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Providing forwarding assurance in multi-hop wireless networks ☆

Sungwon Han [a], Euiyul Ko [a], Hyunsoo Yoon [a], Ikjun Yeom [b,*]

[a] Department of Computer Science, Korea Advanced Institute of Science and Technology, Daejeon, South Korea
[b] Department of Computer Engineering, Sungkyunkwan University, Suwon, South Korea

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The Assured Forwarding (AF) service in a differentiated services network offers different levels of forwarding assurance for IP packets. In the wired Internet, the AF service is implemented as a queue management scheme associated with a drop policy in each router. In multi-hop wireless networks, however, queue management in each node is not enough to provide service differentiation globally since resource is shared among neighboring nodes. Hence, several studies for the AF service in wireless networks have proposed to manipulate the contention window size to provide service differentiation. The contention window size based differentiation scheme provides the AF service by transmitting packets in a higher class with a smaller contention window. However, since network congestion in multi-hop wireless networks causes packet losses at the link layer, the contention window size based differentiation scheme is not sufficient to provide the AF service. In this paper, we propose a rate control scheme to control load on the shared channel by adjusting sending rate according to the estimated channel state and the amount of AF traffic. We further propose a RTS retry limit adaptation scheme for loss rate differentiation at the link layer. Through extensive simulations, we show that proposed schemes are effective to differentiate forwarding assurance in multi-hop wireless networks.

1. Introduction

IEEE 802.11 [13,14,16] is a successful technology for wireless networks. Wireless LANs (WLANs) based on it have successfully replaced wired access links in recent years. Currently, it is attempted to extend it for multi-hop wireless networks such as ad-hoc networks and mesh networks. As more traffic is delivered over wireless networks, QoS (Quality of Service) support for them becomes an important issue. In single hop WLANs, IEEE 802.11e [15] is a key technique to provide QoS, but there are still arguments for how to apply it for multi-hop wireless networks [21].

In the Internet, the differentiated services (diffserv) network [5] is a well-known proposal for providing QoS. Among several services in the diffserv network, the Assured Forwarding (AF) service [11] has been paid wide attention due to its simple and scalable solution for service differentiation. In the AF service, each packet is marked either one of AF or BE (Best Effort, or non-AF) when it enters the network. When a link in the network becomes congested, BE packets are discarded first before dropping AF packets. As a result, AF packets observe much less drop rate than BE packets. Here, the level of congestion in a link is mostly measured by the queue length.

It has been shown that the drop rate differentiation in a local queue is enough to provide the AF service in the wired Internet [6]. However, it may be not directly applicable to multi-hop wireless networks due to the following two reasons: (a) in multi-hop wireless networks, neighboring nodes contend for a shared channel. Therefore local differentiation in each node is not enough to provide the AF service globally; and (b) in multi-hop wireless networks, network congestion causes packet losses not only at the network layer due to buffer overflow but also at the link layer due to link contention [9]. As the offered load increases, the number of backlogged nodes also increases, and consequently more nodes try to transmit a packet. Only some of them, however, can finish their transmission and the others fail to transmit a packet because the number of simultaneous transmissions is limited in multi-hop wireless networks. Hence, it is not effective to use only the local queue based drop policy for the AF service.

Based on the above reasons, the prior attempts to provide service differentiation in multi-hop wireless networks are based on IEEE 802.11e-like differentiation such that a higher priority packet is transmitted with a smaller contention window (*CW*) size [2,3,15,17–19,22]. As a result, higher priority packets have more chances to be transmitted. These schemes may be effective to provide service differentiation in single-cell wireless networks in which all nodes are within their carrier sensing range. In multi-hop wireless networks, however, nodes competing for a wireless channel may not observe the same channel condition since all

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* Corresponding author. Tel.: +82 31 299 4595.
E-mail addresses: sungwon@kaist.ac.kr (S. Han), eyko@cnlab.kaist.ac.kr (E. Ko), hyoon@cs.kaist.ac.kr (H. Yoon), ikjun@skku.edu (I. Yeom).

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nodes are not within each other's carrier sensing range. Then, the ratio of *CW* sizes may not properly differentiate the sending rate of each node. Therefore the contention window size based differentiation scheme is not sufficient to provide the AF service in multi-hop wireless networks.

In this paper, we propose a rate control scheme for the AF service in multi-hop wireless networks. The objective of the proposed scheme is to control load on the shared channel by adjusting sending rate according to the estimated channel state and the amount of AF traffic. Each node measures the amount of AF traffic and the channel utilization. When the utilization exceeds a certain threshold (an indication of congestion), it reduces its sending rate until either the utilization becomes less than the threshold or the sending rate becomes the receiving rate of AF traffic. With this scheme, nodes sending more AF packets maintain higher sending rate than nodes sending less AF packets. If we employ the rate control scheme as the above, the sending rate can be reduced intentionally, and thus the queue may be filled up. For loss rate differentiation at the queue, we simply employ the RED In/Out (RIO) [6] queue management scheme.

We further propose a RTS retry limit adaptation scheme for loss rate differentiation at the link layer. The objective of the RTS retry limit adaptation scheme is to assure forwarding of AF packets. When the network is overloaded, packet losses occur at the link layer due to the failure of RTS transmission. To assure forwarding of AF packets, the RTS retry limit adaptation scheme manipulates the RTS retry limit. If we increase the retry limit, we may expect to reduce the failure of RTS transmission. However, it may increase the overall delay of the network through repeated RTS retries of a packet. To avoid the delay increment, we first analyze the packet loss rate with a given RTS retry limit. Based on this analysis, the proposed scheme maximizes the loss rate differentiation between AF and BE packets while the average loss rate is maintained to be the same as without the differentiation. The proposed schemes are evaluated through extensive ns-2 [1] simulation. It is shown that they can effectively provide different levels of forwarding assurance.

