Efficient Cross-Layer Simulator for Performance Evaluation of Wireless Mesh Networks

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ABSTRACT

Wireless mesh networks (WMNs) are emerging as a promising technology for backhauling data traffic from wireless access networks to the wired Internet that expected to support various types of applications with different quality of service (QoS) requirements. While wireless mesh networking has attracted great industrial and academic interest, many research challenges remain in all protocol layers. Moreover, traditional layered approaches, initially designed for wired networks, have been proven insufficient for WMNs due to the wireless channel variability, co-channel interference and new data traffic peculiarities.

In this work we provide a complete cross-layer solution for WMNs. Our protocol architecture framework comprises a novel joint QoS routing and opportunistic scheduling scheme that exploits the multi-user diversity gain while guarantees end-to-end packet delivery that satisfies the multiple QoS requirements of the underlying applications. On top of this, the interaction of these algorithms with the transport layer is investigated where the suitability of several techniques, such as Explicit Congestion Notification, Explicit Loss Notification and Explicit Rate Notification is considered. In order to assess the performance of the proposed protocols, a realistic and comprehensive simulation platform for WMNs that spans all layers from physical to application has been implemented using OPNET modeler. This paper provides a detailed description of the proposed protocols and the simulation platform together with some performance evaluation results.

Categories and Subject Descriptors

C.2.1 [**Computer-Communication Networks**]: Network Architecture and Design—*Network communications*; C.2.2 [**Computer-Communication Networks**]: Network Protocols—*protocol architecture (OSI model), routing protocols*; D.4.8 [**Operating Systems**]: Performance—*simulation*

General Terms

Algorithms, Experimentation, Performance

Keywords

Performance evaluation, Wireless mesh networks, Cross-layer design, QoS, OPNET

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1. INTRODUCTION

Ubiquitous data access has been for long time the holy grail for service providers. Recent advantages in wireless technologies such as WiFi and WiMAX have enabled fast and convenient data access to users. Nevertheless, the backhaul networks that connect the access points to the core network continue to be a bottleneck both from data capacity and operational cost perspective. Wireless mesh networks (WMNs, [6], see Figure 1), an emerging and promising key technology, are expected to provide a quick and efficient solution to this problem in urban, suburban and even rural environments. WMNs are composed of static wireless nodes/mesh routers (WMR) with ample energy supply. These nodes may operate not only as conventional access point or Internet gateway, but more important, as wireless routers that form an ad hoc network able to relay packets from other nodes without direct access to their destinations.

While many design paradigms for wired or ad hoc networks have been proposed, the differences between wired and wireless communications together with the dynamic and ad hoc nature of mesh networks necessitate the design of new solutions for WMNs [36]. From physical (PHY) layer perspective, intelligent antenna techniques can be used to mitigate the unpredictable and time-varying interferences among wireless links generated by highly dynamic traffic requirements that alter the feasible network capacity regions. From medium access control (MAC) layer perspective, efficient resource allocation schemes shall be developed to exploit the multi-user diversity gain of wireless channels. Moreover, novel routing algorithms shall be designed that harmonically cooperate with the new algorithms at the lower layer in order to provide multi-constrain quality of service (QoS)

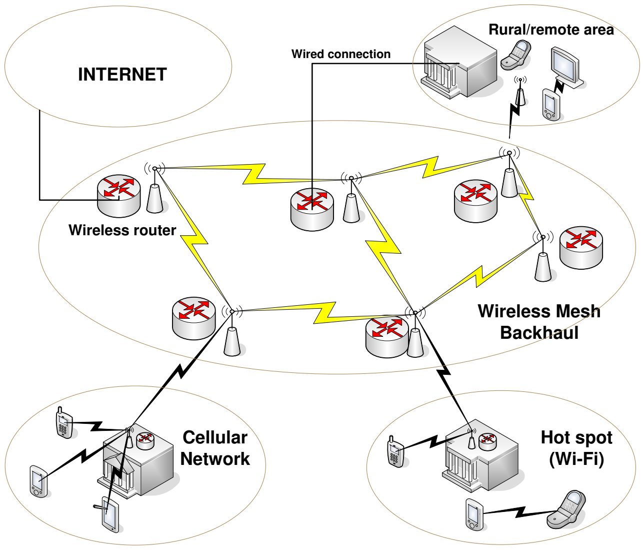

Figure 1: Typical wireless mesh network scenario.

to a wide range of applications along the entire route of communication.

On the other hand, Transport Control Protocol (TCP) should be modified in a way that can acquire useful information of the network state in terms of congestion, packet loss or link failure due to buffer overflow, channel errors, or unavailable data rate in order avoid erroneous decisions. The above operational characteristics in those four layers are primarily the reason why an accurate and efficient cross-layer design paradigm for WMNs should be built, and why traditional division of the communication functionalities where each layer has its own protocol and executes its own task do not perform well.

In this paper, we propose an innovative cross-layer design paradigm for WMNs that allows every layer of intermediate node to have a clear picture of other layers' behaviours of the same node and exploit this knowledge to create synergies in the control functions. The cross-layer techniques proposed and evaluated in this paper include several enhancement techniques for TCP, a multi-constrain QoS routing algorithm, a distributed proportional fair scheduling framework and some PHY layer advanced communications and antenna techniques. We evaluate the proposed cross-layer approaches for WMNs by developing a complex simulation environment using the OPNET [1] modeler application, in which a realistic model of a WMN is created. In this model, all the cross-layer solutions have been implemented, and on top of the network, we have developed a set of real applications, like FTP and voice-over-IP (VoIP) calls. Furthermore, numerous configuration parameters have been defined to provide a great flexibility in the evaluation of the best parameter combinations according to a specific application traffic bundle. Simulation results shows that the proposed design paradigm could successfully provide a significant performance gain with a consequent reduction of the protocol layer redundancies.

