Performance Evaluation of Network-Coded Multicast in Multi-Channel Multi-Radio Wireless Mesh Networks

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***Abstract*—Systems with multiple channels and multiple radios per node have been shown to enhance throughput of wireless mesh networks (WMNs). Recently, network coding has also been proved to be a promising technique for improving network throughput of WMNs. However, the performance of network coding in the context of multicast in multi-channel multi-radio (MCMR) WMNs is still unknown. In this paper, we present a comprehensive performance evaluation of network-coded multicast in MCMR WMNs using extensive simulations, realistic network settings and meaningful performance metrics such as throughput, file completion time, packet end-to-end delay, and packet delivery ratio under a wide range of scenarios.**

I. Introduction

In a wireless mesh network (WMN), wireless routers provide multi-hop wireless connectivity from a host to other hosts either in the same network or in the Internet. The wireless routers are often stationary and form a wireless mesh backbone. Our work in this paper focuses on this mesh backbone, and we will use the terms "routers" and "nodes" interchangeably.

Multicast is a form of communication that delivers information from a source to a group of destinations simultaneously in an efficient manner. The throughput of each multicast source in a random wireless ad hoc network is upper-bounded by $O(1/\sqrt{n^{\epsilon}\log n})$, where $0 \leq \epsilon \leq 1$ and n is the number of nodes in the network [1]. This upper bound indicates that the throughput capacity of multicast in a single-channel WMN becomes unacceptably low as the network size increases. A critical factor that contributes to such a rapid degradation of throughput is the co-channel interference in single-channel networks, worsened by the use of half-duplex radios. A node with a single half-duplex radio is restricted to access one channel at a time, and thus cannot transmit and receive simultaneously. One of the most effective approaches to enhance network throughput is to use systems with multiple channels and multiple radios (MCMR) per node [2]. An MCMR node can transmit on one channel and receive on another at the same time using two different radios. As a result, an MCMR WMN at least doubles the throughput.

Studies on multicast in MCMR networks have been addressed only recently [3], [4], [5], [6], [7]. These algorithms aim at optimizing performance by minimizing the interference among multicast nodes [3], [4], [5], [6] or the number of transmissions incurred in multicast trees [7].

Also recently, network coding [8] has received much attention as a promising technique for improving network throughput. With network coding, a node can combine multiple packets within a single transmission, thus making more efficient use of network bandwidth. Previous studies on the benefits of network coding in wireless multi-hop networks focus mostly on single-channel networks and provide only theoretical bounds [9], [10], [11], making assumptions about a particular coding structure [9] or an unrealistic slotted MAC algorithm [10], or considering only a simple traffic pattern or network topology [11]. The performance of network coding in the context of multicast communication in MCMR systems is still unknown.

This suggests that quantifying the average gain of network coding for multicast in MCMR WMNs under realistic network settings remains an important open issue. Quantifying the practical gain of network-coded multicast not only guides the design of high-performance coding and multicast routing protocols, but also justifies the significant efforts being invested by the research community in exploring this new technology. In this paper, we present a comprehensive simulation-based performance evaluation of network-coded multicast in MCMR WMNs, using realistic network scenarios and useful performance metrics such as throughput, file completion time, packet end-to-end delay, and packet delivery ratio. To the best of our knowledge, our work is the first that studies the performance of multicast with network coding in MCMR WMNs.

The remainder of the paper is organized as follows. A summary of related work is provided in Section II. We describe the system model and our experiment setting in Sections III and IV, respectively. We present our simulation results in Section V and conclude the paper in Section VI.

II. Related Work

Performance analyses of network coding in wireless networks [9], [10], [11], [12], [13], [14], [15], [16], [17] have gained much attention recently. Random linear coding of packets in a multicast flow was first introduced in [12], which provides a lower bound on the probability that all one-hop multicast receivers are able to successfully decode the data sent by the source and shows that this scheme can outperform a traditional store-and-forward routing mechanism. In [13], the authors consider a network with finite queue buffers for storing packets and propose a scheme for coding packets of a single unicast flow that arrive through a random process. They provide a framework that allows for the computations of the delay and queue blocking probability. The work in [14] provides analytical bounds on completion time and stable throughput for random linear coding across multiple multicast flows. In [15], Eryilmaz et al. quantify the performance gains of network coding in terms of completion time from a single source to one-hop receivers with varying channel conditions, modeled as stochastic changes in ON/OFF state.

