

# A Cross-Layer Approach for an Efficient Delivery of TCP/RTP-Based Multimedia Applications in Heterogeneous Wireless Networks

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**Abstract**—Recent trends in the telecommunication industry have been moving toward the development of ubiquitous information systems, where the provision of a plethora of advanced multimedia services should be possible, regardless of time and space limitations. An efficient and seamless delivery of multimedia services over various types of wireless networks is still a challenging task. The underlying difficulty consists of the disparity in the bandwidth availability over each network type. Indeed, the fundamental challenge upon a handoff phenomenon in a heterogeneous wireless network consists of an efficient probing of the bandwidth availability of the new network, followed by a prompt adjustment of the data delivery rate. This paper presents a cross-layer approach that involves five layers, namely, physical, data link, application, network, and transport layers. The three former layers are used to anticipate the handoff occurrence and to locate the new point of attachment to the network. Based on their feedback, the transport layer is used then to probe the resources of the new network using low-priority dummy packets. Being the most widely used protocol for multimedia delivery, this paper addresses multimedia applications based on the Transmission Control Protocol (TCP) and the Real-time Transport Protocol (RTP). The design of the whole cross-layer architecture is discussed, and enhancements to the two protocols are proposed. The performance of the enhanced TCP and the RTP are evaluated and compared against existing schemes through extensive simulations. The obtained results are encouraging and promising for the delivery of multimedia services in heterogeneous wireless networks.

**Index Terms**—Cross layer, heterogeneous wireless networks, next-generation wireless Internet, Real-time Transport Protocol (RTP), Transmission Control Protocol (TCP).

## I. INTRODUCTION

ADVANCED multimedia services are gaining momentum within the communities of both industrial and academic researchers. Indeed, along with the ongoing advances in wire-

less technologies and the exponential growth of the mobile users' community, the provision of multimedia applications in wireless networks is likely to open a promising and strong market for service providers and operators.

Among the protocols used for the delivery of multimedia services, the Transmission Control Protocol (TCP) and the Real-time Transport Protocol (RTP), accompanied with the Real-time Transport Control Protocol (RTCP) [1], are the most notable ones. Being originally designed for wired networks, both TCP and RTP do not perform well in heterogeneous wireless networks for a number of reasons related to the protocols' syntax and semantics. Their current implementations consequently put many stringent constraints on effective multimedia streaming in wireless systems.

In wireless networks, due to users' mobility, mobile nodes freely, and sometimes frequently, change their points of attachment to the network, which is an operation henceforth referred to as handoff. Upon a handoff occurrence, the amount of bandwidth available at the new point of attachment may be different from that at the old one. This bandwidth disparity can be due to differences in traffic load in both wireless cells.

In general, when a mobile node performs a handoff, two scenarios can be envisioned. If the mobile node moves from a higher bandwidth network [e.g., Wireless Local Area Network (WLAN)] to a lower bandwidth network [e.g., general packet radio service (GPRS)] and continues transmitting data without any adjustment to its sending rate, the new network will be congested, and a potential number of packets will be dropped. The connection throughput will eventually be degraded. On the other hand, if the mobile node enters a higher bandwidth network, no adjustment to the sending rate of the mobile node will lead to a waste of the network bandwidth and ultimately to lower network utilization. Such a performance will obviously result in a poor Quality of Service (QoS) and will ultimately affect the credibility of the whole system.

Ideally, mobile users should be able to anticipate imminent handoff events, should be aware of the next point of attachment to the wireless network, and should get their data download rates promptly adjusted (or should themselves adjust their data sending rates) to meet the available resources of the new access point (AP). As an attempt to realize such an ideal network, this paper proposes a cross-layer architecture that involves five layers, namely, physical, data link, application, network, and

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transport layers. The physical and data link layers monitor signal strengths and detect any impending handoff. They then advertise the event to the application layer. In turn, the application layer refers to personal information on the mobile user, history on its mobility patterns, and, if possible, information on the topology of the wireless network to locate the next AP.

Knowing the next point of attachment, two connections are simultaneously set between the mobile node and its correspondent sender: one through the old AP and another through the new one. Assuming that the coverage areas of both APs overlap with each other, the sender continues transmitting actual data through the old connection. Meanwhile, it sends a number of low-priority dummy segments through the new connection. These dummy packets are used to probe the bandwidth availability of the new network, which is similar in spirit to the idea presented in [2].

The application of the concept to both TCP and RTP is considered. Related issues are discussed, and possible solutions are presented. Extensive simulations are conducted to evaluate the performance of the proposed modifications to TCP and RTP. The results demonstrate that the proposed concept is promising for the guarantee of QoS in wireless networks as it assures fast handoff management, increases the system throughput, and maintains lower packet drop rates.

The remainder of this paper is structured as follows. Section II highlights the relevance of this paper to the state of the art in the context of cross-layer design for wireless mobile networks. The key design philosophy and distinct features that were incorporated in the proposed cross-layer architecture are described in Section III. Section IV portrays the simulation environment and reports the simulation results. Following this, this paper concludes in Section V with a summary recapitulating the main advantages and achievements of the proposed architecture.

## II. RELATED WORK

The traditional Open System Interconnection (OSI) layered architecture, as originally specified, did not specifically provide any interaction among its layers. A cross-layer design aims at enabling such an interaction for the sake of better performance and prompt adaptation of the stack functionality in the presence of changing network conditions. Based on the involved layer, the emergence of several cross-layer interactions has been highlighted in the recent literature [3], [4].

The proposed cross-layer architectures and frameworks can be categorized based on the type of communication used to exchange information among layers. In [5], an architecture based on the use of the Internet Control Message Protocol (ICMP) messages is proposed. The architecture involves the physical/MAC layers, the network layer, and the application layer. In [6], an Interlayer Signaling Pipe (ISP) is used to propagate cross-layer information through packet headers. A drawback of this technique consists of the fact that lower layers are required to read the headers of higher layers, which is an operation that ultimately slows down the execution of the lower layers. As a solution to this issue, the Cross-Layer Signaling Shortcut (CLASS) architecture allows direct communication

between the layers [7]. Other cross-layer architectures consider the addition of new components to the protocol stack. Mobile-Man is a notable example [8]. In [9], a cross-layer manager is designed to handle events and state variables sent by the protocol layers. The state variables are used to appropriately coordinate among the link adaptation, security, QoS, and user mobility.

While the aforementioned systems are relatively generic in their design and consequently add significant complexity to the original design of the protocol stack, a number of other cross-layer approaches simply use information from different layers to optimize the protocol behavior in some circumstances [10], [11]. The RTP protocol is itself an example. Indeed, it integrates functions of both the session and presentation (and in some cases application) layers into a single protocol. By maintaining a large context related to a given multimedia session, it is possible to handle several aspects of real-time communication such as synchronization and adaptive application framing. Another example that falls in this category is the Freeze-TCP [12]. It involves the physical and data link layers as it uses their feedback to detect handoffs or to predict a temporary disconnection. If a handoff occurs, a Freeze-TCP mobile host advertises a zero window size to force the sender into frozen mode. This operation aims to avoid drops of in-flight packets. The sender restarts transmitting data only when the mobile host enters a new point of attachment. For a detailed discussion on other cross-layer mechanisms related to TCP and RTP, the interested reader is referred to the related work sections of [13] and [14], respectively.

## III. OVERVIEW OF THE PROPOSED CROSS-LAYER ARCHITECTURE

This section gives a detailed description of the proposed cross-layer architecture. It first outlines the core ideas behind the architecture and its requirements. It next presents the major components of the architecture. In addition, it portrays the proposed enhancements to the working of TCP and RTP to guarantee an efficient and seamless delivery of multimedia services over wireless networks.

### A. Requirements

First, it should be emphasized that this paper targets wireless networks where cells overlap each other. The considered network is assumed to be end-to-end QoS enabled. In fact, the proposed scheme requires that all network elements along the connection path support some priority disciplines. Currently, most networks are best effort, and most routers in the Internet do not apply any priority policy. However, in the near future, through the use of the Differentiated Service Model (DiffServ) [15], routers will be able to support multiple service classes. Having said that, it should be stressed that the proposed cross-layer design does not specifically require a DiffServ architecture. It simply requires a priority-queuing discipline with two priority levels.

