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Cross-layer Analysis of Transport Control Protocols over IEEE 802.11 Wireless Networks

Toni Janevski[1)] and Ivan Petrov[2)]
[1)]Ss. Cyril and Methodius University in Skopje, Faculty of Electrical Engineering and Information Technologies, Karpos 2 bb, 1000 Skopje, Macedonia,
Email: tonij@feit.ukim.edu.mk
[2)]Makedonski Telekom, Orce Nikolov bb, 1000 Skopje, Macedonia,
Email: Ivan.Petrov@telekom.mk

Abstract

In this paper we provide cross-layer analysis of the impact of mobility and Medium Access Control parameters in IEEE 802.11 wireless network on TCP. Performances of the Internet transport protocols may significantly degrade when end to end connection includes wireless links where packets delays and losses are caused by mobility and transmission errors. We perform analysis of the achievable throughput for different TCP versions, including TCP Tahoe, TCP Reno, TCP New Reno, TCP Vegas and TCP SACK, in IEEE 802.11 wireless networks. The analysis showed that the impact of Medium Access Control parameters, such as number of retransmissions and interface queue length in 802.11 networks on the obtained throughput, is stronger for terminals with higher mobility. Also, the analysis contributed to detection of the most appropriate TCP protocols for different mobility of the users in IEEE 802.11 networks.

Keywords: *TCP, Throughput, Transmission Protocols, Wireless Network.*

1. Introduction

The rapid development of Internet and wireless technologies resulted in their integration. In that manner all IEEE wireless networks are IP native, i.e. they define physical and Medium Access Control (MAC) layers, while the network layer is reserved for IP. On one side Internet is based on TCP/IP protocol suite targeted for usage by non-real-time applications (e.g. web, ftp, email etc.) and UDP for data from real-time applications (e.g. voice over IP, streaming, etc.). On the other side, the most successful IEEE wireless networks so far are IEEE 802.11 standards, which have been introduced in almost every device that requires wireless connectivity. Today, many devices have included 802.11 (i.e. WiFi) wireless interfaces in home, office and even cars and other transportation devices, also for connecting sensors etc. Then, we have IP protocol suite running over the 802.11 protocols on lower layers, and it is not difficult for one to see the importance of understanding the cross-layer relations in such scenarios, which is main subject in this paper.

TCP and UDP, which are part of the TCP/IP protocol suite, were carefully tuned in order to maximize their performance on wired networks where packet delays and losses are caused by congestion [1]-[5]. In the wireless networks, delays and losses are mainly caused by mobility handoffs and transmission errors due to bad wireless channel conditions. With the recent developments in mobile wireless networking, the performance of the Internet transport protocols in mobile wireless environment is becoming more important. We should mention that the protocols for wireless access have been designed in order to maximize the utilization of the wireless channel for web browsing and file downloading applications in an environment with restricted mobility, which is the main reason why the buffers and the local Medium Access Control (MAC) retransmissions are tuned in a way to maximize the throughput and the reliability for this kind of applications. In order to decrease packet delay the transport protocols used to deliver real time services and applications to the end user are simple and do not incorporate traffic control and packet retransmission mechanisms.

We focus our attention at the impact of the diverse MAC layer and buffer settings of IEEE 802.11 wireless access technology of the Internet native transport protocol suite during the distribution of multimedia applications in realistic static and mobile scenario.

The paper is organized as follows: Section II gives brief overview of the transport protocols, discusses some related work and motivates the need for our approach. It also briefly describes the 802.11 MAC protocol. Section III describes our simulation scenario and section IV presents the simulation results. Section V concludes the paper.

2. IEEE 802.11 and transport protocols

Experiments of TCP and UDP over IEEE802.11with different signal levels showed that without retransmission implemented at the link layer, loss rates become unacceptable for any application. It was also shown that the MAC layer retransmissions improve TCP performance. On the other hand, a high number of repeated retransmissions can cause TCP to timeout anyway and retransmit the same data as the MAC layer. Moreover, MAC retransmissions can be wasteful and potentially harmful for time-sensitive applications, such as real time video or audio over UDP. The 802.11 MAC layer protocol attempts to face the packet loss problem by implementing its own retransmission scheme. In particular, lost packets are retransmitted after a certain period of time without having received any corresponding ack. Successive retransmissions for the same packet are repeated up to a maximum number of time, which is by default set to 4 in the standard IEEE 802.11, or until receiving a successful ack. A backoff mechanism determines the retransmission timeouts.

This scheme hides wireless error losses from the TCP congestion control mechanism, thus avoiding deleterious multiple reductions of the data sending window. On the other hand, local retransmissions affect packet delivery delay by increasing its variability and thereby affecting time-constrained applications such as audio or video stream. The first TCP implementations were using cumulative positive acknowledgements and required a retransmission timer expiration to send a lost data during the transport. They were following the go-back-n model. In order to enable good user throughput and to control network congestion a lot of work has been done in order to improve its characteristics and with time TCP has evolved. Today's TCP implementations contain variety of algorithms that enables to control the network congestion and to maintain good user throughput in the wired network. Several variants of TCP can be found in the existing wired networks. TCP Tahoe, TCP Reno, TCP New Reno, TCP Vegas and TCP Sack are few of them that are going to be used in our simulation scenarios. The most used variant of TCP in the real world today is TCP New Reno. However, every one of these TCP variants has unique congestion and flow control mechanisms. A problem is defined in the coexistence of the TCP and UDP traffic in a given wireless channel, caused by TCP congestion control functionality. TCP continuously probes for higher transfer rates, eventually queuing packets in the buffer associated with the bottleneck of the connection. The wireless connection can be shared by several devices and applications. In such case it is obvious that the connection level and the queue lengths may increase, thus delaying the packet delivery and hence jeopardizing the requirements of the real-time applications. Such situation is even worse because the wireless medium allows transmission of only one packet at a time and in most of the wireless networks it is not full-duplex as in wired links [6]-[10]. This means that packets should wait their turns to be transmitted. Interference, errors, fading, and mobility are causing additional packet losses, and the IEEE 802.11 MAC layer reacts through local retransmissions which in turn cause subsequent packets to wait in the queue until the scheduled ones or their retransmissions eventually reach the receiver. The back off mechanism of the IEEE 802.11 introduces an increasing amount of time before attempting again a retransmission. In the past years there was a lot of research regarding the problems that TCP and UDP encounters in a wireless environment [11]-[15].

