Multirate-aware Multicast Routing in MANETs

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***Abstract*— We propose a Rate-Adaptive Multicast (RAM) routing protocol that is multirate-aware. During the process of path discovery, the quality of wireless links is estimated to suggest optimal transmission rates, which are then used to calculate the total transmission time incurred by the mobile nodes on a path. Among several considered paths from a source to a destination, RAM selects the path with the lowest total transmission time. Our simulation results show that RAM outperforms single-rate multicast in terms of packet delivery ratio, packet end-to-end delay, and throughput of the multicast group. We apply RAM to effectively support video multicast in mobile ad-hoc networks.**

I. Introduction

A current trend in wireless communications is to enable wireless devices to transmit at different rates *at the physical layer*. Many standards supporting this capability have been made available, such as 802.11a, 802.11b, 802.11g, and HiperLAN2. For example, 802.11b specifies rates of 1Mbps, 2Mbps, 5.5Mbps and 11Mbps. Rate adaptation is the process of dynamically switching data rates to match the channel conditions in order to obtain the optimal throughput.

The multirate capability is motivated by the fact that consumers desire both high rates and long transmission ranges. However, long range transmissions must operate at low rates. Otherwise, the error rate would be very high. Conversely, high rate transmissions must occur over short distances to maintain a tolerable bit error rate. The multirate capability offers consumers the flexibility of choosing between high speeds and long ranges. Devices and standards have been made to support this tradeoff. In addition, several rate-adaptation protocols have been proposed for wireless local area networks (WLANs) [1]–[4], as well as for mobile ad-hoc networks (MANETs) [5]–[8]. The proposed protocols, however, were designed for unicast (point-to-point) communications. Our work presented in this paper aims at exploiting the multirate capability for use with *multicast* in MANETs.

Multicast refers to one-to-many or many-to-many communications. Important multicast applications include audio/video conferencing, distance education, interactive games, and distribution of financial data, software and newspapers. Wireless ad-hoc networks are well suited for multicast due to their inherent broadcast capability. Many multicast routing protocols have been proposed specifically for MANETs [9], [10].

Currently most multicast routing protocols tend to select the shortest path from a sender to a receiver. Such a path contains a minimum number of hops between the source and the destination. Therefore, the distances between hops are long, and the transmission rates are kept low. A low transmission rate incurs long transmission time, which results in low throughput and high energy consumption.

We expect that another route selection criterion, combined with multi-rate transmission, can improve the situation. For example, we could choose paths that are slightly longer than the shortest paths (i.e., having more hops) but on which the distances between neighboring nodes are shorter. Shorter distances between multicast nodes permit data transmissions at higher rates, thereby, lowering transmission time, increasing network throughput and reducing the energy consumption.

By the above reasoning, it seems that long paths offer better performance. However, that may not always be the case because more hops are needed to reach the destination. Each hop incurs delay due to contention for accessing the wireless channels and also due to congestion, if any. Finding an optimal path for the best performance is not an easy task especially in a large network. It is also not practical or productive in a MANET because as soon as such a path is found, mobile nodes can move leaving that path no longer optimal.

Instead of taking into account all factors that could affect the performance of a route, we consider a relatively simple routing metric. Among the available paths between a sender and a receiver, the routing protocol selects the path with the lowest total transmission time. As previously mentioned, low transmission time helps to increase throughput and reduce energy consumption. Our experimental results show that even with such a relatively simple route selection criterion significant improvements are achieved over single-rate multicast.

Specifically, we propose a Rate-Adaptive Multicast (RAM) routing protocol based on the on-demand multicast routing protocol (ODMRP) [10]. ODMRP is a mesh-based multicast routing protocol that is simple, yet offers high packet delivery ratios and adapts well to high mobility of nodes. RAM is a multirate-aware routing protocol, which works as follows. During the process of path discovery, the quality of wireless links is estimated to suggest optimal transmission rates, which are then used to calculate the total transmission time incurred by the mobile nodes on a path. Among several considered paths from a source to a destination, the routing protocol selects the path with the lowest total transmission time. Experimental results show that RAM outperforms single-rate multicast in almost all scenarios, in terms of packet delivery ratio, packet end-to-end delay, and throughput of the multicast group. We apply RAM to implement a framework for video multicast in MANETs.

To the best of our knowledge, our work is the first that

proposes the use of the multirate capability in multicast. The proposed RAM protocol does not require any modifications to the standards and works with any multirate standards. The paper presents RAM in the context of 802.11b.

The remainder of the paper is organized as follows. Related work is presented in Section II. We provide an overview of ODMRP in Section III. Design issues of the multirate-aware multicast routing are discussed in Section IV, while the proposed RAM protocol is described in Section V. Section VI provides experimental results to illustrate the effectiveness and quantify the performance of the RAM protocol. RAM is applied to a video multicast framework in MANETs. In Section VII, we present the design, implementation and performance evaluation of the proposed video multicast framework. Finally, concluding remarks are made in Section VIII.

II. Related Work

Rate adaption involves two stages: channel quality estimation and rate selection. Several metrics are used as indicators of channel quality such as signal-to-noise ratio, signal strength, symbol error rate, or bit error rate. The rate selection procedure then uses the channel quality estimate to select an appropriate rate. A common technique of rate selection is to compare the value of the channel quality indicator against a list of threshold values representing boundaries between the data rates [11].

