Performance analysis and experimentation of heterogeneous IP networks

Giulio Iannello [a,b], Antonio Pescapè [b,c,*], Giorgio Ventre [b,c] and Luca Vollero [b]

[a] *Università Campus Bio-Medico di Roma*

[b] *Università di Napoli "Federico II", Dipartimento di Informatica e Sistemistica*

[c] *ITEM -Laboratorio Nazionale per l'Informatica e la Telematica Multimediali*

Abstract

Packet loss, delay and jitter degrade the quality of services like VoIP (Voice over IP) or Video Streaming over IP networks. In real networks, an experimental measure of these parameters is fundamental in the planning process of new services over novel network infrastructures. Furthermore, currently networks are heterogeneous in terms of *access network technologies*, *end-users' devices*, *Operating Systems* and finally *end-users' application*. This heterogeneity exacerbates even more the need of a real assessment of Quality of Service metrics. In this work we provide an empirical performance study of a real heterogeneous network with respect to delay, jitter, throughput and packet loss, in UDP and TCP environments, by using an innovative tool for network performance evaluation that we called D-ITG (*Distributed Internet Traffic Generator*). We also introduce the concept of "*Service Condition*" as a mechanism to cope with issue related to dynamically changing network service scenarios. A comparative analysis between our practical results and an analytical model recently presented in literature is presented. Results presented in this paper can be used as performance references for development of wireless communication applications over multiservice heterogeneous networks.

Key words: Heterogeneous Wireless Networks, Performance Analysis, QoS parameters.

* Corresponding Author: Antonio Pescapè, e-mail: pescape@unina.it. Via Claudio 21, 80125, Napoli, ITALY; Phone:+39-081-7683825; Fax:+39-081-7683816.

Email addresses: `iannello@unina.it` (Giulio Iannello), `pescape@unina.it` (Antonio Pescapè), `giorgio@unina.it` (Giorgio Ventre), `vollero@unina.it` (Luca Vollero).

Preprint submitted to Elsevier Science *21 July 2004*

1 Introduction

In the last years network capacity has increased at a dramatic rate. At the same time the proliferation of the web has resulted in an exponential increase in the number of "*surfing users*" supported by the Internet. These users are becoming increasingly sophisticated and demand high-bandwidth, low-delay network services at affordable prices. Currently, this services' request is made over new "*heterogeneous, integrated and mobile*" IP (*Internet Protocol*) networks. Always-on connectivity, location-awareness, and environment-aware products are some of the new paradigms over heterogeneous wireless networks. Smart devices, portable devices, wireless communications appear to be the underlying principles of a new revolution in technology. Pervasive computing deals with a wide range of information access methods enabled by mobility, wireless technologies, small embedded systems and broadband technologies [1]. Integration of fixed and wireless access to IP networks presents a cost effective and efficient way to provide seamless end-to-end connectivity and ubiquitous access in a market where demands of mobile Internet have grown rapidly and predicted to generate billions of dollars in revenue. In this environment, among the many factors that determine the feasibility of a given network scenario for the given set of application requirements, there is network performance. Network performance is generally affected by different aspects at the physical, data link, network, and transport layers. In a generic real network and in particular in a heterogeneous scenario, it is extremely difficult (i) to define a general framework for empirical performance evaluation and (ii) to determine the causes of the experimented performance. This paper focuses on the area of performance evaluation of heterogeneous wireless networks from the application level point of view. First, we introduce a network performance methodology dividing our experimentation on several traffic classes. Second, we measure TCP and UDP performance in more than network scenario where there is interoperability among different network technologies, different end-user devices, different operating systems and finally different user application with different QoS (*Quality of Service*) traffic requirements. The performance evaluation study has been performed following the indication of IP Performance Metrics (IPPM) IETF Working Group [12]. The network behavior has been studied introducing an innovative synthetic traffic generator that we called D-ITG (*Distributed Internet Traffic Generator*) [11] and which provides a set of powerful tools for traffic patterns generation over heterogeneous wireless networks and results analysis. We present our experimental results and at the same time we analyze and compare our results with respect to theoretical assumptions on wireless performance behavior carried out in [2]. Finally, we present a clear definition of which system's elements are responsible of network performances degradation and how the used different protocols impact on the observed network performance.

The paper is organized in 7 sections. After this introduction, in the next Section the "*Service Condition*" concept is presented, while in Section 3 motivations and the reference framework on which our work is based are presented. In this section we present some related works that compare and contrast our work and we emphasize our contribution. The experimental setup where our work has been carried out is presented in Section 4, discussing the main issues related to our heterogeneous scenario and describing the measuring procedure. Some functionalities and main concepts regarding the D-ITG platform are shown. Section 5 reports the obtained experimental results with respect to throughput, delay, jitter and packet loss. As far as achieved throughput, in Section 6 we provide a summary of our results and we compare and comment our conclusions in the framework of Bianchi model. Finally, Section 7 provides some concluding remarks and issues for future research.

2 Service Condition

In a new pervasive and ubiquitous scenario, several questions arise when we want to describe the impact of the several "*bricks*" of a such scenario over the performance. We can summarize these factors in the concept of "heterogeneity": end-user device, access network technology, operating system and end-user application heterogeneity.

(1) *End-user device heterogeneity*: we need to know what are device characteristics that will be used to send/receive the media. These devices can range from high-performance workstations, to PDA, down to smartphones with limited video reproduction capabilities. It is reasonable to expect that future services allow the same user of using a wide collection of terminals and of freely moving from one terminal to another. Of course, this information must be managed to have the content delivered to the user with the format most suitable to the device currently adopted.

(2) *Network heterogeneity*: we need to know the characteristics of the network that will be used to deliver the content, since also this one is a critical factor for the correct definition of a SLA. In the current Internet, even if we consider as dynamically variable only the part of a network infrastructure that is closest to the user (i.e. the so called access network or edge network), we have a quite large number of option to deal with: wireline (*Ethernet, Fast Ethernet, Giga Ethernet, xDSL, ...*), wireless (*WLANs, Bluetooth, ...*), 2.5/3/4 G mobile networks (*GSM, GPRS, UMTS*).

(3) *End-user application heterogeneity*: we need to know the characteristics of the service itself, in terms of media involved (audio, video, graphics), of their format (coding and compression techniques) and in terms of the typology of the service (synchronous, asynchronous, transactional, ...).

(4) *Operating System heterogeneity*: we need to know the operating system

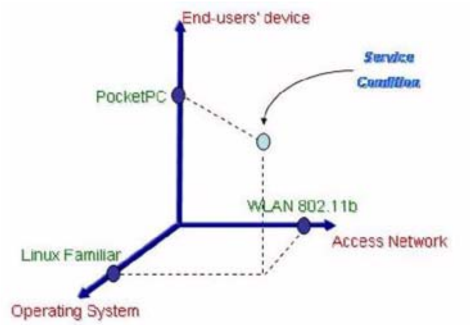

Fig. 1. A real example of "Service Condition"

type in a scenario where more choices are possible (*Unix, Linux, Win 98/NT/2000/XP, Win CE, Linux Familiar, Embedded OS, Symbian, ...*).

In addition, in the previous schema we could place in an additional variable: time. We have therefore a multi-dimensional space where network performance depend on variability of four different technical aspects. Thus, in general, extending this approach we have an *n-dimensional space* where 'n' is the *heterogeneity level*. We call a point in this space a "*Service Condition*". For example, in a three dimensional space we have the result depicted in figure 1. It is clear that to allow future users to have ubiquitous access to novel media services we need to allow them to roam transparently across different network, terminal and service technologies, in the same way today we are allowed to roam across different network operators with GSM/GPRS cellular devices. In our scenario this transparent roaming is reflected in variations in the Service Condition point. The "*Service Condition*" represents a new concept in heterogeneous network architectures. We use this concept in order to dominate the complexity of heterogeneous networks: when an user changes one of the key parameters (access network, terminal, operating system and application) a new point is identified in our three-dimensional space. Following the sequence of the point we can trace a "*User Service Condition Pattern*". In this work we study the network performance in several "*service conditions*". By using our results it is possible to design different multidimesional zone as a function of QoS parameters value: thanks to this model we can quickly determine which application works over a current "Service Condition".

3 Motivation and Related Work

This section compares and contrasts our framework with some other studies. We first briefly discuss the motivation at the base of our work and then we present the work of other alternative approaches.

One of the most innovative concept and, at the same time, the most difficult challenge for all network engineers is actually that of "integration": a unique and pervasive network scenario for the support of all the traffic (data, voice and video). A unique infrastructure but, above all, a unique protocol, the *Internet Protocol*, glue of all applications on different platforms. Among the many innovations introduced in the IP networks, an interesting challenge is to bring services like telephony and video transmission on the same infrastructure used for data traffic. This process relies on using QoS (*Quality of Service*) approach and at same time on the precise characterization of used heterogeneous network scenario. For these reasons, performance and experimental analysis of heterogeneous wireless networks is currently an important research issue. There are several simulation [13] and analytical [14] studies on wireless channel performance, whereas in this work, we test a real heterogeneous mobile environment and present a performance evaluation from the application point of view for a wide range of parameters. Our scenario is heterogeneous in terms of:

- access network technologies (*WLAN 802.11*, *wired Ethernet 10/100 Mbps*)
- end-users' devices (*PDA*, *Laptop*, *PC desktop*)
- end-users' operating systems (*Linux Embedded*, *Linux*, *Windows XP/ 2k/CE*)

Over this heterogeneous scenario we carried out a complete performance study of a real heterogeneous and integrated mobile network. In a situation where a roaming user sends traffic both to another roaming user and to a fixed position, experimental results on throughput - using both UDP and TCP connections - and delay, jitter, packet loss - using UDP connections - are presented. We assess our results showing the different performance (between roaming end-nodes) at different mutual distances.