To enable mobile hosts to simultaneously access two or more different APs, mobile nodes are equipped with multiple wireless interfaces. While having multiple wireless interfaces

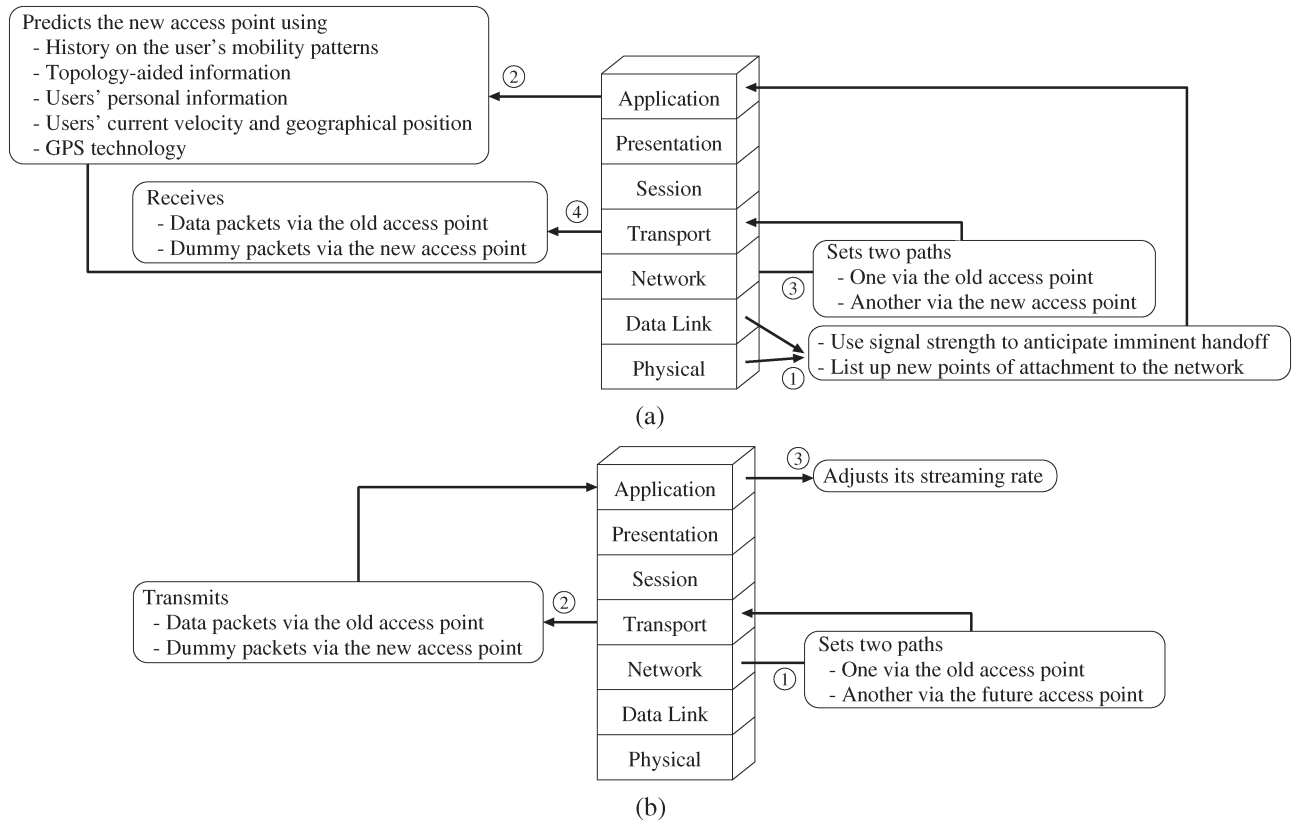


Fig. 1. Envisioned cross-layer design. (a) At the receiver side. (b) At the sender side.

on the same mobile device is impractical, the ongoing advances in the wireless technology have demonstrated that a single physical WLAN interface can be used to simultaneously access multiple WLANs [16]. To allow a mobile node to simultaneously register multiple Care-of-Addresses (CoAs), the Mobile IP (MIP) simultaneous binding option [17], [18] is used. On the other hand, to keep senders always informed of these CoA registrations directly from the mobile nodes, the route optimization option [19] is used. It should be noted that the new CoA of the mobile node in the new cell should be different from the CoA used in the old cell. Finally, at the sender side, applications should be able to adjust their streaming rates by appropriately changing the quality of multimedia contents. As for the type of communication to be used in exchanging information among layers, a wide library of communication types exists, as discussed in the previous section. The proposed cross-layer design can consider implementation of the most adequate one taking into account the required computational load and the communication delay that may result from interactions among the layers.

### B. Cross-Layer Design

A cross-layer optimization can be implemented either at end devices or at intermediate nodes in the network, such as APs or routers. Given the relative easiness and feasibility of the former, this paper focuses on implementing changes on the mobile hosts. Fig. 1 depicts the major procedures of the proposed architecture. At the receiver side (mobile host), the physical

layer of a mobile host instantly measures the radio strength or link quality. When the mobile node moves into the overlapping area of two or more wireless cells, and different signals are consequently detected by the physical and data link layers, a warning message notifying an imminent handoff event, along with a list of new possible APs, is sent to the application layer. In case of multiple APs, the application layer refers to a set of tools to sort out the AP to which the mobile node is most likely going to be connected. Indeed, the application layer may use history on the user's mobility pattern to predict the new AP. Referring to a spatial conceptual map, along with the user's personal information, its current position, and its velocity heading, the application layer can make an accurate prediction of the most probable future AP [20]. Prior knowledge on the topology of the wireless network [21] and the type of application [22] can further increase the accuracy of the prediction. Once the next AP is determined, the sender is informed of the new base station via a new CoA binding update message from the MIP protocol. The network layer then sets two paths: one via the old AP and another via the new point of attachment. The transport layer keeps receiving data packets via the old AP and simultaneously starts receiving dummy packets via the new AP. The dummy packets are used to probe the bandwidth availability of the new network, as will be explained later.

The cross-layer design at the sender side is relatively simpler than that at the receiver side. First, upon receiving information on the new AP, the network layers of both sender and receiver terminals set a new path via the new AP and, at the same time, maintain the old one. The transport layer of the

sender terminal keeps transmitting data packets via the old path and uses dummy packets to probe the bandwidth of the new network. Once the new bandwidth is estimated, the application layer of the sender terminal should accordingly adjust its data streaming rate.

### C. Enhancements to TCP

To cope with issues related to handoff management in heterogeneous wireless networks, a large body of bandwidth-probing techniques has been proposed to make an estimate of the available bandwidth in the new network [23]. Most of these pioneering techniques require accurate measurements of the propagation delay. Under heavy traffic load, an accurate estimate of the propagation delay is usually not possible to obtain: a fact that ultimately leads to erroneous estimates in the available bandwidth. The key concept behind our proposed enhancement to TCP consists of the use of dummy packets for an efficient probing of the bandwidth availability in the new network [13]. Indeed, when the next AP is decided by the proposed cross-layer architecture, as previously explained, and before reaching the middle point of the overlapping area (where the handoff usually takes place), the mobile node keeps on receiving data from the sender using the old connection through the old AP. Meanwhile, the sender sends “*rwnd*” dummy segments to the mobile node through the new AP, where *rwnd* is the receiver window size that limits the maximum value of congestion window. The value of *rwnd* indicates the rate at which the sender transmits dummy segments to the mobile node. The algorithm of the proposed scheme is based on the concept of using these dummy segments to probe the availability of network resources without carrying any new information to the sender. This concept was first proposed in [24] and has been used since then in several works in the recent literature. Notable examples are TCP-Peach [25], the InterPlanetary Transport Protocol (TP-Planet) [26], and the Analytical Rate Control (ARC) [27].

Dummy segments are generated by the sender as a copy of the last transmitted data packet. They are treated as low-priority segments. Accordingly, they do not affect the delivery of the actual data traffic. Indeed, when a router on the connection path is congested, IP packets carrying dummy segments are first discarded. The overhead of these dummy segments in terms of bandwidth consumption should, therefore, not be an issue.

