tel.	140	-	160	100	-	100	-	-	-	-	-	-
	Video tel.	120	-	160	80	-	100	-	-	-	-	-	-
	Video stream.	-	-	-	-	-	200	-	-	-	-	-	-

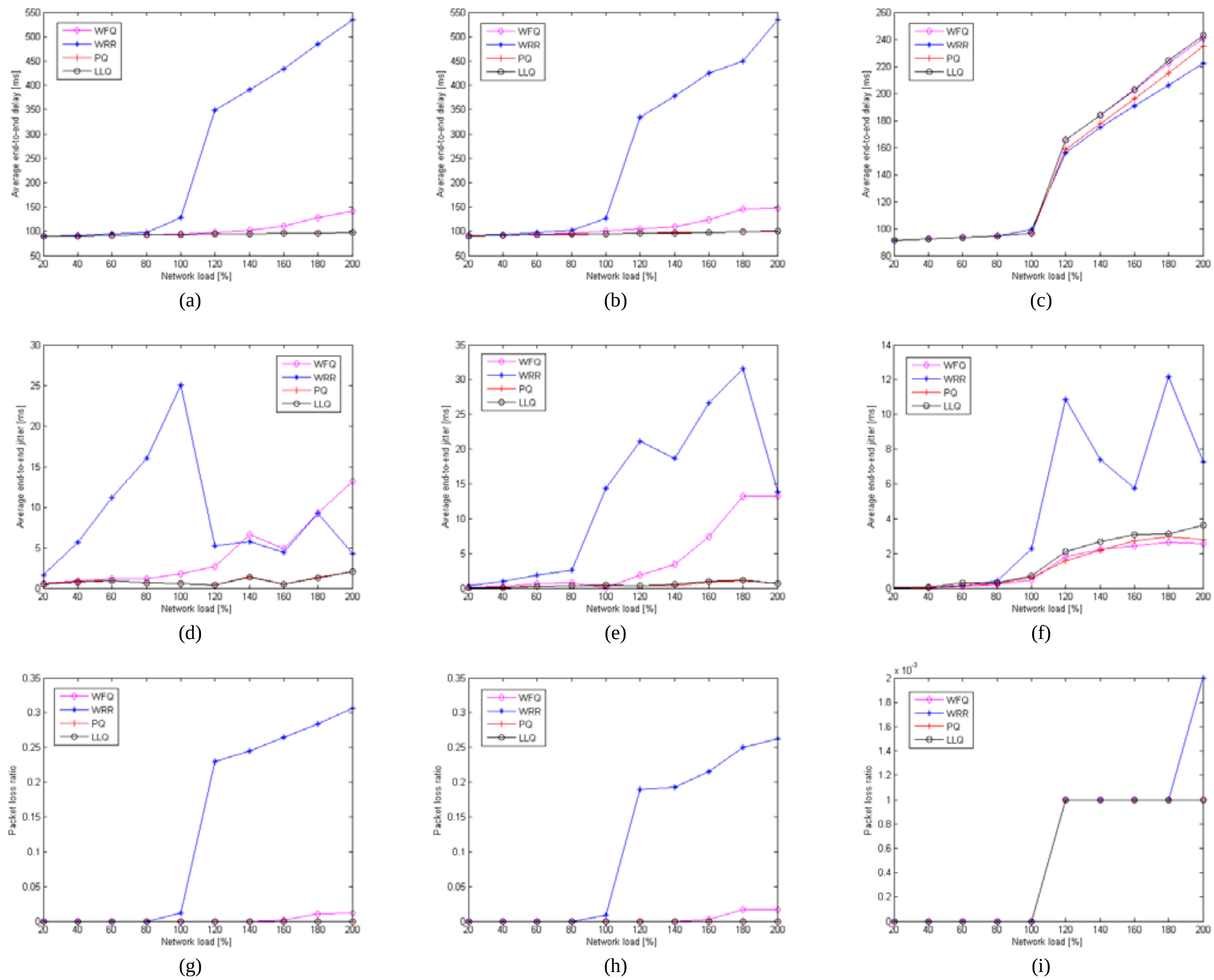

Fig. 4 Simulation results for Exp 2: (a) IPTD of voice telephony; (b) IPTD of video telephony; (c) IPTD of video streaming; (d) IPDV of voice telephony; (e) IPDV of video telephony; (f) IPDV of video streaming; (g) IPLR of voice telephony; (h) IPLR of video telephony; (i) IPLR of video streaming