Other experimental analysis are present in the literature. A performance characterization of ad hoc wireless networks is presented in [3]. The paper examines impact of varying packet size, beaconing interval, and route hop count on communication throughput, end-to-end delay, and packet loss. In [4] a new performance model for the IEEE 802.11 WLAN in ad hoc mode is presented. Three adjustable parameters are presented: packet fragmentation factor, buffer size, and maximum allowable number of retransmissions. In the work there is the measure of the system performance by using three parameters: throughput, delay, and probability of fail to deliver. In [5], three techniques for composite performance and availability analysis are discussed in detail through a queuing system in a wireless communication network. In [6] there is a study on network performance of commercial IEEE 802.11 compliant WLANs measured at the MAC sublayer in order to characterize their behavior in terms of throughput and response time under different network load conditions. A performance study on wireless LAN in a vehicular mobility scenario is presented in [7]. In [8] the performance of a real campus area network are measured. In order to carry out the results the authors used three performance monitoring soft-

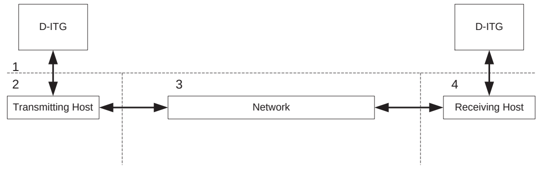

Fig. 2. The Experimental Testbed Infrastructure

ware: CWINS Wireless Benchmarking tool, Harris LAN Evolution Software and WaveLan Diagnostic Software. Performance measuring has been carried out moving on several parameters: received power, walls and floors separating two radio interfaces and finally interfering traffic. In [9] the authors present a comprehensive study on TCP and UDP behavior over WLAN taking into account radio hardware, device drivers and network protocols. [10] presents a performance measurements carried out on a real MAN in order to measure the real throughput. In [15] a discussion on the problems arising when the TCP/IP protocol suite is used to provide Internet connectivity over existing wireless links is presented. [16] studies the capabilities of an IEEE 802.11 wireless LAN. For the test phases, three wireless laptop computers, a wireless and wired desktop computers and an access point (AP) are used.

Our work extends previous works on TCP and UDP performance in many directions. More precisely, for the first time we present a complete evaluation, from the application point of view, of heterogeneous wireless networks in terms of a wide range of QoS parameters: throughput, delay, jitter and packet loss. Measured parameters are obtained for different packet size: in this way we can determine the optimal packet size for each "*Service Condition*". Previous works point their attention only on the wireless channel performance: by introducing the "*Service Condition*" we take into account several factors like Operating Systems, End-Users' Device and Network Technologies and relationships among them. After a measurement phase we place our throughput results in the framework of the model proposed by Bianchi in [2] and we use our results as performance references for development of wireless communication applications over multiservice heterogeneous networks.

4 Testbed Infrastructure, Tools and Experimental Methodology

The goal of our analysis is an empirical performance characterization of real heterogeneous networks in which wireless links are present. In order to pursue this objective a set of experimental setups with similar characteristics has been chosen. All tests can be collapsed in a same general scenario, depicted in figure 2, where two communication entities, a D-ITG transmitter and a D-ITG receiver, are directly connected through an IP network channel. In the context of the "*Service Condition*" concept, we point our attention on

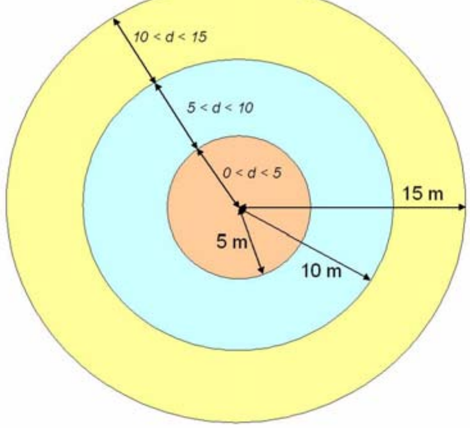

Fig. 3. Layout for Mobile Experiments

the analysis of a specific "*Service Condition subspace*". Indeed, as represented in figure 2, the tests differ for the type of used network, its configuration and the type of host that executes the D-ITG platform; by changing these parameters we tested several strictly related "*Service Conditions*". Others parametric elements, like generated traffic patterns, have not been changed: we used only periodical sources, with fixed Packet Size (PS) and fixed Inter-Departure Times (IDT) between packets since our intention for this study was mainly to focus on the impact of heterogeneity. In table 1 the complete set of parametric elements used in our tests is summarized. In the case of an ad-hoc scenario, we have experimented more configurations, allowing the two communicating hosts to move at various mutual distances: we tested a mobile environment using roaming user in three classes of end-to-end mutual distances ($d \leq 5\ m$, $5\ m \leq d \leq 10\ m$, $10\ m \leq d \leq 15\ m$) (Figure 3).

In the following, the measures are organized such that to distinguish three types of traffic conditions:

- *low* traffic load ($\leq 1.2 Mbps$): in our scenario *low* traffic load means a traffic state in which we are far from the saturated wireless channel condition.
- *medium* traffic load ($\leq 4.0 Mbps$): for *medium* traffic load we mean a traffic state in which we are close to the saturated wireless channel condition.
- *high* traffic load ($\leq 10 Mbps$): in the case of *high* traffic load we have a traffic state in which we are in the saturated wireless channel condition (i.e. every station has always a packet ready for the transmission).

These three traffic conditions are related to three different real traffic loads where we used different packet size. Indeed, in the first traffic profile we used PS equal to {64, 128, 256, 512, 1024, 1500} bytes and IDT equal to $\frac{1}{100}$ (ac-

Table 1: "*Service Conditions*" components

Testbed Element	Variables	Values
1 - D-ITG	Protocol Inter-Departure Size Packet Size	{UDP, TCP} IDT=$\{\frac{1}{100}, \frac{1}{1000}, \frac{1}{10000}\}$ s PS={64, 128, 256, 512, 1024, 1500} bytes
2 - Tx-Host	End users' device	{Workstation, Laptop, Palmtop}
3 - Network	Network Scenario	{Wired2Wired, Wired2Wireless, Wireless2Wireless, with and without Access Point(AP), ...}
4 - Rx-Host	End users' device	{Workstation, Laptop, Palmtop}
5 - Operating System	End users' OSs	{Windows XP, Linux, Linux Familiar}

cording to *low* traffic load). In the second traffic profile we used PS equal to {64, 128, 256, 512} bytes and IDT equal to $\frac{1}{1000}$ (according to *medium* traffic load). Finally in the third traffic profile we adopted PS equal to {64, 128} bytes and IDT equal to $\frac{1}{10000}$ (according to *high* traffic load). For every traffic condition, we organized the data in three types of configurations: (i) a classic configuration, with only laptop and workstation devices, (ii) a second configuration, where the transmitting host is always a Palmtop and (iii) a third configuration, where the receiving host is always a Palmtop.

In order to characterize a system like that one depicted in figure 2, we used the following QoS parameters by using the recommendations of IPPM working group [12]: (i) the (source/destination)-bandwidth (UDP and TCP protocols); (ii) the delay (UDP only); (iii) the jitter (UDP only) and finally (iv) the packet loss (UDP only). For each measured parameter, several trials have been performed in the same operating conditions. The values reported in the following graphics represent a mean value across twenty test repetitions. In our opinion, achieved results represent a good starting point. Indeed, during our current study we are experimenting similar results in other heterogeneous network configurations. In order to achieve our target we needed of an innovative tool for heterogeneous wireless network performance evaluation. Thus to this purpose, we extended the tool presented in the next Subsection and named D-ITG. Beside the statistics provided by D-ITG, we used *nstat* to gather IP, UDP and TCP statistics aggregated across all interfaces, so as to check for unexpected network activity during the experiments.

