RTS Threshold Variation Study on Contention-based IEEE 802.11n EDCA for Real-time Services

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Abstract. As a “wide-deployed” wireless local area networks (WLANs) standard, the IEEE 802.11n was designed to support a high data transmission rate (up to 600 Mbps) and maintain a compatibility with the previous versions. Due to a substantial demand of real-time data services, such as VoIP and video conference, an enhanced quality of service (QoS) mechanism becomes crucially important. The IEEE 802.11n standard inherits the enhanced distributed channel access (EDCA) mechanism from the IEEE 802.11e, as the MAC layer improvement. It classifies the traffic into four different classes of data, called access categories (ACs). However, EDCA still suffers from a data packet collision, caused by hidden nodes problems. The packet loss ratio and end-to-end delay increase along with the escalation of real-time traffic. To address the issue, we conduct a simulation in Riverbed Modeler 17.5 environment with a 128, 256, 512, and 1,024 Bytes of request to send (RTS) threshold variation value to overcome a collision caused by hidden nodes problem. The result shows that the 512 Bytes of RTS threshold value gives the best QoS output among them all.

1. Introduction
The telecommunication traffic through the wireless local area networks (WLANs) and mobile devices will account for more than 63% of total world IP traffic, while wired devices managed for 37% by 2021 [1]. As the latest widely used WLANs technology, IEEE 802.11n [2] standard already gave us a very high data rates at both physical (PHY) and MAC layer by implementing the latest advances in wireless communications technology. The standard operates at 2.4 GHz and 5 GHz of bands frequency, which employed multi-input multi-output (MIMO) antennas for producing maximum throughput up to 600 Mbps [3]. However, IEEE 802.11n WLAN standard still inherit the same roots from its predecessor [4], which initially designed merely for data traffic applications, and not considering a quality of service (QoS) requirements.

The ”legacy” distributed coordination function (DCF) and point coordination function (PCF) MAC schemes mechanism, that was implemented prior to IEEE 802.11n, no longer satisfy the demand of real-time applications [5]. Both of DCF and PCF MAC schemes do not support any prioritization of data traffic [6]. In 2005, IEEE 802.11e was deployed to support quality of service (QoS) over WLANs [7]. Its MAC employs a channel access function called the
hybrid coordination function (HCF), which includes contention-based channel access, called enhanced distributed channel access (EDCA), and contention-free centrally controlled channel access, called HCF controlled channel access (HCCA).

DCF and PCF are soon replaced by HCF, with a new access types to ensure higher bandwidth. It also reduces latency of high priority network traffic, and thus, provides high throughput performance. As an extension of DCF, EDCA provides an enhanced QoS based on the priority classes. It classifies the traffic into four different classes of data called access categories (ACs), as depicted in Fig. 1. EDCA mechanism still inherit the basic principle of CSMA/CA. However, the channel access parameters are defined per ACs, called arbitration inter-frame space (AIFS), instead of distributed inter-frame space (DIFS) that employed in DCF [8]. As shown in Table 1, the value of AIFS (AIFSN) varies to every different ACs. The highest priority AC got the smallest value of AIFSN, and vice versa [8, 9]. Since EDCA give the higher priorities to real-time traffic, e.g., voice and video, so it is likely suitable for such transmission. However, EDCA still suffers from a data packet collision, ultimately caused by the hidden nodes problems. This phenomenon can potentially affect any nodes for cannot hear each other even though they are an adjacent WLANs.

