CCACK: Efficient Network Coding Based Opportunistic Routing Through Cumulative Coded Acknowledgments

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Abstract

The use of random linear network coding (NC) has significantly simplified the design of opportunistic routing (OR) protocols by removing the need of coordination among forwarding nodes for avoiding duplicate transmissions. However, NC-based OR protocols face a new challenge: How many coded packets should each forwarder transmit? *To avoid the overhead of feedback exchange, most practical existing NC-based OR protocols compute offline the expected number of transmissions for each forwarder using heuristics based on periodic measurements of the average link loss rates and the ETX metric. Although attractive due to their minimal coordination overhead, these approaches may suffer significant performance degradation in dynamic wireless environments with continuously changing levels of channel gains, interference, and background traffic.*

In this paper, we propose CCACK, a new efficient NC-based OR protocol. CCACK exploits a novel C*umulative* C*oded* ACK*nowledgment scheme that allows nodes to acknowledge network coded traffic to their upstream nodes in a simple way, oblivious to loss rates, and with practically zero overhead. In addition, the cumulative coded acknowledgment scheme in CCACK enables an efficient credit-based, rate control algorithm. Our experiments on a 22-node 802.11 WMN testbed show that compared to MORE, a state-of-the-art NC-based OR protocol, CCACK improves both throughput and fairness, by up to 3.2x and 83%, respectively, with average improvements of 11-36% and 5.7-8.3%, respectively, for different numbers of concurrent flows. Our extensive simulations show that the gains are actually much higher in large networks, with longer routing paths between sources and destinations.*

1 Introduction

Wireless mesh networks (WMNs) are increasingly being deployed for providing cheap, low maintenance Internet access (e.g. [26, 29, 28]). A main challenge in WMNs is to deal with the poor link quality due to urban structures and interference, both internal (among flows in the WMN) and external (from other 802.11 networks). For example, 50% of the operational links in Roofnet [26] have loss rates higher than 30% [1]. Hence, routing protocol design is critical to the performance and reliability of WMNs.

Traditional routing protocols (e.g., [11, 19, 2]) for multihop wireless networks treat the wireless links as point-to-point links. First a fixed path is selected from the source to the destination; then each hop along the chosen path simply sends data packets to the next hop via 802.11 unicast. *Opportunistic Routing* (OR), as first demonstrated in the ExOR protocol [3], has recently emerged as a mechanism for improving unicast throughput in WMNs with lossy links. Instead of first determining the next hop and then sending the packet to it, a node with OR *broadcasts* the packet so that all neighbor nodes have the chance to hear it and assist in forwarding.

In practice, it is not beneficial if all nodes in the network participate in forwarding traffic for a single flow. Hence, existing OR protocols typically construct a *belt* of forwarding nodes (FNs) for each flow and only members of the belt are allowed to forward packets.

OR provides significant throughput gains compared to traditional routing, however, it introduces a difficult challenge. Without any coordination, all members of the FN belt that hear a packet will attempt to forward it, creating spurious retransmissions, which waste bandwidth. To address this challenge, a coordination protocol needs to run among the nodes, so that they can determine which one should forward each packet.

Recently, [5] showed that the use of *random intra-flow network coding* (NC) can address this challenge in a very simple and efficient manner, with minimal coordination. With NC, the source sends random linear combinations of packets, and each router also randomly mixes packets it already has received before forwarding them. Random

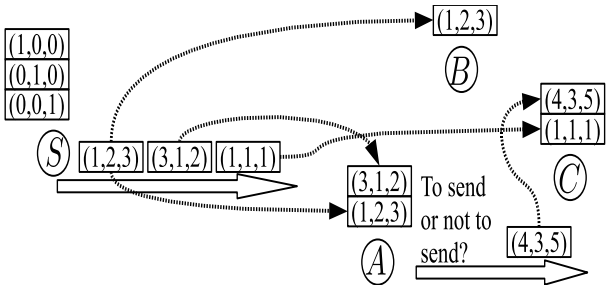


Figure 1. The importance of knowing how many coded packets to transmit.

mixing at each router ensures that with high probability different nodes that may have heard the same packet can still transmit linearly independent coded packets.

NC has significantly simplified the design of OR protocols and led to substantial throughput gains [5] compared to non-coding based protocols. However, the use of NC introduces a new challenge: *How many coded packets should each forwarder transmit?* This challenge, if not efficiently addressed, may prevent NC-based OR protocols from realizing the maximum possible gains.

1.1 The challenge in NC-based OR protocols

We illustrate the main challenge in NC-based OR protocols with the example shown in Figure 1. This figure shows a typical scenario of an NC-based OR protocol. The source S has three downstream FNs A, B, and C. Assume for simplicity that S has three innovative packets X_1, X_2, and X_3 to send. Instead of transmitting the native packets, S transmits three coded packets $X_1+X_2+X_3$, $3X_1+X_2+2X_3$, and $X_1+2X_2+3X_3$ in sequence, which are denoted by the corresponding *coding vectors* $(1,1,1)$, $(3,1,2)$, and $(1,2,3)$. Assume that $(1,1,1)$ coded packet is received by C, and the $(3,1,2)$ and $(1,2,3)$ packets are received by A and by $\{A,B\}$, respectively. The downstream FNs A, B, and C have received a sufficient amount of innovative packets. Collectively, the three FNs can now act as the new source and the original source S should stop transmission. However, it is a non-trivial task for S to know whether its downstream FNs have accumulated a sufficient amount of innovative packets.

The same challenge exists for the intermediate FN A. After transmitting a useful coded packet $(4,3,5)$, which is received by FN C, A has to decide whether it should continue or stop sending coded packets. Furthermore, A has limited knowledge about the reception status of the three packets transmitted by S (e.g., A may not know that C has received $(1,1,1)$ from S), which makes the decision of whether to stop transmission even harder for A than for the source S.

Note that overhearing, a common way of acknowledging non-coded wireless traffic, does not suit network coded traffic. Consider the same example in Figure 1. C generates a coded packet $c_1(1,1,1)+c_2(4,3,5)$. If the randomly chosen coefficients happen to be $c_1=c_2=1$, then a $(5,4,6)$ packet is sent. Suppose A overhears this new packet. If A were aware of the reception of the $(1,1,1)$ packet by C and also knew the coefficients $c_1=c_2=1$, then A could deduce that the previously transmitted $(4,3,5)$ packet was received successfully since $((5,4,6)-1\cdot(1,1,1))/1=(4,3,5)$. Nonetheless, in practice, neither piece of the information is available to A, it is thus impossible for A to know whether the $(4,3,5)$ packet is received or not by only overhearing the $(5,4,6)$ packet sent by C.

One way to address the challenge is to combine individual packet overhearing, as in non-coding based protocols, with a *credit system*, based on coded transmissions, and have the forwarders perform detailed bookkeeping to guarantee credit conservation in the system. This approach is taken in MC^2 [9]. Although theoretically optimal [20], this approach is quite complex in practice. In addition, like every approach that relies on individual packet overhearing, it requires a reliable control plane. In typical WMN environments with high packet loss rates or contention [1], this approach can cause excessive signaling overhead and retransmissions, which can signifi-

cantly limit the performance.

1.2 Loss rate based approaches

Since theoretically optimal solutions are hard to implement in practice, existing NC-based OR protocols use heuristics based on link loss rates, to address the challenge in a simple manner, and to minimize the control overhead.

MORE [5], the first NC-based OR protocol, employs an offline approach which requires no coordination among FNs. In MORE, the source calculates and assigns a *transmission credit* to each FN, using the ETX metric [6], computed from loss rate measurements. Receptions from upstream nodes are then used to trigger new transmissions at the FNs, with precomputed relative frequencies using the transmission credits. Since the ETX metric expresses the *expected* behavior, the approach used in MORE cannot guarantee that the destination will always receive enough packets, due to the randomness of the wireless channel. Hence, the source in MORE keeps transmitting packets from the same batch until it receives an ACK from the destination, unnecessarily increasing interference.

Many other works that improve MORE also use offline measured loss rates as a basic component in their proposed solutions (e.g., [24, 23, 15]).

The drawback of all these approaches is that performance heavily depends on the accuracy and freshness of the loss rate measurements. Loss rate estimates are obtained through periodic probing and are propagated from all nodes to the source. Apparently, the higher the probing frequency, the higher the accuracy, but also the higher the overhead. As a recent study [4] showed, even low-rate control overhead in non-forwarding links can have a multiplicative throughput degradation on data-carrying links.

To reduce this overhead, the authors of MORE collect the loss rates and calculate the credits only in the beginning of each experiment. In practice, this suggests that loss rate measurements should be performed rather infrequently. studies [7, 14] have shown that, *although link metrics remain relatively stable for long intervals in a quiet network, they are very sensitive to background traffic*. For example, in [7], the authors observe that 100 ping packets (one per second) between two nodes in a 14-node testbed caused an increase of 200% or more to the ETT [8] metric of around 10% of the links.[1] Even worse, a 1-min TCP transfer between two nodes in the same network caused an increase of more than 300% to the ETT metric of 55% of the links.

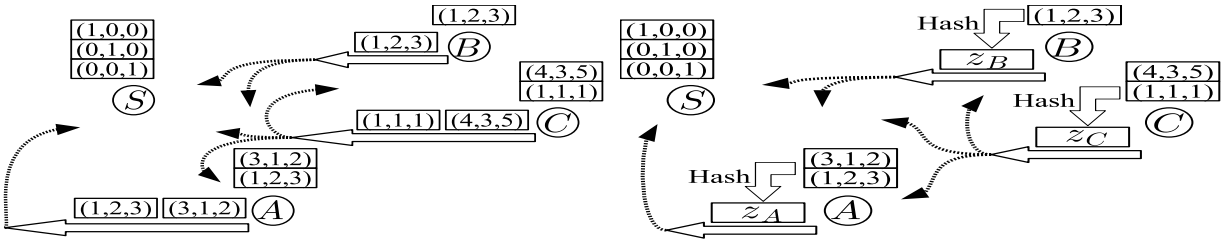
In summary, these approaches suffer from difficulties in accurately estimating loss rates. Overestimated loss rates cause redundant transmissions, which waste wireless bandwidth. On the other hand, underestimated loss rates may have an even worse impact, since nodes may not transmit enough packets to allow the destination to decode a batch. This motivates the need for a new approach, *oblivious* to loss rates.

