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Urgency-based packet scheduling and routing algorithms for delay-sensitive data over MANETs

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Abstract This paper proposes urgency-based packet scheduling and routing algorithms to effectively deliver delay-sensitive data over a multi-hop mobile ad hoc networks supporting IEEE 802.11 multi-rate service. First, packet urgency, node urgency, and route urgency are defined on the basis of the end-to-end delay requirement. Based on these urgency metrics and the estimated transmission delay of each packet by Kalman filter, the proposed packet scheduling algorithm determines the transmission order and drop policy to minimize the node urgency without unnecessary packet drop, and the proposed routing algorithm establishes a route to minimize the derivative of route urgency in order to maximize the number of packets delivered within the required end-to-end delay. Finally, experimental results are presented to evaluate the performance of the proposed joint working algorithms.

Keywords Packet scheduling · Routing · Quality of service · Delay-sensitive data transmission · Mobile ad hoc network · IEEE 802.11 multi-rate service

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1 Introduction

Mobile ad hoc networking technology has attracted considerable interest because nodes can self-configure and dynamically maintain the network topology without the infrastructural support. In recent years, the demand for various multimedia application services such as video conferencing, surveillance system, and video on demand service over Mobile Ad hoc NETworks (MANETs) has been growing rapidly [1, 2]. However, it is not easy to efficiently support data transmission with strict end-to-end delay requirements over MANETs. Furthermore, various issues caused by interference among nodes over the multi-hop network, time-varying network topology caused by the mobility of nodes, and constrained battery power hamper quality of service (QoS) support over MANETs. Routing and packet scheduling algorithms are some of the key functionalities for improving QoS over MANETs. Thus far, many efficient algorithms for routing and packet scheduling over MANETs have been proposed in literature.

A routing algorithm establishes a route between a source node and a destination node via other participating nodes. In general, ad hoc routing protocols are categorized into three types: table-driven, on-demand, and hybrid. Table-driven protocols are essentially proactive because each node maintains an up-to-date routing table by periodically exchanging routing information. Therefore, the delay in constructing a route is negligible because the route is already known when data packets are forwarded. However, a large amount of network resources is required to maintain the latest routing information. Destination-sequenced distance-vector (DSDV) [3], optimized link state routing (OLSR) [4], Cluster-head gateway switch routing (CGSR) [5], and wireless routing protocol (WRP) [6] are examples of table-driven protocols. In contrast, on-demand protocols are fundamentally reactive, because nodes invoke the route construction mechanism only when a route is needed. Thus, the delay until the route is established may be slightly longer. Examples of on-demand protocols include dynamic source routing (DSR) [7], ad hoc on-demand distance vector (AODV) [8], temporally-ordered routing algorithm (TORA) [9], and cluster-based routing protocol (CBRP) [10]. Hybrid protocols combine the advantages of table-driven and on-demand protocols. For example, zone routing protocol (ZRP) [11] proactively maintains topology and link state information within the routing zone and reactively searches for routes beyond the routing zone.

A packet scheduling algorithm determines the transmission order and drop policy at each node. An earliest deadline first (EDF) scheduler [12] determines the packet transmission order by considering the arrival time and the end-to-end delay requirement of each packet. The transmission priority of each packet increases with the amount of time consumed over the network. In a coordinated multi-hop scheduling (CMS) scheduler [13], the transmission priority of a packet at each node is recursively expressed using the transmission priority of the same packet at the previous node along the route. A node will increase the transmission priority of a packet if an excessive delay occurs. Conversely, if a small delay occurs, the node will decrease the transmission priority of a packet in order to serve more urgent packets. In [14], the proposed scheduler is designed to mitigate interference and improve network capacity over wireless mesh networks. The scheduler determines the sets of flows along which transmissions can take place with the least inter-flow interference based on received signal strength. In the drop-tail policy [15], nodes simply drop newly arriving packets if the buffer is full. However, the drop-tail policy leads to a high packet loss rate, a long queuing delay, low utilization of network resources, and long-lasting congestion under a heavy traffic load. Active packet drop policies such as random early detection (RED) [16], BLUE [17], stabilized random early drop (SRED) [18], and dynamic random early detection (DRED) [19] have been proposed to solve these problems. Typically, they operate by maintaining drop/mark probabilities and probabilistically dropping or marking packets before the buffer becomes full. The RED algorithm detects congestion and measures the traffic load level in the buffer using the average buffer size. BLUE uses the instantaneous buffer length and link utilization as indicators of traffic load and congestion. SRED and DRED are more effective in stabilizing the buffer size and controlling the packet loss rate while maintaining high link utilization.

In recent years, many considerable efforts have been focused on a cross-layer approach to effectively utilize the limited and time-varying wireless network resources. In [20], joint scheduling and power control algorithms are proposed to limit multi-user interference and reduce the power consumption over a single hop. QoS-aware routing (QAR) [21] incorporates the admission control scheme and the feedback scheme to determine a route satisfying the bandwidth requirement. In Liang et al. [22], optimizes a lifetime-distance factor that represents the relative weights assigned to the remaining distance and the remaining lifetime of a packet in order to determine its transmission priority over multi-hop wireless networks. A novel software solution between the 802.11 MAC and network layers (Layer 2.5 SoftMAC [23]) is proposed for coordinating the real-time and best-effort packet transmission among neighboring nodes over multi-hop wireless networks. Several effective rate control algorithms at the MAC (Medium Access Control) layer are proposed by considering the time-varying wireless link states [24–26]. Recently, multi-rate based routing algorithms have been proposed in [27, 28]. These routing algorithms determine a route with a higher throughput and lower delay although hop count increases. IEEE 802.11 standards are widely employed as the wireless MAC protocol, which supports multi-rate service by changing the modulation scheme and coding rate based on the received signal strength at the receiver. IEEE 802.11b and IEEE 802.11 g provide data rates of up to 11 and 54 Mbps, respectively. Higher data rates are commonly achieved by selecting more efficient modulation schemes.

In this paper, we propose urgency-based joint working packet scheduling and routing algorithms for delay-sensitive data transmission over multi-rate MANETs. A unique feature of the proposed joint working algorithms is that the packet scheduling algorithm at the MAC layer and the routing algorithm at the network layer are tightly coupled on the basis of the urgency metrics to effectively deliver delay-sensitive data over multi-rate MANETs. The remainder of this paper is organized as follows. The details of the proposed joint working algorithms are described in Sect. 2, the experimental results are presented in Sect. 3, and the concluding remarks are stated in Sect. 4.

2 Proposed urgency-based packet scheduling and routing algorithms

Our objective is to deliver delay-sensitive data packets as many as possible within the end-to-end delay requirement and to distribute the traffic load across the entire network simultaneously over a multi-hop MANET supporting IEEE 802.11 multi-rate service. To achieve this objective, urgency metrics are defined by considering the end-to-end delay requirement.

Definition of packet urgency The packet urgency ($u_{pkt}(t)$) at the jth node along a route ($\vec{R}$) at time t is defined as

$$u_{pkt}(t) = f_{urg}\left(\frac{d_{residual}(t)}{D_{\max}}\right), \tag{1}$$
$$\text{where } d_{residual}(t) = D_{\max} - d^{j}_{acc}(t),$$

where D_{max} is the maximum tolerable end-to-end delay, $d^j_{acc}(t)$ is the cumulative delay from the source node to the jth node, $d_{residual}(t)$ is the residual delay to satisfy the end-to-end delay requirement over the remaining hops, and $f_{urg}\left(\frac{d_{residual}(t)}{D_{\max}}\right)$ is the urgency mapping function between $d_{residual}(t)$ and $u_{pkt}(t)$. In general, $u_{pkt}(t)$ should be inversely proportional to $d_{residual}(t)$ (i.e., a packet with a smaller $d_{residual}(t)$ should be transmitted more urgently for delivery to the destination node in time) [29].