To distinguish dummy segments from actual data packets, dummy segments are marked using one or more of the six unused bits in their TCP headers. A simple modification of the TCP implementation is accordingly required at the end terminals. Upon reception of a dummy segment, the mobile node can thus recognize it. In response to each dummy segment, the mobile node transmits a dummy acknowledgment (ACK) to the sender. Dummy ACK packets indicate the availability of network resources in the new cell. In response to each dummy ACK, the sender transmits, in turn, an actual data packet to the mobile node. ACKs for dummy segments are used to provide an efficient probing of the bandwidth availability in the new network. As a result, senders can adjust their sending rates to the most appropriate value within one round trip time (RTT). They either increase their transmission rates to make full utilization of the new network resources or decrease their

transmission rates to avoid overloading the new network with bursty traffic.

### D. Enhancements to RTP

While TCP dominates most of today’s Internet traffic, RTP represents the core streaming protocol for real-time multimedia services. It does not add any delays to the transmitted data as packet retransmissions are not considered. However, it may congest the network as it does not employ any congestion control. To cope with such an issue, RTP receivers notify senders with statistics on their perceived QoS, such as cumulative packet losses, RTP timestamp, number of packets received, and jitter. This information is periodically reported via signaling messages called Receiver Reports (RRs). Based on these RR messages, the RTP protocol assesses the network condition and accordingly controls its streaming rate. This forms the basic framework of the RTCP protocol.

One important issue that is missing from the design of RTCP pertains to the transmission frequency of RR messages. Indeed, the minimum time interval for transmitting two consecutive RR messages is recommended to be set to 5 s [1]. This aims to meet the 5% fraction of the session bandwidth reserved for RTCP packets. In heterogeneous wireless networks, this policy is inefficient and may largely affect the entire system performance. As a matter of fact, in the case where a mobile receiver performs handoff to a lower-bandwidth network without immediate transmission of an RR packet, by the time the correspondent sender gets notified of the new network conditions and starts accordingly adjusting its streaming rate, the new network may have already been overly congested, and a significant number of packets may have been dropped. In the case of handoff to a higher bandwidth network, no immediate adjustment of the RTP streaming rate may lead to a waste of the new network resources.

As a remedy to this issue, an RTP mobile receiver uses the aforementioned cross-layer design to anticipate an imminent handoff event. It then explicitly notifies its correspondent sender with the event via newly defined RTCP packets. These packets are referred to as RTCP Handoff Notification (HN) packets throughout the remainder of this paper. While they have the same header as ordinary RTCP packets, RTCP HN packets can be distinguished by having their packet type field set to an unused value. It should be reminded that RTCP RR packets are transmitted on a periodic basis, whereas RTCP HN packets are sent only upon detecting degradation in the link quality (in other words, when a handoff event is about to occur).

Upon receiving an RTCP HN packet, the RTP sender probes the available bandwidth in the new network using dummy RTP packets, which is similar to the aforementioned enhancements proposed for TCP. These dummy RTP packets are sent through the AP of the new network at the maximum streaming rate of the multimedia data for a predefined period of time (i.e., less than 1 s). After receiving dummy packets for the predefined period of time, the RTP receiver sends a reception quality feedback to the sender in an RTCP signaling packet. This type of packet is referred to as RTCP Handoff Report (HR) packet throughout this paper. The format of RTCP HR packets

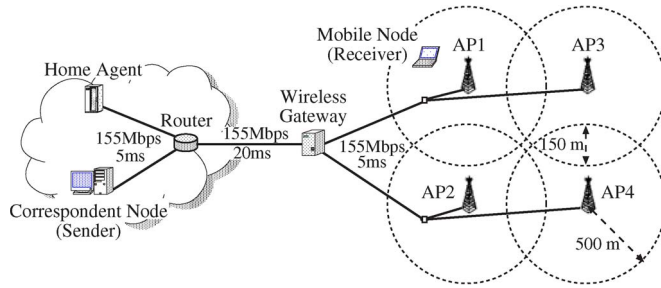


Fig. 2. Abstract configuration of the simulation network topology.

conforms to that of RTCP RR. It includes information on the reception quality measured during the reception of dummy packets. Once the optimal streaming rate of the new wireless network is known to the sender, the receiver starts receiving actual data packets via the new AP and quits its old connection with the sender by issuing an RTCP BYE packet.

#### IV. PERFORMANCE EVALUATION

Having described the details of the proposed cross-layer architecture, we now direct our focus to evaluating its performance. The performance evaluation relies on computer simulation using the network simulator [28]. We first describe the simulation setup, justifying the choices made along the way and next discuss the simulation results.

##### A. Simulation Setup

In the conducted simulations, particular attention is paid to the design of an accurate and realistic environment. Fig. 2 depicts the abstract configuration of the considered network. The wireless part of the network consists of a number of adjacent wireless cells. The coverage radius of each wireless cell is set to 500 m. The distance between two adjacent APs is fixed to 850 m. This makes the longest distance across the overlapping area between two adjacent cells equal to 150 m. In the simulations, the actual distance across the overlapping area is varied from 1 to 150 m. These parameters are chosen with no specific purpose in mind and do not change any of the fundamental observations about the simulation results.

The wireless domain is connected to the wired network through a single wireless gateway. The choice of a single wireless gateway serving all the APs represents a general and simple case. Indeed, considering a topology where APs are served by two different wireless gateways will simply increase the connection RTT and shall have no influence on the overall performance evaluation. To avoid packet drops due to bottlenecks at the wired network, all wired links are given similar capacities equal to 155 Mb/s (e.g., OC3). They have predetermined propagation delays, as indicated in Fig. 2. As for the wireless links, a number of test scenarios were created by setting their capacities to different values. Their delays are minimal and are set to 0.01 ms. All links are presumed to be error free throughout this paper. This assumption is made to avoid any possible confusion between throughput degradation due to packet drops and that due to wireless channel errors.

To best understand the behavior of the proposed enhancements to TCP and RTP, we consider a single handoff between two adjacent APs in the considered topology. Having prior knowledge on the position (coordinates) of each AP, a user refers to its velocity heading and its position to predict the next AP to which it will be connecting. Delay incurred by the computational load of this operation is included in the entire handoff delay. In the simulations, a mobile node receives a new network address from the new AP and issues a CoA binding update message as soon as it enters the cell-overlapping area of two adjacent cells. It accordingly sets up two paths for communication through the old and new APs, respectively. As for the actual handoff, it is performed when the radio strength of mobile nodes or the wireless link quality goes down below a predefined threshold. In the simulations, handoffs are performed when a mobile node reaches the middle line of the overlapping area, which represents the most common case.

In the proposed cross-layer architecture, all the network elements along the connection path need to support some priority disciplines. This operation is enabled using the Weighted Random Early Detection (WRED) scheme [29]. Unless otherwise specified, the queue length of all network elements is set to 50 packets. The size of a data packet is set to 1000 B in TCP and 500 B in RTP. The maximum streaming rate in RTP-related simulations is fixed at 10 Mb/s. All results are an average of multiple simulation runs.

##### B. Analysis: Effects of Handoff on Rate Control

1) *TCP-Based Multimedia Services*: For the sake of a better understanding of the research presented in this paper, we analyze the effects of handoff on the congestion window in the case of TCP-based multimedia services. Different TCP variants are used as comparison terms. These variants include the well-known TCP NewReno [30], Freeze-TCP [12], and TCP Westwood-NR, which is the NewReno-based version of TCP Westwood [31]. While our proposed enhancements can be implemented on any TCP variant, we consider enhancements to TCP NewReno. The reason behind the choice of TCP NewReno among other TCP implementations underlies the fact that TCP NewReno achieves faster recovery from multiple losses within the same window. It also has the potential of improving TCP's performance in the case of bursty losses. For an insightful comparison among the TCP variants, focus is on the performance of schemes during the handoff period. The definition of the handoff period comes later.

In this analysis, we consider two scenarios where handoff occurs. In the first scenario, called H-L, the mobile node moves from a high-capacity (6 Mb/s) cell to a low-capacity (1 Mb/s) cell, whereas in the second scenario, called L-H, the mobile node moves from a low-capacity (6 Mb/s) cell to a high-capacity (11 Mb/s) cell. We focus on the behavior of the congestion window  $cwnd$  in the four TCP versions examined.