To estimate the channel quality, the Autorate Fallback (ARF) rate adaptation scheme, implemented in Lucent's 802.11 WaveLAN-II networking devices, uses the receipts or absence of acknowledgments (ACKs) as an indication of channel quality [1]. The Receiver-Based Autorate (RBAR) protocol [2], on the other hand, uses the strength of received signals. The latter method provides faster and more accurate estimate than the use of ACKs. The protocol proposed by Pavon and Choi for WLANs [3] combines the use of signal strength and the frame error rate to determine the transmission rate. All of the aforementioned protocols were originally designed for WLANs and can also be applied to MANETs [5]. However, these rate-adaptation protocols operate independently of the routing protocol. Therefore, if the routing protocol finds a short path with long hops, the rate-adaptation protocol may not be able to use high rates. The RAM protocol we propose incorporates rate selections into the routing protocol for optimal performance.

The Opportunistic Media Access (OMA) protocol [6] uses RBAR for channel quality estimation and rate selection. Once the rate is calculated and the channel is reserved, the source will send for a duration as if the base rate were used. That is, a flow with a data rate of 11 Mbps will transmit 5.5 times as many packets as a flow with a data rate of 2 Mbps. This scheme aims at achieving temporal fairness, as opposed to throughput fairness in RBAR.

The rate-adaptation schemes proposed by Qiao *et al.* for WLANs [4] and Awerbuch *et al.* for MANETS [7] are based on theoretical models of the attainable throughput in WLANs/MANETS. The models permit us to compute link rates that maximize the throughput. The models, however, make simplifying assumptions (e.g., all nodes in the network are within each other's transmission range [7]), and the rate computation assumes that the packet size is known in advance and that all packets have the same size.

A multirate-aware routing protocol for MANETs was proposed by Seok *et al.* [8]. The protocol requires a new sub-layer called Multirate Aware Sub-layer (MAS) to be added between the network layer and the MAC layer. The route discovery procedure still searches for the shortest path. When the data packets are actually routed, the MAS layer will attempt to find a two-hop sub-path between every pairs of nodes on the shortest path that can offer transmission time shorter than that of the one-hop link. The major drawback of this protocol is the addition of the MAS layer, which requires considerable modifications to the standards.

The RAM protocol we propose is the first multirate-aware routing protocol for multicast. It requires no modifications to the standards and offers significant improvement over single-rate multicast as proved by our simulation results.

III. Overview of ODMRP

ODMRP sources periodically update routing tables and membership information by flooding the network with route refresh packets called `JOIN_QUERY`. When a node receives a non-duplicate `JOIN_QUERY`, it stores the upstream node ID (i.e., backward learning) and rebroadcasts the packet. When the `JOIN_QUERY` packet reaches a multicast receiver, the receiver creates a `JOIN_REPLY` packet and broadcasts it to its neighbors. When a node receives a `JOIN_REPLY`, it checks if the next node ID of one of the entries matches its own ID. If it does, the node realizes that it is on the path to the source and thus is part of the forwarding group. It then broadcasts its own `JOIN_REPLY` built upon the matched entries. The `JOIN_REPLY` packet is thus propagated by each forwarding group member until it reaches the multicast source via the shortest path. This process constructs (or updates) the routes from sources to receivers and builds a mesh of nodes, referred to as the forwarding group.

Figure 1 depicts an example multicast session in which receiver R_1 receives data from both sources S_1 and S_2, while receiver R_2 chooses to receive only from source S_1, and receiver R_3, only from source S_2. The arrows between the mobile nodes indicate the `JOIN_REPLY` packets sent. Figure 1(a) shows the routing tables at the mobile nodes as created by ODMRP.

IV. Design Issues of Multirate-aware Multicast Routing

Besides the routing function, two major issues to be considered in the design of a multirate-aware multicast routing protocol are the mechanism for channel quality estimation and the routing metric. These are described next.

A. Channel Quality Estimation

In unicast communications, the transmission of a data packet is preceded by an RTS/CTS (Request To Send/Clear To Send)

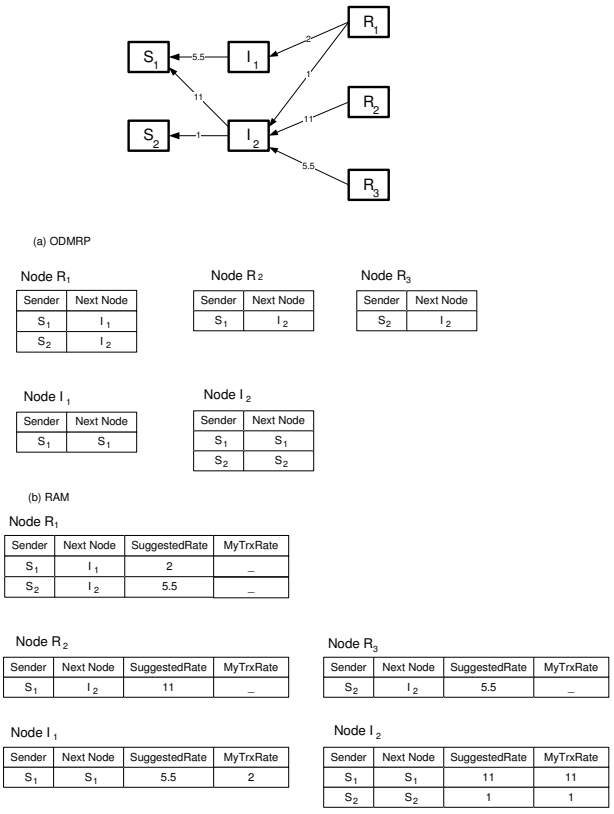


Fig. 1. An example multicast session. The arrows indicate the JOIN_REPLY flows, where the number associated with each arrow denotes the transmission rate of the link in the opposite direction. The routing tables in the JOIN_REPLY packets are the same but without column $MyTrxRate$. In the routing tables of nodes R_1, R_2 and R_3, fields $MyTrxRate$ are left empty since these nodes only receive and do not transmit any data packets.

exchange. The receiving node estimates the channel quality based on the signal strength of the received RTS and sends the rate information to the sending node via the CTS. Since there is no RTS/CTS exchange in multicast, we propose to use the control packets of the routing protocol for estimating the channel quality. Specifically for ODMRP, channel conditions are estimated based on the signal strength of the received JOIN_QUERY packets. The receiving node then sends the rate information to the sending node using the JOIN_REPLY packets. The rate information is thus updated with each route refreshment. This scheme allows the control packets to be used in an efficient manner, for both routing and rate adaptation.