In this Subsection we briefly present an innovative tool, named D-ITG [11], that we introduced for heterogeneous wireless network performance evaluation. The purpose of our *Distributed Internet Traffic Generator* is to build up a suite that can be easily used to generate repeatable sets of experiments by using a reliable and realistic mixture of available traffic typologies. We use the following approach: simulation of real (but synthetic) traffic over real networks by using configurable scenario procedures on individual machines, and by coordinating the actions of these network devices. Typically other traffic generators can generate only UDP traffic with limited generation performance and offer a limited set of random variable distributions. D-ITG implements both TCP and UDP traffic generation according to several probability distributions (*exponential, uniform, constant, pareto, cauchy, normal, ...*) both for IDT (*Inter Departure Times*) and PS (*Packet Size*) random variables. The generation of realistic traffic patterns helps in studying performance issues in todays Internet. D-ITG primary design goals are: (i) *reproducibility of network experiments*: we implemented a method that can use the same seed for different stochastic experiments. It is possible to reproduce experiments by choosing the same seed value for the packets inter-departure and packets size random processes; (ii) *investigation of scaling effects*: using different network loads or different network configurations is possible to study scalability problems; (iii) *increase the generation performance* with respect to existing Traffic Generators; (iv) *increase the available traffic source models* with respect to other Traffic Generators; (v) the *possibility of simulating more complex traffic sources*, repeating many times exactly the same traffic pattern (not only its mean value) and getting information not only about received packets but also about transmitted packets; (vi) *measuring QoS parameters*: jitter, delay (one way delay and round trip time), packet loss and throughput. In a heterogeneous mobile scenario made by communications between PDA or Palmtop, using a distributed generator like D-ITG it is possible to generate high traffic rate on the mobile device and at the same time to store generated and received traffic on a server present in the wired network: this *modus operandi* provide an alternative way to data logging on device where the storage capacity is very small. Due to the nodes' limited resource (RAM, storage capacity, video dimension, ...) in wireless ad hoc networks, scalability is crucial for network operations. More precisely, a distributed approach to network communication using collaborative mechanism allows reaching comparable performance with respect to wired scenario. Indeed using a log server for sender and receiver logging phase we can assure greater performance when we use Palmtop too. Thus, in order to carry out a complete characterization of heterogeneous integrated and mobile networks D-ITG has been ported on several different operating systems: Linux, Windows, and embedded operating systems. With respect to this last platform in our testbed we used PDAs where is running the Linux

Table 2: Picture Legend

Network Scenario	Description
wired2wired	Connection between two workstation through an Ethernet 10/100 Mbps network
wired2wireless	Connection between the workstation and the laptop/palmtop through AP
wireless2wireless (AP)	Connection between laptop and palmtop through AP
wireless2wireless ($d \leq x$)	Connection between laptop and palmtop in ad-hoc mode in a range of x meters

FAMILIAR operating system and currently we are working on a porting on WinCE platform too. Thanks to previous features, D-ITG is the only one traffic generator able to measure real network performance in a heterogeneous scenario where PDA and Palmtop are present.

5 Performance analysis and experimentation

In this section measures obtained in the analyzed "*Service Conditions*" are presented. We organize our results showing the throughput, delay, jitter and packet loss measured in the following subsections. In table 2 the complete reference for the legend used in the following graphs is reported whereas in table 3 details on devices used are depicted.

5.1 Throughput Analysis

We step from showing and analyzing of results for *low* load traffic condition, then we present the results for *medium* and, finally, we show the results for *high* traffic load. As far as the throughput, a deep results analysis is reported in Section 6. In all following figures (except for figures 4 and 5), the first row is related to a situation in which the communication entities are two workstations, one workstation and one laptop, or two laptops; instead, the others two rows are related to a scenario in which the transmitter (second row) or the receiving (third row) host is always a Palmtop, while the transmitting/receiving one is a workstation (wired element) or a laptop (wireless element). First column of each figure represents the behavior observed by the transmitting host, while the second one represents the behavior observed by receiving host.

Table 3: Technical details on the experimental setup

Device	Description
Laptop1	IBM T23, Mobile Intel PIII 1133 Mhz, Main Memory 128 MB, Cache 256 KB, O.S. Linux Red Hat 9.0 kernel 2.4.20-18.9
Laptop2	Acer TravelMate 351 TE: PIII 700 Mhz, Main Memory 128 MB
Workstation1	PC sender, Intel PII 850 Mhz, Main Memory 128 MB, Cache 256 KB, dual boot Operating Systems: Linux(2.4), Windows XP Professional Service Pack 1
Workstation2	PC receiver, Intel C 400 Mhz, Main Memory 64 MB, Cache 128 KB, O.S. Linux(2.4)
Palmtop	Compaq iPAQ H3850, Intel StrongARM 206 Mhz, Main Memory 64 MB, Flash ROM 32 MB, O.S. Linux FAMILIAR kernel 2.4.18
Access Point	Orinoco Ap1000, 11Mbps (802.11b), Multi Channel support
Wireless LAN cards	WiFi ORINOCO 11Mbps GOLD

5.1.1 Low traffic load

In the case of *low* traffic load we are far from the saturated wireless channel condition. Test results for *low* traffic load are depicted in figures 4 and 5. In this case, we show only the graphics related to the behavior of the Palmotp2Laptop communication because it is the only situation where we can appreciate some very low performance degradation: in the case of Laptop2Laptop and Laptop2Palmtop communications both in the UDP and TCP scenario the sent throughput is equal to the received one for each packet size.

The throughput at sender and receiver side is reported in figures 4 and 5, using respectively UDP and TCP transport protocols. A precise results analysis is reported in Section 6.1.

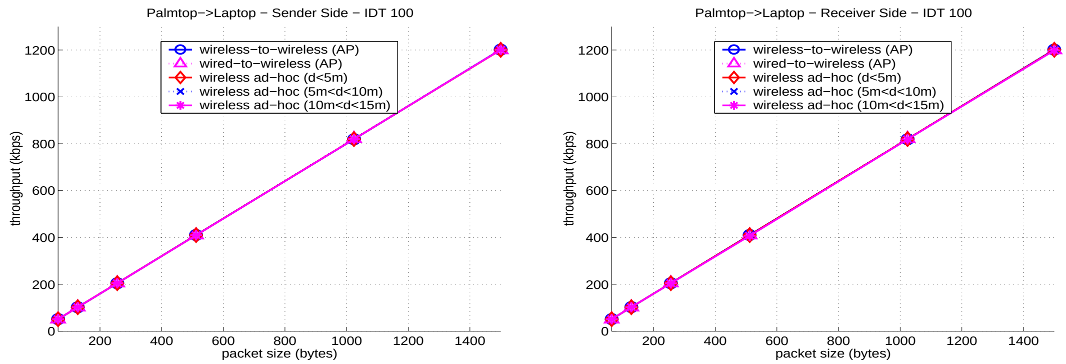

Fig. 4. Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{100}$ s in the case of UDP

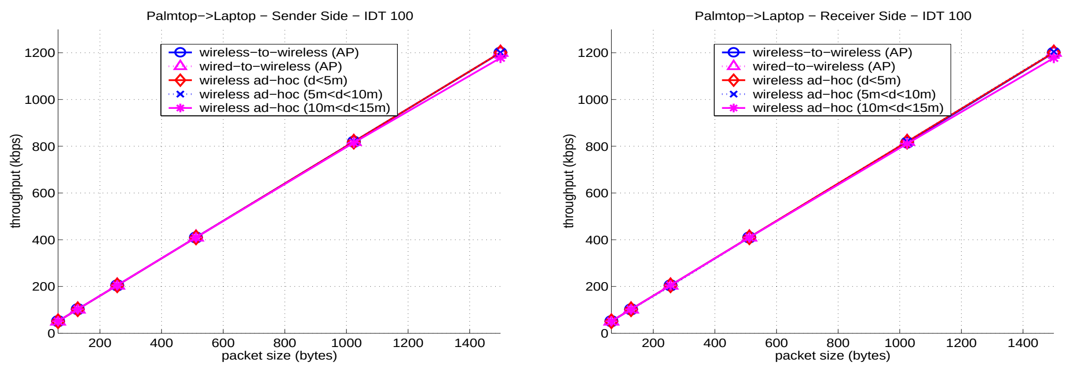

Fig. 5. Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{100}$ s in the case of TCP

5.1.2 Medium traffic load

The test results for *medium* traffic load are depicted in figures 6 and 7. In this case we are close to the saturated wireless channel condition. In order to quantify the proximity to the saturated channel condition, in the diagrams of the throughput it has been brought back also the diagram obtained from the Bianchi theoretical model [2]. In [2] a simple analytical model to compute the saturation throughput performance of the 802.11 is presented. The model assumes a finite number of terminals and ideal channel conditions and it is suited for any access scheme employed. The model shows that performance of the basic access method strongly depends on the system parameters, mainly packet size dimension and number of stations in the wireless network. Such model gives us a bound to the maximum traffic load that can cross the channel at the MAC layer of the ISO/OSI stack, therefore it supplies a useful bound for the traffic at the upper layer. Using our experimental results, we can also provide a practical validation of the Bianchi theoretical model (see Section 6).

In this load condition it turns out with more evidence the dependency from the host typology and the used transport protocol. TCP still demonstrates of being more sensitive to the losses respect to UDP. However, regarding the previous

