A Network Coding Scheme to Improve Throughput for IEEE 802.11 WLAN

Kaikai Chi, Yongchao Wu, Yi-hua Zhu,
School of Computer Science and Technology
Zhejiang University of Technology
Hangzhou, Zhejiang, China 310023

Victor C.M. Leung
Department of Electrical and Computer Engineering
The University of British Columbia
Vancouver, BC, Canada V6T 1Z4

***Abstract*—IEEE 802.11 infrastructure wireless local area network (WLAN) is increasingly popular, in which access points (APs) are applied. In a WLAN with an AP connected to the Internet, the communication between any two nodes is relayed by the AP, i.e., the AP serves all the nodes in the WLAN, which degrades throughput. In this paper, we propose a novel network coding scheme called MPOF that is able to encode multiple packets from different data flows and take data rates of links into account so that throughput is improved.**

Keywords- WLAN; network coding; throughput; IEEE 802.11 standard

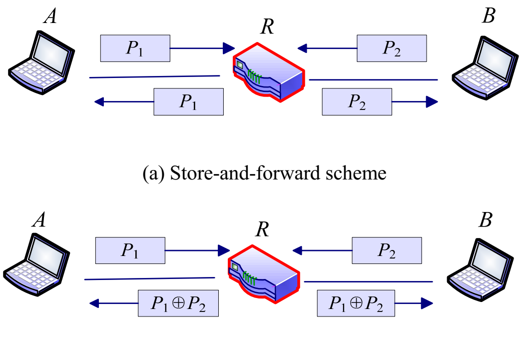
I. Introduction

IEEE 802.11 based wireless local area networks (WLANs) have been widely used with the popularity of portable computing devices such as laptop, smart phone, etc. since their network interfaces use the license-free industrial, scientific, and medical (ISM) radio bands. There are two types of WLANs: infrastructure WLAN and ad hoc WLAN. The main difference between them is in that the former uses access point (AP) whereas the latter does not. Most Wi-Fi hotspots adopt infrastructure WLANs.

In an infrastructure WLAN, AP takes part in all communications among nodes, i.e., the communication between any pair of nodes is relayed by the AP. In other words, the communication involves two hops, which degrades throughput. For example, in the case that a pair of nodes exchanges two data files, the files have to go through the AP one after another. That is, the files cannot be transmitted simultaneously.

To improve throughput of the infrastructure WLAN, we apply network coding, which was proposed by Ahlswede et al. [1] in 2000 and has been proved beneficial in enhancing throughput in the fields of communications and computer network. With network coding, a node in the WLAN can encode multiple received packets to form a single encoded packet and then the generated packet is broadcast to the next-hop nodes, which reduces the number of transmissions so that throughput is improved. It is shown that network throughput is greatly improved by network coding [2][3][4][5].

In [4], Katti et al. proposed the COPE protocol, which is the first practical network coding system whose performance was evaluated in a real wireless test-bed. The main idea of the COPE is illustrated in Fig.1, in which the communication between the nodes A and B needs the relay of node R. Clearly, in the case when nodes A and B exchange packets P_1 and P_2, totally 4 transmissions (i.e., packet P_1 is transmitted to R and then to B, which causes 2 transmission, and packet P_2 goes in the opposite direction and also causes 2 transmissions) are needed when the traditional store-and-forward scheme is applied (see Fig. 1(a)). However, the total number of transmissions can be reduced to 3 when using network coding as shown in Fig.1(b), where the relay node R generates the encoded packet $P_1 \oplus P_2$ after the two packets P_1 and P_2 are received and then broadcast the encoded packet to both nodes A and B. Here, the notation $\oplus$ stands for the network coding operation (e.g., the XOR operation). Upon receiving the encoded packet, node A can recover packet P_2 via the decoding that performs $P_1 \oplus (P_1 \oplus P_2)$, while B can decode P_1 by performing $P_2 \oplus (P_1 \oplus P_2)$. In a word, compared to the traditional scheme, the network coding scheme saves 1 transmission, thus improving throughput. In fact, the throughput can be further improved when R encodes more packets.



