Proposed Enhancement of IEEE802.11e WLAN Through Real Time Simulation Study

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Abstract: *One of the most successful WLAN protocols is IEEE 802.11. That is due to the simplicity and robustness against failures of its medium access control protocol. In addition, IEEE introduces the standard 802.11e for quality of service support. However, this standard is not considered efficient when used for some applications that don't involve data and background in the transmission, but only voice and video. Some of these applications are video conferencing and internal organizations' voice over internet protocols calls. For the purpose of enhancing voice over internet protocols and video streaming over IEEE802.11e WLAN, we have developed an ns-2 patch in C++ which suites our requirements in a simulation based performance enhancement. In this paper, the performance of IEEE802.11e WLAN is evaluated and discussed based on simulation study using the network simulator (ns-2.29) under Linux operating systems fedora core 4. Our simulation results showed enhanced performance for the voice and video traffics over the original IEEE802.11e standard. This shows the effectiveness and efficiency of our simulator of enhancing the performance for voice and video based applications such as video conferencing and voice over internet protocols.*

Keywords: *IEEE 802.11e, medium access control, quality of service, hybrid coordination function controlled channel access, enhanced distributed channel access, coordination function.*

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1. Introduction

There are various versions of IEEE802.11 WLAN in the market, and each applies different modulation technique and operates in different frequency bands. For example, the IEEE 802.11b version provides data rates up to 11 Mb/s on the wireless medium, applying Complementary Code Keying (CCK) and Direct Sequence Spread Spectrum (DSSS) as modulation technique. It operates in the Industrial, Scientific, and Medical (ISM) band at 2.4 GHz. On the other hand, the IEEE 802.11a version operates in the unlicensed 5 GHz band, and provides data rates up to 54 Mb/s on the wireless medium, applying the multicarrier technique Orthogonal Frequency Division Multiplexing (OFDM) as the modulation technique [7]. The 802.11g version applies the same multicarrier modulation technique as 802.11a, but operates in the 2.4 GHz ISM band like 802.11b. However, due to channel conditions and protocol overhead, the maximum achievable throughput on the MAC layer is less than the data rate available on the wireless medium for the mentioned IEEE 802.11 versions [5, 9].

To date, 802.11 WLAN can be considered as a wireless version of Ethernet supporting best effort service (e.g., mail, browsing … ,*etc*). However, the need of wireless networks that support Quality of Service (QoS) has recently grown. In addition, the increasing needs of transmitting voice, video, and other multimedia applications with high-speed Internet access over WLANS made it necessary to have such networks. Relatively, the idea of enhancing the 802.11 MAC protocols and upcoming with the 802.11e (QoS enabled version of IEEE 802.11) was initiated. 802.11e adds QoS features and multimedia applications support to the existing 802.11b and 802.11a wireless standards, while maintaining full backward compatibility with these standards [1, 2].

As the raw data rate at the PHYsical (PHY) layer of IEEE 802.11 standard is now up to 54 Mbps, applications such as VoIP over WLAN and video streaming become feasible. However, the MAC protocol in the original 802.11 standard was designed with best-effort applications in mind and thus cannot meet the basic quality of service QoS requirements for these emerging applications. To address this issue, the IEEE 802.11e working group was established to strengthen QoS support at the MAC layer. Although the IEEE 802.11e has not been finally ratified, it has already received much attention from the research community.

IEEE 802.11e provides a channel access function, called Enhanced Distributed Channel Access (EDCA). It also provides a controlled medium access function, referred to as Hybrid Coordination function controlled Channel Access (HCCA), support applications with QoS requirements [3].

2. Why IEEE802.11e

The widespread of multimedia data and applications transmission over wireless LAN has made it necessity to a QoS support for the IEEE 802.11 standard. Therefore, IEEE 802.11 task group has created a special version, which is 802.11e, which adds a set of QoS enhancement to the original 802.11 MAC [3, 10]. The aim of QoS in wireless networking is to provide priority including channel bandwidth, controlled and bounded jitter and delay (required by some real-time applications such as video streaming and VoIP), minimize the probability of losing and dropping packets. However, providing priority for specific stations to transmit does not mean that the other stations will be ignored [2].