B. Routing Metric

Awerbuch *et al.* proposed to use the *medium time metric* (MTM) for multirate-aware routing for unicast in MANETs [7]. The MTM assigns a weight to each link that is proportional to the amount of medium time used for sending a packet on that link. The medium time includes the time spent backing off during contention, plus the overhead and the transmission time of a full RTS/CTS/DATA/ACK (acknowledgment) exchange. Assuming a network with full interference (i.e., all the nodes are within each other's transmission range), the authors show that a routing protocol that chooses a path having the smallest weight maximizes the network throughput. In practice, the assumption of full interference does not often hold true, especially in large MANETs. As such, the MTM may not deliver the optimal throughput as is assumed. However, simulation results have shown that the use of MTM as a routing metric, rather than the shortest-path metric, indeed improve the network throughput [7]. In the RAM protocol, we use the same concept for minimizing the medium time.

TABLE I

TRANSMISSION TIME (IN μS) OF PACKETS OF VARIOUS SIZES AT DIFFERENT RATES SUPPORTED BY IEEE 802.11B

Packet Size vs. Trx Rate	512 bytes	1024 bytes	1500 bytes
1 Mbps	4096	8192	12000
2 Mbps	2048	4096	6000
5.5 Mbps	745	1490	2182
11 Mbps	372	744	1091

Currently the medium access control for multicast is simply CSMA/CA (Carrier Sense Multiple Access/Collision Avoidance); there is no RTS/CST exchange, nor ACK. Therefore, calculating the medium access time of multicast packets is simpler, which consists of only the contention time and the transmission time T_{trx} of the DATA packet, where T_{trx} is equal to the transmission rate divided by the packet size. We simplify the metric even further by eliminating the contention time component for the following reasons:

– The estimate of the average time spent backing off during contention is 310 μs (the value of half the minimum contention window size multiplied by the slot time) [7]. Compared to the transmission time of data packets, the estimated contention time is small, especially when the link rates are low. See Table I, which lists the transmission times of packets of various sizes at different link rates.

– Without the contention time component, computing the link weights does not require prior knowledge of packet sizes.

– Including the contention time in the routing metric does not guarantee optimal throughput as explained above, since the assumption of full interference does not always hold. Our goal is to improve network throughput in the case of single-rate transmissions, rather than offering absolute optimal throughout. Our simulation results show that this simple routing metric, which considers only data transmission time, offers significantly higher throughput than single-rate multicast.

Therefore, we use the transmission time of data packets as the routing metric in the RAM protocol. Each link is assigned a weight that is inversely proportional to the optimal transmission rate of the link as determined by the signal strength of a JOIN_QUERY. The weight of a path is the sum of the weights of all links on that path. Given several paths between a sender and a receiver, RAM will select the path that has the smallest weight, i.e., the path that will incur the lowest overall transmission time.

V. The RAM Protocol

This section describes the design and implementation of the RAM protocol.

A. Overview of the RAM Protocol

JOIN_QUERY packets are used for estimating the channel conditions, and for accumulating the weights of the links on the paths they traverse in a field called $pathWeight$. Given the optimal transmission rate R of a link (as determined by the signal strength of a received packet), the weight of the link is defined as $1/R$. The weight of a path is the sum of the weights of all the links on that path. After receiving a JOIN_QUERY, every node on a path between a sender and a receiver measures the signal strength of the packet, which is then compared against a set of threshold values to suggest a transmission rate, $SuggestedRate$, using an algorithm similar to RBAR [5]. The $SuggestedRate$ is converted to the link weight, which is then added to the $pathWeight$ recorded in the JOIN_QUERY. When the JOIN_QUERY arrives at a receiver, field $pathWeight$ contains the weight of the source-to-destination path traversed by the JOIN_QUERY.

A node participating in the route discovery/refresh process may receive multiple JOIN_QUERY packets from the same sender that take different paths. The node examines all these JOIN_QUERY packets and selects the JOIN_QUERY, and hence the path, with the minimum $pathWeight$ value. This process is termed as the **JOIN_QUERY consolidation** step.

After a receiver performs the JOIN_QUERY consolidation, the selected path is the source-to-destination path incurring the lowest total transmission time among those examined. The receiver then creates a JOIN_REPLY that records the routing information, and sends the JOIN_REPLY to the source, as in the ODMRP algorithm. To prevent the implosion of JOIN_REPLY packets, every node on the paths from the receivers to the sender consolidates information from several JOIN_REPLY packets it receives from downstream[1] nodes into one JOIN_REPLY, as in ODMRP. This process is termed as the **JOIN_REPLY consolidation** step.

Note that all transmissions of JOIN_QUERY and JOIN_REPLY packets use the base rate (e.g., 1 Mbps in 802.11b) to ensure that they are received correctly over long links. Routing fields in control packets and routing tables at mobile nodes are maintained and processed in the same way as in the ODMRP algorithm. Following is the detailed description of the RAM protocol.

B. Data Structures

In addition to the data structures defined by ODMRP, the RAM protocol uses the following new structures:

– Each JOIN_QUERY contains a new field $pathWeight$ that records the weight of the path traversed by the packet. Field $pathWeight$ is initialized to zero. Every time the JOIN_QUERY traverses a link, the weight of the link is added to the variable $pathWeight$ of the JOIN_QUERY.

