An Efficient Cross-Layer Simulation Architecture for 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. Such networks are expected to support various types of applications with different quality of service (QoS) requirements. In this paper, we provide a complete cross-layer solution for WMNs together with its OPNET simulation architecture that comprises client, server and mesh router models spanning all layers from physical to application. Different TCP algorithm and techniques are implemented on top of a novel joint multi-constrained QoS routing and opportunistic scheduling scheme and their interaction and performance is evaluated through extensive simulations.

1 Introduction

Wireless mesh networks (WMNs [3]) are expected to provide a quick, cheap but also efficient solution for wireless backhauling in both urban and even rural environments. These networks are composed of (usually) static wireless nodes/mesh routers (WMR) with ample energy supply that are able to operate, not only as conventional access points or Internet gateways, but more important, as wireless routers forming an ad hoc network able to relay packets from other nodes without direct access to their destinations.

The need for multi-constrained quality of service (QoS), the fundamental peculiarities of the wireless channel and the dynamic and distributed nature of the wireless mesh networks necessitate the design of new communication solutions [12]. Physical (PHY) layer intelligent antenna techniques can be used to mitigate the unpredictable and time-varying interferences among wireless links that alter the feasible network capacity regions. Medium access control (MAC) layer efficient resource allocation schemes should be developed to exploit the multi-user diversity gain of wireless channels. Moreover, novel routing algorithms must be designed that harmonically cooperate with lower layers to provide multi-constrained QoS to a wide range of applications along the entire routes of communication. On the other hand, Transmission Control Protocol (TCP) should be modified in a way that can acquire useful information of the network state to distinguish among congestion, packet loss or link failures due to buffer overflow, channel errors or unavailable data rate. The above operational characteristics in those four layers are primarily the reason why an accurate and efficient cross-layer WMNs simulation architecture and platform 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.

Despite the importance and necessity of a complete cross-layer architecture and simulation platform for wireless mesh networks, the research outputs towards this direction and commercially available products with configurable protocol layers are very limited. OPNET WiMax model [1] is one of them, but only provides a centralized scheduler with some PHY features. An enhanced NS-2 simulator [9] for WMNs have been also developed to provide more accurate Signal-to-Interference-Ratio (SINR) calculations in PHY and MAC layers, but still missing the rest of the protocol layers.

From an algorithmic point of view, a QoS routing algorithm for wireless networks has been recently proposed [11] that attempts to overcome the NP-complete difficulty of providing optimum routes. Scheduling algorithms for WMNs, both centralized [5] and distributed [6], have also drawn a lot of research attention to overcome NP-completeness [8]. Moreover, due to the characteristics of wireless links and traffic demands

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in WMNs, traditional congestion window based TCP protocols like Reno and New Reno are no longer efficient since they 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. To overcome this problem, several link layer and End-to-End (ETE) solutions have been proposed. The former group tries to improve the algorithms for detecting and correcting bit errors at layer 2. On the other hand, ETE solutions treat the network as a "black box" where the lower layers of the intermediate nodes report explicitly some information, such as data congestion [2], packet loss [4], or available bandwidth [7] to the TCP layer of the end nodes.

In this paper, we propose an novel cross-layer architecture for WMNs where each layer has a clear picture of other layers' behavior and exploits this knowledge to create synergies in the control functions. Moreover, a complex and realistic OPNET [1] simulation environment for WMNs has been developed. Several algorithms, spanning all layers from PHY to Application have been included and numerous configuration parameters have been defined in order to test the interaction of the proposed algorithms and evaluate their performance. Numerical results demonstrate a significant performance gain in terms of goodput, delay and PER.

The rest of the paper is organized as follows. Section 2 provides a brief description of our simulation architecture while the implemented TCP protocols and QoS routing algorithm are described in Section 3 and 4 respectively. The distributed scheduling algorithm is described in 5. The OPNET simulation parameters together with our simulation results are given in Section 6 and finally conclusions are drawn in Section 7.

2 OPNET Architecture

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 some of them are selected to have gateway functionalities. WMR model comprises of TCP, network, MAC and enhanced PHY layers with routing, scheduling, antenna model, fading channel, transmission model, and TCP supports for client and server models, as shown in Figure 1. Client and servers are attached to WMRs generating different application profiles like FTP, HTTP, Video and VoIP that have pre-specified QoS constraints, i.e., end-to-end (ETE) packet delay D_q^r, throughput T_q^r and PER E_q^r. 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.