The rest of this paper is organized as follows: In Section 2, we present related work and motivation of this paper. In Section 3, we propose a rate control scheme and a RTS retry limit adaptation scheme for the AF service in multi-hop wireless networks. In Section 4, we present performance evaluation through ns-2 simulations. We conclude this paper in Section 5.

2. Related work and motivation

There have been many researches on the priority scheduling and the differentiated services in wireless networks. In Refs. [3,17,19,20], delay-based priority scheduling schemes in WLANs have been proposed. They attempt to provide delay differentiation according to the delay requirements of applications. They manipulate either or both the *CW* and the interframe space to provide delay differentiation. In Ref. [22], a distributed contention window control algorithm is proposed to realized a given bandwidth allocation policy in single-cell WLANs. In this algorithm, each node controls its *CW* size according to its utility function and the channel state. In Ref. [2], a diffserv extension of IEEE 802.11, called DIME (Diffserv MAC Extension) has been proposed. In DIME, two modules are provided for EF (Expedited Forwarding) and AF services. For the EF service, it has been proposed to utilize the PCF (Point Coordination Function), and for the AF service, the DCF (Distributed Coordination Function) is modified. In Ref. [10], a QoS scheduling scheme for multi-hop ad-hoc networks has been proposed. In this scheme, each node exchanges an additional information, called priority tag, to assess its priority level relative to other nodes. Based on the priority tag, each node calculates the backoff time to achieve the QoS goal.

The main limitation of the prior works is that they mostly rely on the *CW* to realize service differentiation. It is usually true that a node with a small *CW* may have more chances to transmit its packets in WLANs in where all the nodes experience the same channel condition. However, in multi-hop wireless networks, it may not be hold. In Fig. 1, we depict a topology where competing nodes observe different channel condition. In the figure, node A and node C are within node B's carrier sensing range, and node B has to compete the wireless channel with them. However, node A and node C are not within the other's carrier sensing range, and each of them competes for the channel only with node B. In this scenario, even if we assign the same *CW* for the three nodes, node B has much less chances to transmit its packets than node A and node C. If we want to allocate the bandwidth to the three nodes equally using the *CW*, we have to assign a smaller *CW* to node B. However, the transmission rate is determined by the ratio of the *CW*s of competing nodes rather than specific values of them, and this makes it hard to precisely control the transmission rate using the *CW*. In this paper, we control the transmission rate directly instead of adjusting the *CW*. In the proposed scheme, to provide forwarding assurance to AF packets, each node controls its transmission rate based on the current amount of AF traffic and the channel condition.

3. An assured forwarding service architecture for multi-hop wireless networks

In this section, we propose an AF service architecture for multi-hop wireless networks. Fig. 2 shows the proposed architecture. Between the network and the link layers, we employ a rate control scheme which controls the packet delivery rate from the network layer to the link layer. In the link layer, we adapt the RTS retry limit to provide link layer loss rate differentiation. In the proposed architecture, we use a single queue for both AF and BE packets in order to avoid out-of-order delivery.

While deriving our proposed architecture, we consider TCP as the target traffic of AF service, and focus on per-hop behaviors. Most studies on AF service make the similar assumption in Refs. [6,7]. For UDP based application, we may consider to employ application-layer congestion control schemes in Ref. [8].

3.1. A rate control scheme for AF traffic

The objective of this scheme is to control the transmission rate of each node so that nodes having more AF packets have more chances to transmit its packets. Here note that the scheme does not adjust the *CW* to control the transmission rate as we discussed in Section 2. Each node applies the same *CW*, and the transmission rate is controlled by the packet delivery rate from the network

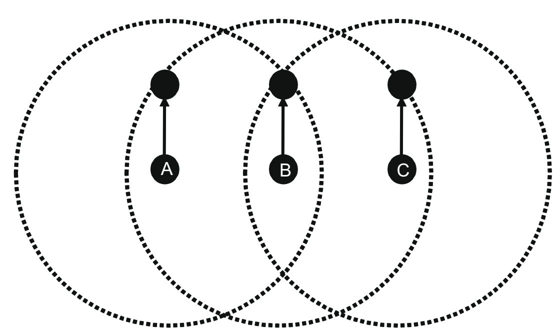


Fig. 1. Flow in the middle topology. (A circle represents the carrier sensing range.)

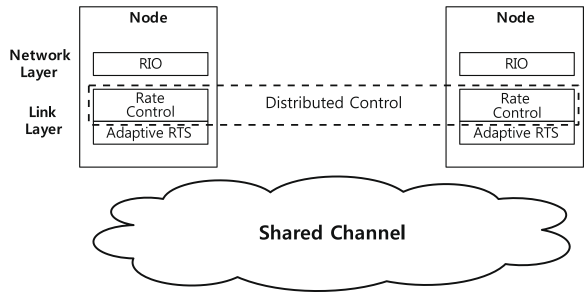


Fig. 2. The proposed assured forwarding service architecture.

layer to the MAC layer since it is easier and more straightforward. When the network bandwidth is sufficient, there is no discrimination between AF and BE packets. When a node detects congestion, however, it reduces its packet delivery rate to the MAC layer, and the network queue is build up. Then, BE packets are discarded first by the RIO queue management policy, and we provide forwarding assurance to AF packets.

The proposed rate control scheme consists of two parts. In the first part, each node controls the delivery rate locally. It monitors the channel state and detects congestion individually. Upon congestion, it reduces its packet delivery rate until the rate becomes the enqueueing rate of AF packets. However, the local control is not enough to provide forwarding assurance to AF packets from other neighbor nodes[1] since, as we discussed in Section 2, channel state of neighbor nodes may not be the same, and each neighbor node may monitor the channel state differently. Then, it may happen that a node observes enough bandwidth to transmit both AF and BE packets while another node cannot transmit even AF packets. In this case, the former node should reduce its transmission rate in order to make room for the other node, but it is hard for the node to detect this situation locally. In the second part, a node, which does not have enough bandwidth to transmit AF packets, sends a source quench signal to neighbor nodes. When a node receives the signal, it reduces its transmission rate if it is observing enough bandwidth for BE packets.