3. Simulation Scenario

The network layout of the simulation scenario that is subject of the analysis in this paper is presented in Figure 1. We have used the network simulator NS2 in order to simulate the outdoor environment presented in Figure 1. Network topology consists of four wired nodes (A0-A3), two wireless base stations (BS0-BS1) and four wireless nodes (n0-n3). The distance between two base stations is set at 20m. The wireless stations are configured to work according the IEEE 802.11g Standard. Wired connections are configured as given in Table 1. Maximum achievable bandwidth rate is set to 20Mbps, instead of the maximal 54Mbps for IEEE 802.11g standard, due to a home environment.

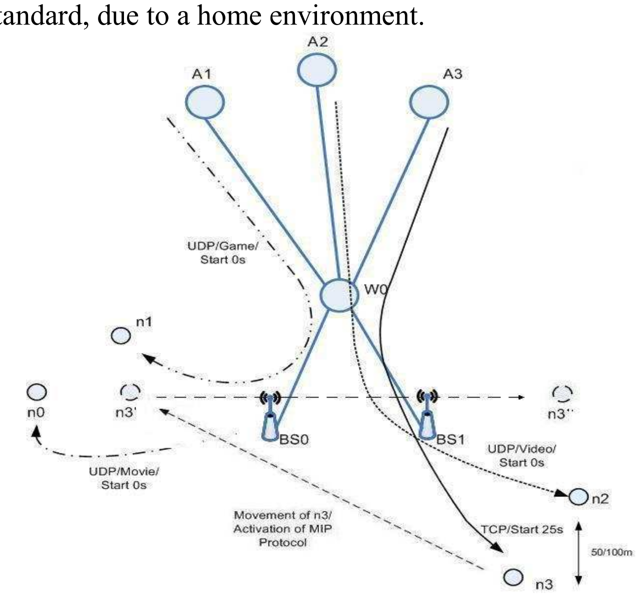

Figure 1 First simulation scenario

The queue size value used in the simulation is calculated by multiplying the longest RTT (Round Trip Time) with the smallest link capacity on the path, which is the 20Mbps throughput effectively available over the wireless link. In Table 2 are presented several applications that are used during the simulation. In the simulation we have used real trace files for video chat and movie traffic. Two VBR H.263 Lecture Room-Cam are used for the Video chat and high quality MPEG4 Star Wars IV trace file is used for the movie [15]. In this simulation the game events have been generated at the client side every 60ms [11].

At the server side updates were transmitted every 50ms towards the client. The payload generated by the client has been set to 42Bytes and the payload generated by the server has been set to 200Bytes. The rest of the packets were set to standard value of 512Bytes for TCP segments. The values for different parameters used in this scenario are listed in Table 3. For the simulation we have used the shadowing model. The shadowing deviation (σ_{dB}) was set to 4 while the path loss exponent (β) was set to 2.7. These parameters are common for urban environment.

Table 1 Configuration of wired links simulated at scenario.

Node 1	Node 2	Delay	Capacity
A1	A0	10ms	100 Mbps
A2	A0	20ms	100 Mbps
A3	A0	30ms	100 Mbps
A0	BS0	10ms	100 Mbps
A0	BS1	10ms	100 Mbps

Table 2 Types of applications and traffic simulated in the presented scenario.

From	To	Type	Transport Protocol	Start	End
BS0	n0	Movie Stream	UDP	0s	110s
A1	n1	Game Traffic	UDP	10s	110s
n1	A1	Game Traffic	UDP	10.1s	110s
A2	N2	Video Chat	UDP	15s	110s
N2	A2	Video Chat	UDP	15.1s	110s
A3	N3	FTP	TCP	35s	110s

Table 3 Simulation parameters.

Parameter	Values	Comments
MAC data retransmissions	1, 2, 3, 4	Default value is set at 4
User-BS distance (m)	5, 10, 50, 100	Common indoor environment
MAC queue pkt. size	25,50,100 pkt	Common values
Velocity (km/h)	2; 4; 10; 15; 25; 50	Random choice
TCP Transport protocol	TCP Tahoe, TCP Reno, TCP Newreno, TCP Vegas, TCP Sack	Commonly used types of TCP protocols in wired networks.

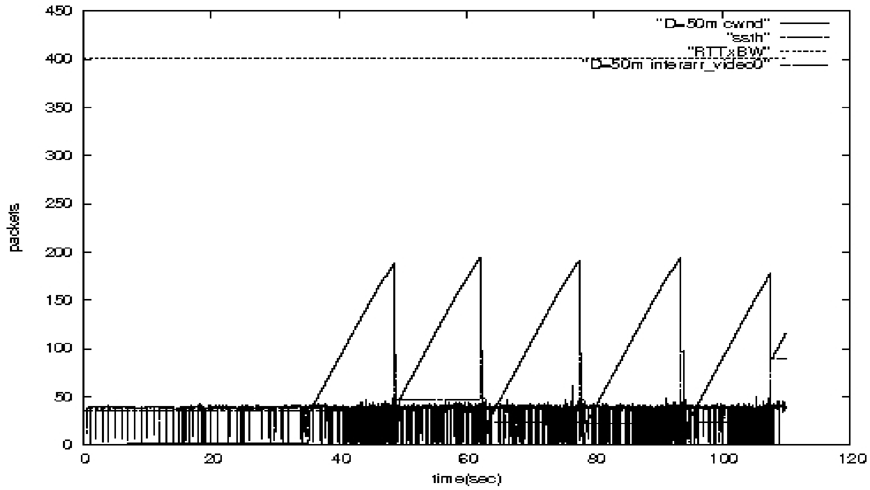

Figure 2 Interarrivals of video packets as a function of the change of TCP congestion window (cwnd)

4. Analysis of transport control protocols

In the following part we will observe simulation results from scenario presented in Figure 1, obtained by using the configuration parameters of the links given in Table 1 and applications defined in Table 2. We study the behavior of the UDP and TCP applications and the TCP impact of the real time applications in the defined network when the node n3 is positioned at 50/100m away from BS1. The rest of the variable parameters are set as follows: MAC data retransmissions are set at 4 and the MAC queue size is set at 50 pkts. As TCP transport protocol is used TCP NewReno. In Figure 2 is presented the changing of the cwnd and the iterarrival of the Video traffic packets when n3 is positioned 50m away from BS1. From these results we can conclude that the TCP application directly impacts the delay and jitter of the packets of the real time applications. In this scenario there is FTP traffic which is saturating the channel and the queues along the path, something that is evident from the significant delays and jitter variation of the real time traffic.