Table 1. Default EDCA access categories.

| Priority | AC | CWmin | CWmax | AIFSN | TXOP Limit | Designation |
|----------|----|-------|-------|-------|------------|-------------|
| Lowest   | BK | 31    | 1023  | 7     | 0          | Background  |
|          | BE | 31    | 1023  | 3     | 0          | Best Effort |
|          | VI | 15    | 31    | 2     | 3.008 ms   | Video       |
| Highest  | VO | 7     | 15    | 2     | 1.504 ms   | Voice       |

The employment of request to send (RTS) and clear to send (CTS) mechanism can overcome a collision caused by hidden nodes problem. As the blind nodes cannot receive any packets from the other nodes, it will lead to collisions and packet loss [10, 11, 12]. Fig. 2 shows a hidden node scenario consists of three nodes, A, B, and C. In this scenario, node B can receive packets
from both nodes A and C, since it’s located in the data range of both nodes. However, node A is hidden from node C and vice versa. In this case, if both nodes A and C send packets to node B at the same time, then it would cause a data packet collision at node B. At this point, RTS/CTS protocol will allow the access points (APs) to regulate the utilization of the air interface media, so the stations (STAs) could see the state of the other stations that will process the packet transmission at the same time. Fig. 3 shows how the protocol works. The source node broadcasts an RTS packet. When the destination node receives the RTS packet, it will wait for the short inter-frame space (SIFS) time interval, then send a CTS packet to the source. When the source node transmits the data, the destination should send an ACK packet. Meanwhile, the network allocation vector (NAV) is used by any stations to determine channel access priority and future traffic on the medium, based on the duration information that announced in RTS/CTS frames before the actual data exchange [13]. The employment of the RTS/CTS mechanism could be done by setting the threshold value.

For a better understanding, the rest of this paper is organized as follows. In section 2, we cover the related works. It is followed by the proposed method in section 3, and results of the simulation is discussed in section 4. Finally, the conclusions are presented in section 5.

2. Related Works
Several works related to RTS/CTS mechanism in IEEE 802.11n have been published in recent years [13, 14, 15, 16]. Singh et al. [13] discuss the performance of the RTS/CTS mechanism in video conference service with RTS threshold variations of 128, 256, 512, and 1,024 Bytes. There are five deployed scenarios with 10, 20, 30, 40, and 50 of users density to observe QoS parameters e.g., throughput, media access delay, retransmission attempts, and end-to-end delay. The results show that 512 Bytes of RTS threshold value gives the best overall output.

In 2015, Maraj et al. [14] discuss the effectiveness of the RTS/CTS mechanism to improve the efficiency in the heavy traffic condition in IEEE 802.11g/n with 20 workstations. The observed output parameters are data dropped and media access delay. The result shows the employment of the RTS/CTS mechanism in IEEE 802.11g can reduce the data drop and improve the network performance better than IEEE 802.11n.

In 2014, Khadrah et al. [15] compared the DCF and EDCA protocol performance to support QoS requirements, e.g., end-to-end delay, packet loss, and jitter in voice and video services. They found a collision was detected on the EDCA access method when carrying out the real-time traffic transmission process. The collision arises along with the traffic escalation, causing
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Figure 4. The network topology.

a delay and packet loss, even though, it still be tolerable in a specific small amount of traffic.

To address the issue, we employ an RTS/CTS mechanism with a 128, 256, 512, and 1,025 Bytes of RTS threshold variation value (as has been done in [13]), to overcome a collision caused by hidden node problem. Since we are only focused on studying the impact of those RTS threshold variations, towards its effect on data packets collision in EDCA (as happened in [15]), we only deployed ten nodes of the user. The key parameters in this study are packet loss ratio, delay, and throughput.

3. Proposed Method
This study was simulated in Riverbed Modeler 17.5 environment. We uses a model to analyze the performance of several RTS threshold variation values in the IEEE 802.11n WLAN standard. In this section, we will elaborate how the simulation is conducted.

3.1. System Design
Fig. 4 shows the deployed networks topology simulated for this project. The center node is the AP, surrounded by 10 nodes of users, while the servers (VoIP and video conference servers) are located outside the circle. A switch connects both servers and access point through 100BaseT (100 Mbps over twisted pair) links. Two another objects located on the top left of the workspace are Application and Profile Definitions (to organize the services, its values, and parameter configurations).

