Performance Evaluation of Heterogeneous Network Scenarios

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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 complex heterogeneous systems (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 the Quality of Service metrics. Stepping fro this consideration, in this Chapter we provide an empirical performance study of real heterogeneous wireless network with respect to delay, jitter, throughput and packet loss, in UDP and TCP environments, by using the innovative tool D-ITG (*Distributed Internet Traffic Generator*).

We also use the concept of "*Service Condition*" as a mechanism to cope with issues related to dynamically changing network service scenario and to formally define a parametric and systemic approach to the performance evaluation.

Results presented in this Report can be used as performance references for development of wireless communication applications over multi-service heterogeneous networks.

Indeed, we perform a large set of measurements with a wide set of hardware. Then we take the results of our measurements and by analyzing the similarities and the differences we understand what we can generalize and what not. We investigate the impact of mechanisms and protocols on the performance metrics and we try to prove which protocol element is a limiting factor or which devices characteristics affect the measured metrics. In addition, thanks to the wide range of provided results (as well as the public traffic traces, available on request), a reader could actually make use of the information provided in this Report for his purposes.

More precisely, this Report presents results of our performance evaluation activities in the following scenario:

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- HetPerf (*Heterogeneous Networks Performance*): this framework presents experimental results related to heterogeneous networks where both wireless local area networks and cellular networks are present. We study the performance in this integrated and heterogeneous scenario organizing our results in the following way:
  - Wireless Local Area Networks (Wi-Fi WLAN 802.11b)
  - Wireless Wide Area Networks (GPRS and UMTS)
- HandPerf (*Vertical Handoff Performance*): this framework presents experimental results related to an architecture for seamless handoffs.

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 (local and geographic) 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 Chapter focuses on the area of performance evaluation of heterogeneous wireless networks from the application level point of view.

By using the same approach adopted in previous Chapters, the experimental analysis has been carried out over large but controlled network environments. Therefore, also in the case of performance evaluation of heterogeneous wireless networks we used experimental test-beds present in our laboratories. Since we also measured scenario where UMTS and GPRS networks are present, we attached our experimental test-beds to Telecom Operators networks. These are the only two cases where the access networks are out of our control. Therefore, as for the analyzed "*Service Condition*" we tested real heterogeneous wireless networks made a number of complex combination by:

- Access Networks: *Ethernet*, *802.11b*, *GPRS*, and *UMTS*
- End User Devices: *Workstation*, *Personal Computer*, *Laptop*, *Notebook*, and *Personal Digital Assistant (PDA)*
- Operating Systems: *Unix*, *Linux*, *Windows*, and *Linux Familiar*

Over the depicted scenario we tested the behavior of both UDP and TCP.

In order to define a systemic measurement approach we introduced a network performance methodology dividing our experimentation on several traffic classes. The performance evaluation study has been performed by following some of the indications of IP Performance Metrics (IPPM) IETF Working Group [13] and by adopting the guidelines reported in [14]. The network behavior has been studied by using D-ITG (*Distributed Internet Traffic Generator*) . 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].

As said, we show how to use the "*Service Condition*" concept to cope with the issues of a performance evaluation framework in a heterogeneous wireless environment. More precisely, first we present and discuss results of a complete performance evaluation of a heterogeneous wireless scenario. In this first study we show the achieved performance as a function of Access Network, End-User Device, Operating System, and Application (TCP or UDP). We called this framework *HetPerf*. Second, we present results of an experimental analysis of a heterogeneous scenario where vertical handoffs are present. In practice, in this case we study the performance of "Service Conditions" that present the same values of End-User Device, Operating System, and Application but different values in terms of Access Networks. We called this second framework *HandPerf*.

Thanks to this performance evaluation study:

- we are in charge of to demonstrate the goodness of the "*Service Condition*" concept in the field of network measurements;
- we provide a complete study of heterogeneous wireless networks in terms of throughput, delay (OWD and RTT), packet loss, and jitter. It represents a real assessment of QoS metrics over heterogeneous wireless networks;
- 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.

1.1 Test-bed Infrastructure, Tools and Experimental Methodology

The goal of our analysis is an empirical performance characterization of real heterogeneous networks in which several wireless links are present. In order to

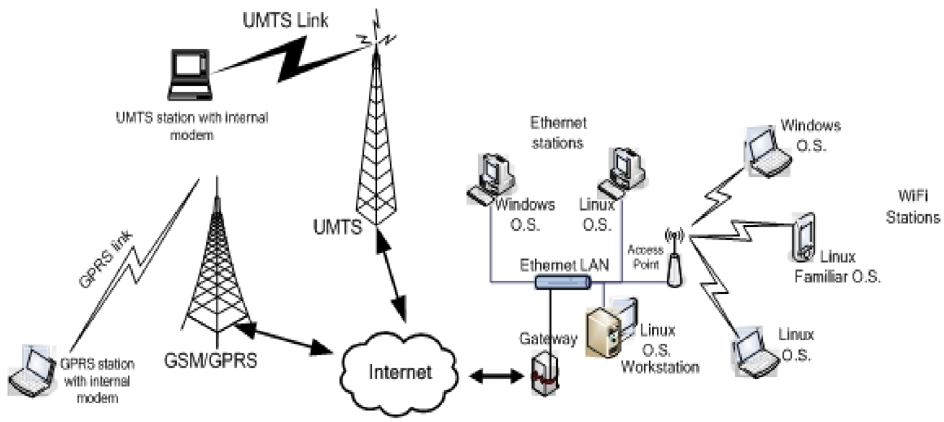


Figure 1: The Experimental Test-bed Infrastructure: Real Network

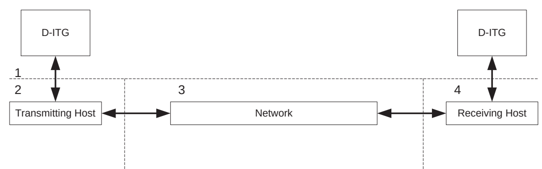


Figure 2: The Experimental Test-bed Infrastructure: Conceptual Schema

pursue this objective a set of experimental setups with similar characteristics has been chosen (see Figure 1).

The first step towards the definition of a systemic measurement approach is the definition of a *network conceptual schema*. All tests can be collapsed in a same general schema, 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. Indeed, the tests differ for the type of used network, its configuration, the type of host that executes the D-ITG platform, and used Operating Systems. By changing these parameters we tested several strictly related "*Service Conditions*". In figure 2 the conceptual schema of the experimental test-bed is depicted.

In this preliminary study, 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 our performance evaluation we organize our measurements such that to distinguish three types of traffic conditions:

- *low* traffic load ($\leq 1.2Mbps$): in our scenario *low* traffic load means a traffic state in which we are far from the saturated 802.11b wireless channel condition.

- *medium* traffic load ($\leq 4.0Mbps$): for *medium* traffic load we mean a traffic state in which we are close to the saturated 802.11b wireless channel condition.

- *high* traffic load ($\leq 10Mbps$): in the case of *high* traffic load we have a traffic state in which we are in the saturated 802.11b 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 $\frac{1}{100}$ (according 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).

2 HetPerf

As for the performance evaluation of the scenario that we called "*HetPerf*", in table 1 the complete set of parametric elements used in our tests is summarized. In addition, 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).

For every traffic condition presented in the previous Paragraph, 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 [13]: (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. 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.

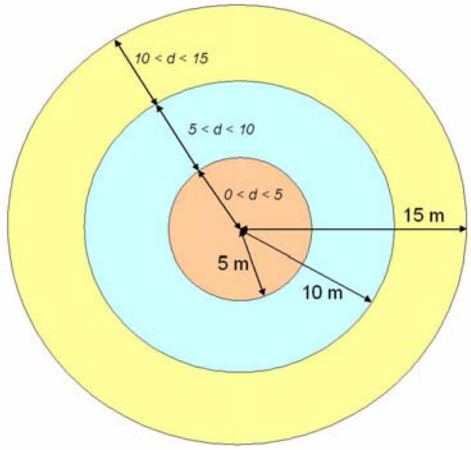


Figure 3: Physical Layout in the case of Experiments with Mobile node

Table 1: "*Service Conditions*" components

Test-bed 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}

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

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

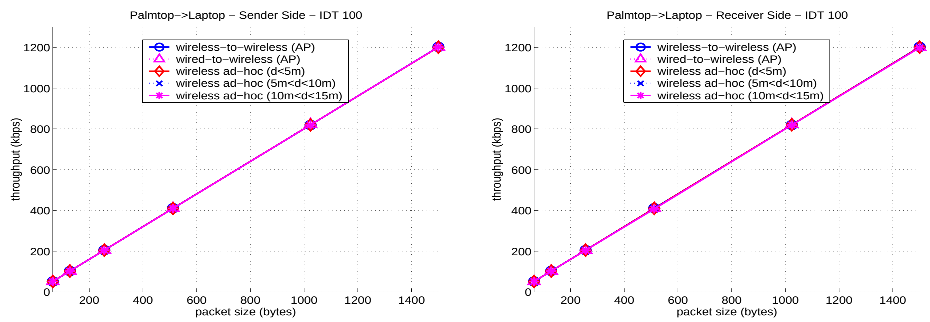


Figure 4: Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{100}$ s in the case of UDP

2.1 Experimentation with Wireless Local Area Network

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.

2.1.1 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 2.1.5. 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.