Definition of node urgency The node urgency ($u_{node}(t)$) is defined as the sum of packet urgency of all the packets in the buffer, i.e.,

$$u_{node}(t) = \sum_{i=1}^{n_{pkt}} u_{pkt(i)}(t), \tag{2}$$

where n_{pkt} is the number of packets in the buffer and $u_{pkt(i)}(t)$ is the packet urgency of the ith packet in the buffer. A larger node urgency implies that more urgent packets are in the buffer.

Definition of route urgency The route urgency ($u_{route}(t)$) is defined as the sum of node urgency of all the nodes along $\vec{R}$, i.e.,

$$u_{route}(t) = \sum_{j\in\vec{R}} u_{node(j)}(t), \tag{3}$$

where $\vec{R}$ denotes the route including all intermediate nodes and $u_{node(j)}(t)$ is the node urgency of the j_{th} node along the route. As the route urgency increases, it may become a congestion route.

The overall architecture of the proposed joint working algorithms is shown in Fig. 1. As shown in the figure, the packet scheduling algorithm at the MAC layer and the routing algorithm at the network layer are tightly coupled on the basis of the urgency metrics to effectively deliver delay-sensitive data over multi-rate MANETs. In the proposed system, the packet scheduling algorithm determines the packet transmission order and the packet drop policy to minimize the node urgency at each node without unnecessarily dropped packets, and the routing algorithm constructs a route to deliver data packets in time with the given node urgency information. These algorithms are presented in detail in Sects. 2.1 and 2.2, respectively.

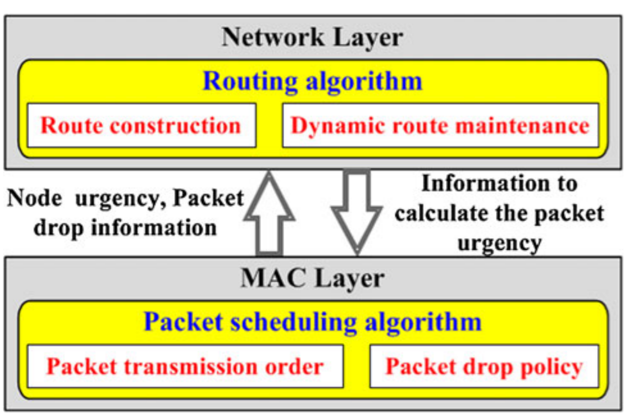


Fig. 1 Overall architecture of proposed joint working algorithms

2.1 Proposed urgency and link condition-based packet scheduling algorithm

First, we describe the additional information required in the data packet header to successfully implement the proposed packet scheduling algorithm. In the proposed algorithms, it is assumed that one-hop neighbor nodes are already time synchronized by periodically exchanging beacons including time stamps [30–32]. An example is shown in Table 1. Each node can estimate the delay over the previous hop using the timestamp (ts^{pre}) at the previous node. Then, the cumulative delay from the source node to the current node ($d^{cur}_{acc}(t)$) is calculated by adding the delay over the previous hop ($ts^{cur} - ts^{pre}$) to the cumulative delay from the source node to the previous node ($d^{pre}_{acc}(t)$).

2.1.1 Transmission delay estimation over IEEE 802.11b multi-rate service

It is assumed that the transmission power and the carrier-sensing threshold are the same at every node, and a modulation scheme is automatically selected on the basis of the received signal strength by receiver-based auto-rate (RBAR) [33]. In other words, a transmitter selects the adequate modulation scheme based on the observed received signal strength and determines the transmission rate. However, instantaneous observed received signal strength is not proper for representing the future wireless link status because the wireless link changes dynamically. Kalman filter [34, 35] is adopted to estimate the received signal strength for the next interval because this filter is a linear dynamic system to effectively remove the non-stationary white Gaussian noise from the measured data. First of all, wireless channel is modeled by

$$rss(t+\Delta) = A\cdot rss(t) + w(t), \tag{4}$$

$$rss_{obs}(t) = H\cdot rss_{est}(t) + v(t), \tag{5}$$

where $rss(t)$ is the state variable of the received signal strength value in the system model at time t, $rss_{obs}(t)$ is the observed received signal strength value at time t, Δ is the updating interval, A is the state transition matrix, H is the matrix for observation model, $w(t)$ is the process white noise, and $v(t)$ is the measurement white noise. A indicates how wireless channel conditions are changing at the next

Table 1 Additional information in data packet header

D_{max}	d_{acc} (t)	ts
End-to-end delay requirement (4 bytes)	Accumulated delay from the source node to the previous node (4 bytes)	Timestamp at the previous node (4 bytes)

state and H represents the relationship between the measurement sample rss_{obs} and the estimated wireless channel state rss_{est} at time t as shown in Eq. (5). After the system model of Kalman filter is determined, the estimated received signal strength value at the node is calculated by

$$\widetilde{rss}_{est}(t+\Delta) = A \cdot \widetilde{rss}_{est}(t) + K(t) \cdot \left(rss_{obs}(t) - H \cdot A \cdot \widetilde{rss}_{est}(t)\right), \quad (6)$$

where $\widetilde{rss}_{est}(t)$ is the estimated received signal strength value at time t and $K(t)$ is the Kalman gain that is determined by considering the stochastic nature of the process and measurement dynamics. It is calculated by

$$K(t) = \left(A \cdot \widetilde{P}(t) \cdot A^T + Q\right) \cdot H^T \cdot \left(H \cdot \left(A \cdot \widetilde{P}(t) \cdot A^T + Q\right) \cdot H^T + R\right)^{-1}, \quad (7)$$

where Q is the covariance matrix of the process white noise, R is the covariance matrix of the measurement white noise, and $\widetilde{P}(t)$ is the covariance of the prediction error. Those are obtained by

$$Q = E\left(w(t) \cdot w(t)^T\right), \quad (8)$$

$$R = E\left(v(t) \cdot v(t)^T\right), \quad (9)$$

$$\widetilde{P}(t) = \left(A \cdot \widetilde{P}(t-\Delta) \cdot A^T + Q\right) - K(t-\Delta) \cdot H \cdot \left(A \cdot \widetilde{P}(t-\Delta) \cdot A^T + Q\right). \quad (10)$$

Now, we can estimate the transmission delay $\tilde{d}_{trans}(t)$ at the current node over IEEE 802.11b multi-rate service as follows [36].

$$\begin{aligned}\tilde{d}_{trans}(t) &= \left(DIFS + 3SIFS + BO + T_{RTS} + T_{CTS} + T_{DATA}\left(\widetilde{rss}_{est}(t)\right) + T_{ACK}\right) \cdot 10^{-6} \\ &= \left(1542 + \frac{8 \cdot L_{DATA\_MAC}}{R_{tr}\left(\widetilde{rss}_{est}(t)\right)}\right) \cdot 10^{-6}, \end{aligned} \quad (11)$$

where $DIFS$ is the distributed inter-frame space, $SIFS$ is the short inter-frame space, BO is the average back-off time, T_{RTS}, T_{CTS}, $T_{DATA}\left(\widetilde{rss}_{est}(t)\right)$, and T_{ACK} are the transmission time of RTS (Request-To-Send), CTS (Clear-To-Send), data when $\widetilde{rss}_{est}(t)$ was given, and ACK (ACKnowledgment) packets at the node, respectively, L_{DATA_MAC} is the data length at the MAC layer, and $R_{tr}\left(\widetilde{rss}_{est}(t)\right)$ is the transmission rate over the wireless link at the node.

