Efficiency Study of TCP Protocols in Infrastructured Wireless Networks checked Abstract Traditional TCP protocols treat all packet loss as a sign of congestion. Their inability to recognize non-congestion related packet loss has significant effects on the communication efficiency in the wireless networks. Recently proposed protocols such as Freeze-TCP, TCP-Probing, TCPWestwood/Westwood+, TCP Veno, TCP-Jersey, and JTCP, all imp rove over th e traditional TCP p ro to cols. This per reports a quantitative comparison of recent protocols against the currently most often used namely, TCP SACK, TCP NewReno, and TCP Vegas. Simulation tests were designed for various network layouts, and with differing external interferences in an attempt to most accurately simulate real-life scenarios. To carry out these comparisons, the performance of each protocol was measured based on three benchmark	Nghiên cứu hiệu suất của các giao thức TCP trong các mạng không dây ở chế độ Infrastructure chế độ Infrastructure (Infrastructure mode): chế độ sử dụng trung tâm phát sóng là Access point Tóm tắt Các giao thức TCP truyền thống xem mọi dạng mất gói tin là dấu hiệu của tình trạng tắt nghẽn. Sự bất lực của chúng trong việc nhận ra hiện tượng mất gói tin không liên quan đến tắt nghẽn có ảnh hưởng đáng kể đến hiệu suất truyền thông trong các mạng không dây. Gần đây, một số giao thức được đề xuất chẳng hạn như Freeze-TCP, TCP-Probing, TCPWestwood/Westwood+, TCP Veno, TCP-Jersey, và JTCP, đều có những tính năng cải thiện hơn so với các giao thức TCP truyền thống. Bài báo này trình bày các kết quả nghiên cứu về so sánh định lượng của các giao thức gần đây với các giao thức được sử dụng thường xuyên nhất, cụ thể là TCP SACK, TCP NewReno, và TCP Vegas. Chúng tôi thiết kế các thử nghiệm mô phỏng cho các sơ đồ mạng khác nhau, và với các nhân tố can thiệp bên ngoài khác nhau nhằm hướng đến việc mô phỏng chính xác nhất tình huống thực tế. Để thực hiện những so sánh này, chúng tôi đo hiệu suất của mỗi giao thức dựa trên ba chỉ tiêu phẩm chất tiêu chuẩn: thông lượng (lượng thông	

metrics: throughput, average congestion window, and time to complete a file transfer. 1 Introduction Even though the Internet was originally designed to support its operation over various transport media [3], most of its components were optimized for wired networks. The TCP protocol, which facilitates the majority of the Internet services (Web, FTP, Telnet) is one of those mechanisms that are, by its design, inherently inefficient in the wireless networks. This is the motivation behind the continuous research in this field. Different paradigms have been used to develop solutions to the problem of TCP in wireless networks, but only few of them are actually possible to implement. A well researched qualitative study of such solutions is presented in [12]. This paper focuses on evalu-ating and comparing implementable solutions such as TCP Westwood/Westwood+, TCP Veno, TCP-Jersey, and JTCP aimed at improving the efficiency of the TCP protocol in wireless and	tin truyền qua một mạng trong một đơn vị thời gian), cửa sổ tắt nghẽn mạng trung bình, và thời gian hoàn thành truyền tập tin. 1 Giới thiệu Mặc dù ban đầu, Internet được thiết kế để có thể hoạt động trên các môi trường truyền tải khác nhau [3], đa số thành phần của nó được tối ưu cho các mạng dây (mạng có dây). Giao thức TCP, tạo điều kiện thuận lợi cho phần lớn các dịch vụ mạng (Web, FTP, Telnet) là một trong những cơ chế vốn đã không hiệu quả trong các mạng không dây do thiết kế của nó. Đây là động lực chính thúc đẩy những nghiên cứu liên tục trong lĩnh vực này. Người ta đã sử dụng các mô hình khác nhau để tìm ra giải pháp cho vấn đề TCP trong các mạng không dây, nhưng chỉ một số ít trong chúng khả thi. Một nghiên cứu định lượng tốt về những giải pháp này được trình bày trong tài liệu tham khảo [12]. Bài báo này tập trung vào vấn đề đánh giá và so sánh các giải pháp khả thi chẳng hạn như TCP Westwood/Westwood+, TCP Veno, TCP-Jersey, và JTCP hướng đến việc cải thiện giao thức TCP trong các mạng không dây và không đồng nhất. Các giao thức này được kiểm tra và đối chiếu với các giao thức TCP được sử dụng thường xuyên nhất TCP SACK, TCP NewReno, và TCP Vegas trong các sơ đồ mạng khác nhau, trong các trường hợp khác	