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

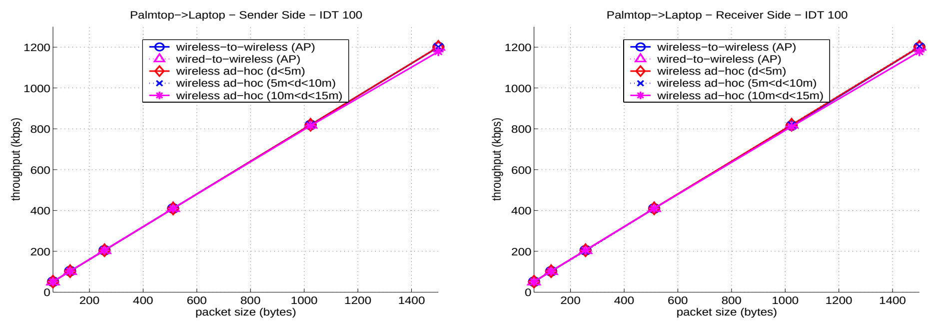


Figure 5: Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{100}$ s in the case of TCP

is reported in Section 2.1.5.

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 2.1.5).

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 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 2.1.5.

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.

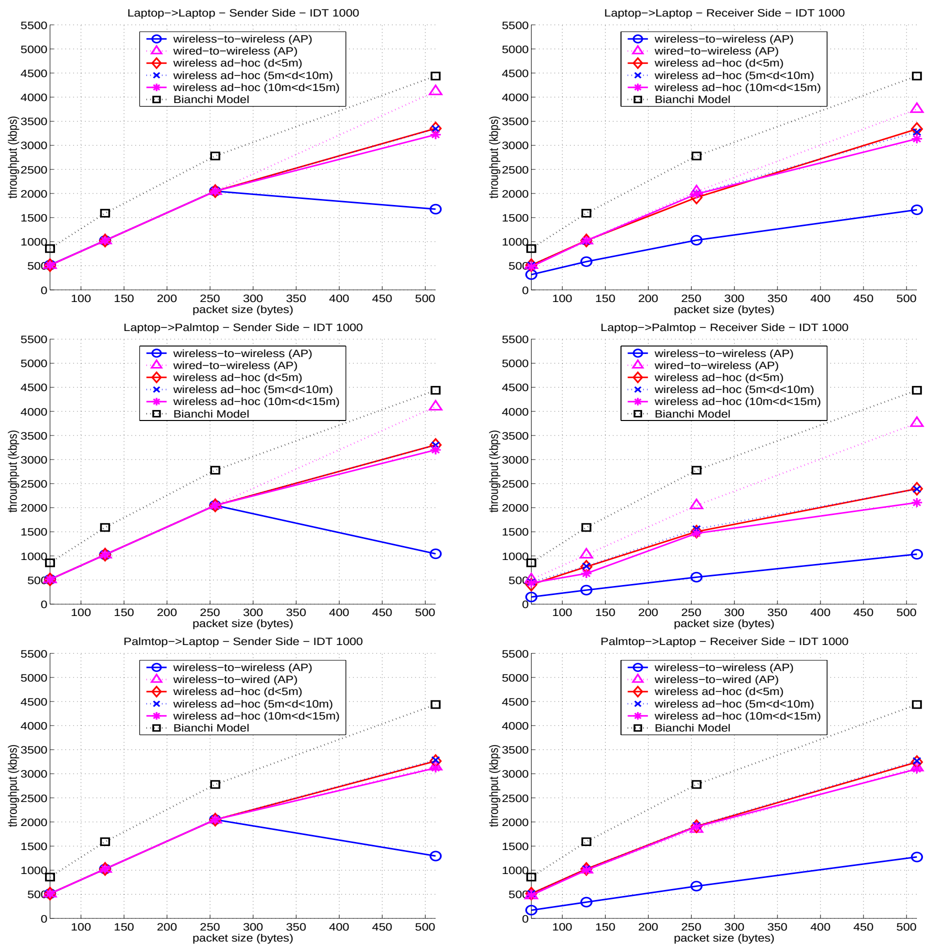


Figure 6: Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{1000}$ s in the case of UDP

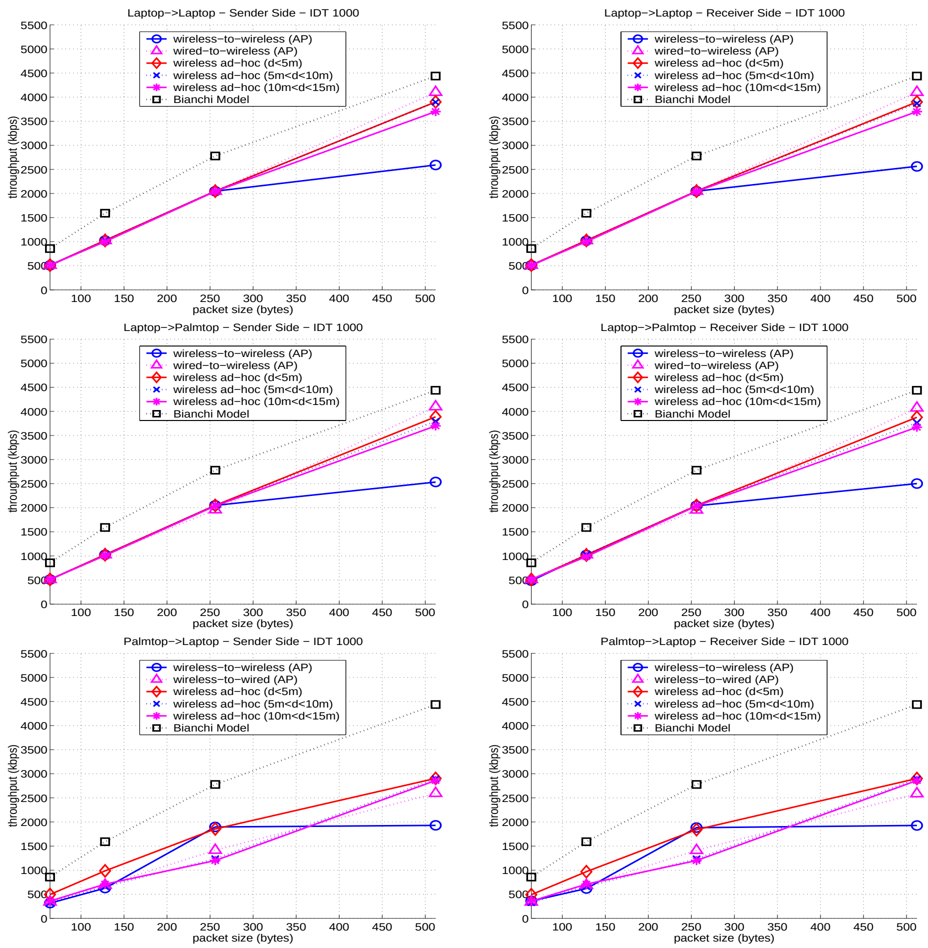


Figure 7: Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{1000}$ s in the case of TCP

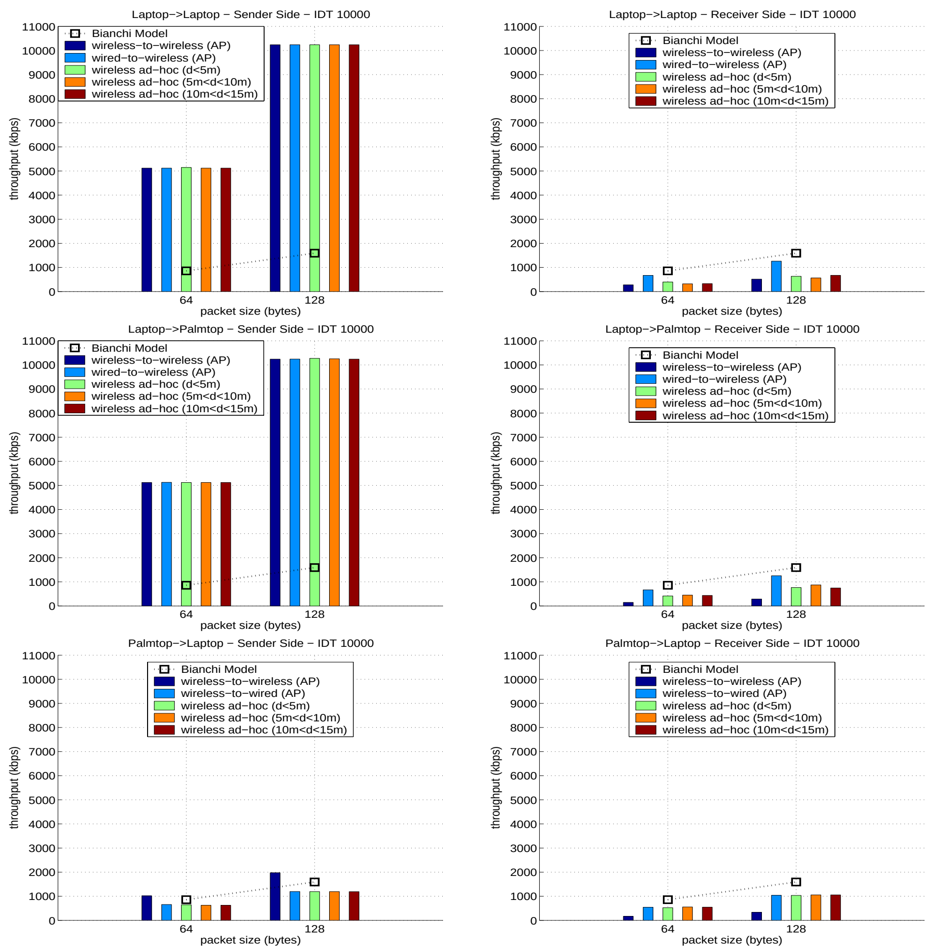


Figure 8: Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{10000}$ s in the case of UDP