In [16], Katti et al. propose COPE, a practical protocol that allow nodes to combine packets together by exploiting the wireless broadcast advantage through opportunistic listening, and show its coding gain over a non-coding scheme in a wireless testbed. Subsequent studies [9], [10], [11] indicate that the coding gain highly depends on network topology, traffic load and traffic pattern. In [17], Koutsonikolas et al. perform a simulation-based study of practical coding gains of unicast flows in single-channel WMNs.

None of the work above, however, considers the performance of multicast flows combined with network coding in MCMR wireless environments. In addition, existing studies consider only the throughput, one-hop latency and completion time metrics. In this paper, we present a comprehensive performance evaluation of network-coded multicast in MCMR WMNs using a wide range of useful metrics including throughput, file completion time, packet end-to-end delay, and packet delivery ratio.

III. System Model and Assumptions

We use the following system model and assumptions in our experiments.

A. Multicast Communication Model

We assume that each multicast source or destination is associated with a different wireless mesh router. That is, a multicast group with d destinations consists of d distinct destination routers and one source router, since we are interested in the multicast performance of routers in the mesh backbone.

Multicast packets are delivered to multicast destinations in a multi-hop manner by following the multicast structure constructed by a multicast routing protocol [18], [19], [20]. Although the operations of different multicast protocols vary from one protocol to another, in general, the multicast structure of a multicast protocol is often defined by its *forwarding set*, a group of nodes designated to forward multicast packets. A multicast source by default is part of this set. Similar to the Internet multicast model, each multicast group is identified by the source ID s and group ID G. Every node maintains a routing entry of the form $[(s, G), f]$, where f is a boolean variable called *forwarding flag*. Nodes that belong to the forwarding set of the multicast structure have their forwarding flags set to true and, upon receiving non-duplicate multicast packets for the multicast group (s, G), re-broadcast them.

B. The 802.11 MAC Model for Multicast

The medium access control (MAC) for multicast uses the basic access procedure of the IEEE 802.11 distributed coordination function (DCF) with carrier sense multiple access and collision avoidance (CSMA/CA) without RTS (request-to-send), CTS (clear-to-send) and ACK (acknowledgment) [21]. The 802.11 standard currently does not implement the RTS/CTS/ACK mechanism for multicast due to the following reasons. First, multiple CTS/ACK packets concurrently sent by multicast receivers of a transmitter have a very high probability of colliding at the transmitter. More importantly, it may not be possible for all the multicast receivers to agree on a common time slot for the transmission of a packet, and the delay would be very long to either reach a transmission time agreement or receive ACKs from all the receivers. Although there exist some works in the literature that propose RTS/CTS mechanisms for multicast, they either incur very long delay (e.g., by polling multicast receivers one by one) [22], [23] or require extensive modifications to the 802.11 MAC protocol [24], [25], [26].

C. Network Coding Model

We use the following *intra-flow* random linear coding [27] to combine packets within a single multicast flow.

1) Multicast Source: When a multicast source with network coding function has a file to deliver to its multicast group, it breaks up the file into a set of ω batches, each having K packets. These K uncoded packets are called *native packets* and K is called the *batch size*. When the source is ready to send, it creates a random linear combination of the K native packets and broadcasts the coded packet. A coded packet x' is computed as:

$$x' = \sum_{i=1}^{K} c_i x_i,$$

where c_i values are random coefficients chosen from a finite field of size q, and x_i entities are native packets.

2) Multicast Forwarders: Nodes listen to all transmissions within its sensing range. When a node overhears a multicast packet, it checks whether it is a multicast forwarder (by looking up the routing entry of the multicast group for the forwarding flag value). If so, the node checks whether the packet is an *innovative packet*. A packet is innovative if it is linearly independent from the packets the node has previously received. Checking for linear independence can be done using Gaussian elimination [28]. The node keeps innovative packets and drops non-innovative ones.