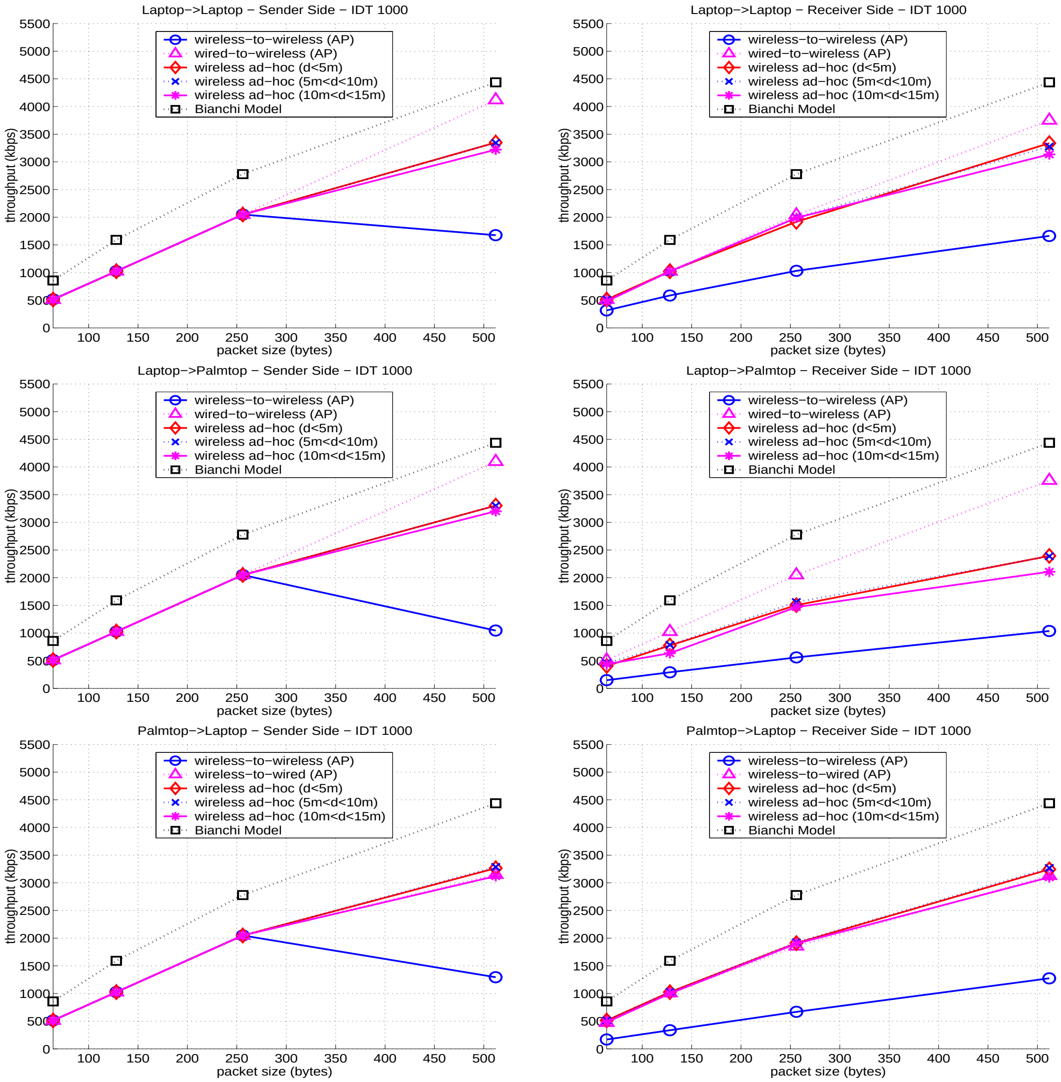

Fig. 6. Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{1000}$ s in the case of UDP

case we can observe the greater sensitivity respect to packet dimension of the wireless configurations, especially of those with the Palmtop. A detailed results analysis is reported in Section 6.2.

5.1.3 High traffic load

Test results for *high* traffic load are depicted in figures 8 and 9. In this case we are in the saturated wireless channel condition. With respect to previous cases we have analyzed a transmission condition where the packet size is equal to 64 bytes and equal to 128 bytes. Indeed, for whichever packet dimension the channel turns out saturated: longer packets carry to a greater channel busy time for delivered or collided packet, and it only leads to a greater number of losses from the sender side for network interface saturation. The organization

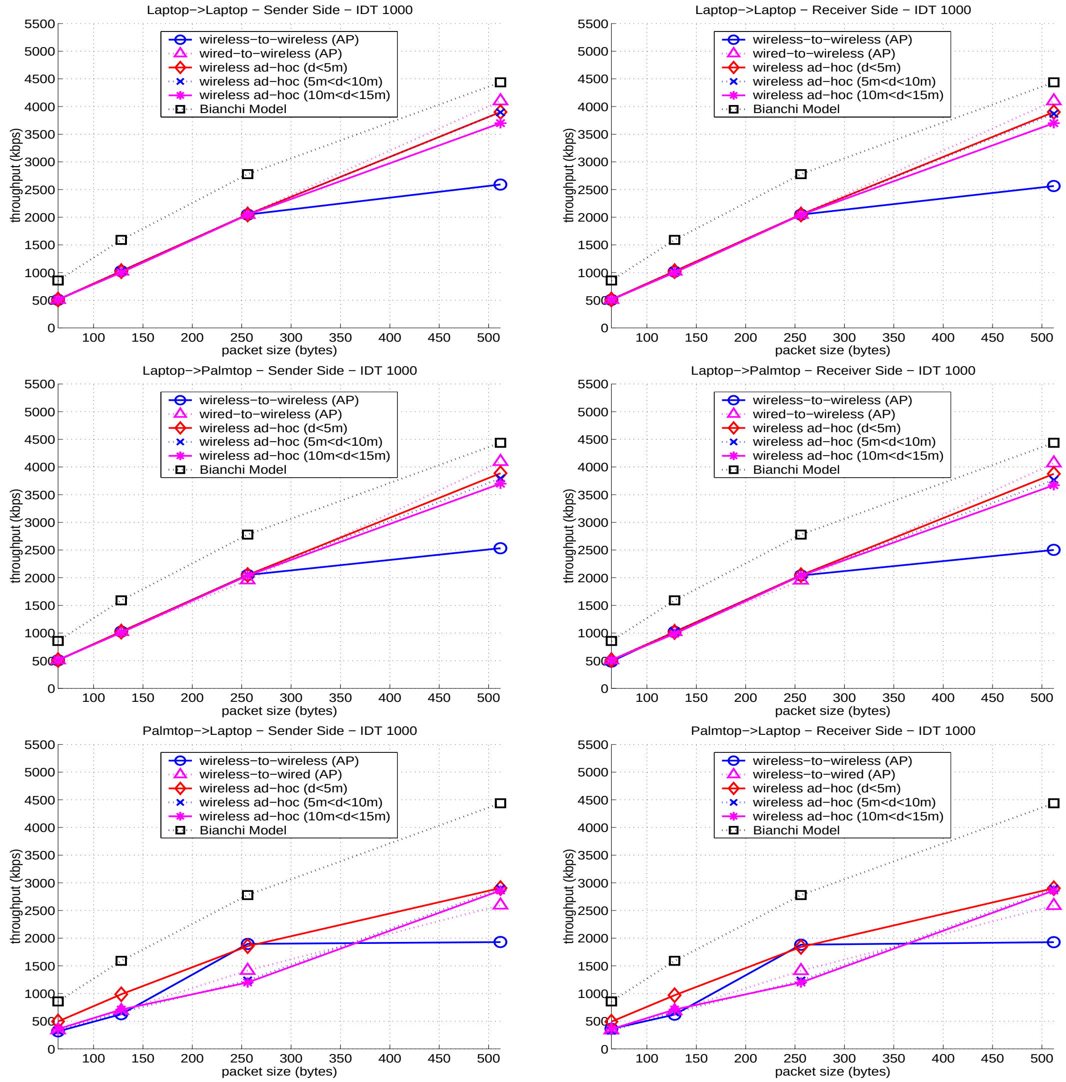

Fig. 7. Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{1000}$ s in the case of TCP

of the diagrams is the same one of the previous cases, the only difference is in having brought back the transmission and reception plots using histogram diagrams (in this case we have changed the figures layout because we have only two packet dimensions). It is interesting to notice the behavior of UDP and TCP in the several analyzed configurations: TCP reacts to the saturation condition limiting the demanded transmission bandwidth, while UDP endures a highest packet loss. This behavior is caused from the presence of a flow-control mechanism in the first protocol, and from the ability to the congestion control of TCP to optimize the use of a high loaded channel. Also in this case a deep analysis is reported in Section 6.3.