2. RELATED WORK

The well-known ad hoc on-demand distance vector (AODV) [27] protocol, a reactive approach for route discovery and maintenance that finds the routes with minimum number of hops from source to destination in ad-hoc networks, is not suitable for high throughput and delay-sensitive applications. [34, 17] have addressed extensively on multi-constrained QoS routing algorithms in wired network based on network state [26, 19] to overcome the NP-complete difficulties of providing optimum routes that guarantee multiple QoS constraints [31]. Meanwhile, QoS routing algorithms for wireless ad-hoc networks have been previously explored in [37, 22, 21, 12, 11]. However, they either overlook the multi-hop queueing delays since only the packet processing time was considered or simply calculate the available bandwidth in terms of slot and reserved for QoS flows that fails to exploit the opportunistic scheduling gain in fast-fading channels.

On the other hand, scheduling for wireless mesh networks has drawn a lot of research attention recently. Due to the fact in [10, 28, 18] that finding a perfect match with the highest network throughput is NP-complete [25, 14] for centralized scheduling algorithms, various distributed scheduling algorithms were proposed. Recently, [15, 16] proposed a distributed opportunistic scheduling algorithm for backhaul networks, which provides multi-user diversity gain in the wireless environments, enforces resource allocation in the long run and maintains strong temporal correlation for interference, without which channel quality and interference cannot be tracked and predicted with reasonable accuracy.

The Transmission Control Protocol (TCP, [4]) is the standard transport protocol for IP networks, independently from the type of MAC and physical layer behaviors. It was originally designed for wired networks, in which TCP assumes that the PER is extremely low. In this case, TCP considers the reason of every occurring loss or timeout is due to buffer overflows in the intermediate nodes. Therefore, by using a congestion window [5], the sender is aware of the receiver buffer capacity and the available bandwidth in the network to control the sending rate. It limits the number of transmitted but not yet acknowledged segments, in which way TCP tries to reduce the number of congestion events due to buffer overflows, and thus packet losses are reduced. In order to check the network condition, TCP starts with a small value of the congestion window size, usually 1 Maximum Segment Size (MSS), and then increases the value of the window, probing for the existence of additional unused link bandwidth along the entire route towards the receiver. This continues until a loss or a timeout occur, when TCP reduces the window to a safe level and starts again probing for unused bandwidth. In the literature can be found many different congestion control approaches that are suitable for different network environments, like the TCP Reno and the TCP New Reno [3].

Unfortunately, the characteristics of wireless links in WMNs demand efficient design of TCP protocol collaboratively with lower layers, due to many uncontrollable quality-affecting factors, such as weather conditions, urban obstacles, multipath interferences, large moving objects, and mobility of wireless end devices [30, 13, 33]. Therefore, in a network with wireless links, the packet losses are often caused by corruption due to link errors or high PER, but not by congestion. Therefore, TCP's congestion control will mistakenly and unnecessarily reduce the sending rate, causing a degradation of the network throughput, inefficiency of network resource utilization and continuing interruptions of data transmission. Moreover, in a wireless environment there could be multiple random packet losses within a single Round Trip Time (RTT), because errors on bit occurs in short bursts, causing problem in the most used TCP congestion control algorithm like TCP Reno. Another issue is the link asymmetry, i.e., the uplink and the downlink may have different capacity, and thus the downlink acknowledgement (ACK) packet can be affected by delay and losses due to bit corruption at TCP layer, causing again a reduction of the transmission rate. Finally, the RTT in wireless environment can be very variable that may cause unnecessary retransmissions of the data packets, called spurious retransmissions.

Various solutions has been proposed in order to solve the issues of TCP in wireless networks [7]: they can be classified into two main groups, the link layer solutions and the end-to-end (ETE) solutions. The link layer solutions are composed of methods based on improvements of FEC (i.e., try to improve the algorithms for detecting and correcting errors on bits at layer 2) and automatic repeat request (ARQ, i.e., layer 2 tries to mange packet retransmissions). Both approaches introduce complexity in the system, even if they are transparent to the upper layers. The ETE solutions, or cross-layer solutions, change the approach that TCP traditional congestion control algorithms adopt, in which the network is considered a "black box". The idea is to let the intermediate nodes lower layers to report explicitly some information to the end node TCP layers, which can react accordingly. In this group we can distinguish the reasons of using explicit notifications into three scenarios, due to: congestion [2], loss [8, 35]), or available bandwidth estimation (TCP TIBET [9] and TCP Jersey [20, 32]).

The rest of the paper is organized as follows. In Section 3, the network configurations and PHY/link layer models are introduced. QoS routing algorithm in the network layer is described in Section 4. Section 5 provides a thorough description of the used MAC layer distributed scheduling framework. Cross-layer TCP protocols are described in Section 6. OPNET simulation environment setup and extensive simulation results are given in Section 7. Finally, conclusions are drawn in Section 8.

3. SYSTEM MODEL

Consider a wireless mesh network which comprises a set of n_r number of WMRs, denoted as $V_R = \{v_r | r = 1, 2, \ldots, n_r\}$ and a set of n_g number of gateways denoted as $V_G = \{v_g | g = 1, 2, \ldots, n_g\}$. QoS flow with index q is generated with a set of constraints, ETE packet delay D_q^r, throughput T_q^r and PER E_q^r. If further consider an arbitrary node i, it has K_i number of one-hop neighbors within fixed transmission range, where these neighbors are $k = 1, 2, ..., K_i$. Meanwhile, a separate queue is attached in each mesh router for each direction of transmission, and multi-hop packets are queued into a specific queue according to pre-found routing sequence.

A route k from a source WMR s to a destination gateway t within the route set Ω_{st} is concatenated by a set of links $\biguplus(v_i, v_j) = \{(v_s, v_{s+1})(v_{s+2}, v_{s+3})...(v_i, v_j)...(v_{t-1}, v_t)\}$, for all $v_i, v_j \in V_R \bigcup V_G$. Therefore, we could formally express the route k from s to t as in (1), where total m candidate routes exist. In the following discussions, we use the term session and flow for the traffic input, (v_i, v_j) and (i, j) for the link between v_i and v_j interchangeably.

$$k = \left\{ \biguplus(v_i, v_j) | \forall v_i, v_j \in V_R \cup V_G, k = 1, 2, \ldots, m \right\} \quad (1)$$

The network runs under a time-division multiple access (TDMA) slotted framework, and we assume all nodes are perfectly synchronized. The time frame consists of f fixed-size time slots, during which the channel remains the same if we assume block fading. Scheduling decisions are taken by all nodes in the network simultaneously at the beginning of each time frame, and stay unchanged until the next frame.