Upon receiving an innovative packet, the forwarding node creates a new coded packet by generating a random linear combination of the innovative coded packets it has received, and broadcasts it. Note that a linear combination of coded packets is also a linear combination of the corresponding native packets. In particular, suppose that the forwarder has received m coded packets, each in the form of:

$$x'_j = \sum_{i=1}^{K} c_{ji} x_i,$$

where x_i is a native packet. The forwarder then linearly combines these coded packets to create a new coded packet as follows:

$$x'' = \sum_{j=1}^{m} a_j x'_j,$$

where a_j's are new random coefficients. The resulting coded packet x'' can be expressed in terms of the native packets as follows:

$$x'' = \sum_{j=1}^{m} a_j (\sum_{i=1}^{K} c_{ji} x_i) = \sum_{i=1}^{K} (\sum_{j=1}^{m} a_j c_{ji}) x_i,$$

thus, it is also a linear combination of the native packets.

3) Multicast Destinations: Upon receiving a packet, a multicast destination checks whether the packet is innovative and discards the packet if it is not. Once the destination receives K innovative packets, it can decode the batch and obtain the native packets using a simple matrix inversion:

$$\begin{pmatrix} x_1 \\ \vdots \\ x_K \end{pmatrix} = \begin{pmatrix} c_{11} & \cdots & c_{1K} \\ \vdots & \ddots & \vdots \\ c_{K1} & \cdots & c_{KK} \end{pmatrix}^{-1} \begin{pmatrix} x'_1 \\ \vdots \\ x'_K \end{pmatrix},$$

where x'_i is a coded packet whose coefficients are $c_{i1}, ..., c_{iK}$.

D. Multi-Channel Multi-Radio Systems

We consider multi-channel WMNs with multiple radios per node. Two nodes u and v are directly connected and form a communication link (u, v) if they are within the transmission range of each other and share a common channel. Each node is equipped with the same number of radios and the network has C orthogonal (non-overlapping) channels. In our simulations, a channel assignment algorithm such as [29], [30], [31], [32], [33], [34], [35] is first applied to the network. We then build a multicast routing tree using each of the following algorithms: shortest path trees based on Dijkstra's algorithm [36], the Steiner tree heuristic in [37] and the minimum data overhead multicast routing algorithm by Ruiz et al. [18].

IV. Experiment Setting

We refer to a multicast session without network coding as **Re**gular **M**ulticast (ReM) and a multicast session in combination with network coding as **Net**work-**Co**ded **M**ulticast (NetCoM). Using Qualnet [38], a software that provides scalable simulations of wireless networks, we simulate networks of static nodes uniformly distributed in areas of different sizes. The channel bandwidth at the physical layer is 11 Mbits/s. The transmission range of the wireless routers is 315m, according to the specifications of wireless routers manufactured by Tropos [39]. The path loss propagation model is two-ray and there is no channel fading.

We follow the medium access control (MAC) for multicast as defined by IEEE 802.11 standards, which is IEEE 802.11 DCF CSMA/CA protocol without RTS/CTS/ACK exchange [21]. At the MAC and transport layers, multicast packets are neither acknowledged nor retransmitted if being lost or damaged. At the transport layer, we do not use any flow or congestion control mechanisms in order to test the network performance under very high loads. The multicast group has one source placed at the center of the network, while the destinations are randomly selected. All destinations join the multicast group at the beginning and stay until the end of the simulation. The underlying routing algorithm is hop-count based, shortest path trees, built by applying Dijkstra's algorithm [36] to each source-destination pair.

In all experiments, the ReM and NetCoM multicast sources each sends a file of size 12 megabytes at a specified constant bit rate (CBR) at the application layer. The (native) data packet size, excluding header size, is 512 bytes. The queuing policy at routers is first-in-first-out. Random coefficients for each linear combination in NetCoM are chosen from a Galois field of size $q = 2^8$, as used in [27], [40]. Each data point in the graphs is averaged from 50 runs using different network topologies and random seeds, and plotted with a confidence interval of 95%.

We consider the following performance metrics:

- *Average throughput*. The throughput of a multicast destination is defined as the total number of native packets the destination receives divided by the interval starting from the time the multicast source begins transmitting the first packet to the time the destination receives its last packet. The average taken over the throughputs of all multicast destinations is the average throughput of the group.
- *Average file completion time*. The completion time of a file received at a multicast destination is defined as the time it takes the destination to finish the reception of the file. The average file completion time of the multicast group is the average of the file completion times of all destinations.
- *Average packet end-to-end delay*. The end-to-end delay (EED) of a (native) packet received at a multicast destination is defined as the latency between the time the packet is transmitted from the multicast source and the time the packet is received at the destination. The average EED of ReM is the average of the EEDs of all the (native) packets received at all multicast destinations. For NetCoM, the EED of a native packet received at a multicast destination is effectively the EED of its associated batch, measured from the time the source transmits the first coded packet of the batch to the time the batch is successfully decoded at the destination. The average EED of NetCoM is the average of the EEDs of all the native packets (or, effectively, of all the batches) received at all multicast destinations.
- *Average packet delivery ratio*. The packet delivery ratio (PDR) of a multicast destination is the ratio of the number of native packets received by the destination divided by the number of native packets the multicast source has sent. The average PDR of a multicast group is the average of the PDRs of all multicast destinations in the group.

