Mobile TFRC: a Congestion Control for WLANs

Lei Zhang, Patrick Sénac, Emmanuel Lochin
ISAE - LAAS/CNRS
Université de Toulouse
Toulouse, France
{firstname.lastname}@isae.fr

Michel Diaz
LAAS/CNRS
Université de Toulouse
Toulouse, France
diaz@laas.fr

Abstract

Based on an identification and evaluation of the subtle counterproductive interactions between the WLANs MAC layer and the transport layer, this paper shows a new approach towards congestion control for WLANs. We introduce a specialization of TFRC (MTFRC: Mobile TFRC), which is adapted to wireless access networks. This TFRC specialization requires only slight changes to the standard TFRC protocol. Simulation results show substantial improvements for applications over TFRC in scenarios where the bottleneck situates on the MAC layer of the mobile nodes.

1 Introduction

TFRC protocol (RFC 3448) has been proved to be able to offer a smooth, low delay and TCP-Friendly packet transmission in a wired network. To improve this mechanism over wireless networks, numerous research work aimed to find efficient differentiation algorithms (LDAs) to distinguish congestion errors from channel errors [2, 1]. However, to date, few studies have focused on the influence of the contention based mechanism CSMA/CA to the TFRC protocol. In this paper we show that the rate processed at transport layer can strongly diverge from the rate offered by the Wifi MAC layer. This discrepancy between these two layers induces losses due to MAC buffer overflow. Indeed, as given in Fig. 1, simulation and experimental results show that the throughput at the transport layer (UDP flows) can surpass the maximum bandwidth that the MAC layer can support (802.11a) and can lead to massive packet loss rate. In this paper, we argues that coordinating the transport sending rate with the rate delivered by the WLAN MAC layer can entail an important loss reduction thus lowering end to end delays and jitter variations. We introduce a new specialization of TFRC: Mobile TFRC (MTFRC), which efficiently adapts the sending rate from transport layer to the MAC layer. Furthermore, our method allows to solve the unfairness issues between upstream and downstream flows.

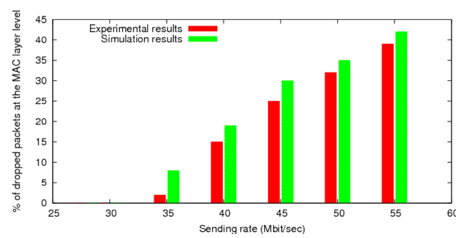

Figure 1. Percentage of packets lost on the MAC buffer as a function of the sending rate

The rest of this paper is organized as follows: in section 2, we give an analytical method to calculate the maximum throughput supported by the MAC layer for generic WLAN cases. We detail our proposed method MTFRC in section 3. We validate our analysis by means of simulation in section 4, and section 5 concludes this study.

2 Calculation of bandwidth capacity limited by MAC layer for generic 802.11b scenarios

In this section, we introduce an analytical model to calculate the maximum bandwidth capacity delivered to mobile nodes by the MAC layer (X_m) for generic 802.11b scenarios. The proposed model pushes further the approach proposed in [3] by considering on one side the diversity of mobile nodes' transmission rate profiles and on the other side, the specificities of TFRC flows. We suppose N mobile nodes in 802.11b coverage are classified into four groups $N_i (i = 1, 2, 3, 4)$ according to their transmission rates (11/5.5/2/1Mb/s). We denote that S is the MAC-layer frame length in bits; T_i^{tr} represents the duration to transmit a data frame at a certain transmission rate R_i. T_i^{ov} is a constant overhead which comprises DIFS, SIFS, two times of the PLCP preamble and the header transmission time as well as the MAC acknowledgment transmission time t^{ack}. We also denote T^{cont}, the average duration of backoff process; $P_c(N)$, the proportion of collisions experienced for each packet successfully acknowledged at the MAC level[3]. Then we have T_i the overall duration for sending one data frame for each node in the group i:

$$T_i = T_i^{tr} + T_i^{ov} + T^{cont}(N) \tag{1}$$

For the N mobile nodes, we define *greedy nodes*, whose throughputs can reach or surpass the maximum bandwidth that the MAC layer can support. Then, in contrast, we define *rate sparing nodes*, whose throughputs are limited by their application layer or by congestion over the network (i.g. a VoIP node that requires a relatively low bandwidth).For each group N_i, we also suppose that there are K_i rate sparing nodes among N_i mobile nodes using the transmission rate R_i. The average throughput of these K_i nodes limited by their application or the congestion in the network is $X_i^j (j = 1, 2, \ldots, K_i)$.