1.3 Our approach – Cumulative Coded Acknowledgments

In this paper, we present a novel approach to NC-based OR and propose CCACK, a new efficient NC-based OR protocol. Unlike existing protocols, FNs in CCACK decide how many packets to transmit in an online fashion, and this decision is completely oblivious to link loss rates.[2] This is achieved through a novel **C**umulative **C**oded **ACK**nowledgment scheme that allows nodes to acknowledge network coded traffic to their upstream nodes in a simple and efficient way, with practically *zero overhead*. In other words, unlike existing NC-based OR protocols which use NC to avoid sending feedback, CCACK *encodes* feedback to exploit its benefits while hiding its overhead. Feedback in CCACK is not required strictly on a per-packet basis; this makes the protocol resilient to individual packet loss and significantly reduces its complexity, compared to [9].

[1] The ETT metric estimates the quality of a link taking into account both the loss rate (through the ETX metric) and the link bandwidth.

[2] By "oblivious to link loss rates" we mean here that loss rates are not taken into account in determining how many packets each FN should transmit. We still use MORE's loss rate based offline algorithm in CCACK to build the FN belt, for a fair comparison between the two protocols. We note though that the coded feedback mechanism in CCACK is orthogonal to the belt construction.



(a) Uncoded Feedback

(b) Coded Feedback

Figure 2. Different types of feedback for network-coded traffic.

Take the scenario in Figure 1 as a continuing example. One naive approach to ensure that S (resp. A) knows when to stop transmission is through the use of *reception reports*, for which each node broadcasts *all the basis vectors* of the received linear space to its upstream nodes, as illustrated in Figure 2(a).[3]

An obvious drawback of this approach is the size of the feedback messages. For practical network coding with symbol size $\mathrm{GF}(2^8)$ and batch size 32, each coding vector contains 32 bytes. To convey a space of dimension $\kappa \gg 1$ thus requires κ 32-byte vectors, which is too large to piggyback to normal forward traffic. The unreliability of the wireless channel further exacerbates the problem as the $\kappa \times 32$-byte feedback messages need to be retransmitted several times until they are overheard by all the upstream nodes.

In contrast, in CCACK each node uses *a single coded feedback vector* to represent the entire space, which may consist of $\kappa \gg 1$ basis vectors. In the broadest sense, the three coded acknowledgment vectors z_A to z_C in Figure 2(b) serve as a hash for their corresponding spaces. As will be explained in Section 3, we have devised a simple mechanism that successfully *compresses* (most of) the space information into a single vector, say z_A for node A, while allowing upstream nodes to *extract* the original space from z_A without exchanging any additional control information. Each single vector z_A can be easily piggybacked to the forward data traffic. This compressed/coded acknowledgment is critical to the efficiency since in CCACK overhearing any of the data packets of A with piggybacked coded ACK will convey to the upstream nodes the entire space (or most of the space) of A. This thus drastically reduces the need of retransmitting feedback information over the unreliable wireless channel.

In addition to efficiently solving the challenge of how many packets each FN should transmit, the cumulative coded acknowledgment scheme in CCACK enables us to develop an efficient rate control algorithm. In contrast, MORE has no explicit rate control mechanism and its performance degrades as the number of flows in the network increases [5, 20, 24, 23].

To evaluate CCACK, we first compare its performance against MORE, using extensive realistic simulations. Our simulations use a realistic physical model, with random signal variations due to fading, take into account the additional packet header overhead introduced by the use of NC and OR, and are conducted over a variety of network topologies. Our results show that CCACK improves both throughput and fairness over MORE, by 27-45% and 5.8-8.8%, respectively, on average, with different number of flows. For some challenged flows which completely starve under MORE, CCACK increases throughput by up to 21x and fairness by up to 124%. In addition, the coding and memory overheads of CCACK are comparable to those of MORE, making it easily deployable on commodity hardware.

To demonstrate this, we present an application layer implementation of CCACK and MORE on Linux and their performance evaluation on a 22-node 802.11 WMN testbed deployed in two academic buildings at Purdue University. Although the small size of our testbed along with the limitations of our implementation limit the potential gains, our testbed results show that CCACK improves both throughput and fairness, by up to 3.2x and 83%, respectively, with average improvements of 11-36% and 5.7-8.3%, respectively, for different numbers of

[3]We sometimes refer to the linear space spanned by the received vectors as the *knowledge space*.

flows, validating the benefits of our approach.

In summary, we make the following contributions:

- We identify the main challenge present in the newly emerged class of NC-based OR protocols: *How many coded packets should each forwarder transmit?* We discuss the inefficiencies of existing loss-based approaches to addressing this challenge and show, through our simulation and testbed evaluations the severe impact such approaches can have on the performance of NC-based OR protocols.

- We propose CCACK, a new efficient NC-based OR protocol. Unlike existing protocols, FNs in CCACK decide how many packets to transmit in an online fashion, and this decision is completely oblivious to link loss rates. Central to the design of CCACK is a novel Cumulative Coded ACKnowledgment scheme that allows nodes to acknowledge network coded traffic to their upstream nodes in a simple and efficient way, with practically zero overhead. In addition to efficiently solving the challenge of how many packets each FN should transmit, the cumulative coded acknowledgment scheme in CCACK enables us to develop an efficient rate control algorithm.

- CCACK brings a shift to the design paradigm of NC-based OR protocols. Existing NC-based OR protocols have identified feedback overhead as a main cause for performance degradation of practical wireless routing protocols and used NC to eliminate the need for feedback exchange, resorting to offline loss-based heuristics. On the contrary, CCACK *encodes* feedback to exploit its benefits and avoid the drawbacks of offline heuristics (e.g., stale information), and at the same time to hide its overhead.

- We present extensive simulations with a realistic physical model showing that CCACK offers significant throughput and fairness improvements over the state-of-the-art MORE. We identify the reasons for these benefits and the scenarios mostly benefited from CCACK, and discuss the relationship between these two performance metrics. We quantify the header, memory, and coding overheads of CCACK, and show that they are comparable to those of MORE, making CCACK easily deployable on commodity hardware.

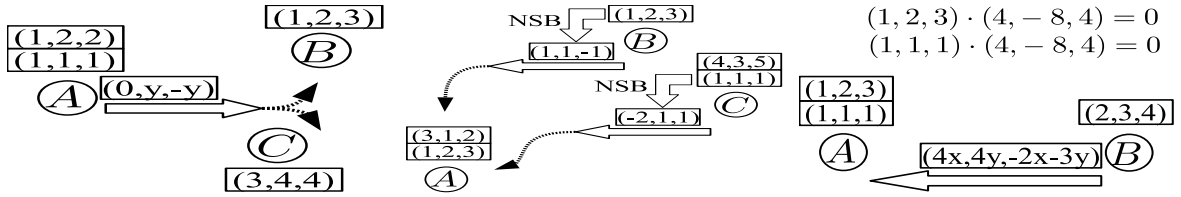
- We present an application-layer implementation of CCACK and MORE and their evaluation on a 22-node 802.11 WMN testbed deployed in two academic buildings at Purdue University. Our testbed experiments confirm the simulation findings.

The remaining of the paper is organized as follows. In Section 2, we introduce the basic principles of coded feedback through a simple existing coded feedback scheme. We identify two problems with this scheme which motivate the design of CCACK, presented in Section 3. Section 4 evaluates the performance of CCACK and MORE through extensive simulations and Section 5 describes the implementation and evaluation of CCACK and MORE on a wireless testbed. Finally, Section 6 concludes the paper.

2 Existing Coded Feedback Scheme

Coded feedback has been used in the past in a different context; in [18], a null-space-based (NSB) coded feedback scheme is used to enhance reliability of an NC-based multicast protocol for multimedia applications in mobile ad hoc networks. In this section, we review this scheme and identify two problems when trying to apply it to reliable unicast OR: the collective space problem and the false positive problem. These two problems motivate the need for a new cumulative coded feedback scheme which is a major component of CCACK.

Take Figure 3(a) for example. A batch of 3 packets are coded together and nodes A to C need to decode all three packets. Let B_v denote the buffer containing the innovative coding vectors received by A (which contains two vectors $(1, 2, 2)$ and $(1, 1, 1)$ in Figure 3(a)).



(a) NSB Feedback for unreliable multicast.

(b) NSB Feedback for reliable unicast OR.

(c) The false positive error of the NSB feedback.

Figure 3. Null-Space-Based (NSB) feedback for unreliable multicast, for reliable unicast OR, and the corresponding false positive event.

Since A has received fewer than 3 innovative packets, it informs its neighbor nodes that it needs more packets by appending to each coded packet a vector z_A satisfying

$$z_A \cdot v = 0, \quad \forall v \in B_v \tag{1}$$

Namely, the inner product between z_A and $v \in B_v$ is zero. There may be multiple choices of z_A that satisfy (1) (e.g., in Figure 3(a), z_A can be any vector of the form $(0, y, -y)$). z_A is then chosen *uniformly randomly* among all valid vectors satisfying (1). Let $S_A = \langle v : v \in B_v \rangle$ denote the linear space spanned by vectors in B_v. One can easily show that:

Lemma 1 *With the above random construction of z_A, any vector $v' \in S_A$ must satisfy $z_A \cdot v' = 0$. Moreover, for any vector $v'' \notin S_A$ we have* $\mathsf{prob}(z_A \cdot v'' = 0) = \frac{1}{2^8}$ *assuming the* $\mathrm{GF}(2^8)$ *finite field is used.*

From the above lemma, node B (resp. C) simply needs to compute the inner product of its own innovative vectors with z_A. In Fig. 3(a), suppose that z_A is chosen as $(0, 1, -1)$. Since $(0, 1, -1) \cdot (1, 2, 3) = -1 \neq 0$, node B must contain innovative packet for A. B can broadcast its innovative packet and once A receives it, A will be able to decode the entire batch.

