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EFT: a high throughput routing metric for IEEE 802.11s wireless mesh networks

Md. Shariful Islam · Muhammad Mahbub Alam · Md. Abdul Hamid · Choong Seon Hong · Sungwon Lee

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Abstract In this paper, we present a throughput-maximizing routing metric, referred to as expected forwarding time (EFT), for IEEE 802.11s-based wireless mesh networks. Our study reveals that most of the existing routing metrics select the paths with minimum aggregate transmission time of a packet. However, we show by analyses that, due to the shared nature of the wireless medium, other factors, such as transmission time of the contending nodes and their densities and loads, also affect the performance of routing metrics. We therefore first identify the factors that hinder the forwarding time of a packet. Furthermore, we add a new dimension to our metric by introducing traffic priority into our routing metric design, which, to the best of our knowledge, is completely unaddressed by existing studies. We also show how EFT can be incorporated into the hybrid wireless mesh protocol (HWMP), the path selection protocol used in the IEEE 802.11s draft standard. Finally, we study the performance of EFT through simulations under different network scenarios. Simulation results show that EFT outperforms other routing metrics in terms of average network throughput, end-to-end delay, and packet loss rate.

Keywords Wireless mesh networks · Routing metric · Medium access time · Hybrid wireless mesh protocol · 802.11s

Md. S. Islam · M. M. Alam · C. S. Hong (✉) · S. Lee
Department of Computer Engineering, Kyung Hee University, 1 Seocheon, Giheung, Yongin, Gyeonggi 449-701, Korea
e-mail: cshong@khu.ac.kr

Md. S. Islam
e-mail: sharif@networking.khu.ac.kr

M. M. Alam
e-mail: mahbub@networking.khu.ac.kr

Md. A. Hamid
Department of Information and Communications Engineering, Hankuk University of Foreign Studies, 89 Wangsan-ri, Mohyun-myon, Cheoin-Gu, Yongin-si, Kyonggi-do 449-791, Korea
e-mail: hamid@hufs.ac.kr

S. Lee
e-mail: drsungwon@khu.ac.kr

1 Introduction

Wireless mesh networks (WMNs) have revolutionized the way wireless access is available to end users. Due to their favorable characteristics, such as self-organization, self-configuration, self-healing, easy maintenance, high scalability, and reliable services, WMNs have gained acceptance from both industry and academia as a cost-effective approach to support high-speed last-mile connectivity and ubiquitous broadband access [1–3]. A schematic example of WMN architecture is presented in Fig. 1 that includes four different entities or network elements: mesh point portal (MPP), mesh access point (MAP), mesh point (MP), and legacy station (STA) [4]. An MPP is a gateway node that connects the WMN to a wired infrastructure, possibly to the Internet or local networks. There can be multiple MPPs deployed in a WMN. An MP is responsible for relaying traffic to other MPs or MAPs, whereas an MAP performs both the functions of an MP and an access point.

The architecture of WMN has some unique characteristics (i.e., use of a shared wireless medium, station-

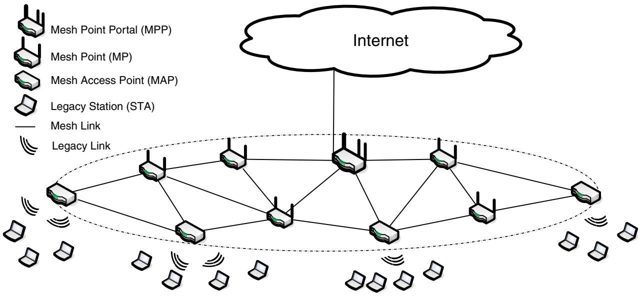

Fig. 1 Architecture of a typical WMN

ary wireless nodes, heterogeneous traffic patterns, both access point and relay functionalities) that differentiate a WMN from other networks and demand revisiting of existing routing protocols for use in WMN. The current draft D2.02 of IEEE 802.11s [4] has defined a path-selection protocol (this term is used to differentiate it from layer-3 routing) for WMN and termed as hybrid wireless mesh protocol. Hybrid wireless mesh protocol (HWMP) can operate in both proactive and reactive modes. The proactive mode is used when nodes are primarily static and traffic flows mainly to/from the Internet via the MPP to/from MAPs or MPs (inter-mesh traffic). In contrast, the on-demand mode is used mainly for traffic between MPs[1] within a mesh network (intra-mesh traffic). Both proactive and on-demand modes select paths using the *Airtime* metric, which takes into account the different data rates of different links but does not consider the effect of packet loss due to interfering transmissions/collisions from contending MPs and the traffic loads of contending neighbors.

In this study, we focus on designing a new routing metric that, together with HWMP, can be used for path selection in IEEE 802.11s-based wireless mesh networks. A routing metric can be defined as a parameter, value, or weight associated with a link (termed a link metric) or a path (termed a path metric). A routing protocol makes the routing decision based on the routing metric, and thus, it plays a critical role in determining the performance of a routing protocol.

Designing a routing metric for WMNs requires consideration of WMNs' unique architecture and the associated wireless networking environment. Unlike traditional wireless local area networks (WLANs), which have a centralized structure, WMNs have no hierarchy. To create a wireless mesh network, neighborhood MPs must overlap. MPs can operate autonomously, are capable of forwarding traffic, and can be a final source or destination. Therefore, redundant paths between MP and MPP or MP and MP can exist in a single WMN domain. Our goal is to design a routing metric that leverages the routing protocol to find the best path among the available paths for a specific flow.

Existing routing metrics usually select paths that require the minimum aggregate transmission time (which does not reflect the actual end-to-end delay) for a packet in a path and, thus, select the links with high data and success rates. However, a high-throughput path is one that can deliver a packet with the shortest end-to-end delay. Hence, a routing metric needs to consider factors other than the time that a packet uses the wireless medium (successful or unsuccessful transmission). However, for better utilization of the medium, network traffic need to be spread over the network, and thus, a routing metric needs to balance the network load to the available paths. Furthermore, the presence of high-priority packets might starve packets of low-priority flows, which a routing metric needs to consider. All the factors addressed above motivate us to design a routing metric that considers the forwarding time of a packet in a node. We define the forwarding time of a packet as the time that a node requires to successfully deliver it to the next hop.

In this paper, we propose a new routing metric, expected forwarding time (EFT), which selects the path that has the lowest end-to-end forwarding delay. The main contributions of our work are as follows. (1) Through detailed analyses and results, we first determine the factors that dominantly affect the EFT of a packet, and then, we extract the parameters that estimate EFT with low cost effectively. (2) We design the proposed routing metric based on extracted parameters and propose methods for low-cost estimation of the parameters. (3) We show how EFT can be incorporated with IEEE 802.11s's HWMP for path selection. (4) Finally, through simulations, we show that our proposed metric outperforms existing routing metrics in terms of average network throughput, end-to-end delay, and packet loss-rate.

The remainder of this paper is organized as follows: We discuss the major routing metrics designed for wireless mesh networks in Section 2. In Section 3, we present the detailed design of the proposed routing metric, and in Section 4, we investigate the performance of our proposed metric. Finally, we present our conclusions in Section 5.

2 Routing metrics for WMNs

In this section, we present and analyze some good examples of routing metrics specifically designed for wire-

[1]Hereafter, unless mentioned explicitly, the term MP is used for both MPs and MAPs.

less mesh networks. Comprehensive surveys of WMN routing metrics can be found in [5, 6].

2.1 Expected transmission count

Expected transmission count (ETX) [7] is one of the first routing metrics designed for wireless mesh networks. It is a link metric that estimates the number of transmission attempts (including retransmissions) required for a successful transmission on a particular wireless link. The ETX of a link is defined by Eq. 1 and the weight of a path is determined by summation of ETX values of all links along the path. ETX can be calculated as

$$\mathrm{ETX} = \frac{1}{d_{\mathrm{f}} \times d_{\mathrm{r}}}, \tag{1}$$

where d_{f} and d_{r} denote the delivery ratio in the forward and reverse directions, respectively. Though the computation of ETX is simple and captures the effect of packet loss rates, this routing metric has several limitations. ETX does not consider the impact of varying transmission rates of different wireless links and sizes of data packets. Another limitation of ETX is the use of an active broadcast-based probing scheme to measure the packet loss rates. This does not reflect the link quality accurately, since probing packets are small and broadcast packets use lower data rates than those of actual data packets [8]. ETX also does not consider the impact of intra-flow and inter-flow interference.[2]

2.2 Weighted cumulative expected transmission time

Draves et al. [9] proposed the weighted cumulative expected transmission time (WCETT) as a path metric for routing in multi-radio multi-channel WMNs. First, they proposed the expected transmission time (ETT) metric to address the issue of varying data rates of different wireless links. They calculated the time required to transmit a packet of size S on a link with a data rate B (raw data rate) using S/B and obtained the bandwidth-adjusted ETX or ETT using Eq. 2. Thus, the ETT of a link is the duration of time a node uses the medium to successfully deliver a packet to the next hop. ETT of the i-th link is defined by

$$\mathrm{ETT}_i = \mathrm{ETX}_i \times \frac{S}{B_i}, \tag{2}$$

where B_i is the data rate of the i-th link. Note that neither ETT nor ETX considers the presence of multiple channels. Furthermore, ETT characterizes the transmission time in the absence of interference.