[1]The downstream direction is from a sender towards a receiver. Conversely, the upstream direction is from a receiver to a sender.

– As illustrated in Figure 1(b), a new field $SuggestedRate[s]$ is added to each entry s of every routing table and every JOIN_REPLY packet, where s denotes the ID of the sender. When a node n receives a JOIN_QUERY from an upstream node p, it suggests an optimal $SuggestedRate$ based on the signal strength of the JOIN_QUERY. However, node n will not be able to send this rate information back to p until later when it receives a JOIN_REPLY from downstream. Node n thus saves the $SuggestedRate[s]$ value in its own routing table. Later, when n receives a JOIN_REPLY from downstream that contains routing information related to source s, it copies the $SuggestedRate[s]$ value from the routing table to field $SuggestedRate[s]$ of the JOIN_REPLY and sends the JOIN_REPLY upstream to p. Node p then uses the $SuggestedRate[s]$ value in the JOIN_REPLY to determine the rate it should use for retransmitting data sent by source s.

– As illustrated in Figure 1(b), a new field $MyTrxRate[s]$ is also added to each entry s of a routing table. During a JOIN_REPLY consolidation interval, an intermediate node may receive several JOIN_REPLY packets from downstream with different $SuggestedRate[s]$ values for the same source s. For example, in Figure 1, node R_1 suggests to I_2 an optimal rate of 1 Mbps for data sent by source S_2, while node R_3 suggests a rate of 5.5 Mbps. Node I_2 should use the lower rate to transmit data packets of S_2 in order to accommodate the downstream link having the lower quality, link (I_2, R_1) (at the expense of a low rate and thus low throughput on link (I_2, R_3)). Therefore, I_2 selects the minimum among the $SuggestedRate[S_2]$ values it sees during a JOIN_REPLY consolidation interval, 1 Mbps in this example, and stores the minimum value in the variable $MyTrxRate[S_2]$. I_2 then uses this rate to transmit data from source S_2 to its downstream neighbors until the next route refreshment interval when a new rate is computed. Note that the recorded rates may become inaccurate before the next route refreshment round due to the mobility of the nodes. However, these rate values still offer better performance than single-rate multicast, as shown by our experimental results in Section VI.

C. JOIN_QUERY Consolidation

When a node receives a non-duplicate JOIN_QUERY (i.e., the first JOIN_QUERY of a route refreshment broadcast) sent by a source s, it delays the packet for a certain period of time, termed as $ConsolidationInterval[s]$, in order to consolidate the JOIN_QUERY packets that take different paths from source s to this node. The goal is to look for a path with a $pathWeight$ as small as possible. When the timer for $ConsolidationInterval[s]$ expires, the JOIN_QUERY having the smallest $pathWeight$ value is selected and broadcasted. The node updates its routing table as specified by the ODMRP algorithm, and copies the $SuggestedRate[s]$ value from the selected JOIN_QUERY to the routing table. After the timer expires, all duplicate JOIN_QUERY packets originating from source s are discarded.

In our experiments, the $ConsolidationInterval$ timer is set to 10 ms. This value was chosen heuristically and should

not be long in order to speed up the route refreshment process and to eliminate paths with too many hops, If a duplicate JOIN_QUERY follows a long path with too many hops, it is unlikely that it will reach the node within the 10 ms slot due to long contention time resulting from having to traverse a large number of hops (JOIN_QUERY packets are transmitted at the base rate). In general, the larger the network, the longer the average source-to-destination path length, and thus the longer the $ConsolidationInterval$ timer should be set.

D. JOIN_REPLY Consolidation

To minimize feedback implosion, every intermediate node p in ODMRP collects JOIN_REPLY packets from downstream nodes during a $ReplyConsolidationInterval$. The node combines routing information for several sources from the collected JOIN_REPLY packets, records all the information into a single JOIN_REPLY, and then broadcasts the aggregate JOIN_REPLY. Our RAM protocol takes advantage of the $ReplyConsolidationInterval$ in ODMRP to consolidate $SuggestedRate[s]$ values from different downstream neighbors in order to compute the transmission rate $MyTrxRate[s]$ that node p will use to transmit data packets from source s, as explained in Section V-B. In the ODMRP algorithm, the $ReplyConsolidationInterval$ value is set to 20 ms.

VI. Experimental Results

We present a comprehensive set of experiments to verify that RAM outperforms single-rate multicast in terms of packet delivery ratio, average packet end-to-end delay, and throughput.

A. Simulation Environment

Our experiments are carried out using QualNet [12], a software that provides scalable simulations of wireless networks and a commercial version of GloMoSim [13]. We compare the performance of the RAM protocol with ODMRP in which data is transmitted at either a single rate of 1Mbps (ODMRP-1Mbps), or 2Mbps (ODMRP-2Mbps), respectively. We use the following metrics in the comparison:

– **Packet delivery ratio** (PDR): is the ratio of the number of data packets actually delivered to the receivers versus the number of data packets supposed to be received. This metric indicates the effectiveness of a multicast protocol.

– **Average end-to-end delay** (EED): is the mean of the end-to-end delays of the packets received at all receivers.

– **Average throughput** : The *receiver throughput* is defined as the total amount of data a receiver R actually receives from all the senders of the multicast group divided by the time it takes for R to get the last packet. The average taken over all the receivers is the *average receiver throughput* of the multicast group. The *average throughput* is defined as the average receiver throughput divided by the number of senders.