2.1 Problems of the NSB Coded Feedback for Unicast OR

The goal of using coded feedback in the context of unreliable multicast is different from in the context of reliable unicast OR. In the former, coded feedback is used by a node to inform neighbors that *they have to send more packets*. In contrast, in the latter, we want to use coded feedback so that nodes can inform their upstream nodes that *they should not send any more packets*. This fundamental difference causes two major problems when trying to apply the above NSB coded feedback scheme to reliable unicast OR protocols, like MORE.

Problem 1: The collective space problem. Take Figure 1 for example. Nodes B and C would like to convey their space information to A so that A can stop packet transmission. Based on the NSB concept, B and C send $z_B = (1, 1, -1)$ and $z_C = (-2, 1, 1)$, respectively, which are orthogonal to their local innovative vectors (see Figure 3(b)). The idea is to hope that, upon the reception of z_B and z_C, A will know that the knowledge spaces of B and C have *collectively* covered the local knowledge space of A and thus will stop transmission.

Nonetheless, when A checks the inner product of the coded feedback and its own innovative packets, we have

$$z_B \cdot (3, 1, 2) = 2 \neq 0 \text{ and } z_C \cdot (3, 1, 2) = -3 \neq 0.$$

Therefore A *thinks that the coding vector* $(3, 1, 2)$ *is innovative to both its downstream nodes and thus continues transmission even when collectively B and C already have enough information.* This misjudgment is caused by

that the NSB coded feedback does not convey the collective space of all downstream nodes but only the space relationship between the individual pairs (e.g., A vs. B and A vs. C). Therefore, if we apply the NSB coded feedback as in [18] to unicast OR, A will not stop transmission until one of its downstream nodes has a local knowledge space that completely covers the local knowledge space of A. This defeats the purpose of OR.

Problem 2: Non-negligible false-positive probability. Take Figure 3(c) for example. A wants to send two packets to B and a network coded packet has been received by B already. To convey its local knowledge space back to A, B sends an orthogonal vector z_B satisfying (1), which is randomly chosen to be any vector of the form $z_B = (4x, 4y, -2x - 3y)$. Suppose that B chooses $z_B = (4, -8, 4)$ and A receives such z_B. *Since z_B is orthogonal to all the innovative vectors of A, A will wrongfully conclude that the knowledge space of B covers the local knowledge space of A. A thus attempts no further transmission.* Although Lemma 1 guarantees that this false positive event happens only with probability $\frac{1}{2^8}$, its impact to the system performance is significant. The reason is that in a multi-hop transmission, any single hop that experiences this false positive event will cause an upstream node to stop transmission prematurely. The communication chain is thus broken and the destination may not be able to receive enough independent packets for decoding. Although one can fix this false-positive issue by retransmitting another z_B vector, the necessary timer management for the unreliable feedback channel and the additional interference caused by retransmission easily negate the benefits of sending coded feedback.

Note that the false positive event has little impact in a multicast scenario. More explicitly, even if A misinterprets the z_B vector, the downstream node B knows that it has not received all three packets of the current batch. Therefore, it will simply keep appending feedback vectors in the code packets it transmits in order to request more information from A (or from any other neighbor nodes). In contrast, the intermediate node B in Fig. 3(c) does not need to decode/receive all three packets. The decision that A stops transmission can now be caused either by a false-positive event or by the fact that B indeed has received enough packets (although not all three packets). B thus faces the following dilemma: whether/when to retransmit another z_B that causes additional interference; or to stay quiet but risk unsuccessful batch decoding.

3 CCACK design

In this section, we present the design of CCACK. We begin with an overview of the protocol and then we describe its two main components: construction of a novel cumulative coded feedback scheme which addresses the two problems we discussed in Section 2.1, and a rate control algorithm, built upon this coded feedback scheme.

3.1 CCACK Overview

The source and the intermediate FNs in CCACK use intra-flow random linear NC. We selected a batch size of $N = 32$ packets and the random coefficients for each linear combination are selected from a Galois Field (GF) of size 2^8, same as in [5, 9, 15].

Nodes in CCACK maintain per flow state, which includes the *current batch* of the flow, a *credit counter* (Section 3.4), and three buffers: a packet buffer B_v, and two coding vector buffers B_u and B_w. With the exception of B_u and B_w, all the other information is also maintained in MORE. The size of B_v is equal to the batch size N, since the number of innovative packets is bounded by the batch size. The size B_u and B_w can be larger, since these buffers only store 32-byte coding vectors and not whole packets. In our implementation we used a size equal to $5 \times N$. Similar to MORE, this information is soft-state and it is flushed if no packet for a flow is received for 5 minutes.

The source and the FNs broadcast randomly mixed packets and store the coding vectors of these packets in B_w. Whenever a node overhears a packet, the node first checks whether the packet is innovative by comparing the coding vector to those of the existing packets in B_v. If innovative, the newly received packet is stored in B_v, similarly to MORE and other existing NC-based OR protocols. Regardless being innovative or not, the node also

checks whether the newly received packet is from an upstream node. If yes, then it stores the forward coding vector in B_u.

Each forward coding vector in B_u and B_w can be marked as H (heard by a downstream node) or $\neg\mathsf{H}$ (not heard). A coding vector is marked as $\neg\mathsf{H}$ when initially is inserted in either of the two buffers, since the node has no information at that time whether any downstream node has heard the packet or not.

Similar to [18], nodes in CCACK embed an additional *ACK vector* in the header of each coded data packet of the forward traffic to report a subset of the packets (or coding vectors) they have received (heard) in the past from their upstream nodes. For the following, we use the terms *forward coding vectors* and *ACK coding vectors* to denote the coding coefficients used to encode the payload of the packets and the feedback vectors used to acknowledge the space, respectively. The construction of the ACK coding vector using the vectors stored in B_u is described in Section 3.3. Nodes mark forward coding vectors as H, using the inner product of these vectors and the ACK coding vectors they receive from downstream nodes, as we explain in Section 3.3.

The destination periodically broadcasts coded feedback to its upstream nodes in the form of ACK vectors (without any payload). This is necessary to inform its upstream nodes whether they should temporarily stop transmitting, since the destination sends no data packets. Once it receives N innovative packets for a batch, it decodes the batch to obtain the N original packets. It then sends an end-to-end ACK back to the source along the shortest ETX path in a reliable manner.[4]

3.2 Solving the collective-space problem

In contrast to the NSB coded feedback scheme in [18], nodes in CCACK construct the ACK coding vectors using *all the received forward coding vectors stored in* B_u, and not only the innovative vectors stored in B_v. Also, when an upstream node A overhears a packet from a downstream node, it uses the ACK coding vector of that packet to decide whether any of the coding vectors in $B_u \cup B_w$, instead of B_v, have been heard by the downstream node.

Nodes keep checking the rank of the B_u and B_w vectors marked as H. When this rank becomes equal to the rank of innovative packets in B_v for a node A, A stops transmitting either temporarily, until it receives another innovative packet, or permanently if the rank of the B_v vectors is already equal to N. In both cases, the downstream nodes have received a sufficient number of packets that cover the innovative packets of A from the knowledge space perspective.

Focusing on B_u and B_w vectors instead of B_v, this new structure solves the collective-space problem of the NSB coded feedback. Continue our example in Figure 3(b). For node A, B_u contains the received coding vectors $(1, 2, 3)$ and $(3, 1, 2)$ while B_w contains the transmitted vector $(4, 3, 5)$. Suppose we reuse the NSB coded feedback for nodes B and C. Then by checking inner products with z_B and z_C, A knows that the $(1, 2, 3) \in B_u$ and $(4, 3, 5) \in B_w$ have been received. Since the rank of $(1, 2, 3)$ and $(4, 3, 5)$ is the same as the rank of B_v vectors, A stops transmission.

3.3 Solving the false positive problem

We now describe our new ACK design that drastically reduces the false-positive probability from $\frac{1}{2^8}$ to $\left(\frac{1}{2^8}\right)^M$ for any integer $M \geq 1$.

Each node maintains M different $N \times N$ *hash matrices* H_1 to H_M where $N = 32$ is the batch size and each entry of the matrix is randomly chosen from $GF(2^8)$. All nodes in the network are aware of the H_1 to H_M matrices of the other nodes. This is achievable by using the ID of a node as a seed to generate the H_1 to H_M

[4] In our current implementation, similar to MORE, the source moves to batch $i + 1$ only when it receives the end-to-end ACK from the destination for batch i. As [15] showed, a better approach is for the source to move to batch $i + 1$ immediately after it stops transmitting packets for batch i. In the future we plan to incorporate this feature in CCACK.

matrices. We assume that all vectors are row vectors and we use the transpose u^{T} to represent a column vector (constructed from the row vector u).

To improve the efficiency of our feedback mechanism, we associate a $usage\_count$ with every vector in B_u. When a vector is placed in B_u, its $usage\_count$ is set to 0. Every time this vector is selected in the feedback construction algorithm, its $usage\_count$ is incremented by 1. The ACK vector is always constructed using those vectors in B_u with the lowest counts. This will reduce the probability that the same vectors are repeatedly acknowledged many times.

Nodes construct the ACK vectors using the following algorithm:

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§ CONSTRUCT THE ACK VECTOR

1: Start from a $0 \times N$ matrix Δ.
2: **while** The number of rows of $\Delta \leq N - 1 - M$ **do**
3: Choose the u with the smallest $usage\_count$ from B_u. If more than one such u exist, choose one randomly.
4: **for** $j = 1$ to M **do**
5: **if** the $1 \times N$ row vector uH_j is linearly independent to the row space of Δ **then**
6: Add uH_j to Δ.
7: Perform row-based Gaussian elimination to keep Δ in a row-echelon form.[5]
8: **end if**
9: **end for**
10: Increment the $usage\_count$ of u by 1.
11: **end while**
12: Choose randomly the coding coefficients c_1 to c_N such that the following matrix equation is satisfied:

$$\Delta(c_1, \cdots, c_N)^{\mathrm{T}} = (0, \cdots, 0)^{\mathrm{T}}.$$

Remark 1: We also require that the randomly chosen coefficients c_1 to c_N are not all zero.
Remark 2: By Line 3 there will be at least 1 degree of freedom when solving the above equations. Since Δ is in the row-echelon form, it is easy to choose c_1 to c_N.
13: Use the vector $(c_1, \cdots, c_N)$ as the ACK vector.

