SIMULATIONS AND ANALYSIS OF 3G AND WLAN INTERWORKING

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SUMMARY

Seamless interconnection with wireless LAN and 3G technologies is essential for the future wireless environment. The Wireless Local Area Network (WLAN) integrated mobile device is designed to extend the reach of enterprise applications and to create new collaboration environments. Also very rapid new service development has started to change traffic mixes in the cellular networks towards IP dominating carriers like GPRS (General Packet Radio Service) and WLAN. These drastic changes require new research in the network as well as system interworking areas for both the cellular and WLAN technology areas. This is the main challenge our research is trying to solve giving answers to rising interworking and interoperability questions.

Keywords: *3G, WLAN, Interworking, QoS, 802.11e*

1. INTRODUCTION

In the future the co-existence of several QoS capable radio access technologies is possible. The users have several network interfaces in one mobile terminal and thus the opportunity to choose which access technology to use. WLAN connections [1], [5] can offer faster connections than 3G (2Mb/s) with cheaper prices. For example the IEEE 802.11a version can achieve data rates of up to 54 Mbit/s at the wireless medium using the *Orthogonal Frequency Division Multiplexing* (OFDM) modulation technique in the unlicensed 5 GHz frequency band [6].

Quality-of-Service (QoS) supported WLAN accesses will be soon available, which makes possible the usage of these access networks with 3G mobile devices. Several technologies have been introduced to improve the QoS of the IEEE 802.11 DCF (Distributed Co-ordination Function) protocol; like IEEE 802.11e, Blackburst and DFS (Distributed Fair Scheduling) [8], [9]. The problem with WLAN networks is the high error rate probability. It can rise up to 40% causing trouble especially to the streaming type of applications. IEEE 802.11e standard is trying to correct the situation by enabling the use of a maximum of eight separate priority queues for prioritising higher priority traffic compared to other traffic [4].

The 3GPP has defined four traffic (QoS) classes that have their own QoS profile and attributes [3]. All traffic in the 3/4G network will be handled according to the operator's requirements for the each of the traffic classes. These 3GPP traffic classes are mapped to the QoS policies in the core network to enable end-to-end QoS. The main QoS method to be used in core network is proposed to be DiffServ [1], but also flow-based control by RSVP [3] has been proposed for Rel. 6. RSVP truly provides the requested QoS, but it has scalability problems in large networks. DiffServ architecture addresses this problem.

The 3GPP defined traffic (or QoS) classes are *Conversational* class for voice and RT multimedia messaging, *Streaming* class for streaming type of applications (Video On Demand (VOD) etc.), *Interactive* class for interactive type of applications (eCommerce, WEB browsing, etc.) and *Background* class for background type applications (email, FTP, etc.).

Table 1 presents the current QoS attributes for the radio bearer [3].

2. IEEE 802.11E QOS ISSUES AND MAPPINGS TO 3G TRAFFIC

IEEE 802.11e employs a new MAC protocol called the *Enhanced Distributed Co-ordination Function* (EDCF) and it is the basis for the *Hybrid Co-ordination Function* (HCF) [4]. The QoS support is done with the *Traffic Categories* (TC). MAC service data units will be delivered through multiple back-off instances within one station, each back-off instance parameterised with TC-specific parameters. A single station can have up to eight transmission queues realized as virtual stations inside a station, with QoS parameters that determine their priorities. If the counters of two or more parallel TCs in a single station reach zero at the same time, a scheduler inside the station avoids the virtual collision. The scheduler grants the *Transmission Opportunity* (TXOP) to the TC with highest priority, out of the TCs that virtually collided within the station. There is then still a possibility that the transmitted frame collides at the wireless medium with a frame transmitted by other stations [4].

When we are using strict priority queuing, the mappings between IEEE 802.11e and 3GPP traffic classes can be done as follows [10,12]. We can use four strict priority queues of which each corresponds to one of the 3GPP traffic classes. We can also use the three THPs (Traffic Handling Priority) used for the *Interactive* class to further sub classify *Interactive* class traffic by inserting it to three separate queues.

Traffic class	**Conversa-tional class**	**Streaming class**	**Interactive class**	**Background class**
Maximum bit rate (kbps)	<= 16 000 (2) (7)	<= 16 000 (2) (7)	<= 16 000 - overhead (2) (3) (7)	<= 16 000 - overhead (2) (3) (7)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)
SDU format information	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	$5*10^{-2}$, 10^{-2}, $5*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}	$5*10^{-2}$, 10^{-2}, $5*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6)
SDU error ratio	10^{-2}, $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-1}, 10^{-2}, $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-3}, 10^{-4}, 10^{-6}	10^{-3}, 10^{-4}, 10^{-6}
Transfer delay (ms)	80 – maximum value	250 – maximum value		
Guaranteed bit rate (kbps)	<= 16 000 (2) (7)	<= 16 000 (2) (7)		
Traffic handling priority			1,2,3	
Allocation/Retention priority	1,2,3	1,2,3	1,2,3	1,2,3
Source statistic descriptor	Speech/unknown	Speech/unknown		
Signaling Indication			Yes/No	

Table 1 3GPP QoS attributes for radio access bearer services

1) Void.
2) The granularity of the bit rate attributes shall be studied. Although the UMTS network has the capability to support a large number of different bit rates, the number of possible values, shall in practice be limited, in order to prevent introducing unnecessary complexity into terminals, charging and other inter-working functions. An exact list of the supported values shall be defined together with S1, N1, N3 and R2 working groups.
3) Impact from layer 2 protocols on maximum bit rate in non-transparent RLC protocol mode shall be estimated.
4) In case of PDP type = PPP, maximum SDU size is 1502 octets. In other cases, maximum SDU size is 1 500 octets.
5) Definition of possible values of exact SDU sizes, for which UTRAN can support transparent RLC protocol mode, is the task of RAN WG3.
6) Values are derived from CRC lengths of 8, 16 and 24 bits on layer 1.
7) In case of GERAN the highest bit rate value is 473.6 kbps.

Traffic class	**PHB actions (mapped into DSCP)**
Conversational	EF
Streaming	AF 1
Interactive (THP 1)*	AF 21
Interactive (THP 2)	AF 22
Interactive (THP 3)	AF 23
Background	AF 3
* Traffic Handling Priority	

Table 2 3G Traffic class mappings into DSCP

All control plane traffic can be handled similarly mapping it into one or several IEEE 802.11e queues. The mapping process can be policy based controlled and the mapping can be indicated at the IP level by the DSCP (DiffServ Code Point) inserted to the TOS field by DS classifier/marker mechanism or by the actual application that generates the control plane traffic. Table 2 shows the PHB (Per-Hop-Behaviour) actions with DSCP mappings [11].

3. RESEARCH PROBLEM

Because WLAN provides faster connections with cheaper prices than 3G networks, the interest towards 3G and WLAN interworking has risen lately. Though the wireless channel is the biggest packet error and therefore packet loss cause in wireless communication systems, several efforts to improve the QoS in WLAN networks have been proposed. In high error rate conditions the radio channel should offer better QoS to higher service classes.

The packet size affects the traffic delay and throughput in an erroneous radio channel, thus the QoS level variation due to the radio channel can be compensated by changing the traffic class related packet sizes dynamically.

To ensure the end-to-end QoS, the whole path between the sender and receiver must be QoS capable. QoS management should be IP-based, requiring some QoS mappings between OSI layers 2 and 3. IntServ and DiffServ QoS architectures have been standardized for IP level QoS management solutions. Problems arise due the differences in QoS mechanisms of different access and core technologies.

The goal of our work is to study the interoperability of 3G and IEEE 802.11e wireless networks and the suitability of DiffServ and RSVP core networks to provide efficient end-to-end QoS control in wireless communication systems. The throughputs, delays and dropping rates are studied with QoS mapping and both core networks. Another goal of the work is to find out how much the throughput of the traffic decreases with various wireless channel error rates, while changing the traffic mix and average packet sizes of the individual traffic class. We also try to find optimal packet size combination for each of the traffic classes.