2.1.2 Proposed packet transmission order determination

The transmission priority of each packet is determined on the basis of its packet urgency and wireless link condition. A packet with a high transmission priority is transmitted earlier than a packet with a low transmission priority to minimize the node urgency. In the proposed packet scheduling algorithm, the transmission priority of a data packet is determined according to the variation of node urgency after the packet transmission. The node urgency after the kth data packet transmission can be expressed by

$$\begin{aligned}u_{node}\left(t + \Delta\tilde{T}_{pkt(k)}\right) &= u_{node}(t) + \Delta u_{node}(t,k) \\ &= u_{node}(t) + \sum_{\substack{i=1\\ i\neq k}}^{n_{pkt}} \left\{u_{pkt(i)}\left(t + \Delta\tilde{T}_{pkt(k)}\right) - u_{pkt(i)}(t)\right\} - u_{pkt(k)}(t), \end{aligned} \quad (12)$$

where $\Delta u_{node}(t,k)$ is the increasing amount of the node urgency when the kth data packet is transmitted at time t, and $\Delta\tilde{T}_{pkt(i)}$ is the transmission delay of the ith data packet estimated by using Kalman filter. Now, the increasing amount of the node urgency after the transmission of the kth data packet at time t is represented as follows. The equation is simplified by using the first order Taylor series expansion under the assumption of $\left|\Delta\tilde{T}_{pkt(k)}\right| < \varepsilon$, where ε is a small real number.

$$\begin{aligned}\Delta u_{node}(t,k) &= u_{node}\left(t + \Delta\tilde{T}_{pkt(k)}\right) - u_{node}(t) \\ &\cong \sum_{\substack{i=1\\ i\neq k}}^{n_{pkt}} \left\{u_{pkt(i)}(t) + \Delta\tilde{T}_{pkt(k)} \cdot \frac{du_{pkt(i)}(t)}{dt}\right\} - \sum_{\substack{i=1\\ i\neq k}}^{n_{pkt}} u_{pkt(i)}(t) - u_{pkt(k)}(t) \\ &= \sum_{\substack{i=1\\ i\neq k}}^{n_{pkt}} \left\{\Delta\tilde{T}_{pkt(k)} \cdot \frac{du_{pkt(i)}(t)}{dt}\right\} - u_{pkt(k)}(t) \\ &\cong \Delta\tilde{T}_{pkt(k)} \cdot \sum_{\substack{i=1\\ i\neq k}}^{n_{pkt}} \left\{\frac{\Delta u_{pkt(i)}(t)}{\Delta\tilde{T}_{pkt(i)}}\right\} - u_{pkt(k)}(t), \end{aligned} \quad (13)$$

where $\Delta u_{pkt(i)}(t)$ is the decreasing amount of the packet urgency when the ith data packet is transmitted. Therefore, the proposed packet scheduling algorithm selects a packet from the packets in the buffer to maximize the decreasing amount of the node urgency normalized by its transmission time, i.e.,

$$k^* = \arg\min_{1\le k\le n_{pkt}} \frac{\Delta u_{node}(t,k)}{\Delta\tilde{T}_{pkt(k)}} = \arg\min_{1\le k\le n_{pkt}} \sum_{\substack{i=1\\ i\neq k}}^{n_{pkt}} \left\{\frac{\Delta u_{pkt(i)}(t)}{\Delta\tilde{T}_{pkt(i)}}\right\} - \frac{u_{pkt(k)}(t)}{\Delta\tilde{T}_{pkt(k)}}. \quad (14)$$

If $\Delta\tilde{T}_{pkt(i)} \neq \Delta\tilde{T}_{pkt(j)}$ for $i\neq j$, a sequence of packets should be considered simultaneously to obtain the optimal solution. In this case, the wireless link condition in the relatively far future is required to transmit multiple packets. However, it is non-causal and very difficult to predict the condition accurately. As a feasible solution, we select only a packet with the steepest decreasing rate (i.e. the smallest derivative of the packet urgency) of the node urgency at time t. Because routing control packets such as RREQ (Route REQuest), RREP (Route REPly), and RERR (Route ERRor) play a critical role in route construction and maintenance, the proposed packet scheduling algorithm assigns the highest transmission priority (maximum packet urgency) to them in the implemented system.

Since the buffer size at each node is limited, a packet drop policy is required when the buffer overflows. In this paper, packets are dropped in the following two cases: (I) To avoid unnecessary packet transmission, delay-sensitive packets are immediately dropped when the sum of their accumulated delay and estimated transmission delay is larger than the required end-to-end delay regardless of buffer occupancy and network congestion level. (II) When buffer overflow is caused by newly arriving packets, each node in the proposed system drops the data packet having the highest probability of not arriving at its destination node in time. That is, when the buffer overflows, a node discards the data packet having the maximum packet urgency in the buffer. Figure 2 shows a flowchart for the proposed packet drop policy.

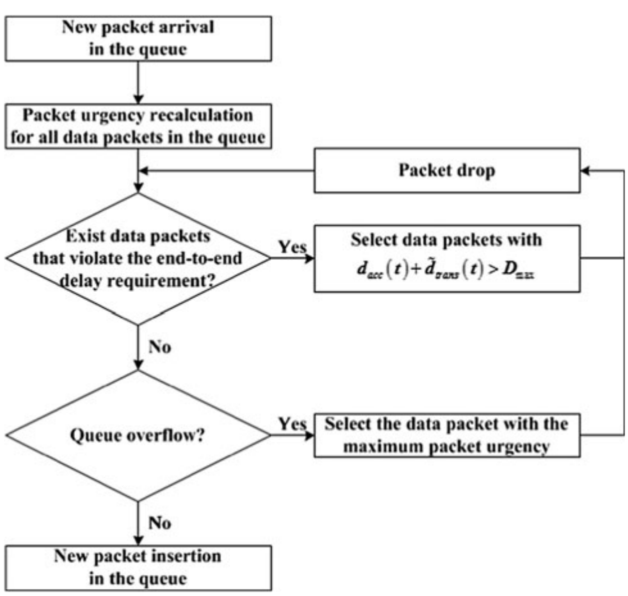


Fig. 2 Flowchart for proposed packet drop policy

2.2 Proposed urgency-based routing algorithm

As compared to existing routing algorithms, one of the most significant and distinctive features of the proposed routing algorithm is the fact that a route is established to maximize the number of arriving packets at the destination in time instead of the shortest or fastest routes. If the urgency mapping function is non-increasing and differentiable in $0<\frac{d_{residual}}{D_{\max}}<1$, the route urgency increases when more packets are staying at the intermediate node buffer. Based on the phenomenon, the derivative of route urgency is selected as the cost function of the below routing problem. Now, the routing problem is formulated as follows to take into account the number of packets and their urgency over the route simultaneously.

Routing problem formulation Determine a route $\vec{R}$ to minimize

$$\frac{du_{route(\vec{R})}(t^-)}{dt}. \quad (15)$$

When the urgency mapping function is convex, the established route includes the nodes holding packets with big residual delay values. When the urgency mapping function is linear, the derivative of the urgency mapping function is constant value, and thus the routing algorithm considers only numbers of packets stored in the buffer at the nodes over the constructed route. At last, if the urgency mapping function is concave, the created route consists of nodes having packets whose delay values are so big that it would be dropped shortly without arriving at the destination on time.