Therefore, to find paths with less intra-flow interference and channel diversity, the authors in [9] proposed WCETT, which is defined by

$$\mathrm{WCETT} = (1-\beta) \times \sum_{i=1}^{n} \mathrm{ETT}_i + \beta \max_{1 \le j \le k} X_j, \tag{3}$$

where X_j is the summation of ETT of the links in a path p operating on channel j, k is the number of orthogonal channels available, and $0 \le \beta \le 1$ is a tunable parameter. Note that the first component of WCETT defines the end-to-end delay experienced in a particular path, while the second component accounts for channel diversity and finds the path with less intra-flow interference. However, WCETT does not consider the link quality and traffic loads of the contending nodes. Moreover, WCETT does not explicitly consider the effect of inter-flow interference. Therefore, in the presence of multiple flows in the network, it may end up finding paths through more dense areas where congestion is more likely and overall network throughput degrades.

2.3 Airtime

The Airtime metric is the default routing metric specified in the draft 2.02 of IEEE 802.11s [4]. This metric defines the amount of channel resources (C_{a}) consumed by transmitting the frame over a particular link and is calculated as

$$C_{\mathrm{a}} = \left[O + \frac{S_{\mathrm{t}}}{r}\right] \times \left[\frac{1}{1-e_{\mathrm{f}}}\right], \tag{4}$$

where O and S_{t} are constants that define channel access overhead and number of bits in the test frame, respectively. The input parameters r and e_{f} are the data rate in megabits per second and the frame error rate for a test frame, respectively. The rate r represents the data rate at which a node would transmit a frame of standard size S_{t} based on current conditions, and its estimation is dependent on local implementation of rate adaptation. The frame error rate e_{f} is the probability that, when a frame of standard size S_{t} is transmitted at the current transmission rate r, the frame is corrupted due to transmission error. The path metric is the sum of metric values of all links in the path.

This metric only takes the transmission rate and transmission error rate into consideration. In reality, the frame error rate due to transmission error does not reflect the actual link quality of a wireless link. A closer

[2]Intra-flow interference occurs when nodes in a single path attempt to transmit packets of the same flow and interfere with each other. Inter-flow interference is the interference suffered among concurrent flows.

look at the Airtime metric reveals that it is analogous to the ETT metric, where the first part of Eq. 4 reflects the transmission time and the second part measures the number of retransmissions required, like ETX. Thus, Airtime, like ETT, by not addressing the behavior of contending nodes, can route traffic to congested areas of the network because links with a higher data rate will always be given higher priority.

2.4 Metric of interference and channel-switching

In [10], the authors presented a new metric that incorporates both intra-flow and inter-flow interference. The metric of interference and channel-switching (MIC) for a path p is defined as follows:

$$\mathrm{MIC}(p) = \frac{1}{N \times \min(\mathrm{ETT})} \sum_{\mathrm{link}_l \in p} \mathrm{IRU}_l + \sum_{\mathrm{node}_i \in p} \mathrm{CSC}_i, \tag{5}$$

where N is the total number of nodes in the network and min(ETT) represents the smallest ETT in the network. IRU_l represents the interference-aware resource usage and CSC_i accounts for the channel switching cost.

However, MIC has some major drawbacks in terms of implementation. First, it assumes that all the nodes located in the collision domain of a particular link contribute to the level of interference, irrespective of whether those nodes are actually generating interfering traffic or not. Second, it requires up-to-date information regarding the ETT of each link—this requires significant overhead and may degrade the overall network performance.

3 Proposed routing metric

3.1 Problem description and motivation

In wireless mesh networks, multiple paths between any source–destination pair usually exist. End-to-end delay experienced by a packet on different paths might be different because of the shared nature of wireless links. Obviously, the path that will take less time to deliver a packet from a source to a destination is the better choice to route the packet. Our goal is to design a routing metric that can estimate the end-to-end delay experienced by a packet for the available paths, to enable the routing protocol to select the best path. If all the packets can be delivered with minimum delays, the overall network throughput will increase. In the following, we discuss the factors that affect the forwarding time of a packet in a node.

- *Transmission rate:* Traffic routed via a link with a higher transmission rate take less transmission time and should outperform links with lower transmission rates. Furthermore, due to the shared nature of the wireless medium, the transmission rates of neighbors (i.e., contending nodes) also affect the forwarding time. This is because a node has to wait for the medium to be free while neighbors keep the medium busy.
- *Success rate:* This represents the number of MAC layer transmission attempts required for a successful transmission. A lossy wireless link or a link that experiences more collisions results in multiple transmissions of a single packet on that link. This has a negative impact on the routing metric value.
- *Contending neighbors and their loads:* The number of contending/interfering nodes and their traffic loads in the neighborhood of a forwarding node has a large impact on the performance of WMNs. Furthermore, low overhead availability of this information is a challenge for WMNs.
- *Load awareness:* This represents the traffic load of the forwarding nodes in a path. If the forwarding nodes are loaded, their queues build up quickly and the queueing delays of the packets increase. In contrast, selecting a lightly loaded path balances the network load.
- *Traffic priority:* Wireless mesh networks are supposed to provide QoS to the flows (for example, EDCA [11] is used in the MAC layer) by supporting different classes of traffic. Therefore, packets have different priorities depending on the traffic class of the flows, and consequently, the forwarding time of a packet varies according to the traffic class. A low-priority packet might be starved in the presence of high-priority packets. A longer path or relatively worse path might produce better throughput for low-priority packets. Moreover, distributing the high-priority packets in different paths might also balance the load in the network.

Most of the routing metrics proposed in the literature and discussed in Section 2 (for example, ETT, MIC, and Airtime) determine end-to-end delay by summing up the value of ETT of the links in a path, as proposed by Draves et al. [9]. However, ETT does not address all the factors discussed above. More specifically, the medium access time depends not only on the number of retransmissions (or transmissions), but also on the time required for each transmission attempt. For example, a forwarding node with more contending nodes experiences more interruptions during the backoff process, which results in a greater medium access delay. Further, a contending node with low channel quality (or low transmission rate) or packets of larger size might force a node to freeze its backoff counter for a longer period. Moreover, a low-priority packet might starve for a long time, because of the presence of several high-priority packets. As a result, a path with the minimum expected delay for that traffic class is the best path for the packet, even though this might not be the best path in terms of transmission rate and success rate, i.e., ETT. This also ensures load-balancing in the networks.

Moreover, ETT might direct all the packets of the network towards a single path (i.e., the best path), leading to increased congestion and contention. As a result, the queuing delay of packets may increase and the network throughput may decrease due to under-utilization of network resources, if lightly loaded parallel paths exist with comparatively low quality. Furthermore, due to increased congestion and contention, a better path will turn into a worse path and ETT will choose a new path. But, due to the nature of ETT, it will forward all the packets to that new better path and the new path will in turn become worse.

The main challenge in designing a routing metric is therefore to find the parameters that incorporate the aforementioned factors; furthermore, the routing metric should take into account the availability of the value of each parameter with minimum overheads.

3.2 Design considerations

As mentioned previously, our goal is to design a routing metric based on the EFT of a packet in a node. In this section, we explore and justify the parameters that affect the forwarding time of a packet.

3.2.1 Number of retransmissions and transmission rate

The forwarding time of a packet for any outgoing link depends on the number of transmission attempts required to successfully deliver a packet for this link and the transmission rate of the link. Like ETT [9] and Airtime [4], EFT also considers both of these parameters. Furthermore, the size of the contention window (i.e., number of backoff slots) varies with the number of retransmissions.