3. IEEE802.11e Original Standard MAC Functions

3.1. Enhanced Distributed Channel Access and Coordination Function

IEEE 802.11e provides a channel access function, called Hybrid Coordination Function (HCF), to support applications need QoS requirements. The HCF includes both a contention-based channel access and a centrally controlled channel access. The contention-based channel access of the HCF is also referred to the EDCA.

A new concept, transmission opportunity (TXOP), is introduced in IEEE 802.11e. A TXOP is a time period when a station has the right to initiate transmissions onto the wireless medium. It is defined by a starting time and a maximum duration. A station cannot transmit a frame that extends beyond a TXOP. If a frame is too large to be transmitted in a TXOP, it should be fragmented into smaller frames.

The EDCF works with four Access Categories (ACs), which are virtual Distributed Coordination Functions (DCFs), where each AC achieves a differentiated channel access. This differentiation is achieved through varying the amount of time a station would sense the channel to be idle and the length of the contention window during a backoff.

The EDCF supports eight different priorities, which are further mapped into four ACs, shown in Table 1. Access Categories are achieved by differentiating the Arbitration InterFrame Space (AIFS), the initial window size, and the maximum window size [2, 3]. Each priority level in EDCF has a different backoff increment function. Assigning a short contention window to higher priority stations ensures that in most cases, high priority stations are more likely to access the channel than low-priority ones.

Each station has a different Distributed coordination function InterFrame Space (DIFS) according to its priority level. In IEEE 802.11, ACKnowledgement (ACK) packets have higher priorities than data packets. An ACK packet is sent after sensing the medium for a time of SIFS, whereas the medium has to be sensed for a longer time (equal to DIFS) to send an data packet. This relation is shown in Figure 1.

Table 1. EDCF supports eight different priorities mapped into four ACs [2].

Priority	Access Category	Designation
1	0	Background
2	0	Background
0	0	Best Effort
3	1	Video Probing
4	2	Video at 1.5 Mbps
5	2	Video at 1.5 Mbps
6	3	Voice at 64 Kbps
7	3	Voice at 64 Kbps

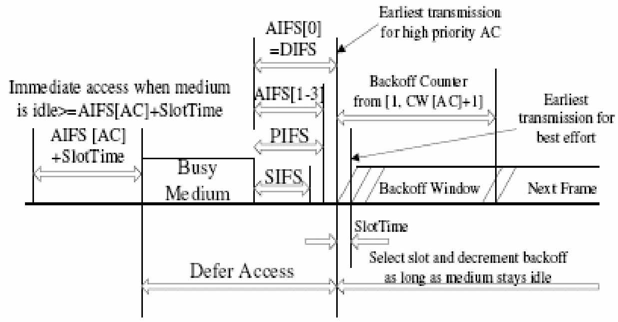

Figure 1. EDCF channel access IFS relation [8].