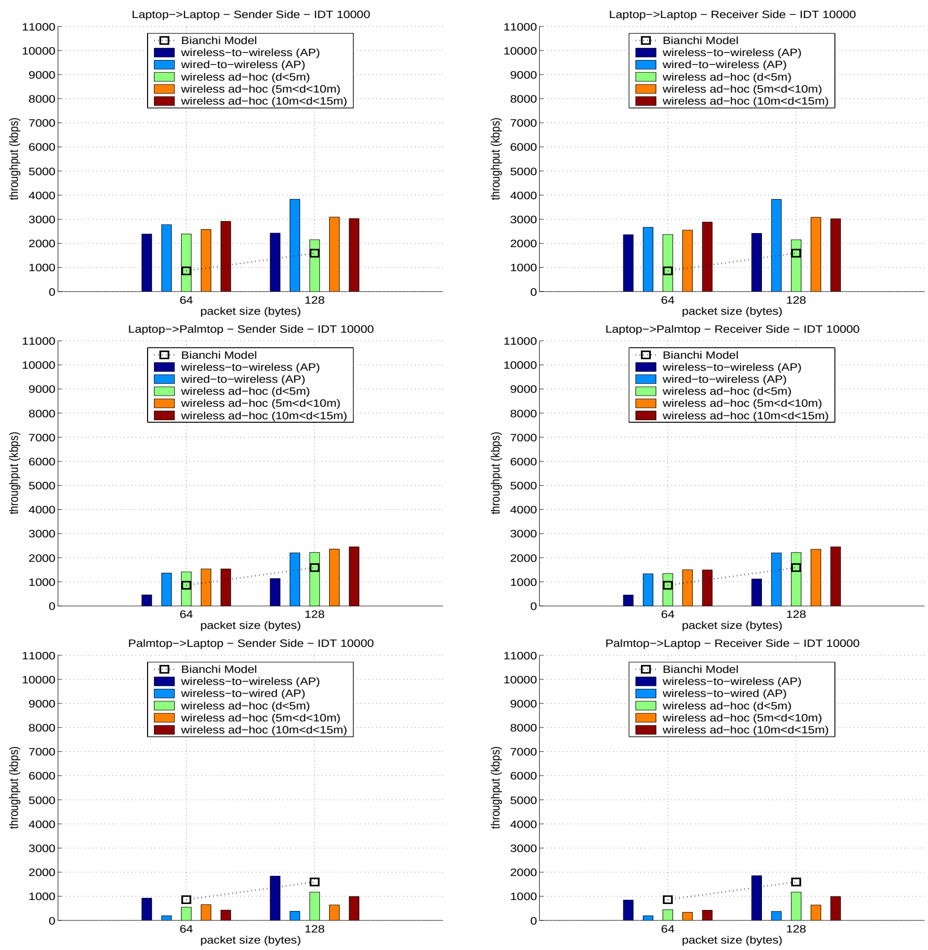


Figure 9: Throughput analysis at sender (left) and receiver (right) side for IDT $= \frac{1}{10000}$ s in the case of TCP

The organization 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 2.1.5.

2.1.2 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 [13]; (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 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 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 configura-

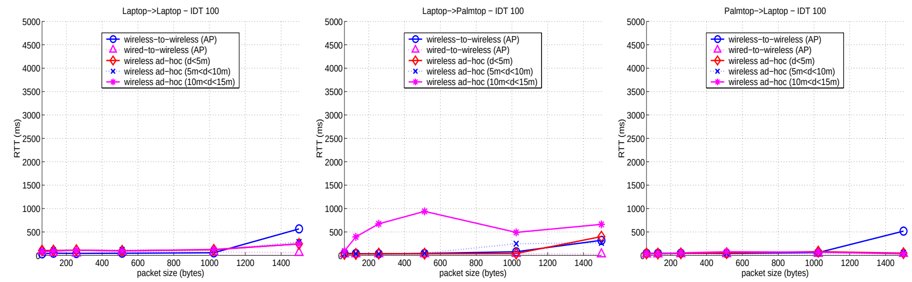


Figure 10: UDP delay for IDT $= \frac{1}{100}$ s

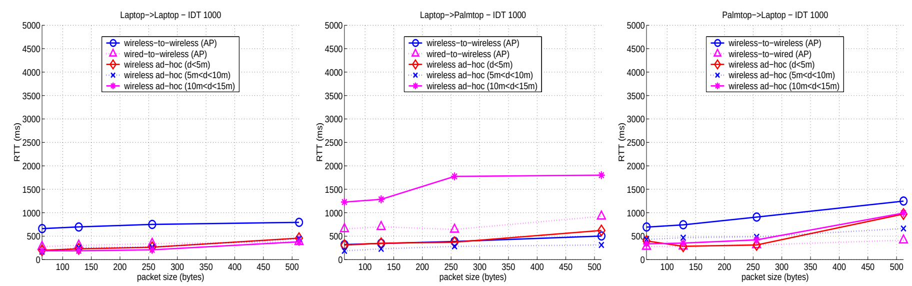


Figure 11: UDP delay for IDT $= \frac{1}{1000}$ s

tion 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.

2.1.3 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 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 $4ms$ (we experimented a jitter equal to

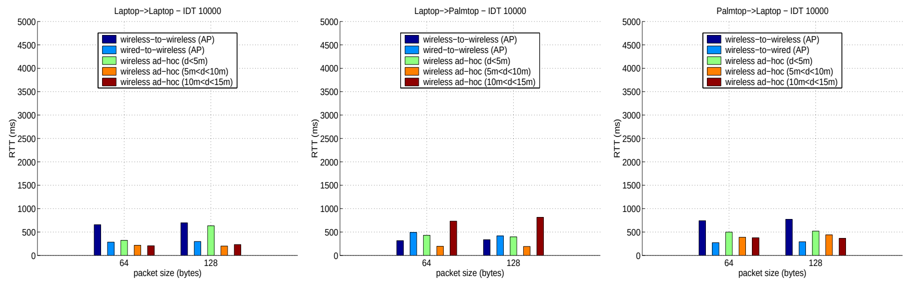


Figure 12: UDP delay for IDT $= \frac{1}{10000}$ s

2.5ms in the Laptop2Laptop configuration); (ii) in the *medium* traffic condition the worst case is under the 8ms (we experimented a jitter equal to 2.5ms in the Laptop2Laptop configuration); (iii) in the *high* traffic condition the worst case is under the 4ms (we experimented a jitter equal to 0.5ms 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. The jitter behavior is associated to packet loss behavior and experimented throughput. We present the packet loss trend in the subsection 2.1.4.

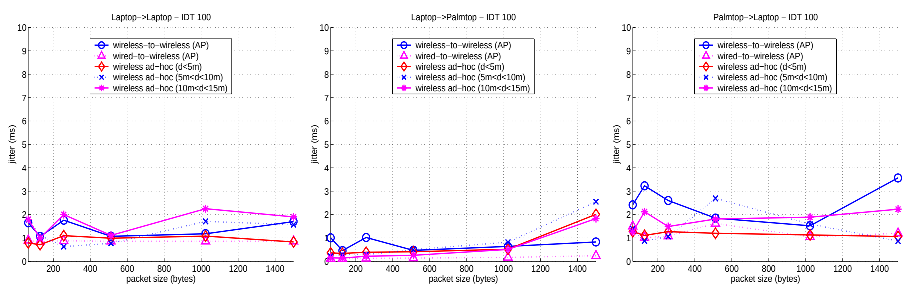


Figure 13: UDP jitter for IDT $= \frac{1}{100}$ s

2.1.4 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

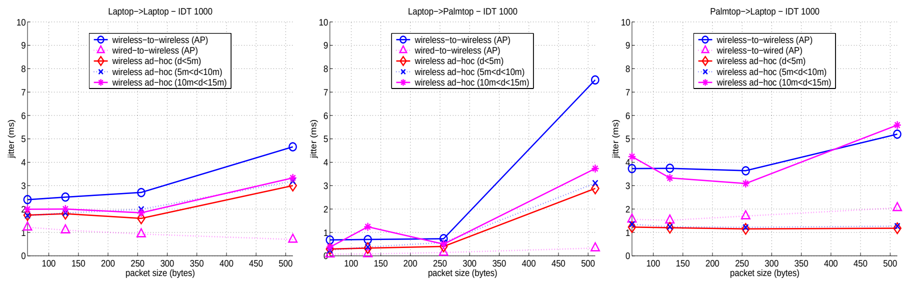


Figure 14: UDP jitter IDT $= \frac{1}{1000}$ s

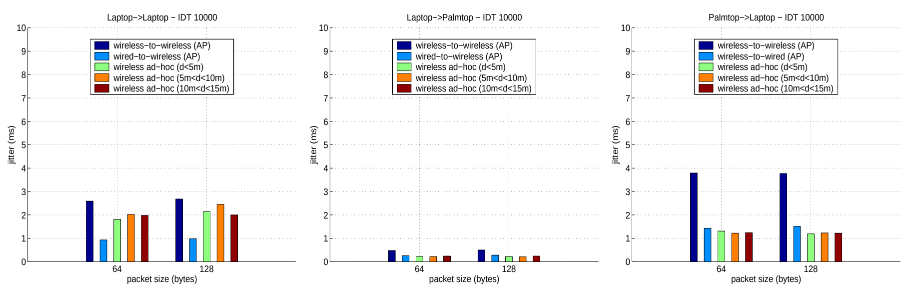


Figure 15: UDP jitter for IDT $= \frac{1}{10000}$ s

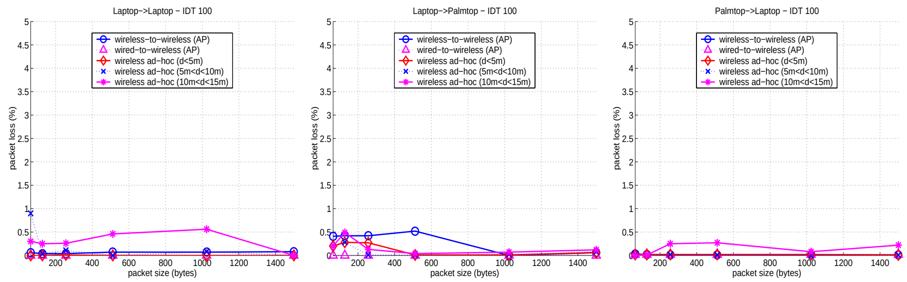


Figure 16: UDP packet loss for IDT $= \frac{1}{100}$ s

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 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 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.