3.2.2 Multiple traffic classes

IEEE 802.11s-based WMNs allow multiple classes of traffic (such as background, best-effort, video, voice, etc.) to co-exist, and thus, the medium access delay of a packet depends on the traffic class of the packet. If a high-quality path is dominantly accessed by the high-priority flows, the transmission of a low-priority packet might not defreeze its backoff counter due to consecutive transmissions of high-priority flows, and hence, its defer time may become very long. Therefore, the forwarding time of a packet is severely affected by the defer time, especially for low-priority flows. As a result, inclusion of the defer time in the forwarding time will select the path with the shortest delay for a particular traffic class. Such a path might not have the best ETT or ETX. However, this will select the best path for that traffic class and balance the load of the network for different traffic classes.

The medium access delay of the packets of the j-th traffic class, denoted as d_{a}^{j}, is given by

$$d_{\mathrm{a}}^{j} = \sum_{i=0}^{M'-1} \left[d_{\mathrm{d}}^{j} + d_{\mathrm{b}}^{j}(i) \right] + \sum_{i=1}^{M'-1} d_{\mathrm{c}}^{j}, \tag{6}$$

where M' is the maximum retransmission limit, d_{d}^{j} and d_{c}^{j} are the defer time and unsuccessful transmission time for the j-th traffic class, respectively, and $d_{\mathrm{b}}^{j}(i)$ is the backoff time of a packet of the j-th traffic class at the i-th transmission attempt. Note that each interruption during the backoff process is followed by a defer time.

The defer time for the j-th traffic class is given by

$$E[d_{\mathrm{d}}^{j}] = \begin{cases} \mathrm{AIFS}[j] + E[H] \times E[T], & j = 1 \\ \mathrm{AIFS}[j], & j = 2, 3, \end{cases} \tag{7}$$

where AIFS[j] ($j = 1, 2, 3$, and a higher value of j indicates high priority) is the arbitrary inter frame space according to the 802.11e [11], $E[H]$ is the expected number of interruptions by neighbors with higher-priority packets in a single defer time and $E[T]$ is the expected interruption time (see Appendix A for more details).

The analytical results in Fig. 2 show the impact of defer time on the average medium access delay for different traffic classes. As shown in the figure, a node with low-priority traffic experiences higher backoff delay in the presence of high-priority traffic. Further, the medium access delay increases as the number of high-priority contending nodes increases—due to the increase of defer time. Thus, the quality of a link for a particular traffic class is affected not only by the conventional parameters (like ETX, ETT), but also by the contending high-priority traffic.

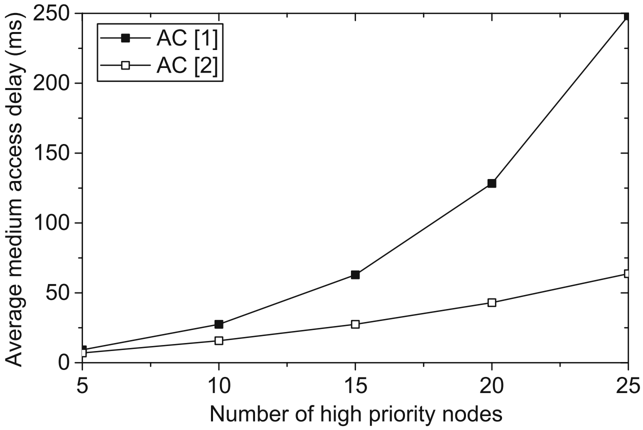

Fig. 2 Average medium access delay for different traffic classes due to the variation in defer time. Each node uses the same data rate and packets of same size

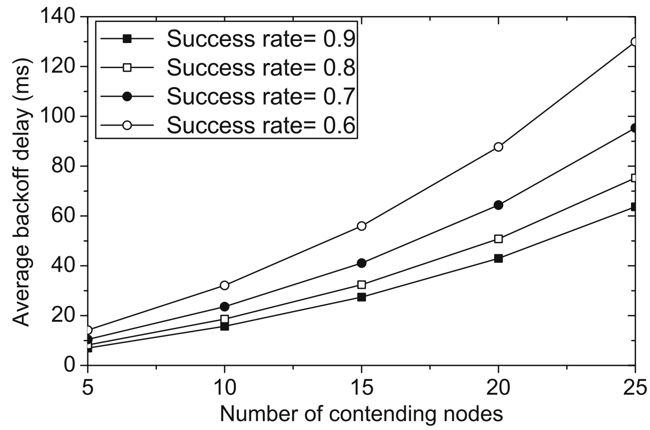

Fig. 3 Impact of number of contending nodes on the average backoff delay. Each node uses the same data rate and packets of same size

3.2.3 Neighbor density and loads

A packet forwarded in a path with lightly loaded and/or a fewer numbers of contending nodes experiences less delay. However, it is not trivial to obtain the information about contending neighbors. Existing routing metrics (for example, MIC [10]) count the number of contending nodes and cannot differentiate between neighbors and contending nodes. This is because a neighbor's queue might be empty, and hence, such a node will not contend for the link. Further, the loads of the neighbors are updated by periodic explicit messages [12], which incur a huge control overhead. We use a novel method to implicitly measure the number of contending nodes and their loads.

In CSMA/CA networks, nodes go through a backoff procedure before accessing the medium. A node is forced to freeze its backoff counter upon hearing a transmission from the neighbors. The number of interruptions in a transmission attempt depends on the number of contending nodes and the probability at which the nodes attempt to access the medium, where the load of a contending node determines the probability at which it accesses the medium. A node experiences more medium-access delay if it has a dense neighborhood with heavily loaded nodes. The expected backoff delay in the i-th transmission attempt, $E[d_{\mathrm{b}}^{j}(i)]$, can be calculated as

$$E[d_{\mathrm{b}}^{j}(i)] = E[w_i^j] \times d_{\mathrm{s}} + E[b_i^j] \times (d_{\mathrm{b}} + d_{\mathrm{d}}^{j}), \tag{8}$$

where $E[w_i^j]$ is the expected number of idle slots for the j-th traffic class and d_{s} is the duration of a generic slot, $E[b_i^j]$ is the expected number of interruptions for the j-th traffic class, and d_{b} is the expected duration of an interruption. Therefore, the expected backoff delay, $E[d_{\mathrm{b}}^{j}]$, can be calculated as

$$E[d_{\mathrm{b}}^{j}] = \frac{p_{\mathrm{s}}}{1-(1-p_{\mathrm{s}})^{M'}} \sum_{i=0}^{M'-1} (1-p_{\mathrm{s}})^{i}.E[d_{\mathrm{b}}^{j}(i)], \tag{9}$$

where p_{s} is the probability of success of a transmission. Appendix B details the derivation of Eqs. 8 and 9.

Figure 3 shows that the average backoff delay of a node increases with an increase in the number of contending nodes, because an increased number of contending nodes results in more interruptions. In contrast, a better success rate requires fewer retransmissions, and hence, the backoff delay decreases with increasing success rates.

Figure 4 shows the average backoff delay of a node with different offered loads. As the load of a neighbor of a node increases, the neighbor attempts to access the medium with higher probability, and hence, the backoff

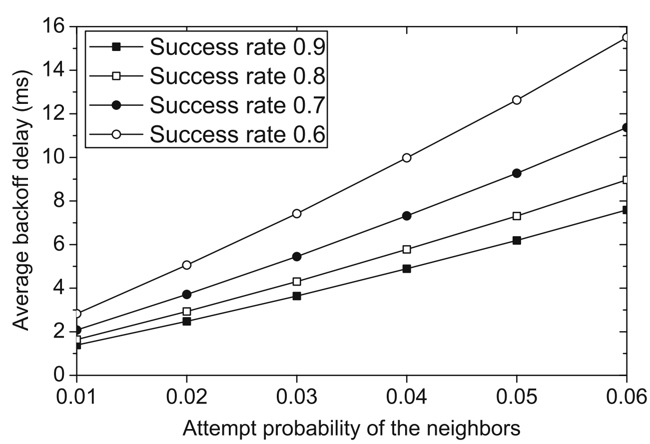

Fig. 4 Impact of loads of contending nodes on the average backoff delay. Each node uses the same data rate and packets of same size

delay of the node increases with increasing neighbors' loads. Therefore, a node experiences less backoff delay with lightly loaded contending nodes.

A closer look at Eq. 8 indicates that the backoff delay is determined dominantly by the number of interruptions in a single transmission attempt. Therefore, the expected number of interruptions implicitly determines the density and load of the neighborhood of a node.

3.2.4 Transmission rate and packet size of neighbors

The medium access delay (i.e., backoff delay) depends not only on the density and loads of the contending nodes (as mentioned in Section 3.2.3), but also on the transmission time (i.e., the busy time in Eq. 8) of each interruption. To the best of our knowledge, none of the existing routing metrics address this issue.