We conducted four sets of experiments and measured the above performance metrics as functions of

Parameter	Set 1	Set 2	Set 3	Set 4
Network size (nodes)	50	50	50	25, 50, 75, 100
Number of channels	1-7	3	3	3
Group size (destinations)	30	30	10-40	15, 30, 45, 60
Source rate (packets/s)	250	100-350	250	250

TABLE I: Experiment parameters

1) *number of orthogonal (non-overlapping) channels*. We increase the number of channels from one to seven. The number of radios per node is one for one channel, two for two channels, and three for three to seven channels.
2) *multicast source rate*. ReM and NetCoM source rates at the application layer increase from 100 to 350 packets/s.
3) *multicast group size*. The number of multicast destinations varies from 10 to 40.
4) *network size*. We vary the network size while maintaining the same node density of about 35 nodes/km^2. Specifically, we created four networks: 25 nodes in a 850m $\times$ 850m area, 50 nodes in a 1200m $\times$ 1200m area, 75 nodes in a 1450m $\times$ 1450m area, and 100 nodes in a 1700m $\times$ 1700m area. The multicast group size is 60% of the network size. In particular, there are 15, 30, 45, and 60 multicast destinations in the 25-node, 50-node, 75-node, and 100-node networks, respectively.

Additional simulation parameters are set as follows. In all the experiments,

- the network size is 50 nodes (except in set 4 as stated above), distributed uniformly over an area of size 1200m $\times$ 1200m.
- the number of orthogonal channels is three and the number of radios per node is also set at three (except in set 1 as stated above).
- each multicast group has 30 destinations selected randomly (except in sets 3 and 4).
- the multicast source rate at the application layer is 250 packets/s (except in set 2).
- the NetCoM batch size is 32, a common batch size used in network coding experiments [27], [40].

Table I summarizes the set of experiments and their respective simulation parameters. For each scenario, we ran ReM and NetCoM *separately* using the same experiment configuration.

V. Experimental Results

The results are illustrated by the graphs in Fig. 1 to Fig. 4. We observe the following facts common to all the experiments.

First, in all cases the average file completion time of NetCoM is shorter than that of ReM, by about 2-3 % (Figs. 1(d) to 4(d)). The completion time of ReM does not vary much when the parameters change (except in set 2), and varies around the 100 seconds mark, which is the time for a source to complete the transmission of the 12Mbytes file at the application layer plus a small end-to-end delay (See Figs. 1(d), 3(d) and 4(d). Figs. 1(e), 3(e) and 4(e) show magnifications of ReM completion time). The completion time of NetCoM varies by only 1% to 2% (except in set 2). This suggests that file completion time is not affected much by the number of channels, group size and network size.

Second, network coding offers higher throughputs than regular multicast in most cases (Fig. 1(a) to Fig. 4(a)). The throughput gain is about 25-30 %, much lower than the gains reported in most previous papers on network coding in wireless networks, which assume very dense networks. However, this throughput gain is consistent with the results in [17], which use realistic network settings as we do in this paper. (Paper [17] focuses on unicast performance in single-channel networks.) Furthermore, the throughput of network-coded flows can even be lower than that of regular flows in some cases (Figs. 3(a) and 4(a)), as also reported in [17].

The throughput gain of NetCoM, if achieved, comes at the expense of much longer packet end-to-end delay (Fig. 1(c) to Fig. 4(c)). This third fact has not been reported in any existing work, and can be explained as follows. NetCoM forwarding nodes require additional time for coding the received packets before transmitting. Moreover, upon receiving a coded packet, a NetCoM destination may not be able to decode it right away. It has to wait to receive at least K innovative packets, where K is the batch size, before decoding them. The delay to obtain the native packets is thus longer compared with ReM.