Table 2 Additional information in RREQ packet header

$\sum_{j\in\vec{R}_{interim}} \frac{du_{node(j)}(t)}{dt}$
Derivative of the interim route urgency (4 bytes)

2.2.1 Route construction mechanism

In the proposed routing algorithm, the basic procedure for route construction is similar to AODV [6] but additional information is required in the RREQ packet header, as shown in Table 2. Based on the information, the optimal route of the above problem is obtained by

$$\vec{R}^* = \underset{\vec{R}\in R_{avail}}{\arg\min} \frac{du_{route(\vec{R})}(t^-)}{dt} = \underset{\vec{R}\in R_{avail}}{\arg\min} \left\{ \sum_{j\in\vec{R}} \frac{du_{node(j)}(t^-)}{dt} \right\}, \quad (16)$$

where R_{avail} is the set of all available routes between the source node and the destination node. A source node floods an RREQ packet only when a route is needed. When an intermediate node receives the RREQ packet, it recalculates the derivative of the interim route urgency over the interim route by adding its node urgency derivative, and then updates the RREQ packet. The intermediate node rebroadcasts only the updated RREQ packet to its neighbor nodes for an interim route with the minimum derivative of the route urgency in order to reduce the number of RREQ packets. These steps are repeated until the RREQ packet arrives at the destination node. When RREQ packets arrive at the destination node, the route with the minimum derivative of route urgency is selected, and then, an RREP packet for the selected route is sent back to the source node. The obtained route is optimal in terms of derivative of route urgency without loss of generality. Figure 3 shows a flowchart for the proposed route construction mechanism.

An example of the route selection mechanism is presented in Fig. 4. It is assumed that a source node S floods an RREQ packet for the route discovery to a destination node D. When the intermediate node B receives the RREQ packet, it rebroadcasts the updated RREQ packet only for the interim route {S, A, B}, which has a smaller derivative of the interim route urgency than the interim route {S, G, B}. When RREQ packets arrive at the destination node through the routes {S, A, B, C, D} and {S, E, F, D}, the destination node calculates their derivative of route urgency using Eq. (16). The destination node selects the route {S, A, B, C, D} with the minimum derivative of route urgency, even though the number of hops in this route is greater than the number of hops in {S, E, F, D}. As shown in Fig. 4, the proposed routing algorithm selects the route having the minimum derivative of route urgency, which is constructed along nodes with a relatively light traffic load. Consequently, we can achieve load balancing over the entire network.

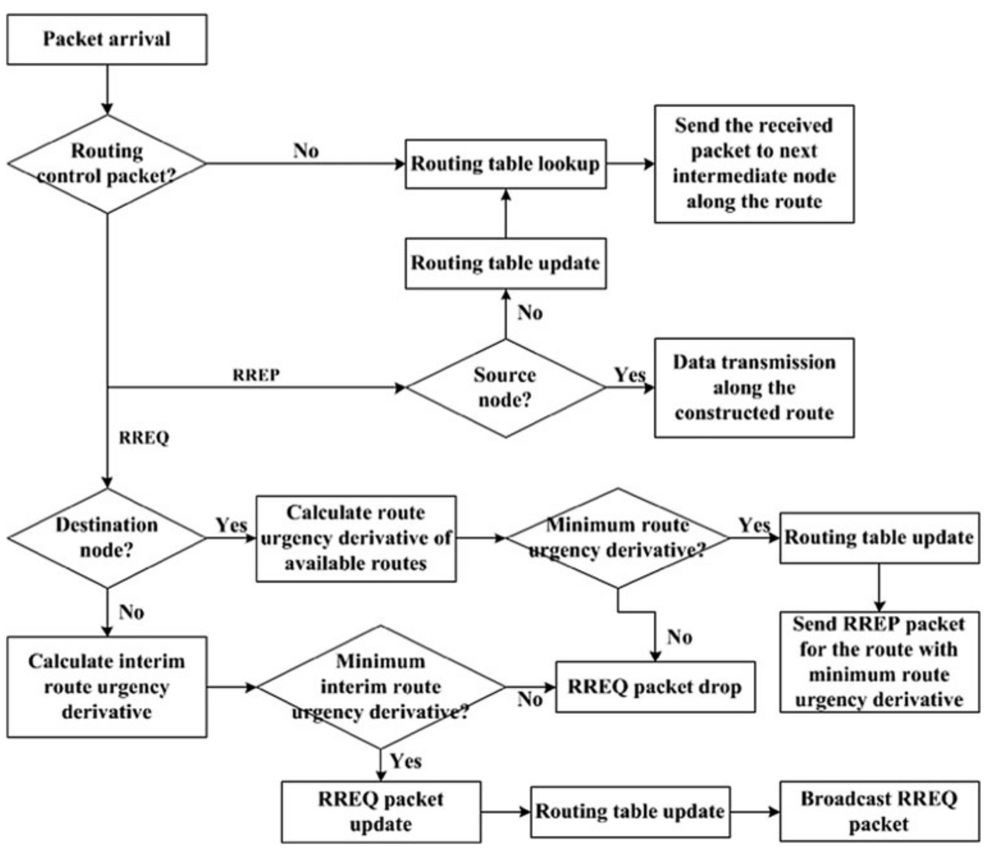
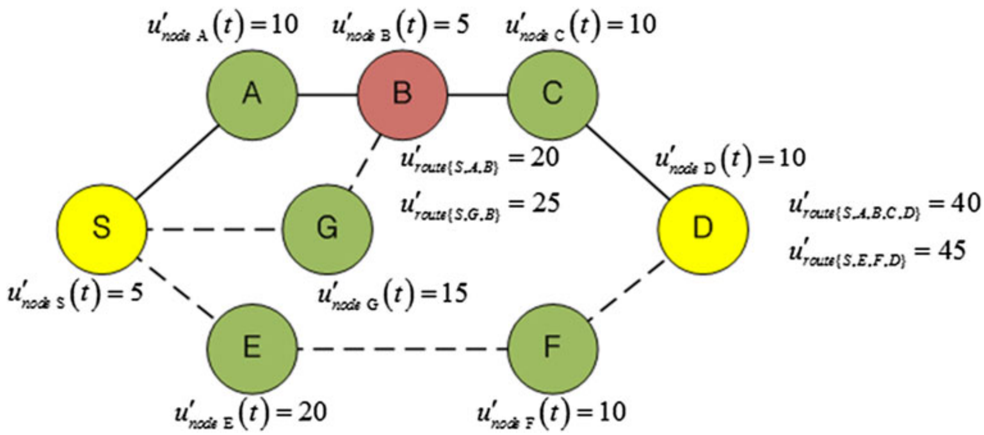


Fig. 3 Flowchart for proposed route construction mechanism

Fig. 4 Example of route selection



2.2.2 Dynamic route maintenance mechanism

The service satisfaction of users may deteriorate drastically if there are frequent packet drops along an unstable route. Therefore, dynamic route maintenance must be considered in order to effectively transmit delay-sensitive data. In this paper, the route maintenance mechanism considers the following two factors: (I) the estimated signal strength of each flow and (II) the packet drop rate of each flow. When the estimated signal strength obtained from Eq. (6) is less than the pre-determined threshold, the intermediate node notifies the source node by sending an RERR packet and a new route is established before the actual link failure occurs. As a result, the delay jitter is decreased because it effectively reduces the link failures associated with the node mobility. Meanwhile, the nodes record the number of dropped packets within a packet drop counting interval and the corresponding flows. Then, the nodes select the flow whose packets are dropped with the highest frequency. If the number of dropped packets for the selected flow is larger than a pre-determined threshold, an RERR packet is transmitted to the source node to request a new route, even though the current route is still available. If the source node desires a new route after receiving the RERR packet, it can reinitiate the route construction mechanism.