The PHY layer employs the adaptive modulation and coding techniques (AMC), where there are a finite number V of transmission modes, each of which corresponds to a unique modulation and coding scheme and one particular interval of the received signal-to-interference-plus-noise ratio (SINR). The transmission rate at each mode is proportional to its spectral efficiency, i.e., transmission mode v can transmit maximum c_v packets in one time slot, where $v = 1, 2, ..., V$, or $H = fc_v$ packets in a frame. The Rayleigh fading Model [29] is used for the wireless channel representation while the required PER is derived based on SINR curves for the used adaptive modulation and coding scheme. At given time t, the receiving SINR γ_{ij}^t for the transmitter-receiver pair (v_i, v_j) is given by (2),

$$\gamma_{ij}^t = \frac{P_{ij} C_{ij}^t d_{ij}^{-\alpha}}{\sum_k P_{kj} C_{kj}^t d_{kj}^{-\alpha} + \mathcal{N}_0} \quad (2)$$

where P_{ij}, C_{ij}^t and $d_{ij}^{-\alpha}$ are transmission power, channel gain (the antenna gain has been also included here) and path loss between link (v_i, v_j) respectively. Typical value for path loss exponential factor is 3.5. $\mathcal{N}_0$ is the single-sided power spectrum density for additive white Gaussian noise. Power control is not considered in this phase, i.e., all the nodes have the same fixed transmission power P_{ij}.

In order to reduce the interference to adjacent concurrent transmissions and increase the frequency reuse and channel capacity, the WMRs are equipped with directional antennas.

4. QOS ROUTING PROTOCOL

The problem of providing optimum routes that guarantee multiple QoS constraints has been proven to be NP-complete [31], and therefore, in order to overcome this difficulty we define a new utility function based on the "dissatisfaction ratio" $\mathbf{R}$ that experienced by each QoS metric. More specifically, we define the ratio $\mathbf{R}$ for each of the QoS requirement as follows:.

(1) $\mathbf{R}_k^D$: ETE packet delay dissatisfaction ratio for route k is defined as the actual delay measurement (accumulated delay hop by hop), $\sum_{(i,j)\in k} D_{ij}^a$, over the QoS delay requirement D_q^r, i.e.,

$$\mathbf{R}_k^D(q) = \frac{\sum_{(i,j)\in k} D_{ij}^a}{(1-\beta_D) D_q^r}. \quad (3)$$

(2) $\mathbf{R}_k^T$: Throughput dissatisfaction ratio is formulated as the ratio between the throughput requirement T_q^r and actual *bottleneck* link throughput, $\min_{(i,j)\in k} T_{ij}^a$, the minimum of all one-hop throughputs along route k, i.e.,

$$\mathbf{R}_k^T(q) = \frac{(1+\beta_T) T_q^r}{\min_{(i,j)\in k} T_{ij}^a} \quad (4)$$

Table 1: Resource reservation and indication factors for different traffic patterns.

	$\mathbf{I}_D$	$\mathbf{I}_T$	$\mathbf{I}_E$	β_D	β_T	β_E
voice-over-IP	1	1	0	var	var	-
Data (FTP etc.)	0	1	1	-	var	var

(3) $\mathbf{R}_k^E$: PER dissatisfaction ratio is defined as the multiplication of all one-hop error rate, $1-\prod_{(i,j)\in k}(1-E_{ij}^a)$, over PER requirement E_q^r since this is a multiplicative constrain, i.e.,

$$\mathbf{R}_k^E(q) = \frac{1-\prod_{(i,j)\in k}(1-E_{ij}^a)}{(1-\beta_E)E_q^r} \tag{5}$$

A resource reservation margin factor has been introduced as β_D, β_T and β_E for delay, throughput and PER respectively. In other words β represents the additional resources that we reserve beyond the QoS requirements in order to provide a safe guard for imperfect resource estimations and system fluctuations.

Since a session has to fulfil the set of QoS requirements, a source-to-gateway route will be feasible if and only if all defined ratios are less than one, $(\mathbf{R}_k^D(q), \mathbf{R}_k^T(q), \mathbf{R}_k^E(q)) \leq 1$. However, some constraints may not be critical in some applications (for instance, broadband data services are not sensitive in delay). In order to efficiently cope with this issue we introduce the indication function $\mathbf{I}_p$, where $p = D, T, E$, expressed as,

$$\mathbf{I}_p = \begin{cases} 1 & \text{if parameter } p \text{ is critical in QoS flow } q \\ 0 & \text{otherwise} \end{cases} \tag{6}$$

An example of the resource reservation margin factors and indication functions chosen for two types of QoS flows in the network, namely, voice-over-IP and broadband data services respectively, is demonstrated in Table 1.

Our multi-constrained QoS performance index for route k can be formulated as,

$$\mathbf{U}_k = \max\left[\mathbf{I}_D\mathbf{R}_k^D(q), \mathbf{I}_T\mathbf{R}_k^T(q), \mathbf{I}_E\mathbf{R}_k^E(q)\right] \tag{7}$$

and the proposed multi-objective function in order to take an optimum heuristic decision is given by

$$\mathbf{S} = \min_{\forall k\in\Omega_{st}} [\mathbf{U}_k] \tag{8}$$

5. MAC PROTOCOL

Opportunistic proportional fair scheduling, using SINR as a scheduling utility, has been proven beneficial in providing higher throughput by exploiting the multi-user diversity gain of wireless networks. However, such scheme comes with an inherent drawback, i.e., in many occasions it fails to guarantee the required QoS performance in a long run. This is because opportunistic decisions usually introduce more fluctuating instantaneous performance at individual incoming and outgoing links. In order to overcome this difficulty and enforce QoS, the distributed opportunistic proportional fair scheduler proposed in [15, 16] is considered in our cross-layer framework. In this scheme, the utility function (or scheduling metric) combines both routing and scheduling parameters and in a way that not only achieves opportunistic gain but can also supports quality of service as committed by the routing algorithm in use. This scheduling scheme has been proven not only to achieve a network throughput improvement but at the same time to allow for more accurate channel predictions by providing high level of temporal correlation of interference. This property is of paramount importance for the long-term prediction of channel quality required for the optimum performance of the routing algorithm as it will be described in the following.