When a node A overhears a packet with an ACK vector z from a downstream node, it uses again the inner product to check all its vectors in B_u and B_w and determine whether any of them has been heard by the downstream node. More explicitly, a vector $u \in B_u$ (or B_w) is marked H if and only if u passes *all the following M different "H_tests"* (one for each H_j):

$$\forall j = 1, \cdots, M, \quad uH_j z^{\mathrm{T}} = 0, \tag{2}$$

where H_1 to H_M are the hash matrices of the downstream node of interest.

Remark: In our practical implementation, instead of choosing completely random hash matrices H_1 to H_M (each with N^2 random elements), we simply choose H_1 to H_M as "random diagonal matrices", with the N diagonal elements for each H_j randomly chosen from 1 to 255 (excluding zero) and all other elements being zero. This simplification improves the efficiency as the matrix multiplication uH_j can be performed in linear instead of N^2 time.

We now quantify the false positive probability (passing all M tests simultaneously) with this new coded feedback scheme.

Proposition 1 *Consider an upstream/downstream node pair A_U and A_D, and A_U receives an ACK vector z_0 from A_D. The hash matrices H_1 to H_M of node A_D are chosen uniformly randomly. For any w vector in $B_u \cup B_w$ of*

[5]Since Δ is always of row-echelon form, it is easy to check whether the new vector is linearly independent to the row space of Δ.

the upstream node A_U, if such w is in the space of the u vectors selected by the downstream node A_D, then it is guaranteed that such w vector will pass all M tests in (2)*. If such w is not in the space of the selected u vectors, then the false-positive probability (passing all M tests) is $\left(\frac{1}{2^8}\right)^M$.*

Proof: Let S_B denote the linear space spanned from the u vectors selected by A_D. If w in $B_u \cup B_w$ of A_U is in S_B, then $w = \sum_i \alpha_i u_i$ is the linear combination of the selected u vectors (indexed as u_i). Since by construction $u_i H_j z_0^{\mathrm{T}} = 0$ for all selected u_i, we have $w H_j z_0^{\mathrm{T}} = \sum_i \alpha_i u_i H_j z_0^{\mathrm{T}} = 0$. The proof is complete.

Suppose that w in $B_u \cup B_w$ of A_U is not in S_B. Conditioning on the non-zero z_0 vector, for any j the $H_j z_0^{\mathrm{T}}$ vector must be randomly distributed over the null space of S_B, since $u_i H_j z_0^{\mathrm{T}} = 0$ for all selected u_i vectors and since H_j is chosen randomly. Moreover, conditioning on the non-zero z_0, $H_1 z_0^{\mathrm{T}}$ to $H_M z_0^{\mathrm{T}}$ are independently distributed over the null space of S_B. As a result, even though only a single vector z_0 is transmitted, the CCACK scheme has the same effect of using the NSB coded feedback M times, sending out M independently randomly selected vectors ($H_1 z_0^{\mathrm{T}}$ to $H_M z_0^{\mathrm{T}}$) from the null space of S_B. By Lemma 1, the overall false-positive event is when all M tests return false positive. The overall false positive probability becomes $\left(\frac{1}{2^8}\right)^M$. The proof is complete. □

The M value represents a tradeoff between how many vectors one can acknowledge $\frac{32}{M}$ and the false-positive probability $(2^{-8})^M$. Since *any false alarm event for any packet over any link will trigger the land-sliding cost of breaking the communication chain*, we observe in our experiments that any $M \leq 3$ will severely jeopardize the reliability of the CCACK. In our implementation we thus choose $M = 4$, which gives a false positive probability of 2.23×10^{-10} that is necessary for the effectiveness of CCACK.

It is worth noting that a naive way of avoiding false-positive events is to increase the underlying finite field size $\mathrm{GF}(2^b)$, which is not viable for WMNs. One reason is that to achieve the level of false-positive probability needed in our CCACK scheme ($M = 4$), we need $b = 32$, which uses 4 bytes to represent a single coding symbol. The size of each forward coding vector and each coded feedback vector thus grows from 32×1 bytes to 32×4 bytes, which substantially increases the overhead. An even bigger challenge is that each addition and multiplication coding operation now operate on $\mathrm{GF}(2^{32})$. A table look-up method has to have $2^3 2 \times 2^3 2$ 4-byte entries, which takes prohibitively 4 million terabytes to store. Since table look-up is impossible, one thus has to use online polynomial-based computation each time a coding operation needs to be performed, which is far beyond the capability of today's silicon technology.

3.4 Rate control

The cumulative coded feedback scheme in CCACK helps nodes to determine when they should stop transmitting packets for a given batch, but it does not tell anything about *how fast* nodes should transmit before they stop. Unlike in MORE, in CCACK we cannot use receptions from upstream to trigger new transmissions, since the goal is exactly to stop the upstream nodes, when the downstream nodes have sufficiently enough packets. In addition, we want to apply rate control to the source as well, and not only to the FNs.

The rate control algorithm in CCACK uses a simple credit scheme, which is oblivious to loss rates but aware of the existence of other flows in the neighborhood, and leverages CCACK's cumulative coded acknowledgments.

For each flow f at a node, we define the "differential backlog"[6] as:

$$\Delta Q^f = \dim(B_v^f) - \dim(B_H^f) \tag{3}$$

[6]Our solution is inspired by the theoretical backpressure based rate control algorithms [21]. The difference is that, instead of queue lengths, we use innovative coded packets to define a *cumulative differential backlog* for flow f at every node with respect to all its downstream nodes for that flow.

where B_H^f is the set of vectors in $B_u^f \cup B_w^f$ marked as H, and $\dim(S)$ denotes the number of linearly independent packets in the set S. Note that $\dim(B_H^f) \leq \dim(B_v^f)$. ΔQ^f is the difference between the number of innovative packets at a given node and the cumulative number of innovative packets at its downstream FNs for flow f. As we saw in Section 3.2, when $\Delta Q^f = 0$, i.e., $\dim(B_v^f) = \dim(B_H^f)$, the node stops transmitting packets for flow f. Note that for the destination of flow f, $\Delta Q^f = 0$.

We also define the relative differential backlog ΔQ_{rel}^f for each flow f as:

$$\Delta Q_{rel}^f = \frac{\Delta Q^f}{\Delta Q^f + \Delta Q_N} \tag{4}$$

where, ΔQ_N is the total differential backlog of all the neighbor nodes for all flows, calculated as follows. Every time a node n_j broadcasts a coded data packet, it includes in the packet header its current total differential backlog $\Delta Q_{n_j}^{tot}$ of all flows crossing that node. All nodes that hear this packet update their ΔQ_N as an exponential moving average:

$$\Delta Q_N = 0.5 \times \Delta Q_N + 0.5 \times \Delta Q_{n_j}^{tot} \tag{5}$$

Every node in CCACK (including the source and the destination) maintains a credit counter for each flow. Every time there is a transmission opportunity for a node A, one flow f is selected in a round robin fashion, among those flows with $\Delta Q^f > 0$, and the credit counter of that flow is incremented by $\alpha \times \Delta Q_{rel}^f + \beta$. If the counter is positive, the node transmits one coded packet for flow f and decrements the counter by one, otherwise it selects the next flow. The credit increment $\alpha \times \Delta Q_{rel}^f + \beta$ is larger for flows with large "backpressure", thus packets of such flows will be transmitted more frequently. For our implementation we selected $\alpha = 5/6$ and $\beta = 1/6$. If $\Delta Q_{rel}^f = 1$, then $\alpha \times \Delta Q_{rel}^f + \beta = 1$ and the credit counter will always remain equal to 1, effectively allowing the node to always transmit.

4 Evaluation

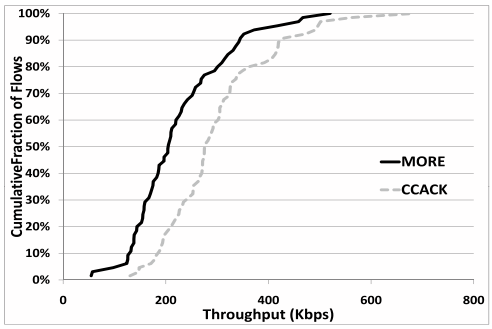
4.1 Methodology

We evaluated the performance of CCACK and compared it against MORE using extensive simulations. We used the Glomosim simulator [22], a widely used wireless network simulator with a detailed and accurate physical signal propagation model. Glomosim simulations take into account the packet header overhead introduced by each layer of the networking stack, and also the additional overhead introduced by MORE or CCACK. For the implementation of MORE, we followed the details in [5].

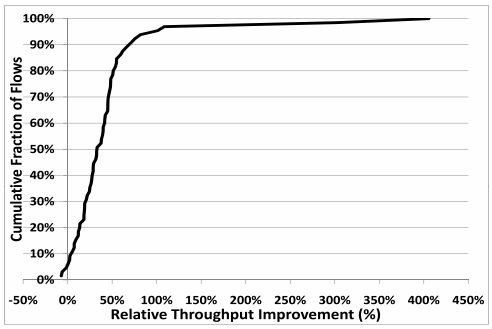
We simulated a network of 50 static nodes placed randomly in a $1000m \times 1000m$ area. The average radio propagation range was 250m, the average sensing range was 460m, and the channel capacity was 2Mbps. The *TwoRay* propagation model was used and combined with the Rayleigh fading model to make the simulations realistic. Because of fading, transmission and sensing range are not fixed but vary significantly around their average values.

We simulated each protocol in 9 different randomly generated topologies, i.e., placement of the 50 nodes. We varied the number of concurrent flows from 1 up to 4. For a given number of flows, we repeated the simulation 10 times for each topology, selecting randomly each time a different set of source-destination pairs, i.e., we had a total of 90 different scenarios for a given number of flows. In each scenario, every source sent a 12MB file, consisting of 1500-byte packets.

Following the methodology in [5, 3], we implemented an ETX measurement module in Glomosim which was run for 10 minutes prior to the file transfer for each scenario to compute pairwise delivery probabilities. There was no overhead due to loss rate measurements during the file transfer.