heterogeneous networks. These protocols are tested against the currently most often used protocols TCP SACK, TCP NewReno, and TCP Vegas in various network layouts, under different circumstances, and with differing external interferences in an attempt to most accurately simulate real-life scenarios. The ultimate goal is to isolate the most efficient solution to the non-congestion packet loss problem of the TCP protocol in wireless networks. The rest of the document is organized as follows. Section 2 offers an overview of the traditional TCP protocol and piats-mechanisms. Section 3 discusses existing problems that TCP protocol faces in wireless networks. Section 4 introduces the most recent solutions to the wireless packet loss TCP problem. Section 5 elaborates on the complexities of the research and the methods used in its implementation. Section 6 looks at the results and singles out efficient protocols and their mechanisms. Section 7 maps out the future research goals.	nhau, và với các nhân tố tác động bên ngoài khác nhau để mô phỏng chính xác nhất các tình huống thực tế. Mục tiêu cuối cùng là thu được các giải pháp hiệu quả nhất cho vấn đề mất gói tin không tắt nghẽn của giao thức TCP trong các mạng không dây khác nhau. [redacted]	

2 Original Design of TCP Protocols TCP identifies a transport layer protocol that provides a reliable and in-order delivery of data between two hosts. TCP is a defensive protocol highly sensitive to network congestion. To ensure a reliable communication, TCP uses an acknowledgement packet (ACK) as a response to a successfully delivered packet. ACKs are cumulative; each ACK carries the sequence number of the next data octet expected to be received. In case of a lost packet, the next packet received will return the ACK of the packet received prior to the loss, causing the sender to recognize two identical ACKs. These are called duplicate ACKs and are considered a signal of a packet loss. The two most common TCP distributions, TCP Tahoe [10] and TCP Reno [11], have mechanisms to compensate for the efficiency drop due to the congestion related packet loss. These mechanisms are proven to be successful in wired networks [6]. In fact,	[redacted] **ban đầu** [redacted] giao tiếp [redacted] tin [redacted] có tính chất [redacted] của [redacted] dữ liệu tiếp theo [redacted] [redacted] [redacted] phân phối [redacted]	

both protocols were designed for wired networks only. Consequently, the only major type of packet loss the wired networks experience is assumed to be caused by network congestion. Random loss occurs less than 1% of the time [11]. Thus, Tahoe and Reno interpret every loss as a sign of congestion and invoke the mechanisms to control it. 3 Non-Congestion Packet Loss In wireless networks, the occurrence of packet loss does not necessarily imply congestion.The reliability of the wireless links depends on the conditions of the environment in which they are located and the objects (mobile or stationary) that obstruct signal propagation. Two basic types of wireless non-congestion related loss can be identified. The first one is the random packet loss that manifests itself through bit corruption. Such packets are discarded by the routers or the end hosts. The second type of packet loss is the disconnection packet loss that occurs when the mobile host com-pletely disconnects from the wireless network. This type	[redacted] xuất hiện ít hơn [redacted] [redacted] gói tin [redacted] hoàn toàn với [redacted]	