The routing algorithm used in Section 4 estimates the QoS routing demand for the session q in a certain future (e.g., for the whole duration of a data session) and passes the scheduling the throughput allocation target a_{ij}^q for the link (v_i, v_j). For instance, the routing algorithm may ask for $a_{ij}^q = (1+\beta_T)\cdot T_q^r$ amount of bandwidth resources to be reserved on the link (v_i, v_j). The scheduling scheme at node i will generate the throughput allocation target vector $\overrightarrow{a_i} = (a_{i1}, a_{i2}, ..., a_{il})$ with the demands of all l incoming and outgoing links and activate the appropriate link for transmission-reception each time based on the following utility function,

$$\mathbf{U}_{ij} = a_{ij}^q \frac{\rho_{ij}}{\varphi_{ij}} \tag{9}$$

where ρ_{ij} and φ_{ij} are the instantaneous throughput and channel capacity in the long run respectively. Results in [15, 16] prove that, from MAC perspective, by choosing the proposed link utility metric (9) the scheduler guarantees the proportional target QoS throughput as well as fairness among links. However, in that work it is not clear how the localized and short term scheduling decisions may affect a more long term routing performance and guarantee multiple QoS requirements.

The performance of the proposed scheme highly depends on the accurate estimation of multiple system parameters required for the optimum routing and scheduling decisions. For this reason, each node keeps a table with measurement of previous transmissions from all its neighbors. The measured parameters include link throughput, link SINR and queuing/tranmission delay. The throughput statistics are passed to the scheduling scheme for the estimation of the long run channel capacity φ_{ij}. The SINR statistics for each link (i,j) and the queuing delay statistics in each node are used to estimate the expected PER and ETE packet delay, respectively, for the routing decisions.

6. CROSS-LAYER TCP PROTOCOLS

Traditional TCP has been proven successful in wired networks to provide end-to-end reliable communication and assure ordered delivery of packets by flow and error control mechanisms. However, wireless mesh networks impose new demands for efficient TCP protocol design that should be able to distinguish congestion from non-congestion loss, such as frequent link failures and channel fluctuations. Existing variations of TCP, like Reno, New Reno and Jersey react to packet losses can assist in configuring the congestion window size more properly. However, those protocols have been designed for low PER conditions and they do not interact or exploit any possible information from lower layers. Furthermore, advanced communications techniques, such as AMC, directional antenna beam-forming and opportunistic scheduling in lower layers can generate further throughput variations that need to be considered in the design on efficient TCP protocols for WMNs. In order to avoid unnecessary reductions in link bandwidth utilizations, significant throughput degradation and high interactive delays, we need to adopt a cross-layer approach where TCP exploits possible information from lower layers. Explicit Congestion Notification (ECN), Explicit Loss Notification (ELN), and Explicit Rate Notification (ERN) are the three techniques that we consider in conjunction with New Reno TCP in the following of this work.

6.1 Explicit Congestion Notification (ECN)

If a transmitted packet reaches a backhaul mesh router with congestion or buffer overflow, this intermediate node sets the congestion-experiencing bit in the IP header of the packet. When the considered packet finally reaches the destination gateway, the destination IP layer notifies the congestion at its TCP layer as shown in Figure 2(a). The TCP layer, in turn, sets the corresponding Explicit Notification Echo (ENE) bit in the TCP header of the corresponding ACK packet. When the TCP sender receives the ACK packet with the set ENE bit, it reduces the congestion window according to the traditional fast recovery and fast retransmission algorithms, and signals back to the TCP receiver using the Congestion Reduced Window (CRW) bit. When the TCP receiver receives packets with the CRW bit set, it stops setting the ENE bit in the ACK packet. This two-way approach is used also by TCP Jersey in conjunction with the estimation of the available bandwidth.

6.2 Explicit Loss Notification (ELN)

This is introduced when packet loss happens due to link failure. Consider an arbitrary chosen link $l = (v_i, v_j)$ at time t experiences channel fading and thus corrupt the transmitted packet from v_i. If no ARQ scheme is used in MAC layer, node v_j doesn't discard the packet even if it contains erroneous bits, but sends it to the IP layer. Then the IP layer sets the Loss bit in the IP header and forwards the packet to the destination as shown in Figure 2(b). When the packet is received by the gateway, it reports to the TCP layer that the corresponding packet is lost due to channel errors and the gateway will discard this erroneous packet. To implement this scheme, we introduce a new flag in the TCP segment, ELN bit, which, when it is set it means the sequence number in the "Acknowledgement number" field is the sequence number of the next segment expected that has been lost due to channel errors. When the TCP sender receives the ACK packet with the ELN bit set, it does not reduce the congestion window, but simply retransmits the lost packet. The ELN bit can work fine also in case of burst losses, because it selectively indicated the segment that must be retransmitted because of channel errors.

6.3 Explicit Rate Notification (ERN)

Due to the fact of large channel variability and co-channel interferences among adjacent nodes, the link throughput will change from time to time. These unexpected throughput fluctuations in the backaul network may result in low endpoint TCP performances and buffer overflow. ERN is the cross-layer scheme proposed to tackle this problem as shown in Figure 2(c). In each intermediate backhaul mesh router, the MAC layer periodically informs the IP layer of the available bandwidths on the link. The relay packet records this available bandwidth value in the IP packet header before