First, we describe how to detect congestion and control the delivery rate locally. Network congestion is defined as a state in which the packet arrival rate exceeds the network capacity, and, in wired networks, it is usually detected by the queue length at the link. In multi-hop wireless networks, however, due to the shared nature of wireless links, each node has to monitor all the queues of neighbor nodes to detect congestion by the queue length. In this scheme, to detect congestion locally, we monitor the wireless link instead of the queue length. In the IEEE 802.11 MAC protocol, each node defers its transmission for a random number of idle time slots to avoid collision as shown in Fig. 3, and this number of idle time slots can be used to detect congestion. Let x_i be the probability that node i has at least one packet to transmit for a given idle slot, and it is given by

$$x_i = \begin{cases} \frac{a_i}{C} & \text{if } s_i = a_i \\ 1 & \text{if } s_i < a_i \end{cases} \tag{1}$$

where C is the channel capacity, and s_i and a_i are the sending rate and the arrival rate[2] of node i, respectively. $\mathcal{N}$ is the set of neighbor nodes, and m is the number of neighbor nodes. Since a node attempts to transmit a packet with a probability of $2/(CW+1)$ [4], where CW is the contention window size, the probability P_I that a given slot is idle can be calculated by

$$P_I = \prod_{i \in \mathcal{N}} \left(1 - x_i \cdot \frac{2}{CW+1}\right) \tag{2}$$

Using (2), the average number of idle slots n_{ave} between two busy periods (transmission or collision) is calculated by

$$n_{ave} = \frac{\prod_{i\in\mathcal{N}}\left(1 - x_i \cdot \frac{2}{CW+1}\right)}{1 - \prod_{i\in\mathcal{N}}\left(1 - x_i \cdot \frac{2}{CW+1}\right)} \tag{3}$$

From (3), we can observe that if $\sum_{i\in\mathcal{N}} x_i$ increases, i.e. nodes have more packets to transmit, then n_{ave} decreases. In the proposed scheme, we compare n_{ave} with n_{target} to detect congestion. Here, n_{target} is the number of idle slots for $\sum_{i\in\mathcal{N}} x_i = 1$, i.e. $\sum_{i\in\mathcal{N}} a_i = C$. It reduces the arrival rate when $n_{ave} < n_{target}$ (which indicates that $\sum_{i\in\mathcal{N}} a_i > C$). When reducing the arrival rate, we protect the minimum sending rate which is corresponding to the arrival rate of AF traffic so that a link sending more AF traffic obtains more channel bandwidth.

For a given aggregate arrival rate, n_{ave} can be observed differently depending on the distribution of x_i. To detect congestion regardless of the distribution of x_i, we use the upper bound of n_{ave} as n_{target}. Here note that n_{target} is defined very conservatively, and it does not necessarily mean that the link utilization is maximized. We adjust the CW as well to improve the link utilization. Given $\mathbf{X} = \{x_i, i \in \mathcal{N}\}$, n_{ave} has the maximum value when $x_i = x_j$, $\forall i,j \in \mathcal{N}$. To calculate n_{target}, we evaluate (3) with $x_i = \frac{1}{m}$, $\forall i \in \mathcal{N}$ as follows

$$n_{target} = \frac{\left(1 - \frac{1}{m}\cdot\frac{2}{CW+1}\right)^m}{1 - \left(1 - \frac{1}{m}\cdot\frac{2}{CW+1}\right)^m} \tag{4}$$

Fig. 4 shows how n_{target} changes as m increases. It shows that n_{target} is a quasi constant for a large m. Therefore, we can precalculate n_{target} with a given CW.[3]

Here note that n_{target} can be considered as the time spent for contention. To increase the effective channel capacity, we may consider to use a smaller CW to decrease n_{target}. However, a smaller CW may increases the collision probability. To maximize the effective capacity, we should minimize n_{target} without increasing the collision probability. Based on Refs. [12,22], the average number of time slots F between two consecutive transmissions can be estimated by

$$F = \frac{\left(1 - \frac{1}{m}\cdot\frac{2}{CW+1}\right)^m + \frac{T_c}{aSlotTime}\left(1 - \left(1 - \frac{1}{m}\cdot\frac{2}{CW+1}\right)^m - \frac{2}{CW+1}\cdot\left(1 - \frac{1}{m}\cdot\frac{2}{CW+1}\right)^{m-1}\right)}{\frac{2}{CW+1}\cdot\left(1 - \frac{1}{m}\cdot\frac{2}{CW+1}\right)^{m-1}} \tag{5}$$

where $sSlotTime$ is a duration of an idle slot, and T_c is a duration of an RTS collision period. To maximize the effective capacity, F should be minimized. CW for minimizing F can be calculated by derivative of (5). In Fig. 5, we present n_{target} and CW for maximizing the effective capacity. It is observed that CW and n_{target} are converged when m is sufficiently large. In the proposed scheme, we use 4.49 for n_{target} and 8 for CW.[4]

So far, we have described how the proposed scheme detect congestion and controls the transmission rate locally. In the second part of the proposed scheme, a node, which does not have enough bandwidth for its AF packets, sends a signal to neighbor nodes, and

[1] In this paper, neighboring nodes are defined as the nodes competing for a wireless channel.

[2] Here the arrival rate means the packet rate arriving at the MAC layer, and it is corresponding to the delivery rate of the network layer. In this paper, we use both interchangeably.

[3] In the proposed scheme, we do not use the binary exponential backoff mechanism because nodes adjust its sending rate according to the channel state to control load on the channel.

[4] We set $aSlotTime = 20$ μs and $T_c = 685.6$ μs. (IEEE 802.11b).