Our simulations model networks of sizes ranging from 25 to 200 nodes that are placed randomly within a 1000m $\times$ 1000m area. A two-ray propagation model is used in our experiments, with the free space path loss set to (2.0, 0.0) for near sight and the plane earth path loss set to (4.0, 0.0) for far sight. The transmission power is set constant at 15dBm. We implemented PHY802.11b at the physical layer, which uses a preconfigured BER-based packet reception model. The MAC802.11 with Distributed Coordination Function (DCF) was chosen as the medium access control protocol. Multicast and broadcast transmissions used carrier sensitive multiple access with collision avoidance (CSMA/CA). Unicast transmissions also used RTS/CTS in addition to CSMA/CA.

TABLE II
PARAMETERS USED IN OUR EXPERIMENTS

Function of	NS	NR	SR	MS	ND
ND, Fig.2(a)	5	15	2	1	25-200
NS, Fig. 2(b,c)	1-20	15	(*)	1	50, 100
NR, Fig. 3(a)	5	1-20	2	1	50
MS, Fig. 3(b)	5	15	2	0-20	50
TL, Fig. 3(c)	5	15	1-10	1	50

where
ND: Node density (nodes); NS: Number of senders;
NR: Number of receivers; SR: Sender's rate (packets/s);
MS: Node mobility speed (m/s); TL: Network traffic load;
and TL = NS $\times$ SR;
$(*)$ SR = 10 packets/s $\div$ NS

We model node movements using the random waypoint mobility model [14] with mobility speed ranging from 0m/s to 20m/s (0km/hr to 72km/hr). A node randomly selects a destination and moves towards that destination at a predefined speed. Once the node arrives at the destination, it stays in its current position for a pause time between 0 and 30s. After that it selects another destination and repeats the same procedure.

One multicast group is simulated in all the experiments. Senders and receivers of a multicast group were selected with uniform probability among the mobile hosts. The members joined the multicast group at the beginning of the simulation and stayed on until the whole group was terminated. The senders of a multicast group transmitted at a constant bit rate specified for each experiment. The packet size excluding the header size was 512bytes. The queue size at every mobile node is 50Kbytes. The refresh interval in ODMRP is 20s.

In each experiment, every source sent data for 600s of simulated time. The simulation then continued for another 300s to allow all packets to be processed and routed. Each experiment was run several times using different random seed numbers, and the collected statistics were averaged over those runs. We measure the above metrics as functions of node density, node mobility, the number of multicast senders, the number of multicast receivers, and network traffic load. Additional parameters used in each of the experiments described below are listed in Table II. Note that ND, the node density, is indicated by the number of nodes placed randomly within a 1000m $\times$ 1000m area and TL, the network traffic load, is the sum of all the senders' rates.

B. Simulation Results

Figure 2(a) shows the graphs of PDRs, average throughputs and average EEDs of the three protocols under different *node densities*: 25, 50, 75, 100, 150 and 200 nodes in a 1000m

$\times$ 1000m area. The PDRs and average throughputs of RAM are higher than those of ODMRP-1Mbps and ODMRP-2Mbps, and remain almost constant, independent of the node density. The average EEDs of RAM are significantly lower, especially when compared with those of ODMRP-1Mbps.

When the node density is low, ODMRP-1Mbps gives higher PDRs than ODMRP-2Mbps. When the network is sparse, distances between the nodes are longer. When a high transmission rate, 2Mbps in this case, is used over long ranges, the bit error rates are high. Therefore, ODMRP-2Mbps gives lower PDRs than ODMRP-1Mbps. As the node density increases, distances between nodes are shortened, and the error rates of ODMRP-2Mbps transmission are improved. When the network becomes dense, the PDRs of ODMRP-2Mbps increases, and even surpass the PDRs of ODMRP-1Mbps. In ODMRP-1Mbps, the low transmission rate means longer transmission time; thus more packets are waiting to access the channels. Higher number of contentions implies a higher probability of collision, which leads to lower PDRs.

The PDRs of RAM are always the highest, above 97%, because RAM can always choose the optimal transmission rates. When the node density is sparse, it uses low transmission rates for long distances to ensure low error rates. When the node density is dense, it chooses high rates so that wireless channels are optimally utilized, resulting in less contentions, fewer collisions, and higher throughputs.

The end-to-end delay of a packet is determined by three factors: transmission time, propagation time, and processing time at mobile nodes, which includes delay due to channel contention. Given the same network configuration, propagation time can be considered to be the same in all the experiments. The deciding factors are, therefore, the transmission time and channel contention delay. The higher the transmission rate, the shorter the transmission time. Also, shorter transmission time implies less contentions, leading to lower contention delay at different hops. On the other hand, in our experiments, the average path lengths from the three protocols do not differ much. Consequently, ODMRP-2Mbps offers lower average EEDs than ODMRP-1Mbps. RAM performs better than both of them for the same reasons, because RAM can transmit at a higher rate (e.g., 5.5Mbps or 11 Mbps), whenever possible.

Since the PDRs of RAM are higher and the EEDs are lower, the throughputs given by RAM are higher than those from ODMRP-1Mbps and ODMRP-2Mbps. Note that the average throughput graph is consistent with the PDR graph.

In the following experiments, we observe that RAM outperforms ODMRP-1Mbps and ODMRP-2Mbps for the reasons explained above. Figures 2(b) and (c) provide PDRs, average throughputs and average EEDs as functions of the *number of multicast senders*. RAM shows good scalability in terms of PDRs and average EEDs as the number of senders increases. When the node density is high, the PDRs of ODMRP-1Mbps and ODMRP-2Mbps decline noticeably as the number of senders grows. More senders imply more JOIN_QUERY packets to be flooded, which creates more traffic and therefore more collisions. The problem is more serious when nodes are close to each other, i.e., when the network is dense. RAM is able to mitigate the increased traffic due to the use of high transmission rates whenever possible, resulting in excellent scalability. RAM also performs better than both single-rate protocols in terms of average receiver throughput. The performance difference is bigger when the network is dense, as illustrated in Figure 2(c).