Throughout this paper, the *handoff period* is defined as the time period during which a mobile node travels over the cell-overlapped area. If we consider two time instants  $t_0$  and  $t_2$  with  $t_0 < t_2$ , we suppose that handoff starts at time  $t_0$  and ends



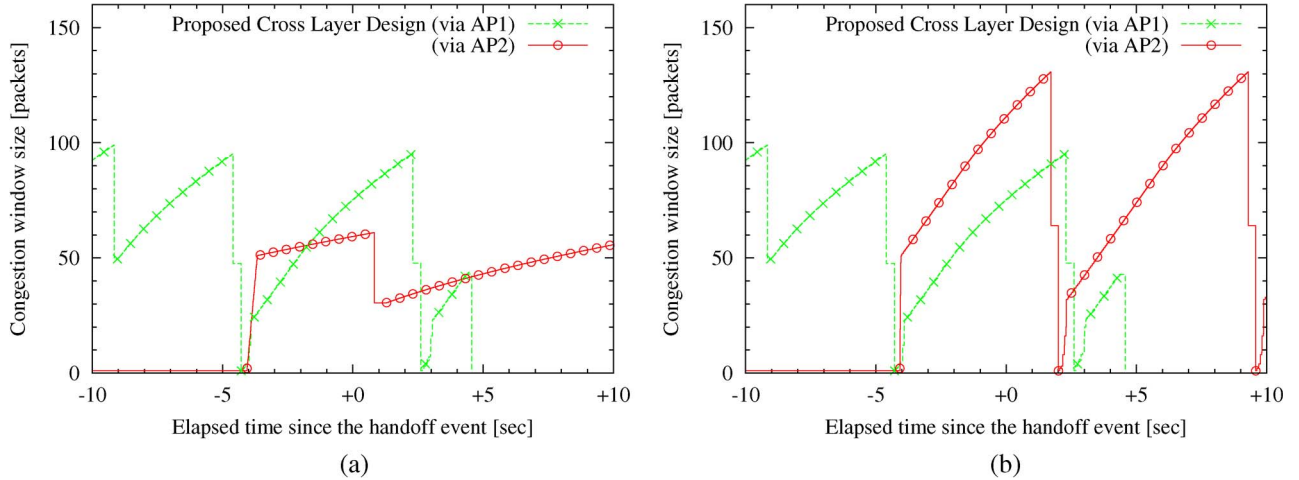


Fig. 3. Variation of the congestion window size during handoff in the case of the proposed cross-layer design. (a) H-L scenario (6 Mb/s  $\rightarrow$  1 Mb/s). (b) L-H scenario (6 Mb/s  $\rightarrow$  11 Mb/s).

at time  $t_2$ . Handoff occurs at time  $t^*$  when the middle of the overlapping area is reached, i.e.,  $(t^* = (t_0 + t_2)/2)$ .

*Cross-layer approach:* First of all, we explain the proposed cross-layer behavior during the handoff period in detail.

- 1)  $t_0 \leq t < t_1$ : At time  $t_0$ , the mobile node enters the cell-overlapping area. During the time interval  $[t_0, t_1]$ , with  $(t_1 < t^*)$ , the sender sends *rwnd* dummy segments to the receiver through the new AP, where *rwnd* is the maximum value for the congestion window that is specified by the receiver. The receiver transmits a dummy ACK in response to each dummy packet received.
- 2)  $t_0 + \text{RTT} \leq t < t_1 + \text{RTT}$ : After one RTT elapsed from the first dummy segment transmitted, the ACKs related to the dummy segments reach the sender. In response to each dummy ACK, the sender transmits an actual data packet to the receiver. Note that we assume  $(t_1 + \text{RTT} < t^*)$ , which is an assumption that will be later confirmed by simulation results.
- 3)  $t = t_1 + \text{RTT}$ : At this time, due to the previously shown mechanisms, the congestion window of the connection established via the new AP results in  $(cwnd(t) := n_{\text{ACK}})$ , where  $n_{\text{ACK}}$  is the number of dummy ACKs received so far. This means that in both scenarios (H-L and L-H), the sender adjusts its sending rate to the most appropriate value within one RTT.
- 4)  $t > t_1 + \text{RTT}$ : The classical TCP NewReno algorithms are used.

In Fig. 3, we show the variations of the congestion window size of the proposed cross-layer approach during the time period from 10 s before to 10 s after the handoff phase. Let AP1 and AP2 be the old and new APs, respectively. We reproduce both H-L and L-H scenarios in Fig. 3(a) and (b), respectively. These plots have been obtained assuming  $t_0 = -4.5$  s and  $t_2 = 4.5$  s. Consequently, the handoff happens at  $t^* = 0$  s. It is observed that, in both scenarios, starting from  $(t > t_1 + \text{RTT})$ , the *cwnd*, whose value is greater than the slow start threshold, linearly increases according to the bandwidth estimated in the new cell. Moreover, we observe that when a

loss is detected, the original recovery algorithms of NewReno are used.

*Freeze-TCP:* Here, we describe the behavior of Freeze-TCP during the handoff.

- 1)  $t = t^*$ : At this time instant, the receiver (knowing that a handoff is occurring) advertizes a zero window size to the sender. This operation is performed to compel the sender into a frozen mode to prevent the congestion window from dropping to one.
- 2)  $t = t_3$ : Let  $t_3 (t^* < t_3 < t_2)$  denote the time instant when Freeze-TCP starts retransmitting data to the new AP. The value of congestion window will not be changed from the last value; thus,  $(cwnd(t) := cwnd(t^*))$ .
- 3)  $t > t_3$ : If the bandwidth of the new network  $b_{\text{new}}$  is greater than that of the old one  $b_{\text{old}}$ , such as in the L-H scenario, the congestion window increases by  $(1/cwnd)$  for each ACK received, i.e.,  $(cwnd := cwnd + (1/cwnd))$  until a congestion occurs. On the other hand, if  $b_{\text{new}} < b_{\text{old}}$  (H-L scenario), the new network suddenly gets overloaded with a large number of data packets. This congests, in turn, the transmission queue at the bottleneck link's router and eventually results in the discard of a large number of packets. As a result, TCP almost immediately decreases its *cwnd* to 1.

The underestimation or overestimation of the bandwidth availability in the new network in both H-L and L-H scenarios have been examined in Fig. 4(a) and (b), respectively. Here, we show the congestion window size in packets during the handoff period. Comparing the results against those obtained in the case of the cross-layer approach, we notice that Freeze-TCP exhibits a slow increase of *cwnd* in the L-H scenario after a loss, as well as a drastic reduction of *cwnd* in the H-L scenario as the new network gets overloaded, and a number of packet drops occur.

*TCP NewReno:* Here, we analyze the TCP NewReno behavior in detail.

- 1)  $t = t^*$ : At this time, the handoff is performed. The old connection is closed, and a new connection is opened

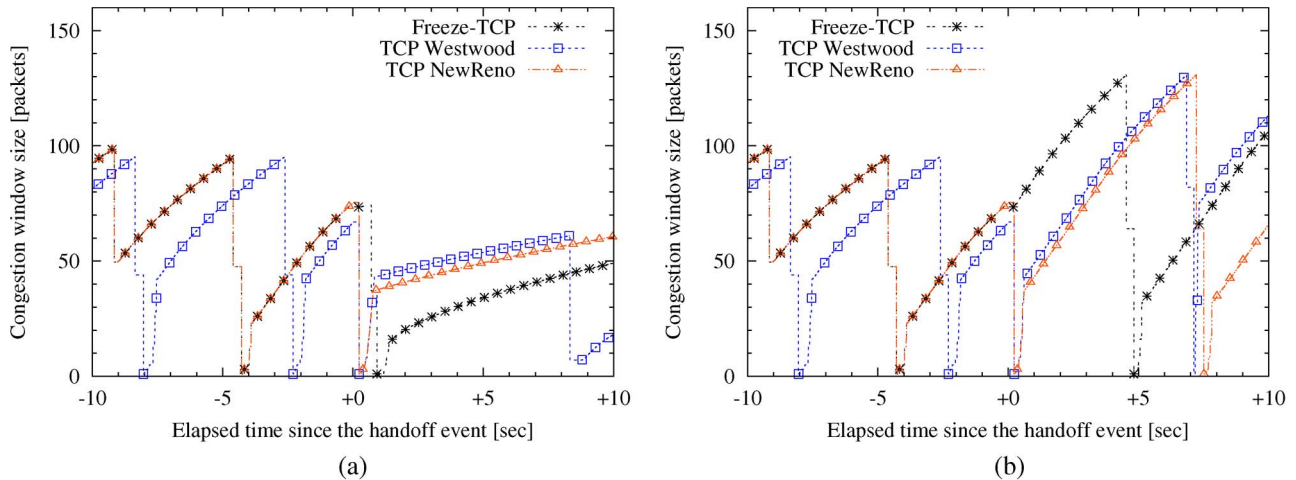


Fig. 4. Variation of the congestion window size during handoff in the case of TCP NewReno, TCP Westwood-NR, and Freeze-TCP schemes. (a) H-L scenario (6 Mb/s  $\rightarrow$  1 Mb/s). (b) L-H scenario (6 Mb/s  $\rightarrow$  11 Mb/s).

via the new AP. Consequently, after a timeout, TCP NewReno sets ( $cwnd := 1$ ).