Each of the $(N - \sum_{i=1}^{4} K_i)$ greedy nodes fully uses the maximum throughput X_m delivered by the MAC layer. The aggregated bandwidth X between all the mobile nodes and the access point is given by:

$$X = \sum_{i=1}^{4} \sum_{j=1}^{K_j} X_i^j + (N - \sum_{i=1}^{4} K_i) * X_m + X_{AP} \tag{2}$$

We define $P_i^j = X_i^j / X (j = 1, 2, \ldots K_i)$, the proportion of throughput used by each rate sparing nodes in group i and $P_{AP} = \frac{X_{AP}}{X}$, the proportion of the aggregated throughput used by the AP.

The proportion of the throughput for each of the $(N - \sum_{i=1}^{4} K_i)$ greedy mobile nodes is:

$$P_b = X_m / X \tag{3}$$

CSMA/CA protocol allows all the greedy mobile nodes to share fairly the radio channel. Theoretically, the average time T between the two successive emission packets sent by greedy nodes comprises the following 4 parts: (1) the time required for sending one packet by each of the greedy node with different transmission rate: $\sum_{i=1}^{4} T_i * (N_i - K_i)$; (2) the time required for sending packets by the sparing nodes with different transmission rate. According to the above analysis on the rate proportion, every time a packet is sent out by a greedy node, there should be $(\sum_{i=1}^{4} \sum_{n=1}^{K_i} P_i^n)/P_b$ packets sent by all the rate sparing nodes; (3) the time required for sending packets (i.e. TFRC feedback packets or downloading data) from the AP to N mobile nodes. Similarly, every time a packet is sent out by a greedy node, there should be (P_{AP}/P_b) packets sent by AP; (4) the time spent in collisions (T_{col}).

$$T = \sum_{i=1}^{4} T_i * (N_i - K_i) + \frac{\sum_{i=1}^{4} \sum_{n=1}^{K_i} P_i^n * T_i}{P_b} + T_{col} + T_{AP} * \frac{P_{AP}}{P_b} \tag{4}$$

T_{AP} and T_i can be calculated with equation (1), and we have a total of $(N + 1)$ contention mobile node.

$$T_{col} = P_c(N + 1) * t_{jam} * (N + 1) \tag{5}$$

t_{jam} represents the average time spent in collision for each node in case of collision.

Since the average time between the two successive emission packets is T, we can calculate X_m, the maximum throughput supported by the MAC layer for greedy nodes with the following equation:

$$X_m = S/T \tag{6}$$

With S the length of MAC layer packet in bits.

The maximum available throughput at the transport layer is:

$$X_t = S_t/T \tag{7}$$

With S_t the length of transport layer packet in bits.

Our analytical model has been validated by a set of simulations under OPNET. Figure 2 shows the evolution of X_t (with analytical and simulation results) in UDP case in terms of number of the greedy uploading mobile nodes ($N = [4, 30]$) in four different scenarios. In the first scenario, all the mobile nodes have a transmission rate of $11 Mbit/s$. Then, for each other three scenarios, one among N mobile nodes has respectively a transmission rate of $5.5 Mbit/s$, $2 Mbit/s$ and $1 Mbit/s$.

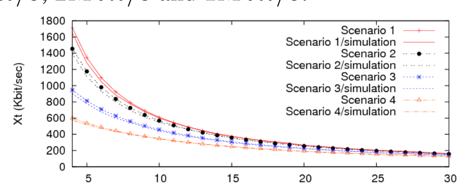

Figure 2. Evolution of X_t in UDP mode as a function of the number of uploading nodes

3 Cross-layered Congestion Control

3.1 Rate adaptation

In section 2, we have given formulas to find the maximum throughput supported by contention based MAC layer for the mobile nodes. When the sending rate from transport layer (eg. estimated by TFRC equation) becomes higher than the bandwidth offered by the MAC layer, packets can be lost in the MAC buffers. These losses increase the loss event rate p processed by the TFRC protocol and degrade the TFRC sending rate. However, following a phase without losses, the TFRC sending rate will increase until it exceeds the available MAC layer rate again, thus inducing harmful variations and unstable oscillations of the sending rate. In order to illustrate this behavior, we simulate two mobile nodes uploading data to remote servers with a transmitting rate of $5.5Mb/s$ where congestion occur at the MAC layer. Fig 3 gives the result of the TFRC throughput and the maximum available throughput at the transport layer (X_t). We can observe that TFRC obtains unstable rate variations around X_t and that MAC buffer overflow occurs. The transport layer throughput has a standard deviation of $115.8Kb/s$ for an average throughput of $1.94Mb/s$ after $t = 25sec$.