Figures 3(a), (b), and (c) show PDRs, average throughputs and average EEDs as functions of the *number of receivers*, *node mobility speed*, and *network traffic load*, respectively. The graphs indicate that RAM outperforms the single-rate protocols in all cases.

We arrived at the same conclusion from experiments using other node densities (25 and 100 nodes in a 1000m $\times$ 1000m area). Also, as the node density increases, the performance improvement of RAM over single-rate multicast is further magnified, similarly to the cases illustrated by the graphs in Figures 2(b) and (c). We do not show those results here due to space limitation.

VII. An Application: A Framework for Video Multicast in MANETs

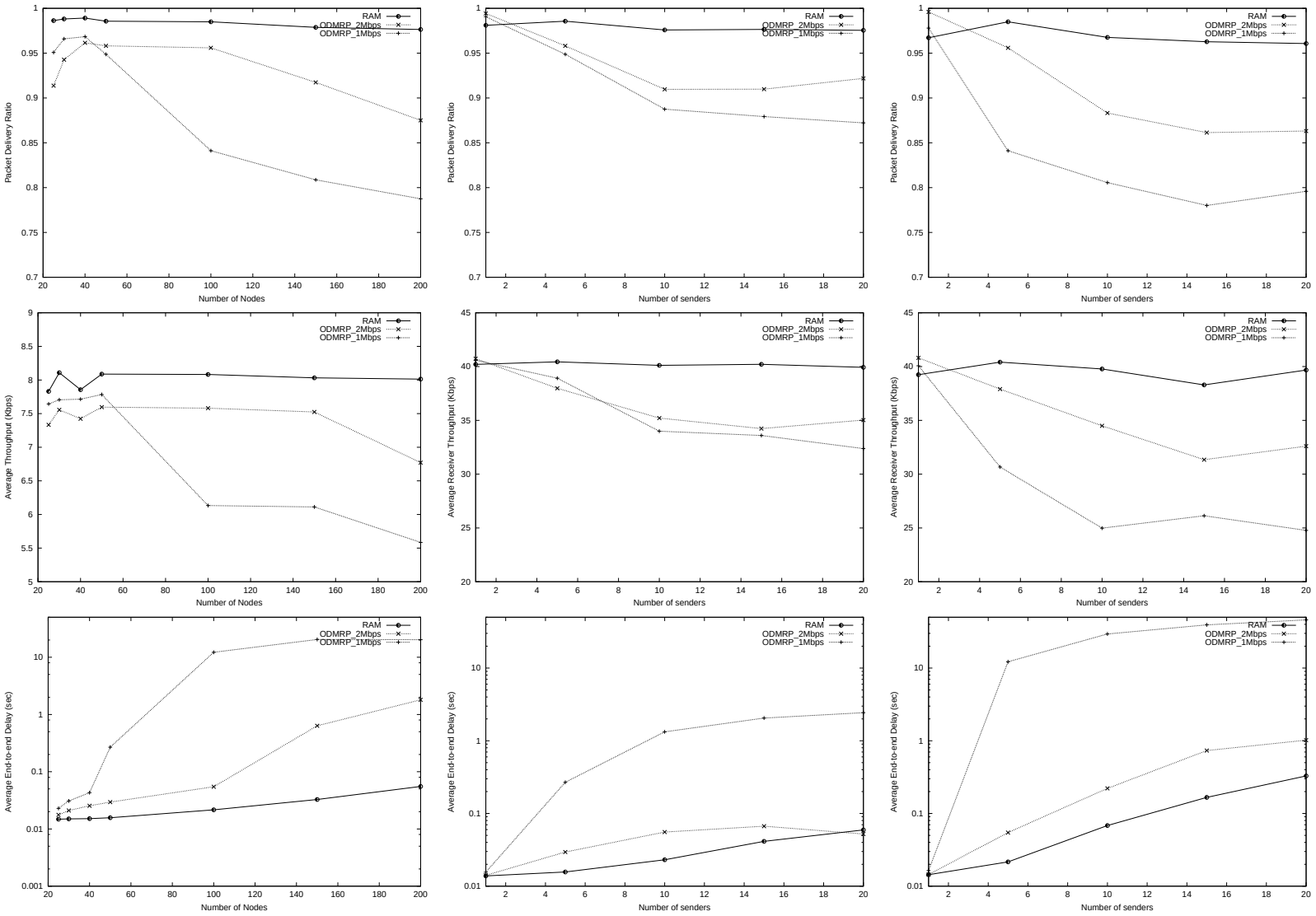
Video multicast over MANETs is a challenging problem due to heterogeneities and constraints such as low-bandwidth wireless channels and limited battery power. Video applications, on the other hand, are demanding in terms of bandwidth and delay constraints. We propose a framework for multicasting video over MANETs, which employs the RAM protocol instead of single-rate multicast to meet the requirements of video applications. This section describes the design, implementation, and performance evaluation of the proposed framework.

A. Framework Design

We solve the heterogeneity and constraint issues of video multicast over MANETs as follows.