The rest of the article is organized as follows. In section 4 the simulation model, three scenarios and the achieved results are presented. One access point scenario is extended to cover end-to-end situation with DiffServ and IntServ core networks. In section 5 the results are analysed and section 6 concludes this article.

4. SIMULATION MODEL AND SIMULATION RESULTS

With the evolution of QoS-capable 3G wireless networks, the wireless community has been increasingly looking for a framework that can provide effective network-independent end-to-end QoS control. One big problem arises with these kind of diverse networks: namely the dissimilarity of traffic characteristics and QoS management methods in access and core networks. The problem with WLAN networks is the high error rate probability. IEEE 802.11e standard has been applied trying to correct the situation by enabling the use of a maximum of eight separate priority queues for prioritising higher priority traffic compared to other traffic [4]. QoS supported WLAN uses the *Enhanced Distributed Co-ordination Function* (EDCF), which is the basis for the *Hybrid Coordination Function* (HCF) [4].

RSVP has been used in domains, where there is no direct radio interface. In the RAN (Radio Access Network) case we have assumed that the radio interface between BTS (Base Transceiver Station) and UE (User Equipment) in RAN will be handled similarly to WLAN but with different methods defined by 3GPP standardization. As RAN is based on ATM, the basic assumption has been that the RAN is correctly dimensioned to carry all traffic coming from and going to UE direction, so by default RAN QoS is out of scope of the scenarios considered in this article.

4.1 Mapping QoS attributes to cross domain interfaces

3GPP has defined four traffic (QoS) classes and three subclasses (Interactive THP, Traffic Handling Priority) that can have their own QoS attributes [3]. All traffic in the 3G network will be handled according to the operator and service's requirements for each of these traffic classes. The main QoS method to be used at the core network is supposed to be DiffServ [1]. In addition to that 3GPP has defined RSVP as an additional UE originated QoS method in 3GPP Rel6 between UE-SGSN (UE Serving GPRS Support Node) and GGSN (Gateway GPRS Support Node). RSVP can be used in the situations, where scalability problems will not arise (e.g. small networks). 3G traffic classes are: *Conversational* class (for voice and real-time multimedia messaging), *Streaming* class (for streaming applications like Video On Demand (VOD) etc), *Interactive* class (interactive applications like eCommerce, WEB-browsing, etc.) and *Background* class (for background applications such as email and FTP). QoS values for each traffic classes are defined in [2]. In DiffServ domain four priority queues can be implemented for each of the 3G traffic classes. The three THPs (Traffic Handling Priority) are also available for *Interactive* class to further sub-classify class traffic by inserting it to three separate queues. 3G to DiffServ mapping process can be policy based controlled and the mapping can be indicated at the IP level by the DSCP (DiffServ Code Point) inserted to the TOS field by DS classifier/marker mechanism or by the actual application that generates the control plane traffic. Table 2 shows the PHB actions with DSCP mappings.

The nature of RSVP functionality differs significantly from DiffServ. RSVP uses end-to-end signalling enabling a single UE to reserve end-to-end transport capacity from the network or RSVP can be used by Bandwidth Broker and COPS-PR (Common Open Policy Server Policy Provisioning) protocol to set appropriate traffic filters to routing nodes to achieve similar capacity reservation than by UE signalling.

4.2 General issues about simulations

The goal for the simulations is to study how much the throughput of the traffic decreases with various error rates, while changing the traffic mix and average packet sizes of the individual traffic class.

We used network simulator version 2 (ns-2) with IEEE 802.11 EDCF functionality implemented by Ni Qiang et al. in the Planete Project-INRIA [7]. To emulate the process of packet transmission errors, we extended the simulator by implementing a two-state Markov model to the air interface.

MMPP (Markov Modulated Poisson Process) is a doubly stochastic process, where the intensity of a Poisson process is defined by the state of a Markov chain (Figure 1). The transition matrix Q of the modulating Markov chain is defined by

$$Q = \begin{bmatrix} -\omega_1 & \omega_1 \\ \omega_2 & -\omega_2 \end{bmatrix}.$$

In our error scenario, the channel switches between a "good state" and a "bad state". In Figure 1 the λ1 is the good state and λ2 the bad state. Packets are transmitted correctly, when the channel is in state λ1, and errors occur, when the channel is in state λ2. When the channel is in state λ1, it can either remain in this state with probability 1-ω1 or make the transition to state λ2, with probability ω1. Likewise, if the channel is in state λ2, it remains in this state with probability 1-ω2 and changes state with probability ω2.

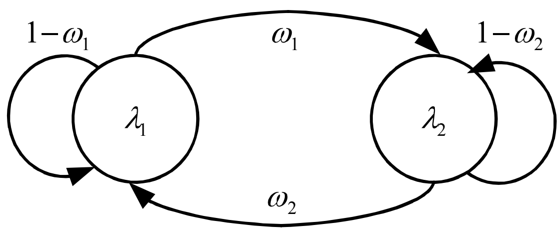

Fig. 1 State machine for a 2-state MMPP

4.3 One access point simulation case

In the first simulation scenario we concentrate on WLAN radio connections with only one access point (AP) [10]. The simulation scenario is illustrated in Fig. 2. It includes five WLAN stations, of which one is the access point. Station 1 has the highest priority and station 4 the lowest priority. The total traffic load is 2.5 Mbit/s per station, thus the WLAN radio channel (11Mbps) is almost fully loaded. Station 1 and station 3 have both five flows all generating 500 kbit/s constant bit rate (CBR) traffic. Stations 2 and 4 have also five flows each generating an average of 500 kbit/s variable bit rate (VBR) traffic. So each station generates the same amount of traffic. VBR parameters of both of these stations are as shown in Table 3. Four priority levels were used in our simulations. EDCF parameters of different TCs are shown in Table 4.

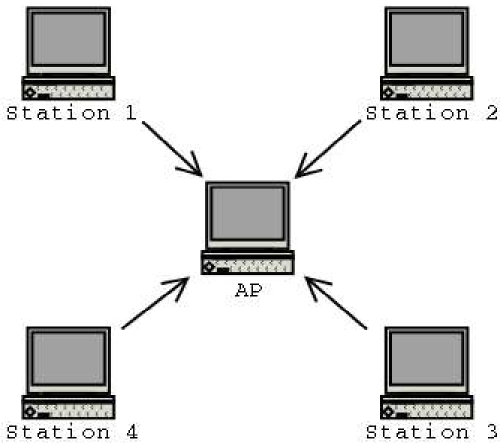

Fig. 2 Simulation scenario

Rate	500 kbit/s
Rate deviation	0.25
Rate time	2.0
Burst time	1.0
No. of changes	10
Time deviation	0.5
Maxrate	648 kbit/s

Table 3 Traffic properties of the VBR traffic

Traffic class	1	2	3	4
CWMin	7	10	15	127
CWMax	7	31	255	1023
AIFS (CWOffset)	2	4	7	15

Table 4 Used EDCF Parameters

We varied the packet sizes of the traffic classes by iterating through all the packet size combinations between 100 and 1500 bytes using six 280 byte steps. The possible packet sizes were thus 100, 380, 660, 940, 1220 and 1500 bytes. The iteration was as follows: at the iteration round 0 packet size of all the classes is 100 bytes. The lowest priority traffic class packet size is increased by 280 bytes at every iteration. If the packet size is 1500 bytes, it is reset to 100 bytes and the next lowest traffic class packet size is increased by 280 bytes. This way the lowest priority traffic class iterates through all the packet size possibilities within 6 iterations and the next lowest priority traffic class within $6^2 = 36$ iterations. The second highest traffic class goes through all the packet sizes within $6^3 = 216$ iterations and the highest within $6^4 = 1296$ iterations. This enables us to study the effect of the packet size to the QoS parameters of each traffic class.