The transmission time of a neighbor (i.e., the duration of an interruption) depends on the channel quality and the packet size of the contending nodes. If a contending node experiences bad channel quality (and, hence, transmits at a lower rate) or transmits a very large packet, the transmitting node might have to wait for a longer period of time in a freezing state. However, the availability of neighbors' transmission rates and packet size at any node requires periodic control packet exchange among neighbors, which incurs control overheads and might not be feasible for a rapidly changing wireless environment.

Figure 5 shows the impact of the transmission rate and packet size of the neighboring nodes on the backoff delay of a transmitting node. The medium-access delay of a node increases if its neighbors transmit at a lower rate or they transmit larger data packets. The reason for this is that a node needs to stay in the freezing state for longer periods in both cases. This strongly suggests that the neighbors' transmission rates and packet size should be included in the routing metric; these factors are largely ignored in existing routing metrics. Therefore, we include the expected freezing time of an interruption in our routing metric.

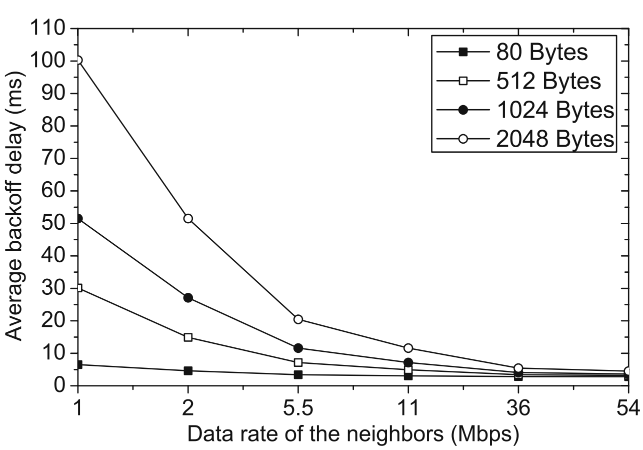

Fig. 5 Impact of transmission rate and packet size of the contending nodes on the average backoff delay. Number of contending nodes and their transmission probabilities are fixed

3.2.5 Queueing delay of forwarding nodes

The queuing delay is the amount of time a packet spends waiting in the transmitter's queue before it gets the chance of transmission. Therefore, if a packet is scheduled to forward through a node that already has enough packets in the transmission queue, it will have to wait until other packets in the queue finish their transmission successfully. If we consider only service time as the routing metric, most of the packets will be forwarded on paths with less service time. This will increase the queue size and, eventually, the queuing delay of a packet. However, if the same packet is transmitted through a lightly loaded node, it will experience less queuing delay, and the end-to-end delay will decrease. Therefore, traffic should be split in such a manner that load-balancing can be achieved in the forwarding nodes. This motivates us to incorporate the queuing delay of a packet of a particular traffic class in our proposed routing metric.

3.3 EFT: the proposed metric

In this section, we explain the proposed routing metric. The routing metric assigns a weight value, which represents the required EFT of a packet to successfully deliver it from a node to the next hop. In Section 3.2, we already explored the parameters that incur delays to a packet in a node. The EFT of a packet is the sum of these delays. The associated delays are narrated briefly in the following:

- The transmission time of a packet in a single transmission attempt is the duration wherein the packet uses the wireless medium. The transmission time depends on the data rate and the packet size.
- The idle time is the duration of the backoff slots that a node requires to access the wireless medium. This also increases with the increasing number of retransmissions.
- The waiting time due to traffic priority is represented by the defer time and is the durations at the beginning of a transmission and before resuming the backoff after each interruption.
- The freezing time represents the contending neighbors and their loads. The freezing time in each

interruption is the sum of the defer times and the duration of the interruptions (i.e., the duration of the transmission of the neighbor). The number of interruptions (i.e., transmission by the contending neighbors) also affects the freezing time.

– The queueing delay of a packet is the time that a packet waits in the queue before it starts any transmission attempt.

A successful transmission of a packet might require a number of transmission attempts. The EFT of a packet is the sum of the times that a packet requires both successful and unsuccessful transmission attempts. Let ET_i^j denote the expected time required for a packet of the j-th traffic class in the i-th transmission attempt. Thus, ET_i^j is the sum of aforementioned delays and is given by

$$\mathrm{ET}_i^j = d_d^j + \sum_{k=0}^{E[w_i^j]} d_s + \sum_{k=0}^{E[b_i^j]} [d_\mathrm{b} + d_d^j] + d_\mathrm{t} + d_q^j \tag{10a}$$

$$= E[w_i^j] \times d_s + E[b_i^j] \times d_\mathrm{b} + (1 + E[b_i^j]) \times d_d^j + d_\mathrm{t} + d_q, \tag{10b}$$

where d_t is the transmission time of a packet and d_q^j is the queueing delay of a packet of the j-th traffic class.

For a successfully delivered packet, the EFT is the sum of the expected time in each transmission attempt. If the number of the transmission attempts (i.e., ETX) required for a successfully delivered packet is M, the EFT of a packet of the j-th traffic class for the l-th link, EFT_l^j, is given by

$$\mathrm{EFT}_\mathrm{l}^j = \sum_{i=1}^{E[M]} \mathrm{ET}_i^j \tag{11}$$

Combining Eqs. 10b and 11, we have the following:

$$\mathrm{EFT}_\mathrm{l}^j = \underbrace{E[M] \times E[w_i^j] \times d_s}_{\text{Expected Idle time}} + \underbrace{E[M] \times E[b_i^j] \times d_\mathrm{b}}_{\text{Expected freezing time}} + \underbrace{E[M] \times (1 + E[b_i^j]) \times d_d^j}_{\text{Expected defer time}} + \underbrace{E[M] \times d_\mathrm{t}}_{\text{ETT}} + \underbrace{E[d_q^j]}_{\text{Queue delay}}. \tag{12}$$

In Eq. 12, the first term indicates expected duration of the idle slots that a node needs to wait during backoff; this value is dominant when a node requires a large number of retransmissions.

The second term indicates two factors mentioned in Section 3.2: (1) the number of interruptions during the medium access period, which provides an indication of the number of contending nodes and their loads, and (2) the channel quality (data rate) and the packet size of the contending nodes. The third term indicates the defer time for a packet, and it has a significant influence on the flow of lower traffic classes. The fourth term indicates the ETT metric mentioned in [9], and finally, the fifth term indicates the average queueing delay of a packet of a particular traffic class.

Now, the path metric referred to as cumulative expected forwarding time (CEFT) for a flow of the j-th traffic class for a path of h hops can be given by

$$\mathrm{CEFT}^j = \sum_{l=1}^{h} \mathrm{EFT}_\mathrm{l}^j. \tag{13}$$

Note that CEFT is an additive metric, and as the number of links in a path increases, the value of CEFT increases. Among multiple paths between a particular source–destination pair, the path having the lowest CEFT will be the best path.

3.4 Implementation issues

3.4.1 Estimation of the required parameters

To calculate the value of EFT, a forwarding node needs to estimate the values of the parameters presented in Eq. 12. The value of d_t depends on the selected rate of a link, and we assume that a rate adaptation mechanism that selects the rate for each link is running. There are many rate adaptation techniques available in the literature (for example, [13–17]). However, we opt to use the automatic rate fallback (ARF) [13] technique to estimate the data rate because of its ease of implementation and wide acceptance by different WLAN vendors. Therefore, we assume that each forwarding node knows the value of $E[M]$ and d_t for each of the outgoing links.

Each node measures the number of interruptions in a single transmission attempt and the duration of the interruptions for each traffic class. We use exponentially weighted moving average (EWMA) to estimate the expected value of the parameters given by

$$E[b_i^j(t)] = \alpha \times b_i^j(t) + (1 - \alpha) \times E[b_i^j(t-1)] \tag{14a}$$

$$E[d_\mathrm{b}^j(t)] = \beta \times d_\mathrm{b}^j(t) + (1 - \beta) \times E[d_\mathrm{b}^j(t-1)], \tag{14b}$$

where α and β are the tuning parameters to smooth the estimated value. Following the same procedure, the average queueing delay is estimated by measuring the average queue size and queue service time for each traffic class. Accordingly, each node estimates the defer time of each traffic class.