(a) Store-and-forward scheme

(b) Network coding scheme

Fig.1. Nodes A and B exchange packets P_1 and P_2

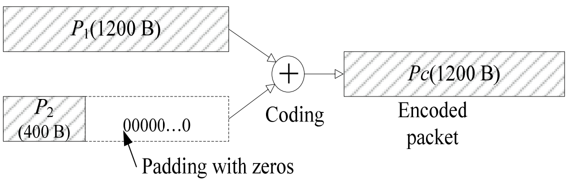
Hitherto, a lot of research on wireless network coding has been conducted to design more efficient coding schemes in order to improve the throughput. Zhao et al. [6] found that the key to COPE's high performance lies in the interaction between COPE and the MAC protocol after analyzing the performance of COPE, and they proposed a round-robin scheduling scheme, which leads to significant gain in throughput. Fang et al. [7] compared the analytical and

QSHINE 2014, August 18-20, RHODES, Greece

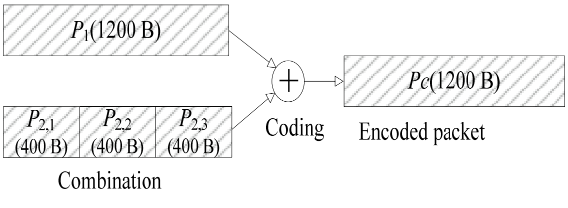
DOI 10.4108/icst.qshine.2014.256194

experimental performance of COPE-style network coding in IEEE 802.11 ad-hoc networks and showed that the gap between the theoretical and practical gains is due to the different channel qualities of sending nodes. Dong et al. [8] proposed a loop coding scheme to efficiently reduce packet loss rate so as to improve TCP throughput. In order to achieve a fine tradeoff between throughput and overhead, Nage et al. [9] designed a scheme to adaptively control the waiting time of packet pool in different network traffic conditions. Kim et al. [10] considered the interplay between rate adaptation and network coding, and then jointly designed the rate adaptation and network coding policy to improve the throughput of wireless network. Additionally, they proposed a coding scheme able to determine which decoding receiver acknowledges the reception of an encoded packet. Kim et al. [11] designed a distributed framework that facilitates the choice of the best rate on each link while using network coding. Zhang et al. [12] investigated the benefits of applying a form of network coding to unicast applications in disruption-tolerant networks. Jones et al. [13] considered the joint design of optimal routing, scheduling, and network coding strategies to maximize throughput in wireless networks.

The above surveyed studies related to network coding have the feature that an intermediate node, i.e., a coding node, picks at most one packet from one flow to form the encoded packet, which is referred to as one-packet-one-flow (OPOF) scheme below. In fact, the OPOF scheme fails in making full advantage of network coding that some packets can be piggybacked without consuming extra capacity of wireless channel because all the shorter packets are padded with zeroes to make their lengths identical in the case when the lengths of the packets being coded differ. As an example, consider the coding case shown in Fig. 2(a), where two flows have coding chance and the sizes of data packets P_1 and P_2 from them are 1200 B and 400 B, respectively. As a result, the size of the encoded packet P1 ⊕ P2 is 1200 B, where only 400 B in packet P_2 is piggybacked on P_1. In fact, up to 800 B more data can be also piggybacked on P_1 if P_2 has a larger size.



(a) COPE



(b) Improved coding scheme

Fig.2. MPOF scheme

The above shortcoming can be overcome by letting more packets from each flow participate in network coding. For example, as shown in Fig. 2(b), three packets in the lower flow, say $P_{2,1}$, $P_{2,2}$, and $P_{2,3}$, can be picked and combined into one packet as P_2, which is then encoded with P_1. Thus, four packets $P_{2,1}$, $P_{2,2}$, and $P_{2,3}$, and P_1 are transmitted in one transmission without expending extra capacity of the channel, thus improving the throughput. Motivated by this, we propose a novel network coding scheme. The main contribution of the paper is as follows:

1) The network coding scheme aiming at encoding packets with each consisting of one or more packets from a flow is presented to improve throughput, which has the feature that Multiple Packets in One Flow (MPOF) may be combined to form a packet with larger size to be encoded. Accordingly, we refer to the proposed network coding scheme as MPOF network coding scheme below.

2) The MPOF takes data rates of the links connecting the receivers into account so that links with lower data rates are not included to avoid the encoded packets can only be transmitted in a too low data rate.

3) The numeric experiments show the throughput under MPOF scheme can be improved as high as 20% compared with the existing OPOF policy.

The rest of this paper is organized as follows. The MPOF network coding scheme is presented in Section II. Simulation results and performance analysis are presented in Section III. Section IV concludes the paper.