Fig. 3. The IEEE 802.11 MAC protocol.

the neighbor nodes respond to the signal. When the arrival rate to the MAC layer in a node becomes the enqueueing rate of AF packets to the network layer (then, eventually, all the packets from the node belong to AF class due to the RIO policy), it sends a signal to neighbor nodes as depicted in Fig. 6. Instead of defining a new message format for the signal, we simply exchange additional multiple CTSs in order to minimize the implementation cost. We insert two pairs of CTS packets in the middle of an exchange of RTS/CTS to distinguish a source quench signal from an ordinary exchange of RTS/CTS for packet transmission. If we use only a pair of CTS packets for the signal, then sender-side neighbor nodes (who can receive packets only from the sender) can sense only a pair of RTS/CTS (sent by the sender), and cannot distinguish them. When a node receives one RTS and two CTS packets (sender-side neighbor nodes) or three CTS packets (receiver-side neighbor nodes) consecutively with the interval of SIFS + CTS + SIFS, it reduces its arrival rate if its arrival rate is greater than the enqueueing rate of AF packets. Actually, the neighbors of a receiver do not need to reduce their rate when only a sender observes the congestion. However, it is not trivial to distinguish which node is congested upon a transmission failure. In the proposed scheme, neighbors of both the sender and the receiver reduce their rate to resolve the congestion. Here note that the Network Allocation Vector (NAV) of the first RTS and CTS packets should include two additional CTS packets. Neighbor nodes not within the transmission range of the node sending the signal cannot decode RTS and CTS packets, but they can still detect the signal by the length of transmissions (20 byte RTS, and 14 byte CTS). Even though the NAV value of the first RTS or CTS packet cannot be decoded, the subsequent CTSs in a signal are not interrupted by them since EIFS (Extended IFS) is larger than the interval of SIFS + CTS + SIFS.

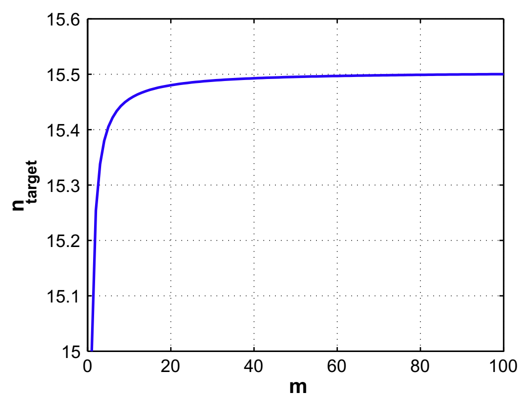
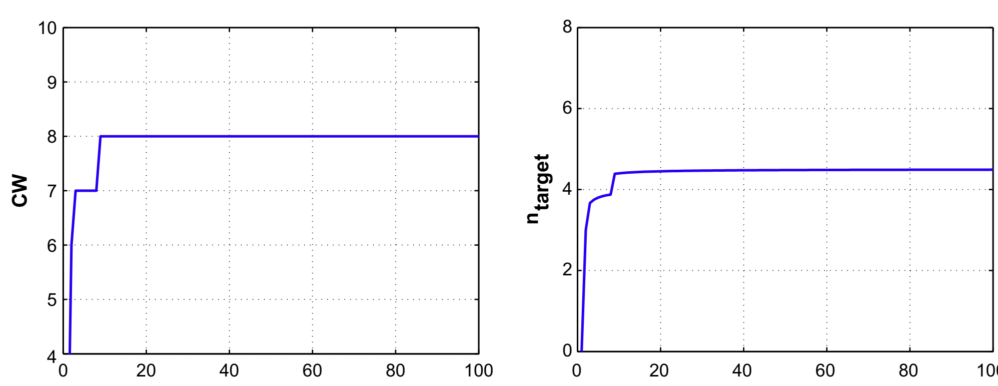


Fig. 4. n_{target} with respect to m ($CW = 31$).

In Fig. 7, we present an algorithm for the proposed rate control scheme. In this algorithm, each node individually monitors n_{ave}. For each monitoring interval, δ, each node compares n_{ave} with n_{target}, and checks whether it received a signal from its neighbor nodes in the last interval. If n_{ave} is greater than n_{target}, and there was no signal received, then it increases its arrival rate, r_{total}, linearly in order to maximize the link utilization. Otherwise, it reduces the arrival rate proportional to the arrival rate of BE packets. Then, a node sending more BE packets releases more bandwidth. If it



(a) CW (b) n_{target}

Fig. 5. Optimal CW and n_{target} with respect to m.

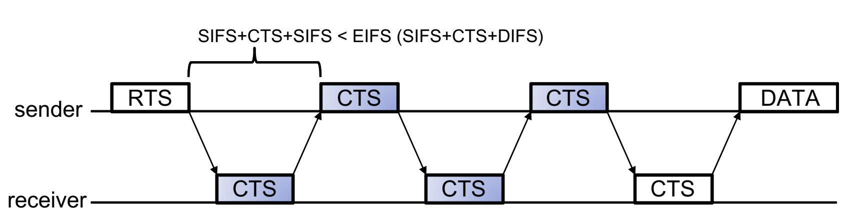


Fig. 6. A source quench signal.

For each monitoring interval δ
if $n_{ave} > n_{target}$ AND $r_{total} > r_{af}$ AND not received a signal **then**
  $r_{total} = r_{total} + a$
else
  if $r_{total} > r_{af}$ **then**
    $r_{total} = r_{total} - b(r_{total} - r_{af})$
  else
    send a signal to neighbor nodes
  end if
end if
n_{ave}: the average number of idle time slots
n_{target}: the target number of idle time slots
r_{af}: the arrival rate of AF traffic
r_{total}: the aggregate arrival rate of AF and BE traffic
a, b: the AIMD coefficients

Fig. 7. A rate control algorithm for AF traffic.

does not send any BE packet ($r_{total} = r_{af}$), it does not reduce the arrival rate, and sends a signal to neighbor nodes.