3.4.2 Incorporating EFT in HWMP

The HWMP is the default path selection protocol defined in draft 2.02 of IEEE 802.11s. As specified in [4] and [18], only a single active routing protocol and a corresponding path selection metric should be used in a single WMN. HWMP uses the Airtime metric as the default routing metric. Instead, we incorporate EFT as the routing metric to be used in HWMP. In the current draft, the metric identifier value of the Airtime metric is set to 0, whereas values 1–254 are reserved for future use. We therefore use the value 1 for EFT in the active path selection metric identifier field. This information is embedded in the "mesh configuration" element that is used to advertise mesh services. Whenever an MP establishes a link with another MP, it uses peer link open and peer link confirm frames. The "mesh configuration" element is contained on these frames and also in beacon frames transmitted by the MPs. Thus, all the MPs can be notified about the default routing protocol (HWMP) and the metric (EFT) they are going to use.

HWMP is a blend of on-demand (reactive) and proactive routing mechanisms. In the on-demand mode, a source MP broadcasts a path request (PREQ) message requesting a route to the destination with the EFT field initialized to zero. Note that an MP may receive multiple copies of the same PREQ from a source through different paths. After receiving a PREQ, an intermediate MP creates a path to the source or updates its current path if the PREQ contains a greater sequence number, or the sequence number is the same as in the current path and the PREQ offers a better EFT value than the current path. If a new path is created or an update occurs, the PREQ is then re-broadcasted with an updated metric field that reflects the CEFT of the path to the source. The destination MP unicasts a path reply (PREP) message after creating or updating a path to the source.

In contrast, the *proactive tree-building mode* can be executed in two ways to let the MPs in the WMN create a path with the root or portal MP (MPP). First, in the *proactive PREQ* mechanism, the root MP periodically broadcasts a proactive PREQ message with increasing sequence number, destination address set to all 1s, and the EFT value initialized to zero. After receiving a proactive PREQ, each MP creates or updates its forwarding information to the root MP, updates the EFT value and hop-count of PREQ, and retransmits the updated PREQ. A proactive PREP from an MP establishes the path from the root MP to that MP.

In the case of the *proactive RANN mechanism*, the root MP starts to periodically broadcast a root announcement (RANN) message that propagates the metric information across the network. Upon reception of a RANN message, an MP that wants to create or refresh a path to the root MP sends a unicast PREQ to the root MP. The root MP then unicasts a PREP in response to each PREQ. The unicast PREQ creates the reverse path from the root MP to the originating MP, while the PREP creates the forward path from the MP to the root MP.

3.5 Discussions

Some minor details of the proposed routing metric are worth mentioning. First is the computation cost of the metric, which includes the time complexity and the message overhead. Each of the parameters used in the metric contributes little computation cost and may or may not require message overhead. All the existing mechanisms need to determine the expected number of retransmissions and the data rate, and this depends on the specific algorithm used to measure the parameters. We have used ARF, which does not need any extra message to exchange among the neighbors and only needs to count the number of retransmissions based on local data and accordingly, selects the desired data rate. The other parameters used in the metric are locally calculated by the nodes and, thus, do not need to exchange any control message. Furthermore, each node uses a local variable to store the value of the parameters, and the computational cost associated with measuring, updating, and averaging the values is negligible.

Second is the energy consumption for the overheads associated with the metric computation. Since the computational cost of the metric is negligible and it requires no extra control message (depends on the rate selection mechanism), the metric is assumed to be energy efficient. Furthermore, the mesh nodes are supposed to be static and are not energy-constrained [1]. Thus, energy consumption might not be so critical for the mesh networks.

Third is the stability of the proposed metric and the performance of the routing protocol. EFT is a dynamic metric and its value changes with the change of the network states. When EFT is used with an on-demand routing protocol, the changed metric value only affects the selection of new paths and the existing paths remain unchanged unless any of these paths is broken. Therefore, we assume that incorporating EFT with an on-demand routing protocol provides stable network operations. In contrast, a dynamic (or network-status-aware) metric might affect the stability when used with a proactive routing protocol. Therefore, the operation of a proactive routing protocol with a dynamic metric requires careful handling. To combat that, EFT smoothes the abrupt changes of the metric value by using EWMA to estimate the value of each parameter. Besides, HWMP can increase the stability by increasing the duration of the route update intervals. Moreover, we have restricted HWMP to enforce a route change only if a new path provides 20% improvement in the metric value. We have found in the simulation that the above-mentioned methods provide route stability for most of the simulation runs, and the improved results found in the simulation justify that.

Finally, we have validated the performance of EFT only by simulations. Considering the similarities of the EFT and Airtime metrics, we believe that EFT can be implemented with HWMP. The performance of EFT in a real implementation will further justify the effectiveness of the metric, and our future plan is to implement this with an enhancement of the routing protocol.

4 Performance evaluation

We have done extensive simulations to evaluate the performance of EFT and compare it with the well known existing routing metrics, namely, ETX, ETT, Airtime, and MIC, using ns-2 [19]. To measure and compare the results, we have used a mesh network with 50 stationary mesh routers (MP) randomly deployed in an area of 1,500 × 1,500 m. We have used both intra-mesh flows and flows that have an end-point outside the WMN (i.e., inter-mesh flow). We have considered a packet size of 1,024 bytes. The transmission range is set to 250 m and the interference range (carrier sense range) is set to 550 m. We consider UDP to be the transport layer and we assume that all flows generate data at a constant rate. The sources of the flows are randomly chosen for both intra- and inter-mesh traffic, whereas the destination is the gateway node for inter-mesh traffic. The on-demand mode of HWMP is used for intra-mesh traffic and the proactive mode is used for inter-mesh traffic. Each simulation run has been executed 15 times, and the average results are plotted in the graphs. The error bars in the graphs parallel to the y-axis indicate the variations of the obtained results from the presented average values and, thus, show the minimum and maximum values obtained among the simulation runs. For each simulation run, the active nodes are randomly selected (as long as the number of nodes is less than 50), and the source node for each of the flows is also randomly selected. Thus, the variations in the obtained results mainly occur due to the randomness of the topology.

We have considered the following performance metrics: (1) average network throughput—the sum of the size of the total data packets received by the destinations (MPP, in the case of the outgoing inter-mesh traffic) per unit time, (2) average end-to-end delay—the average delay experienced by all successfully delivered packets (for inter-mesh traffic, delay is measured only within the mesh network), and (3) packet loss rate—this indicates the ratio of the packets that are lost in the path to the number of packets generated by the sources. Simulation results for different network scenarios are shown to demonstrate the impact of network size, traffic load, variable link quality, and traffic classes on the performance metrics.

4.1 Scenario 1: impact of network size

In this scenario, we have assumed that 20 flows exist in the network with a sending rate of 5 pkts/s. To determine the impact of network size, we gradually increase the number of nodes in the network from ten to 50 nodes. We consider the case of a multi-radio, multi-channel environment, where each mesh node is equipped with two 802.11b radios and each radio operates on one of the three available channels. As shown in Fig. 6, EFT outperforms other existing metrics in terms of average network throughput with an increasing number of nodes. The average network throughput achieved using ETT and Airtime metrics are almost identical, as both consider the link with higher data rate as the better one and do not consider the effect of interference. MIC, on the other hand, performs slightly better than the aforementioned metrics as it chooses the next hop based on the number of neighbors. EFT outperforms MIC as it considers the actual number of

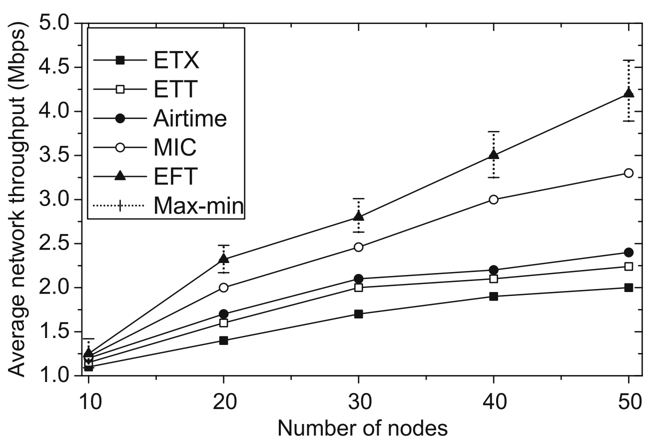

Fig. 6 Average network throughput for multi-radio, multi-channel node with IEEE 802.11b interface. Each flow maintains a rate of 5 pkts/s