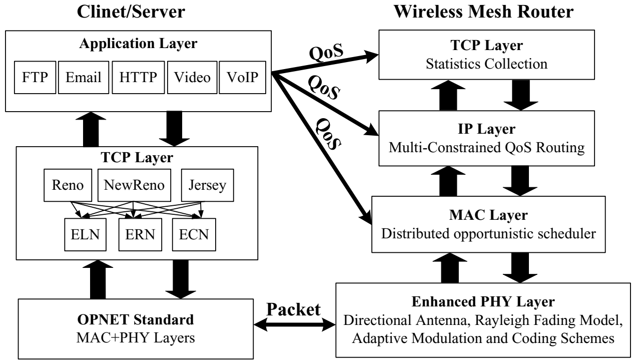


Figure 1. OPNET WMNs architecture.

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 fixed-size time slots, during which the channel remains the same if we assume block fading. Scheduling decisions are taken by all nodes 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), and Rayleigh fading model [10] 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. Power control is not considered in this phase, but 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.

3 Client/Server: TCP Protocols

Although TCP protocols has been proven successful in wired networks to provide ETE reliable communications and assure ordered delivery of packets by flow control and error control mechanisms, WMNs demand new and efficient TCP protocol design to distinguish congestion loss from frequent link failures and channel fluctuations. These are because advanced communications technologies like AMC, directional antenna beam-forming and distributed scheduler provide variable throughput due to instantaneous channel conditions. Explicit congestion notification (ECN), explicit loss notification (ELN), and explicit rate notification (ERN) with tradition TCP protocols like Reno, New Reno and Jersey are the ones we propose as follows.

3.1 ECN

If a transmitted packet reaches a backhaul mesh router with congestion, 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 2a. The TCP layer, in turn, sets the corresponding Explicit Notification Echo bit in the TCP header of the corresponding ACK packet. When the TCP sender receives the ACK packet with the set 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.

3.2 ELN

As shown in Figure 2b, this is introduced when packet loss happens due to link failure. Consider the link (v_i, v_j) that 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. When it 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, ELN bit, in the TCP segment, when it is set it means the sequence number in the "ACK 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 indicates the segment that must be retransmitted because of channel errors.

3.3 ERN

ERN is the cross-layer scheme proposed to tackle unexpected throughput fluctuations as shown in Figure 2c. In each backhaul mesh router, the MAC layer periodically informs the IP layer of the available bandwidths on the link. The relay packet records this value in the IP packet header before sending it to the next hop. The subsequent intermediate nodes along the path 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 ETE available bottleneck bandwidth for the whole TCP connection is recorded in the header in order to avoid network overload. Then, instead of using an estimated bandwidth value to calculate the congestion window as the bandwidth delay product, the TCP receiver can use the actual bottleneck bandwidth information from backhaul nodes' IP layer. The updated window field value is sent back to the sender that will react accordingly to reduce congestion window.

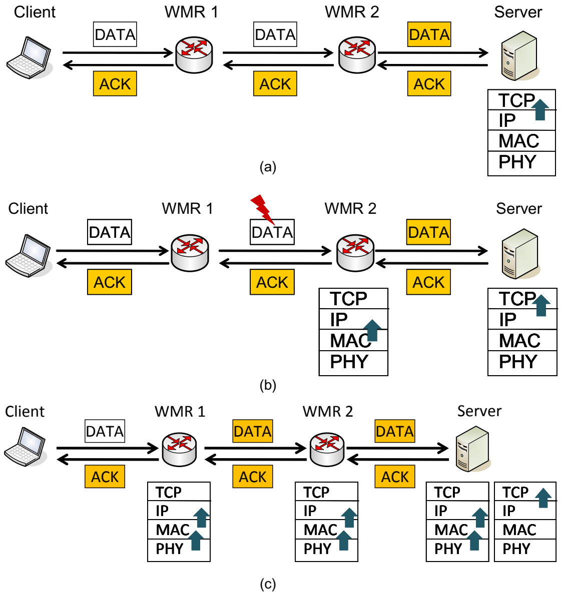


Figure 2. (a) ECN (b) ELN (c) ERN schemes.

4 WMR: QoS Routing Protocol

The problem of providing optimum routes that guarantee multiple QoS constraints has been proven to be NP-complete [11], and therefore, in order to overcome this difficulty we define a new utility function based on the "dissatisfaction ratio" **R** that experienced by each QoS metric. More specifically, we define the ratio **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.,

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

(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 throughput along route k, i.e.,

$$\mathbf{R}_k^T(q) = \frac{T_q^r}{\min_{(i,j)\in k} T_{ij}^a} \tag{2}$$

(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.,

$$\mathbf{R}_k^E(q) = \frac{1-\prod_{(i,j)\in k}(1-E_{ij}^a)}{E_q^r} \tag{3}$$

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, $\left(\mathbf{R}_k^D(q), \mathbf{R}_k^T(q), \mathbf{R}_k^E(q)\right) \leq 1$. Our multi-constrained QoS performance index for route k can be formulated as,

$$\mathbf{I}_k = \max\left[\mathbf{R}_k^D(q), \mathbf{R}_k^T(q), \mathbf{R}_k^E(q)\right] \tag{4}$$

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

$$\text{route } k^* \text{ is chosen } \Leftarrow \min_{\forall k\in\Omega_{st}} \mathbf{I}_k \tag{5}$$

where we assume a set of routes Ω_{st} have been found between a source s and a destination gateway t.