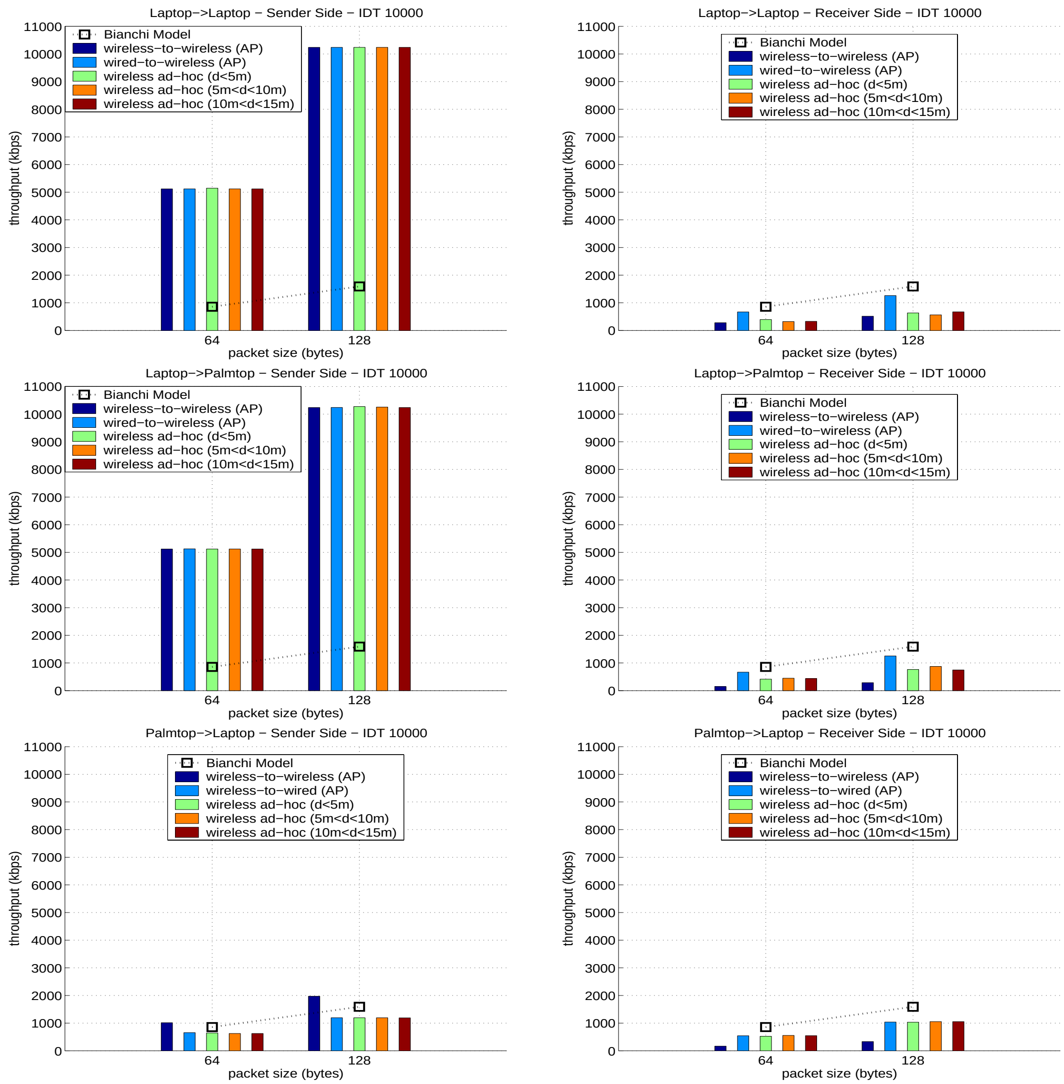

Fig. 8. Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{10000}$ s in the case of UDP

5.2 Delay Analysis

Delay is the amount of time that a packet takes to travel from the senders application to reach the receivers destination application. For example, in an Internet Telephony scenario, one-way delay requirement is stringent for VoIP to maintain good interaction between end-nodes. In order to have an upper bound for the one way delay we measured the Round Trip Time (RTT). This is due to: (i) we perform our measurement activities in the field of IPPM recommendations [12]; (ii) our measurements are carried out over wireless local links and not over geographical wireless links. The organization of the diagrams follows the same layout of the previous sections. As far as RTT thanks to our experimentation we learned that: (i) in the case of *low* traffic condition the configuration with Access Point presents the lowest performance for high

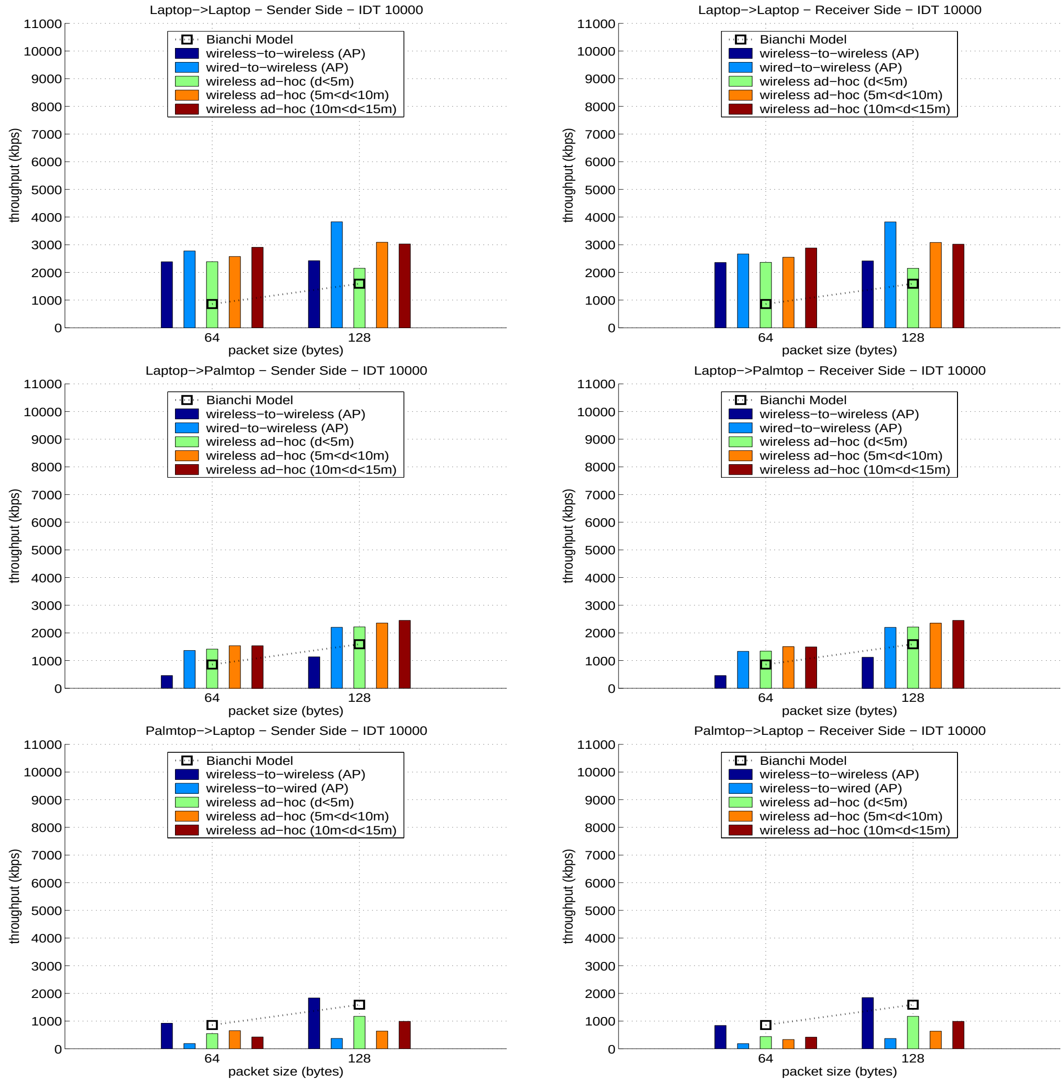

Fig. 9. Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{10000}$ s in the case of TCP

packet size (for PS equal to 1500 bytes we measured RTT $\approx$ 500 ms); (ii) the previous statement is not true in the case of Laptop2Palmtop configuration where we experimented the lowest performance in the case of ad-hoc configuration, with a distance d between sender and receiver equal to $10\ m \leq d \leq 15\ m$ (in this case we measured RTT until to 1000 ms). In the case of *medium* and *high* traffic condition we experimented the same behavior described for *low* traffic condition with same trend at a much high values. More precisely, as far as *medium* traffic condition: (i) the configuration with Access Point remains the scenario with lowest performance in the case of Laptop2Laptop and Palmtop2Laptop communications; (ii) in the case of Laptop2Laptop communications the RTT for the configuration with Access Point is under the 900 ms ($700\ ms \leq RTT \leq 900\ ms$); (iii) in the case of Palmtop2Laptop communications the RTT for the configuration with Access Point reaches RTT $\approx$ 500 ms

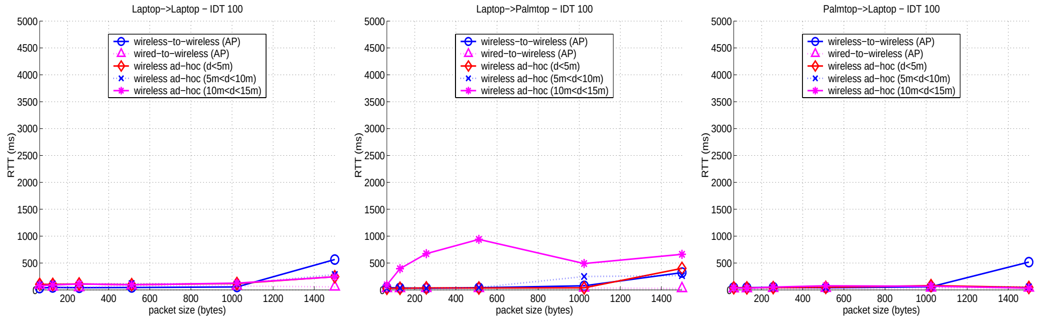

Fig. 10. UDP delay for IDT $= \frac{1}{100}$ s

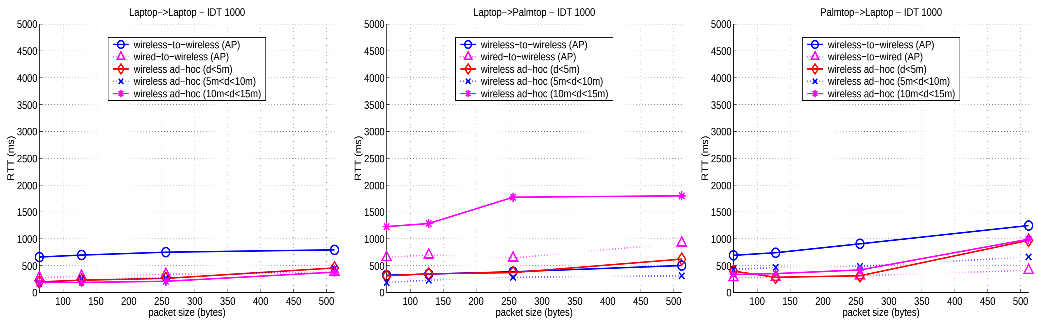

Fig. 11. UDP delay for IDT $= \frac{1}{1000}$ s

for PS equal to 1500 bytes; (iv) in the case of Laptop2Palmtop configuration we experimented the lowest performance in the case of ad-hoc configuration, with a distance d between sender and receiver equal to $10\ m \leq d \leq 15\ m$ (in this case we measured an $1300\ ms \leq RTT \leq 1800\ ms$). As far as *high* traffic condition we have the same behavior of the *medium* traffic condition with the following differences in terms of achieved results: (i) in the case of Laptop2Laptop communication lower RTT performance have been reached by using the Access Point configuration and obtaining RTT $\approx$ 700 ms for PS equal to 128 bytes; (ii) in the case of Palmtop2Laptop communication lower RTT performance have been reached by using the Access Point configuration and obtaining RTT $\approx$ 800 ms for PS equal to 128 bytes; (iii) in the case of Laptop2Palmtop communication lower RTT performance have been reached by using the ad-hoc configuration, with a distance d between sender and receiver equal to $10\ m \leq d \leq 15\ m$ and obtaining RTT $\approx$ 800 ms for PS equal to 128 bytes.