We now discuss the results specific to each of the four experiment sets described above.

A. Varying the Number of Orthogonal Channels

The graphs in Fig. 1(a) show that when the number of channels used in the network increases, the average throughputs of ReM and NetCoM both increase, as expected, since using more non-overlapping channels reduces interference, packet collision and contention time at the MAC layer, resulting in smaller loss rates, lower delay, and consequently higher throughput. In addition, as the number of channels increases from 1 to 4, the throughput of NetCoM is higher than that of ReM by 10-50 %. However, as the number of channels increases further from 5 to 7, while the number of radios per node remains at 3, the throughputs of ReM and NetCoM are very similar. This is the result of a problem specific to network coding in MCMR networks, which we term "overhearing degradation". Network-coding forwarders and destinations exploit the wireless broadcast advantage to overhear as many packets as possible in order to collect innovative packets. (A destination needs K innovative packets in order to decode a batch.) Having a large number of channels (compared with the number of radios per node) degrades the overhearing quality because coded packets are now scattered over many channels. Therefore, NetCoM has little or no gain over ReM when the number of channels is large, above four in this experiment.

Similarly, as the number of channels increases, the PDRs of REM increase accordingly, from 50% to 90%, thanks to MCMR reducing interference, contention and collisions (Fig. 1(b)). The PDRs of NetCoM also increase (thanks to MCMR) and are higher than those of ReM (thanks to network coding advantage), but only up to four channels. As the number of channels increases from five to seven, the PDRs of NetCoM do not improve and are lower than those of ReM (85% vs. 90% of ReM) due to the ‘overhearing degradation” problem discussed above.

The graphs in Fig. 1(c) show that the higher the number of orthogonal channels are available in the network, the lower the EEDs for both ReM and NetCoM, as expected. Adding more channels reduces contention among nodes within an interference area, and thus lowers the contention delay that packets spent at intermediate nodes.

We notice that the improvement of NetCoM over ReM from 4-7 channels is not as dramatic as that from 1-3 channels. When there are 1-3 channels, the number of radios per node matches the number of channels, maximizing the degree of parallelism. In the case of 4-7 channels, the number of radios is fixed at three and less than the number of available channels, reducing the degree of parallelism and thus the NetCoM gains.

B. Varying the Multicast Source Rate

The PDRs of both ReM and NetCoM decrease as the multicast source rate increases, as expected (Fig. 2(b)). As the source rate goes up, more packets are transmitted during a period of time, increasing the collision probability among packets in an interference region and thus leading to more packet losses. A higher source rate also causes more packets to be dropped by forwarding nodes due to full queues.

It is important to note that when the rates are low (100-150 packets/s), NetCoM PDRs are lower than ReM PDRs, by about 5%. One ReM packet lost on the way implies one native packet lost at a destination. However, one *coded packet* lost may prevent the whole batch from being decoded, causing the loss of K native packets at a destination, where K is the batch size. When the traffic load in the network is low, the main source of packet loss in both ReM and NetCoM is channel errors (and possibly occasional collisions due to random backoff of IEEE 802.11 MAC). Given the same loss rate, NetCoM destinations will observe more native packet losses than ReM destinations, resulting in lower NetCoM PDRs.

Nonetheless, when the source rates are high (above 200 packets/s), NetCoM offers higher PDRs than ReM, by 10% to 15%. Under high traffic loads, the queues at forwarding nodes and destinations will become full, causing packets to be dropped. NetCoM, thanks to the capability of combining multiple packets, allows the queues to clear faster, resulting in less packets to be dropped and thus higher PDRs.

When the multicast source rate increases, the average throughputs of both ReM and NetCoM increase (Fig. 2(a)). A higher source rate implies that more data is transmitted over a period of time, and thus more is received per second, leading to a higher throughput. At high traffic loads of over 200 packets/s, NetCoM has higher throughputs than ReM, by 20-40 %, thanks to better PDRs.

In terms of packet end-to-end delay (Fig. 2(c)), we observe the following two facts. First, the EED of NetCoM is always higher than that of ReM, the reason for which was explained earlier. Second, as the source rate increases, the EEDs of both ReM and NetCoM increase. A higher packet arrival rate leads to longer queues at forwarding nodes, and thus longer waiting and processing time. This results in longer end-to-end delays.