5 WMR: MAC Protocol

Traditional distributed opportunistic scheduler like using SINR as the scheduling utility comes with the inherent difficulty to guarantee the required QoS performance in a long run. This is because opportunistic approaches usually introduce more fluctuating instantaneous performances at individual incoming and outgoing links. In order to overcome this difficulty and enforce QoS, the distributed opportunistic proportional fair scheduler proposed in our cross-layer framework is described as follows.

The QoS routing algorithm used estimates the 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,

$$\mathbf{U}_{ij} = a_{ij}^q \frac{\rho_{ij}}{\varphi_{ij}} \tag{6}$$

where ρ_{ij} and φ_{ij} are the instantaneous throughput and channel capacity in the long run respectively. [6] proves that, by choosing the proposed link utility metric (6) the scheduler guarantees the proportional target QoS throughput as well as fairness among links.

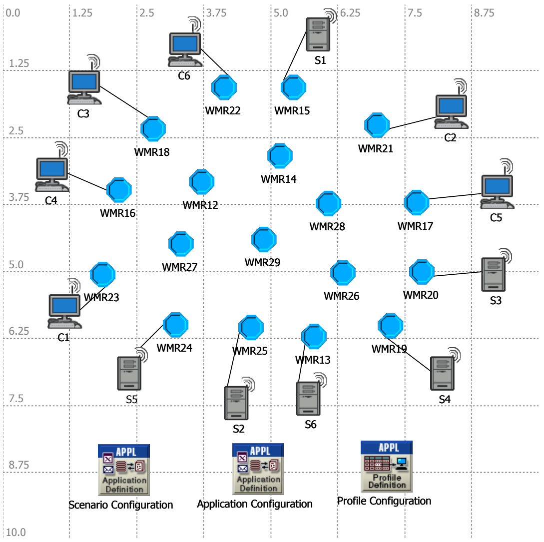


Figure 3. Simulation scenario example.

6 Numerical Results

We use OPNET [1] modeler to create an integrated WMNs simulation environment, where a number of WMRs are randomly and independently deployed on a two dimensional space consisting the backhaul network. A variable number of clients and servers are connected with the WMN through WMRs, as shown in Figure 3. The developed WMR models representing the functionalities of different layers are shown in Figure 4a. We enhance the simplified OPNET model of PHY layer with the Rayleigh fading channel model [10], and the required PER is derived based on SINR curves for the used AMC scheme. Furthermore, in order to reduce the interference to 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) with all the OSI protocol layers and the cross-layer improvements we have proposed. The network configuration parameters in our simulation environment are summarized in Table 1.

Some other processes have been added as seen in Figure 3. The “Scenario Configuration” configures the

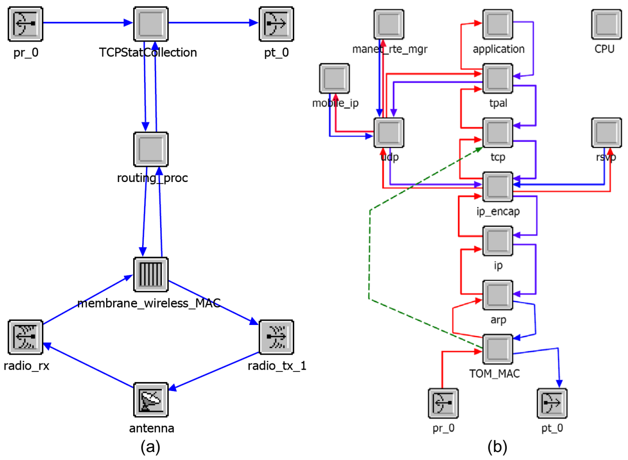


Figure 4. (a) Protocol layer models for WMRs, (b) The client and server models.

global simulation settings, such as the used TCP approach and TDMA synchronization among backhaul routers. Two processes, namely "Application Configuration" and "Profile Configuration", define the application profiles, such as FTP and VoIP. Both nodes allow us to select and configure multiple application models and usage patterns, such as the statistics of how often an application is used, its duration, the number of users etc.