- 2)  $t > t^*$ : The sender enters the slow start phase. Upon reaching the slow start threshold ( $ssthresh$ ), the sender switches to the congestion-avoidance phase [30].

In Fig. 4(a) and (b), we observe that starting from  $t \geq 0$ , TCP NewReno exhibits poor performance when compared to the cross-layer approach. This is intuitively due to the slow delivery of data packets when the new connection is established.

**TCP Westwood:** Finally, we examine the behavior of TCP Westwood in the two scenarios examined.

- 1)  $t = t^*$ : At this time, the handoff causes timeout expiration. As a consequence, TCP Westwood sets ( $cwnd := 1$ ) and ( $ssthresh := BWE$ ), where  $BWE$  is the connection BandWidth Estimate that is defined as the rate at which data are delivered to the TCP receiver [31]. The estimate is based on the rate at which ACKs have been received and on their payload. Note that these values are obtained before the handoff happens ( $t < t^*$ ).
- 2)  $t > t^*$ : A slow start phase starts until the value of  $ssthresh$  is reached. It is then followed by the congestion avoidance phase similar to NewReno [30]. Note that because  $ssthresh$  is set to a value calculated before handoff occurrence, TCP Westwood aggressively behaves in the H-L scenario. It, therefore, can result in a large number of packet drops. On the other hand, in the case of the L-H scenario, TCP Westwood underestimates the new bandwidth at least at the very beginning (establishment time) of the new connection.

From Fig. 4(a) and (b), we observe that TCP Westwood behaves in the same way as TCP NewReno. It thus exhibits poor performance compared with that of the proposed cross-layer approach.

Compared with other TCP versions, the proposed cross-layer approach demonstrates the best performance as it exhibits a very low packet loss rate in the H-L scenario and makes an efficient use of the new bandwidth in the L-H scenario.

2) **RTP-Based Multimedia Services:** In RTP-related simulations, the third operation of the cross-layer design at the

sender side (streaming rate adjustment in Fig. 1) is performed using the loss-delay-based adjustment (LDA+) algorithm [32]. The reason behind this choice underlies the fact that LDA+ achieves relatively good TCP friendliness, even when RTCP generates feedback messages at low frequencies. In this context, it should be noted that while frequent transmissions of control messages are beneficial for quick adaptation to sudden changes in network conditions, it incurs overhead in terms of bandwidth consumption. Another reason behind the choice of LDA+ consists of the fact that we assumed all wireless links to be error free in the conducted simulations. Indeed, in case of low bit error rate (BER) environments, the use of LDA+ as a rate control method suffices. However, in high BER environments, the LDA+ scheme can be substituted by more adequate schemes such as ARC [27], TCP Friendly Rate Control (TFRC) [33], or the Rate Control Scheme (RCS) [34]. As for the underlying protocol, RTP can be implemented on any network type. It can, indeed, work on TCP/IP, ATM, or frame relay. In the conducted simulations, User Datagram Protocol (UDP) is used as the transport protocol and IP as the network protocol.

In this analysis, we compare the transition of the streaming (sending) rate in the proposed approach with that of the standard RTP. Fig. 5(a) and (b) shows the transitions of the streaming rate in H-L and L-H scenarios, respectively. In these plots, the handoff begins at  $t^* = 0$  s.

**Cross-layer approach:** Here, we show the streaming rate control during handoff.

- 1)  $t = t^*$ : At this time, the handoff operation starts. The receiver transmits an RTCP HN packet to the RTP sender. In response, the sender transmits dummy RTP packets through the new AP at the maximum streaming rate of the data.
- 2)  $t = t^* + \text{RTT}$ : After an RTT elapsed since the transmission of the RTCP HN packet, the RTP receiver begins to receive the dummy RTP packets. Note that the sending rates of dummy packets are not reflected in Fig. 5 since they do not convey the actual data.
- 3)  $t = t_4$ : Let  $t_4$  ( $t_4 > t^*$ ) denote the time instant when the RTP receiver transmits an RTCP HR packet to the sender.

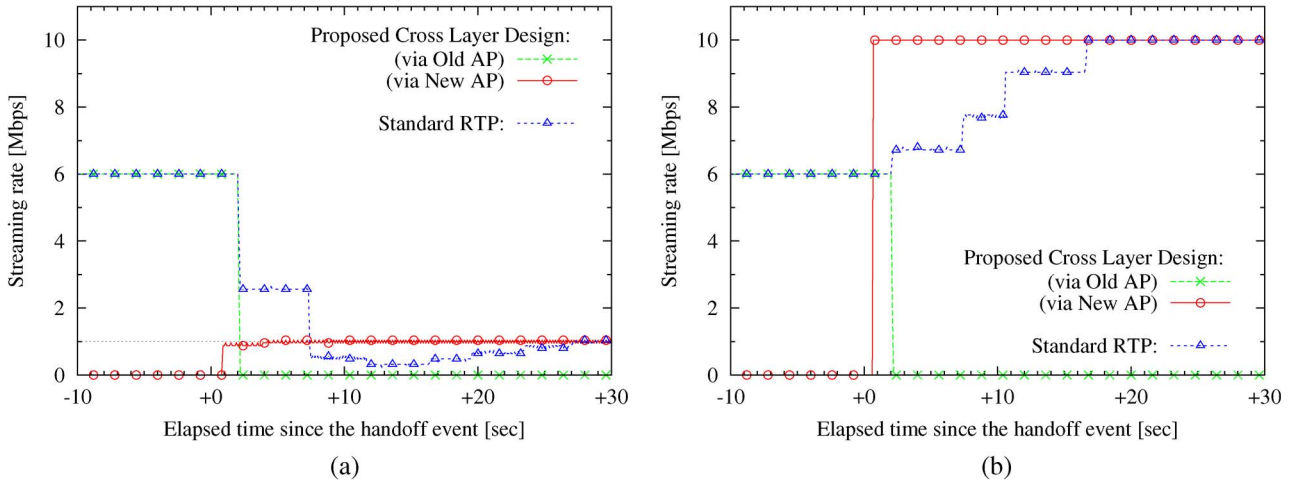


Fig. 5. Transition of the streaming rate during handoff in the case of the proposed cross-layer design and standard RTP. (a) H-L scenario (6 Mb/s  $\rightarrow$  1 Mb/s). (b) L-H scenario (6 Mb/s  $\rightarrow$  11 Mb/s).

After receiving dummy packets for a predefined period of time, the RTP receiver sends an RTCP HR packet through the new AP.

- 4)  $t > t_4$ : Upon receiving the RTCP HR packet, the RTP sender calculates the appropriate streaming rate from the information included in it, as in the case of receiving an RTCP RR packet. It then transmits normal RTP packets through the new AP at the computed rate.

Fig. 5(a) and (b) shows that, in both scenarios, the proposed approach sends data at the appropriate streaming rate according to the bandwidth estimated in the new cell after handoff.

*Standard RTP:* Here, we analyze the standard RTP behavior.

- 1)  $t = t^*$ : The RTP sender starts transmitting packets through the new AP but keeps the sending rate, which was adapted to the previous cell. Therefore, in the L-H scenario, its rate is below the available bandwidth in the new cell. On the other hand, in the H-L scenario, the network in the new cell falls into congestion, and a lot of packet drops are caused.
- 2)  $t = t_5$ : Let  $t_5 (t_5 > t^*)$  denote the time instant when the RTP sender receives the first RTCP RR packet since the handoff event. The RTP sender calculates the streaming rate from the information included in the RTCP RR packet and adjusts the sending rate to the computed rate.
- 3)  $t > t_5$ : Even after receiving the RR packet, the new streaming rate can be inaccurate. This is due to the fact that the first RR packet includes the old information that is not valid for the new cell. Furthermore, if the streaming rate falls below the available bandwidth, it takes a long period of time to achieve the appropriate rate to the new cell because LDA+ incrementally increases it. For instance,  $t_5$  is equal to 2.039 s in Fig. 5(a) and (b).