(a) CDF of throughputs achieved with MORE and CCACK.



(b) CDF of relative throughput improvement of CCACK over MORE.

Figure 4. Throughput comparison between CCACK and MORE – single flow.

It is generally known that the full benefit of OR over traditional routing is exposed when the destination is several hops away from the source [5]; in those cases, OR reduces the overhead of retransmissions incurred by high loss rates and increased self-interference. Hence, for the single-flow experiment, among the 90 flows we simulated, we show the results of the 65 flows for which the destination was not within the transmission range of the source (with ETX shortest paths of 3-9 hops). For the evaluation with multiple flows, we kept scenarios with flows of shorter paths, when those flows interfered with other flows. On the other hand, we do not show the results for scenarios where the multiple flows were out of interference range of each other, since those scenarios are equivalent to the single-flow case. We were left with 68 scenarios with 2 flows, and 69 scenarios with 3 and 4 flows.

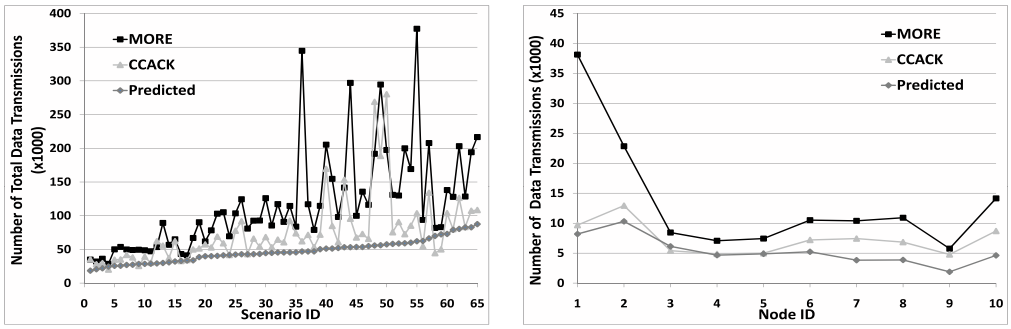
4.2 Single flow

We begin our evaluation with a single flow. Figure 4(a) plots the Cumulative Distribution Function (CDF) of the throughputs of the 65 flows with MORE and CCACK. We observe that CCACK outperforms MORE; the median throughput with CCACK and MORE is 276Kbps and 205Kbps, respectively.

Figure 4(b) plots the CDF of the relative throughput improvement of CCACK over MORE for all 65 flows, defined as $\frac{T^f_{CCACK} - T^f_{MORE}}{T^f_{MORE}} \times 100\%$, where T^f_{CCACK}, T^f_{MORE} are the throughput of flow f with CCACK and MORE, respectively. We observe that CCACK achieves a higher throughput than MORE for 95% of the flows. The median gain of CCACK over MORE is 34%. However, for some challenged flows with the destination 7-9 hops away from the source, the throughput with CCACK is 2-5x higher than with MORE.

Where does the gain for CCACK come from? Figure 5(a) plots the total number of data transmissions with CCACK and MORE in each of the 65 scenarios, as well as the predicted number of transmissions in each scenario using MORE's offline ETX-based credit calculation algorithm. The 65 scenarios are sorted with respect to the predicted number of transmissions.

We observe that nodes with MORE perform a higher number of transmissions than the predicted number in all 65 scenarios. The actual number is often more than twice the predicted number, and in some scenarios up to 6-7x the predicted number. This shows that the credit calculation algorithm based on offline ETX measurements mispredicts the required number of transmissions even in the absence of background traffic. The cause is self-interference which changes the loss rates, which in most cases become higher than in a quiet network, where only probing traffic exists [14]. Moreover, the source in MORE keeps transmitting packets until it receives an ACK from the destination. With long paths, this may result in a large number of unnecessary transmissions, as the ACK travels towards the source.



(a) Total number of data transmissions per scenario. (b) Total number of data transmissions per node for one scenario.

Figure 5. Total number of data transmissions with MORE and CCACK, and predicted number of transmissions, based on MORE's credit calculation algorithm, with a single flow.

In contrast, the number of data transmissions with CCACK is much lower than with MORE in all but 2 scenarios. In most scenarios it is close to the predicted number, and in some cases, it is even lower. This shows the effectiveness of the coded feedback mechanism in CCACK, combined with the online rate control mechanism of Section 3.4.

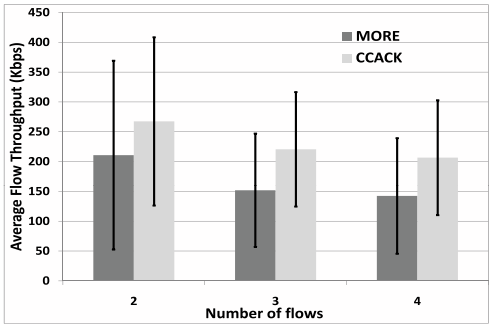
Figure 5(b) shows an example (one scenario) of how data transmissions are distributed over the FNs. Nodes are sorted with respect to their ETX distance to the destination, i.e., node 1 is the source and node 10 is the FN closest to the destination. With MORE, the source and the FN closest to the source, perform many more transmissions than the remaining FNs, (2.5-7.6x and 1.4-4.6x, respectively). In contrast, CCACK ensures that these nodes stop transmitting when the remaining downstream FNs have received enough innovative packets. Overall, with CCACK, all 10 nodes perform fewer transmissions than with MORE. The savings range from 17% (for node 9) up to 74% (for the source).

4.3 Multiple flows

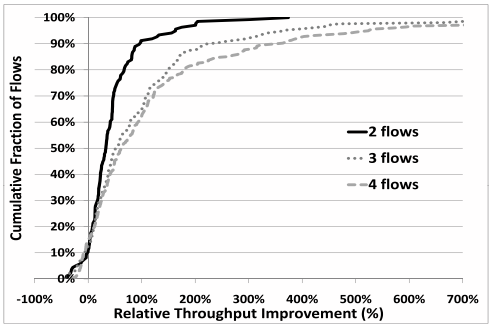
We now evaluate CCACK and MORE with multiple concurrent flows. Here, in addition to throughput, we compare the two protocols in terms of fairness, using *Jain's fairness index* (FI) [10]. Jain's FI is defined as $(\sum x_i)^2/(n \times \sum x_i^2)$, where x_i is the throughput of flow i and n is the total number of flows. The value of Jain's FI is between 0 and 1, with values closer to 1 indicating better fairness.

Throughput Comparison Figures 6(a), 6(b) compare throughput with CCACK and MORE with 2, 3, and 4 flows. Figure 6(a) plots the average per-flow throughput with the two protocols as a function of the number of flows. We observe that CCACK outperforms MORE by 27% on average in the 2-flow case, and by 45% on average in the 3-flow and 4-flow cases. Note that the gain of CCACK is higher with a larger number of flows, when the congestion level becomes higher causing substantial changes to the ETX values. given flow at nodes whose downstream nodes have collectively received a sufficient number of packets, a large amount of bandwidth is saved which can be used by the nodes or their neighbors for transmitting packets for other flows. In contrast, the gain of MORE over traditional routing in [5] drops as the number of concurrent flows increases.

Figure 6(b) plots the CDF of per-flow relative throughput improvement with CCACK over MORE, as defined in Section 4.2, with 2, 3, and 4 flows. CCACK improves per-flow throughputs for more than 85% of the flows in all 3 cases (with 2, 3, and 4 flows). The median improvement is 33%, 55%, and 62%, respectively, with 2, 3, and 4 concurrent flows. Similar to the single flow experiments, some starving flows with MORE show a several-fold

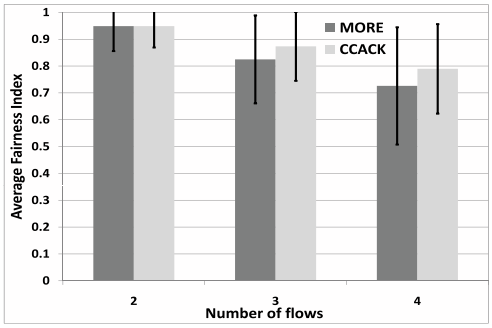


(a) Average per-flow throughputs (bars) and standard deviations (lines).

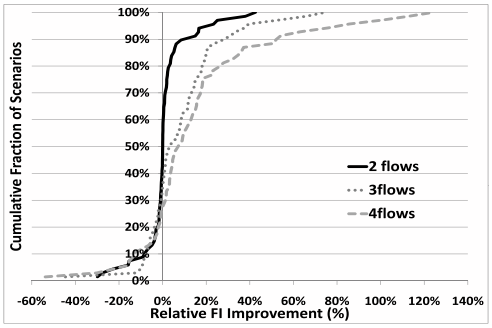


(b) CDF of relative throughput improvement of CCACK over MORE.

Figure 6. Throughput comparison between CCACK and MORE – multiple flows.



(a) Average per scenario FIs (bars) and standard deviations (lines).



(b) CDF of relative FI improvement of CCACK over MORE.

Figure 7. Fairness comparison between CCACK and MORE – multiple flows.

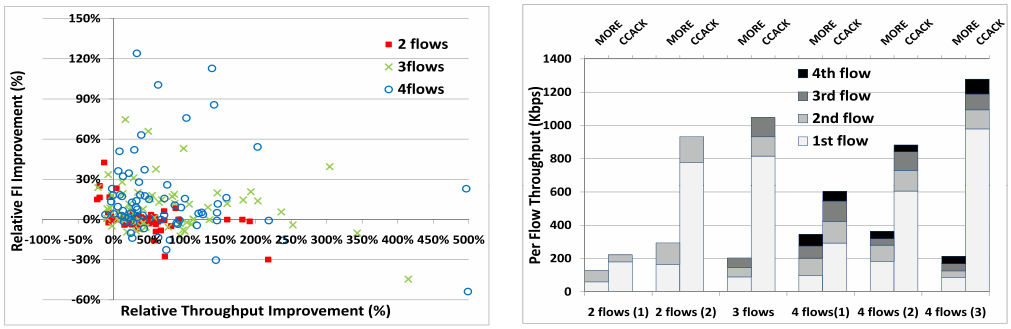
improvement with CCACK, up to 4.7x, 9.1x, and 21.4x, in the 2-, 3-, and 4-flow cases, respectively.[7]

Fairness Comparison Figures 7(a), 7(b) compare fairness with CCACK and MORE in case of 2, 3, and 4 concurrent flows. Figure 7(a) plots the average FI with the two protocols. We observe that the average FI is the same with the two protocols in the 2-flow case, but is higher with CCACK in the 3-flow, and 4-flow case by 5.8% and 8.8%, respectively.