5.3 Jitter Analysis

Jitter is the variation in delay of the packets arriving at the receiving end. It can be considered as the standard deviation and it might be caused by congestion, insufficient bandwidth, varying packet sizes in the network, out of order packets. In an Internet Telephony architecture, excessive jitter may

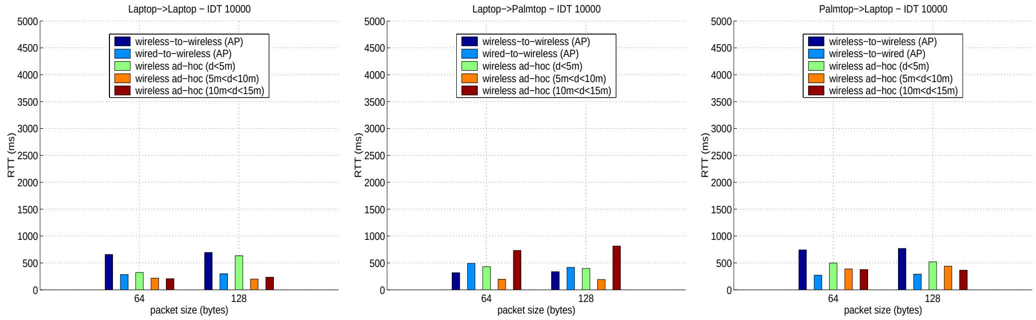

Fig. 12. UDP delay for IDT $= \frac{1}{10000}$ s

cause packet loss in the receiver jitter buffers thus affecting the playback of the voice stream. In figures 13, 14 and 15 we show the UDP jitter experimented in the three traffic conditions (*low*, *medium*, *high*). In almost all the analyzed "*Service Conditions*" and for each packet size we experimented the worst case in the configuration with Access Point. The jitter diagrams confirm that there is a weak sensitivity of the jitter as a function of the used configuration and the used hosts. Digging into details, the experimented jitter values are the following: (i) in the *low* traffic condition the worst case is under the 4 ms (we experimented a jitter equal to 2.5 ms in the Laptop2Laptop configuration); (ii) in the *medium* traffic condition the worst case is under the 8 ms (we experimented a jitter equal to 2.5 ms in the Laptop2Laptop configuration); (iii) in the *high* traffic condition the worst case is under the 4 ms (we experimented a jitter equal to 0.5 ms in the Laptop2Palmtop configuration). Highest values of the jitter have been experimented in the case of *medium* traffic load for high values of packet size (512 bytes) and for communications between Laptop2Palmtop and Palmtop2Laptop: this behavior is due to low capacity of Palmtops. Jitter behavior is associated to packet loss behavior and experimented throughput: we present the packet loss trend in the next subsection.

5.4 Packet Loss Analysis

Packet loss is a measure of packets discarded deliberately or non-deliberately by intermediate links, nodes and end-systems along a given transmission path between sender and receiver. In this section we present a packet loss analysis following the same "*modus operandi*" of the previous section: (i) UDP scenario; (ii) different PSs; (iii) different IDTs; (iv) different network technologies and end nodes. Figure 16 shows the UDP packet loss for IDT $= \frac{1}{100}$ s. Except some singularities in the ad-hoc configuration with $10\ m \leq d \leq 15\ m$, all considered "*Service Conditions*" showed a packet loss under the 0.5% and substantially equal to 0. More precisely, only when at receiver side a Palmtop is present and the packet size is lower than 512 bytes we measured a packet loss diverse

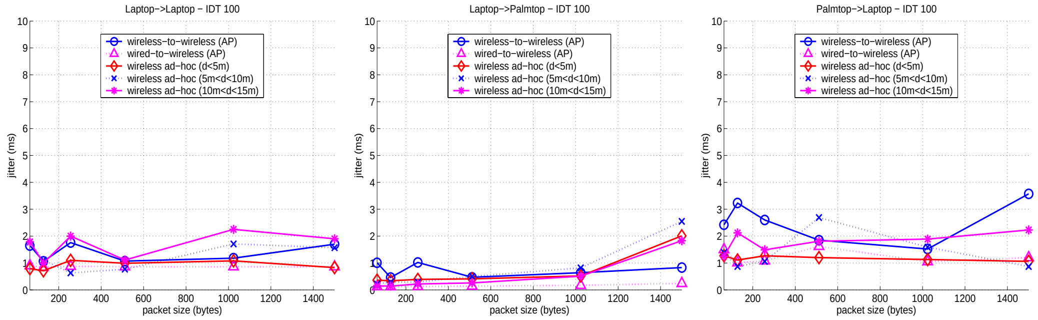

Fig. 13. UDP jitter for IDT $= \frac{1}{100}$ s

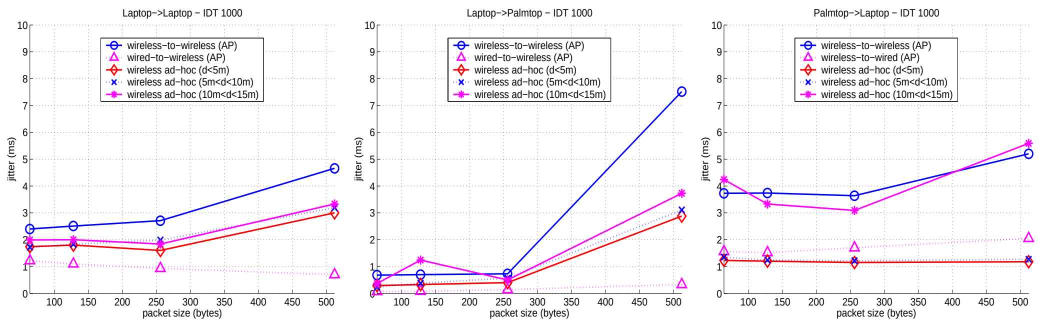

Fig. 14. UDP jitter IDT $= \frac{1}{1000}$ s

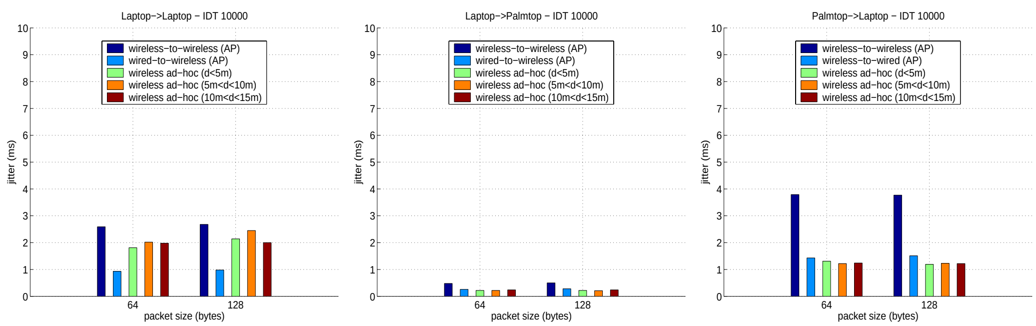

Fig. 15. UDP jitter for IDT $= \frac{1}{10000}$ s

from 0 and in all case lower than 0.5%. At the opposite site, Figure 17 and Figure 18 show dramatic values for the packet loss: in the case of *medium* and *high* traffic load the configuration with Access Point presents the lowest performance in terms of packet loss. More precisely, we experimented: (i) UDP packet loss for IDT $= \frac{1}{1000}$ s up to 70%; (ii) UDP packet loss for IDT $= \frac{1}{10000}$ s up to 95%.