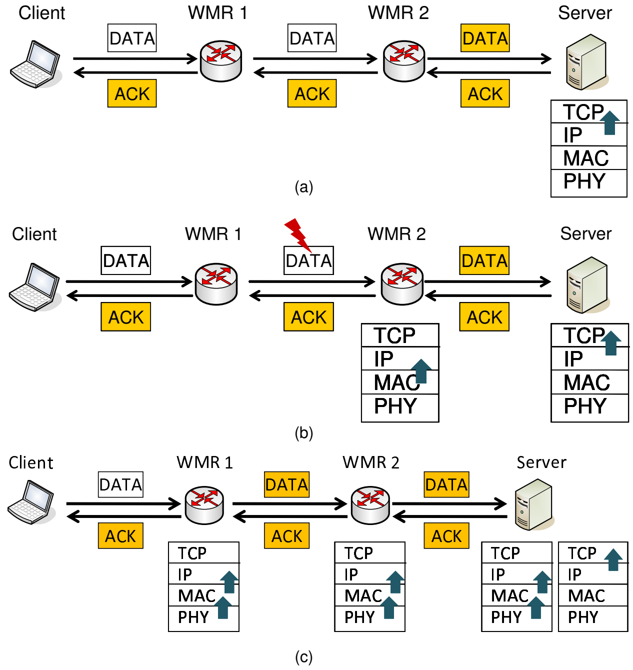

Figure 2: (a) Explicit congestion notification mechanism, (b) Explicit loss notification mechanism, and (c) Explicit rate notification mechanism.

sending it to the next hop. The subsequent intermediate nodes along the path towards the destination compare their notified available bandwidths with the one recorded in the receiving packets, and if lower, they update the value in the packet. Therefore, the end-to-end available bottleneck bandwidth for the whole TCP connection is recorded in the header in order to avoid network overload. In the TCP congestion control algorithms that use the bandwidth estimation for improving TCP performance, like TCP TIBET and Jersey, the TCP layer calculates the congestion window as the bandwidth delay product. In this case, instead of using an estimated value, the TCP receiver can use the information regarding the rate coming from the IP layer and multiply it with the current RTT. The obtained value is compared with the value to be used in the window field in the ACK packet and, if lower, it will be updated in the window field and sent back to the sender. The TCP sender reacts accordingly reducing the number of bytes to be sent in the network.

7. SIMULATION RESULTS

We use OPNET [1] modeler to create an integrated WMNs simulation environment where the performance and interaction of our proposed algorithms and techniques is evaluated. In this environment, eighteen wireless mesh routers are randomly and independently deployed on a two dimensional space consisting the backhaul network where variable number of client and server couples are connected, as shown in Figure 3. The developed wireless mesh router models representing the functionalities of different layers are shown in Figure 4(a). In PHY layer, the Rayleigh fading channel model [29] is adopted while the required PER is derived based on SINR curves for the used adaptive modula-

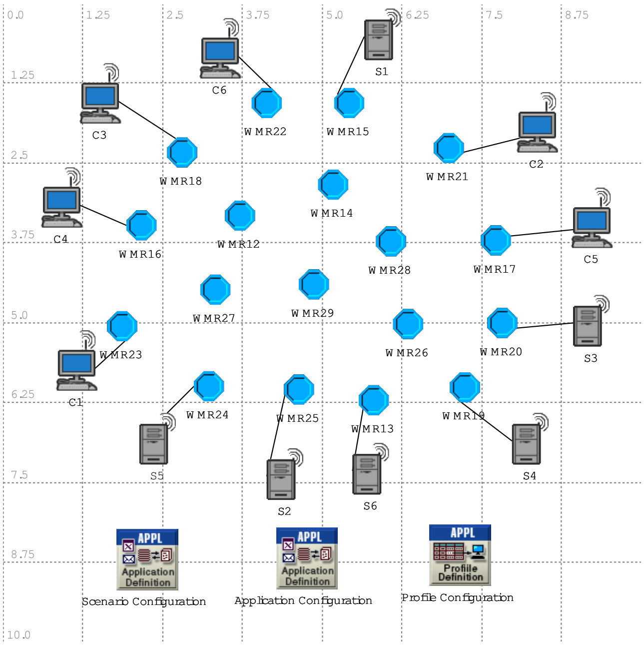

Figure 3: Example of the standard scenario used for the simulation campaign. Eighteen wireless mesh routers consist of the backhaul network, where six client/server pairs connect to.

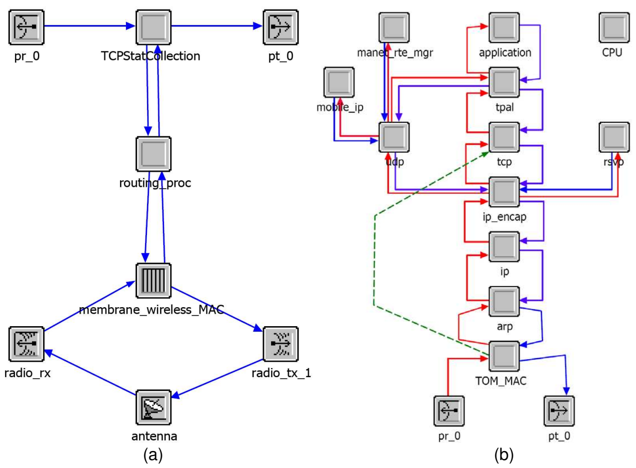

Figure 4: (a) Protocol layer models for wireless mesh routers, (b) The client and server models.

tion and coding scheme. Furthermore, in order to reduce the interference to/from adjacent concurrent transmissions and increase the frequency reuse and channel capacity, the nodes are equipped with directional antennas. The client and server models are shown in Figure 4(b) where all the protocol layers have been implemented following the OSI protocol stack including our own algorithms, improvements and modifications. The network configuration parameters in our simulation environment are summarized in Table 2.

The "Scenario Configuration" process (in Figure 3) configures the global parameters of the simulation (such as the type of cross layer approach used by TCP) and provides perfect TDMA synchronization among all backhaul wireless mesh routers. The "Application Configuration" and "Profile Configuration" processes define the application profiles,

Table 4: Overall network performances on different scenarios

	Scenarios	Goodput	PER	delay
1	1 flow 3 hops	10.0 Mbps	$< 10^{-5}$	≈ 0.01 s
2	1 flow 5 hops	3.5 Mbps	$10^{-2} - 10^{-3}$	≈ 0.02 s
3	3 flows 3 hops	7.5 Mbps	10^{-5}	≈ 0.01 s
4	3 flows 5 hops	3.0 Mbps	$10^{-2} - 10^{-3}$	≈ 0.03 s
5	6 flows 3 hops	4.0 Mbps	10^{-5}	≈ 0.01 s
6	6 flows 5 hops	1.4 Mbps	10^{-2}	≈ 0.1 s

such as FTP and VoIP (Table 3 indicates the most important parameters), application usage patterns (like how often the application is used, the usage during each session, the number of users and the usage fluctuations etc) and various QoS constraints (e.g. ETE packet delay, throughput and PER).