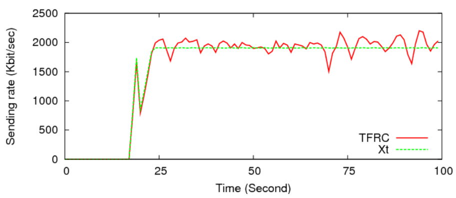

Figure 3. Performance comparison between TFRC and X_t

In order to improve the QoS delivered to TFRC flows on WLAN access networks, we introduce a specialization of the TFRC protocol (MTFRC) to WLANs based on a cross-layer interaction between the transport and the mac layers. More precisely, we propose to constraint the TFRC rate equation with the MAC layer available rate processed as defined in the previous section. The algorithm for processing the MAC limited threshold X_t is inserted in every access point AP, this threshold can be calculated according to different dynamical parameters collected by AP in real time. Sending and receiving rate profiles are also taken into account in the case when rate sparing nodes exist. Every mobile node then compares its processed TFRC equation based sending rate X_{tfrc} with the threshold X_t. If the calculated X_{tfrc} is higher than X_t, the sending rate X_{send} is then adjusted to X_t to avoid congestion and losses in the MAC layer as follows:

$$X_{send} = min(X_{tfrc}, X_t) \tag{8}$$

3.2 Fairness improvement

Since the access point, AP, is considered as a normal contention-based mobile node, it has the same opportunity of sending packets (to all the download mobile nodes) as any of the other upload mobile nodes. Indeed, if we suppose there are U TFRC uploading nodes and D TFRC downloading nodes in the AP coverage, the average throughput of each upload flow is equal to the aggregate throughput of the download flows sent by AP. We introduce in this section a rate-equalization mechanism that makes it possible for the downloading flows to gain a fair share of the throughput delivered by the MAC layer.

Following the analysis in section 2, since AP is considered as a normal transmission mobile node, the average uploading bandwidth supported at MAC layer X_u can be estimated from equation (6) (where $N = U + 1$). Note that X_u is equal to $X_m = D * X_d$ where X_d is the average bandwidth for each downloading flows. The aggregated bandwidth (X) exchanging between the AP and all the mobile nodes is given by:

$$X = U * X_u + D * X_d = (U + 1) * X_m \tag{9}$$

The object of the proposed rate equalization mechanism is to assign this total bandwidth X more fairly to each of the mobile node (download or upload mobile nodes). So each mobile node can get a bandwidth of

$$X_{fair} = \frac{X}{(U + D)} = \frac{X_m * (U + 1)}{(U + D)} \tag{10}$$

Therefore, by combining both the fair share constraint and the MAC rate constraint in equation (8), we obtain the constrained sending rate:

$$X_{send} = min(X_{tfrc}, X_t, X_{fair}) \tag{11}$$

This specialization of TFRC limits the sending rate of each of the U upload mobile node to X_{fair}. Therefore, the contention based mechanism allows AP to gain more sending opportunities, which corresponds to an additional bandwidth of ($X - U * X_{fair}$) for the AP. Thus, each download node can get a bandwidth of X_{fair}.

4 Simulation and validation

We have simulated under OPNET a set of wireless scenarios to validate our proposed method. We present in this section two typical scenarios. We set the link bandwidth capacity of the access router to $C = 10Mb/s$ in order to have $C >> \sum_1^N X_m$ with N the number of mobile nodes. As a result, X_m is considered as the bottleneck between the mobile node and the destination.

We suppose that several TFRC mobile nodes send data packets to their corresponding servers via an 802.11b access router. The buffer size in the access point is $20KByte$ and the buffer size of MAC layer of each mobile node is $256Kbit$ (default setting in OPNET). The size of data frame

(S_t) is equal to $8192bit$ (MPDU size $S = 8614bit$). In all the simulations, the traffic generation starts at $t = 15sec$.