Figure 7(b) plots the CDF of per-scenario relative FI improvement with CCACK over MORE, defined similarly to the relative throughput improvement in Section 4.2, with 2, 3, and 4 flows. We observe that CCACK improves fairness in more scenarios as the number of flows in the network increases – in 40% of the 2-flow scenarios, 65% of the 3-flow scenarios, and 72% of the 4-flow scenarios. Similar to the throughput results, the improvement is very large for some scenarios: up to 74% with 3 flows, and up to 124% with 4 flows. This shows again that CCACK helps some challenged flows, which completely starve with MORE. The improvement in fairness with CCACK is a result of the rate control algorithm, which calculates the transmission credits of the nodes online, taking into account their differential backlogs. In contrast, in MORE credits are calculated offline for each flow and are oblivious to the presence of other flows.

Throughput vs. Fairness We now investigate more closely the relationship between throughput and fairness. Figure 8(a) shows the scatterplots of the relative total throughput improvement per-scenario vs. the relative FI improvement per-scenario, in the 2-, 3-, and 4-flow experiments.

[7]Note that the heavy tails of the 3-flow and 4-flow curves are not shown in Figure 6(b) for better clarity.



(a) Scatterplot of relative throughput improvement vs. relative FI improvement with 2, 3, and 4 flows.

(b) Per-flow throughputs with MORE and CCACK for the 6 scenarios with the largest FI decrease under CCACK.

Figure 8. Investigating the relationship between throughput and fairness.

We observe that CCACK improves at least one of the two metrics in all but two scenarios (two points in the 3rd quadrant of Figure 8(a)). There are a few points in the 2nd quadrant for all three cases; these are scenarios, where CCACK improves fairness, at the cost of a small total throughput decrease. The majority of the points for the 2-flow case are gathered in the 1st and 4th quadrants, i.e., CCACK either improves throughput at the cost of a (typically) small decrease in fairness, or it improves both metrics. The majority of the points are gathered in the 1st quadrant for the 3-flow and 4-flow cases. This shows that as the number of flows increases, CCACK improves both throughput and fairness in most scenarios.

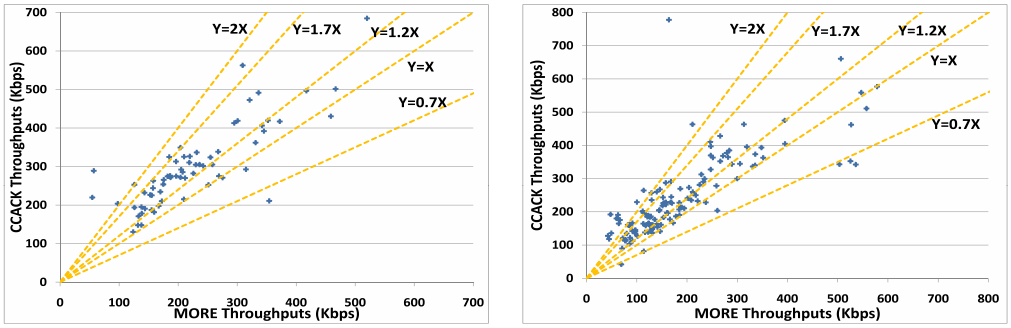
We now take focus on a few points in the fourth quadrant in Figure 8(a), corresponding to scenarios where FI is reduced by more than 20% with CCACK. There are two 2-flow, one 3-flow, and three 4-flow scenarios (points). Note that all 6 these points correspond to large throughput improvements, from 72% up to 499%. One may wonder if these improvements are achieved at the cost of compromising the fairness, i.e., throughput of only one flow increases significantly, causing starvation to the remaining flows.

Figure 8(b) shows that this is not the case. This figure plots the individual per-flow throughputs with MORE and CCACK for these 6 scenarios. We observe that in all but 2 cases, CCACK improves throughput of *all* flows involved. The reduction in the FI actually comes from the fact that throughput improvement is much higher for some flows than for some others, and not as a result of starvation of some flows. Take the last scenario (*4 flows (3)*) as an example. CCACK improves throughput of the first flow by 11x (from 85Kbps to 978Kbps), but also improves throughputs of the other 3 flows by 183%, 108%, and 113%.

A closer look at the topology of that scenario revealed an interesting situation. We found that the first flow was a 1-hop flow, whose FN belt overlapped with the FN belt of the 4-th, 9-hop flow, near the source of the 4th flow. With MORE, it took a long time for the destination of the 4th flow to decode each batch; during that time the source as well as every other FN kept transmitting packets for that batch. Since routers in MORE serve flows in a round robin fashion, they kept switching between the 1st and the 4th flow. In other words, a short flow was starved because of a long flow(!), achieving a throughput of only 85Kbps, although there was no need for the routers to forward packets of the long flow. In contrast, with CCACK, the coded acknowledgment scheme quickly caused the source of the 4th flow to stop transmitting packets, once the remaining FNs had enough packets. Hence, the FNs near the source were able to forward packets only for the 1-hop flow, increasing its throughput to 978Kbps.

4.4 Which flows benefit most from CCACK?

Finally, we examine which flows benefit most from CCACK in single-flow and multi-flow scenarios. Figures 9(a), 9(b), 9(c), and 9(d) show the scatterplots for the individual flow throughputs achieved with MORE and



(a) Single flow.

(b) Two flows.

(c) Three flows.

(d) Four flows.

Figure 9. Scatterplots of per-flow throughputs achieved with MORE and CCACK with 1, 2, 3, and 4 concurrent flows.

CCACK with 1, 2, 3, and 4 concurrent flows in the network.

Single flow In Figure 9(a), we observe that the majority of the points in the single-flow case lie between the lines $Y = 1.2X$ and $Y = 1.7X$, i.e., the throughput gain of CCACK over MORE for a large range of absolute MORE throughput values (100-350Kbps), is typically 20-70%, independent of the absolute throughput value of MORE. In other words, when a single flow is present in the network, CCACK benefits equally both low-throughput and medium-throughput flows. For flows of higher absolute MORE throughput (>350Kbps), the gain of CCACK is smaller; for these flows, the destination is 3-4 hops away from the source and MORE itself can realize most of the gains over traditional routing. On the other hand, for a few flows with very long path lengths (>7hops), the gains are higher than 100% – these are the points on the left of the $Y = 2X$ line in Figure 9(a). For these flows, throughput with MORE can be as low as 55Kbps; in contrast, with CCACK there is no flow with throughput lower than 130Kbps.

These high-gain points reveal an additional benefit of CCACK. Recall that with both protocols, the destination sends an end-to-end ACK to the source after decoding a batch to trigger the beginning of the next batch. In MORE, this ACK has to compete with coded traffic as it travel towards the source, since nodes never stop transmitting. With long paths, it may take a long time for the ACK to reach the source, and this can lead to significant throughput degradation for these flows, as has also been shown in [15]. In contrast, with CCACK, the ACK can quickly travel towards the source without any contention if there is no other flow in the network, since all nodes have already stopped transmitting, thanks to the coded feedback.

Multiple flows In Figures 9(b), 9(c), and 9(d), we observe that, as the number of flows increases, more points are gathered on the left of the $Y = 1.7X$ line; in the 4-flow case, in Figure 9(d), a large fraction of points is

Table 1. Coding overhead in CCACK in terms of $\mathrm{GF}(2^8)$ multiplications. Operations marked with (*) are common in MORE and CCACK.

Operation	Avg.	Std. Dev.
Packet Transmission		
Coded pkt construction (src/FNs)*	48000/27240	0/13128
ACK vector construction	11584	5369
Total (src/FNs)	59584/38824	5369/10021
Packet Reception		
Independence check*	326	156
H_tests	428	316
Rank of H pkts in $B_u \cup B_w$	292	169
Total	1046	416

gathered on the left of the $y = 2X$ line. Note that the absolute MORE throughputs for many of these points are very low; in particular in the 3-flow and 4-flow cases, throughputs with MORE are as low as only 5Kbps; in contrast, with CCACK, there is only one flow with 41Kbps, and all the remaining flows achieve throughputs higher than 50Kbps. In other words, many flows starve with MORE, as the number of flows in the network increases, and CCACK significantly benefits those flows, with the gains being as high as 21x. In contrast to the single-flow case, these are not necessarily flows with long routing paths, as we saw in the example of Figure 8(b).

On the other hand, there is no clear trend for flows of medium to high throughput with MORE. The gain of many of those flows remains between 20% and 70%, as in the single-flow case, since there is no room for further room for improvement in a congested network. However, in the 3- and 4-flow scenarios, we also observe gains higher than 70% for some flows of medium MORE throughput (200-400Kbps). On the other hand, for many of those flows, and also for flows of higher MORE throughput, throughput is slightly reduced with CCACK – the number of points between the lines $Y = X$ and $Y = 0.7X$ increases with the number of flows. This is because these flows typically maintain high throughput with MORE by causing starvation to some other flows. CCACK's rate control algorithm reduces the throughputs of those flows in favor of the most challenged flows, improving overall fairness in the corresponding scenarios.

4.5 CCACK's overhead

Finally, we estimate CCACK's overhead compared to MORE. Similar to [5], we discuss three types of overhead: coding, memory, and packet header overhead.

Coding overhead. Unavoidably, CCACK's coding overhead is higher than MORE's, since routers have to perform additional operations both when transmitting and when receiving a packet. However, all the additional CCACK operations are performed on N-byte *vectors* instead of the whole K-byte *payload*. Therefore, in practical settings (e.g., with $N = 32$ and $K = 1500$), the coding overhead of CCACK is expected to be comparable to that of MORE.