The packet drop counting interval is dynamically adjusted on the basis of observed packet drop information in order to reduce the route maintenance overhead. If the number of dropped packets for the selected flow is below a pre-determined threshold, a node realizes that the current routes passing through it are suitable for delay-sensitive data transmission and increases the packet drop counting interval by a fixed step size, and vice versa. Lower and upper bounds for the packet drop counting interval are required to avoid unnecessary route reconstructions and to effectively reflect the current network state, respectively. To reduce the unnecessary routing overhead, the node with a large node urgency value does not broadcast RREQ and RREP packets to its neighbor nodes since it may not deliver the delay-sensitive data on time. It is called RREQ/RREP packet forwarding control mechanism in the followings. Figure 5 shows a flowchart for the proposed dynamic route maintenance mechanism.

3 Experimental results

Experimental results are provided to demonstrate the performance of the proposed joint working algorithms. During the experiment, the NS-2 simulator [37] is employed. The experimental environment is set up as follows:

(1) Thirty nodes are randomly located in a 400 m × 400 m rectangular area and seventy nodes are randomly located in a 600 m × 600 m rectangular area. All nodes move at a speed of 1–5 m/s according to a random waypoint model. The number of connections is 1–20. Pairs of source and destination nodes are arbitrarily determined. A two-ray ground model is adopted for the wireless channel, and nodes try to access the medium according to the IEEE 802.11 MAC protocol [38].
(2) The target bandwidth of each data flow is set to 40.96 Kbps. That is, all source nodes generate 10 packets/s, and the packet size is 512 bytes.
(3) The buffer size is fixed to 50 packets at every node. The packet drop counting interval is dynamically adjusted in the range of 1.0–5.0 s with a fixed step size of 1.0 s. The threshold for dynamic route maintenance is set to a 20 % packet drop rate. D_{max} is set to 1.0 s.
(4) The transmission power is set to 0.0127 W. The carrier sense threshold and receptions threshold are 2.25×10^{-11} W and 3.652×10^{-10} W, respectively. The transmission rates according to the distance are determined as shown in Fig. 6.

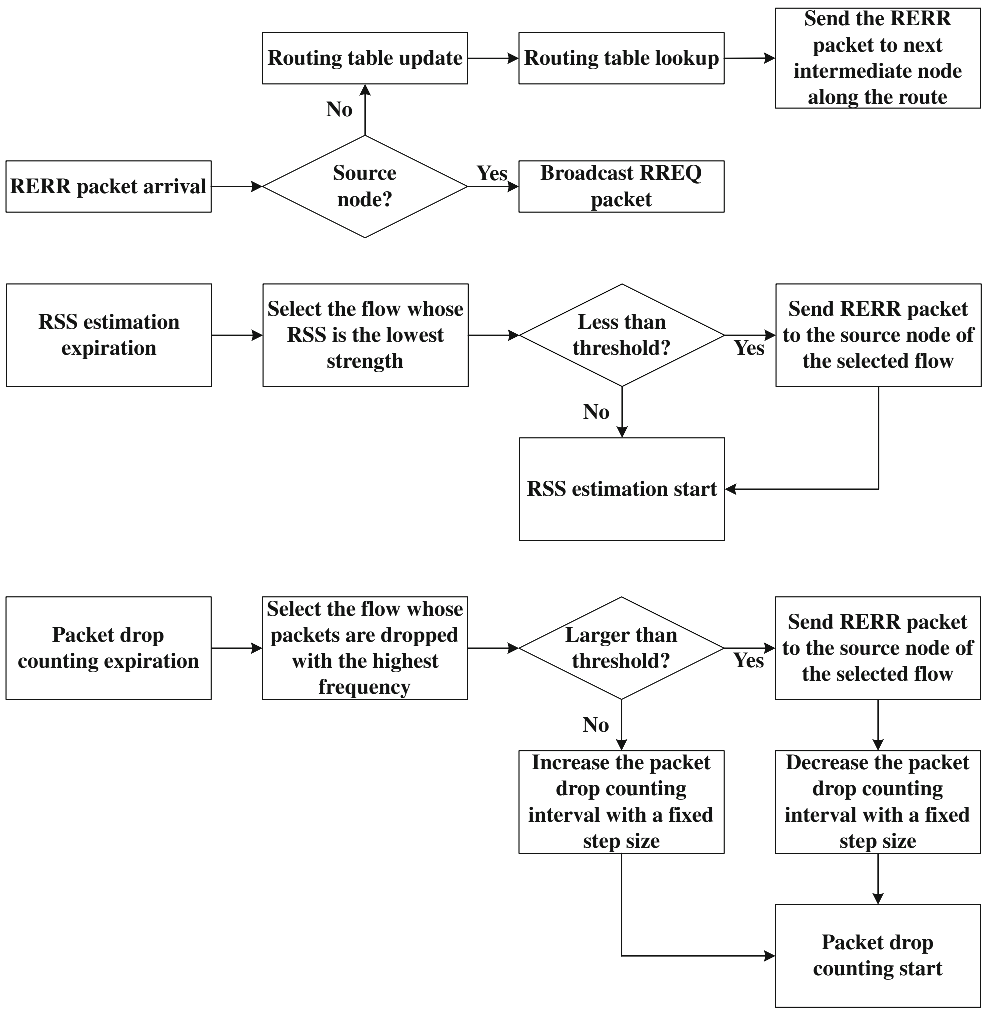


Fig. 5 Flowchart for proposed dynamic route maintenance mechanism

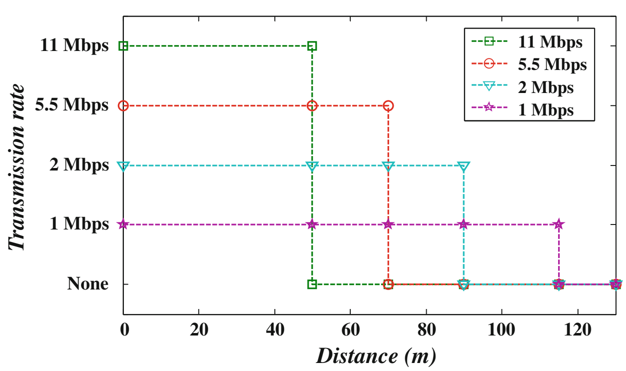


Fig. 6 The transmission rates according to distance in IEEE 802.11b multi-rate service

(5) A, H, Q, and R of Kalman filter are set to 1, 1, 10, and 0.5, respectively. The initial values of $\widetilde{rss}_{est}(0)$ and $\widetilde{P}(0)$ are fixed to 0.

(6) To verify the superiority of the proposed joint working algorithms, they are compared with four other methods: (I) AODV with FCFS/Drop-tail/RBAR (the routing algorithm at the network layer is AODV, the packet scheduling algorithm at the MAC layer is FCFS (First-Come, First-Served) service discipline and drop-tail policy, and the MAC layer adopts RBAR (Receiver-Based Auto-Rate)), (II) DSR with FCFS/Drop-tail/RBAR, (III) QAR with FCFS/Drop-tail/RBAR, and (IV) the proposed routing algorithm with FCFS/Drop-tail/RBAR.