4.1 Scenario1

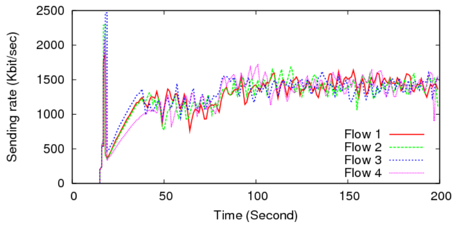

Figure 4. TFRC sending rate

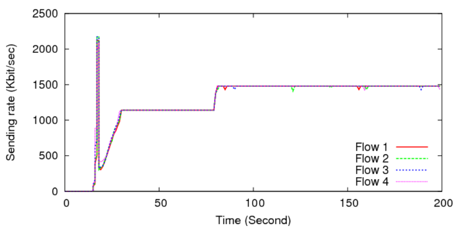

Figure 5. MTFRC sending rate

In scenario 1, we suppose that two of the mobile nodes always share a transmission rate of $11Mb/s$. The other two mobile nodes have a transmission rate of $5.5Mb/s$ between $t = [15sec, 80sec]$. When they move towards the access router, their transmission rates turn to $11Mb/s$ at $t = 80sec$. Between $t = [15sec, 80sec]$, according to equation (7) (with $S_t = 1024Byte$, $N_1 = 2, N_2 = 2, N = 4$), X_t is equal to $1.14Mb/s$ between $t = [80sec, 200sec]$ and X_t rises to $1.48Mb/s$ with $N_1 = 4$. In this scenario, the sending rate X_{send} always equals to X_t because the bottleneck always situates on the MAC layer of each mobile node. Fig. 4 and 5 represents the sending rate of TFRC and MTFRC. We can see that our proposal efficiently avoids the losses on the MAC layer and substantially improves the quality of the transmission.

4.2 Scenario 2

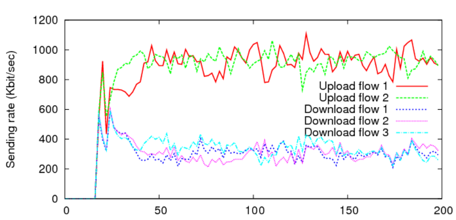

Figure 6. TFRC sending rate

In scenario 2, we address fairness issues between upload and download flows. For illustration purpose, we consider that two mobile nodes upload TFRC flows (of which the transmission rates are respectively 11Mb/s and 2 Mb/s) and three mobile nodes download TFRC flows with transmission rate of 5.5 Mb/s. Fig. 6 shows that by default, each upload flow occupies much more bandwidth (average ratio of three) than each download flow. Conversely, when using MTFRC improved with the proposed fairness mechanism, we observe a fair share of the bandwidth between the upload and download flows (Fig. 7). Indeed, in this case, when applying equation (7) (10) and (11), since the AP is considered as a upload node we have $N = 3, N_1 = N_3 = 1, N_2 = 1$ (which corresponds to the aggregated 3 download nodes) and we get $X_{up} = X_m = 968Kb/s$ and $X_{fair} = X_{up} * 3/5 = 581Kb/s$. The sending rate for each of the upload mobile nodes is then limited to X_{fair} to allow download nodes sharing the same bandwidth.

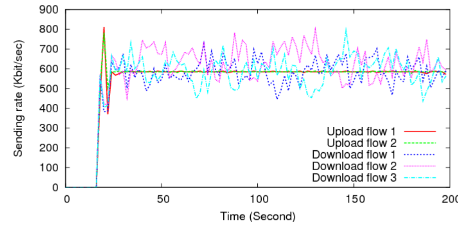

Figure 7. Sending rate with MTFRC

5 Conclusion

In this paper, we introduce an analytical model of calculating dynamically the available throughput supported by MAC layer. The result is given as a parameter to the transport layer mechanisms (e.g. TFRC rate calculation) in order to suppress losses in MAC buffers, therefore improves significantly the transmission quality. Moreover, we pushed forward the idea of rate adjustment to improve the fairness between uploading and downloading flows. Our future work aims to determine a light cost and efficient information collection method involved in the base station. Furthermore, we will investigate the potential impacts of the proposed approach in the context of handover managements.

References

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