Since the improvements on the network performance of the novel joint routing/scheduling scheme have been already demonstrated in our previous works [15, 16, 24, 23], the results in this paper are manly focus on the impact of the TCP protocols on the proposed lower layer schemes. In order to evaluate the performance of the traditional TCP while it is placed on top of the modified lower layers, we implement different scenarios on variable number of concurrent TCP flows with different average number of hops between clients and servers. Table 4 shows a synthesis of the obtained results for each scenario: the achievable single connection goodput, the average PER and packet delay. Later, the proposed cross-layer TCP protocols "New Reno+ECN", "New Reno+ELN", and "New Reno+ERN" are compared with other conventional schemes "Reno", "New Reno", and "Jersey", in terms of average per connection goodput, end-to-end delay and jitter. Goodput is the application throughput, i.e., the amount of useful bits per time unit successfully forwarded by the network from a certain source to a certain destination. Jitter is defined as the variance of the inter-arrival time between two consecutive packets. Its importance relies on the performance assessment for real-time traffics like VoIP, since higher values indicates poorer voice quality. Consider two consecutive packets if they leave the source node with time stamps $t1$ and $t2$, and are played out at the destination node at time $t3$ and $t4$ respectively. Therefore, we have, Jitter $= |(t4 - t3) - (t2 - t1)|$.

It is shown in Table 4 that if we increase the number of active flows in the network, we may potentially generate more interferences from time slot to time slot and meanwhile limited network resources must be shared, thus worsen overall achievable performances for any single connection in terms of goodput and average packet delay. On the other hand, if we increase the number of hops between any client/server pair, longer packet delay is expected not only because of longer transmission range, but also due to increased self-interferences inside the flow, more TCP retransmissions should be expected for increased PER. Overall, we may conclude that although both increasing number of hops and concurrent TCP flows will cut down the single flow achievable goodput, the self-interferences within the flows has more severe effect than the cross-interferences among adjacent routes.

Table 2: Network configuration parameters

Parameter	Value	Parameter	Value
Channel Model	Rayleigh fading model	Path Loss Coefficient	3.5
Directional Antenna Pattern	Side lobe: -25dB Main lobe: 30°	Adaptive Modulation and Coding Schemes	BPSK-1/2, QPSK, 16QAM 64QAM, 128QAM
Doppler Frequency	25Hz	System Bandwidth	50MHz
Slot Duration	$80\mu s$	Slots per Frame	100
Frame Duration	8ms	MAC Packet Length	1024 bytes
Number of WMR	18	Number of client/server pair	6
Network Size	10 km × 10 km square	Transmission Range	2 km
TCP Maximum Trans. Unit	816 bytes	Queue Length	100 packets
TCP Maximum Segment Size	776 bytes	Traffic Patterns	FTP and VoIP

Table 3: FTP and VoIP application profiles

FTP Parameter	FTP Value	VoIP Parameter	VoIP Value
Inter-Request Time	Poisson distribution with $\lambda = 2s$	Encoder scheme	G.729A
File Size	Constant 1 MBytes	Voice frame per packet	1
Type of Service	Best Effort	Type of Service	Interactive Voice
Start time	Constant 0.2s	Start time	Constant - 0s
Duration	End of profile	Duration	10s
Repeatability	Once at start time	Repeatability	Unlimited

Figure 5 demonstrates average connection goodput with respect to average PER for the case of six concurrent flows in the network, each of which has average five hops between any client and server pair. It shows that TCP New Reno with ERN gives the best overall performance due to the precise bandwidth estimation in the intermediate backhaul mesh node. From transport layer perspective, TCP New Reno is able to recover efficiently and fast from network congestion due to buffer overflow by using fast recovery and fast retransmission mechanisms. On the other hand, if only New Reno is use, it shows similar performance to Jersey, but slightly lower due to the fact that it does not have the bandwidth estimation algorithm. New Reno with ECN and Reno perform worse than New Reno and Jersey. This is because New Reno performs aggressively even if with ECN since it is unable to completely avoid congestion and the consequent bandwidth reduction; Reno, however, does not perform well because it does not have the fast recovery mechanism. It is interesting to see that when the PER values varies between 10^{-2} and 10^{-1}, the New Reno with ELN becomes the best performer, because of the increased number of losses due to bit errors. In this case, New Reno with ELN can exploit its mechanism of requesting the retransmission of erroneous packets only. Furthermore, at high PER, the congestion events are less frequent and the advantage of bandwidth estimation is reduced.

Figure 6a shows the end-to-end delay distributions of six concurrent clinet/server pair coexisting in the network but configured with different traffic patterns. We notice that 53% of the overall ETE delay distributed below 100ms if only VoIP connections are present, but it increases to 500ms if both FTP and VoIP traffics are present. This is achieved by multi-constrained QoS routing algorithms that tries to prioritize/give more resources to delay-sensitive voice traffics, but as for FTP, although ETE delay is not a big concern, our proposed cross-layer design paradigm could still achieve relatively lower delay all below 3s.