(7) The total simulation time is set to 500 s.

3.1 Test of urgency mapping functions

To verify the performance of the proposed joint working algorithms, we preformed experiment with a variety of non-increasing functions. As shown in Fig. 7, we examined three types (e.g. convex, linear, and, concave) of non-increasing and differentiable urgency mapping functions in $0 < d_{residual}(t)/D_{\max} < 1$. The relation between the mapping functions and the corresponding routes is

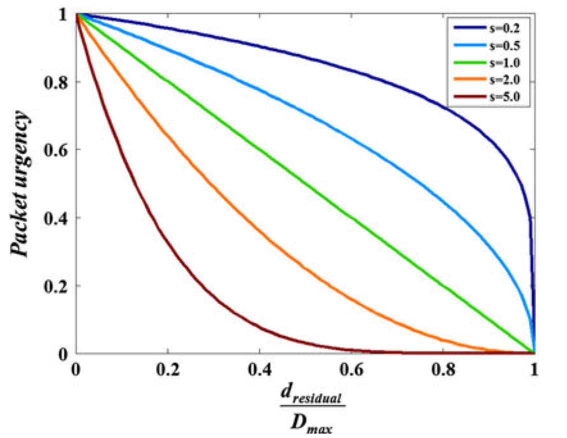


Fig. 7 The packet urgency mapping function: $u_{pkt} = f_{urg}\left(\frac{d_{residual}}{D_{\max}}\right) = \left(1 - \frac{d_{residual}}{D_{\max}}\right)^{s}$

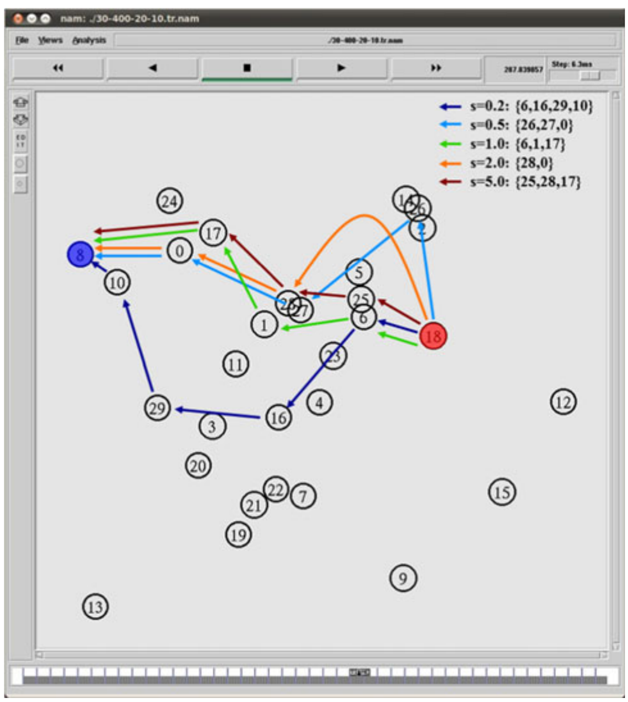


Fig. 8 Snapshot of routing paths at about 287 s

presented in Fig. 8, and the histograms of accumulated packet urgency values over the routes are shown in Fig. 9. It is obviously shown in Fig. 9 that the route contains more packets with large packet urgency values as the urgency mapping function becomes more concave, vice versa. Based on the observation, the convex mapping function with s = 2 is selected as the urgency mapping function of the proposed joint working algorithms in the followings.

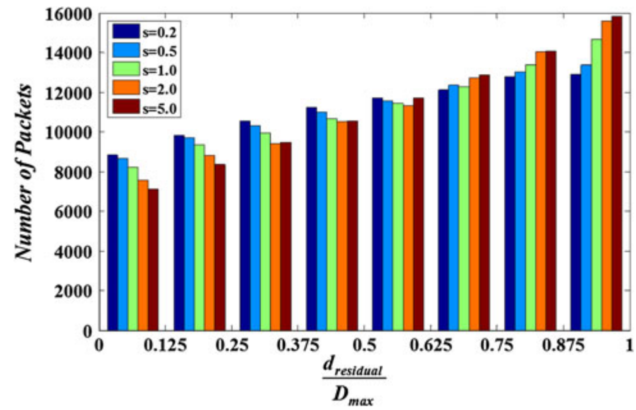


Fig. 9 Histograms of accumulated packet urgency values over the routes

3.2 Performance comparison with respect to end-to-end QoS metrics

In this section, the packet arrival rate within D_{max}, end-to-end delay, and routing overhead are adopted as end-to-end QoS performance metrics. The packet arrival rate within D_{max} is given in Fig. 10. It is observed that DSR with FCFS/Drop-tail/RBAR has significantly decreasing packet arrival rates within D_{max} as the number of flows increases. Actually, DSR with FCFS/Drop-tail/RBAR shows the worst performance because intermediate nodes may transmit RREP packets through stale cached routes, and then pollute local caches at the other nodes over MANETs. Consequently, the packet arrival rate within D_{max} decreases. QAR with FCFS/Drop-tail/RBAR also exhibits worse performance than the proposed joint working algorithms because it accounts for only the available bandwidth of a new route without considering existing routes. Thus, it may degrade the QoS of the existing routes by accepting a new route. Under a heavy traffic load, the proposed routing algorithm with FCFS/Drop-tail/RBAR presents worse performance than QAR with FCFS/Drop-tail/RBAR, as shown in Fig. 10(a). The reason is that the proposed routing algorithm with FCFS/Drop-tail/RBAR increases the network stress by transmitting packets that violate the end-to-end delay requirement, and it does not reflect the current network state effectively during the route establishment stage. In contrast, the proposed joint working algorithms provide better performance than any other method because they distribute the traffic load over the entire network and effectively control the packets accumulated in the buffer at each node.

The resulting end-to-end delay is presented in Fig. 11. As shown in the figure, the end-to-end delay curve of DSR with FCFS/Drop-tail/RBAR increases significantly when the number of flows increases. In fact, DSR with FCFS/Drop-tail/RBAR exhibits the worst performance,

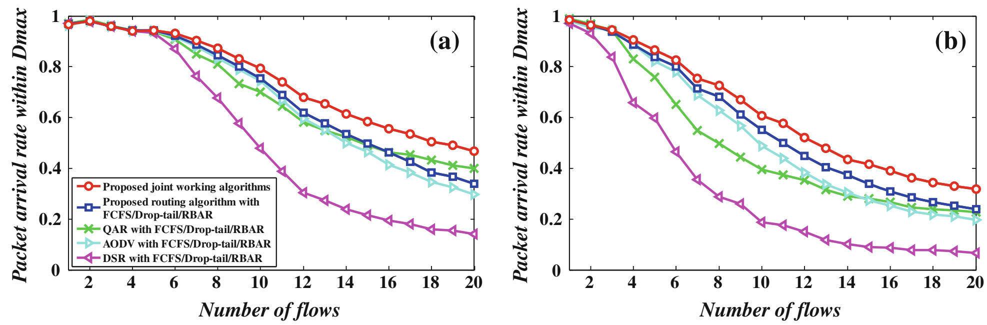


Fig. 10 Packet arrival rate within D_{max}: **a** 400 m × 400 m network topology with 30 nodes and **b** 600 m × 600 m network topology with 70 nodes