The queue size at the MAC layer and the number of MAC layer retransmissions impact the TCP throughput. In Figure 3 we show analyses of the throughput of an FTP application as a function of the distance, queue size and the number of MAC layer retransmissions. The results are shown for TCP New Reno. From the results one may conclude that the queue size of the MAC layer does not impacts the throughput for a given value of the number of MAC layer retransmissions (the curves for different queue sizes and some other parameters are overlapping). Hence, if we increase the number of the MAC layer retransmissions we shall obtain better throughput. The best throughput in this case is obtained when the number of MAC layer retransmissions is set to 4 retransmissions. One may notice that the queue size (i.e. IFQ) at the MAC layer drastically impacts the throughput. It is obvious that if we increase the queue size we will obtain better throughput. It is also obvious that the same throughput is achieved for given values of the MAC layer queue size when the number of the MAC layer retransmissions is set to 3 and 4 retransmissions. The best throughput is achieved when Interface Queue (IFQ) has value of 100 packets. So far now we are able to conclude that the throughput for all of these transport protocols (i.e. TCP versions) is decreasing as a function of the distance and is increasing as a function of IFQ buffer size and the number of MAC layer retransmission. Drastically lower throughput is achieved at longer distances (i.e.

100m) for all used transport protocols. The queue size at distance of 100m does not impact the throughput as it is a case when the distance is shorter (50m). Highest throughput is achieved for up to four MAC retransmissions.

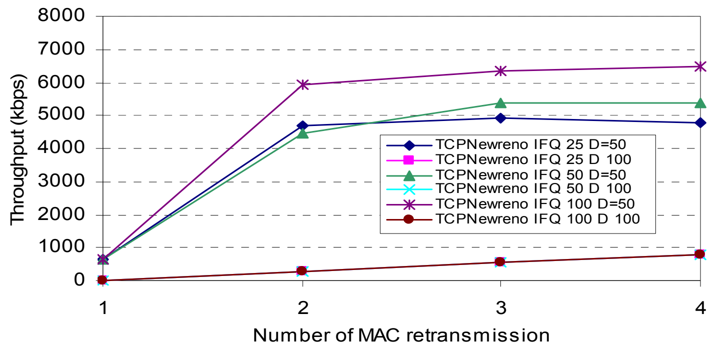

Figure. 3. TCP NewReno throughput for different access point distances; different MAC queue sizes.

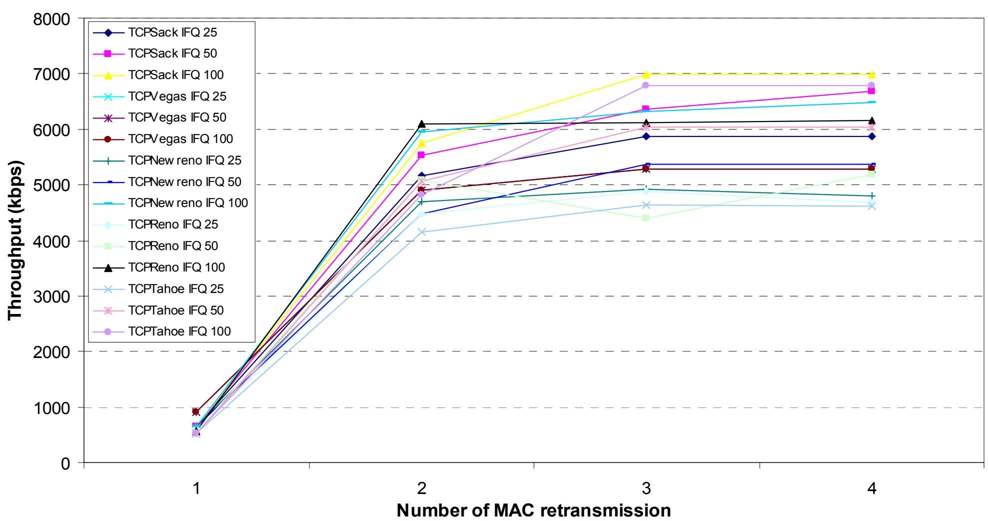

Figure. 4. Throughput of variety TCP protocols for different MAC queue sizes and different number of MAC retransmissions; Distance between n3 and the AP, BS1 is 50m.

Further, TCP New Reno has slightly better performance than TCP Reno. The overall worst performances are achieved with TCP Vegas excluding the case when the MAC queue size has value of 25 packets (in such case the throughput achieved with TCP Vegas is the second best, after the one achieved when TCP SACK is used as a transport protocol).

In Figure 5 we present the throughput achieved when the distance between wireless terminal and base stations is 100m. In this scenario, one may conclude that when are needed four MAC retransmission to achieve maximum performances regarding the wireless network and the TCP version.

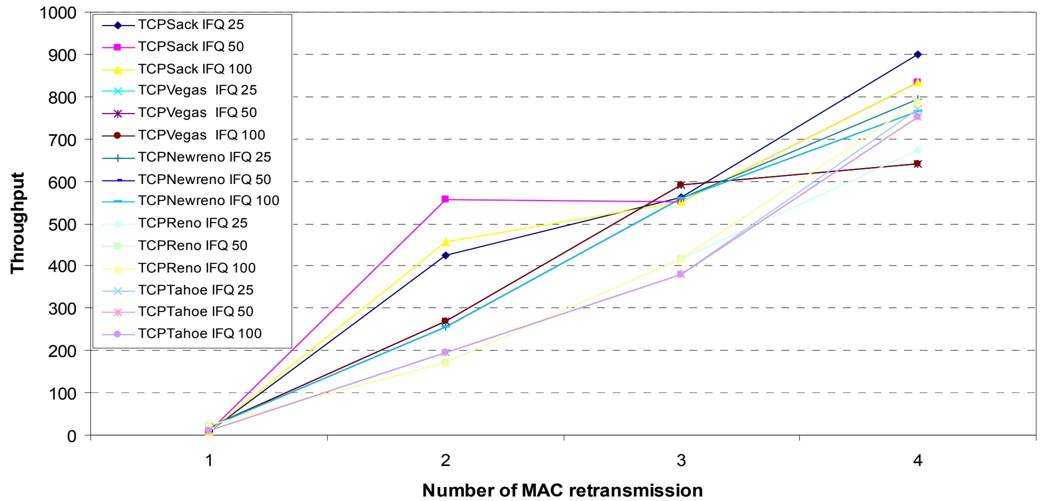

Figure. 5. Throughput of variety TCP protocols for different MAC queue sizes and different number of MAC retransmissions; Distance between n3 and the AP, BS1 is100m.