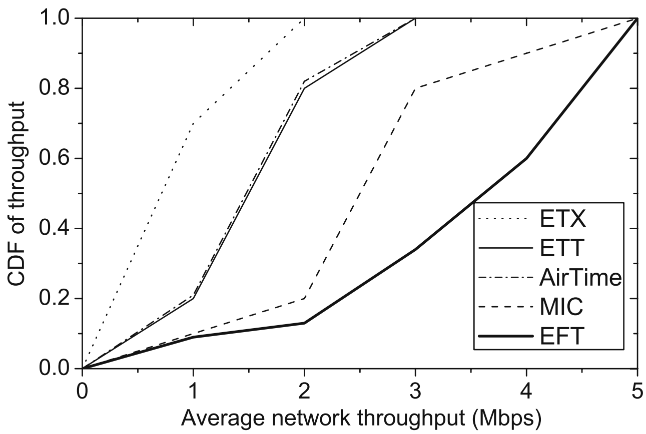

Fig. 7 CDF of average network throughput, where CDF is defined as the probability that throughput is less than or equal to a given value

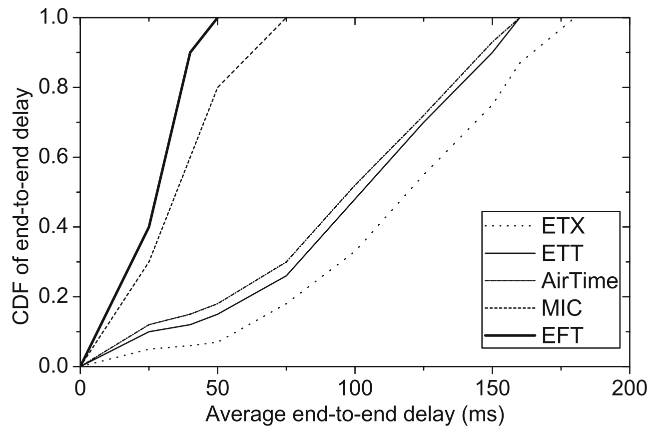

Fig. 9 CDF of end-to-end delay, where CDF is defined as the probability that delay is less than or equal to a given value

interfering nodes and their loads by incorporating the total forwarding delay. As the number of nodes and their loads increase, the EFT values of nodes with more contending nodes increase, which allows EFT to choose paths around less-congested areas. This also indicates that EFT achieves load-balancing by making efficient use of the medium. We also analyze the simulation results using the help of cumulative distribution function (CDF), where CDF plots for throughput in Fig. 7 show that, for EFT, the throughput is above 4.5 Mbps 80% of the time, whereas for MIC and ETT, the throughput is higher than 2.8 and 2 Mbps, respectively, 80% of the time.

Figure 8 demonstrates that EFT outperforms all the other metrics in terms of average end-to-end delay. Unlike the existing metrics, where in most cases only the transmission time of a packet is considered, EFT includes every possible delay in a node. Therefore, the end-to-end delays of the packets in EFT are minimal compared to the other metrics. The CDF plot in Fig. 9 also shows that, for 90% of the time, the end-to-end delay for EFT is less than 50 ms, whereas it is much higher for ETX, ETT, Airtime, and MIC metrics.

Figure 10 shows that the average packet loss rate achieved using EFT is lower than that of other metrics. In the initial stage, when the number of nodes is less (i.e., network connectivity is low), packet loss rates are higher for most of the metrics as the number of available paths is less. As the number of nodes increases, available paths also increase and packet loss rates tend to decrease. However, due to the efficient distribution of the traffic and the fact that EFT prefers paths through less congested network areas, it achieves better results in terms of packet loss rate.

4.2 Scenario 2: impact of traffic load

In this scenario, we have assumed that the network size is fixed and there are a fixed number of flows in

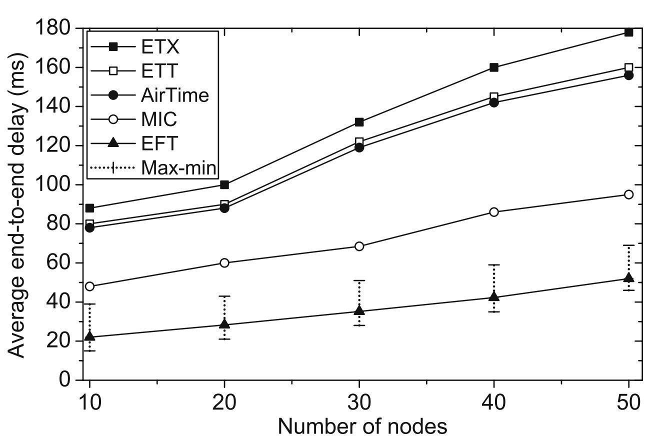

Fig. 8 Average end-to-end delay for 20 active flows with varying numbers of nodes

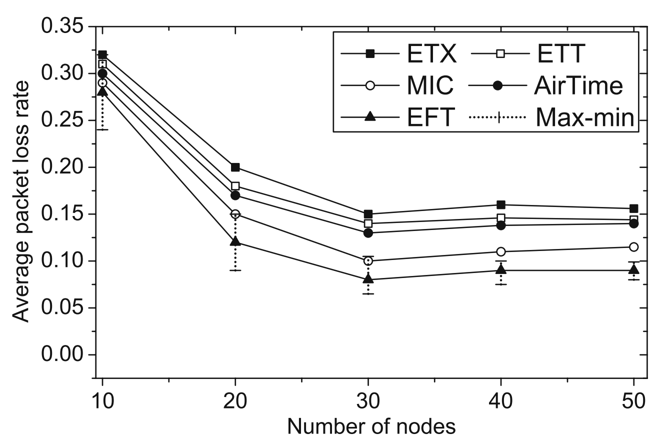

Fig. 10 Average packet loss rate for varying numbers of nodes with 20 flows

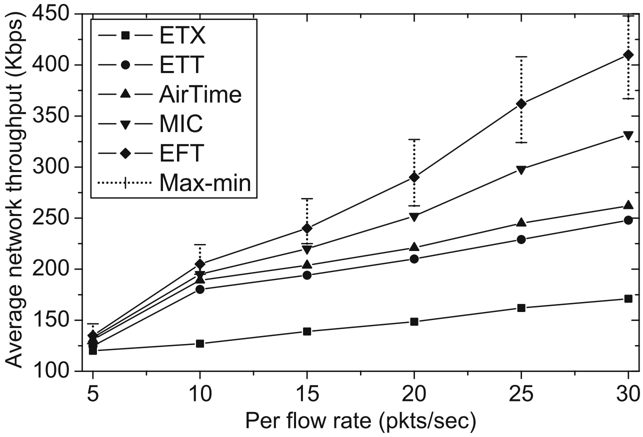

Fig. 11 Average network throughput for different traffic loads of the flows

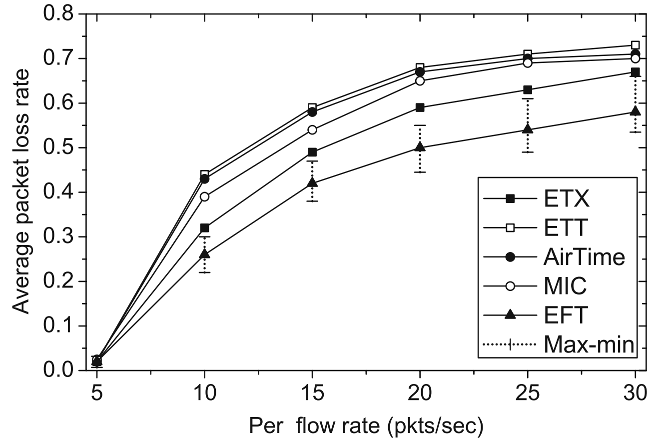

Fig. 13 Average packet loss rate with increasing loads of the active flows

the network. There are 50 nodes in the network, and 20 flows are generating data with randomly selected sources and destinations. We gradually increase the data generation rates of the flows from 5 to 30 pkts/s to measure the impact of traffic load on the routing metrics. Note that the nodes are operated in multi-radio and multi-channel mode. Figures 11, 12, and 13 show the average network throughput, end-to-end delay, and packet loss rates, respectively, for different metrics under various network loads.