Very high source rates (250-350 packets/s) reduce the completion time of both ReM and NetCoM (Figs. 2(d) and 2(e)). These rates cause a large number of packet losses (Fig. 2(b)). It thus takes a destination less time to receive a file that is only 60-70% of the original size, hence shorter completion time.

C. Varying the Multicast Group Size

We observe a striking difference between ReM and NetCoM performance variations. As the group size increases, the performance of ReM degrades, while the performance of NetCoM improves (except for the completion time), as shown in Fig. 3. For ReM, as the group size increases, the multicast routing trees become more dense, creating more interference and contention among forwarding nodes, and thus negatively impacting the EED and throughput. More multicast nodes also create more traffic and thus cause more collisions and packet drops from full queues, lowering the PDR.

Although more contention and collisions also affect NetCoM, their impacts are negated by the “opportunistic listening” property [16] of network coding: as the number of nodes involved in the coding process increases (i.e., the multicast tree becomes more dense), a NetCoM node collects innovative packets from overhearing its neighbors *faster*. This leads to lower EED (Fig. 3(c)), and allows the queues at multicast nodes to clear faster, resulting in higher throughput and PDRs (Figs. 3(a) and 3(b)), as the group size increases. This implies that the gain of NetCoM over ReM is significant in dense networks, in agreement with previous studies [16], [17].

D. Varying the Network Size

In this set of experiments, the number of multicast destinations in each network is 60% of and increases with the network size. As a result, we observe similar trends as in the above group-size experiments. Specifically, the performance of ReM degrades as the network size increases (Fig. 4). A larger network implies longer source-to-destination paths, which increases ReM EED (Fig. 4(c)). Longer path lengths also mean a higher probability that a packet will be lost or damaged due to channel errors or collisions. Thus, the PDR of ReM goes down from 98% to 65% (Fig. 4(b)), leading to a throughput decrease from 250 packets/s to 160 packets/s (Fig. 4(a)), as the network becomes larger.

Although longer path lengths also affect the PDRs of NetCoM, their impacts are negated by a higher number of overhearing opportunities in a larger network, as explained above. This lowers NetCoM EED and improves the PDR in larger networks. The PDR of NetCoM increases from 72% to 87%, leading to a throughput improvement of 15%, from 200 packets/s to 230 packets/s.

To further confirm the validity of the above four sets of results, we repeated the experiments using other multicast routing algorithms such as the Steiner tree heuristic in [37] and the minimum data overhead multicast routing algorithm by Ruiz et al. [18] (results are not shown here due to space limitation). The results and observations from these two algorithms are consistent with those from the above shortest path tree experiments.

VI. Conclusion

We present a comprehensive performance evaluation of multicast flows with and without network coding in MCMR WMNs. The results confirm findings from previous studies that network coding improves the performance of multicast flows in large and dense networks. On the other hand, our paper also presents new findings that have not been reported in literature. First, the use of multiple channels and multiple radios helps improve the performance of NetCoM as it is supposed to. However, a large number of channels will lead to the "overhearing degradation" problem and no longer benefit NetCoM. Second, the packet EED of NetCoM is significantly higher than that of regular multicast due to additional time incurred by the coding at forwarding nodes and the destinations waiting to collect enough innovative packets before decoding. Third, destinations' waiting time to collect innovative packets is a major source of overhead that increases the packet EED and completion time. We can shorten this waiting time by increasing the group density and/or network density.

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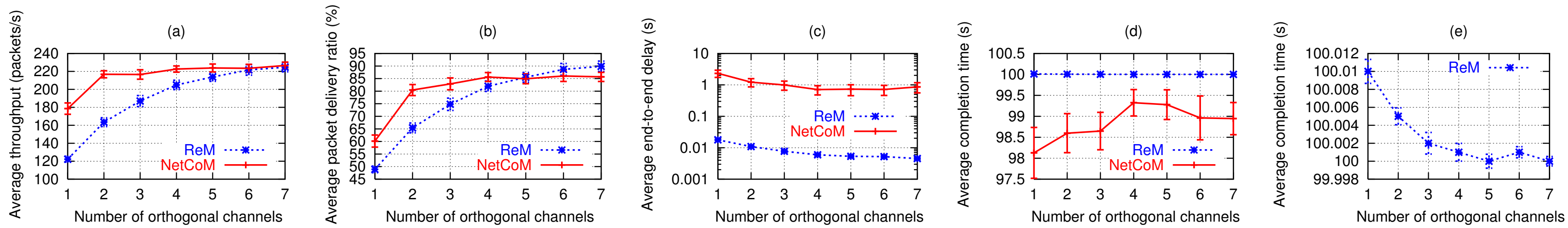