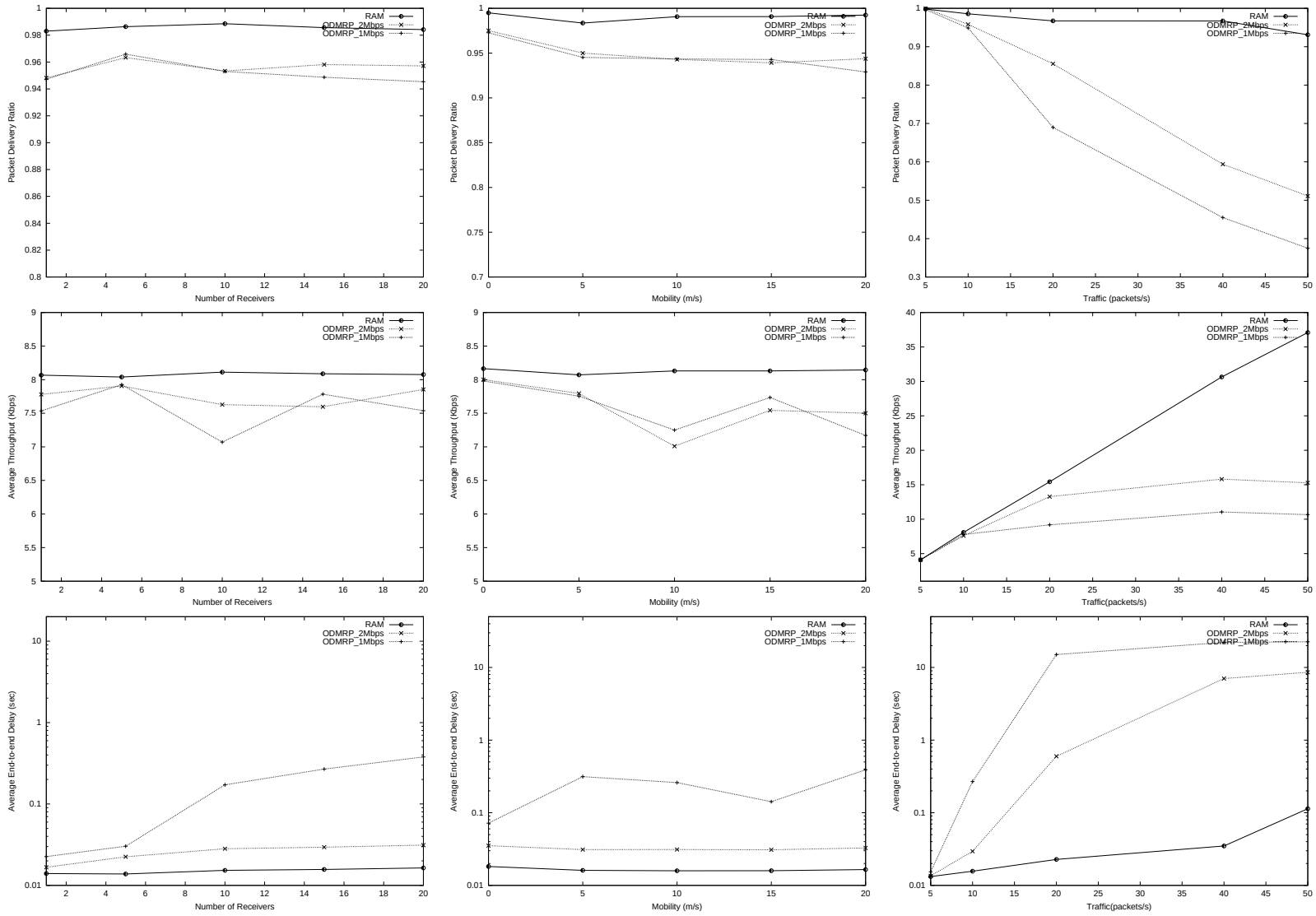
– **Limited bandwidth**. The video multicast framework uses the scalable, noncausal predictive codec with vector quantization and conditional replenishment (SNP/VQR) [15], which offers superior video quality than the standard codecs including MPEG4 and H.264 at low bit rates. Use of SNP/VQR allows transmission at a lower rate than required by the standard codecs with no perceivable difference in the video quality. To further overcome the bandwidth limitation, we employ the RAM protocol instead of single-rate multicast to achieve higher packet delivery ratio and higher throughput.

– **Network/receiver heterogeneities**. In order to accommodate a wide range of mobile devices, SNP/VQR offers layer coding where the video is encoded into multiple layers of data streams. A subset of the compressed bit stream provides the base quality. Additional enhancement layers build on to the video quality in both the spatial and temporal domains. Receivers with powerful batteries and access to higher bandwidth channels can choose to receive all layers, thus acquiring video of the highest quality. Other receivers may opt for a reduced number of layers to save energy and network bandwidth. Similarly, when a router detects the onset of congestion or



(a) F(node density) (b) F(number of senders), density = 50 nodes (c) F(number of senders), density = 100 nodes

Fig. 2. Functions of node density and number of senders



(a) F (number of receivers) (b) F (node mobility) (c) F (traffic load)

Fig. 3. Functions of the number of receivers, node mobility, and network traffic load (network density = 50 nodes)

a drained battery, it can select to transmit only the most important layer(s) using *priority packet dropping* [17].

– **Limited battery power**. This problem is solved in part by using layered video coding and priority packet dropping as mentioned above. In addition, our RAM protocol allows routers to save energy by minimizing the transmission time.

B. Framework Implementation

A SNP/VQR encoder [15] at the sender compresses the video sequence frame by frame. The sender then transforms the compressed frames into an array of data packets, which are transmitted over the MANET. In our framework, we implement three levels of quality of service: gold, silver and bronze. The bronze service uses only the base (the most important) layer. The silver service couples the base and first enhancement layers, while the gold service requires all three layers. A receiver can decide how many layers it wants to receive. After the selected layers are received, the SNP/VQR decoder reconstructs the compressed video frames. In case of transmission errors, the SNP/VQR decoder applies error concealment techniques [16] to the damaged pixels.