2.1.5 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 UDP and TCP behavior over wireless scenario. TCP over wireless issues have been extensively discussed and several innovative proposals have been presented [17] [19]. 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

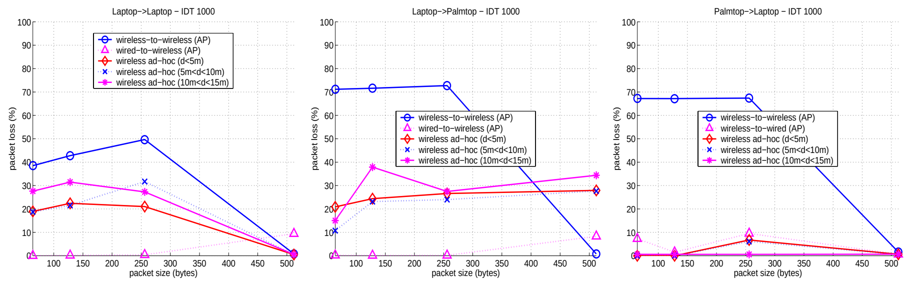


Figure 17: UDP packet loss for IDT $= \frac{1}{1000}$ s

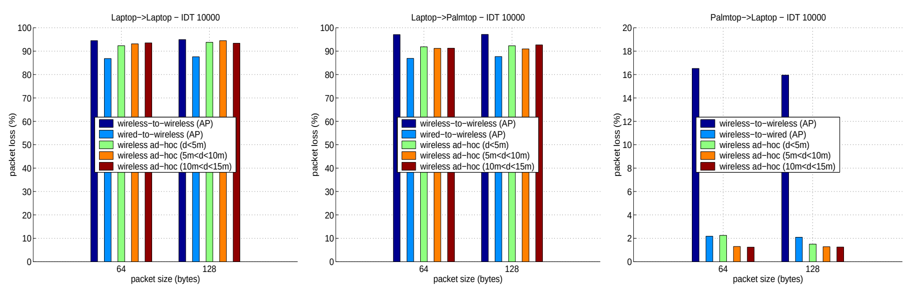


Figure 18: UDP packet loss for IDT $= \frac{1}{10000}$ s

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 [20] 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 [17] 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 2.1.1.

Low traffic load As we have anticipated in Section 2.1.1, 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 $1500bytes$ 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.

Medium traffic load In our opinion, results obtained in the *medium* traffic load analysis represent one the most important contributes of this work. 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 $256bytes$; (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 $3.5Mbps$; (iv) at higher packet size ($PS > 512bytes$) 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 ($PS < 512bytes$) 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);

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 (2.1.5): 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 do not provide this graphics because we have a low achieved throughput (in the case of PS = $1500 bytes$ and IDT = $\frac{1}{10000}$ we have a data rate equal to $120 Mbps$ over a $11 Mbps$ channel).

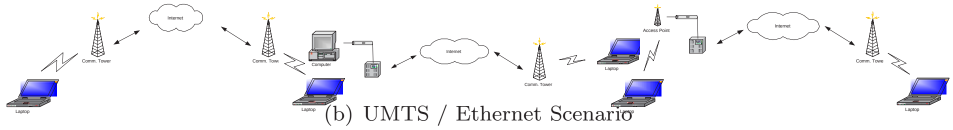
2.2 Experimentation with Wireless Wide Area Network

2.2.1 Scenario

In this Paragraph we analyze the case where one the end is represented by an UMTS node. More precisely, we study the performance related to the communication between an UMTS device and (i) Ethernet node, (ii) WLAN 802.11b node, and (iii) GPRS node. Table 4 summarizes the analyzed "*Service Conditions*". Therefore, we analyzed a subset of the considered "*Service Conditions*" set (see Figure 1). With the term "UMTS Uplink" we mean a scenario where at sender side there is an UMTS station. Also, with the term "UMTS Downlink" we mean a scenario where at receiver side there is an UMTS station.

Table 4: Wireless Wide Area Experimentations: Analyzed *Service Conditions*

Sender Station	Receiver Station
UMTS	Ethernet
UMTS	Wi-Fi WLAN 802.11b
UMTS	GPRS
Ethernet	UMTS
Wi-Fi WLAN 802.11b	UMTS
GPRS	UMTS



(a) UMTS / GPRS Scenario (b) UMTS / Ethernet Scenario (c) UMTS / Wi-Fi Scenario

Figure 19: Wireless Wide Area Experimentation Scenario

In the case of “UMTS Uplink” we measured both UDP and TCP scenario. Due to the private addressing of UMTS network, in the case of “UMTS Downlink” we measured only the TCP scenario. Indeed, thanks to D-ITG feature it is possible to open the TCP in both the directions.

In this kind of experimentation we prefer to use internal modems. Due to the supported GPRS and UMTS internal modem drivers we performed our experimentations over Windows XP platforms. This choice was also adopted after a survey on the diffusion of UMTS and GPRS internal modems.

According to the general network scenario presented in Figure 1, in Figure 19(a), Figure 19(b), and in Figure 19(c) we show the *Service Conditions* under our attention. Table 5 specifies the parameters characterizing each analyzed *Service Conditions*.

Table 5: Devices Description

Devices	Description
Laptop 1	Toshiba Satellite Pro 4300, Intel PIII 650 Mhz Main Memory 186MB, Cache 256KB, Windows 2000 Prof. O.S.
Laptop 2	Toshiba Satellite S5200-801, Intel P4 2,0 Ghz, Main Memory 512MB, Cache 512KB, O.S. Windows XP
PC 1	Intel P4 2,6 Ghz Main Memory 1024MB, Cache 512KB, Windows XP O.S.
GPRS Modem	Merlin G301 - Novatel Wireless
UMTS Modem	Merlin U530 - Fast Mobile Card 3
WLAN NIC	DLink Air-Plus
Ethernet NIC	3Com EtherLink XL 10/100

We used UMTS modem on ’Laptop 2’ and the GPRS modem and the Wi-Fi card on ’Laptop 1’. ’PC 1’ represents the station attached to Ethernet network. Also in this case we provide a complete characterization of throughput, packet loss, jitter and delay (round trip time). In addition, in this Paragraph we provide an analysis of TCP measuring two different situation. In the first one we used the Nagle algorithm; the second one is characterized by traffic generation with Nagle algorithm disabled. Moreover, at the end of this section we show another point of view of our results. We show the behavior of measured parameters as a function of the time. More precisely, after presenting the average value of throughput, delay, jitter and packet loss, we present their instantaneous value during the experiment interval time.

2.2.2 Experimental Parameters

The tested *Service Conditions* presented in Table 4 have been analyzed in the case of *low* traffic load ($\leq 1.2Mbps$). This choice is due to nominal bandwidth of used UMTS network ($384Kbps$ in the download link and $64kbps$ in the upload link)

To achieve the generated bit rate we used IDT equal to $\frac{1}{100}$ and PS varying in the interval $32, 15000bytes$. Table 6 reports the theoretical generated bit rate for each pair PS, IDT. The rows of Table 6 represent the points where we measured the average value of throughput, jitter, packet loss, and delay. We provide this kind of measurement in graphs where the measured parameter is showed as function of the packet size.

Table 6: Traffic Parameters

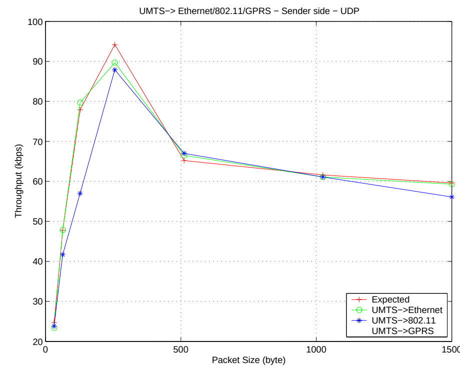
IDT	PS	Generated Bit Rate
1/100 s	32 bytes	26,1 Kbps
1/100 s	64 bytes	51,2 Kbps
1/100 s	128 bytes	102,4 Kbps
1/100 s	256 bytes	204,8 Kbps
1/100 s	512 bytes	409,6 Kbps
1/100 s	1024 bytes	819,2 Kbps
1/100 s	1500 bytes	1,200 Kbps

By using the D-ITG features we stored log files at both sender and receiver side. In this case we provide a complete characterization at both end of the communication. The sender side analysis give us the possibility to understand and isolate the device dependencies and the network dependencies. Obviously, the sender side log provide us the difference between the theoretical and real generated bit rate. For each combination of IDT and PS, each experiment was repeated ten times. In order to avoid measurement errors due to unattended external phenomena (especially over UMTS and GPRS networks) we performed the measurements interleaving for both packet size and "*Service Conditions*". The measurement interval was $09:00a.m. - 07:00p.m.$ Finally, the duration of each test was equal to $30s$.

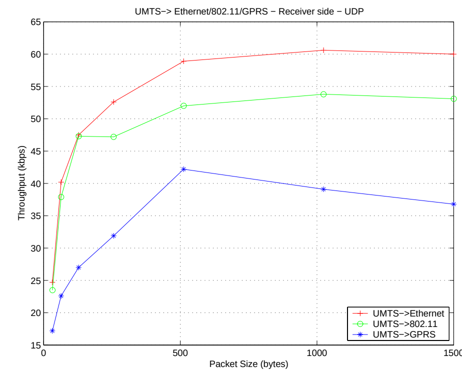
2.2.3 Throughput Analysis

Graphs in Figure 20 show the throughput behavior for each "*Service Conditions*". Each graphic contains three plots (UMTS/Ethernet, UMTS/GPRS, UMTS/WLAN 802.11b).