Fig. 1: Functions of number of orthogonal channels

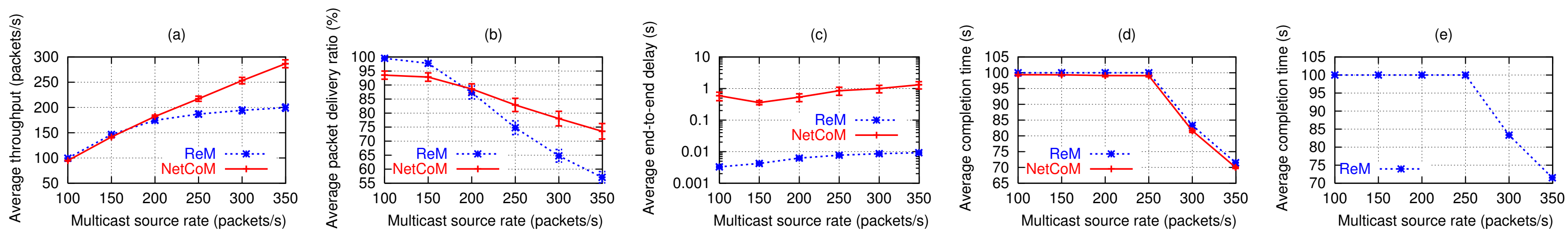

Fig. 2: Functions of multicast source rate

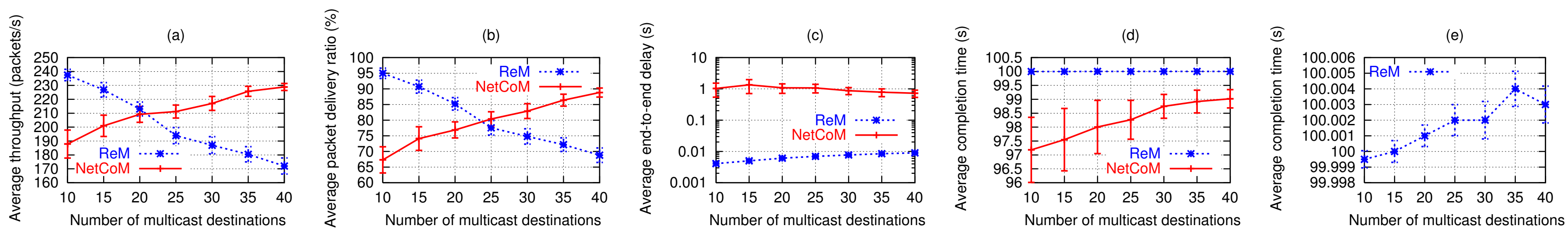

Fig. 3: Functions of multicast group size

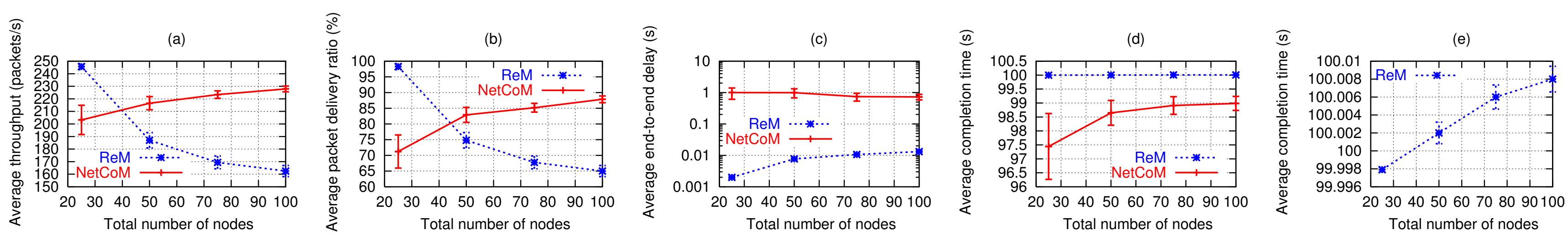

Fig. 4: Functions of network size