To multicast layer coded video over a network, either of the following approaches can be used: priority packet dropping (PPD) [17], and receiver-driven layered multicast (RLM) [18]. Using PPD, packets from the base layer are assigned the highest priority, while packets from enhancement layers are given progressively lower priorities. When there is not enough bandwidth, routers discard packets based on their assigned priorities. Using RLM, a sender transmits each video layer over a separate multicast group. A receiver probes for available bandwidth by periodically joining a higher layer group. If packet loss exceeds some threshold after the join experiment, the receiver leaves the experimented group. Otherwise, it stays at the new subscription level. We find that RLM is not suitable for layered multicast in MANETs due to the following reasons. First, the join experiments adds adversely to the network traffic, especially when the size of the multicast group is large. Second, RLM does not perform well in tracking available bandwidth in a bursty network [17]. Third, PPD is simpler to implement than RLM. We thus chose PPD for multicasting layered video in our framework.

We also implemented the RAM protocol in the framework, which provides the routing function of ODMRP and rate adaptation. The RAM protocol significantly enhances the performance of multicast in MANETs, as demonstrated by simulation results in Section VI. In the next section, we present experimental results that show the effectiveness of the proposed framework.

C. Performance Evaluation

We evaluated the performance of our video multicast framework using several video sequences under different network conditions. We present here the results from one set of experiments in which a video sequence with 120 frames was encoded and transmitted over the simulated MANET. The simulated network consisted of 50 mobile nodes randomly placed within a $1000\text{m} \times 1000\text{m}$ area with each mobile host moving at a speed of 1m/s. One node was randomly selected as the sender of a multicast session having 20 receivers. The receivers were selected with uniform probability among the mobile hosts. The members joined the multicast group at the beginning of the simulation and stayed onuntil the whole group was terminated. The output from the SNP/VQR codec was compressed video packets marked as "gold", "silver" and "bronze" packets. The sender transmitted the packets over three respective multicast groups, each corresponding to a layer. The sending rate for each group is 60 packets/s, which corresponds to the output rate of the SNP/VQR encoder. The packet size excluding the header size varied from 99 to 250 bytes. The size of the queue at mobile nodes is 50 Kbytes. PPD was implemented at every mobile host using several queues with different priorities. Within each queue, packets were discarded upon congestion using first-in-first-out scheduling. Other network parameters were set as described in Section VI.

TABLE III
DISTRIBUTION OF VIDEO PACKETS LOST

Layer	Packets	Packets Lost in *Light* Traffic		Packets Lost in *Heavy* Traffic	
		Number	Rate	Number	Rate
Bronze	4,880	70	1.43%	196	4.02%
Silver	4,880	99	2.03%	320	6.56%
Gold	4,800	175	3.65%	3,276	68.25%
Total	14,400	344	2.70%	3,795	26.28%

We measure the packet loss rates for the three video layers under both light and heavy traffic conditions. The results are shown in Table III, where the packet loss rates averaged over the twenty receivers are given. We can see that the majority of lost packets in the heavy load network are from the gold layer (the least important layer) due to priority packet dropping.

The quality of reconstructed video frames is assessed by computing the average peak signal-to-noise ratio (PSNR), defined as

$$\text{Average PSNR} = \frac{1}{R \times N_K} \sum_{r=1}^{R} \sum_{k=1}^{N_K} \text{PSNR}(k, r), \qquad (1)$$

where $\text{PSNR}(k, r)$ is defined as the PSNR of frame k reconstructed by receiver r, N_K is the total number of frames in the video sequence, and R is the number of receivers included in the multicast session. Table IV lists the average PSNRs under three different network conditions: no packet loss, low packet loss under light traffic, and moderate packet loss under heavy traffic. We observe that the quality of the reconstructed video in the light load network is comparable to that in an ideal setting with no packet loss. The video quality remains reasonably good even in the network with high traffic.

To provide a subjective evaluation of the quality of the reconstructed video, Figure 4 includes several frames reconstructed by the SNP/VQR decoder at different receivers. Since the quality of the reconstructed video varies from one receiver to another, we chose frames from video sequences whose average PSNR is closest to the PSNR values included in



(a) Original frame (b) Light traffic, bronze service (c) Light traffic, silver service (d) Light traffic, gold service

(e) No loss, gold service (f) Heavy traffic, bronze service (g) Heavy traffic, silver service (h) Heavy traffic, gold service

Fig. 4. Perceived video quality from bronze, silver and gold services

TABLE IV
AVERAGE PSNRS FOR BRONZE, SILVER, AND GOLD SERVICES

Network Conditions	Packets Lost	Quality of Service		
		Bronze	Silver	Gold
Ideal (no loss)	0%	28.75dB	29.58dB	29.99dB
Light traffic	2.70%	28.59dB	29.32dB	29.69dB
Heavy traffic	26.28%	28.36dB	28.88dB	29.00dB

Table IV. The sample frames demonstrate that our framework offers very good perceptual visual quality, even in congested networks. This is due in large part to the high performance provided by the proposed RAM protocol.

VIII. CONCLUSION

We propose a multirate-aware routing protocol for multicast in MANETs. Channel conditions are estimated, and the resulting optimal transmission rates are communicated using the control packets of the routing protocol. Among several paths available between a sender and a receiver, the routing protocol selects the path with the lowest total transmission time. Our simulation results show that the proposed RAM protocol outperforms single-rate multicast in terms of packet delivery ratio, end-to-end delay, and throughput. RAM's high performance enables effective video multicast in MANETs, as demonstrated by our proposed video multicast framework. RAM works with any multirate standards and, to the best of our knowledge, is the first multicast routing protocol that exploits the multirate capability of mobile devices.

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