In the proposed algorithm, we employ the AIMD (Additive Increase Multiplicative Decrease) mechanism to adjust the arrival rate since the state of a node in multi-hop wireless networks is highly dynamic, and it is hard to find an optimal rate at a given time. In many network-related studies, especially on TCP, it has been shown that the AIMD mechanism is robust and effective for fair resource sharing. In the algorithm, a and b are the AIMD coefficients such that the arrival rate of a link decreases by $b(r_{total} - r_{af})$ upon congestion, and increases a packets for each monitoring interval. In general, it is known that the AIMD mechanism is not much sensitive to its coefficients.

3.2. A RTS retry limit adaptation scheme

In multi-hop wireless networks, packet transmission can be easily corrupted by interferences from various sources, and the proposed rate control scheme may not be enough to provide forwarding assurance to AF packets. In this section, we present a complementary scheme to protect AF packets. This scheme is independently operated with the rate control scheme, and can cooperate with other schemes. The proposed scheme differentiates the RTS retry limit for AF and BE packets to assure forwarding of AF packets. While deriving the scheme, we assume that packets are dropped due to the RTS transmission failure based on the observations in the previous section.

Then, the packet loss rate p is simply given by

$$p = (p_r)^l \tag{6}$$

where p_r is the probability of RTS collision, and l is the RTS retry limit. By applying a larger l for AF packets, they can observe less loss rate than BE packets. However, we cannot increase l for AF packets arbitrarily since it may increase overall packet delay due to repeated retries of a packet. The objective of the proposed scheme is to find the maximum RTS retry limit for AF packets without changing the average loss rate. The following equation defines the problem of the proposed scheme.

$$(p_r)^l = \alpha(p_r)^{l_{af}} + (1-\alpha)(p_r)^{l_{be}} \tag{7}$$

where α is the proportion of AF traffic, and l_{af} and l_{be} are the retry limits for AF and BE packets, respectively. The proposed scheme finds either the maximum l_{af} or the minimum l_{be} and correspondingly the other with given p_r, l, and α. Here l is the retry limit without differentiation, and the default value is seven in the IEEE 802.11 standard. Here note that, when $\alpha = 1$ or 0, l_{af} or l_{be} is simply equal to l, respectively. Hereafter, we assume that α is a value greater than 0 and less than 1.

From (7), l_{af} is calculated by

$$l_{af} = \log_{p_r} \mathscr{A} \tag{8}$$

where

$$\mathscr{A} = \frac{(p_r)^l - (1-\alpha)(p_r)^{l_{be}}}{\alpha} \tag{9}$$

In (8), $\mathscr{A}$ should be greater than 0, and then we have

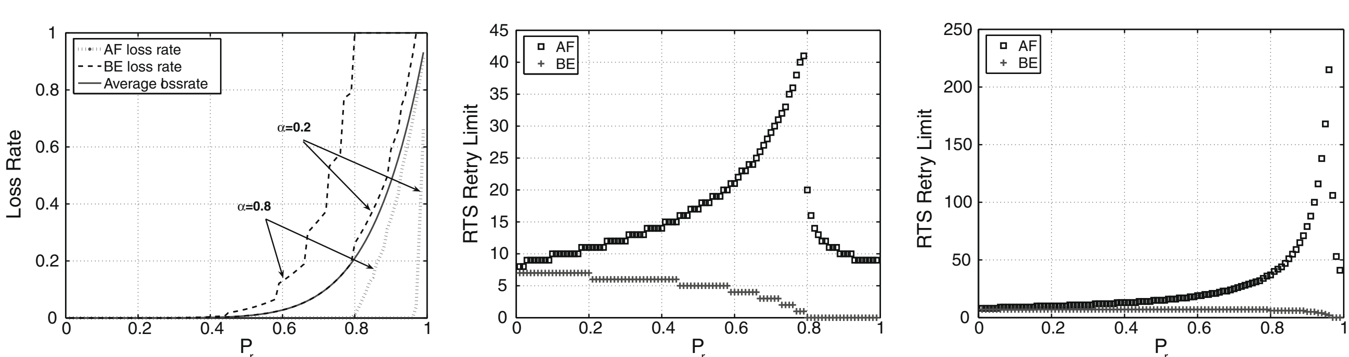
$$l_{be} > l - \log_{p_r}\{1-\alpha\} \tag{10}$$

Eq. (10) should be satisfied to maintain the average loss rate. Since l_{be} cannot be negative, l_{be} for maximizing l_{af} is set to

$$l_{be} = \begin{cases} l - \log_{p_r}\{1-\alpha\} + \epsilon & \text{if } l > \log_{p_r}\{1-\alpha\} \\ 0 & \text{otherwise} \end{cases} \tag{11}$$

where ϵ is a small number for inequality. Eq. (11) shows that as α increases (which means that the proportion of AF traffic increases), l_{be} decreases to maintain the average loss rate. When $\log_{p_r}\{1-\alpha\} \geqslant l$, l_{be} is set to 0, and all the BE packets are dropped. By applying (11) to (8), we can calculate the corresponding l_{af}.

In Fig. 8, we present the impact of α, the retry limit (l), and the collision rate (p_r) on the loss rate. In Fig. 8(a), we increase p_r and observe the loss rate. The solid line represents the loss rate without differentiation. If we apply the loss rate differentiation, the loss rate is impacted by the proportion of AF traffic (α). When $\alpha = 0.8$, all the BE packets and some of AF packets are dropped for $p_r > 0.8$. In Fig. 8(b), the retry limits for AF and BE packets (when $\alpha = 0.8$) are plotted. As p_r increases to 0.8, l_{af} increases to avoid loss, and l_{be} decreases to maintain the average loss rate. For



(a) Loss rate (b) RTS retry limit ($\alpha = 0.8$) (c) RTS retry limit ($\alpha = 0.2$)

Fig. 8. Loss rate and RTS retry limit with the RTS retry limit adaptation scheme.