Longer the distance between the BS and the terminals means more MAC retransmissions to achieve the maximum performance in the 802.11 wireless networks. Again, best throughput is achieved with TCP Sack. The second best is achieved with TCP Reno except when the IFQ size is 25 packets (in such case TCP New Reno and TCP Tahoe show better throughput performances). The worst throughput is achieved with TCP Vegas for average of four MAC retransmissions. On the other side, when we use three MAC retransmissions the best throughput is achieved with TCP Vegas, while in such case the worst performance is achieved with TCP Tahoe. In the following part of this section we provide analysis of the UDP traffic (generated by game application) in presence of background TCP traffic (generated by an FTP flow). If we analyze the results presented in Figure 6 we will notice that the average throughput of the game traffic between the nodes A1-n1, when FTP flow is enabled, has constant value for different TCP versions when the number of the MAC layer retransmissions is bigger than two, for all possible scenarios. The results of the average packet delay for game traffic (UDP traffic) when different TCP versions are used for the background TCP traffic (i.e. the FTP flow) are shown in Figure 7.

Table 4. Average delay of the game traffic (A1-n1) in ms when the MAC retransmissions are set at value of three and the node n3 is 50m away from BS1.

L=3	IFQ (pkts)	25	50	100
D=50m	TCP Tahoe	21.1711	21.6210	21.9826
	TCP Reno	21.2408	21.2902	21.9826
	TCP NewReno	21.2953	21.4861	21.7785
	TCP Vegas	21.0952	21.0952	21.0952
	TCP Sack	21.4726	21.9197	21.9386

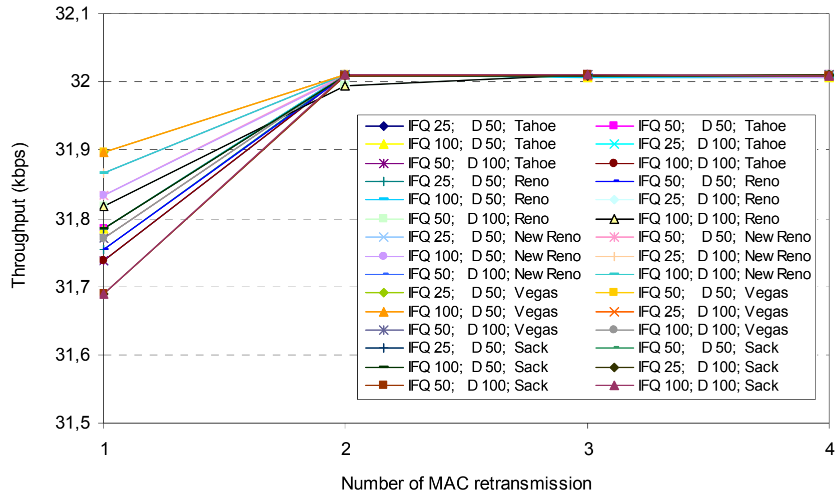

Figure. 6. Comparation of average throughput of the game traffic between the nodes A1-n1 when FTP flow is enabled for different TCP protocols.

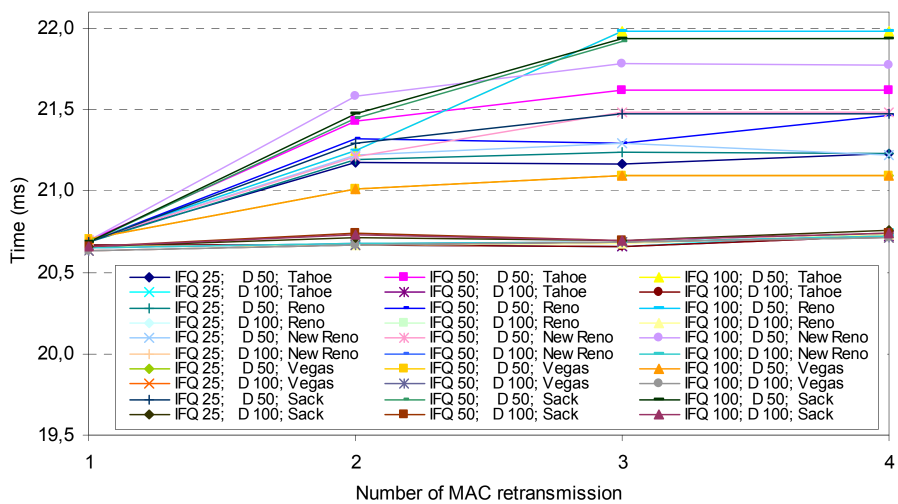

Figure. 7. Comparation of average delay of the game traffic between the nodes A1-n1 when FTP flow is enabled for different TCP protocols.

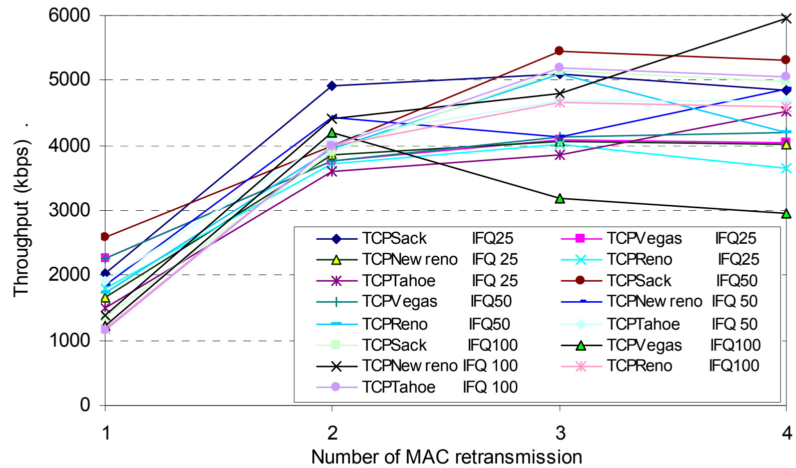

Figure 8. Average Throughput of the FTP traffic when the node n3 is moving toward BS0 with speed V=4m/s