Most existing routing metrics only consider link quality, and thus, the network throughput does not increase at the expected rate with increasing loads. As shown in Fig. 11, for ETX, ETT, and Airtime metrics, the network throughput increases very slowly with increasing loads, because all these metrics forward the packets toward the best path. In contrast, MIC selects the path based on the number of neighbors of

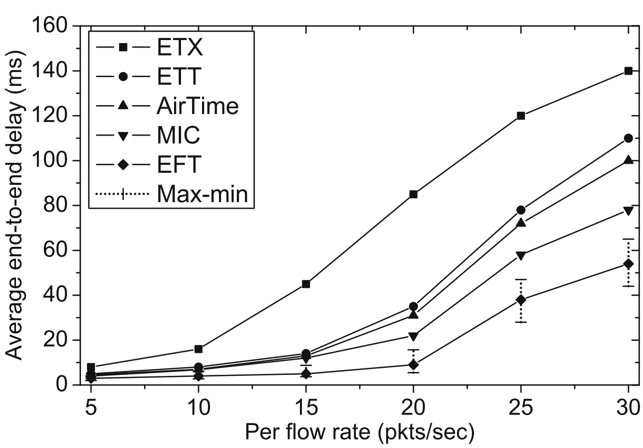

Fig. 12 Average end-to-end delay with increasing loads of the active flows

the forwarding node and, hence, achieves a slightly better network throughput than others. However, MIC cannot differentiate a neighbor and a contending node and, therefore, cannot estimate the ultimate load of the neighbors. On the contrary, EFT selects paths considering the number of contending nodes and their loads (i.e., the expected number of interruptions in each transmission attempt) and, therefore, outperforms all the existing metrics in terms of achieved throughput. Note that EFT balances the loads of the network by directing the traffic toward the lightly loaded zone of the network, and hence, it spreads the traffic over the network.

Figure 12 shows the average end-to-end delays for different metrics. EFT performs better in terms of average end-to-end delay than ETX, ETT, Airtime, and MIC. Because ETT and Airtime only consider the data rates of links and prefer links that have higher data rates, both these metrics tend to forward all packets to the same path, which results in network congestion. Also, MIC does not balance the traffic load over the network nodes and, thus, creates congested regions. Therefore, both queuing delay and medium access delay are greater when these routing metrics are used, which have negative effects on the average end-to-end delay. In contrast, the average end-to-end delay for a packet that uses EFT is less, as this metric chooses paths with less medium access, transmission, and queuing delays (i.e., considers all the factors that impact the forwarding time of a packet).

Figure 13 shows the average packet loss rate of the network with increasing traffic loads. In general, as the traffic load increases, the packet loss rate of all the metrics tends to increase. Due to their inability to address load balancing, most of the packets tend to choose paths with high data rate when ETT, Airtime, or MIC metrics are used. This results in buffer drops in the intermediate nodes. Moreover, as ETT and Airtime do not address the effects of interference, packets are also lost due to interference or collision from contending nodes. In contrast, EFT prefers paths in less congested regions of the network, and packets experience only low to medium contention. Therefore, the packet loss rate when using EFT is lower than that when using the other metrics.

4.3 Scenario 3: impact of link quality

We consider a simple scenario in Fig. 14 to show that the quality of a link depends not only on the transmitting node's data rate, but also on the data rates of the nodes in its neighborhood; a lower data rate contributes negatively to the overall throughput. As shown in the figure, nodes A, B, D, E, and F transmit data at a rate of 54 Mbps, whereas node C transmits data at 48 Mbps and G uses the lowest rate of 6 Mbps. We assume that all stations are tuned to a single channel and that the transmitting power of nodes is reduced so that nodes can hear transmission only from their one-hop neighbors. There are three existing UDP flows from nodes $G \rightarrow F$, $A \rightarrow C$, and $B \rightarrow C$. We start a new flow from node $D \rightarrow E$. Based on their properties, the routing metrics might choose any of the paths from $D \rightarrow C \rightarrow E$ and $D \rightarrow F \rightarrow E$.

Figures 15 and 16 show the average throughput and end-to-end delay achieved by the new flow, respectively. It is evident that EFT achieves a significantly higher throughput than ETX, ETT, Airtime, and MIC, as EFT always chooses the best throughput path $D \rightarrow C \rightarrow E$. This is because the transmission from G to F at a lower data rate keeps node F busy for a longer period of time and the medium access time in node F increases. This makes path $D \rightarrow F \rightarrow E$ a lower throughput path than $D \rightarrow C \rightarrow E$; by incorporating the transmission rate and packet size of the contending nodes (i.e., the expected busy period of the contending nodes), EFT identifies this fact. However, because of the high data rate, ETT and Airtime will more often choose the low-throughput path $D \rightarrow F \rightarrow E$. Based on the packet loss probability of the links, ETX might switch between the two available paths. Interestingly, MIC in this scenario achieves a lower throughput as it always prefers $D \rightarrow F \rightarrow E$ as the better path because node F has fewer interfering neighbors than node C. For all these reasons, the average end-to-end delay achieved by EFT is also lower than that achieved using the other metrics. Figure 16 shows that the average end-to-end delay of EFT is 62% lower than that of ETX,

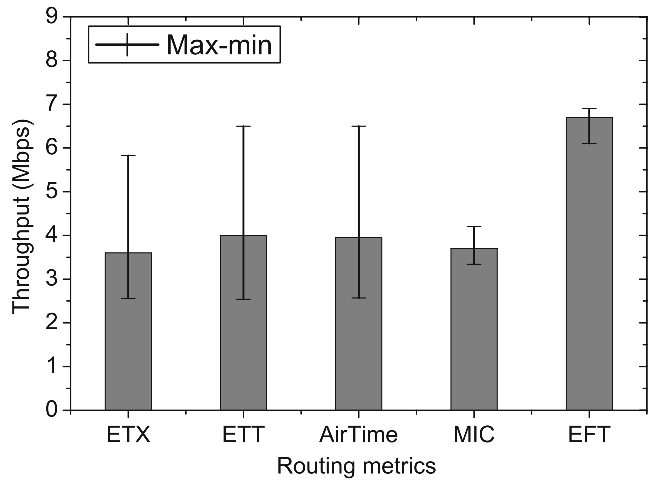

Fig. 15 Average throughput of a new flow for different routing metrics in scenario 3

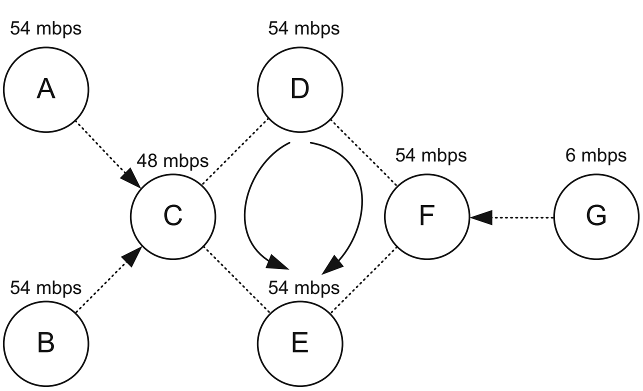

Fig. 14 Impact of contending nodes' data rates and packet sizes in path selection of a new flow

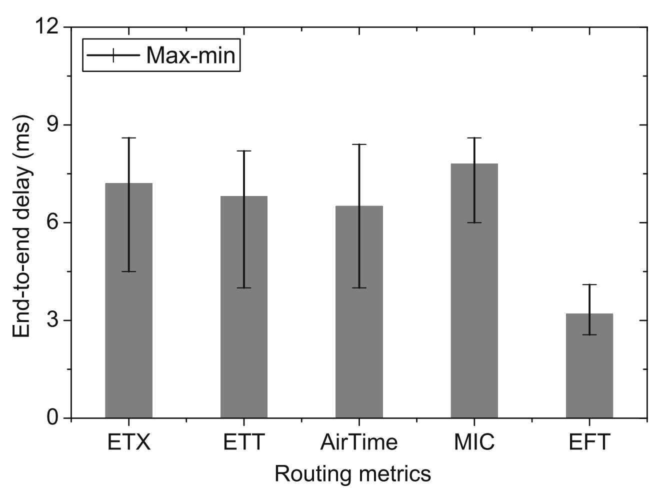

Fig. 16 Average end-to-end delay of a new flow for different routing metrics in scenario 3

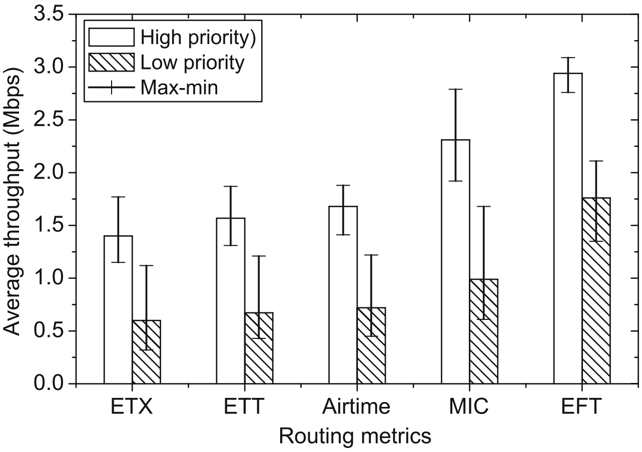

Fig. 17 Average throughput of flows with different traffic priorities

59% lower than that of ETT, 56% lower than that of Airtime, and 65% lower than that of MIC.