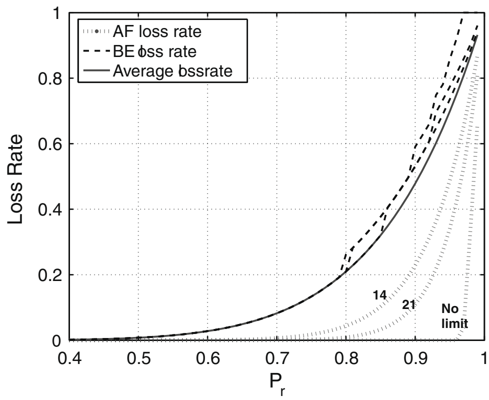


Fig. 9. Loss rate with different upper limits ($\alpha = 0.2$).

$p_r > 0.8$, l_{be} becomes zero, and then l_{af} finally decreases. When $\alpha = 0.2$, most packets are BE, and AF packets can be protected for $p_r < 0.97$ without decreasing the average loss rate. As shown in Fig. 8(c), however, the retry limit for AF increases up to 215. The increased retry limit can protect AF packets from being dropped, but it may also increase delay due to the extensive retries of an RTS. Hence, we need to set an upper limit on the retry limit for AF packets. In Fig. 9, we show that the drop rate and the retry limit when the upper limit is set to 14 and 21, respectively. Although the drop rate of AF packets increases as the upper limit decreases, it is shown that the proposed scheme can still provide notable differentiation in loss rate between AF and BE packets.

Based on the analysis in the above, we present an algorithm for RTS retry limit adaptation in Fig. 10. It is shown that the proposed scheme can be implemented with a simple algorithm.

```
for each packet transmission do
    if the packet belongs to AF class then
        α = λα + (1 − λ)
        calculate l with α and p_r using Eq. (8)
    else
        α = λα
        calculate l with α and p_r using Eq. (11)
    end if
    try to send an RTS until reaching l
    if the RTS is successfully transmitted then send the packet
    else drop this packet
    update p_r
end for
α: the proportion of AF traffic
λ: the moving average weight
l: the RTS retry limit
p_r: the RTS collision rate
```

Fig. 10. A RTS retry limit adaptation algorithm.

4. Performance evaluation

In this section, we evaluate the proposed AF service architecture through ns-2 [1] simulations. In the simulation, we use IEEE 802.11b MAC with 11 Mbps bandwidth. The transmission range and the interference range are set to 250 and 550 m, respectively. The default RTS retry limit is set to seven, i.e., a packet is dropped after seven unsuccessful RTS initiations. The maximum RTS retry limit is set to 14. *CW* is set to 8. *a*, *b*, and δ for the rate control scheme are set to 1 packet, 0.25, and 0.1 s. RIO parameter set $\{min_q, max_q, max_p\}$ is $\{10, 30, 0.1\}$ for BE packets and $\{30, 40, 0.05\}$. The packet marker is implemented on each node so that packets are marked from the source node.

We compare performance of the proposed schemes with performance of IEEE 802.11e as a *CW* based differentiation scheme. AF packets are mapped onto the highest access category AC_VO ($CW_{min} = 7$, $CW_{max} = 15$) and BE packets are mapped onto access category AC_BE($CW_{min} = 31$, $CW_{max} = 1023$). In the original IEEE 802.11e, each access category has a separate queue with the assumption that packets from a flow belong to an access category. In our scenario, packets from a flow can be marked either AF and BE based on the temporal sending rate, and our-of-order delivery may occur if we use separate queues for them. In this paper, we use a single queue for both AF and BE packets to avoid out-of-order delivery.

4.1. Preliminary evaluations

We begin with a simple topology depicted in Fig. 11. In the figure, there are two 2-hop chains. We inject a TCP flow to each chain. In this scenario, two TCP flows contend for a shared channel because all nodes are within each other's carrier sensing range. For Flow 1, we increase the contract rate for the AF service from 0 to 1 Mbps. Flow 2 sends only BE packets. The objective of this simulation is to show how the proposed rate control scheme achieves forwarding assurance when nodes with different amounts of AF traffic contend for a shared channel. The results are presented in Fig. 12. It is shown that Flow 1 achieves its contract rate without

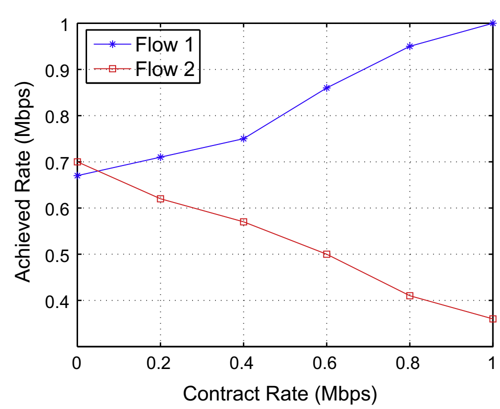
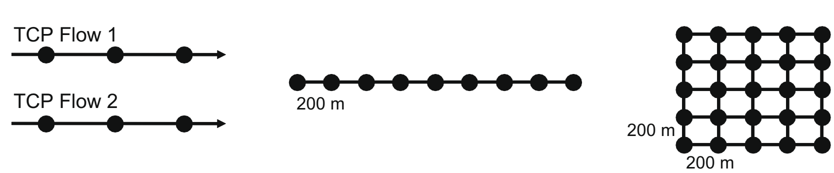
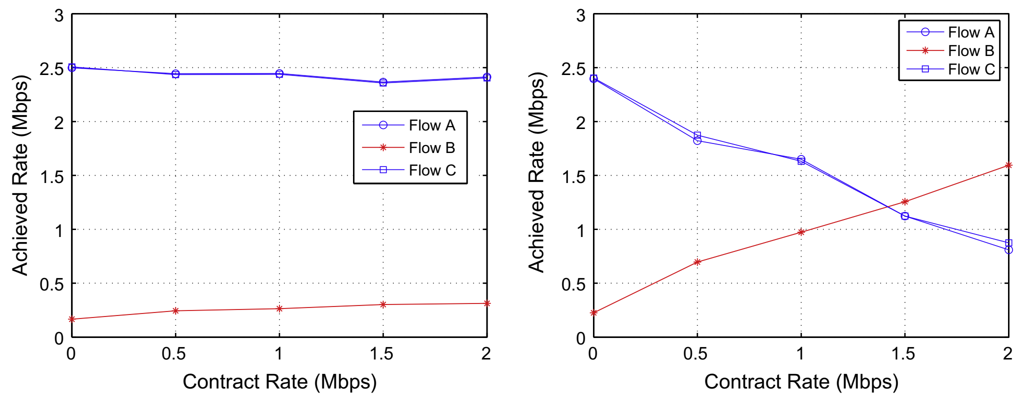