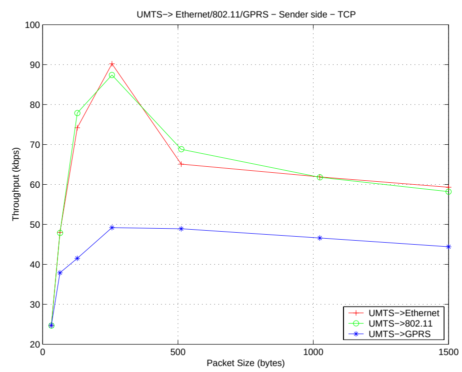
The left column of Figure 20 represent the sender behavior, the right column the receiver one. The first row is related to UDP, the second to TCP.



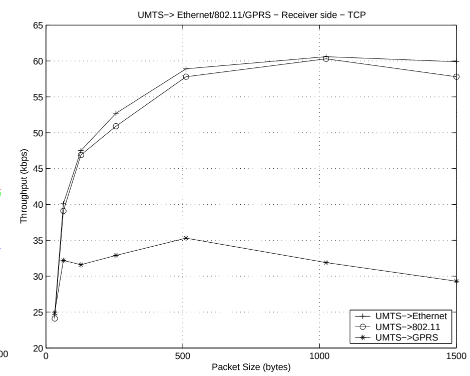
(a) UMTS to Ethernet/802.11b/GPRS, UDP (Sender Side)



(b) UMTS to Ethernet/802.11b/GPRS, UDP (Receiver Side)



(c) UMTS to Ethernet/802.11b/GPRS, TCP (Sender Side)



(d) UMTS to Ethernet/802.11b/GPRS, TCP (Receiver Side)

Figure 20: 'UMTS Uplink' Throughput Behavior

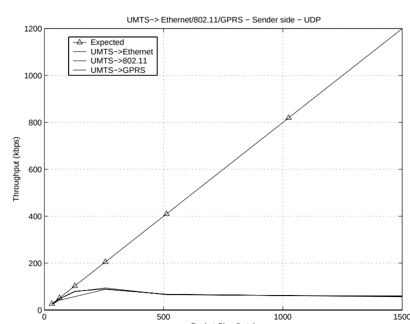


Figure 21: UMTS to Ethernet/802.11b/GPRS, UDP (Comparison between theoretical and real bit rate)

In the case of UDP, there is a significant difference between the generated and received traffic. At sender side the maximum bit rate was equal to ca. $90kbps$ in correspondence of PS equal to $256byte$. At receiver side the maximum bit rate was equal to ca. $64kbps$ in correspondence of PS equal to $1024byte$. This bit rate value is related to the UMTS uplink bandwidth. When at receiver side a GPRS end is present, we measured a maximum bit rate equal to ca. $46kbps$ (this value represents the GPRS downlink bandwidth). As for the understanding of device and network dependencies, Figure 21 shows the difference between the theoretical and real generated bit rate.

In the case of TCP (Figures 20(c) and 20(d)), we measured a maximum bit rate equal to ca. $90kbps$ in correspondence of PS equal to $256byte$. It is worth noting that in the case of TCP there is a lower bit rate than that one measure in the UDP scenario when a receiver side there is a GPRS end. TCP suffers the dynamics of GPRS network.

Figures 22 and 23 show the behavior of TCP and UDP throughput as function of the time. Instantaneous maximum values were found in the case of $256byte$. For this PS value we measured $115kbps$ in the case of TCP and $150kbps$ in the case of UDP.

Figures 24(a) and 24(b) present the sender and receiver behavior in the case of 'UMTS downlink'. In the case of Ethernet/UMTS communication the sender and receiver trend presents the same profile. This is not true in the case of Wi-Fi or GPRS networks. Independently of the technology used at sender side the maximum throughput was obtained in the case of $1500byte$. The maximum values was measured in the case of Ethernet and it was equal to $160kbps$

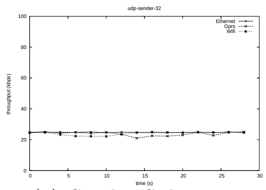
Figure 25 shows the behavior of TCP throughput as function of the time in the case of 'UMTS downlink'.

2.2.4 Jitter Analysis

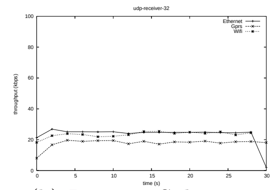
Figures 26(a), 26(b), and 26(c) show the jitter behavior in the analyzed "*Service Condition*".

As showed in Figures 26(a) and 26(b) the jitter grows when PS grows. This behavior is confirmed to the plots depicted in Figure 27 (in this Figure the jitter is depicted as a function of the time for each packet size). More precisely, for a packet size equal to $1500byte$ we measured an average maximum jitter equal to $0.340s$ and an instantaneous maximum jitter equal to $0.7s$.

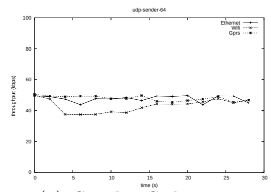
Figure 26(c) presents the jitter behavior in the case of 'UMTS downlink' scenario. In the case of Ethernet and Wi-Fi the jitter is always lower than $0.100s$. When we used the GPRS network, for a packet size equal to $1500byte$ we measured an average maximum jitter equal to $0.810s$ and an instantaneous maximum jitter equal to $2.2s$.



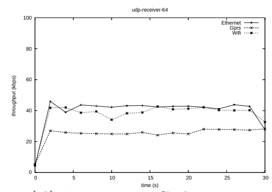
(a) Sender Side -c 32



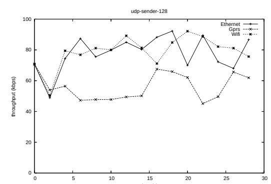
(b) Receiver Side -c 32



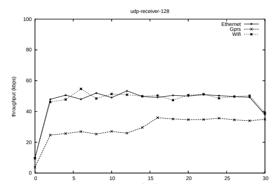
(c) Sender Side -c 64



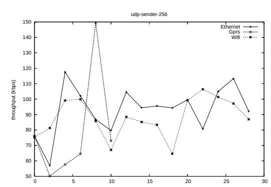
(d) Receiver Side -c 64



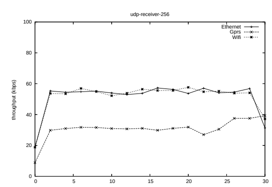
(e) Sender Side -c 128



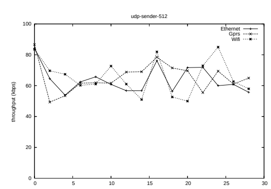
(f) Receiver Side -c 128



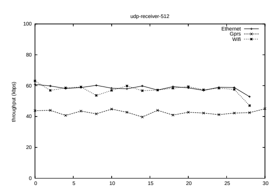
(g) Sender Side -c 256



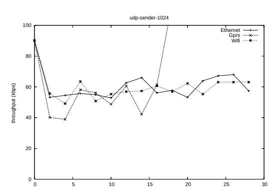
(h) Receiver Side -c 256



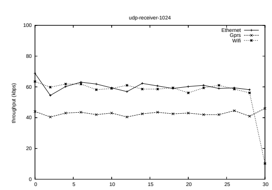
(i) Sender Side -c 512



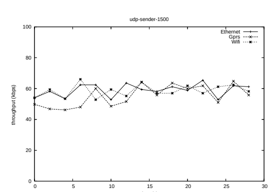
(j) Receiver Side -c 512



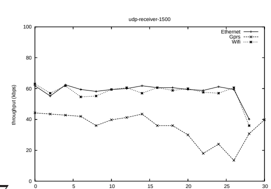
(k) Sender Side -c 1024



(l) Receiver Side -c 1024

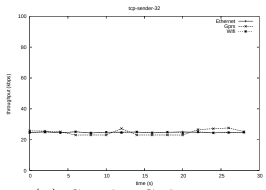


(m) Sender Side -c 1500

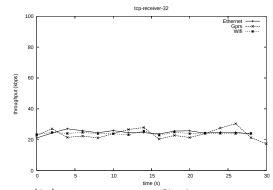


(n) Receiver Side -c 1500

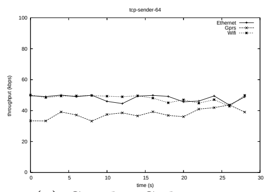
Figure 22: Throughput instantaneous values during the experiment interval time ('UMTS uplink' and UDP)