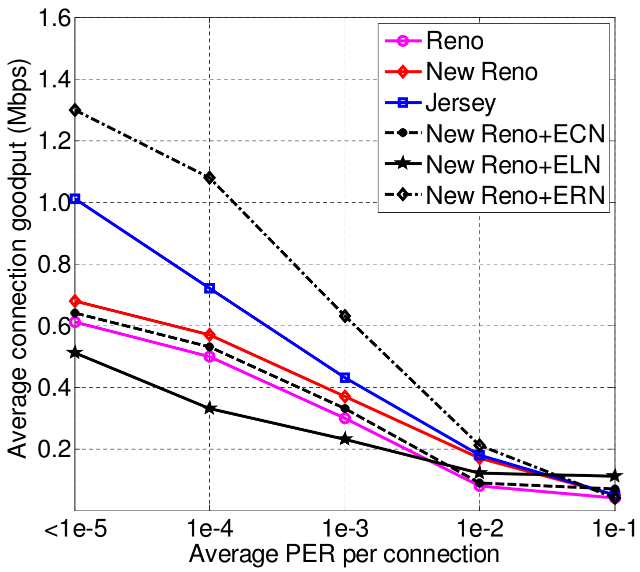

Figure 5: Average connection goodput vs. average PER in case of six TCP flows and average five hops for each connection.

The distribution of jitter values in case of only voice traffic and both voice and data traffics are shown in Figure 6b. We notice that it does not experience significant change with the introduction of the data traffic, i.e., only 2.98% of the overall jitter exceed jitter constraints 200ms for voice traffic, but seen a slight increase to 3.27% when both traffics are present. This shows that the QoS routing algorithm is working effectively in the network layer so that voice jitter constrains remain satisfied even though data traffics attempt to share the bandwidth resources.

Finally, we compare the MAC layer retransmission of packets with errors with the end-to-end solution we have proposed at TCP layer (Explicit Loss Notification). In this case, we consider a simulation scenario with six concurrent flows and an average of three hops between sources and destinations. We compare the maximum achieved goodput in case of ELN enabled or MAC layer retransmission enabled. In every of the above cases we have selected the TCP New Reno as congestion control algorithm. We found that there is a reduction of the goodput from 3.45Mbps for "New Reno+ELN" to 2.15Mbps in case of the use of MAC layer retransmission. This happens because MAC layer retransmission increases end-to-end delay, in case of erroneous MAC packets. Therefore, TCP timeout may expire and recovery procedure starts.

8. CONCLUSIONS

In this paper we propose and demonstrate the performance of a complete cross-layer architecture solution for wireless backhaul mesh networks. Several novel algorithms spanning all layers from PHY to TCP have been implemented in an integrated OPNET simulation platform. Previous simulation results have shown that distributed scheduler takes advantage of multi-user diversity gain, while the routing algorithm uses MAC layer statistics to guarantee multi-constrain QoS requirements such as delay, throughput and PER. In this work the impact of various TCP schemes and enhancements (namely, explicit congestion notification, explicit loss notification, and explicit rate notification in conjunction with traditional TCP protocol New Reno) on the overall network performance has been evaluated. Simulation results show that the optimized TCP protocol for our architecture turns out to be TCP New Reno and ERN when average end-to-end PER is relatively low, i.e., in good channel conditions, because it helps end-point users to control the sending rate to avoid possible congestions and packet loss due to buffer overflow or variable link throughput. Nevertheless, TCP New Reno and ELN scheme outperforms all other standard techniques when the channel conditions become poor because it successfully distinguish the link failure from network congestion in the wireless environment. Overall, our simulation results demonstrate that the proposed layer 1, 2, 3, and 4 techniques could guarantee QoS to real-time traffic like VoIP with various delay and jitter constraints.

9. ACKNOWLEDGMENTS

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10. REFERENCES

[1] OPNET Inc., http://www.opnet.com/.

[2] The addition of explicit congestion notification (ECN) to IP, RFC 3168.

[3] The newreno modification to TCP's fast recovery algorithm, RFC 2582.

[4] Transmission control protocol, RFC 793.

[5] Window and acknowledgement strategy in TCP, RFC 813.

[6] I. F. Akyildiz and X. Wang. A survey on wireless mesh networks. *IEEE Communications Magazine*, 43(9):S23–S30, Sept. 2005.

[7] E. Amir, H. Balakrishnan, S. Seshan, and R. Katz. Efficient tcp over networks with wireless links. In *Proceedings of Fifth Workshop on Hot Topics in Operating Systems*, pages 4–5, May 1995.

[8] G. Buchholcz, T. Ziegler, and T. V. Do. Tcp-eln: On the protocol aspects and performance of explicit loss notification for TCP over wireless networks. In *Proceedings of First International Conference on Wireless Internet 2005*, 10-14 July 2005.

[9] A. Capone, L. Fratta, and F. Martignon. Bandwidth estimation schemes for TCP over wireless networks. *IEEE Transactions on Mobile Computing*, 3(2), April-June 2004.

[10] L. Chen, S. H. Low, J. C. Doyle, and M. Chiang. Cross-layer congestion control, routing and scheduling design in ad hoc wireless networks. In *Proceedings of IEEE INFOCOM 2006*, Barcelona, Spain.

[11] R. Draves, J. Padhye, and B. Zill. Comparisons of routing metrics for static multi-hop wireless networks. In *Proceedings of ACM Annual Conf. Special Interest Group on Data Communication (SIGCOMM)*, pages 133–144, Aug. 2004.

[12] E. Felemban, C. G. Lee, R. Boder, and S. Vural. Probabilistic qos guarantee in reliability and timeliness domains in wireless sensor networks. In *Proceedings of IEEE INFOCOM 2005*, Mar. 2005.

[13] Z. Fu, H. Luo, P. Zerfos, S. Lu, L. Zhang, and M. Gerla. The impact of multi-hop wireless channel on TCP performance. *IEEE Transactions on Mobile Computing*, 4(2), March-April 2005.

[14] M. R. Garey. *Computers and intractability: a guide to the theory of NP-completeness*. W. H. Freeman, 1979.

[15] Y. Hou and K. K. Leung. Framework of opportunistic allocation of wireless resources. In *Proceedings of IEEE PacRim 2007*, Victoria, B.C., Canada, Aug. 2007.

[16] Y. Hou and K. K. Leung. A novel distributed scheduling algorithm for mesh networks. In *Proceedings of IEEE Globecom 2007*, U.S.A., 2007.

[17] J. M. Jaffe. Algorithms for finding paths with multiple constraints. *IEEE Networks*, 14:95–116, Apr. 1984.

[18] K. Jain. Impact of interference on multi-hop wireless network performance. *Wireless networks*, 11(4):471–487, 2005.