of packet loss is a characteristic of infrastructured networks and occurs either when a mobile host becomes physically too distant from the base station or when it moves between two adjacent wireless networks (handoff). Unfortunately, both Tahoe and Reno cannot address non-congestion loss and will interpret such packet loss as a sign of congestion, triggering its defensive and conservative congestion control mechanism. Hence, the main focus in all TCP solutions for wireless and heterogeneous networks is the ability of the protocol to distinguish between these types of packet loss and respond appropriately. 4 End-to-End Solutions This section focuses on four solutions proposed in the last four years. These solutions represent pure end-to-end protocols. 4.1 TCP Westwood TCP Westwood (TCPW) [5] does not rely on the traditional additive increase multiplicative decrease (AIMD) algorithm but instead on a more aggressive estimation of the available bandwidth after a	[redacted] các mạng kiểu infrastructured [redacted] và sự mất mát gói tin như thế [redacted] kiểm soát [redacted] bảo vệ và duy trì [redacted] [redacted] trong việc [redacted] mất gói tin [redacted] [redacted] thuần túy [redacted] tăng cộng giảm nhân [redacted] băng thông có sẵn [redacted]	

loss event has occurred. Thus, Westwood relies on a dynamic algorithm that inferes the network state from the received ACKs. This information is used in an optimistic statistical estimation of the available bandwidth. Since the bandwidth changes with each packet sent, Westwood performs bandwidth estimation upon the reception of each ACK. 4.2 TCP-Jersey TCP-Jersey [15] not only addresses the problem of non-congestion random loss, but also attempts to deal with the congestion loss more efficiently. To explicitly differentiate between the congestion and non-congestion loss, TCP-Jersey uses two fundamentally different and separate mechanisms: one for the aggressive modification of the congestion window in case of congestion related losses and the other for dealing with non-congestion related packet losses. To deal with congestion loss, TCP-Jersey uses a dynamic algorithm for changing the size of the congestion window. Much like Westwood, it attempts	[redacted] phỏng đoán [redacted] [redacted] linh hoạt [redacted] [redacted]	

to aggressively estimate the congestion window after the loss has occurred using the available bandwidth estimator (ABE) algorithm [15]. The second mechanism detects the type of packet loss via a modified version of the explicit congestion notification scheme (ECN) [7]. ECN works in cooperation with ran-dom early detection (RED) [8] to probabilistically mark the packets with the congestion bit when the router queue exceeds the minimum threshold and drop every packet when the queue exceeds the maximum threshold. 4.3 TCP Veno TCP Veno [9] focuses on solving the random noncongestion loss problem. It is very similar to TCP Vegas [4] which is an improvement on TCP Reno by introducing a proactive response to the network behavior. TCP Veno uses two main parameters: the expected rate, defined as the ratio of the congestion window size over the best RTT, and the actual rate, defined as the ratio of the congestion window size over the last measured RTT	[redacted] [redacted] của [redacted] phát hiện ngẫu nhiên sớm [redacted] theo kiểu xác suất [redacted] mọi [redacted] [redacted] tính chất [redacted] một nhân tố dấu [redacted]	

to calculate the backlog in the router queue that is used as a sign of congestion.	[redacted]	
4.4 JTCP	[redacted]	
JTCP [14] assumes that the network congestion can be inferred from the difference in the interarrival times of successive packet ACKs. This is the same paradigm used in TCP Veno. JTCP tackles only random wireless loss.	[redacted]	
One basic concept used in JTCP is the interarrival jitter. It is defined as the time difference of two packets on the sender side and the time difference of the same two packets on the receiver side. If the interarrival jitter is greater than zero, that means that the second packet traveled through the network longer than the first one. Thus, some time was lost in the queues of the network routers.	[redacted]	
A second important concept is the jitter ratio (Jr). Note that if the arrival rate of packets at the router is greater than its service rate, a queue is going to form at that router. Jitter ratio can be defined as the variance of the queue length and provides for the ability to detect whether the packets	[redacted]	