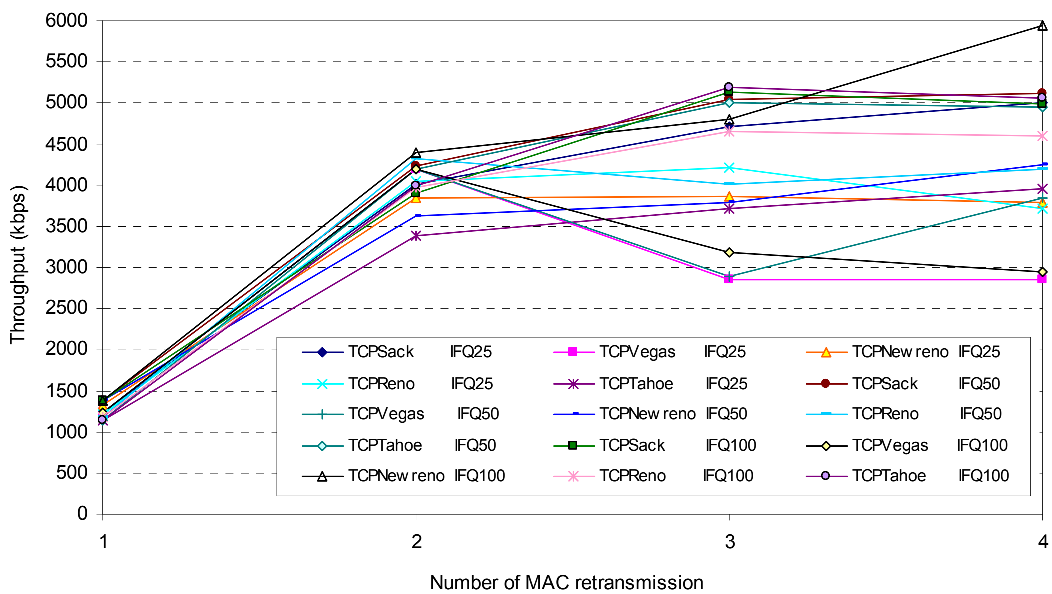

Figure 9. Average Throughput of the FTP traffic when the node n3 is moving toward BS0 with speed V=7m/s

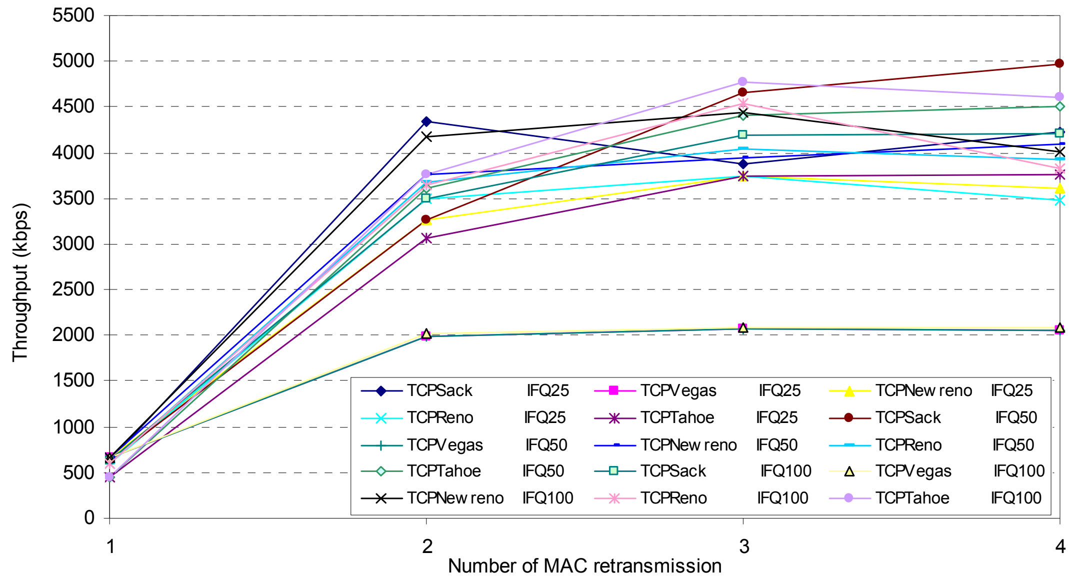

Figure 10. Average Throughput of the FTP traffic when the node n3 is moving toward BS0 with speed V=14m/s

Numerical results for the packet delay of the game traffic are given in Table 4. Lower delays are obtained for larger IFQ buffers and vice versa. The optimal number of MAC retransmissions for smaller IFQ buffers regarding all TCP versions is three, while two retransmissions are good choice for larger buffers, which is similar to conclusions regarding the throughput of the UDP game flow (Figs. 6 and 7). Hence, for UDP game traffic, the optimal number of MAC retransmission is two, which is independent from the distance (i.e. the same results are obtained for different distances between the wireless node and the AP). After we have finished the simulations when the node has static position we have conducted the same analysis of the traffic when the node n3 is mobile. We will study the behavior of the TCP applications and the TCP impact of the real time applications in the defined network when node n3 is mobile. During this analysis we should notice that n3 begins to move 55s after simulation starts. Like we have stated in Table 3 the mobile node n3 is moving with three different velocities towards n3'. We will observe scenarios when n3 is moving with speed of 15km/h, 25km/h and 50km/h. At Figs. 8, 9 and 10 is presented the change of the average throughput of the TCP traffic achieved in the simulation environment accordantly. From these figures we can evident that if we increase the number of MAC retransmissions the average throughput will increase too. Best throughput values are achieved when the number of MAC retransmissions in set at value of three. From Figs. 8, 9 and 10 we can notice that best performance has been achieved when IFQ is set at value of 50 pkts and TCP SACK is used as a transport protocol. We can evident that for smaller speed we are achieving better results when we use queue with size of 50 pkts. This is the case when n3 is moving with speed of 15km/h and 25km/h.

In Figure 11 we have compared the behavior of the average throughput as a function of the MAC queue size when n3 is moving with the defined speed and MAC data retransmissions are set at three. It becomes obvious that the throughput of the flow decreases with increasing of the speed of moving. As TCP transport protocol is used TCP SACK. It is also obvious that we are achieving best results if we set the IFQ buffer size at 50 pkts. Shortly we can resume that in this kind of scenario we will achieve best throughput if we use TCP SACK as a transport protocol, we set the number of MAC retransmissions at three and the IFQ buffer size at 50 pkts.

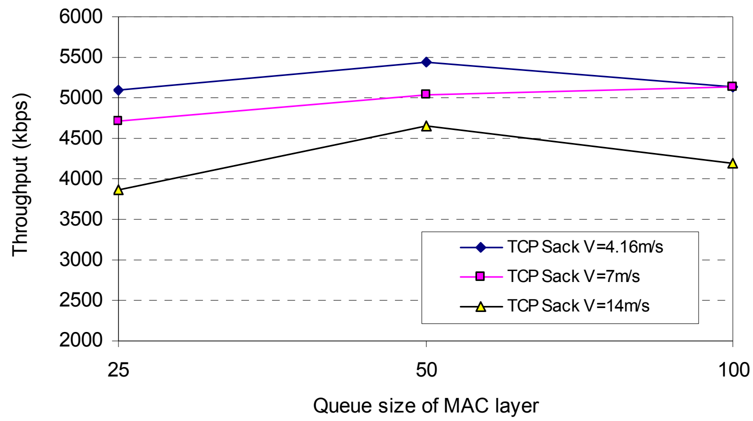

Figure 11. Average throughput of the FTP flow when the number of MAC retransmissions is set at three and TCP SACK is used as a transport protocol.