[19] T. Korkmaz and M. Krunz. Bandwidth-delay constrained path selection under inaccurate state information. *IEEE/ACM Trans. on Networking*, 11(3):384–398, Jun. 2003.

[20] S. Li and N.Ansari. TCP-jersey over high speed downlink packet access. In *Proceedings of IEEE Global*

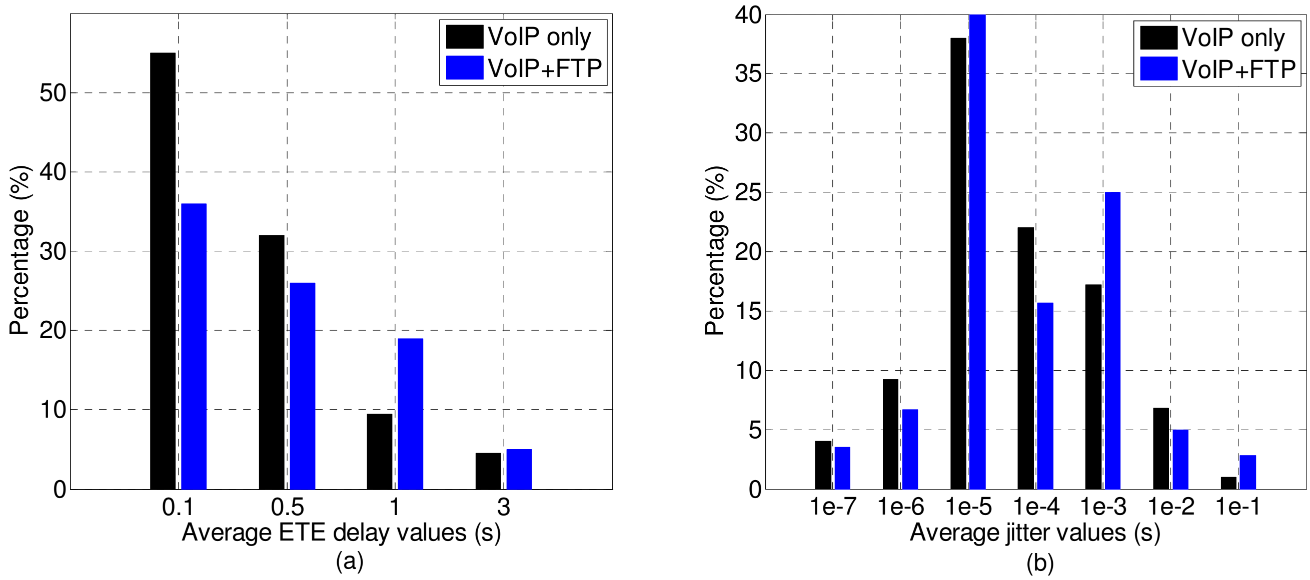

Figure 6: Simulation results on: (a) distribution of the voice ETE delay values (b) distribution of jitter values, both are shown in case of only voice traffic and both voice and data traffic are present in the network.

Telecommunications Conference, 2005, volume 6, Nov 28-Dec 2 1995.

[21] C. R. Lin. On-demand QoS routing in multihop mobile networks. In *Proceedings of IEEE INFOCOM 2001*, volume 3, pages 1735–1744, Apr. 2001.

[22] C. R. Lin and J. Liu. QoS routing in ad hoc wireless networks. *IEEE JSAC*, 17(8):1426–1438, 1999.

[23] C. H. Liu, A. Gkelias, and K. K. Leung. A cross-layer framework of QoS routing and distributed scheduling for mesh networks. In *Proceedings of IEEE VTC 2008 Spring*, Singapore.

[24] C. H. Liu, K. K. Leung, and A. Gkelias. A novel cross-layer QoS routing algorithm for wireless mesh network. In *Proceedings of IEEE ICOIN'08*, Busan, Korea, 2008.

[25] L. Lovasz. *Matching theory*. North-Holland, 1986.

[26] P. V. Mieghem and F. A. Kuipers. On the complexity of qos routing. *Computer Communications*, 26(4):376–387, Mar. 2003.

[27] C. Perkins and E. Royer. Ad-hoc on-demand distance vector routing. In *Proceedings of WMCSA'99*, pages 90–100, San Jose, CA, USA, 1999.

[28] S. Ramanathan. Scheduling algorithms for multihop radio networks. *IEEE/ACM Trans. on Networking*, 1(2):166–177, 1993.

[29] B. Sklar. Rayleigh fading channels in mobile digital communication systems .I. Characterization. *IEEE Communications Magazine*, 35(7):90–100, Jul 1997.

[30] Y. Tian, K. Xu, and N. Ansari. TCP in wireless environments: problems and solutions. *IEEE Communications Magazine*, 43(3), March 2005.

[31] Z. Wang and J. Crowcroft. Quality-of-service routing for supporting multimedia applications. *IEEE JSAC*, 14(7):1228–1234, Sept. 1996.

[32] K. Xu, Y. Tian, and N. Ansari. TCP-jersey for wireless ip communications. *IEEE Journal on Selected Areas in Communications*, 22(4), May 2004.

[33] S. Xu, T. Saadawi, and M. Lee. On tcp over wireless multi-hop networks. In *Proceedings of IEEE Military Communications Conference, MILCOM 2001*, volume 1, pages 28–31, Oct. 2001.

[34] X. Yuan. Heuristic algorithms for multiconstrained quality-of-service routing. *IEEE/ACM Trans. on Networking*, 10(2):244–256, Apr. 2002.

[35] B. Zhang and M. Shirazi. Implementation of explicit wireless loss notification using mac-layer information. In *Proceedings of IEEE Wireless Communications and Networking, WCNC 2003*, volume 2, 16-20 March 2003.

[36] Q. Zhang and Y.-Q. Zhang. Cross-layer design for QoS support in multihop wireless networks. *The Proceedings of the IEEE*, 96(1):64–76, Jan. 2008.

[37] Y. Zhang and T. Gulliver. Quality of service for ad hoc on-demand distance vector routing. In *Proceedings of WiMob'2005*, volume 3, pages 192–196, Aug. 2005.