The radio channel packet error rate was varied also from 0% to 60% with steps of 20%. Table 5 shows the transition probabilities we used to achieve different packet error rates in the air interface.

Error rate	ω_1	ω_2
20%	0.16	0.63
40%	0.40	0.60
60%	0.90	0.60

Table 5 Transition probabilities for 2-state MMPP

Fig. 3, Fig. 4, Fig. 5 and Fig. 6 show the obtained throughputs, when channel error rate increases from 0% up to the 60% of totally lost packets. Fig. 7 depicts the total throughput of these four different error rate scenarios. As can be seen from the graphs EDCF works quite well with different channel error rates. Highest traffic class will get significantly more capacity than lower priority classes with every simulated channel error rate. The need for classification increases with the channel error rate. This shows also the fact that EDCF is suitable for using with 3G to support QoS in those situations, when 3G devices will use WLAN access to the core IP network.

One interesting issue arises with the fact that the highest priority traffic class will get more bandwidth, when the packet size and the channel error rate are increasing. With the proposed 3G traffic classification specifications this can lead to the situation that the highest class uses all the capacity under certain network conditions. Our simulation results proved this issue clearly. Based on this issue, it is important to find out the limits, where packet size and channel error rate can change. If we think for example of the situation, where we have small size high priority packets (voice traffic) under heavily loaded network (i.e. channel error rate is high), we will lose also those small size high priority packets.

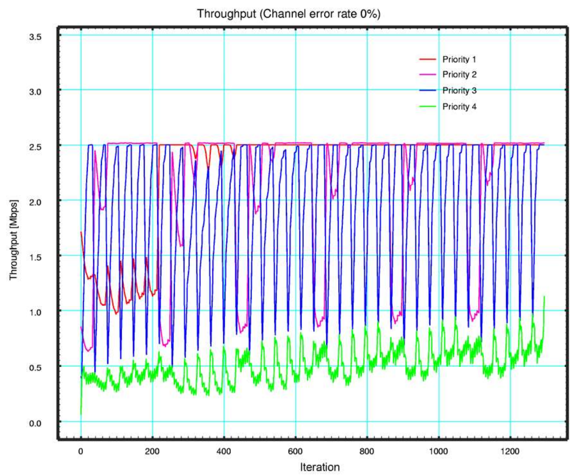

Fig. 3 Throughput of the 4 traffic classes in relation to the packet size combination iteration, when the channel error rate is 0%

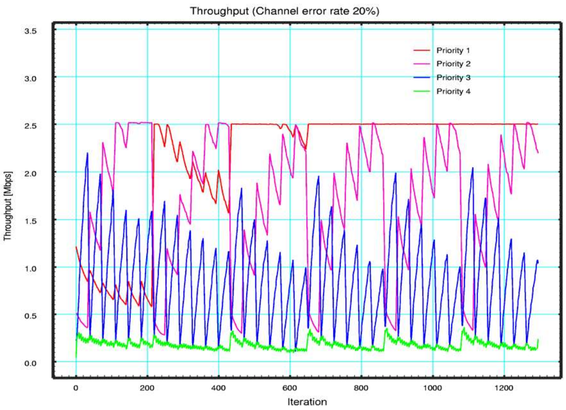

Fig. 4 Throughput of the 4 traffic classes with channel error rate 20%

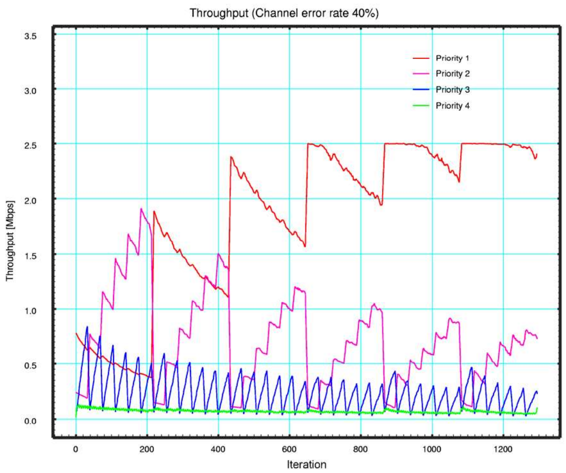

Fig. 5 Throughput of the 4 traffic classes with channel error rate 40%

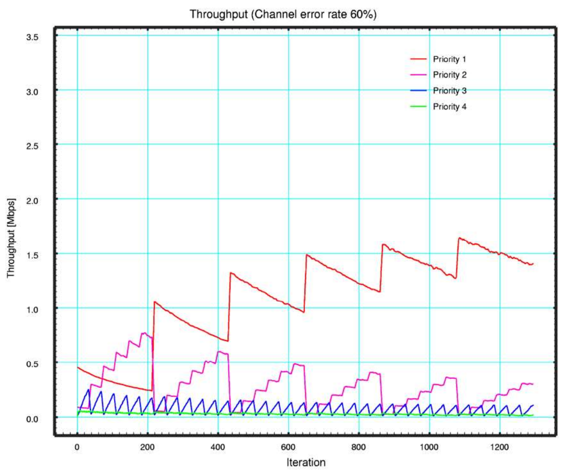

Fig. 6 Throughput of the 4 traffic classes with channel error rate 60%

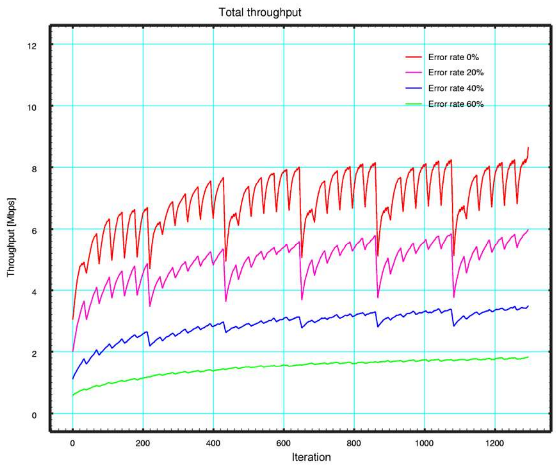

Fig. 7 Total throughput of the 4 different error rate scenarios

4.4 Extended end-to-end simulation case

Here we extend our studies to the end-to-end scenario [11]. We use the same WLAN scenario as in [10] and previous section, but now we also take into account DiffServ core network. The goal for the simulations is to study how much the throughput of the traffic decreases with various error rates, while changing the traffic mix and average packet sizes of the individual traffic class.

The simulation scenario is illustrated in Fig. 8. It includes eight WLAN stations. Stations 1, 2, 3 and 4 are located in the coverage area of the access point AP1. Station 1 has the highest and station 4 the lowest priority. Stations 5, 6, 7 and 8 are connected to access point AP2. The link between AP2 and the core network is 10 Mbit/s link with 5 ms delay. Link between LSR5 and LSR6 routers is 15 Mbit/s link with 13 ms delay and the link between LSR6 and LSR7 15 Mbit/s link with delay 14 ms. All other wired links in core network are 10 Mbit/s links with negligible delay.

DiffServ QoS architecture is used in core network. LSR4 is a DiffServ Edge router and LSR5 is a DiffServ Core router. In LSR4 there are four physical DropTail RED (Random Early Detection) queues with two precedence levels in each queue; one level for packets in profile and the other for packets out of profile. The queue sizes are shown in Table 6. A Token bucket traffic policer is used. The parameters of the policer are shown in Table 7.