The results in this paper mainly focus on the impact of the TCP protocols on the proposed lower layer schemes. We implement different scenarios on variable number of concurrent TCP flows with different average number of hops between clients and servers. Table 2 shows a synthesis of the obtained results for each scenario, i.e., the achievable single connection goodput, the average PER and packet delay. It is shown that if we increase the number of active flows in the network, we may potentially generate more interference, thus worsen the overall achievable performance in terms of goodput and average packet delay. On the other hand, if we increase the number of hops communication pairs, longer packet delay is expected, not only because of higher number of relays, 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. Overall network performances

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

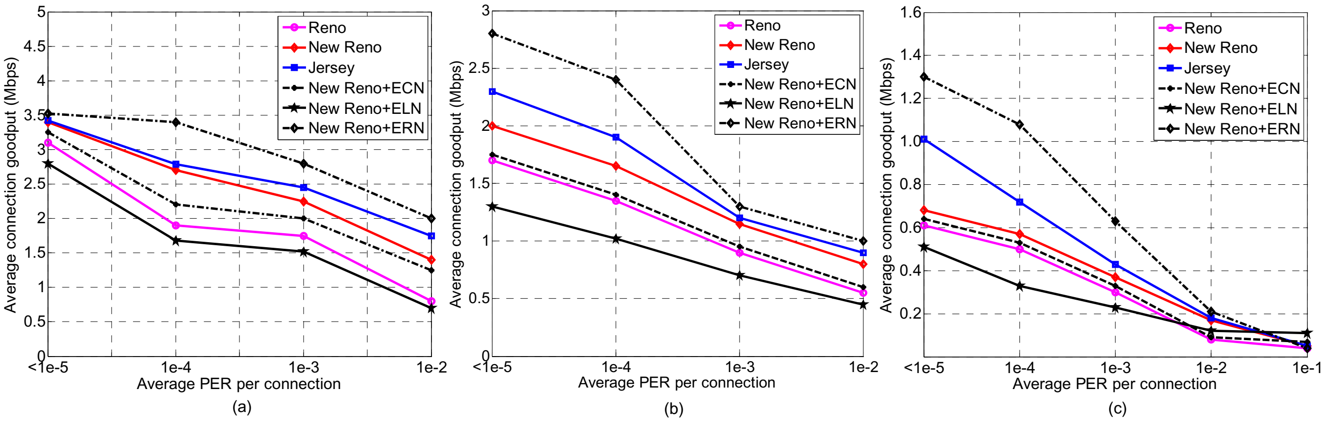
The cross-layer TCP protocols "New Reno+ECN", "New Reno+ELN" and "New Reno+ERN" are compared with some conventional schemes, such as, "Reno", "New Reno" and "Jersey" for the case of one/three/six concurrent flows in the network, each of them has an average of five hops between any client and server pair (Figure 5). It can be seen that "New Reno+ERN" always gives the best overall performance due to the precise bandwidth estimation in the intermediate backhaul mesh node with ERN scheme. Meanwhile, 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. Nevertheless, if only New Reno is used, it shows similar performance to Jersey, but slightly lower since it does not have the bandwidth estimation algorithm. New Reno with ECN and Reno perform worse than New Reno and Jersey, because New Reno acts aggressively (even if ECN is included) since it is unable to completely avoid congestion and consequent bandwidth reduction. Finally, Reno does not perform well because it does not have a fast recovery mechanism. It is interesting to see that when PER values vary between 10^{-2} and 10^{-1} in Figure 5c, "New Reno+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 more important.

7 Conclusions

In this paper we propose and demonstrate the performance of a complete simulation architecture for wireless mesh networks. Several novel algorithms spanning all layers from PHY to Application have been implemented in an integrated OPNET simulation platform. The impact of various TCP schemes and enhancements on the overall network performance has been evaluated. Results show that the optimized TCP protocol for our architecture turns out to be TCP New Reno and ERN when average ETE PER is relatively

Table 1. Network configuration parameters

Parameter	Value	Parameter	Value
Channel Model	Rayleigh fading model	Path Loss Coefficient	3.5
Directional Antenna Pattern	Main lobe: 30°, Side lobe: -25dB	AMC	MPSK, MQAM
Doppler Frequency	25Hz	System Bandwidth	50MHz
Slot Duration	$80\mu s$	Slots per Frame	100
Frame Duration	8ms	MAC Packet Length	1024 bytes
Network Size	10 km × 10 km square	Transmission Range	2 km
TCP Max. Trans. Unit	816 bytes	Traffic Patterns	FTP and VoIP



(a) (b) (c)

Figure 5. Average connection goodput vs. average PER in case of: (a) one TCP flow, (b) three TCP flows, (c) six TCP flows. All of them have average five hops for each connection.

low, i.e., in good channel conditions. This is because it helps the 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.

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