Table 4: Statistical analysis of QoS parameters for experiment 1 – LLQ scheduler

	Regression model	b_0	b_1	b_2	b_3	r^2	r	p
Voice telephony IPTD	$y=b_0+b_1t+b_2t^2+b_3t^3$	90.015	0.010	-1.135×10^{-5}	8.790×10^{-9}	0.979	0.989	>0.05
Video telephony IPTD	$y=b_0+b_1t+b_2t^2+b_3t^3$	89.172	0.087	0	2.186×10^{-6}	0.675	0.821	>0.05
Video streaming IPTD	$y=b_0+b_1t+b_2t^2+b_3t^3$	130.355	-2.186	0.028	-7.108×10^{-5}	0.963	0.981	>0.05
Voice telephony IPDV	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.284	0.011	0	7.558×10^{-7}	0.738	0.859	>0.05
Video telephony IPDV	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.562	-0.019	0	-7.241×10^{-7}	0.773	0.879	>0.05
Video streaming IPDV	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.951	-0.054	0.001	-2.237×10^{-6}	0.969	0.984	>0.05
Voice telephony IPLR	-							-
Video telephony IPLR	-							-
Video streaming IPLR	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.048	-0.003	4.113×10^{-5}	-1.231×10^{-7}	0.782	0.884	>0.05

Table 5: Statistical analysis of QoS parameters for experiment 2 – LLQ scheduler

	Regression model	b_0	b_1	b_2	b_3	r^2	r	p
Voice telephony IPTD	$y=b_0+b_1t+b_2t^2+b_3t^3$	89.550	0.026	0	-5.725×10^{-7}	0.998	0.999	>0.05
Video telephony IPTD	$y=b_0+b_1t+b_2t^2+b_3t^3$	89.500	0.066	0	4.807×10^{-7}	0.999	1	>0.05
Video streaming IPTD	$y=b_0+b_1t+b_2t^2+b_3t^3$	129.624	-2.141	0.028	-7.060×10^{-5}	0.966	0.983	>0.05
Voice telephony IPDV	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.045	0.030	0	1.355×10^{-6}	0.693	0.832	>0.05
Video telephony IPDV	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.253	-0.010	0	-5.242×10^{-7}	0.781	0.884	>0.05
Video streaming IPDV	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.763	-0.044	0.001	-1.895×10^{-6}	0.973	0.987	>0.05
Voice telephony IPLR	-							-
Video telephony IPLR	-							-
Video streaming IPLR	$y=b_0+b_1t+b_2t^2+b_3t^3$	0.057	-0.003	4.383×10^{-5}	-1.271×10^{-7}	0.899	0.948	>0.05

The output trace file from each simulation scenario in experiments 1 and 2 is used to measure average IPTD, IPDV and IPLR values obtained for Conversational (voice and video telephony) and Streaming traffic classes. As we can see from Figures 3a and 3b, as well as from Figures 4a and 4b, for voice and video telephony respectively, average end-to-end delay stays within 100 ms only when PQ and LLQ schedulers are implemented on network nodes. In Experiment 1, the average IPTD exceeds the target value when the bottleneck link load between nodes CORE1 and CORE2 is above 160% in Scenario 1 and above 100% in Scenario 2 for voice telephony. For video telephony IPTD exceeds the target value when the bottleneck link load is above 120% in Scenario 1, and above 100% in scenario 2. In Experiment 2, the average IPTD exceeds the target value when the network load is above 140% in Scenario 1 and above 100% in Scenario 2 for voice telephony. For video telephony IPTD exceeds the target value when the network load is above 120% in Scenario 1, and above 80% in scenario 2.

Considering the effect of different schedulers on average end-to-end delay for Streaming traffic, which is depicted in Figures 3c and 4c, we can notice that all schedulers have almost the same performances and provide satisfactory level of QoS. The average IPTD does not exceed 400 ms regardless of the network/bottleneck link load.

Results from Figures 3d and 3e, as well as from Figures 4d and 4e show that jitter stays below 50 ms for voice and video telephony in all experiments regardless of scheduler type implemented on network nodes. The same behavior is also observed in Figures 3f and 4f, for Streaming traffic. Results for jitter show non-monotonic behavior: increasing with the network load, reaching some maximum and then decreasing. More network latency is necessary in order to deliver a stream due to network congestion.

In Figures 3g and 3h for first experiment, and Figures 4g and 4h for second experiment, packet loss rate for voice and video telephony is presented. Network congestion has the greatest influence on WRR scheduler, which does not perform satisfactory when network/bottleneck link is overloaded more than 100 %. In Experiment 1, the average IPLR exceeds the target value when the bottleneck link load is above 160% in Scenario 1 and above 100% in Scenario 2 for voice telephony.