From these figures, in both scenarios, the proposed cross-layer approach shows better performance than the standard RTP as it appropriately adjusts the streaming rate immediately after handoff.

### C. Simulation Results

1) *TCP-Based Multimedia Services:* To evaluate the performance of the cross-layer architecture in delivering TCP-based multimedia services, two quantifying parameters are used, i.e., average throughput and loss rate. Throughput indicates the number of bytes received by a mobile node during the handoff period. The loss rate is the ratio of dropped packets to the aggregate sent packets during the handoff period. As dummy packets do not carry any new information, they are considered in neither the computation of the loss rate nor the throughput.

First, to investigate the robustness of the proposed cross-layer design in anticipating handoff events and promptly adjusting the transmission rate to the available bandwidth in the new wireless cell, we envision a scenario where the bandwidth of the old cell is set to 6 Mb/s and the bandwidth of the new cell is varied from 1 to 11 Mb/s. The moving speed of the mobile node is set to 50 km/h.

Fig. 6(a) compares the throughput of the proposed cross-layer design with that of the three other TCP variants for different disparities in the available bandwidth. The figure demonstrates that the proposed cross-layer design achieves the highest throughput compared with TCP Westwood-NR, Freeze-TCP, and TCP NewReno. It also shows an abrupt increase in throughput achieved by the proposed cross-layer design. When the available bandwidth in the new network is lower (the range of negative values on the  $x$ -axis), the four simulated schemes exhibit smaller throughputs. This is simply due to the fact that the bandwidth in the new network becomes less available. On the other hand, when the new cell has a higher bandwidth (the range of positive values on the  $x$ -axis), the proposed cross-layer design gains up to more than 200% over the three other schemes. This significant gain is mainly due to the fact that dummy segments inform the sender of the extra bandwidth becoming available in the new cell within a single RTT and accordingly stimulate it to increase its sending rate.

Fig. 6(b) illustrates the performance of the four schemes in terms of packet drops. The packet loss rate is plotted as a function of the difference between the available bandwidths in both the new and old networks. The results show that the



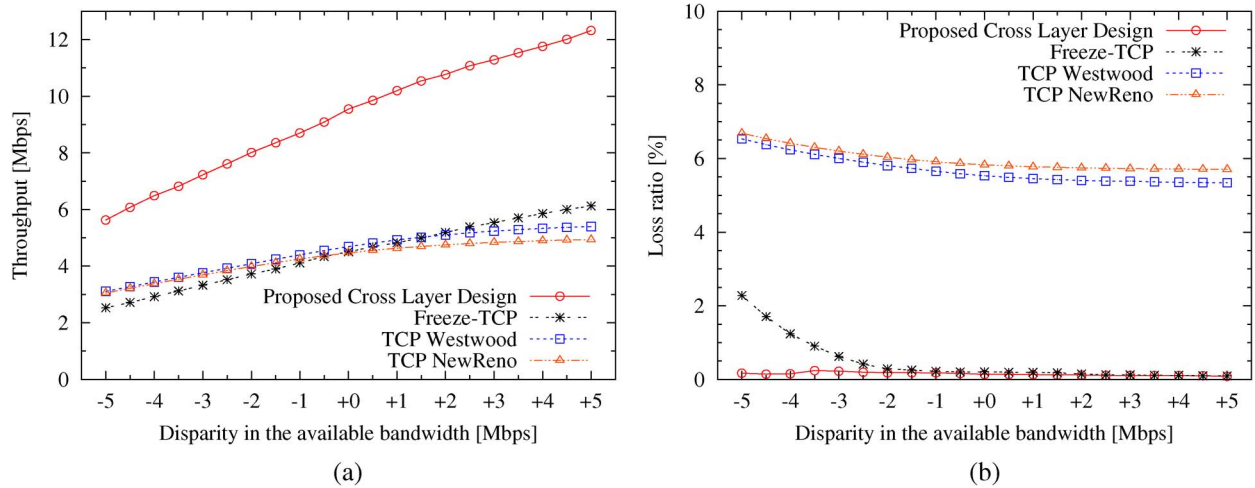


Fig. 6. Transmission efficiency for different disparities in the available bandwidth (mobile node speed = 50 km/h). (a) Throughput. (b) Packet drop rate.

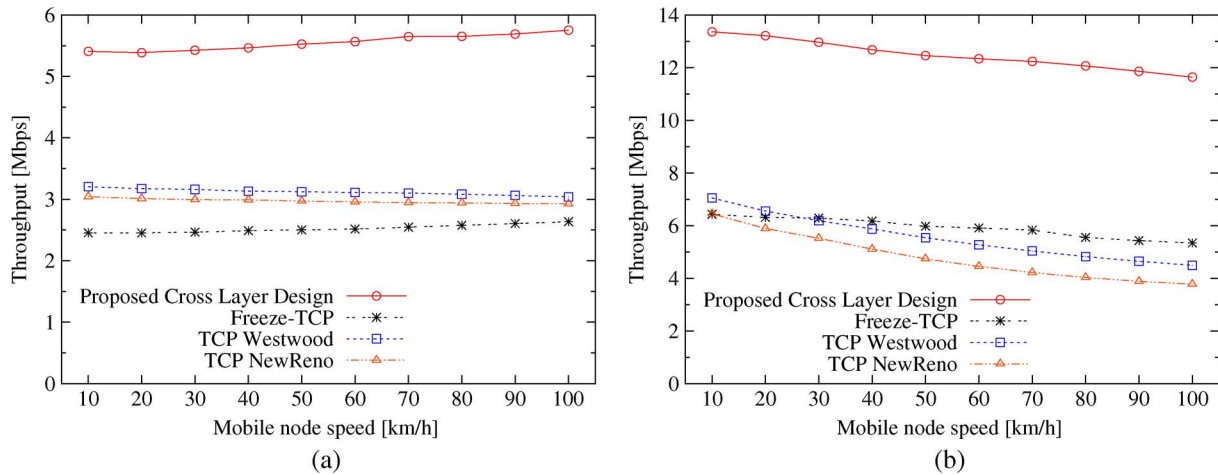


Fig. 7. Throughput variation for different mobile node speeds. (a) H-L scenario (6 Mb/s  $\rightarrow$  1 Mb/s). (b) L-H scenario (6 Mb/s  $\rightarrow$  11 Mb/s).

proposed cross-layer architecture and Freeze-TCP achieve the lowest packet drop rate. The proposed scheme further outperforms the Freeze-TCP scheme and achieves almost zero drops, regardless of the available bandwidth in the new cell. The main reason for this performance is in the intrinsic characteristic of the proposed scheme. Indeed, the proposed cross-layer design uses dummy segments to estimate the optimum rate at which the sender should send data. Accordingly, the sender avoids overloading the network with data packets that would ultimately be dropped otherwise.

On the other hand, while TCP Westwood-NR and TCP NewReno exhibit a throughput that is relatively equal to that of the proposed cross layer when the bandwidth in the new cell becomes less available ( $\ll 3$  Mb/s), their achieved throughput comes at the price of significant packet drops. This remark is illustrated in Fig. 6(b). Indeed, the results of the figure indicate that the two schemes experience the highest packet drop rate. This poor performance is mainly due to their bursty nature. In fact, both schemes keep transmitting data at window sizes that cannot be accommodated by the new network. This leads to congestion and ultimately higher drops. In summary, since the proposed cross-layer design uses dummy segments to probe the available bandwidth in the new cell, it achieves the highest

throughput and maintains the lowest drop rate compared with the other three schemes.

In light of the narrow surface of the cell overlapping area, the length of the handoff period becomes shorter as the mobile node speed increases. This decrease in length of the handoff period may influence the working of the proposed cross-layer design as the time required by the cross-layer architecture to manage handoff and to probe for bandwidth availability becomes shorter. To investigate such an impact, we vary the speed of the mobile node from 10 to 100 km/h. We envisage two scenarios, i.e., H-L and L-H scenarios. Throughputs of the proposed cross-layer design and the other three TCP variants for different mobile node speeds are graphed in Fig. 7. The figure confirms the impact of the mobile node speed on the throughputs of the four methods as their throughputs decrease with an increase in the mobile node speed. Nevertheless, it shows that the throughput of the proposed cross-layer design remains the highest in both scenarios and for all considered speeds.