3.2. Hybrid Coordination Function Controlled Channel Access

The controlled medium access of the HCF, referred to as HCF HCCA extends the EDCA access rules by allowing the highest priority medium access to the Hybrid Coordinator (HC) during the CFP and the CP. The details about the controlled medium access are summarized in this section. A TXOP can be obtained by the HC via the controlled medium access. The HC may allocate TXOPs to itself to initiate MSDU Deliveries whenever it requires, after detecting the medium as being idle for PIFS, and without backoff. To give the HC higher priority over legacy DCF and EDCA access, Arbitration InterFrame Space Number (AIFSN) must be selected such that the earliest medium access for EDCA stations is DIFS for any AC. During CP, each TXOP of an 802.11e station begins either when the medium is determined to be available under the EDCA rules, that is, after AIFS plus the random backoff time, or when a backoff entity receives a polling frame, the QoS CF-Poll, from the HC. The QoS CF-Poll from the HC can be transmitted after a PIFS idle period, without any backoff, by the HC. During CFP, the starting time and maximum duration of each TXOP is also specified by the HC, again using the QoS CF-Poll frames. During CFP, 802.11e backoff entities will not attempt to access the medium without being explicitly polled, hence, only the HC can allocate TXOPs by transmitting QoS CF-Poll frames, or by immediately transmitting downlink data. During a polled TXOP, a polled station can transmit multiple frames that the station selects to transmit according to its scheduling algorithm, with a SIFS time gap between two consecutive frames as long as the entire frame exchange duration is not over the allocated maximum TXOP limit [2]. The HCCA mechanism is designed for the parameterized QoS support, which combines the advantages of PCF and DCF.

4. Limitations of the Original IEEE802.11e

IEEE802.11e uses four queues with eight different priorities as mentioned previously in Table 1. This will not be efficient for some organizations which utilize most of their wireless networks for VoIP and video conferencing applications. According to the original standard, two queues will be used for background and best effort data with three different priorities. On the other hand, if we consider a scenario where ten stations are transmitting VoIP and video with one station transmitting best effort data, it will not be efficient to use two queues with three different priorities for the best effort station. In the next sections, we propose our ns-2 simulator which will overcome the mentioned limitations of the original standard when uses for VoIP and video applications.

5. Performance Enhancement Using ns-2 Based Simulation

We conducted our experiment under Linux Operating system (Fedora Core 4) using Network simulator 2. NS (version 2.29) is an object-oriented, discrete event driven network simulator developed at UC Berkely written in C++ and OTcl. NS is used for simulating local and wide area networks. In our simulation, we have considered three queues to maximize the utilization of the VoIP and video applications in the network. We have also changed some of the simulation parameters such as CWmin, CWmax, and AIFSN in the original IEEE802.11e standard [6].

Table 2 shows the IEEE 802.11e MAC parameters values used in the simulation. We choose IEEE 802.11b PHY layer, and the PHY data rate is set 11 Mbps. The simulation parameters are shown in Table 3.

Table 2. IEEE 802.11e MAC Parameters.

Parameter	Value
Slot time	20 us
Beacon interval	100 ms
Fragmentation threshold	1024 Bytes
RTS threshold	500Bytes
SIFS	20 us
PIFS	40 us
DIFS	60 us
MSDU (Voice and Video)	60 ms
MSDU (data)	200 ms
Retry limit	7
TXOP limit	3000 us
CAP rate	21 us
CAP max	8000 us
CAP timer	5120 us

Table 3. Simulation parameters of the first scenario.

Simulation Parameter	Voice	Video	Best effort
Transport protocol	UDP	UDP	UDP
CWmin	3	7	15
CWmax	7	15	1023
AIFSN	2	2	3
Packet size (bytes)	160	1280	1500
Packet interval (ms)	20	10	12.5
Data rate (kbps)	64	1024	960

6. Performance Metrics

6.1. Average End-to-End Delay

End-to-end delay is the difference between the time when the packet is received by the end user and the time it been sent at. For multimedia applications such as voice and video, and to meet QoS standards, packet delay must be limited and bounded by certain values to result in high performance.

The average end-to-end delay is calculated using the following equation:

$$AVERAGE\ END\text{-}TO\text{-}END\ DELAY = \frac{1}{S}\sum_{i=1}^{s}(r_i - s_i) \qquad (1)$$

where, S is the number of successfully received packets.

i is the unique packet identifier.

r_i is the time at which a packet with unique identifier is received.

S_i is the time at which a packet with unique identifier i.