In the following part we observe simulation results from scenario presented in Figure 1, obtained by using configuration parameters of the links given in Table 1 and applications defined in Table 2. Like in the previous case we continue to analyze the behavior of the UDP/TCP applications and the TCP impact on real time applications in 802.11 wireless networks regarding the throughput as the most important performance metric for non-real-time flows (which use the TCP on transport layer). In this situation the distance between the base stations is set at 20m and n3' is moving towards n3'' with lower speed.

The queue size at the MAC layer and the number of MAC layer retransmissions in the given situation impact the TCP throughput. In Figure 12 we show analyses of the throughput of an FTP application as a function of the distance, queue size and the number of MAC layer retransmissions. From the results one may conclude that the queue size of the MAC layer impacts the throughput for a given value of the number of MAC layer retransmissions. If we increase the number of the MAC layer retransmissions in this situation we are not obtaining better throughput. One may notice that the queue size (i.e. IFQ) at the MAC layer drastically impacts the throughput. It is obvious that if we increase the queue size we will obtain better throughput. It is also obvious that the same throughput is achieved for given values of the MAC layer queue size for given MAC layer retransmissions. The best throughput is achieved when Interface Queue (IFQ) has value of 100 packets.

So far now we are able to conclude that the throughput for all of these transport protocols (i.e. TCP versions) is decreasing as a function of the distance and is increasing as a function of IFQ buffer size. Because we are analyzing relatively small distances and defined simulation scenario where we have strong signal coverage and small attenuation, errors caused by the channel condition are small and the MAC layer have no need to activate its retransmission mechanism in order to improve the throughput.

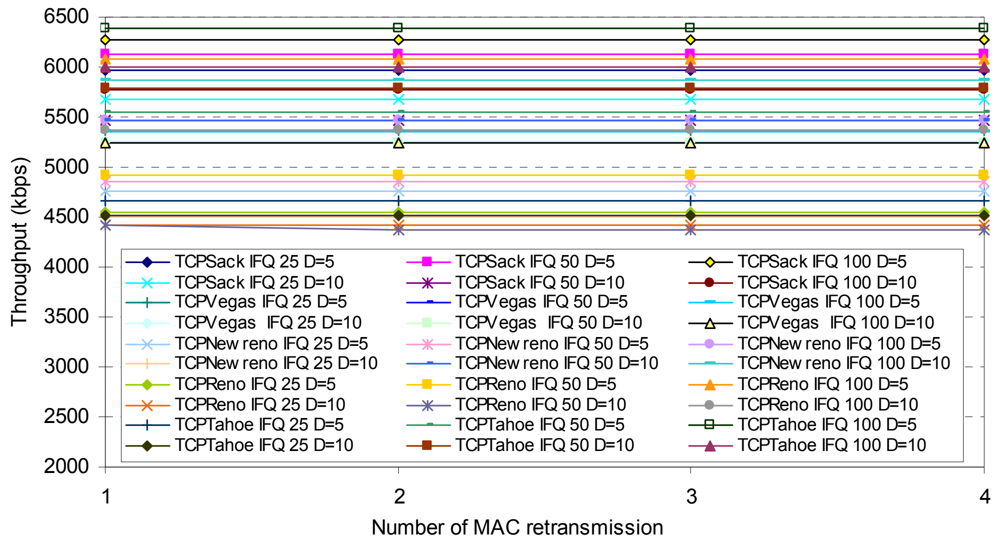

Figure 12. Throughput of variety TCP protocols for different MAC queue sizes and different number of MAC retransmissions; Distance between n3' and the AP, BS1 is 5/10m.

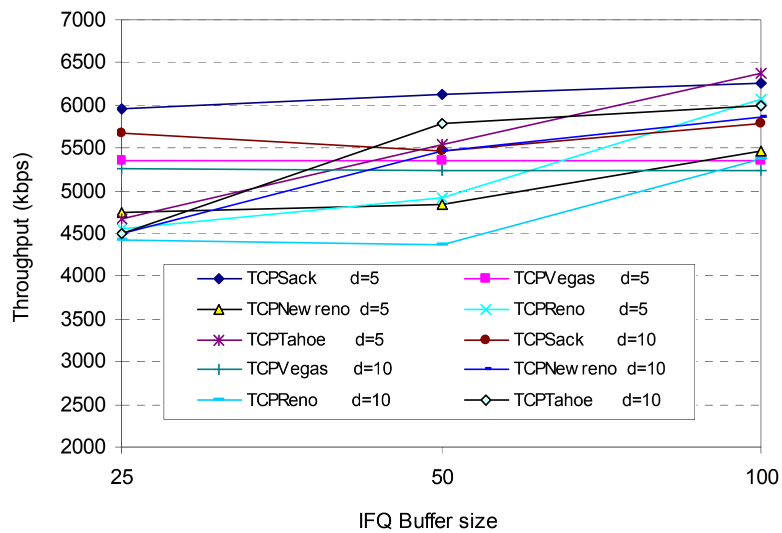

Figure 13. Throughput of variety TCP protocols for different MAC queue sizes when the number of MAC retransmissions is set at 3; Distance between n3 and the AP, BS1 is set at 5/10m.

At this point one may conclude that only IFQ buffers size strongly influence the throughput. Furthermore, the best throughput is achieved with TCP SACK as a transport protocol (d=5m). TCP Tahoe (IFQ=50;d=10m) is the second best. When IFQ receives value of 100 packets, then TCP Tahoe outperforms TCP SACK. On the other side, TCP Tahoe performs very poor for small IFQ values, i.e. it shows the almost worst performances for IFQ=25 (the smallest IFQ value in our analysis) when compared with all other cases in Figure 13. Further, TCP New Reno has slightly better performance than TCP Reno. TCP Vegas is on third place excluding the case when IFQ receives value of 25 pkts when it is the second best protocol. TCP Reno has better performances than TCP NewReno when IFQ is set at value of 100 pkts which is not the case for smaller size of the IFQ buffer. When the distance between n3 and BS1 is set at 10m, performances of the transport protocols are different from the previous. In this case best performance is achieved with TCP Tahoe when the IFQ buffer size set at 50/100pkts (it slightly outperforms TCP Sack) excluding the case when the IFQ buffer is set at 25 (in such case the throughput achieved with TCP Vegas has the same value with the one achieved with TCP NewReno and TCP Reno).