4.4 Scenario 4: impact of traffic priority

We now investigate the impact of traffic priority on path selection. We make use of a network comprising 50 nodes and 20 flows. Among the flows, ten flows are high-priority flows while the remaining ten flows are low-priority flows. Figures 17 and 18 show the average throughput and end-to-end delay achieved by the different traffic classes using different metrics, respectively. As shown in the figures, the high-priority traffic not only achieves better performance using EFT than other metrics, but the low-priority traffic also achieves a very high throughput and low delay. Because the existing metrics forward traffic without considering their priorities, paths are not selected for particular classes. Therefore, low-priority traffic starves in the MAC layer for getting access to the medium. Furthermore, the low-priority traffic contends with the high-priority traffic and reduces the throughput and increases the delay of the high-priority traffic. In contrast, EFT forwards the traffic by considering the best path for a particular traffic class. Thus, for most cases, the low-priority packets avoid the paths used by the high-priority packets, resulting in shorter delays and high throughput. This also reduces the loads and contentions of the high-priority traffic, and they achieve better performance.

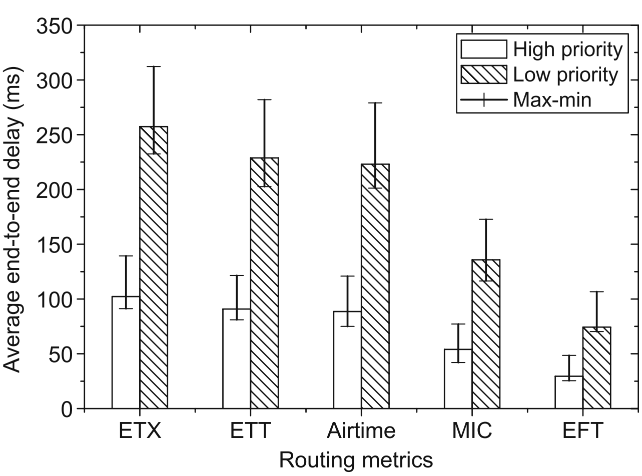

Fig. 18 Average end-to-end delay of flows with different traffic priorities

5 Conclusions

In this paper, we have presented a new routing metric, EFT, that addresses all factors (transmission rate, success rate, contenting neighbors and their loads, load-awareness, and traffic priority) that affect the forwarding time of a packet in a node. EFT chooses the path that has minimum end-to-end delay, which, along with transmission delay, also includes medium access and queuing delays. EFT can capture the effect of the traffic loads of neighbors and chooses paths through less congested areas of the network and balances traffic loads among network nodes. The EFT metric treats packets of different traffic classes differently and selects paths suitable for the particular traffic class. The proposed metric is incorporated with IEEE 802.11s's HWMP routing protocol, and simulation results demonstrate that it performs significantly better than the existing routing metrics.

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Appendix

A. Defer time with traffic priority

For simplicity of analysis, we consider only three classes of traffic, and according to [11], for $j = 2, 3$, defer time (i.e., AIFS[j]) is equal to the duration of DIFS and is constant. However, for low-priority traffic (i.e., $j = 1$), $\text{AIFS}[1] = \text{DIFS} + d_s$. Because a node with high-priority traffic can initiate a transmission during the extra slot of the low-priority traffic, a node with low-priority traffic cannot defreeze its backoff counter. Thus, d_d^1 depends on the number of nodes associated with the higher-priority traffic and the probability at which they attempt to transmit. The defer time of traffic class 1 will include the transmission time of the high-priority packet initiated in the extra slot. Furthermore, the number of interruptions can be more than one. Therefore, the defer time of a packet of traffic class 1 is

$$d_d^1 = \text{AIFS}[1] + H \times T, \tag{15}$$

where, H is a random variable representing the number of interruptions in a single defer time and T is a random variable representing the amount of time of each interruption, given by

$$T = \text{AIFS}[2/3] + d_\text{t} + \text{SIFS} + d_\text{ACK}. \tag{16}$$

The expected value of T is given by

$$E[T] = \text{AIFS}[2/3] + E[d_\text{t}] + \text{SIFS} + d_\text{ACK}. \tag{17}$$

Let, μ_2 and μ_3 denote the probability at which a node with traffic classes 2 and 3, respectively, attempts to transmit in any randomly selected slot. The number of interruptions in a defer time is geometrically distributed with parameter μ, where μ is the probability that at least one high-priority packet transmits. Let n_2 and n_3 be the number of neighbors with traffic classes 2 and 3, respectively, and τ_2 and τ_3 be the probability of their transmissions, respectively. We therefore have

$$\mu_2 = 1 - (1-\tau_2)^{n_2} \tag{18a}$$

$$\mu_3 = 1 - (1-\tau_3)^{n_3} \tag{18b}$$

$$\begin{aligned}\mu &= 1 - (1-\mu_2)(1-\mu_3)\\ &= 1 - (1-\tau_2)^{n_2}(1-\tau_3)^{n_3}\\ &= 1 - (1-\tau)^{n_2+n_3}\end{aligned} \tag{18c}$$

The expected number of interruptions by a node with a higher-priority packet is therefore $E[H] = \frac{\mu}{1-\mu}$.

B. Backoff delay

The backoff time at each transmission attempt depends on the size of the backoff counter (i.e., the size of the contention window), the number of interruptions in the backoff process (which depends on the number of contending neighbors) due to transmissions (either successful or unsuccessful) by the neighbors and the channel quality (transmission rate) of the transmitting neighbors, as well as the defer time after each freezing. Thus, the backoff delay in the i-th transmission attempt (where, $i = 0, 1, ..., M'-1$) of the j-th traffic class $d_\text{b}^j(i)$ is given by

$$d_\text{b}^j(i) = \sum_{k=1}^{w_i^j} d_s + \sum_{k=1}^{b_i^j} (d_\text{b} + d_d^j) \tag{19}$$

At the beginning of the backoff process of the i-th transmission attempt, a node uniformly chooses a backoff value w_i^j in the range $(0, 2^i \times CW_{\min}(j))$, where $2^i \times CW_{\min}(j)$ defines the size of the current contention window for traffic class j. Therefore, the expected size of the backoff counter is $E[w_i^j] = \frac{w_i^j}{2}$ [20].

If the number of busy slots at the i-th transmission attempt is b_i^j, then a node has to wait $B_i^j = w_i^j + b_i^j$ slot times so that its backoff value w_i^j reaches zero and it can start transmission. Let there be n neighbors of a tagged node (the node that is in backoff). Let p_t be the probability that there is a transmission in a slot. The probability of a slot being idle is the probability that none of the neighbors transmit in that slot, given by $(1-p_\text{t})$. Now, the expected number of B_i^j slots required to get w_i^j idle slots is found by using Pascal distribution [20] with parameters w_i^j and $(1-p_\text{t})$. Therefore, the expected number of slots is $E[B_i^j] = \frac{E[w_i^j]}{1-p_\text{t}}$. The expected number of busy slots can then be calculated as

$$E[b_i^j] = E[B_i^j] - E[w_i^j] = \frac{w_i^j \times p_\text{t}}{2(1-p_\text{t})}. \tag{20}$$

The expected backoff delay for a packet of the j-th traffic class at the i-th transmission attempt can be found using Wald's equation [21] as

$$E[d_\text{b}^j(i)] = E[w_i^j] \times d_\text{s} + E[b_i^j] \times (d_\text{b} + d_\text{d}^j). \tag{21}$$

The number of transmission attempts (M) required for a successfully delivered packet has a truncated geometric distribution, with the PMF given by

$$p[M=i] = \frac{p_\text{s}(1-p_\text{s})^i}{1-(1-p_\text{s})^{M'}}[i = 0, 1, 2, ..., M'-1]. \tag{22}$$

Thus, by combining Eqs. 21 and 22, the expected backoff delay for a packet of the j-th traffic class is

$$E[d_\text{b}^j] = \frac{p_\text{s}}{1-(1-p_\text{s})^{M'}} \sum_{i=0}^{M'-1} (1-p_\text{s})^i . E[d_\text{b}^j(i)]. \tag{23}$$

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