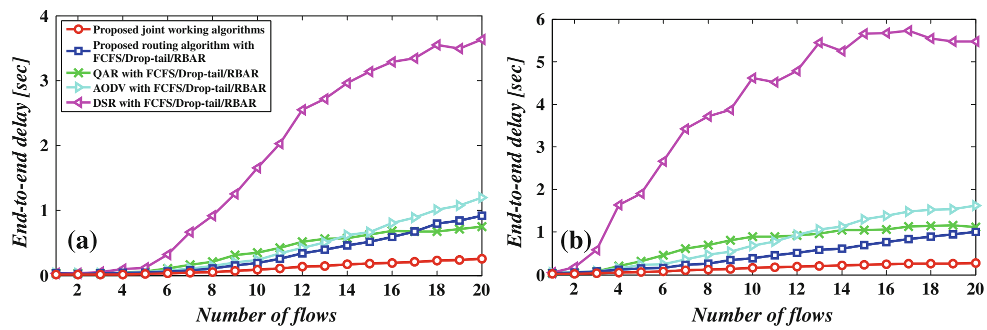


Fig. 11 End-to-end delay: **a** 400 m × 400 m network topology with 30 nodes and **b** 600 m × 600 m network topology with 70 nodes

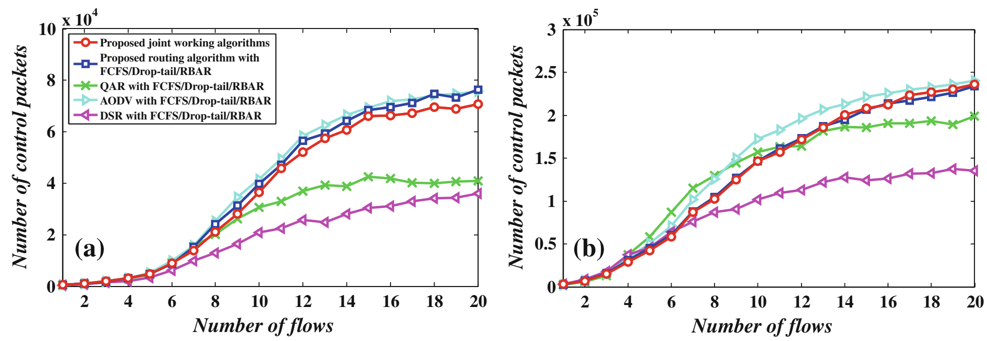


Fig. 12 Number of control packets: **a** 400 m × 400 m network topology with 30 nodes and **b** 600 m × 600 m network topology with 70 nodes

as observed in Fig. 11. Compared to AODV with FCFS/Drop-tail/RBAR and the proposed routing algorithm with FCFS/Drop-tail/RBAR, QAR with FCFS/Drop-tail/RBAR supports a low end-to-end delay under a heavy traffic load. However, the end-to-end delay of QAR with FCFS/Drop-tail/RBAR is greater than that of the proposed joint working algorithms because it does not consider the end-to-end delay requirement of each packet. The proposed joint working algorithms maintain the end-to-end delay of each packet within a tolerable range as far as possible. On the basis of these observations, the proposed joint working algorithms can provide better delivery service for delay-sensitive data transmission over multi-rate MANETs.

Now, we consider the routing overhead, which it is one of the most important factors in protocol design over multi-rate MANETs. Figure 12 shows the number of routing control packets. DSR with FCFS/Drop-tail/RBAR has the smallest routing overhead because it uses source routing instead of relying on a routing table at each node and route caching can reduce the route discovery overhead. It is observed in Fig. 12 that the control overhead of the proposed joint working algorithms is almost same as AODV with FCFS/Drop-tail/RBAR by adopting RREQ/RREP packet forwarding control mechanism.

3.3 Performance comparison with respect to proposed urgency metrics

In this section, we compare the proposed joint working algorithms with the other four methods in terms of the proposed urgency metrics when the number of connections is 14. The sum of node urgency values is provided in Fig. 13. It is apparently observed that DSR with FCFS/Drop-tail/RBAR has the highest sum of node urgency values among all algorithms, furthermore this curve

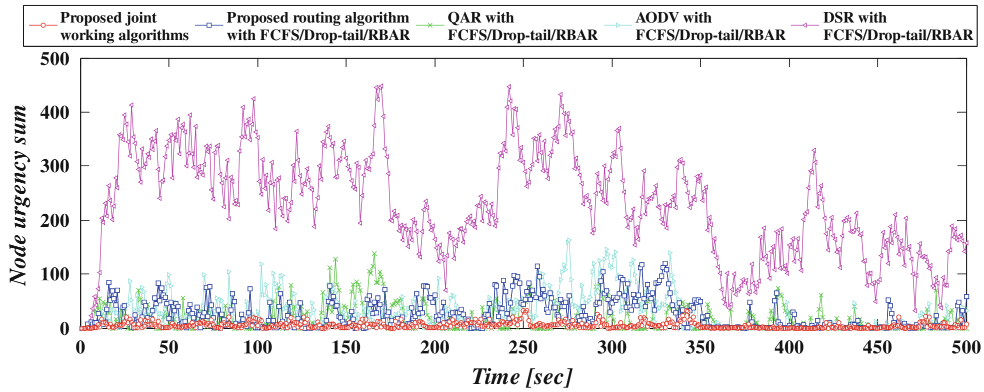


Fig. 13 Sum of node urgency values over the entire network

Table 3 Statistics of node urgency sum

Methods	Average node urgency sum	Maximum node urgency sum	Standard deviation of node urgency sum
Proposed joint working algorithms	5.163	32.159	5.955
Proposed routing algorithm with FCFS/Drop-tail/RBAR	24.318	119.997	27.288
QAR with FCFS/Drop-tail/RBAR	16.995	138.000	24.545
AODV with FCFS/Drop-tail/RBAR	28.713	162.667	33.324
DSR with FCFS/Drop-tail/RBAR	230.926	448.571	95.524

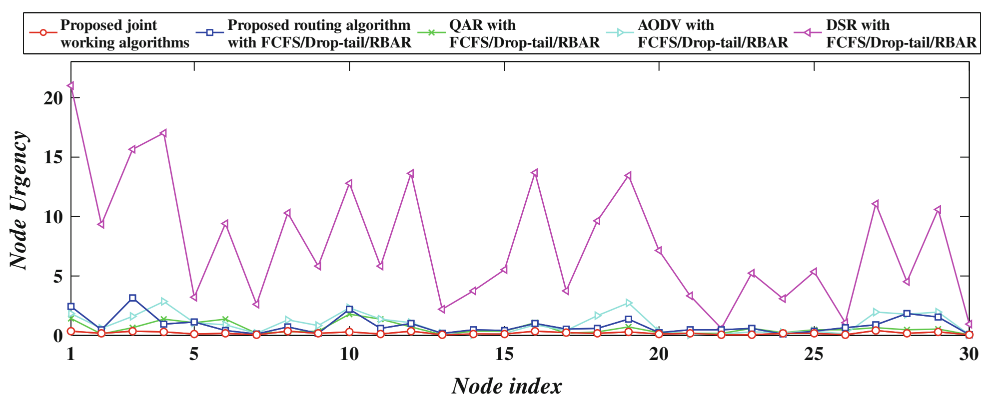


Fig. 14 Node urgency at each node

Table 4 Statistics of node urgency at each node

Methods	Average node urgency	Maximum node urgency	Standard deviation of node urgency
Proposed joint working algorithms	0.172	0.409	0.106
Proposed routing algorithm with FCFS/Drop-tail/RBAR	0.811	3.141	0.734
QAR with FCFS/Drop-tail/RBAR	0.567	1.767	0.470
AODV with FCFS/Drop-tail/RBAR	0.957	2.857	0.835
DSR with FCFS/Drop-tail/RBAR	7.698	21.021	5.218