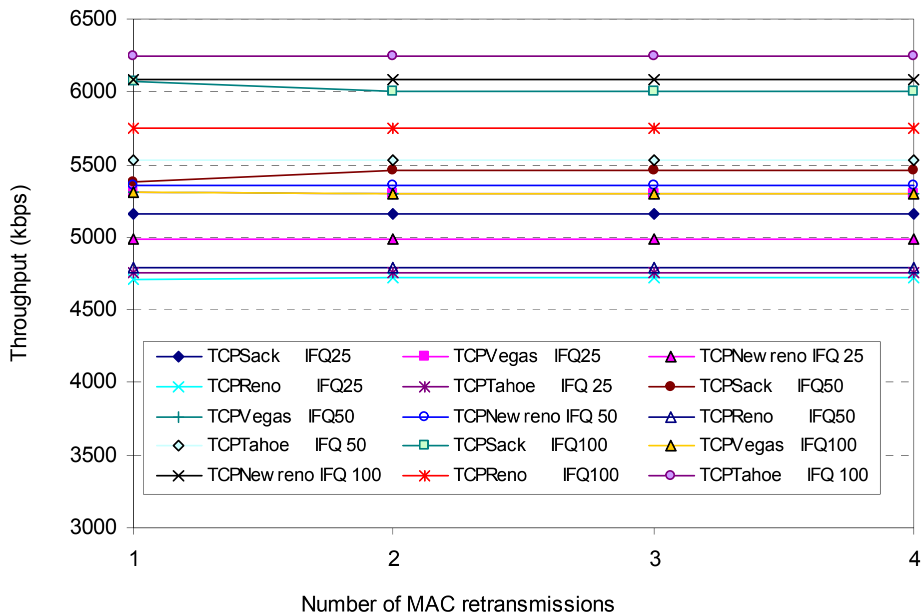

Figure 14. Comparation of average throughput when FTP flow is enabled with different TCP protocols when n3 is moving towards BS0.

The performances of TCP SACK and TCP NewReno are overleaping (IFQ=50/100) except when IFQ receives value of 25pkts when TCP SACK outperforms TCP NewReno. The overall worst performances are achieved with TCP Reno. From Figure 13 one may conclude that TCP SACK outperforms TCP Tahoe as well as other TCP implementations for the case of indoor environment (smaller distances between the terminal and the access point). If we analyze the rest of the parameters, such as average delay and jitter, one may conclude that the number of MAC retransmissions for the specific scenario doesn't influence the throughput while the IFQ buffer size impacts all of the mentioned parameters. According to the results, in order to make the right choice of the IFQ size without worsening the performances of the UDP applications, we should set the IFQ buffer size at value of 50. In Table 5 are presented numerical results regarding average packet delay of a TCP flow for different TCP versions and different IEEE 802.11 queue sizes (i.e. IFQ values), while the number of MAC retransmission is set to three (according to the previous discussions in this paper).

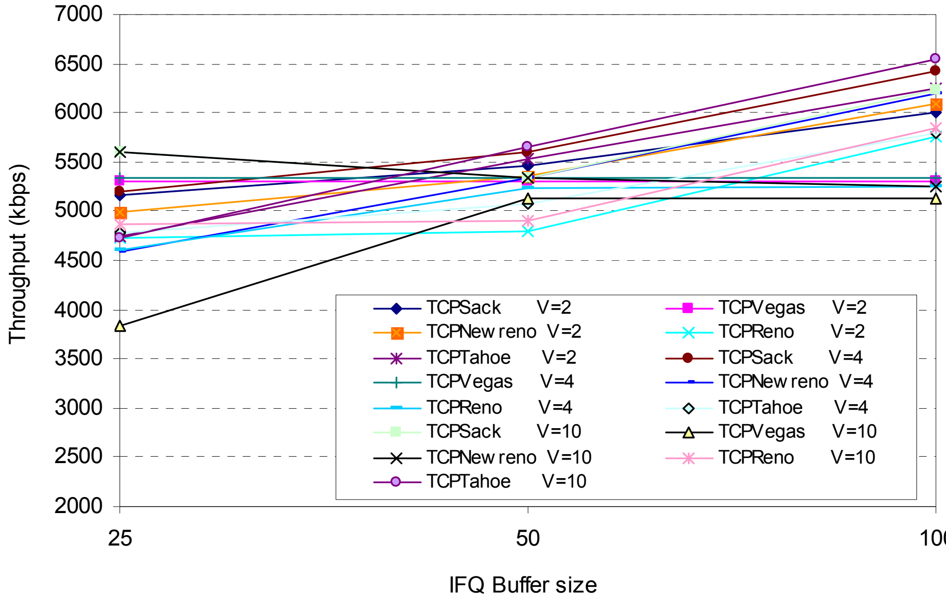

Figure 15. Average throughput of the FTP flow when the number of MAC retransmissions is set at three, the traffic is provided with different transport protocols and n3 is moving toward BS1 with speed of V=2/4/10 kmph.

After we have performed detailed analysis of the throughput in the simulation scenario when the nodes are static we have performed the same analysis for mobile scenario. For that purpose we have incorporated Mobile IPv4 protocol to handle the user mobility. In the first case we were analyzing static scenario when n3 was distanced from BS0 at 5 and 10m. In this case we analyze scenario when n3 is in the radio coverage area of the BS1 and performs handoff from BS0 to BS1. In Figure 14 we present the average throughput for different TCP version in home environment with low mobility (2km/h). The results show that the number of MAC retransmissions does not impact the achieved throughput, which is again due to lower losses in the wireless channel because of smaller distance between the access point (i.e. BS) and the terminal as well as due to low mobility. However, the MAC queue size influences the available throughput and it is different for different TCP versions (Figure 14 and Figure 15). Bigger queue size (i.e. buffer) leads to higher throughput, and the best results are obtained for this scenario with TCP Tahoe (IFQ = 100 packet).

However, although the throughput is the most valuable performance metric for non-real-time flows, the delay is also important. In Table 5 we provide results of the average packet delay regarding the given scenario where the delay budget includes mainly the wireless link delay of the IEEE 802.11 network (of course, including buffering). Overall average packet delay, using different queue sizes, has lowest value for TCP SACK (d=10m) and TCP NewReno (d=10m), while highest average delay is obtained for TCP NewReno (d=5m). For instance, we note that TCP NewReno appears for the lowest and for the highest overall average packet delays (Table 5) and this is due to the higher values for average delay when we are using longer IFQ queue sizes. This is leads to a conclusion that TCP NewReno, which is the most used today is very sensitive to the buffer length, which affects its performances such as average delay. Regarding the average delay results shown in Table 5 leads to a conclusion that TCP SACK provides overall best results regarding lower packet delay in the IEEE 802.11 network in indoor environment.