In the remainder of this section, we envision a scenario whereby a TCP connection competes for bandwidth with  $N$  TCP connections in the new cell after handoff. A roaming TCP receiver performs handoff from a 6-Mb/s cell to an 11-Mb/s

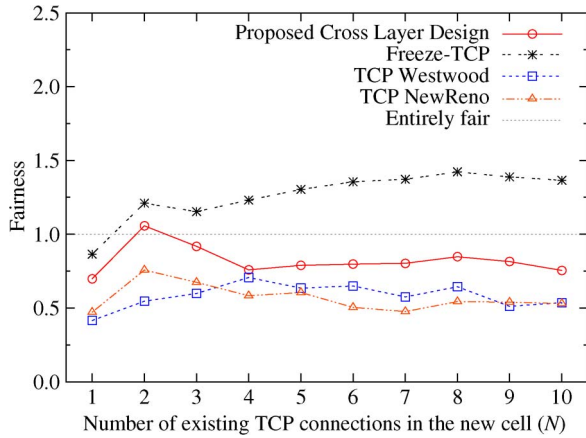


Fig. 8. Fairness to existing TCP connections after handoff.

cell. The other users remain in the same cell. As a fairness index, we use the following metric:

$$F_T = \frac{r_{hTCP}}{r_{eTCP}} \quad (1)$$

where  $r_{hTCP}$  and  $r_{eTCP}$  denote the throughput achieved by the roaming user via the new AP and the average throughput of the other  $N$  TCP connections, respectively. Each throughput is measured for 5 s after the handoff occurrence time.  $F_T = 1$  means that the newly coming user and the already-existing users are evenly sharing the bandwidth.  $F_T > 1$  indicates that the newly coming user conquers a portion of the cell bandwidth that is higher than that used by the old users. Fig. 8 shows that the proposed cross-layer design achieves better fairness than the other TCP variants. This is due to the fact that the proposed scheme estimates the available bandwidth while taking into account the network dynamics. In contrast, Freeze-TCP affects the other traffic in the new cell as it does not adjust the window size after handoff.

2) *RTP-Based Multimedia Services*: To highlight the efficiency of the proposed enhancements to RTP when implemented over the proposed cross-layer design, we compare its performance against that of standard RTP. In the case of standard RTP, we ignore both the delay that is due to the handoff operation and the in-flight packet drops that may happen during the handoff operation. The rationale behind this setting is to investigate the system performance in terms of packet drops due only to delay in adjusting the streaming rate and not delay in the management of the handoff. In the performance evaluation, two metrics are used, i.e., throughput achieved by the receiver and packet losses that occurred along the communication path. Here, packet losses are computed every 100 ms. They do not include dummy packet drops.

To investigate the interactions of RTP with the proposed cross-layer architecture, we consider an RTP mobile receiver moving between a higher bandwidth cell (6 Mb/s) and a lower bandwidth cell (1 Mb/s). Fig. 9 plots the transition of the experienced packet losses and the actual throughput achieved by the mobile node when the node performs handoff to a cell with less bandwidth. The figure shows that the standard RTP achieves a slightly higher throughput than the proposed cross-layer design. This performance, however, comes at the

price of significant packet drops, as indicated in Fig. 9(b). This performance is attributable to delay in the adjustment of streaming rate. Indeed, until reception of an RR packet message, the standard RTP sender keeps transmitting data at high rates that cannot be accommodated by the resources of the new cell, as shown in Fig. 5(a). The figure shows that even after receiving an RR packet message (2.039 s after the handoff event), the new streaming rate is not accurately computed and largely exceeds the available bandwidth at the new cell. This is due to the fact that the computation of the new streaming rate is based on old information that is not valid for the new cell. In the case of the proposed cross-layer design, when a handoff is about to occur, the sender gets notified of the event via an RTCP HN message. In response, it starts transmitting dummy packets to the receiver via the new AP. The receiver uses these dummy packets to make an accurate estimate of the bandwidth of the new network and reports it to the sender via the RTCP HR message. The sender promptly adjusts its streaming rate to the bandwidth of the new cell. This helps to avoid overloading the network with data packets and to accordingly elude packet drops, as indicated in Fig. 9(b).

To ensure that the proposed cross-layer design makes efficient use of the network resources when more bandwidth becomes available in the network, we consider a scenario where a mobile node roams from a lower bandwidth cell to a higher bandwidth cell (6 Mb/s  $\rightarrow$  11 Mb/s). Fig. 10 plots the transition of the mobile node's throughput 10 s before and 30 s after the handoff occurrence time. In this simulation, as packet drops were observed neither in the proposed cross-layer design nor in the standard RTP protocol, we do not graph packet losses. Fig. 10 shows that the proposed cross-layer scheme achieves higher throughput compared with the standard RTP immediately after the handoff event. This demonstrates the robustness of the proposed cross-layer scheme to adapt to changes in the wireless network environment. It also indicates the accuracy in probing bandwidth availability using dummy packets. On the other hand, the performance of standard RTP remains limited as the sender keeps streaming data at rates far below the available bandwidth in the new network, which is for a fairly long period of time after the handoff occurrence. Moreover, the bandwidth estimation of standard RTP is inaccurate as it is based on old information provided by RTCP RR messages. The inaccuracy of the bandwidth estimation is manifested in the stair-step shape of the streaming rate graph of standard RTP (Fig. 10). In this example, the sender needed nearly 20 s after the handoff event until it could be able to stream data at the available bandwidth of the new network.

Finally, we evaluate the performance of the proposed cross-layer design in a scenario whereby an RTP connection shares bandwidth with  $N$  TCP connections in the new cell after a handoff. In this scenario, an RTP receiver performs handoff from 6 to 11 Mb/s. As a friendliness index, we use the following metric:

$$F_R = \frac{r_{hRTP}}{r_{eTCP}} \quad (2)$$

where  $r_{hRTP}$  and  $r_{eTCP}$  denote the throughput of the RTP connection via the new AP after handoff and the average throughput of the  $N$  TCP connections, respectively. Each throughput is

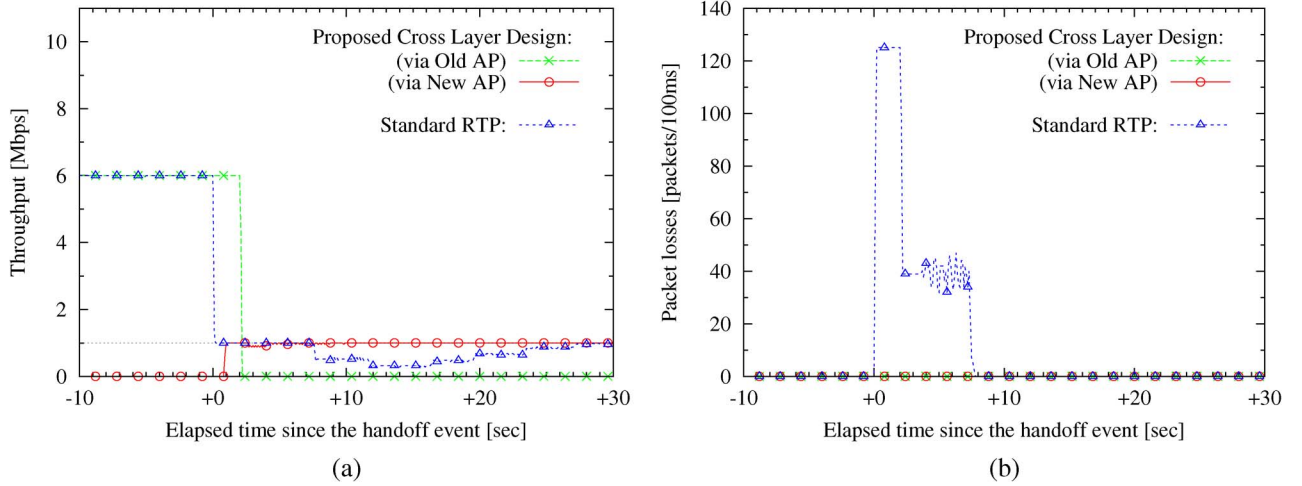


Fig. 9. Transmission efficiency in H-L scenario (6 Mb/s → 1 Mb/s). (a) Throughput. (b) Packet losses.