6.2. Throughput

Throughout can be defined as the number of bits successfully received by the receiver divided by the total transmission period of time in seconds. In fact, for multimedia applications such as video and audio, it is necessary to have sufficient throughput to meet the requirements of QoS standards [11]. The throughput of a single communication link is calculated using the following equation:

$$THROUGHPUT = \gamma = \frac{1}{C}\sum_{F=1}^{c}\frac{1}{(t_{end} - t_{start})} * (\text{N*S*8bits}) \qquad (2)$$

where,

γ is the throughput in Mbps, t_{end} is the transmission end time in seconds, t_{start} is the transmission start time in seconds, C is the total number of flows in the network, N is the _ No. of packet received successfully, S is the successfully delivered packet size.

6.3. Packet Loss

Packet loss is defined for a receiving Mobile station as the number of packets lost or dropped during transmission. While one usually assumes that packet losses are directly proportional to latency it will be shown that this is not true in some cases. We have studied separately the packet losses due to increasing of system load, the packet losses due the limited buffer size of the access point and the packet loss due long distance and week signal. We study packet losses for various values of link delays and random movement of mobile stations within the coverage of one access point.

7. Simulation Results

For experimentation purposes, we have designed two types of scenario. Simulation scenario results are discussed in the following subsections.

7.1. First Simulation Scenario

This scenario includes a single access point with variable number of mobile stations moving randomly within its coverage area. The number of mobile stations is increased form 3 to 15 with 3 stations at a time. Every three QoS stations transmits three different types of flows (voice, video and best effort data) to the same destination, which is the access point. We choose IEEE 802.11b PHY layer, and the PHY data rate is set 11 Mbps.

We start with the throughput results for the first scenario, which is shown in Figures 2 and 3. In Figure 2, the graph illustrates the effect of increasing the number of active QoS stations transmitting data to the access point on the throughput values for the three data flows. The sending rate in this simulation is 11 Mbps, while the CWmin and CWmax size and AIFSN values as stated in Table 4.

In comparison, Figure 3 illustrates the effect of increasing the number of active QoS stations transmitting data to the access point on the throughput values for the three data flows using IEEE 802.11e standard [4] CW size and AIFSN values shown in Table 4. Our CW size and AIFSN values provide better results considering the voice and video flows, but not the best effort data flow. This is clearly observed from Figures 2 and 3. In both cases, it is clearly seen from the graphs that IEEE 802.11e provides service differentiation for different priorities when the system is heavily loaded by increasing the number of stations. When the number of stations is 3 or 6, all the data flows have equal channel capacity. However, in the case of 9, 12 and 15 stations, the channel is reserved for higher priority data flows. As we mentioned in the previous sections, voice flow has the highest priority among the others, while the best effort data flow has the lowest priority.

The average end-to-end delay is another important performance metric that should be taken into account. Figures 4 and 5 represent the results obtained from our simulation using different CW size and AIFSN values.

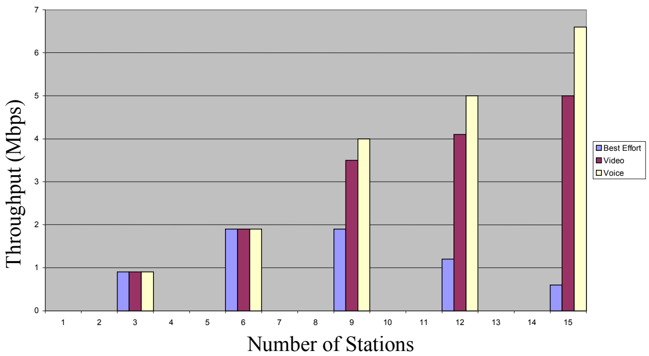

Figure 2. Effect of network load on throughput for different access categories using our CW size and AIFSN values.