Traffic class	Conversa-tional	Streaming	Interactive	Background
In profile	30	60	30	30
Out of profile	0	30	30	0

Table 6 Queue sizes

Traffic class	Conversa-tional	Streaming	Interactive	Background
CIR	3.0 Mbit/s	2.5 Mbit/s	2 Mbit/s	1.5 Mbit/s
MBS	3 KB	2 KB	1.5 KB	1.5 KB

Table 7 Token bucket policer parameters

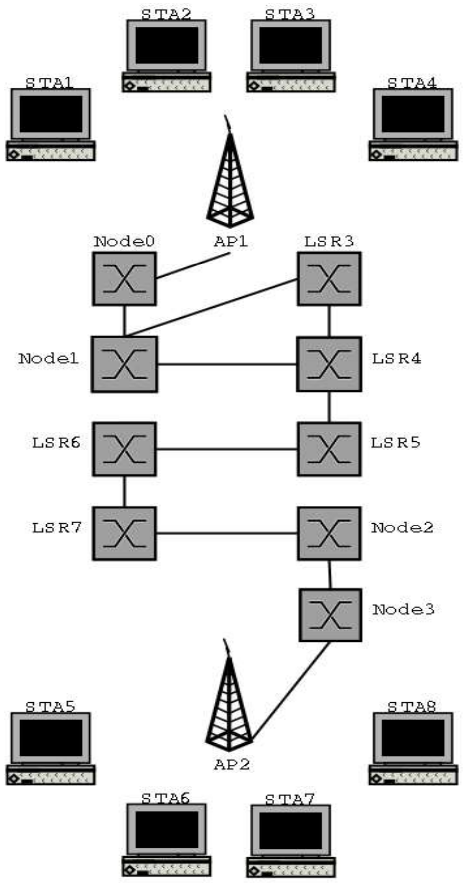

Fig. 8 End-to-end network simulation configuration

In our simulations stations 1, 2, 3 and 4 are sending nodes and stations 5, 6, 7 and 8 are receiving nodes. Station 1 sends to station 5, and so forth. Each station generates the same amount of traffic. The total traffic load is 2.5 Mbit/s per station. Station 1 and station 3 have both five flows all generating 500 kbit/s constant bit rate (CBR) traffic.

Rate	500 kbit/s
Rate deviation	0.25
Rate time	2.0
Burst time	1.0
No. of changes	10
Max. rate	648 kbit/s

Table 8 Traffic properties of the VBR traffic

Stations 2 and 4 have also five flows each generating an average of 500 kbit/s variable bit rate (VBR) traffic. VBR parameters of both of these stations are shown in Table 8. We used four priority levels in our simulations. EDCF parameters of different TCs are shown in Table 9.

Traffic class	**Conversa-tional**	**Streaming**	**Interactive**	**Backgroun d**
CWMin	7	10	15	127
AIFS	2	4	7	15
CWMax	7	31	255	1023

Table 9 Used EDCF Parameters

In this scenario we varied the packet sizes of the traffic classes iterating through all the packet size combinations starting from 500 bytes to 1500 bytes with 500 byte steps. The possible packet sizes were thus 500, 1000 and 1500 bytes. The iteration was following: at iteration 0 packet size of all the classes is 500 bytes. The lowest priority traffic class packet size is increased by 500 bytes every iteration. If the packet size is 1500 bytes, it is reset to 500 bytes and the next lowest traffic class packet size is increased by 500 bytes. This way the lowest priority traffic class iterates through all the packet size possibilities in 3 iterations and the next lowest priority traffic class in $3^2 = 9$ iterations. The second highest traffic class goes through all the packet sizes in $3^3 = 27$ iterations and the highest in $3^4 = 81$ iterations.

The radio channel packet error rate was varied also from 0% to 60% with 30% steps. Table 10 shows the transition probabilities we used to achieve different packet error rates in the air interface.

Error rate	ω_1	ω_2
30%	0.27	0.62
60%	0.90	0.60

Table 10 Transition probabilities for 2-state MMPP

Fig. 9 shows obtained throughput when channel error rate is 0% and Fig. 10 shows obtained throughput when channel error rate is 30% of totally lost packets. Fig. 11 and Fig. 12 depict the delays of these same error rate scenarios, and Fig. 13 and Fig. 14 show the dropped packets. As can be seen from the graphs EDCF works quite well with different channel error rates. Highest traffic class will get significantly more capacity than lower priority classes in all our simulation scenarios. The delays and packet drops are also in the correct order. This shows also the fact that EDCF is suitable for using with 3G to support QoS in those situations when 3G devices will use WLAN access to the core IP network.

Same interesting fact with the highest priority class and big packet sizes as noticed in one access point case was noticed in the end-to-end scenario as well. When the packet size and channel error rate increases, WLAN station will get increasingly bandwidth.

Interesting fact is that the packet size that gives the best throughput to the highest class is dependent on the error rate. For example, the maximum throughput of the highest class in 0% and 30% error scenario is achieved when the highest-class packet size is about 700 bytes but as the error rate increases, the larger packet size will give the best results, when the lower priority packet sizes are small. When the error rate is very high (tested with 60%), the maximum throughput is achieved with largest 1500 byte packet size. This is quite obvious, but it should be carefully thought when building up WLAN hot spots. By using larger packet sizes the network delay (due to queuing) also increases, and that can cause several problems with real time applications. The results depict also that if suitable packet and error rate combination can be fixed, the different QoS requirements can be met. It is up to the network configuration how to tune these parameters suitably for different applications.