are being queued at the router or not. JTCP uses the jitter ratio in combination with the traditional loss events to determine the type of loss in the network.	[redacted]	
5 Simulation Experiments	[redacted]	
There are two main methods of testing the modifications to any networking system: simulations and live testing. Each has its advantages and faults [2]. Simulation allows for the widest range of testing scenarios no possible with live tests. Therefore simulations were implemented using the ns-2 simulator version 2.28 [1] on a Linux Fedora Core 4 platform [13].	[redacted] kiểm tra rất nhiều trường hợp mà các phép kiểm tra trực tiếp không làm được [redacted] do [redacted]	
5.1 Design of the Simulations	[redacted]	
Simulation tests were divided according to three changing network parameters: type of wireless loss, total packet delay, and network congestion state.	[redacted] sự trì hoãn gói tin toàn phần [redacted] mạng [redacted]	
5.1.1 Type of Wireless Loss	[redacted]	
The behavior of the protocol is simulated under random packet loss and disconnection loss. Simulation tests subjects the protocol to 0%, 0.1%, 0.5%, 1%, 5%, and 10% random	Tính chất của giao thức [redacted] áp dụng trên giao thức [redacted]	

packet loss. Disconnection loss subjects the protocol to a period of 100% packet drop for the length of 0, 0.5, 1, 2, 5, and 10 seconds. In random loss testing, the packets are dropped at the same rate on all wireless channels, while in the disconnection testing the loss occurs only on one wireless channel. Of course, the loss occurs in both directions, meaning that the packets and their ACKs face the same danger of being dropped. 5.1.2 Total Packet Delay To test the behavior of presented protocols in real life networks, local area networks (LANs) and wide area networks (WANs) environments were simulated. • LAN Networks - The simulation model of the LAN network as shown in Figure 1 includes four hosts of which two serve as routing devices, i.e. base stations. The topology has two wireless LAN links where wireless hosts are in close proximity to the base station causing very short propagation delay. The two base stations are connected via wired connection of high bandwidth and low	[redacted] được kiểm tra trên giao thức đang trong giai đoạn thả gói tin 100% [redacted] được thả [redacted] đó, trong các phép [redacted] hướng [redacted] thả [redacted] Sự trì hoãn gói tin toàn phần [redacted] đặc tính [redacted] diện [redacted] làm sự trì hoãn truyền rất nhỏ [redacted]	

delay. Figure 1. LAN Network. • WAN Networks - The WAN model is a little more complex. A heterogeneous environment is targeted primarily characterized by long overall delays. Hence, the topology is a combination of two wireless WAN links (e.g. satellite link), one wired WAN link, and one long-delay LAN link as shown in 2. Figure 2. WAN Network. The LAN topology was simulated for 300 seconds in both random and disconnection loss scenarios. Similar logic has been used for the simulation of the WAN topology, but tests were 600 seconds long. 5.1.3 Network Congestion State One point of interest is to observe the ability of the protocol to distinguish between congestion loss and wireless noncongestion loss. All tests were run under the following three conditions: • No Congestion — The TCP protocol flow exists alone in the network. The flow never fills the	[redacted] thời gian trì hoãn toàn phần dài [redacted] [redacted] [redacted] việc [redacted] [redacted] tồn tại [redacted]	

queue of the router; hence, no packet is lost due to the network congestion. • UDP-based congestion — Running a single congestion flow in the network is an unrealistic scenario. To test the behavior of the protocol in congested environments, a UDP flow was introduced at a rate that takes 80% of the bandwidth of the bottlenecked link. To add more variability to the simulation, a level of randomization is added to the sending rate of the UDP flow. • TCP-based congestion — In this scenario, all 7 protocols were run in simultaneous flows. All of the flows start at the same time and fight over the available bandwidth. In such conditions, the congestion is imminent. This condition allowed to observe the aggressiveness/friendliness character of the protocols. 5.2 Fixed Parameters Some of the network parameters were preset and left constant in all tests: • Packet Size — The packet size was set to 1024 bytes, i.e. 984 bytes of data and 40 bytes of a packet	[redacted] [redacted] [redacted] [redacted]	