As far as these last two traffic conditions, we experimented acceptable packet loss values: (i) for a *medium* traffic load only in the case of Laptop2Laptop and Laptop2Palmtop configuration, packet size up to 256 bytes and wired2wireless connection; (ii) in the case of *high* traffic load only when the sender was the Palmtop: this behavior is due to low transmission rate of the Palmtop that guarantees the reception of almost all sent packets. Finally, by analyzing

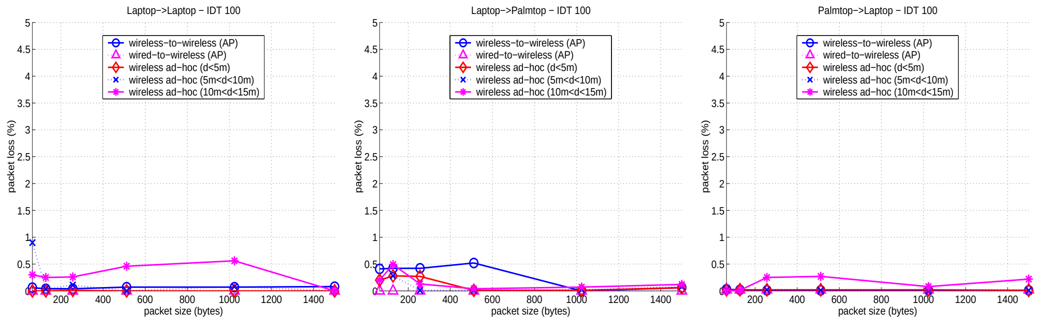

Fig. 16. UDP packet loss for IDT $= \frac{1}{100}$ s

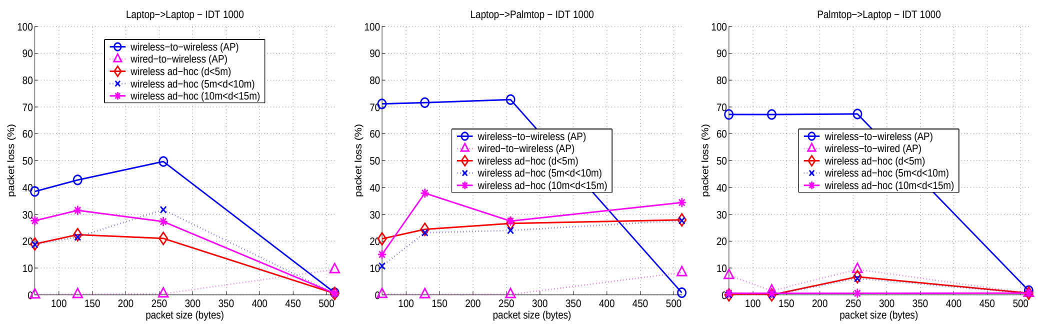

Fig. 17. UDP packet loss for IDT $= \frac{1}{1000}$ s

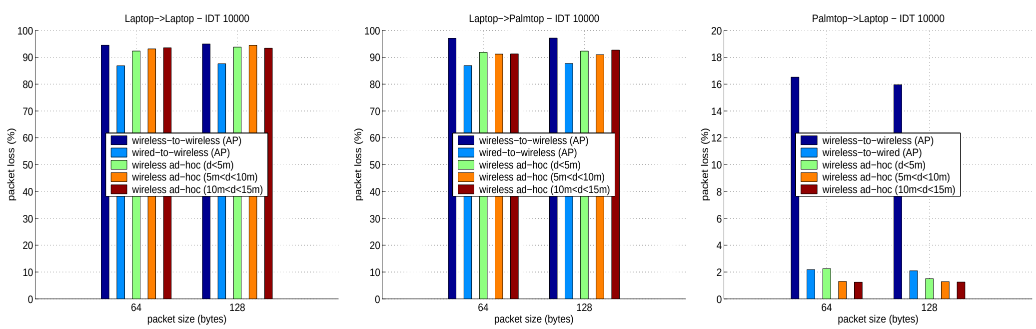

Fig. 18. UDP packet loss for IDT $= \frac{1}{10000}$ s

packet loss behavior we learned that: (i) the lowest packet loss values are obtained for high packet size; (ii) the worst case is obtained in the case of Palmtop at receiver side; (iii) with the exception of a Palmtop at sender side, at higher data rate the bottleneck are the wireless links (both in ad-hoc and with Access Point) and not the the end-users' device. The experimented packet loss results are strictly related to throughput behavior. In the next Section we present a deep analysis on achieved throughput.

6 Summary of Results

The analysis presented in this Section permits to understand the applicability of the Bianchi model to a real heterogeneous wireless network and to better know the TCP behavior over wireless scenario. TCP over wireless issues have been extensively discussed and several innovative proposals have been presented [15] [17]. Despite this situation, TCP performance analysis and characterization, from the user perspective, over a real heterogeneous wireless network represent an open issue. We present novel results that take into account a wide range of factors: different devices, different OSs and different network technologies are considered. Over this complex environment TCP performance are extremely difficult to understand. The TCP assumption that all losses are due to congestion becomes quite problematic over wireless links. In [18] G.T. Nguyen *et al.* show that (i) WLAN suffers from a frame error rate (FER) of 1.55% when transmitting 1400 byte frames over an 85 ft distance, with clustered losses and that reducing the frame size by 300 bytes halves FER there is an increase of framing overhead; (ii) mobility also increases FER for the WLAN by about 30%; (iii) FER is caused by the frequent invocations of congestion control mechanisms which repeatedly reduce TCP's transmission rate; (iv) if errors were uniformly distributed rather than clustered, throughput would increase. In addition, in [9] G. Xylomenos *et al.* show that in shared medium WLANs, forward TCP traffic (data) contends with reverse traffic (acknowledgments). In the WLAN this can lead to collisions that dramatically increase FER. As fas as maximum throughput, in [15] G. Xylomenos *et al.* show that the maximum throughput over a single wireless link, using either an IEEE 802.11 (2 Mbps) or an IEEE 802.11b (11 Mbps) WLAN is respectively equal to 0.98 Mbps and 4.3 Mbps. Thus, in the case of IEEE 802.11 there is an efficiency equal to 49% whereas in the case of IEEE 802.11b the efficiency is equal to 39.1%. This behaviour is due to higher speed links are affected more by losses, since TCP takes longer to reach its peak throughput after each loss.

In addition to these already know phenomena we present our innovative results that highlight the dependencies with (i) an high heterogeneity level, (ii) the properties of Palmtop device and (iii) three different traffic classes made by several combinations of IDTs and PSs. Furthermore, we present the TCP performance over wireless link varying the "*application level*" packet size: thanks to this "*modus operandi*" we can simple highlight which is the real TCP behavior over heterogeneous wireless network for different packet size values. Comparing the behavior for the same "*application level*" packet size, our analysis permits to clarify the conditions in which TCP performs better than UDP. In this section we analyze and comment our results with respect to achieved throughput. In order to give more readability to our analysis, we divide this Section in the same way of the Subsection 5.1.

6.1 Low traffic load

As we have anticipated in Section 5.1.1, in this case we show only the results related to the Palmtop2Laptop configuration. As far as the throughput analysis, in the case of UDP protocol, from Figure 4 we learn that in the case of *low* traffic load there is substantially the same behavior in all considered configurations. In the case of TCP protocol (Figure 5) we observed a similar behavior, with the following difference: in the case of a Palmtop at sender side and in the case of the ad-hoc configuration, with a distance d between sender and receiver equal to $10\ m \leq d \leq 15\ m$, a light throughput reduction (starting from a packet size equal to 1024 bytes) was experimented. Thus, in this case for the several configurations two aspects are clearly depicted: (a) the communication is reliable and (b) the light degradation of the performance is due to the smaller computational power of the adopted devices (PDAs). Also in this case, we have demonstrate that TCP suffers the losses mainly, having a different behavior with respect to UDP; TCP, indeed, interprets the losses like due to congestion phenomena and reacts consequently, reducing the maximum transmittable rate and emphasizing the phenomenon of bandwidth reduction. Indeed, of particular interest is the case of 1500 bytes packets, where the packet dimension exceeds MTU (*Maximum Transfer Unit*), the maximum allowable dimension of a MAC data unit. The fragmentation produces the duplication of the total number of transmitted packet and it exacerbates the throughput reduction of the wireless channel. Thus due to this behavior we experimented a (little) throughput reduction in the case of *low* traffic load. Finally, in the *low* traffic low and with a packet size close to the MTU, UDP performs better than TCP.

6.2 Medium traffic load

Probably results obtained in the *medium* traffic load analysis represent one the most important contributes of this paper. Indeed, we learned that in the case of *medium* traffic load there is a throughput behavior strictly coupled with network, device and traffic characteristics. In this case we are close to the Bianchi model hypothesis. Thanks to our results we can demonstrate that: (i) the Bianchi model represents an optimal upper bound; (ii) due to network dynamics present among TCP/IP application and data link layer and due to heterogeneity of considered elements there is a divergence between the theoretical Bianchi results and our real measures.