The Application Definition has a role as a declaration node for several employed services, as depicted in Fig. 5 and 6. In Fig. 5, we could see the VoIP service configurations uses a G.711 encoder scheme, which has 160 Bytes of payloads (1 packet = 160 Bytes). Therefore, we set the “voice frames per packet” value to 5, since each voice frame has a size of 32 Bytes [17]. The type of service (ToS) value is set to interactive voice (6), which indicates the AC category

Figure 5. VoIP application definition.  Figure 6. Video conference application definition.
according to voice service on the EDCA access method. Meanwhile, on video conference service configuration, we use low resolution frame (frame inter-arrival time is 10 frames/sec), with frame size information for $128 \times 120$ pixels, which has a frame equivalent to 17,280 Bytes of payloads, as shown in Fig. 6. The value of ToS is set to interactive multimedia (5) which indicates that video conference has a lower priority compared to VoIP in EDCA access methods.

After the services has been declared in the Application Definition, they must be set in the Profile Definition node. Table 2 shows the Profile Definition table configurations. The simulation is started at 70th second, shown by the accumulation of start time (Constant(60)) and start time offset (Constant(10)).

| Parameter                | VoIP          | Video Conference |
|--------------------------|---------------|-------------------|
| Profile name             | VoIP          | Video conference  |
| Application name         | VoIP application | Video conference application |
| Start time offset (sec)  | Constant(10) | Constant(10)      |
| Start time (sec)         | Constant(60) | Constant(60)      |
| Duration (sec)           | End of profile | End of profile    |
| Repeatability            | Once at start time | Once at start time |
| Operation mode           | Simultaneous | Simultaneous      |

The RTS mechanism configurations in the AP are summarized in Table 3. We employ an RTS threshold values variations of 128, 256, 512, and 1,024 Bytes. The AP works in IEEE 802.11n WLAN standard with 2.4 GHz of frequency bands and a data rate of 65 Mbps / 600 Mbps. The RTS threshold range aims to determine the maximum size of the data packet. If the sent packet is higher than the RTS threshold value, then the station will send the RTS packet to the intended wireless node.
3.2. Scenarios
The performance parameters tested in this simulation are the packet loss ratio, delay, and throughput. These three variables are then simulated towards RTS threshold variations. The detailed scenarios of this work are summarized in Table 4.

| Scenario | RTS Threshold (Bytes) | Services | QoS Parameters |
|----------|------------------------|----------|----------------|
| 1        | 128                    | VoIP &   | Packet loss, delay, |
| 2        | 256                    | Video conference | & throughput |
| 3        | 512                    |          |                |
| 4        | 1,024                  |          |                |

4. Results and Discussion
In this section, we evaluate the effect of RTS threshold variation on VoIP and video conference services based on the arrangement in Table 4.

4.1. Packet Loss Ratio
As depicted in Fig. 7, VoIP service has an obvious lower average packet loss ratio compared to video conference. Since video conference got a lower priority compared to VoIP, it must wait for being transmitted, resulting in a higher packet loss ratio compared to VoIP. From all of the deployed scenarios, only in 512 Bytes of RTS threshold value employment has the lowest packet loss ratio for both services, as its traffic sent and received shown in Fig. 8. Hence, it shows that the 512 Bytes RTS threshold value gives a better effect to downturn the packet loss by preventing the occurrence of a collision. However, the overall results of the packet loss ratio in all scenarios are categorized good based on TIPHON standard [18].

Figure 7. Packet loss ratio of real-time services.

Figure 8. Packet loss ratio on 512 Bytes RTS threshold.
4.2. Delay
Fig. 9 shows that the most considerable delay occurred in video conference service compared to the VoIP. Since the video conference has more data packet to transmit i.e., image and sound, it requires a longer delivery time than VoIP service, which makes it vulnerable to being influenced by delay. From all of the deployed scenarios, the 512 Bytes of RTS threshold value employment in scenario 3, has the lowest delay for both services, as its fluctuation along the simulation time is shown in Fig. 10. The overall results of the delay in all scenarios are categorized good based on TIPHON standard [18].