header. For variability, the size of the UDP packets used in the simulations was set to 512 Bytes. • Queue Size — All simulated routers were assumed to have queue sizes of 50 packet loads, i.e. 50KB. Similarly, all routers assumed the drop tail queuing mechanism, which means that all packets coming to the full router queue are dropped. The only exception is the simulation of TCP-Jersey which by design requires a slightly modified version of the drop tail queuing mechanism that implements a simplified version of the congestion warning mechanism [15]. • Delayed ACKs — Most of the currently implemented TCP protocols use the delayed acknowledgement method. 5.3 Benchmark Metrics To gain an understanding of the behavior of the tested protocols, the following benchmark metrics were used: • Throughput — This is defined as the ratio of total data transferred to the time it took to transfer it. • Average Congestion Window — This metric is determined by the sum of all	[redacted] được [redacted] [redacted] tất cả các hàng đợi/drop tail [redacted] thả [redacted] [redacted] đang được thụ thi trong hiện tại [redacted] trì hoãn [redacted] [redacted]	

congestion window sizes divided by the number of transmissions. This parameter provides an idea of the protocol’s resilience to loss and its ability to recover. • Time to Complete — This is the time required to transfer a continuous block of memory (file). Note that the simulation for each network condition has been performed only once. However, simulations were performed for a significant periods of time generating several hundred thousand packets. With such a sample size the true state of the simulated network can be assumed. Multiple simulations required not only time but also an extraordinary space overhead, especially for high bandwidth/low latency conditions. 6 Results and Discussion The simulation tests returned an abundance of information. For a full analysis of the results, please refer to [13]. A subset of these results is transcribed in this section. 6.1 Behavior Characteristics Figure 3. Throughput under	[redacted] [redacted] bộ nhớ [redacted] [redacted] Thực hiện nhiều mô phỏng [redacted] [redacted] [redacted] [redacted] [redacted]	

random loss in long UDP congested LAN. There was little difference in the general behavior of the protocols between identical tests in LANs and WANs. However, while the tendencies are the same, the protocols tend to show results closer to each other. The superiority of TCP Westwood and JTCP becomes more apparent as the number of randomly lost packets gets larger, i.e. the congestion windows becomes bigger as shown in Figure 4. The same random loss rate will drop fewer packets in WANs than in LANs because fewer packets are released into the network. Therefore, while it is much more costly to lose a packet in WAN, apparently all protocols exhibit a comparable efficiency in recovering from the loss of a relatively small number of packets. This logic is strengthened by the results of TCP Westwood that does not perform as impressively in WAN environments as it does in LANs. Figure 4. Throughput under random loss in UDP congested WAN. G	[redacted] [redacted] [redacted]	

ested and UDP congested environments. The throughputs and average congestion windows are not significantly altered by longer disconnection periods. In addition, the performance of the protocols under UDP congestion is very similar (Figure 5). This can be explained by realizing the disconnection loss does not tax performance because it allows the network to recover from congestion. This is probably due to the differences in the retransmission timeout (RTO) algorithms of the protocols as some protocols test the network more frequently than others. Therefore, in these conditions there is a much greater difference between the performances of the protocols. TCP NewReno is the top performer, but in the disconnection loss it is accompanied by TCP SACK. JTCP once again performed poorly in the throughput and average congestion window

[redacted]

benchmarking.

Figure 5. Throughput under disconnection loss in UDP congested LAN.

The WAN topology shows a more stable and uniform performance. This is manifested in much smaller absolute differences between the solutions' benchmark parameters as shown Figure 6.

6.2 Best Performing Protocols

The results of the simulations do not show a clear winner amongst the protocols analyzed. With each tested environment having its set of unique attributes, it was actually expected that different protocols might be better suited to deal with different network conditions. However, a clear gain from these results is a good idea of what protocols are dominant. Three of them stood out the most: JTCP, TCP Westwood, and TCP SACK. All three of these solutions are rather unique in their design, having relatively little in common with each other.

JTCP proved itself highly efficient under both random and disconnection loss in both LANs and WANs. This

[redacted]

[redacted]

[redacted]

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would indicate that the congestion anticipatory feature of JTCP performed quite well. However, it cannot be generalized that all of the congestion anticipation mechanisms perform favorably under wireless packet loss because of the somewhat disappointing performance of TCP Veno. With both of these solutions operating fairly similarly, it is altogether possible that the edge that JTCP gained over TCP Veno was simply because of a better tuned settings for the congestion window modification. Still, JTCP has a shortcomings — it performed rather poorly while competing for bandwidth with the other TCP flows.