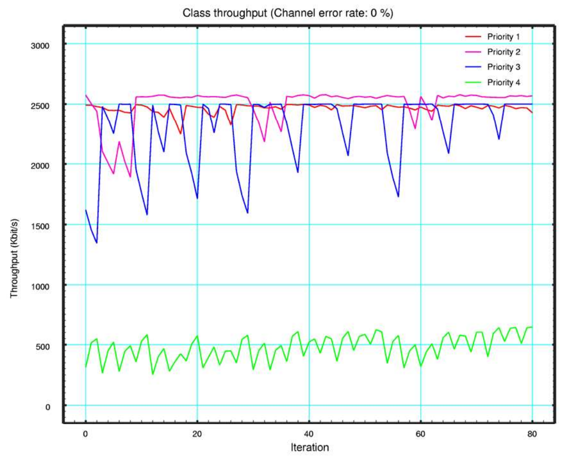

Fig. 9 Throughput of the 4 traffic classes with channel error rate 0%

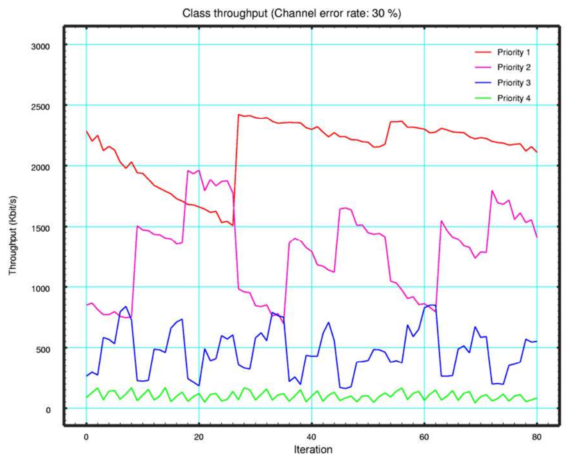

Fig. 10 Throughput of the 4 traffic classes with channel error rate 30%

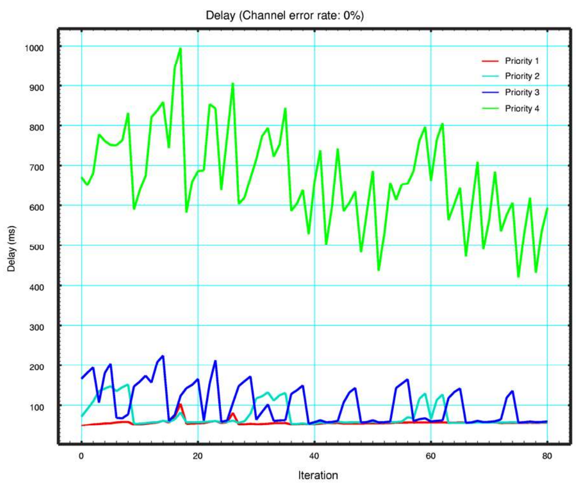

Fig. 11 Delay of the 4 traffic classes with channel error rate 0%

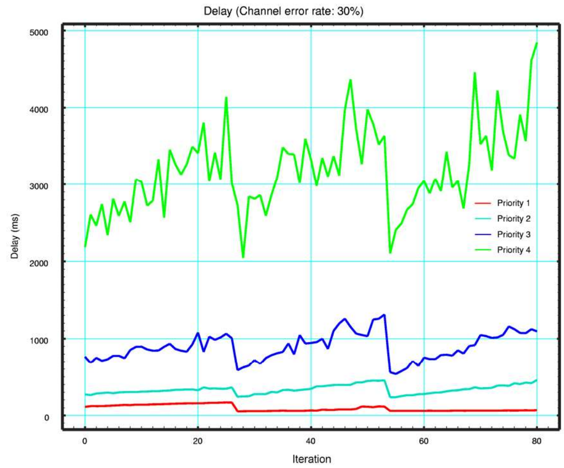

Fig. 12 Delay of the 4 traffic classes with channel error rate 30 %

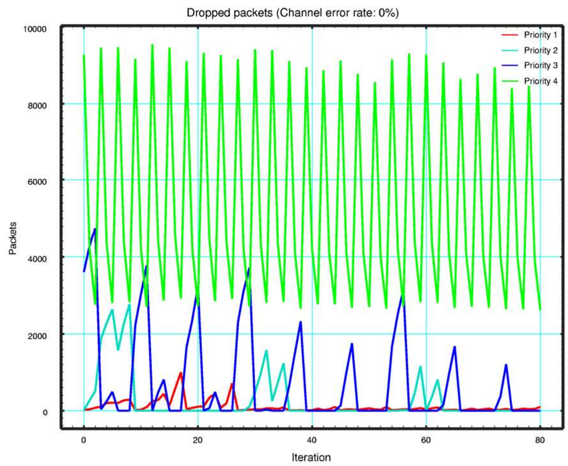

Fig. 13 Packet drops of the 4 traffic classes with channel error rate 0%

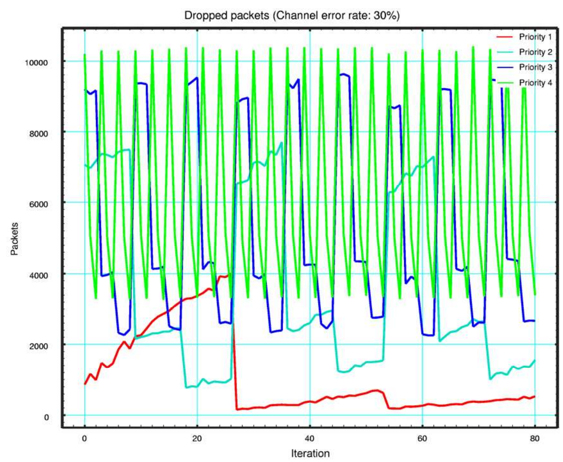

Fig. 14 Packet drops of the 4 traffic classes with channel error rate 30%

4.5 RSVP and DiffServ comparison

The goal is to study what are the throughputs, delays and dropping rates in RSVP and DiffServ core cases [13]. Simulation environment in Fig. 15 consists of six core network nodes (R), which build up a ring. Each of the core routers has three connected 3G/WLAN access points and each of the APs have four connected UEs with different priorities (conversational, streaming, interactive, background). So there are 18 access points and 72 UEs. The UEs below the dashed line (connected to APs 1-9) each send data at the rate of 2.5Mbps to a random UE above the dashed line (connected to APs 10-18). When considering one specific AP, stations 1 and 3 generate 2.5Mbps CBR traffic and stations 2 and 4 2.5Mbps VBR traffic. The WLAN stations start sending at time 3 - 4.5 seconds randomly. Simulation time is 40 seconds and the used packet size is 1000 bytes for all stations. Available bandwidth within the core network was 8 Mbps. In the core network all wired capacity was reserved for RSVP use and best effort queue size was 5000 bytes in every node. We used the traffic parameterization shown in Table 11 for RSVP.

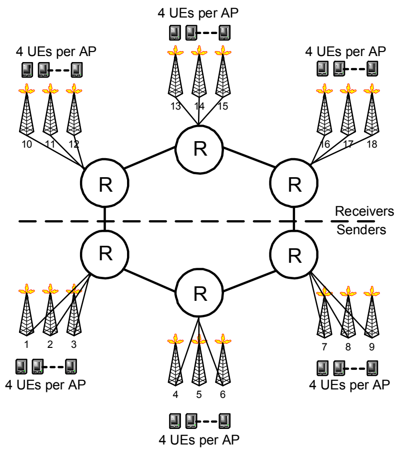

Fig. 15 Simulation environment

3GPP Traffic class	Bandwidth Mb/s	Bucket size bytes
Conversational	3.0	3000
Streaming	2.5	2000
Interactive (3 THPs)	2.0	1500
Background	2.0	1500

Table 11 RSVP parameterization

As link capacity is small compared to number of reservations, some of the reservations do not succeed and traffic related to them goes in the network as best effort traffic. With RSVP we used *Weighted Fair Queuing* (WFQ) method. DiffServ uses Token Bucket policers and its parameterization is presented in Table 12.

DiffServ uses RED queuing in DropTail mode. In-profile packet queue lengths are 30 packets for each class and out-of-profile packet queues are 60 packets long. We used four priority levels in both scenarios. EDCF parameters of different Traffic Classes are shown in the Table 13.

To emulate the process of packet transmission errors we used the same two-state Markov model in the air interface as in the previous sections. The transition probabilities are presented in Table 14. We ran several different error rate simulations but we find 0 and 20% error rates most illustrative.

3GPP Traffic class	CIR Mb/s	Bucket size bytes
Conversational	3.0	3000
Streaming	2.5	2000
Interactive (3 THPs)	2.0	1500
Background	2.0	1500

Table 12 DiffServ token bucket parameterization

Traffic class	Conversa-tional	Streaming	Interac-tive	Back-ground
CWMin	7	10	15	127
AIFS (CWOffset)	2	4	7	15
CWMax	7	31	255	1023

Table 13 Used EDCF parameters

Error rate	ω_1	ω_2
0%	0	1
20%	0.16	0.63

Table 14 Transition probabilities for 2- state MMPP

4.5.1 Scenario 1: RSVP core

In this chapter the througputs, delays and packet drops in the RSVP core network are evaluated. All these QoS parameters are calculated as mean values of all the class-specific end-to-end traffic flows, thus the results include the wireless networks at both ends and the RSVP core network.

RSVP throughputs

As can be seen in Fig. 16 *Interactive* class has higher throughput than *Streaming* class. This is caused by the random nature of reservation signalling. Table 15 explains the class-specific reservation probabilities as a function of available link bandwidth in the RSVP core. For example, in case there is already 6Mbps reservation for two *Conversational* class flows only *Interactive* and *Background* classes can reserve the rest of the bandwidth, assuming that all priorities try to make a reservation with the same probability.

Other traffic characteristics follow very well expectations on throughput delay. Throughput is best and delay follows the throughput being higher than in other classes due to the high throughput. It can also be seen in Fig. 16 and Figure 17 that the traffic flows are smoother in lower error environment. Average throughputs on each traffic class also follow well our expectations. Throughputs are in preferable order, *Conversational* class has highest throughput and *Background* class the lowest. Fig. 18 shows also slight rise of throughput in *Interactive* class and corresponding declining in

Background class for higher error rates. This can be caused by differences in reservation success between classes. It should be noted that the throughput of the *Background* class is zero after time 25s. UEs sending *Background* traffic are not allowed to send data at all to the wireless network due the ECDF MAC protocol. As can be seen from Figures 21 and 22 the packet loss is quite constant the whole simulation time.