Fig. 12. Achieved throughput with various contract rates in a two 2-hop chain topology.



(a) Two2-hopchaintopology (b) Chaintopology (c) Gridtopology

Fig. 11. Simulation topologies.



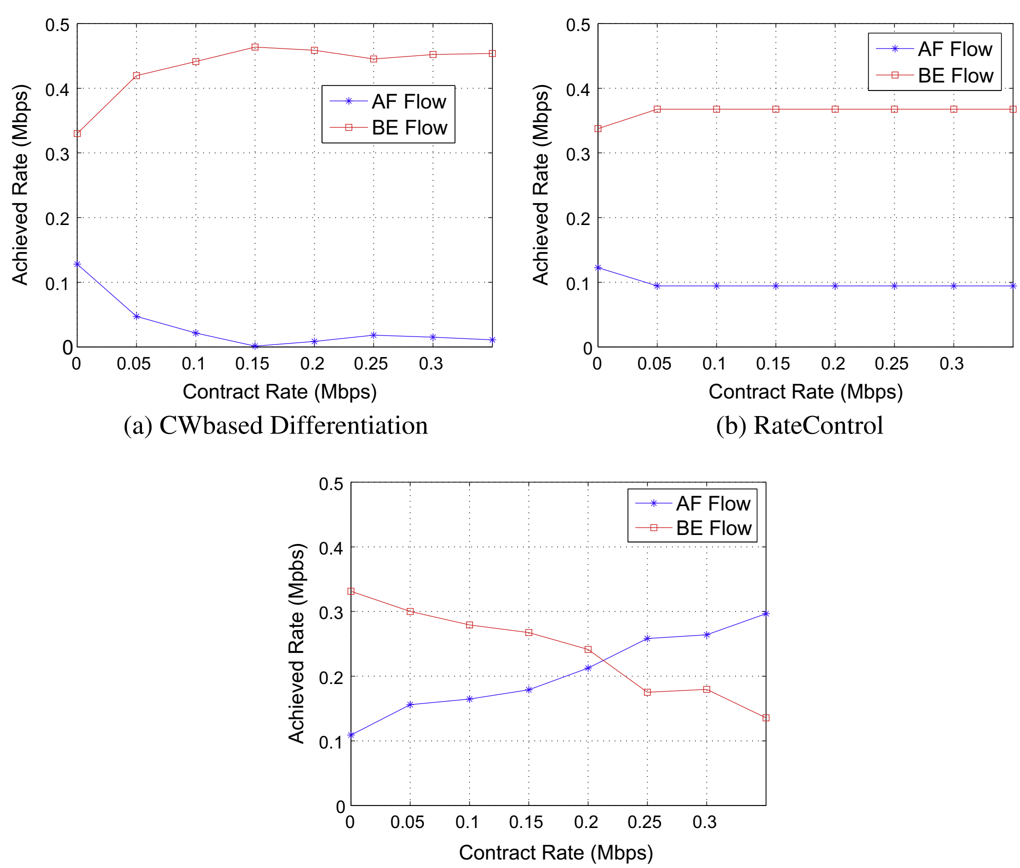
(a) CW based differentiation (b) Rate control + Control signal

Fig. 13. Achieved rates with various contract rates in a flow in the middle topology.

degradation of the aggregate throughput. The aggregate throughput (1.4 Mbps) may look low compared to the link bandwidth (11 Mbps), but it is due to multi-hop transmission. When we perform the simulation with the legacy IEEE 802.11b in the same topology, we observe the similar aggregate throughput. As the amount of AF traffic increases on the shared channel, the throughput of Flow 2 decreases because it only sends BE traffic. Here note that there is no packet loss due to RTS collision in this scenario since all the nodes are within each other's carrier sensing range, and the throughput differentiation is contributed by the rate control scheme.

4.2. Evaluations in a flow in the middle topology

In this section, we perform simulations in the depicted in Fig. 1. We inject a TCP flow to each link. For Flow B, we increase the contract rate for the AF service from 0 to 2 Mbps. Other flows send only BE packets. In Fig. 13(a), we present the achieved rate with *CW* based differentiation. Even though the *CW* for BE packets is much larger than the *CW* for AF packets, it is observed that Flow B (sending AF packets) achieves much less throughput than the others, and this shows that service differentiation in forwarding assurance is hardly realized by *CW* based differentiation. With a



(c) RateControl+AdaptiveRTS

Fig. 14. Achieved rates with various contract rates in a chain topology.

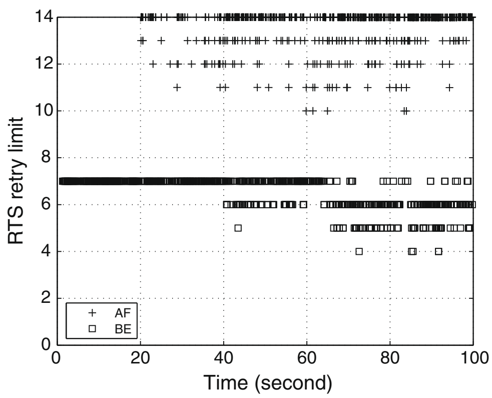


Fig. 15. The change of the retry limit when the contract rate increases from 0 to 0.4 Mbps.