![Figure 9. Delay of real-time services.](image1.png)

![Figure 10. Delay on 512 Bytes RTS threshold.](image2.png)

4.3. Throughput
Throughput has an inverse relation towards the packet loss ratio parameter. Since the packet loss ratio in VoIP services is much smaller than the video conference, so it gains a better throughput, as depicted in Fig. 11. From all of the deployed scenarios, the 512 Bytes of RTS threshold value employment in scenario 3, has the highest throughput for both services, as its fluctuation along the simulation time is shown in Fig. 12. The overall results of the throughput in all scenarios are categorized good based on TIPHON standard [18].

![Figure 11. Throughput of real-time services.](image3.png)

![Figure 12. Throughput on 512 Bytes RTS threshold.](image4.png)

4.4. Discussion
From all four examined scenarios, the detailed packet loss ratio, delay, and throughput that affected by RTS threshold value variation are summarized in Table 5. The escalation of the
RTS threshold value slowly improves the network performance in real-time services, although it is not significant. The best RTS threshold value performance is shown in scenario 3, with 512 Bytes. However, not every increment will give a good impact on the network performance, as shown by scenario 4 with 1,024 Bytes. Overall performance of the output parameter calculation from all scenarios are excellent, as set by TIPHON standard [18].

The employment of the RTS/CTS mechanism can guarantee the transmission of data packets through the network without experiencing a collision. However, if there is no hidden nodes issue found in the network, the RTS/CTS mechanism instead will cause an overhead, that will resulting in degradation of throughput. Therefore, when applying a 1,024 Bytes of RTS threshold value in scenario 4, the threshold value will only cause a network overhead, which adversely affects the output parameters. Those value will disrupt the traffic delivery process, resulting in increased delay and packet loss, and decreasing throughput value. According to [19], if the network already experienced an eminent performance, marked by the lowering of the packet loss ratio and delay, followed by the increase of throughput, then upgrading the RTS threshold value is not necessary.

| AC  | Scenario | Delay (ms) | Packet Loss Ratio (%) | Throughput (Mbps) |
|-----|----------|------------|-----------------------|-------------------|
| Voice | 1 | 0.97813 | 3.20492 | 3.15923 |
|      | 2 | 0.89916 | 3.18112 | 3.16542 |
|      | 3 | 0.84842 | 2.96250 | 3.27884 |
|      | 4 | 0.98688 | 3.72496 | 3.04485 |
|      | Average | 0.92815 | 3.26837 | 3.16209 |
| Video | 1 | 3.41018 | 8.23321 | 2.78940 |
|      | 2 | 3.29818 | 8.10772 | 2.80799 |
|      | 3 | 2.52427 | 7.87113 | 2.90866 |
|      | 4 | 3.46027 | 8.32533 | 2.69520 |
|      | Average | 3.17322 | 8.13435 | 2.80031 |

5. Conclusion
In this paper, we have simulated the RTS threshold value variation for 128, 256, 512, and 1,024 Bytes in Riverbed Modeler 17.5 simulator environment. Based on the results and discussion in section 4, the employment of 512 Bytes of RTS threshold value gives the best QoS output, due to the active RTS/CTS frame in reducing the number of collisions. The improvement of network performance characterized by the reduction of delay in the video conference and VoIP services for 2.52427 ms and 0.84842 ms, respectively. Packet loss ratio increases for 7.87114 % in video conference service, and 2.96250 % in VoIP. Meanwhile, the throughput value in video conference and VoIP also escalates for 2.90866 Mbps and 3.27885 Mbps, respectively. On the other hand, the worst scenario resulted by 1,024 Bytes RTS threshold value. This thing happened due to the occurrence of the networks overhead, which lead to quality degradation. The delay in video services escalates for 3.46027 ms, while in VoIP rises for 0.9281475 ms. The packet loss ratio for video conference and VoIP services also increase for 8.32533 % and 3.73496 %, respectively — meanwhile, the throughput of VoIP decline for 3.27885 Mbps, and 2.90866 Mbps for video conference.
Acknowledgement
The authors would like to thank Lembaga Penelitian dan Pengabdian Masyarakat (LPPM), Institut Teknologi Telkom Purwokerto for funding this project.

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