Table 5. Average delay (ms) of TCP flows for different IFQ buffer sizes (MAC retransmissions are set at three).

L=3	d (m)	Delay (ms) IFQ=25	Delay (ms) IFQ=50	Delay (ms) IFQ=100	Overall average (ms)
TCPSack	5	52,04	53,48	62,16	55,89
TCPVegas		60,67	60,67	60,67	60,67
TCPNewreno		61,39	58,37	66,65	62,14
TCPReno		61,44	53,51	64,88	59,94
TCPTahoe		53,67	54,58	59,84	56,03
TCPSack	10	47,25	52,31	53,56	51,04
TCPVegas		55,05	55,08	55,08	55,07
TCPNewreno		45,44	47,27	60,62	51,11
TCPReno		55,66	52,77	59,63	56,02
TCPTahoe		54,17	49,66	64,84	56,22

Table 6. Average throughput of TCP traffic for different terminal velocities (MAC retransmission is set at value of three and the IFQ buffer has value of 50pkts).

L=3; IFQ50	v=2km/h	v=4km/h	v=10km/h
1	TCP Tahoe	TCP Sack	TCP Tahoe
2	TCP Sack	TCP Vegas	TCP Sack
3	TCP NewReno	TCP NewReno	TCP NewReno
4	TCP Vegas	TCP Reno	TCP Vegas
5	TCP Reno	TCP Tahoe	TCP Reno

We may also summarize TCP protocols regarding the throughput as the most importance performance metric for non-real-time flows, which indeed are using the TCP as transport protocol. So, in Table 6 we have ordered the transport protocols according to the showed performance for different wireless terminal velocities (when IFQ is set at 50pkts and we use three IEEE MAC retransmissions). From Figure 15 and Table 6 one may conclude that best performances are obtained with TCP Tahoe, which is followed by TCP SACK on the second place, and the bronze goes to TCP NewReno.

We can summarize that for the given indoor scenario the number of MAC layer retransmissions has no impact on the throughput. In the outdoor scenario we have set the MAC layer retransmissions at three because in that case we have achieved best throughput. This is the main reason why we have adopted this value for the indoor scenario. Analyzes has shown that in that case we will obtain best throughput when IFQ receives value of 50 pkts. From Figure 15 one may conclude that if we increase the IFQ buffer size we will achieve better throughput for TCP i.e. non-real-time traffic. This analysis leads to necessity of development of an open transport protocol layer for IEEE 802.11 environment in home or office as well as in outdoor environment, so the best transport protocol will be chosen to run in every possible scenario.

5. Conclusion

In this paper we have performed detailed traffic analyses regarding the performances of different TCP versions in 802.11 wireless networks. We have compared different transport protocols by using the throughput as a merit. The results showed the high importance of the Medium Access Control (MAC) parameters in 802.11 wireless networks regarding the throughput. From the given results of the analyses we are able to conclude that distance between nodes directly impacts the TCP traffic flow and its throughput. The throughput decreases as a function of the distance between the nodes in the wireless environment. The number of the MAC layer retransmissions for small distances has no influence on the TCP and UDP traffic. MAC layer queue size i.e. IFQ directly affects the traffic flows in the wireless environment especially for shorter distances between the wireless nodes so it has to be carefully tuned in order to be achieved higher traffic flows. The choice of the TCP transport protocol affects the throughput in the network.

These simulations have shown that overall best performances are obtained in all scenarios if we set the maximum number of MAC layer retransmissions at value of three, because we have shown that at bigger distances obtained throughput is the same when the number of MAC retransmissions increases to higher values than three. Then the optimal MAC queue size is 50 packets (in order to optimize the performances of the UDP applications). Mobility directly impacts the throughput. The throughput decreases as a function of the terminal velocity.

Best results regarding the throughput are achieved with TCP SACK in static and with TCP Tahoe in mobile environment. However, the results were different when we have analyzed the average packet delay as the performance metric where TCP SACK has shown the best overall performance. However, the throughput is the most important metric for the non-real-time traffic, which in fact uses TCP on the transport layer.

Finally, with analysis in this paper we have contributed to the detection of the TCP behavior in IEEE 802.11 wireless networks, for different settings of the IEEE 802.11 MAC layer (regarding the queue size and MAC retransmissions) and different mobility of the terminals (with implemented Mobile IP protocol for handling the terminal mobility). This cross-layer analysis, by using the protocols on layer 2 and layer 4 of the OSI protocol model, lead to possibility and necessity for creation of an open transport protocol layer, especially for the case of IEEE 802.11 networks in the home or in the office, as well as outdoor wireless networks. Hence, our future work is targeted to solutions for open transport layer protocols in future wireless terminals, which is a target for wireless local networks, but it is not limited to them in the continuously evolving wireless world.

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Authors

Toni Janevski is a Professor at the Faculty of Electrical Engineering and Information Technologies, Ss. Cyril and Methodius University in Skopje, Macedonia. He received his Dipl. Ing., M.Sc. and Ph.D. degrees from Faculty of Electrical Engineering, Ss. Cyril and Methodius University in Skopje, in 1996, 1999 and 2001, respectively. In the past, during 1996-1999 he has worked at the Macedonian GSM 900 mobile operator Mobimak on the cell planning and dimensioning of cellular network, as well as traffic and performance analyses. From 1999 he is with Faculty of Electrical Engineering and Information Technologies in Skopje. In 2001 he has conducted research in optical communication at IBM T. J. Watson Research Center, New York. He is an author of a book titled "Traffic Analysis and Design of Wireless IP Networks", which is published in 2003 by Artech House Inc, Boston, USA. Also, he has published over 100 scientific papers. His research interests include Mobile, Wireless and Multimedia networks, Internet technologies, Information Technology, Traffic Theory, Quality of Service, as well as Dimensioning, Planning and Optimization of Telecommunication Networks. He is a Senior Member of IEEE.

Ivan Petrov is with telecom operator Makedonski Telekom, Skopje, Macedonia. He received his Dipl.Ing. and M.Sc. from the Faculty of Electrical Engineering, Ss. Cyril and Methodius University in Skopje, in 2005 and 2009, respectively. From 2009 he is working towards his Ph.D. degree under supervision of Professor Toni Janevski. His research interests include transport protocols and cross-layer optimization techniques in heterogeneous wireless IP networks.