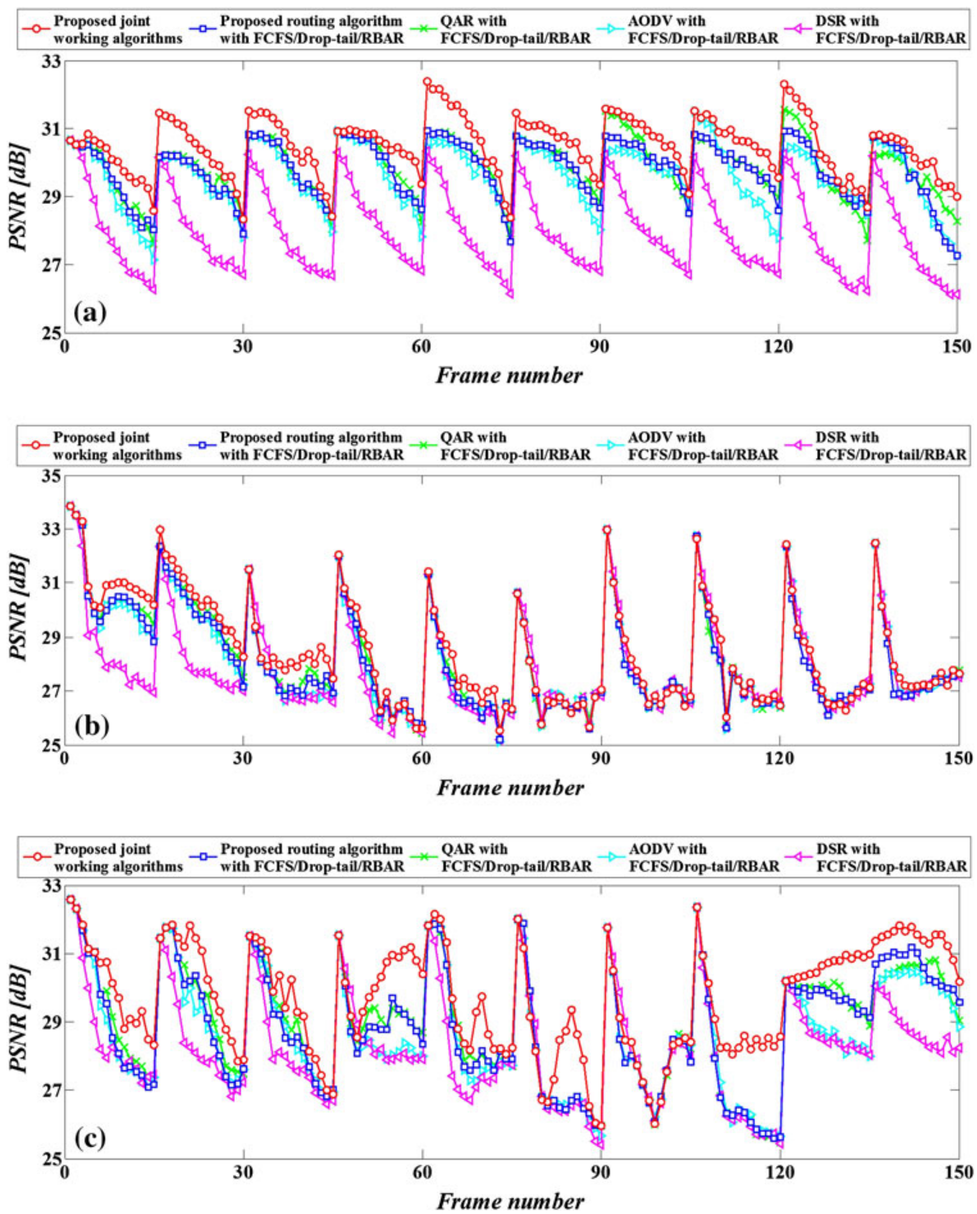


Fig. 15 PSNR values according to frame number: **a** City, **b** Crew, and **c** Foreman video sequences

Table 5 Average PSNR value comparison

Methods	Test video sequences (dB)		
	City	Crew	Foreman
Proposed joint working algorithms	30.490	28.469	29.713
Proposed routing algorithm with FCFS/Drop-tail/RBAR	29.807	28.137	28.859
QAR with FCFS/Drop-tail/RBAR	29.875	28.174	28.929
AODV with FCFS/Drop-tail/RBAR	29.615	28.109	28.642
DSR with FCFS/Drop-tail/RBAR	27.952	27.844	28.262

suddenly increase when the traffic is concentrated at only some nodes and/or when network congestion occurs. AODV with FCFS/Drop-tail/RBAR, QAR with FCFS/Drop-tail/RBAR, and the proposed routing algorithm with FCFS/Drop-tail/RBAR have an even higher sum of node urgency, although their curves are considerably lower and smoother than one of DSR with FCFS/Drop-tail/RBAR. However, the proposed joint working algorithms significantly reduce the sum of node urgency values. The results are summarized statistically in Table 3.

Fig. 16 Subjective video quality comparison of the 57th frame of Foreman video sequence: **a** Proposed joint working algorithms, **b** proposed routing algorithm with FCFS/Drop-tail/RBAR, **c** QAR with FCFS/Drop-tail/RBAR, **d** AODV with FCFS/Drop-tail/RBAR, and **e** DSR with FCFS/Drop-tail/RBAR

In addition, the node urgency at each individual node is presented in Fig. 14. This shows that the proposed joint working algorithms can maintain consistently low node urgency among active nodes, whereas some active nodes participating in data forwarding obviously have high node urgency in the other methods. The statistical results in Table 4 indicate that the proposed joint working algorithms can significantly decrease the average, maximum, and standard deviation of node urgency by distributing traffic over the entire network.

3.4 Performance comparison with respect to achievable video quality

It is well known that video traffic is very sensitive to delay and delay jitter. In this section, the achievable video quality is measured for 14 connections. During the experiment, the H.264/AVC JM 15.1 [39] video codec is employed. The test video sequences are the QCIF (Quarter Common Intermediate Format)-sized City, Crew, and Foreman. The video stream is encoded at 15 fps, and its target bandwidth is set to the value of the product of the packet size and the average number of packets received per second that satisfies the end-to-end delay requirement at the destination node. A group of pictures (GOP) consists of 15 frames (IPPPPPPPPPPPPPP). The performance measure is the achievable peak signal to noise ratio (PSNR) value without scene freezes. The resulting PSNR curves are presented in Fig. 15, and the results are summarized in Table 5. As shown in the table, it is obviously observed that the proposed joint working algorithms provide better PSNR values than the other methods. For the subjective quality comparison, some captured frames are presented in Fig. 16. It is apparently observed in the figures that the proposed joint working algorithms can support much better subjective video quality than those of the existing routing algorithms.

4 Conclusion

In this paper, we have proposed urgency-based joint working packet scheduling and routing algorithms that effectively support delay-sensitive data transmission over multi-rate MANETs. Basically, packet urgency, node urgency, and route urgency have been defined on the basis of the end-to-end delay requirement. Effective tightly coupled packet scheduling and routing algorithms have been designed based on these metrics. The experimental results have shown that the proposed joint working algorithms provide better service for delay-sensitive data transmission over multi-rate MANETs than the other methods, by distributing the traffic load over the entire network and effectively controlling the packets accumulated in the buffer of each node.

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