II. MPOF NETWORK CODING SCHEME

In this section, we propose the MPOF scheme for IEEE 802.11 infrastructure WLANs. As in [4] and [14], we assume that the coding node maintains a dedicated queue able to hold M packets for each flow passing through the node and the buffered packets can be encoded together. For a given flow F_i, we use L_i, R_i, and N_i to denote the size of the packet from the flow, the data rate of the link from the coding node to the receiver in the flow, and the number of the flow's packets buffered in the coding node.

We observe that the time consumed to transmit a single encoded packet may be larger than the total time for transmitting all the packets taking part in the encoded packet, especially when there exists a flow with a much lower transmission rate and a smaller packet size than the other flows. For example, suppose that there are two flows, where flow F_1 is with L_1 = 1000 B and R_1 = 10 Mbps, which takes $1000\times8/(10\times10^6)$ = 0.0008 s to transmit one packet, and flow F_2 is with L_2 = 600 B and R_2 = 1 Mbps, which takes $600\times8/10^6$ = 0.0048 s for transmitting one packet. Thus, the total time consumed in transmitting one packet from each flow without network coding is 0.0056 s. but the time consumed by transmitting an encoded packet is $1000\times8/10^6$ = 0.008 s since the encoded packet must satisfy the property that its size is equal to that of the largest packet, i.e., 1000 B, and its transmission rate is the lowest one, i.e., 1 Mbps so that any receiver is able to receive the encoded packet successfully.

To avoid the above observed problem, the proposed MPOF scheme encodes packets selected from several suitable flows even when much more flows can be encoded together. We use transmission time (TT) to measure the influence of transmission rate on throughout. That is, in the MPOF scheme, once the coding node obtains the transmission chance, it selects some non-empty flows to encode according to the principle that a new flow is chosen only when the TT of the newly encoded packet generated after the new flow joins is less than the sum of the TT of the original encoded packet without the new flow and the total TTs of the packets being coded in the new flow. The algorithm underlies the proposed MPOF scheme is described as the following, in which L^* and R^* stand for the size of the current encoded packet and the transmission rate of the encoded packet, respectively, and S_f and S_p are the set of the selected flows and the set of the packets to be encoded together, respectively.

Step 1. Find out all the non-empty coding flows with each corresponding to a non-empty queue and list them as $F_1, F_2, \ldots, F_K$, where K is the number of the non-empty flows. Choose the flow with the maximum packet size, say F_x ($x \in \{1, 2, \ldots, K\}$), and initialize $L^* = L_x$, $R^* = R_x$, $S_f = \{F_x\}$ (i.e., the flow with the maximum packet size is definitely to be encoded). Moreover, put the first packet in the queue of F_x into S_p.

Step 2. Set $k = 1$.

Step 3. If $k = x$, go to Step 6.

Step 4. Let $n_k \equiv \min\left\{\left\lfloor L^* / L_k \right\rfloor, N_k\right\}$, which is the number of flow F_k's packets being combined to form one packet participating in encoding, where $\lfloor \ \rfloor$ is the floor function. Additionally, check whether the following condition holds:

$$\frac{L^*}{\min\{R_k, R^*\}} < \frac{L^*}{R^*} + \frac{n_k L_k}{R_k} \tag{1}$$

where the item on the left side represents the TT of the encoded packet after the packets from flow F_k joins, and the first and second items on the right side represent the TT of the original encoded packet exclusive of flow F_k and the TT for the packets in the flow F_k to be combined into one packet so as to take part in the network coding.

If Eq.(1) holds, which means the TT is reduced when the flow F_k participates in the encoding, then set $R^* = \min\{R_k, R^*\}$ and $S_f = S_f \cup \{F_k\}$ (i.e., flow F_k is chosen to be encoded with the existing flows in S_f). Moreover, combine the first n_k packets in the queue of flow F_k to form a new packet, which is added into S_p. Go to Step 6.

Step 5. Check if the number of the buffered packets for the flow F_k is greater than or equal to $M/3$. If so, choose the flow F_k for coding, i.e., let $R^* = \min\{R_k, R^*\}$ and $S_f = S_f \cup \{F_k\}$. Moreover, combine the first n_k packets in the queue of flow F_k to form a new packet, which is added into S_p. (This step is used to prevent the queue of flow F_k from overflow resulting from long time of no packet being chosen for encoding.)

Step 6. Let $k = k + 1$. If $k = K+1$, then go to Step 7, otherwise go to Step 3.

Step 7. Encode together all the packets in S_p to generate an encoded packet, which is then transmitted with the rate of R^*.