Table 1
Average per-flow loss rates and RTTs with the proposed loss rate differentiation scheme.

Contract rate (Mbps)		0	0.1	0.2	0.3
Loss rate (%)	AF flow	4.1	2.2	3	2
	BE flow	4.2	5.7	5.7	8.6
RTT (ms)		546	600	539	506

higher PHY data rate such as 54 Mbps of 802.11g, Flow B may achieve a larger proportion of the channel capacity than with 11 Mbps of 802.11b, because the channel time captured by Flow A and C would be decreased. However, it is still hard for Flow B to achieve its contract rate due to the topology. In Fig. 13(b), it is shown that the proposed rate control scheme is effective to provide forwarding assurance to AF packets.

4.3. Evaluations in an 8-hop chain topology

In this section, we perform a set of simulation with an 8-hop chain topology as shown in Fig. 11(a). We inject four TCP flows, and one of them is subscribed to the AF service with the contract rate from 0 to 0.35 Mbps. Other flows send only BE packets. In Fig. 14, we compare our two proposed schemes, the rate control scheme and the RTS retry limit adaptation scheme (labeled as Adaptive RTS in the figure) with *CW* based differentiation. In the figure, we show the achieved throughput of the AF flow and the aggregate achieved throughput of BE flows.

Unlike Fig. 12, it is observed that the AF flow does not achieve its contract rate with the rate control scheme. It is due to that the rate control scheme is not effective in this single chain topology since each node has the same amount of AF traffic. For the same reason, the *CW* based differentiation scheme also fails to provide service differentiation. On the contrary, it is shown that the RTS retry limit adaptation scheme effectively provides throughput differentiation through loss rate differentiation at the link layer when the network is congested.

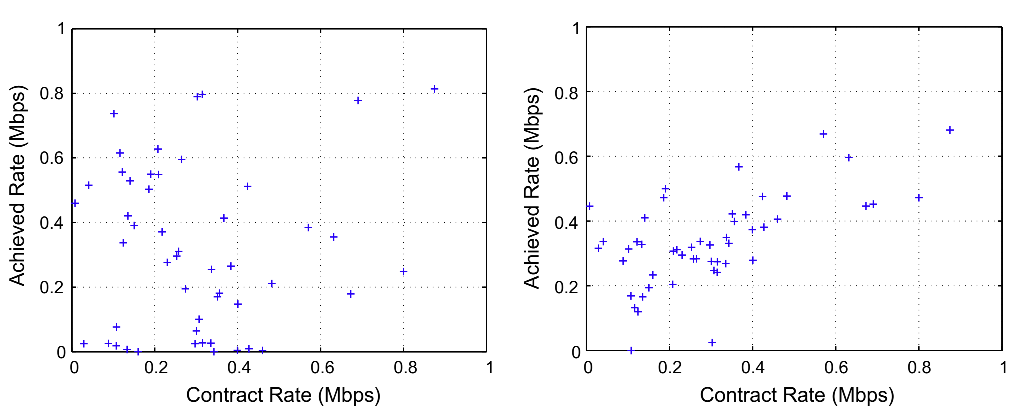
In Fig. 15 and Table 1, we observe the detail operation of the RTS retry limit adaptation scheme. Fig. 15 shows the change of the RTS retry limit over time when we increase the contract rate from 0 to 0.3 Mbps for every 20 s. As the contract rate increases, it is shown that the RTS retry limit for AF packets also increases, and the retry limit for BE packets decreases. Consequently, as shown in Table 1, the packet loss rate of the AF flow decreases as the contract rate increases while the loss rate of the BE flows increases. Here, we may notice that RTT does not increase regardless of the contract rate, and this confirms that the RTS retry limit adaptation scheme can reduce AF packet losses without increasing the average delay.

4.4. Evaluations in a grid topology

In this section, we perform simulations with 25-node (5×5) grid topologies. Each node is 200 m apart from its neighbors as in Fig. 11(b). We randomly pick five pairs of nodes and inject a TCP flow to a pair of nodes. We set randomly picked contract rates from 0 to 1 Mbps to five TCP flows. We repeat this simulation ten times with different contract rates, and then represent the result in Fig. 16. The result shows that the proposed scheme can provide service differentiation in multi-hop wireless networks. Table 2

Table 2
The number of packets sent and loss rates in a grid topology

Run	AF		BE	
	Loss rate (%)	Packets sent	Loss rate (%)	Packets sent
1	0.11	7846	1.41	3764
2	0.2	7642	0.78	4383
3	0.01	8059	3.02	2885
4	0.02	8050	3.24	2096
5	0	7082	2.13	4696
6	0.06	7792	2.91	2028
7	0	6751	1.92	4168
8	0.24	7150	0.37	4571
9	0.04	7282	1.93	3984
10	0	8136	2.68	3167



(a) CWbaseddifferentiation (b) Ratecontrol+RTSretrylimitadaptation

Fig. 16. Achieved rates of five TCP flows with randomly picked contract rates from 0 to 1 Mbps in a grid topology.

shows the number of packets sent and loss rates at the link. It is observed that the proposed scheme can protect AF packets from being lost.

5. Conclusion

In this paper, we have proposed a rate control scheme and a RTS retry limit adaptation scheme for the AF service in multi-hop wireless networks. While most prior approaches for the AF service have employed delay based differentiation, the proposed architecture provides throughput differentiation through rate control and loss rate differentiation. Through extensive simulation in various topologies, it has been shown that proposed schemes are effective to provide the AF service.

Acknowledgements

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