To verify this, we measured the per-packet cost of the various operations performed upon a packet transmission/reception averaged over all packets transmitted/received at all nodes in the 90 simulation scenarios of Section 4.2. Table 1 provides the average values and the standard deviations. The costs are given in terms of $\mathrm{GF}(2^8)$ multiplications, which are the most expensive operations involved in coding/decoding [5].

Construction of an ACK vector in CCACK requires on average 11584 multiplications. The total coding cost in transmitting a packet (i.e., constructing a coded packet and an ACK vector) in CCACK is only 24% higher than

MORE's, assuming the worst case cost for packet encoding (48000 multiplications). If we use instead the average packet encoding cost at FNs (27240 multiplications), the total cost of transmitting a packet in CCACK is only 38824 multiplications, i.e., lower than MORE's encoding cost at the source.[8]

When receiving a packet, the cost of checking for independence (also in MORE) requires on average only 326 multiplications. The additional operations of performing the H_tests (if the received packet comes from downstream) and maintaining the rank of the H pkts in $B_u \cup B_w$ (if a received packet from downstream passes all M H_tests) require on average only 428 and 292 multiplications, respectively, i.e., their costs are comparable to the independence check cost. The total cost of packet reception operations in CCACK is only 1.7% of the total packet transmission cost. Hence, the bottleneck operation in CCACK is preparing a packet for transmission at an FN with 32 innovative packets in B_v.

In [5], the authors found that the bottleneck operation in MORE (packet encoding at the source) takes on average $270\mu s$ on a low-end Celeron 800MHz, limiting the maximum achievable throughput with MORE to 44Mbps with a 1500 byte packet. In CCACK, the cost of the bottleneck operation is 24% higher, so we can expect a maximum achievable throughput of 35Mbps. Note that this value is still higher than the effective bitrate of current 802.11a/g WMNs [12].

Memory overhead. Same as in MORE, routers in CCACK maintain an innovative packet buffer B_v for each flow, and also a 64KB look up table for reducing the cost of the GF(2^8) multiplications [5]. With a packet size of 1500 bytes, the size of B_v is 48KB. The extra overhead in CCACK comes from the two additional buffers B_u and B_w, which store, however, only 32-byte vectors, and not whole packets. In our implementation, the total size of B_u and B_v is $2 \times 5 \times 32 \times 32 = 10KB$, which is relatively small compared to the size of MORE's structures.

Header overhead. The N-byte ACK vector and the total differential backlog $\Delta Q_{n_j}^{tot}$ are the two fields we add to the MORE header. The differential backlog per flow is bounded by the batch size N. With $N = 32$, two bytes are enough to support up to 2048 flows, and the total size of the two fields is equal to 34 bytes. However, in CCACK, we do not include in the packet header the transmission credits for the FNs, which are required in MORE. This can potentially make CCACK's header smaller than MORE's depending on the number of FNs.

5 Protocol implementation and testbed evaluation

In this section, we describe an implementation of CCACK on a WMN testbed and present experimental results comparing CCACK and MORE.

5.1 Testbed description

Our testbed, Mesh@Purdue (MAP) [25], currently consists of 22 mesh routers (small form factor desktops) deployed on two floors of two academic buildings at Purdue University. A schematic of the testbed is shown in Figure 10. Each router has two radios. For this study, we used one of them: the Atheros 5212 based 802.11a/b/g wireless radio operating in b ad hoc mode. Each radio is attached to a 2dBi rubber duck omnidirectional antenna with a low loss pigtail to provide flexibility in antenna placement. Each mesh router runs Mandrake Linux 10.1 (kernel 2.6.8-12) and the open-source *madwifi* driver [17] is used to enable the wireless cards. IP addresses are statically assigned. The testbed deployment environment is not wireless-friendly, having floor-to-ceiling office walls, as well as laboratories with structures that limit the propagation of wireless signals and create multipath fading.

[8] Note that the source in CCACK does not have to construct an ACK vector, and hence the cost at the source is the same as in MORE.

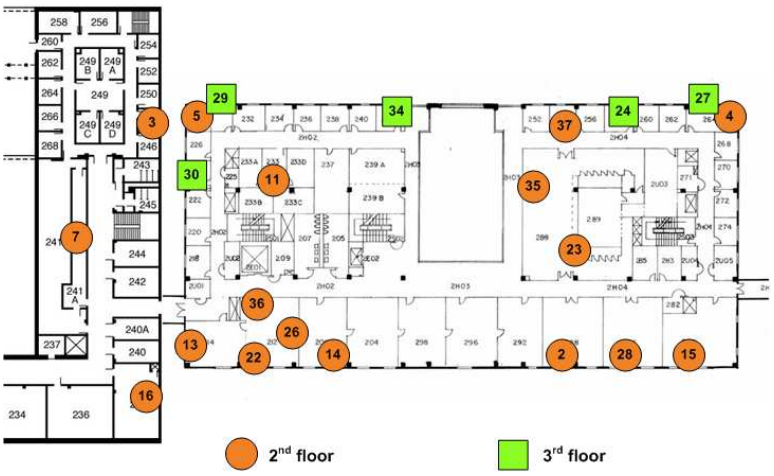


Figure 10. A schematic of MAP.

5.2 Implementation details

NC-based wireless protocols (e.g., [5, 13]) are typically implemented as a shim between the IP and the MAC layer, i.e., at layer 2.5. Here, for ease of debugging, deployment, and evaluation, we implemented CCACK at the application layer, using broadcast sockets. For a fair comparison, we also implemented MORE at the application layer, following all the details in [5]. We note that such an implementation unavoidably results in some performance degradation for both protocols, compared to an implementation closer to the MAC layer, from crossing the kernel-user boundary. Actually, the degradation is larger for CCACK because its credit mechanism is closely coupled with the MAC layer, as we explain later in this section.

Our implementation handles only synthetic traffic, i.e. data packets are generated within the MORE or CCACK application, similarly as the implementation in [27], in which packets are generated within Click. The layer-2.5 header of MORE or CCACK is part of the application layer packet payload. The source initially generates k random payloads for the current batch and mixes them every time it wants to transmit a packet. It then appends the MORE or CCACK header and delivers the resulting packet to the IP layer, which in turn delivers the packet to the MAC for transmission. Packets are broadcast at the MAC layer, and every neighbor node can hear them. When a node receives a packet, it extracts and processes the protocol-specific header from the payload; if the node is an FN (i.e., it finds its ID[9] in the FN list in the header), it also uses the coding coefficients (also included in the header) to check for linear independence. If the received packet is innovative, the rest of the payload is stored for future mixing (if the node is an FN) or for decoding (if the node is a multicast receiver).

5.2.1 Removing the dependence on the MAC layer

In an ideal implementation at layer 2.5, a node running either MORE or CCACK transmits a packet when (1) *the 802.11 MAC allows* and (2) *the credit counter is positive*. In our application layer implementation, we cannot get any feedback from the MAC, and hence, we had to modify the transmission policy for the two protocols.

In our implementation of MORE, the application instead delivers packets to the IP when only the second condition holds and there is enough space in the socket buffer; from the IP layer, the packets are delivered to the wireless driver stored at the card's queue for transmission at a later time. Similar to a layer 2.5 implementation,

[9]To reduce the header overhead, we used 1-byte IDs instead of 4-byte IP addresses.

the credit counter is incremented every time a packet is received from an upstream node, and decremented after every transmission.

Unlike in MORE, the credit counter in CCACK is incremented every time the MAC layer signals a transmission opportunity. Since the application cannot know when there is a transmission opportunity without access to the MAC layer, we approximate the number of transmission opportunities via the following heuristic. A node increments its credit counter every time it hears a data packet transmission from another node by a fraction of $1/N$ of the actual increment determined by the rate control algorithm, where N is the number of nodes in the node's neighborhood. The intuition behind this is that with a fair MAC layer every node in a neighborhood would roughly get an equal number of transmission opportunities. To avoid possible deadlock situations, where every node in a neighborhood is waiting for another node to transmit, we also use a timeout equal to one data packet transmission time, after which a node always increments its credit counter.

5.2.2 Dealing with queue sizes

With a layer-2.5 implementation [5] of an NC-based protocol, a pre-coded packet is always available awaiting for transmission. If another innovative packet is received before the pre-coded packet is transmitted, the pre-coded packet is updated by multiplying the newly received packet with a random number and adding it to the pre-coded packet. This approach ensures that every transmitted packet includes information from all the received innovative packets, including the most recent ones.

In contrast, in our implementation, we have no control over a packet, once it leaves the application layer, and we cannot update the coded packets buffered at the socket buffer or awaiting for transmission at the card's queue, if a new innovative packet is received. This inefficiency can have a significant impact on the performance of the two protocols. If a packet is queued either at the IP or at the MAC layer for a long time, it may not contain information from all the received packets so far. Even worse, the downstream nodes may have already received enough packets from the current batch, in which case the enqueued packets should not be transmitted at all. This is true in particular at the MORE sources which may create packets at a rate faster than the (actual) MAC's transmission rate; in contrast, in CCACK the sources are also paced down by the rate control algorithm. To avoid this problem, we limit the socket buffer size to one packet and the card's queue length to three packets, in order to minimize the time from the moment a packet is created at the application layer till the moment the packet is actually transmitted.

5.2.3 Dealing with end-to-end ACKs

In both protocols, a destination sends an end-to-end ACK back to the source every time it decodes a batch. It is critical for the performance of the protocols that these ACKs are propagated to the source in a fast and reliable way, since, otherwise, the source cannot move to the next batch.

ACK reliability. To provide reliability, the ACKs in MORE are *unicast* at the MAC layer. In contrast to 802.11 broadcast mode, 802.11 unicast mode provides a reliability mechanism through acknowledgments and retransmissions. Unfortunately, there is an upper limit to the number of times a packet can be retransmitted at the MAC layer. For our Atheros wireless cards, this limit is 11. In our experiments, we found that 11 retransmissions were not always enough to deliver the packet to the next hop (especially under heavy traffic). Since this particular card does not allow changing this limit through *iwconfig*, we had to implement an additional ACK-retransmission scheme at the application layer.