Step 8. End.

Now let us briefly analyze the computational complexity of the MPOF scheme. In Step 1, finding the queue with the largest packet size takes time $O(K)$ and setting the values of parameters takes constant time. The iterations of Steps 3, 4, 5 and 6 are conducted at most K times. In each iteration of Steps 3, 4, 5 and 6, updating the parameters and checking Eq. (1) take constant time and combining n_k packets of flow F_k at most takes time $O(M)$ as $n_k \leq M$. Thus, the iterations of Steps 3, 4, 5 and 6 totally takes time $O(KM)$. Finally, in Step 7, the coding operation takes time $O(K)$ as at most K packets are encoded together. Therefore, the overall complexity of the MPOF scheme is $O(KM)$, which is linear to the number of coding flows K and the queue size M.

III. SIMULATION AND PERFORMANCE ANALYSIS

We use Matlab to simulate an IEEE 802.11 infrastructure WLAN with an AP, in which a pair of nodes performs communication in two hops, and compare the network throughput of the proposed MPOF scheme with that of the traditional network coding (TNC) scheme [4], i.e., the OPOF scheme.

In our simulation settings, there are N pairs of source transmitters and receivers, which are distributed around the AP, i.e., the coding node. We assume each source node has saturated traffic, i.e., it always has buffered packets to transmit, but the AP does not generate its own packets. As in IEEE 802.11a standard, each node can support 8 different transmission rates in the set $\Omega = \{6, 9, 12, 18, 24, 36, 48, 54\}$ (Mbps). Additionally, all the link rates are randomly chosen from Ω. As in [15], the packet size of each flow is chosen from [40, 1500] B. The probabilities of the packet size being 40 B and 1500 B are 0.4 and 0.2, respectively, and the packet size randomly and uniformly distributes in [41, 1499] B with the total probability of 0.4. We adopt IEEE 802.11 MAC protocol and set the total simulation time to 2 seconds. The other parameters relative to MAC layer are set as follows. SlotTime = 20 μs, DIFS = 50 μs, $CW_{Min} = 31$, and $CW_{Max} = 1023$. Here, SlotTime, DIFS, CW_{Min}, and CW_{Max} stand for a time slot in MAC layer, distributed inter-frame space, minimum contention window, and maximum contention window, respectively. Finally, set $M = 20$. All the results shown in the following figures are the average of 1000 simulation runs.

The average throughputs under MPOF and TNC are shown in Fig. 3. From the figure, we observe that the proposed MPOF scheme outperforms TNC in throughput, which reflects the fact that throughput can be improved by MPOP in which the coding node is able to transmit more packets via one single transmission. However, it should be noted that either in MPOF or TNC, as N increases, throughput increases first and then decrease. The reason is explained as follows. When N is small, both the source node and the coding node are able to seize the channel quickly due to the contention for the channel between the source and coding node is negligible. In a word, competing for the channel does not bring much delay so that the throughput is increased when N grows, which enables data to be simultaneously delivered through multiple flows. However, when N grows continuously (e.g., N grows to 7 in Fig. 3), the channel contention becomes serious and the contention windows of the nodes are enlarged due to packet collisions, causing the nodes involved in packet collision to wait much time for obtaining the chance in each transmission, which decreases the throughput.

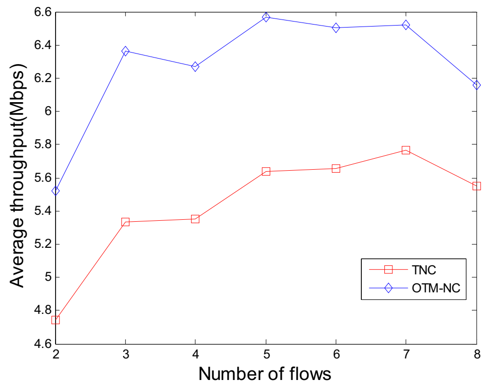


Fig.3. Average throughput of MPOF SCHEME and TNC

Fig.4 plots the cumulative distribution function of the throughput gain, which is defined as the ratio of throughputs of MPOF to TNC, for the case when $N = 5$. Obviously, we prefer the gain larger than 1, which indicates that MPOF has better throughput than TNC. From the figure, we observe that the accumulated probability of the gain being larger than 1.1 is around 0.72 since the accumulated probability of the gain less than 1.1 is around 0.28, which indicates that the gain is larger than 1.1 for most cases. Hence, MPOF considerably outperforms TNC in terms of throughput.