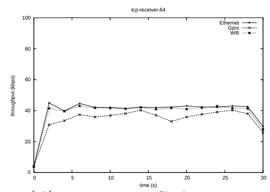
(a) Sender Side -c 32



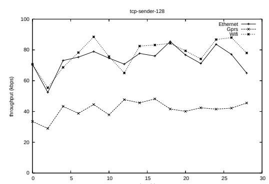
(b) Receiver Side -c 32



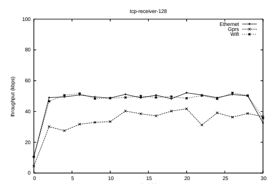
(c) Sender Side -c 64



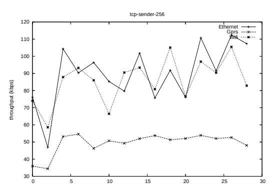
(d) Receiver Side -c 64



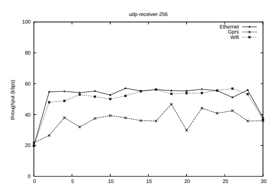
(e) Sender Side -c 128



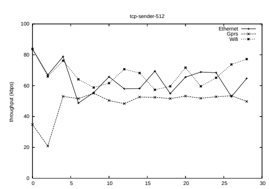
(f) Receiver Side -c 128



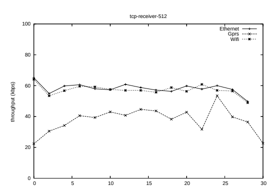
(g) Sender Side -c 256



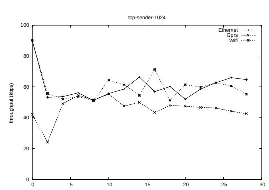
(h) Receiver Side -c 256



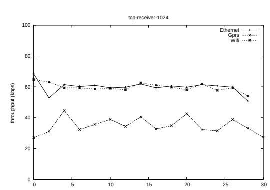
(i) Sender Side -c 512



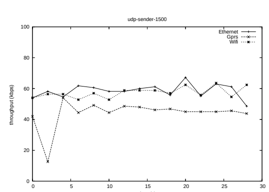
(j) Receiver Side -c 512



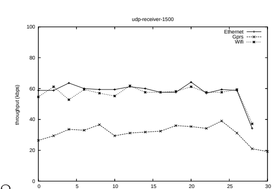
(k) Sender Side -c 1024



(l) Receiver Side -c 1024

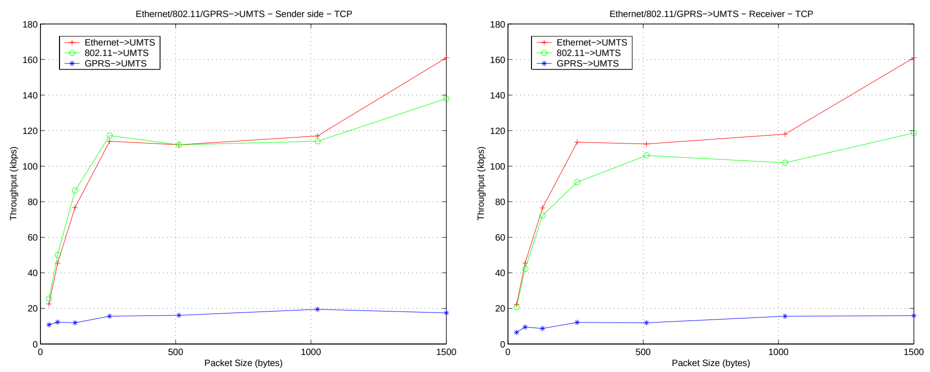


(m) Sender Side -c 1500



(n) Receiver Side -c 1500

Figure 23: Throughput instantaneous values during the experiment interval time ('UMTS uplink' and TCP)



(a) Ethernet/802.11b/GPRS to UMTS, TCP (Sender Side)

(b) Ethernet/802.11b/GPRS to UMTS, TCP (Receiver Side)

Figure 24: 'UMTS Downlink' Throughput Behavior

2.2.5 Packet loss Analysis

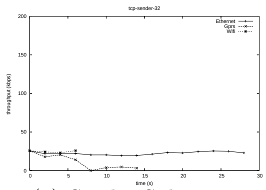
Figure 30 shows the packet loss behavior. This graph presents a gaussian profile in the packet size range $[32, 512]bytes$ for each kind of receiver network technology. We measured the maximum packet loss for a packet size equal to $256bytes$. For this value of packet size, Figure 20(a) indicates the maximum throughput value at sender side. The same point of Figure 20(b) does not present the same throughput value of Figure 20(a). This confirms the experimented packet loss percentage. When the packet size grows the packet loss decreases. This is due to the different bit rate at sender side. Indeed, for high values of the packet size the real sender throughput is low and the receiver is able to receive all sent packets.

2.2.6 Delay (Round Trip Time) Analysis

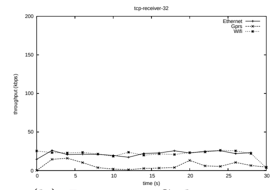
In order to overcame the limitations related to the clock synchronization between the two communicating ends in this preliminary analysis over heterogeneous networks we prefer to use the Round Trip Time (RTT) instead of One Way Delay (OWD).

As previously said, in the case of TCP delay analysis we performed two kind of experimentations. In the first we provide results when the Nagle algorithm is present. The second provide results when the Nagle algorithm is disabled.

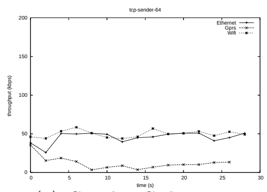
Figure 31 shows the 'UMTS downlink' RTT behavior in the case of TCP. The worst case was experimented in the GPRS case. More precisely, we experimented the maximum RTT when the Nagle algorithm is disabled and the average maximum RTT ($34s$) was measured in the case of $256byte$. Figure 36 confirms that the experiment interval time (in the case of GPRS *to* UMTS) is greater than $30s$. This is due to the high delay experimented by packets in the



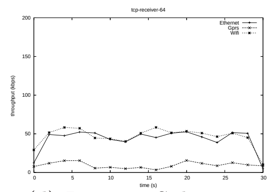
(a) Sender Side -c 32



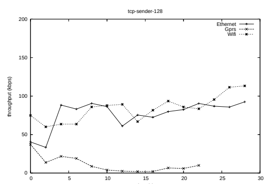
(b) Receiver Side -c 32



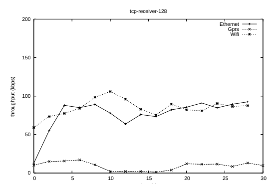
(c) Sender Side -c 64



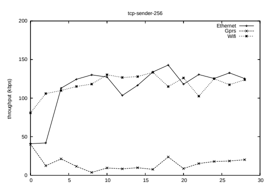
(d) Receiver Side -c 64



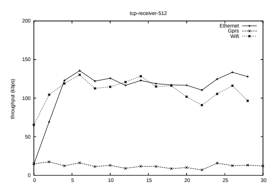
(e) Sender Side -c 128



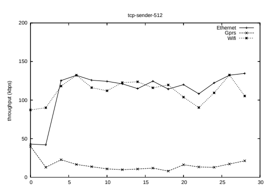
(f) Receiver Side -c 128



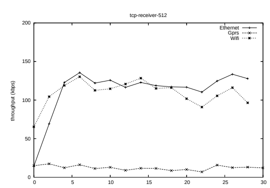
(g) Sender Side -c 256



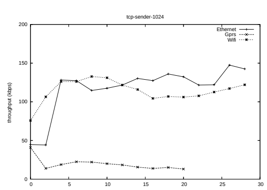
(h) Receiver Side -c 256



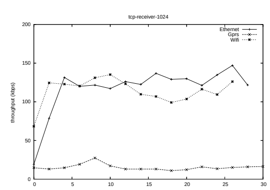
(i) Sender Side -c 512



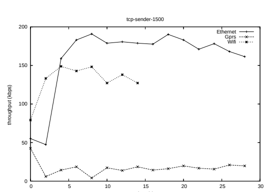
(j) Receiver Side -c 512



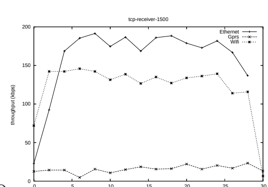
(k) Sender Side -c 1024



(l) Receiver Side -c 1024

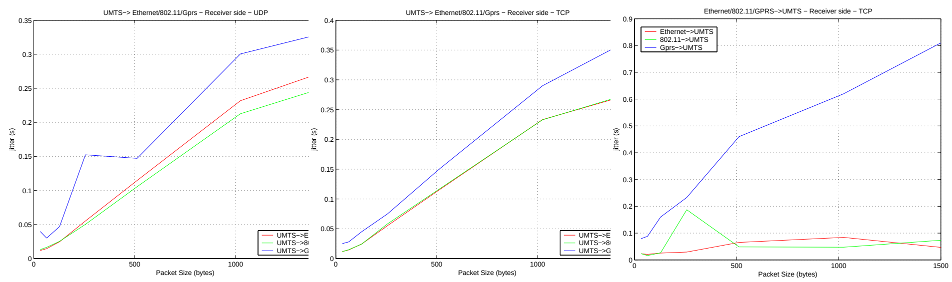


(m) Sender Side -c 1500



(n) Receiver Side -c 1500

Figure 25: Throughput instantaneous values during the experiment interval time ('UMTS downlink' and TCP)



(a) 'UMTS Uplink' (UDP) (b) 'UMTS Uplink' (TCP) (c) 'UMTS Downlink' (TCP)

Figure 26: Jitter Analysis

path receiver to sender.

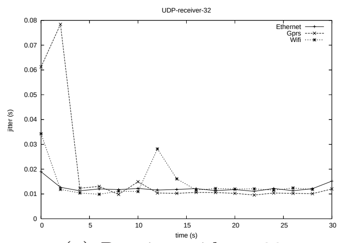
Figure 32 and Figure 34 show the 'UMTS uplink' RTT behavior in the case of TCP. Also in this case we experimented the maximum RTT when the Nagle algorithm is disabled.

Figure 33 shows the 'UMTS uplink' RTT behavior in the case of UDP. This Figure contains results related to UMTS *to* Ethernet/Wi-Fi/GPRS. The left side of Figure 33 shows the RTT behavior related to the UMTS *to* Ethernet/Wi-Fi in the packet size range equal to $[32, 1500] bytes$. In the case of UMTS *to* GPRS we measured RTT results in the packet size range equal to $[128, 1500] bytes$. This is due to frequent network flapping for packet sizes equal to $32 bytes$ and $64 bytes$. Indeed, in the case of these packet sizes we experimented continuous service interruption of the used GPRS network. The maxumum RTT value is equal to $21s$ for packet sizes equal to $1024 bytes$ and $128 bytes$. Also in this case the worst RTT values was measured in the case of UMTS *to* GPRS scenario.

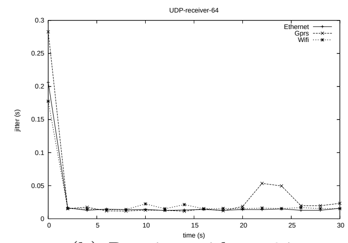
Graphs of Figure 35 show the instantaneous RTT behavior as a function of the time for each packet size.