[redacted]

Figure 6. Throughput under disconnection loss in UDP congested WAN.

[redacted]

TCP Westwood was also one of the best performing protocols. It managed to surface as a solid candidate to replace Reno-based protocols as a networking standard. What is truly amazing about the performance of TCP Westwood is that it is not, inherently, a protocol geared to fix the problems

of wireless non-congestion loss but simply to increase the efficacy of the communication under traditional, mainly wired, infrastructures. This is particularly interesting since TCP-Jersey also uses a dynamic algorithm for the estimation of available bandwidth but implements a congestion warning scheme to differentiate between congestion and non-congestion packet loss. However, it does not come even close to performing as impresivelly as TCP Westwood. Nevertheless, TCP Westwood is not without flaws. It performs convincingly under random wireless loss but fails to impress under disconnection loss. It is also far better in LAN topologies than in WANs. That kind of behavior can be easily explained by its aggressive and quick response to intermittent packet drop but it has difficulty dealing with the long term, systematic loss of packets. Furthermore, the downside of TCP Westwood's aggressive character is its inherent unfriendliness toward other existing TCP flows in the long run.	[redacted] [redacted]	

TCP SACK was the last of the protocols that proved efficient in different types of wireless networks. As TCP SACK is a widely accepted and implemented protocol in most of the current operating systems, we can say that even if none of the proposed protocols that directly tackle the problem of wireless packet loss become widely accepted, the performance of TCP SACK might be a better alternative than we hoped for. Since it is not a true wireless solution, TCP SACK gains in efficacy by simply responding to the loss events by resending the correct packets and inflating the congestion window to avoid a drop in the transmission rate. That is why most of TCP SACK's superb performance comes in the environments facing disconnection loss. Its performance under random loss is not as impressive.

7 Conclusion and Future Work

The main purpose of this study was to compare and analyze the performance of the four recently proposed TCP protocols optimized for wireless networks, TCP West- wood/Westwood+,

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TCP-Jersey, TCP Veno, and JTCP. The analysis was performed assuming random and disconnection packet loss in wireless networks. There was no one solution that proved superior in all conditions, but the same small group of protocols appeared on top of the leader board in all simulations. As summarized in Table 1, TCP Westwood and JTCP outperformed their competition under random packet loss in both burst and long flow testing with the realization that the performance of TCP Westwood was much better in LAN than in WAN topologies. JTCP showed a remarkable performance in all environments under both random and disconnection packet loss but showed a significant drop in throughput when competing with other TCP flows. Under disconnection loss we saw two protocols dominating: TCP SACK and, once again, JTCP. TCP Westwood posted average results in disconnection loss simulations. Table 1. Best Performing Protocols. Random Loss	[redacted] [redacted] [redacted]	

Disconnection Loss
LAN No Congestion
TCP Westwood/JTCP
JTCP/TCP SACK
UDP Congestion
TCP Westwood/JTCP
TCP SACK
TCP Congestion
TCP Westwood
TCP NewReno/TCP SACK
WAN No Congestion
JTCP TCP SACK
UDP Congestion
JTCP JTCP
TCP Congestion
TCP NewReno
TCP NewReno

Some basic networking parameters were set to their most often encountered values and held constant to allow for close comparisons of the tested protocols. The different treatment of these parameters (packet size, switch queue size, queuing mechanism, etc.) could possibly provide us with somewhat different results.

[redacted]

This work does not provide enough information to give us a complete insight about the best protocols for wireless networks. Future research needs to consider the behavior of these protocols in ad-hoc networks, especially when the hosts exhibit rapid

[redacted]

movement tendencies. Also the effect of simultaneous random and disconnection packet loss needs to be assessed. Finally, the authors wish to acknowledge the useful comments provided by the anonymous reviewers that lead to a significant improvement of this paper.	[redacted]	