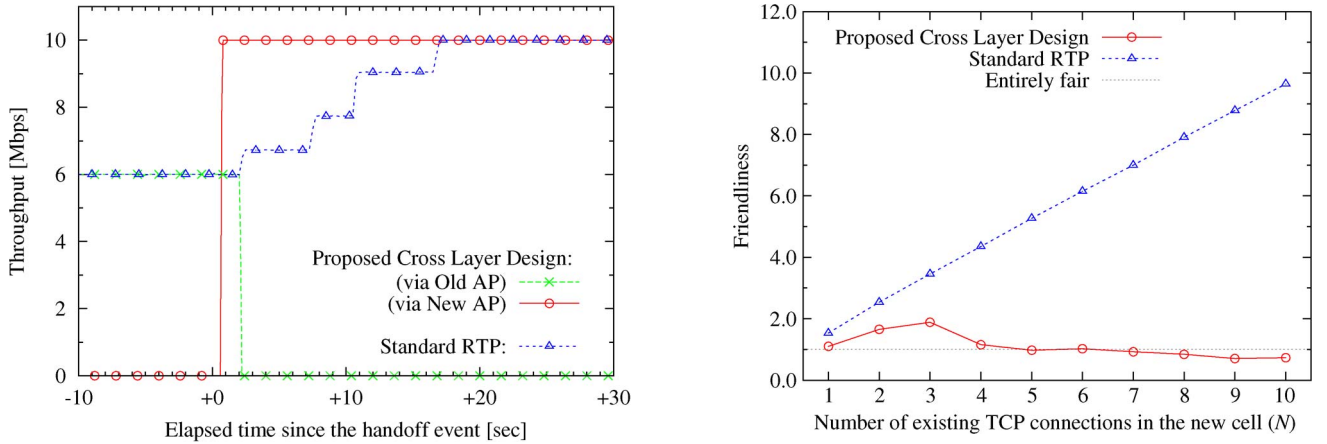


Fig. 10. Transmission efficiency in L-H scenario (6 Mb/s → 11 Mb/s).

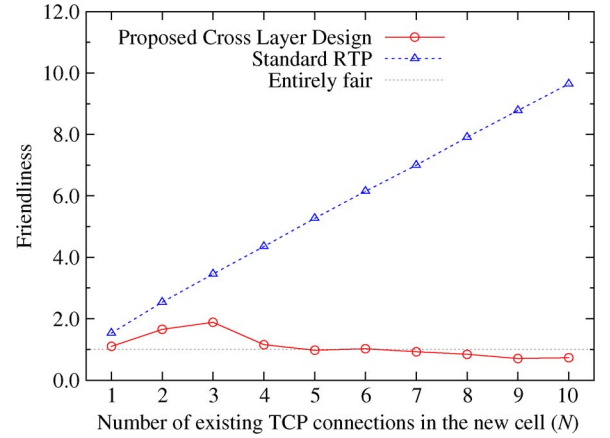


Fig. 11. Friendliness with existing TCP connections after handoff.

measured for 5 s after the handoff occurrence time. Fig. 11 plots the friendliness index of both schemes as a function of the total number of competing TCP connections  $N$ . The figure indicates that the proposed approach achieves TCP friendliness faster than the standard RTP. In standard RTP, RTP traffic causes a large number of packet losses in all the connections in the new cell after handoff and accordingly unfairly degrades the throughput of existing TCP traffic. In contrast, the proposed approach exhibits better friendliness as it adjusts the streaming rate and sets it to moderate rates after handoff.

#### D. Discussion

In the proposed cross-layer approach, bandwidth probing is based on dummy packets. Admittedly, reception of dummy segments and transmission of dummy ACKs by mobile nodes result in additional energy consumption. The proposed cross-layer scheme may thus be seen as costly in terms of reducing the battery life of mobile nodes. However, the performance gains achieved by the proposed cross-layer architecture in terms of both throughput and reduced packet drops are worthwhile and can be used to advocate for this additional cost. Indeed, the high throughput and low packet loss rates of the proposed cross-layer design lead to a significant reduction in the overall

transmission time of a given data file. This intuitively reduces the overall usage time of the mobile node battery and ultimately saves its energy. Moreover, apart from the rare case of mobile nodes flip-flopping over a cell overlapping area, the additional energy consumption due to dummy segments remains minimal.

To illustrate the idea at hand, we consider the following simple mathematical analysis. Let  $N_r$  and  $N_t$  denote the number of received dummy packets and the number of transmitted dummy ACKs by a mobile node. Let  $t_r$  and  $t_t$  denote the time required to receive a dummy packet and the time required to transmit a dummy ACK packet. Denoting by  $I_r$  and  $I_t$  the amount of electric current required to receive a single packet and the amount of electric current required to transmit a single ACK packet, the battery consumption of a mobile node due to dummy packets and ACKs can be expressed as follows:

$$B_{\text{TCP}} = N_r \cdot I_r \cdot t_r + N_t \cdot I_t \cdot t_t. \quad (3)$$

As a mobile node sends back an ACK for each received dummy packet,  $N_t = N_r = N$ . The above equation can thus be simplified as follows:

$$B_{\text{TCP}} = N(I_r \cdot t_r + I_t \cdot t_t). \quad (4)$$

Using the specifications of the WLAN CardBus adapter developed by Cisco [35], when 802.11b is in use,  $I_t$  and  $I_r$  can be set to a maximum of 539 and 327 mA, respectively. Additionally, the times required to receive a data packet with a length of 1000 B and to transmit an ACK packet of 32 B are equal to 727 and 23  $\mu$ s, respectively ( $t_r = 727[\mu\text{s}]$ ,  $t_t = 23[\mu\text{s}]$ ). Using these values, the consumed battery in the case of receiving  $N$  dummy packets and transmitting  $N$  dummy ACKs is simply

$$B_{\text{TCP}} < 0.0067 \cdot N[\text{mA} \cdot \text{min}]. \quad (5)$$

Even in the case of 100 dummy packets, the consumed battery is less than 0.67 mA-min. For a mobile phone with a battery lifetime equal to 730 mA-h (43 800 mA-min), the consumed battery represents a negligible amount. All in all, along with ongoing advances in technologies related to batteries, the use of dummy packets to probe for bandwidth availability shall not be an issue for mobile users.

In a similar way, when a mobile node receives  $N_r$  dummy RTP packets, the battery consumption due to dummy packets can be calculated as<sup>1</sup>

$$B_{\text{RTP}} = N_r \cdot I_r \cdot t_r. \quad (6)$$

If a mobile node receives RTP packets at 10 Mb/s for 0.5 s, the consumed battery is less than 150 mA-min.

## V. CONCLUSION

In this paper, we have proposed a cross-layer design for the efficient delivery of multimedia services in heterogeneous wireless networks. The designed cross-layer architecture involves five layers. The proposed interactions between the layers are simple and practical. The key idea behind the proposed cross-layer architecture is to anticipate imminent handoffs, to notify senders with these events, and to stimulate them to probe for the resources of the new wireless network. Dummy packets are used for this purpose. Two types of multimedia applications are considered, namely, TCP-based and RTP-based applications. For each type, adequate enhancements are proposed.

The performance of the proposed cross-layer architecture is evaluated for both TCP-based and RTP-based applications using computer simulations. Simulation results elucidate the outstanding performance of the proposed cross-layer architecture in achieving high throughputs while reducing packet drops. The results also demonstrate the effectiveness of dummy packets in making accurate estimation of the available bandwidth. The resiliency of the proposed cross-layer design to changing network conditions is also verified. The results are promising for streaming multimedia services in heterogeneous wireless networks where the disparity in the available bandwidth is still a major issue to solve.

From the simulation results, we believe that the proposed cross-layer design represents an important contribution to the field of multimedia delivery in heterogeneous wireless net-

works. It is the authors' hope that the findings in this paper would stimulate further research work in the area.

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<sup>1</sup>Here, we ignore two packets (RTCP HN and HR).



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**Kenichi Kashibuchi** (S'05) received the B.E. degree in information engineering and the M.S. degree in computer sciences in 2005 and 2007, respectively, from Tohoku University, Sendai, Japan, where he is currently working toward the Ph.D. degree.

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