Digging into more details, as far as the throughput analysis, in the case of UDP protocol, from Figure 6 we learn that: (i) there is a progressive throughput reduction, at sender side, starting from PS equal to 256 bytes; (ii) both at sender and receiver side the configuration with Access Point shows lowest performance (indeed, in this case, the generated traffic present a double channel occupation); (iii) in the case of a Palmtop at receiver side there are, in all configuration, lowest performance. This reduction is higher in the case of configuration with Access Point. For example, in this case with PS equal to 512 bytes there is a difference with the model proposed by Bianchi equal to 3.5 Mbps; (iv) at higher packet size (PS $>$ 512 bytes) all tested configurations are far from the values of the model proposed by Bianchi (except for the wired-to-wireless configuration); (v) in the ad-hoc configuration there is a clear dependence between the achieved throughput and the end-nodes distances;

From Figure 7, in the case of TCP protocol, we learn that: (i) TCP shows better performance than UDP: this behavior is due to TCP capacity of putting more data into a single (TCP) segment. When we transmit UDP, our IP frame will carry only 512 bytes. When we transmit TCP traffic, TCP fits more data into the packet before transmitting (if they are available right away). This can happen until the proximity of MTU: in the *medium* traffic load when we reach the MTU, UDP presents better performance than TCP. Digging into numerical details, at low packet size (PS $<$ 512 bytes) TCP presents, in almost all considered configurations, 1 Mbps more than UDP achieved throughput; (ii) also in this case the configuration with Access Point shows lowest performance (but in the case of TCP we reach, for the same reason due to fit more data into a single segment, a better throughput with respect to the same configuration);

6.3 High traffic load

In the case of *high* traffic load, results obtained in this analysis show that the model proposed by Bianchi can not be used as an upper bound in all analyzed configuration. More precisely, in the case of UDP protocol (Figure 8) the Bianchi curve represents still an upper bound. At the opposite site, in the case of TCP protocol (Figure 9), we measured real throughput that overcame the values indicated in the Bianchi model. In the case of UDP protocol, both at sender and receiver side the configuration with Access Point shows lowest performance. The other configurations show substantially the same performance. In the case of TCP protocol, where there is a palmtop at receiver side, the configuration with Access Point shows lowest performance. Finally, using the TCP protocol we observed that all analyzed ad-hoc configuration show best performance. This behavior is due to the same motivation presented in the previous subsection (6.2): in this case we are far from MTU and with a saturated channel. We repeated the experiment with a packet size equal to 1500 bytes and the same IDT and we measured that UDP performs better than TCP. We don't provide this graphics because we have a low achieved throughput (in the case of PS = 1500 bytes, IDT = $\frac{1}{10000}$ we have a data rate equal to 120 Mbps over a 11 Mbps channel).

7 Conclusions and Ongoing Work

All over the world heterogeneous wireless networks are being used to support mobile services and innovative multi-scenario applications. Unfortunately, application performance over heterogeneous networks is severely impacted by problems of high and variable round trip times, fluctuating bandwidths, frequent link outages, burst losses, etc. As a consequence, the end-user experience in such environments is significantly different from the relatively stable wired environments. In this work we presented a general framework for traffic analysis and performance characterization in real heterogeneous mobile networks from a end-user perspective. Our work extends previous works on TCP and UDP performance over WLANs in many directions. Indeed, this work steps from the assumption that a current realistic scenario must consider the fusion of wired and wireless networks, several kinds of user devices, different operating systems and users' applications. We proposed to control this complexity the innovative "*Service Condition*" concept and we presented a general framework for empirical performance study of heterogeneous wireless networks introducing a per "*traffic load*" class analysis: we defined three traffic conditions and we divided our experimentation in three stages: *low* traffic load, *medium* traffic load, *high* traffic load. A number of tests conducted on our real testbed yielded important characteristics such as throughput, delay, jitter and packet loss under various network loads in UDP and TCP scenarios. We carried out our results by introducing an innovative open source traffic generator, named D-ITG. Our results provide a clear and precise characterization of the measured QoS parameters for each analyzed "*Service Condition*". Indeed, one of the contributions of our work was the clear definition of which system's elements are responsible of network performance degradation and how to use different protocols impacts observed on the traffic behavior. We have shown and analyzed the *real* performance of a *real* heterogeneous wireless network. We carried out the dependencies from used device (in particular the limitations when at receiving side a Palmtop is present) and from wireless network configuration (in particular in the case of Access Point configuration and ad-hoc mode scenario). We experimented a better TCP performance than UDP performance when the packet size is under the 512 bytes. We think that this result is particularly interesting when compared with other previous works. This analysis has been conducted with mobile users too. Furthermore, our results have been analyzed with respect to analytical model provided by Bianchi. We have demonstrated that it is useful as an upper bound, but in a real scenario and from the application point of view a tuning of the Bianchi model parameters could be useful: we are working on a revised analytical modeling of Bianchi proposal in order to take into account results shown in this paper. We are moving toward an "*application level Bianchi*" model. In the meanwhile, these results can be used as references for development of wireless communication applications. Indeed in a planning phase of innovative applications over heterogeneous networks is necessary a complete parametric network characterization.

Currently, our testbed allows experiments on a small-scale. We will test obtained results on a heterogeneous network of a much wider-scale. Furthermore, in our ongoing work, we are conducting a similar analysis presented in this paper in a scenario where interference due to Bluetooth and IrDA communications are present. Indeed we believe that interoperability, interference and co-existence of wireless networks of different standards is one of the major issues in the future research. By using D-ITG capabilities we will test a similar scenario using different traffic patterns made by different stochastic IDTs and PS distributions according to several theoretical traffic models: we are studying which are the dependencies between experimented performance and different traffic patterns. Finally, we believe that a complete analysis (from the physical layer to the application layer) is needed, but in a first approximation where a performance analysis is necessary for characterizing end-users application over heterogeneous network, our approach is exceptionally important. An interesting second step could be a deep analysis in order to understand which is the relationship between measured performance at application/transport layer and modeled/measured performance at physical/data link layer.

8 Acknowledgments

This work has been carried out partially under the financial support of the "Ministero dell'Istruzione, dell'Universit e della Ricerca (MIUR)" in the framework of the FIRB Project "Middleware for advanced services over large-scale, wired-wireless distributed systems (WEB-MINDS)".

References

[1] J. P. Munson, M. Viveros. "Pervasive and Mobile Commerce Applications" *IPCCC 2002 - 21st IEEE International Performance, Computing, and Communications Conference*, April 2002 - Phoenix, Arizona

[2] G. Bianchi, "Performance Analysis of the 802.11 Distributed Coordination

Function", *Selected Area in Communications, IEEE Journal on* Volume: 18 Issue: 3, March 2000 Page(s): 535 -547

[3] C. -K. Toh, M. Delwar, D. Allen, "Evaluating the communication performance of an ad hoc wireless network", *Wireless Communications, IEEE Transactions on* , Volume: 1 Issue: 3 , July 2002 Page(s): 402 -414

[4] F. Eshghi, A. K. Elhakeem, "Performance analysis of ad hoc wireless LANs for real-time traffic", *Selected Areas in Communications, IEEE Journal on* , Volume: 21 Issue: 2 , Feb. 2003 Page(s): 204 -215

[5] Ma Yue, J. J. Han, K. S. Trivedi, "Composite performance and availability analysis of wireless communication networks", *Vehicular Technology, IEEE Transactions on* , Volume: 50 Issue: 5 , Sept. 2001 Page(s): 1216 -1223

[6] B. Bing, "Measured performance of the IEEE 802.11 wireless LAN", *Local Computer Networks, 1999. LCN '99. Conference on* , 18-20 Oct. 1999 Page(s): 34 -42

[7] J. P. Singh, N. Bambos, B. Srinivasan, D. Clawin, "Wireless LAN performance under varied stress conditions in vehicular traffic scenarios", *Vehicular Technology Conference, 2002. Proceedings.* VTC 2002-Fall. 2002 IEEE 56th , Volume: 2 , 24-28 Sept. 2002 Page(s): 743 -747 vol.2

[8] A. Messier, J. Robinson, K. Pahlavan, "Performance monitoring of a wireless campus area network", *Local Computer Networks, 1997. Proceedings*, 22nd Annual Conference on , 2-5 Nov. 1997 Page(s): 232 -238

[9] G. Xylomenos, G. C. Polyzos, "TCP and UDP performance over a wireless LAN", *INFOCOM '99. Eighteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings.* IEEE , Volume: 2 , 21-25 March 1999 Page(s): 439 -446 vol.2

[10] J. C. Amaro, R. P. Lopes, "Performance analysis of a wireless MAN", *Network Computing and Applications, 2001. NCA 2001. IEEE International Symposium on*, 8-10 Oct. 2001 Page(s): 358 -361

[11] D-ITG, Distributed Internet Traffic Generator: http://www.grid.unina.it/software/ITG

[12] IP Performance Metrics (IPPP), IETF Working Group http://www.ietf.org/html.charters/ippm-charter.html - RFC 2330: Framework for IP Performance Metrics; RFC 2498: IPPM Metrics for Measuring Connectivity; RFC 2679: A One-way Delay Metric for IPPM; RFC 2680: A One-way Packet Loss Metric for IPPM; RFC 2681: A Round-trip Delay Metric for IPPM; RFC 3393: IP Packet Delay Variation Metric for IP Performance Metrics (IPPM).

[13] M. Carvalho and J.J. Garcia-Luna-Aceves, "Delay Analysis of IEEE 802.11 in Single-Hop Networks" *Proc. IEEE ICNP 03: 11th IEEE International Conference on Network Protocols*, Atlanta, Georgia, November 4-7, 2003.

[14] H. Wu, Y. Peng, K. Long, S. Cheng, J. Ma, “Performance of Reliable Transport Protocol over IEEE 802.11 Wireless LAN: Analysis And Enhancement”, *IEEE INFOCOM'2002*.

[15] G. Xylomenos, G. C. Polyzos, P. Mahonen, M. Saaranen, “TCP Performance Issues over Wireless Links”, *IEEE COMMUNICATIONS MAGAZINE*, VOLUME 39, NUMBER 4, 2001, PP. 52-58.

[16] T. Demir, C. Komar, C. Ersoy, “Measured Performance of an IEEE 802.11 Wireless LAN”

[17] H. Balakrishnan, V. N. Padmanabhan, S. Seshan, R. H. Katz, “A comparison of mechanisms for improving TCP performance over wireless links”, *ACM SIGCOMM '96*, August 1996, PP. 256-267.

[18] G.T. Nguyen, R.H. Katz, B. Noble, and M. Satyanarayanan, “A tracebased approach for modeling wireless channel behavior“, *Winter Simulation Conference*, Dec. 1996, pp. 597-604.