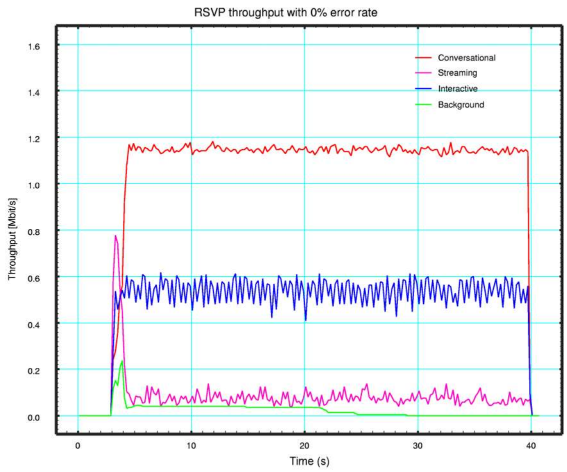

Fig. 16 RSVP throughput with 0% error rate

Available link bandwidth	Conversa-tional	Streaming	Interactive	Back-ground
8	0.25	0.25	0.25	0.25
6	0.25	0.25	0.25	0.25
5.5	0.25	0.25	0.25	0.25
5	0.25	0.25	0.25	0.25
4	0.25	0.25	0.25	0.25
3.5	0.25	0.25	0.25	0.25
3	0.25	0.25	0.25	0.25
2.5	0	0.33	0.33	0.33
2	0	0	0.5	0.5
Average	0.194	0.231	0.287	0.287

Table 15 Reservation probabilities

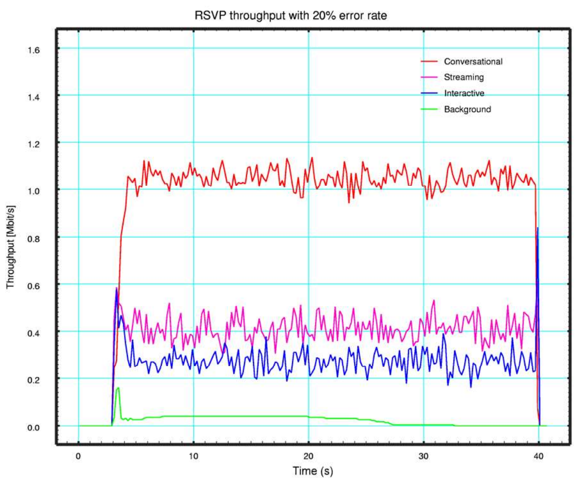

Fig. 17 RSVP throughput with 20 % error rate

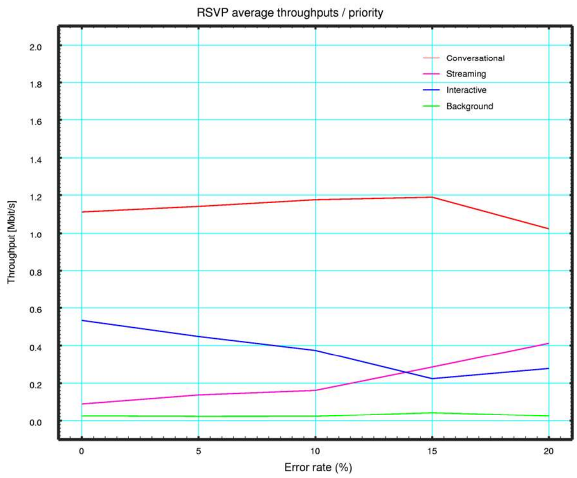

Fig. 18 RSVP Average throughputs/priority

RSVP delays

Delay behaviour (Fig. 19 and Fig. 20) has similar features as throughput. All aggregate flows are in correct order and delay is adequately low (< 0.5 ms) in both *Conversational* and *Streaming* class. Also *Interactive* and *Background* classes are far below

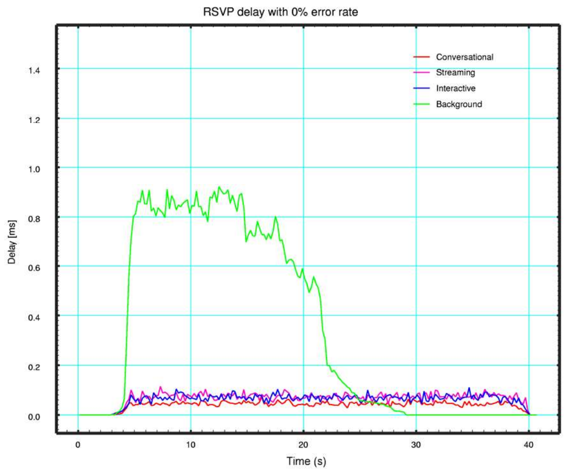

Fig. 19 RSVP delay with 0% error rate

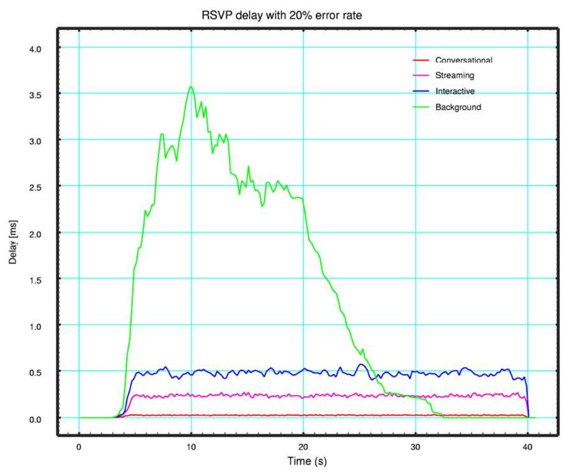

Fig. 20 RSVP delay with 20% error rate

their worst-case scenario values. The delay of the *Background* class varies around 0.8 ms, but after 20 seconds it starts to decrease and finally reaches zero value. Same phenomenon is shown with both channel error rates. This is due to the fact that the RSVP throughput has also decreased to zero, so there is no *Background* traffic in the RSVP core.

RSVP packet dropping

RSVP packet dropping follows the throughput being higher in higher throughput classes. In this case a better describer for packet dropping would probably be percentage value, which would turn the order of curves into opposite order. Dropping rate is very stable, when the error rate is 0% (Fig. 21), but becomes unstable and rising with error rate is 20% (Fig. 22).

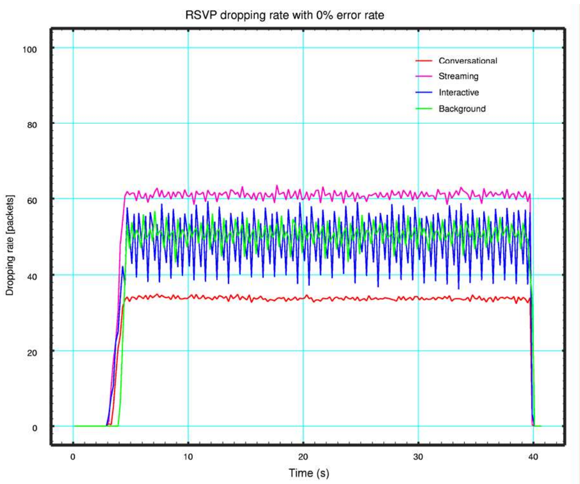

Fig. 21 RSVP dropping rate with 0% error rate

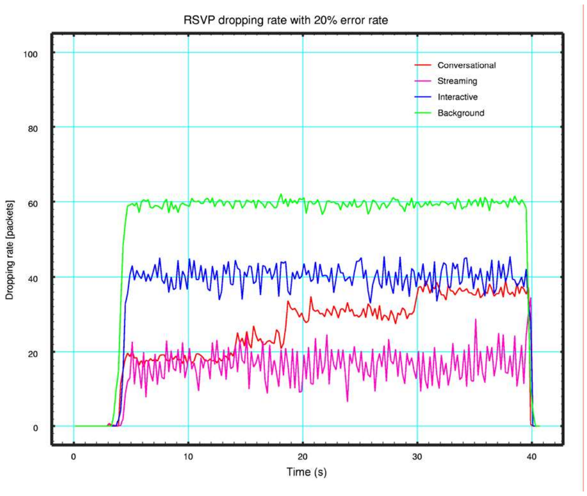

Fig. 22 RSVP dropping rate with 20% error rate

4.5.2 Scenario 2: DiffServ core

In this chapter the throughputs, delays and packet drops in the DiffServ core network are evaluated. All these QoS parameters are calculated as mean values of all the class-specific end-to-end traffic flows, thus the results include the wireless networks at both ends and the DiffServ core network.