Fast ACK propagation. Similar to in [5], ACKs are sent to the source over the shortest ETX path to ensure quick propagation. In addition, in [5], ACKs are prioritized over data transmissions. In addition to ensuring fast ACK propagation, prioritizing ACKs over data packets is critical in our application layer implementation for one more

reason. Since we have no control over a packet once it leaves the application layer, we have to guarantee that an ACK packet will never be dropped if the card's queue is full of data packets.

To implement ACK priority over data packets in our application layer implementation, we leveraged the TOS bits ("TOS filed") of the IP header, which can be set using *setsockopt* at the application layer, and the priority properties in Linux routing [16]. The basic queuing discipline in Linux, *pfifo_fast*, is a three-band first-in, first-out queue. Each band is *txqueuelen* packets long, as configured with *ifconfig*. In our implementation, we set *txqueuelen = 5*, as mentioned in 5.2.2. Packets are enqueued in the three bands based on their TOS bits. The three bands, 0, 1, 2, have different priorities, with band 0 having the highest priority and band 2 having the lowest priority. Packets from a given band are dequeued only when all higher priority bands are empty. By default, the TOS bits are set to 0000 and packets are enqueued in band 1. For ACKs, we set them to 1010. This combination corresponds to "minimum delay + maximum reliability" (or "mr+md") and enqueues the ACKs in the highest priority band 0.

Another factor that caused significant delay to the ACK packets and resulted in very low throughput was the ARP messages. Since ACKs are unicast at the MAC layer, the sender of an ACK first sends an ARP request before the actual transmission of the ACK packet, in order to learn the MAC address that corresponds to the IP address of the next hop. If no reply is received, the ARP request is retransmitted after a timeout (the default is 1 sec). Both the ARP requests and the ARP replies are *broadcast* at the MAC layer. Since 802.11 broadcast implements no reliability mechanism for broadcast frames, ARP messages are susceptible to loss due to poor channel conditions or collisions. Indeed, we observed in our experiments that sometimes ARP requests were retransmitted up to 90 times, which resulted in a 1.5 min delay, before the actual ACK was sent. To deal with this problem, before each experiment, we cached permanently at each node on the shortest ETX path from a receiver to the source, the IP-MAC mapping of the next hop, using the *ip* command, thus completely eliminating the exchange of ARP messages during the experiment.

In addition to the two protocols, we also implemented an ETX measurement module, same as the one we used in our simulations. The source code for the two protocols and the ETX module together is over 7800 lines of C code.

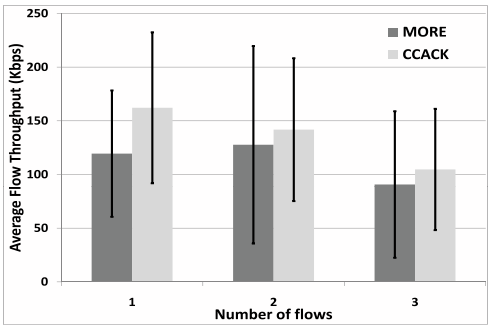
5.3 Experimental setup

In the implementation of the two protocols we used the same parameters as in our simulation study in Section 4. In all the experiments, the bitrate of the wireless cards was set to 2Mbps and the transmission power to 16dBm. We disabled RTS/CTS for unicast frames as most operational networks do. With these settings, the length of the shortest ETX paths between different nodes is 1-5 hops in length, and the loss rates of the links vary from 0% to 91%, with an average value of 36%.

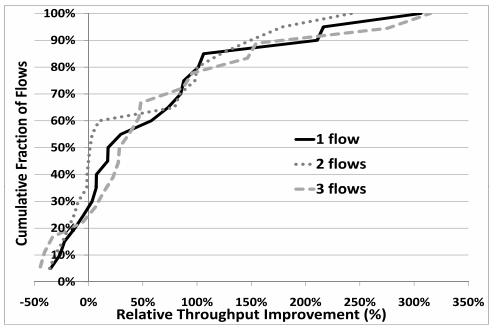
We experimented with 20 single-flow scenarios (i.e., randomly selected source-destination pairs), 10 2-flow scenarios, and 6 3-flow scenarios. For each scenario, we first ran the ETX module to collect the pairwise loss rates and ETX metric for each link of our testbed, and then we ran the two protocols, MORE and CCACK, in sequence. With both protocols, the source sent a 2.3MB file consisting of 1460-byte packets.

As we have explained in Section 4.1, the gain of CCACK over MORE is more pronounced with flows over long paths, where the destination is several hops away from the source. Unfortunately, the size of our testbed limited our choices in flow selection. Hence, in the single-flow experiments described below, we also included flows where the destination was 2 hops away from the source (unlike in Section 4.2, where the minimum source-destination distance was 3 hops). Similarly, the small size of the testbed resulted in a large fraction of the nodes being within sensing range of each other; this prevented us from increasing the total number of flows beyond three, since the medium became congested, resulting in very poor performance for both protocols.[10] These two limitations, along

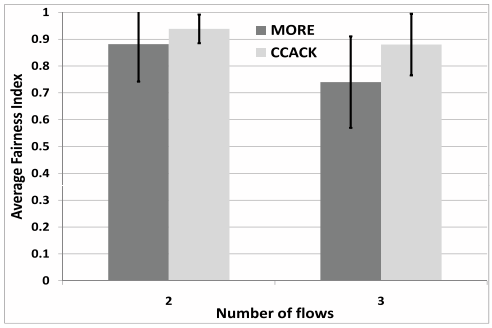
[10] As explained in [5], intra-flow NC based protocols cannot increase the capacity of the network and they can only improve throughput as long as the total load remains below the network capacity.



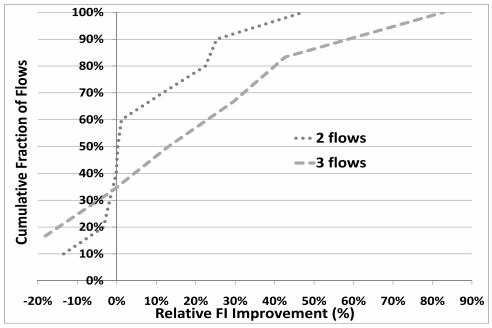
(a) Average per-flow throughputs (bars) and standard deviations (lines).



(b) CDF of relative throughput improvement of CCACK over MORE.



(c) Average per scenario FIs (bars) and standard deviations (lines).



(d) CDF of relative FI improvement of CCACK over MORE.

Figure 11. Testbed evaluation.

with the implementation limitations we discussed in Section 5.2.1, are expected to limit the gains of CCACK over MORE, compared to the simulations results in Section 4.

5.4 Experimental results

The testbed evaluation results are shown in Figures 11(a), 11(b), 11(c), and 11(d).

Figures 11(a), 11(b) compare throughput with CCACK and MORE with 1, 2, and 3 flows. In Figure 11(a), we observe that CCACK outperforms MORE by 36% on average in the 1-flow scenarios, by 11% in the 2-flow scenarios, and by 15% on average in the 3-flow scenarios. Figure 6(b) plots the CDF of per-flow relative throughput improvement with CCACK over MORE, as defined in Section 4.2, with 1, 2, and 3 flows. CCACK improves per-flow throughputs for 72% of the flows in the 1-flow scenarios, 55% of the flows in the 2-flow scenarios, and 75% of the flows in the 3-flow scenarios. The median improvement is 18%, 3%, and 28%, respectively, in the 1-, 2-, and 3-flow scenarios. These gains are lower than the ones observed in the simulation results in Section 4, due to the limitations we discussed in Section 5.3. In spite of these limitations though, our results still demonstrate the benefit of CCACK over MORE in the case of challenged flows. We observe that about 20% of the flows in 1-flow and 2-flow scenarios, and 17% of the flows in the 3-flow scenarios show a several-fold throughput improvement with CCACK, up to 3x, 2.4x, and 3.2x, respectively.

Figures 11(c), 11(d) compare fairness with CCACK and MORE in case of 2, and 3 concurrent flows. Figure 11(c) plots the average FI with the two protocols. We observe that the average FI is higher with CCACK in both the 2-flow, and 3-flow case by 5.7% and 18.9%, respectively. These values are actually higher than the simu-

lation results. Due to the small size of the testbed, the network gets more easily congested, even with 2 flows and CCACK's backpressure-inspired credit mechanism is very effective in allocating the medium's bandwidth fairly among contending flows. Figure 11(d) plots the CDF of per-scenario relative FI improvement with CCACK over MORE. CCACK improves fairness in 60% of the 2-flow scenarios, and 65% of the 3-flow scenarios and the gains can be as high as 83% in some challenged scenarios.

6 Conclusion

The use of random linear NC has significantly simplified the design of opportunistic routing (OR) protocols by removing the need of coordination among forwarding nodes for avoiding duplicate transmissions. However, NC-based OR protocols face a new challenge: *How many coded packets should each forwarder transmit?* To avoid the overhead of feedback exchange, most practical existing NC-based OR protocols compute offline the expected number of transmissions for each forwarder using heuristics based on periodic measurements of the average link loss rates and the ETX metric. Although attractive due to their minimal coordination overhead, these approaches often suffer significant performance degradation in dynamic wireless environments with continuously changing levels of channel gains, interference, and background traffic.

In this paper, we presented a novel approach to NC-based OR through the design of CCACK, a new efficient NC-based OR protocol. Instead of avoiding feedback exchange, CCACK encodes feedback messages in addition to encoding data packets. A novel Cumulative Coded Acknowledgment scheme allows nodes in CCACK to acknowledge network coded traffic to their upstream nodes in a simple and efficient way, oblivious to loss rates, and with practically zero overhead. The cumulative coded acknowledgment scheme in CCACK also enables an efficient credit-based, rate control algorithm. Our experiments on a 22-node 802.11 WMN testbed show that compared to MORE, a state-of-the-art NC-based OR protocol, CCACK improves both throughput and fairness, by up to 3.2x and 83%, respectively, with average improvements of 11-36% and 5.7-8.3%, respectively, for different numbers of concurrent flows. Our extensive simulations show that the gains are actually much higher in large networks, with longer routing paths between sources and destinations.

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