For video telephony IPLR exceeds the target value when the bottleneck link load is above 180% in Scenario 1, and above 100% in scenario 2. In Experiment 2, the average IPLR exceeds the target value for voice and video telephony when the network load is above 160% in Scenario 1 and above 100% in Scenario 2 for voice telephony. On the other side, there is no packet loss in case of PQ and LLQ schedulers for Conversational and Streaming traffic classes no matter how high is overload of bottleneck link or whole external network.

In Figures 3i and 4i, packet loss rate for Streaming traffic is presented; we noticed that it remains below standard defined values for all schedulers except WRR. WRR exceeds target value in Experiment 1 when bottleneck link load is above 160% and in Experiment 2 when the network load is above 200 %. This is presented in Table 3.

As we can see from Table 2, PQ scheduler has the lowest throughput values for Interactive and Background classes in both experiments. When PQ scheduling algorithm is used, lower priority classes are starving and throughput is equal zero for both Interactive and Background traffic classes when network load is higher than 100%. LLQ on the other side, provides fair level of bandwidth for lower priority classes, and at the same time fulfills QoS requirements for Conversational and Streaming traffic classes.

From all these experiments, it can be summarized that only novel approach of mapping, together with LLQ scheduler keeps critical performance parameters for voice and video telephony as well as for Streaming traffic class below standard defined values and at the same time provides fair amount of bandwidth for Interactive and Background traffic classes.

The obtained results are statistically processed using Statistical Package for the Social Sciences (SPSS) version 17.0. The null hypothesis states that QoS parameters can be guaranteed with novel approach. Against the null hypothesis is setup the alternative hypothesis. The 95% confidence interval is chosen, which relates to level of statistical significance of $p<0.05$. The regression analysis is conducted to find the relationship that explains how the variation in IPTD/IPDV/IPLR values for Conversational and Streaming traffic classes, depends on the variation in network overload. Coefficient of correlation (r) is measured to give the true direction the correlation, while the coefficient of determination (r^2) is measured to give the strength of correlation.

Using comparative analysis of different regression models in SPSS, we have decided to use cubic polynomial regression model. The results of regression analysis have been summarized in Table 4 (Experiment 1) and Table 5 (Experiment 2) only for our approach of suggested mapping and using LLQ scheduler, since it has the best performances among all other scheduling mechanisms.

From Tables 4 and 5, it can be noticed that regression dependency between packet loss and link load percentage for Conversational traffic class is not possible to determine. Value of packet loss rate is constant and is always zero for Conversational traffic class no matter how high is traffic overload in external IP backbone network. This result is consequence of handling Conversational traffic with strict priority above all other traffic classes. For all other results depicted in Tables 4 and 5, *p* value (significance) is greater than 0.05, which means that there is almost no statistical relationship between QoS parameters and link load percentage when novel approach is used.

4. Conclusion and Future Work

As today's wireless and wireline networks converge in an IP-based multi-service next generation network (NGN), QoS interworking between the wireless and wireline technologies supporting end-to-end applications will become essential. Motivation behind this research was to meet end-to-end QoS requirements for real-time traffic flows, regardless of network conditions.

In order to satisfy QoS end-to-end concept, it is crucial to define efficient QoS mapping scheme between UMTS services and IP QoS classes in case of DiffServ based networks. This paper presented one example of mapping which was implemented on GGSN and EDGE nodes, as they perform DiffServ edge function in our simulation model. Our novel approach suggested mapping of voice and video telephony into two different EF PHB groups with different DSCP values. Other important problem which was pointed out in our work concerns the implementation of PHB and the choice of traffic scheduling algorithm. We proposed the idea of using LLQ scheduler, with PQ scheduling for both voice and video telephony over all other traffic classes. Default implementation of LLQ scheduler in NS-2 was changed in order to support PQ scheduling of two virtual queues instead of one.

The results obtained from our simulation study indicate that using LLQ provides better performances than using WFQ, PQ and WRR schedulers in terms of QoS parameters such as IPTD, IPDV and IPLR for real-time UMTS traffic in case of network/bottleneck link overload. Only when novel approach is used, it is possible to keep QoS parameter under target values for Conversational and Streaming traffic classes, but at the same time to provide fair level of bandwidth for Interactive and Background traffic classes. Statistical analysis shows that there is almost no statistical relationship between the performance metrics of real-time services and the network load when novel approach is used.

Future avenues of research will focus on hierarchical traffic scheduling in order to perform refined scheduling between voice and video telephony with the main goal of providing better Quality of Service for real-time traffic flows.

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