DiffServ throughputs

At the Diffserv scenario *Conversational* traffic is dominant and other traffic classes are very close to zero. The obvious difference is that RSVP has much better control over lower priority flows and therefore it would be a better solution for interworking QoS control purposes.

As can be seen from Fig. 23 traffic with priorities 3 and 4 disappears within 10 seconds after beginning of the test. This means also that the delay for priorities 3 and 4 becomes 0 (zero), as there is no traffic in priority classes 3 and 4 as shown in next chapter. Difference between throughputs with 0% and 20% error rate is significantly low. Fig. 25 shows that the average throughputs of the classes are the same between different error rates. This indicates that throughput behaviour is very stable, when using DiffServ as opposed to RSVP, which causes large variations in class throughputs between error rates. Still, this stable behaviour is achieved at the cost of lower priority class throughputs, which are close to zero.

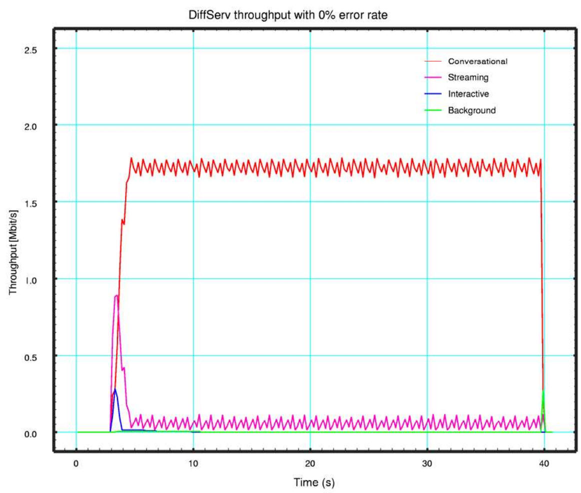

Fig. 23 DiffServ throughput with 0% error rate

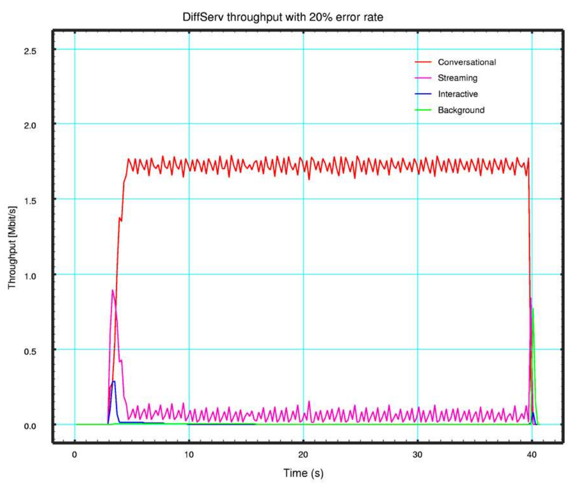

Fig. 24 DiffServ throughput with 20% error rate

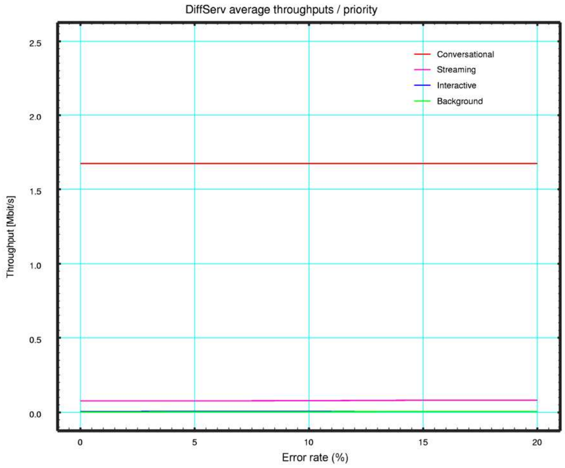

Fig. 25 DiffServ average throughputs/priority

DiffServ delays

As presented in Fig. 26, the delay for flows with priorities 3 and 4 become zero (vanishing from logarithmic scale). This actually means that flows with priorities 3 and 4 are not reaching their target receiver node, but are totally dropped during transmission. Similar effect occurs with 20% error rate in Fig. 27.

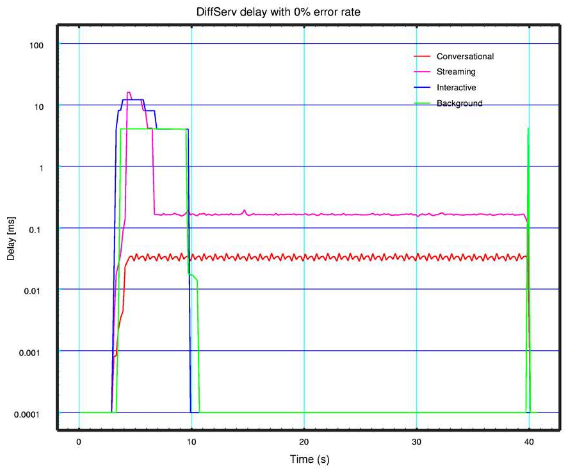

Fig. 26 DiffServ delay with 0% error rate

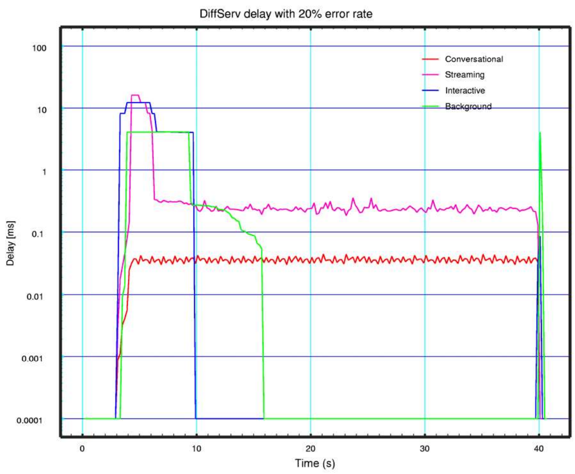

Fig. 27 DiffServ delay with 20% error rate

DiffServ packet dropping

Fig. 28 shows that dropping rates are located as could be predicted according to their priorities. As can be seen in Fig. 29 increased error rate increases dropping rate accordingly. High error rate affects the dropping rates, so that there seems to be lower dropping rate in 20% error rate scenario. As the air interface corrupts packets, few of them reach the wired network. Hence, there is smaller probability for the congestion in the wired network.