2.3 Conclusions and Ongoing Work

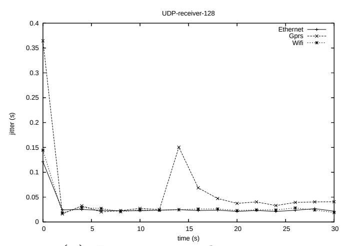
All over the world heterogeneous wireless networks are being used to support mobile services and innovative multi-scenario applications. Unfortunately, application performance over heterogeneous networks is severely impacted by problems of high and variable round trip times, fluctuating bandwidths, frequent 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



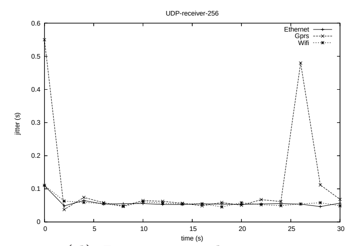
(a) Receiver side -c 32



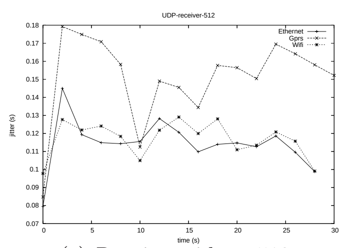
(b) Receiver side -c 64



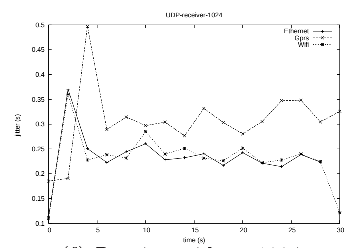
(c) Receiver side -c 128



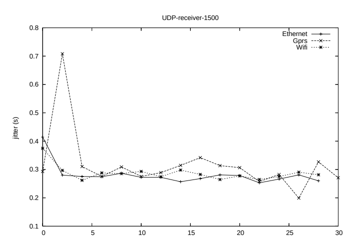
(d) Receiver side -c 256



(e) Receiver side -c 512



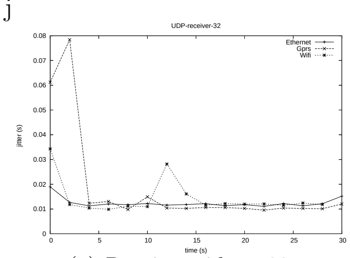
(f) Receiver side -c 1024



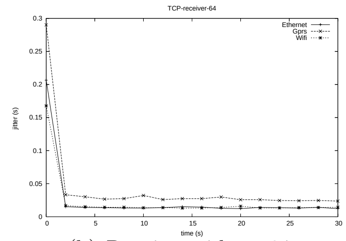
(g) Receiver side -c 1500

Figure 27: Jitter instantaneous values during the experiment interval time ('UMTS uplink' and UDP)

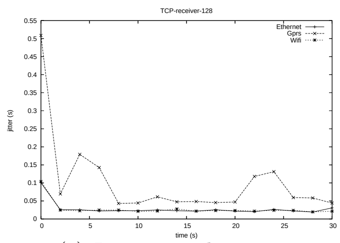
j



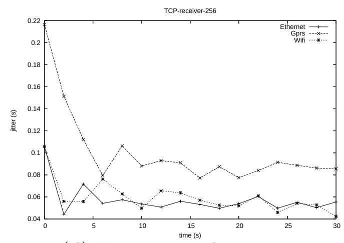
(a) Receiver side -c 32



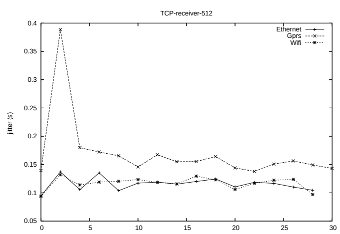
(b) Receiver side -c 64



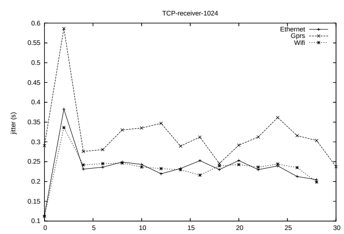
(c) Receiver side -c 128



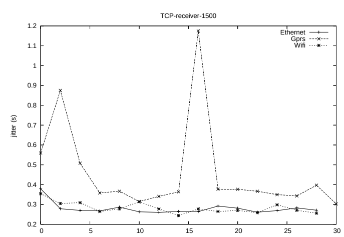
(d) Receiver side -c 256



(e) Receiver side -c 512

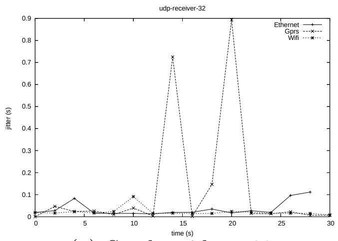


(f) Receiver side -c 1024

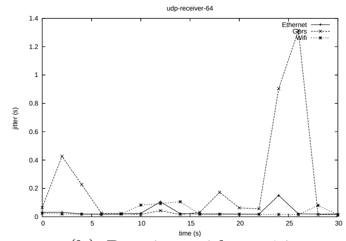


(g) Receiver side - c 1500

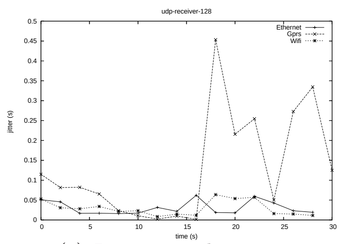
Figure 28: Jitter instantaneous values during the experiment interval time ('UMTS uplink' and TCP)



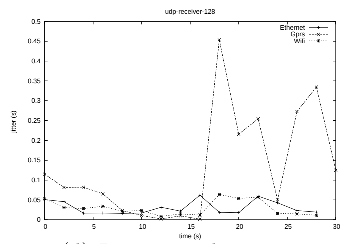
(a) Sender side -c 32



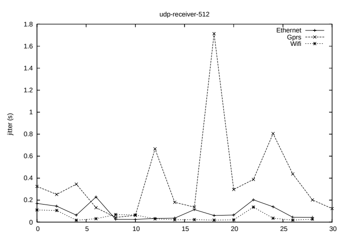
(b) Receiver side -c 64



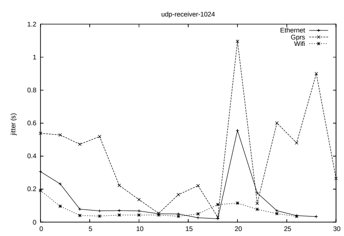
(c) Receiver side -c 128



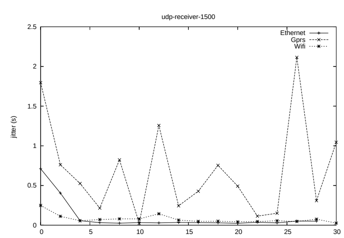
(d) Receiver side -c 256



(e) Receiver side -c 512



(f) Receiver side -c 1024



(g) Receiver side - c 1500

Figure 29: Jitter instantaneous values during the experiment interval time ('UMTS downlink' and UDP)

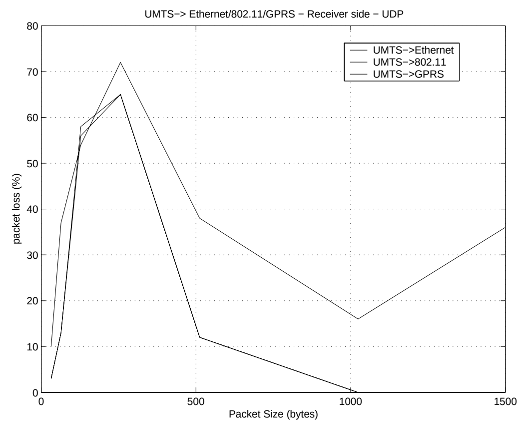
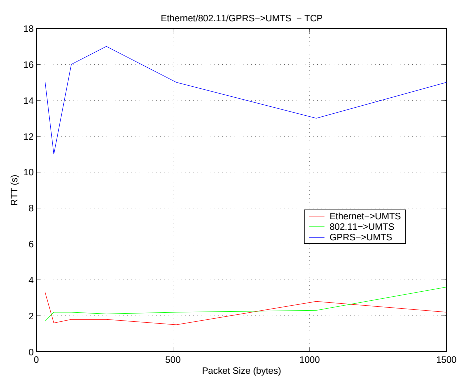
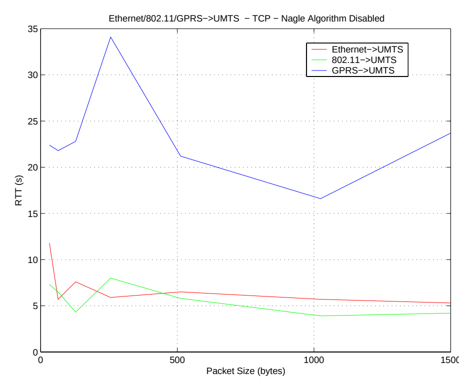


Figure 30: Packet loss analysis, UDP (Receiver Side)

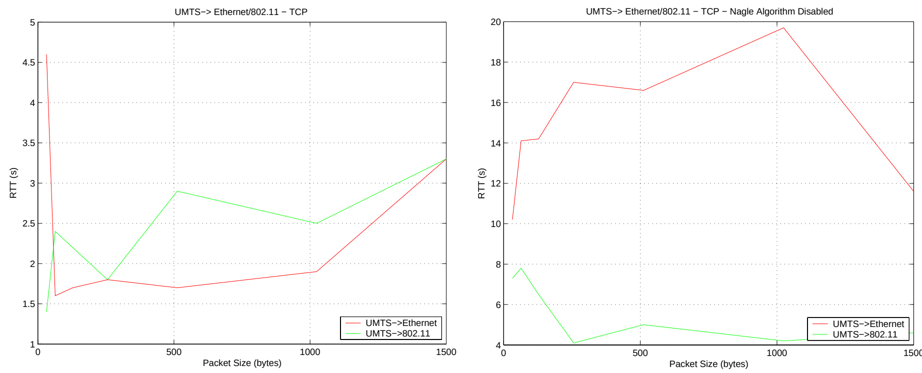




(a) Ethernet/802.11/Gprs to UMTS, TCP (Nagle Algorithm Disabled)