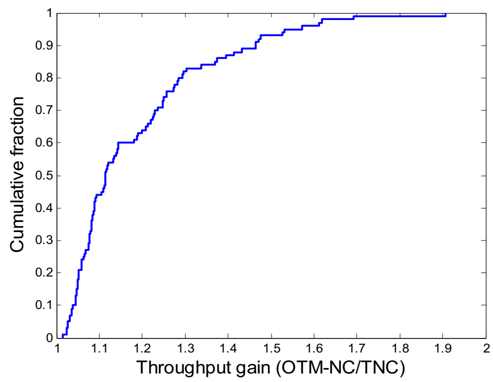


Fig.4. Cumulative distribution function of throughput gains (MPOF/TNC) when N=5.

IV. Conclusion

In this paper, we propose a novel network coding scheme, i.e., the MPOF scheme, to improve throughput for IEEE 802.11 infrastructure WLAN. The proposed scheme has the strengths that it takes data rates of links into account and is able to combine several packets form one flow to generate a larger packet so that more data can be piggybacked in an encoded packet, thus improving throughput considerably compared to TNC.

Acknowledgment

This work was partially supported by the National Natural Science Foundation of China (No. 61379124, 61001126), the Ph.D. Programs Foundation of Ministry of Education of China (No. 20123317110002), the Natural Science Foundation of Zhejiang Province, China (No. LY13F020025), and the Project-sponsored by SRF for ROCS, SEM of China.

References

[1] R. Ahlswede, N. Cai, S.-Y. R. Li, and R. W. Yeung. Network information flow. IEEE Transactions on Information Theory, 2000, 46(4): 1204-1216.

[2] Y. Wu, P. A. Chou, and S.-Y. Kung. Information exchange in wireless networks with network coding and physical-layer broadcast. In Proc. CISS, 2005.

[3] T. Ho and R. Koetter. Online incremental network coding for multiple unicasts. DIMACS Working Group on Network Coding, 2005.

[4] S. Katti, H. S. Rahul, W. Hu, D. Katabi, M. Médard, and J. Crowcroft. XORs in the air: Practical wireless network coding. IEEE/ACM Transactions on Networking, 2008, 16(3): 497-510.

[5] S. Sengupta, S. Rayanchu, and S. Banerjee. An analysis of wireless network coding for unicast sessions: The case for coding-aware routing. In Proc. 26th International Conference on Computer Communication, 2007, pp: 1028-1036.

[6] F. Zhao, and M. Medard. On analyzing and improving COPE performance. In Proc. of Information, 2010, pp: 1-6.

[7] Z. Fang, M. Medard, M. Hundeboll, J. Ledet-Pedersen, S. Rein, F. Fitzek. Comparison of analytical and measured performance results on network coding in IEEE 802.11 ad-hoc networks, NetCod 2012.

[8] Q. Dong, J. Wu, and W. Hu. On the X-ities of network coding in wireless networks. In Proc. of International Conference on Measurement and Modeling of Computer Systems, 2007.

[9] T. Nage, F. R. Yu and M. St-Hilaire. Adaptive control of packet overhead in XOR network coding. In Proc. of ICC, 2010, pp: 1-5.

[10] Y. Kim and G. de Veciana. Is rate adaptation beneficial for inter-session network coding. IEEE Journal on Selected Areas in Communications, 2009, 27(5): 635-646.

[11] T. Kim, I. Broustis, S. Vural, D. Syrivelis, S. Singh, S. Krishnamurthy, T. La Porta, , "Realizing the Benefits of Wireless Network Coding in Multirate Settings," IEEE/ACM Transactions on Networking, vol.21, no.3, pp.950-962, 2013.

[12] X. Zhang, G. Neglia, J. Kurose, D. Towsley, H. Wang, "Benefits of Network Coding for Unicast Application in Disruption-Tolerant Networks," IEEE/ACM Transactions on Networking,, vol.21, no.5, pp.1407-1420, 2013.

[13] N. Jones, B. Shrader, E. Modiano, Optimal routing and scheduling for a simple network coding scheme, IEEE INFOCOM, 2012.

[14] J. Le, J. Lui, and D. M. Chiu, “How many packets can we encode—an analysis of practical wireless network coding,” In Proc. of IEEE INFOCOM, 2008, pp: 371-375.

[15] Internet packet size distributions: Some observations. http://netweb.usc.edu/ rsinha/pkt-sizes/.