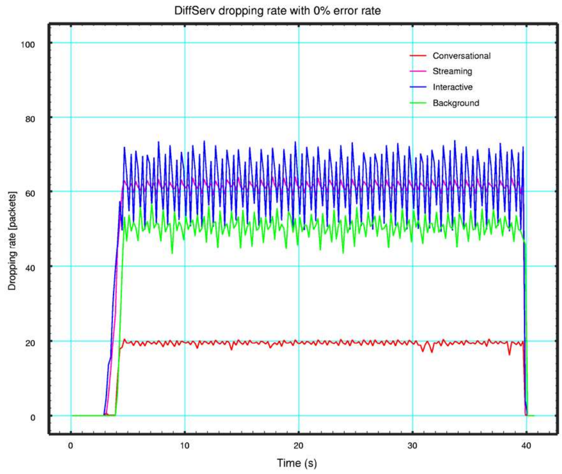

Fig. 28 DiffServ dropping rate with 0% error rate

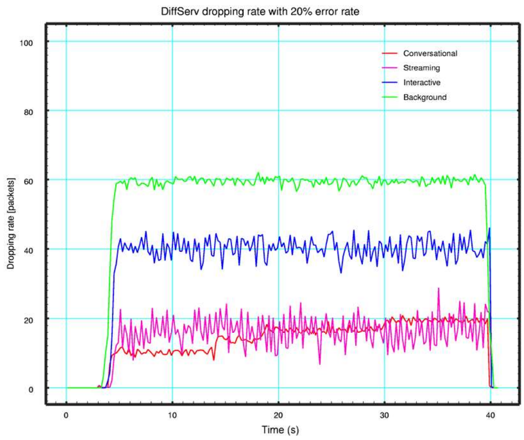

Fig. 29 DiffServ dropping rates with 20 % error rate

5. RESULT ANALYSIS

In this section the achieved results are analysed [12]. Especially the optimal packet size for every class is calculated and RSVP and DiffServ cores compared.

5.1 Optimal packet size

The best packet size combination was found from the simulation results by calculating an emphasized factor to all packet size combinations. The factor was calculated by using the formula (1)

$$S_i = D_i + T_i , \tag{1}$$

where

$$D_i = \sum_{j=1}^{4} 100 * \delta_j * d_j^i \tag{2}$$

and

$$T_i = \sum_{j=1}^{4} 100 * \tau_j * b_j^i . \tag{3}$$

D_i is a total delay factor and T_i is a total throughput factor calculated from the results of all classes. δ_j is the weight coefficient of the delay of class *j* and d_j^i is the factor calculated to the delay of class *j* at the iteration *i*. τ_j is the weight coefficient of the throughput of class *j* and b_j^i is the factor calculated to the throughput of class *j* at the iteration *i*. The delay factor d_j^i is a factor between 0 and 1. It was calculated with formula (4)

$$d_j^i = \frac{d_j^{\max} - d_i}{d_j^{\max} - d_j^{\min}} , \tag{4}$$

where d_i is the measured delay at iteration *i*, $d_j^{\max}$ is the maximum limit of delay of class *j* and $d_j^{\min}$ is the minimum limit of delay of class *j*. The factor of all delays above the maximum limit is 0 and the factor of all delays below the minimum limit is 1. The throughput factor b_j^i is also a factor between 0 and 1. It was calculated with formula (5)

$$b_j^i = \frac{b_i - b_j^{\min}}{b_j^{\max} - b_j^{\min}} , \tag{5}$$

where b_i is the measured throughput at iteration *i*, $b_j^{\max}$ is the maximum limit of throughput of class *j* and $b_j^{\min}$ is the minimum limit of throughput of class *j*. The factor of all throughputs above the maximum limit is 1 and the factor of all throughputs below the minimum limit is 0. The delay and throughput limits are presented in Table 16.

Traffic class	$d_{\min}$ (ms)	$d_{\max}$ (ms)	$b_{\min}$ (kbit/s)	$b_{\max}$ (kbit/s)
Conversational (1)	100	500	-	2400
Streaming (2)	250	1000	-	2200
Interactive (3)	-	-	-	2100
Background (4)	-	-	-	2000

Table 16 Delay and throughput limits for each traffic class

If there was no maximum and/or minimum limit defined for the delay or the throughput, the lowest result was the minimum limit and the highest result was the maximum limit. Coefficients δ_j and τ_j define how much weight are given to the delay and the throughput of a class *j* when calculating the total delay and the throughput factors. The coefficients of all the classes of different error rate scenarios are shown in Table 17.

With low error rates the weights on high priority class traffic characteristics are low, because when error rate is 0% or 20%, the delays and the throughputs of high priority classes are very satisfactory with all packet sizes. Thus, with these error scenarios low priority classes' traffic characteristics can be emphasized, especially throughput, because delay is not an important issue with best effort traffic classes.

As the error rate increases, high priority traffic class traffic becomes more important, and their weights are raised. When the radio channel packet error rate is as high as 40%, best effort traffic class traffic characteristics are given no weight. In addition, the delays of Conversational and Streaming classes are so low that the emphasis is more on classes' throughputs. When the error rate is 60%, the highest priority traffic class gets all the weights.

Error rate	0%		20%		40%		60%	
	δ	τ	δ	τ	δ	τ	δ	τ
Class 1	0	0.1	0.1	0.1	0.15	0.15	0.5	0.5
Class 2	0	0.1	0.1	0.2	0.3	0.4	0	0
Class 3	0	0.4	0	0.25	0	0	0	0
Class 4	0	0.4	0	0.25	0	0	0	0

Table 17 Weight coefficients of traffic classes in different error scenarios

Interesting issue arises from the results with the fact that the highest priority traffic class will get more bandwidth when the packet size and the channel error rate are increasing. With the proposed 3G traffic classification specifications this can lead to the situation that the highest class uses all the capacity under certain network conditions. Our simulation results proved this issue clearly. Based on that it is important to find out the limits where packet size and channel error rate can change. If we think for example the situation where we have small size high priority packets (voice traffic) under heavily loaded network i.e. channel error rate is high, we will lose also those small size high priority packets. In 0% and 20% error scenario, the best results are achieved when the highest-class packet size is about 1500 bytes, but if the error is very high (we tested with 60%), smaller packet size gives the best results.

5.2 DiffServ versus IntServ

Fig. 30 shows that the throughput in DiffServ case is slightly better than in RSVP case. That is expected due to the resource reservation nature of RSVP. In DiffServ case all traffic classes can have unlimited number of flows compared to RSVP's bandwidth limiting functionality and access control.

The difference between these techniques is almost negligible due to the fact that both RSVP and DiffServ achieve the maximum capacity of the network. This is due to the amount of traffic in the network: the flows are sending traffic so intensively that there is always a demand of the bandwidth for best effort traffic and hence the network is quite overloaded all the time.

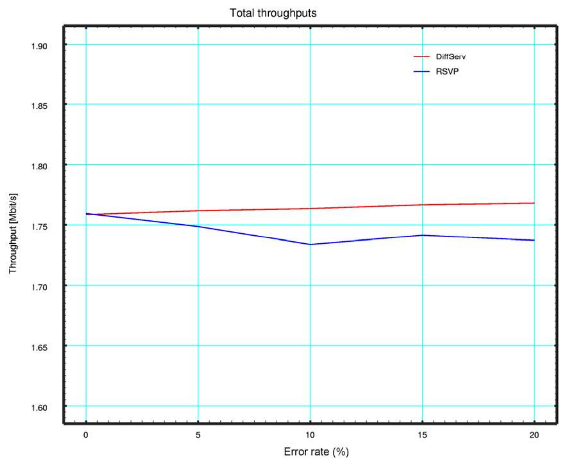

Fig. 30 Comparison of total throughputs with RSVP and DiffServ

6. CONCLUSIONS AND FUTURE WORK

6.1 Conclusions

In this article we provided architecture for end-to-end QoS control in a wired-wireless environment with effective QoS translation. We used DiffServ and RSVP architectures combined with 3G/WLAN interworking and IEEE 802.11e. As can be seen from the results, clearly RSVP can keep delays smaller than the DiffServ core. Results also show that the best and most suitable combination of the QoS control would be RSVP - IEEE 802.11e hybrid. Suitability materializes especially in the control of the lower priority flows enabling them more controllably with bandwidth and delay.

6.2 Future work

Next we will expand the simulations to cover large number of network nodes and study how the parameters can be tuned e.g. by using dynamic policy based management and tuning tools to find optimal operating point of network resources. Also further development of 3G interworking with other access methods is gaining increasingly importance and to achieve solid and robust interworking QoS is the next research challenge.

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