(b) Ethernet/802.11/Gprs to UMTS, TCP (Nagle Algorithm)

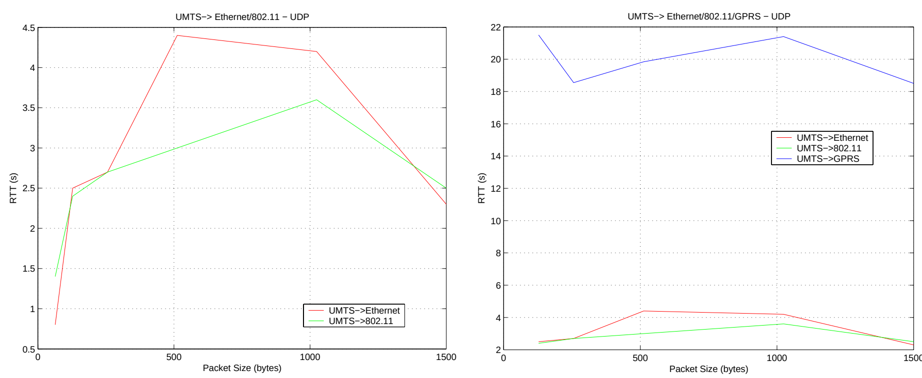
Figure 31: 'UMTS downlink' RTT analysis (TCP)



(a) UMTS to Ethernet/802.11, TCP (Nagle Algorithm Disabled

(b) UMTS to Ethernet/802.11, TCP (Nagle Algorithm

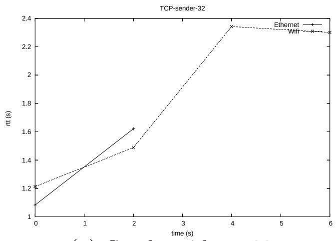
Figure 32: 'UMTS uplink' RTT analysis (TCP)



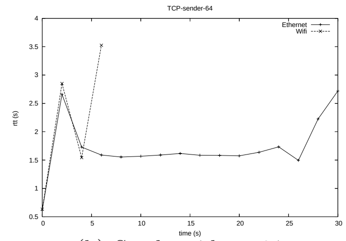
(a) UMTS to Ethernet/802.11b, UDP

(b) UMTS to Ethernet/802.11b/GPRS, UDP

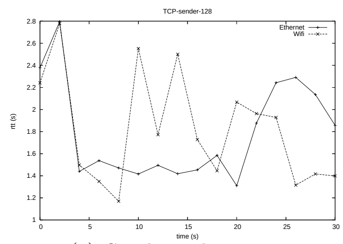
Figure 33: 'UMTS uplink' RTT analysis (UDP)



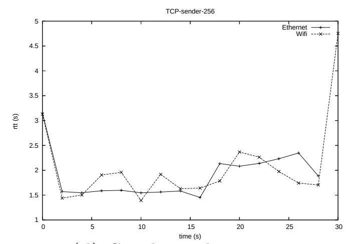
(a) Sender side -c 32



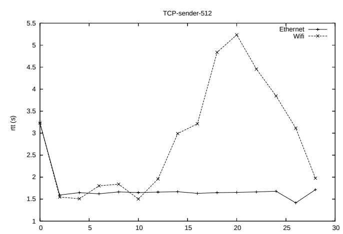
(b) Sender side -c 64



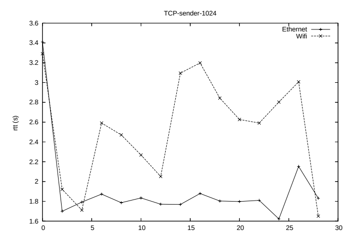
(c) Sender side -c 128



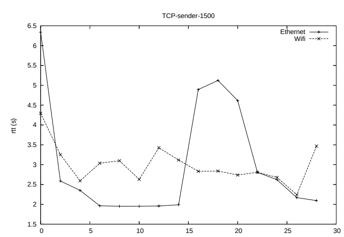
(d) Sender side -c 256



(e) Sender side -c 512

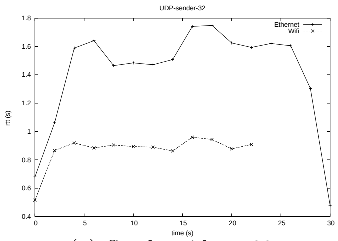


(f) Sender side -c 1024

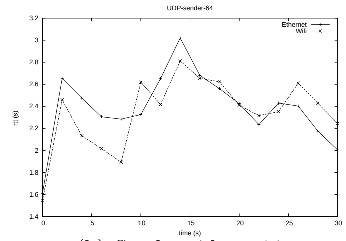


(g) Sender side - c 1500

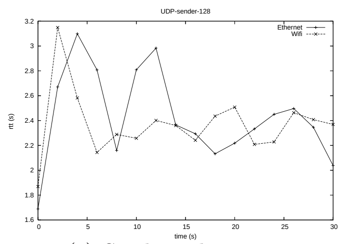
Figure 34: Round Trip Time instantaneous values during the experiment interval time ('UMTS uplink' and TCP)



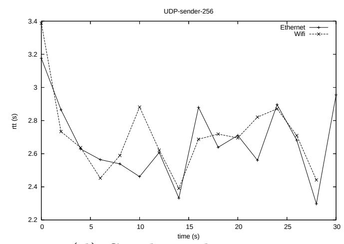
(a) Sender side -c 32



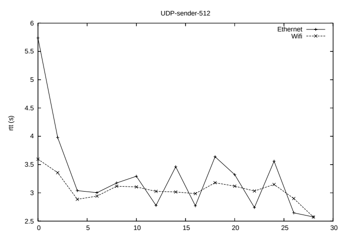
(b) Sender side -c 64



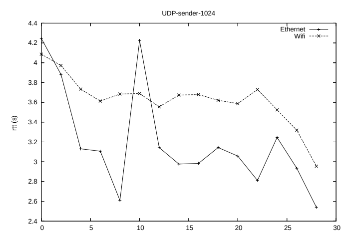
(c) Sender side -c 128



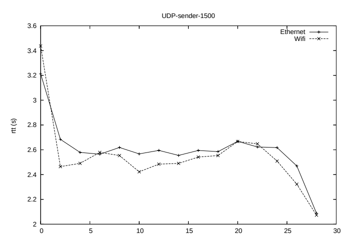
(d) Sender side -c 256



(e) Sender side -c 512

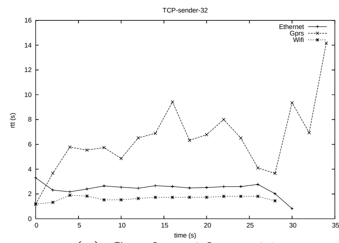


(f) Sender side -c 1024

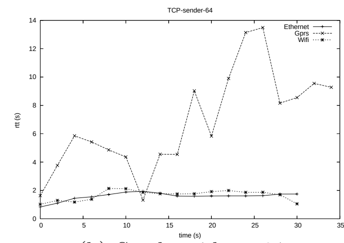


(g) Sender side -c 1500

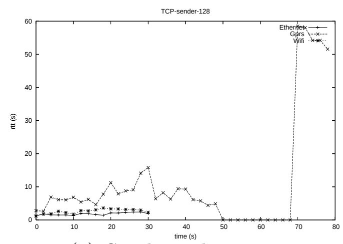
Figure 35: Round Trip Time instantaneous values during the experiment interval time ('UMTS uplink' and UDP)



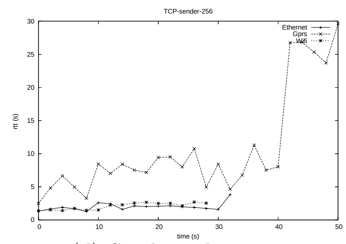
(a) Sender side -c 32



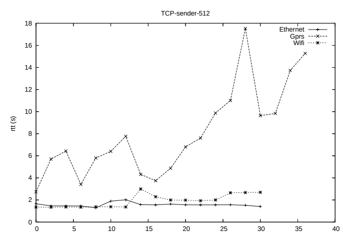
(b) Sender side -c 64



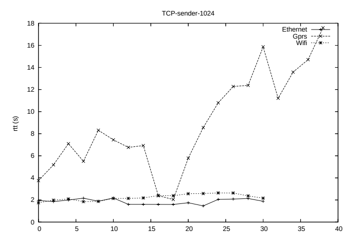
(c) Sender side -c 128



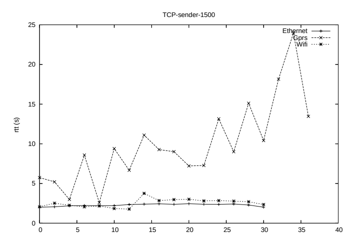
(d) Sender side -c 256



(e) Sender side -c 512



(f) Sender side -c 1024



(g) Sender side- c 1500

Figure 36: Round Trip Time instantaneous values during the experiment interval time ('UMTS downlink' and TCP)

"*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 test-bed 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 work. 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 test-bed 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 work 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.

3 Discussion and Related Work

This section compares and contrasts our performance framework with some other studies. There are several simulation [15] and analytical [16] 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. As for real measurement framework, 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 software: 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 [17] a discussion on the problems arising when the TCP/IP protocol suite is used to provide Internet connectivity over existing wireless links is presented. [18] 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.

To the best of our knowledge, our work extends previous works on TCP and UDP performance in many directions. More precisely, we present a complete evaluation, from the application point of view, of heterogeneous wireless networks in terms of a wide range of QoS parameters. 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 Condi-*

tion" 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.

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