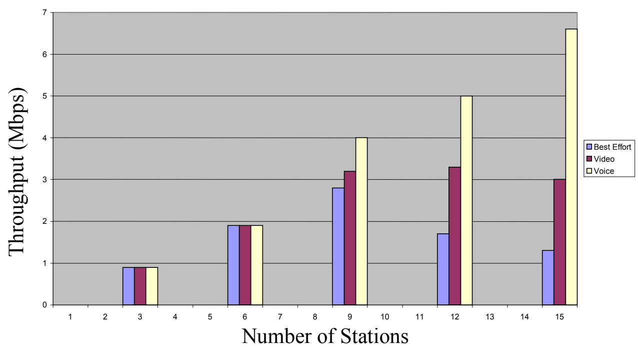

Figure 3. Effect of network load on throughput for different access categories using IEEE 802.11e standard CW size and AIFSN values.

The graphs in Figures 4 and 5 illustrate the effect of increasing the number of active QoS stations transmitting data to the access point on the average end-to-end delay values for the three data flows separately from source (mobile stations) to destination (access point). We modified the first scenario so that all the stations transmit three types of data flows. We vary the channel load by increasing the number of active QoS stations from 1 to 14. Our proposed CW size and AIFSN values enhances the performance with respect to the voice and video flows, but not for the best effort data flow. This is shown in Figure 4 when we have more than 11 active QoS stations.

On the other hand, Figure 5 represents the simulation result using the CW size and AIFSN values in Table 4. However, as shown in Figure 6, these values provide better results than ours with respect to best effort data flow. This is accepted for our project, because our main concern is to enhance the performance for multimedia data flows such as voice and video.

Another important factor that has a great effect on the IEEE 802.11e WLAN performance for QoS support is the packet drop and loss ratio. To calculate the number of packets dropped or lost in the transmission medium, we subtract the number of packet successfully received by the receiver (the access point in our case) from the total number of packets sent by the sender (mobile stations). Table 5 shows the effect of increasing the number of active QoS stations on the packet drop and loss ratio. We vary the network load by 3 stations at a time sending three different data flows. In this simulation, we maintained the same simulation parameters in Table 3.

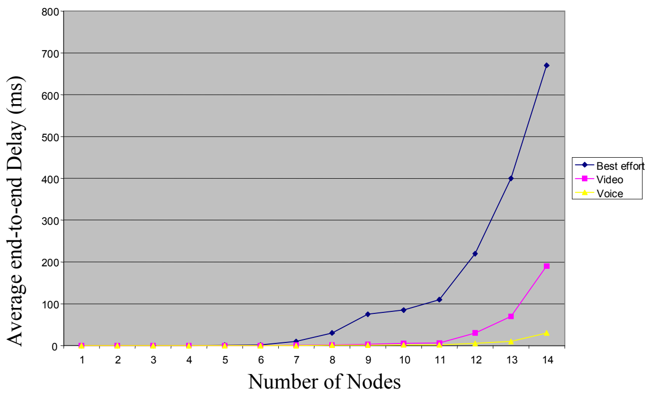

Figure 4. Effect of network load on the average end-to-end delay for different access categories using our proposed CW size and AIFSN values.

Table 4. Original IEEE 802.11e simulation parameters.

Simulation Parameter	Voice	Video	Best Effort Data
CWmin	7	10	31
CWmax	7	31	1023
AIFSN	1	2	3

It is clearly observed from Table 5 the service differentiation between the different data flows according to their priority levels. This difference appears more when the channel is heavily loaded by increasing the number of stations. For the best effort data flow, the packet drop starts when the number of stations is 3. That is due to the fact that best effort data flow has the lowest priority. On the other hand, as the voice flow is considered, the packet drop starts when the number of stations increases to 9. This reflects the fact that voice flow has the highest priority to reserve the channel when it is heavily loaded. The percentage of the packet drop for reaches up to 82% for the maximum channel load considering the best effort data flow, while it reaches up to 20% for the voice flow.

In fact, the system throughput is inversely proportional to the number of dropped and lost packets. In addition, packet drop has great effect on the network average end-to-end delay. Relatively, delay is directly proportional to the number of dropped packets.

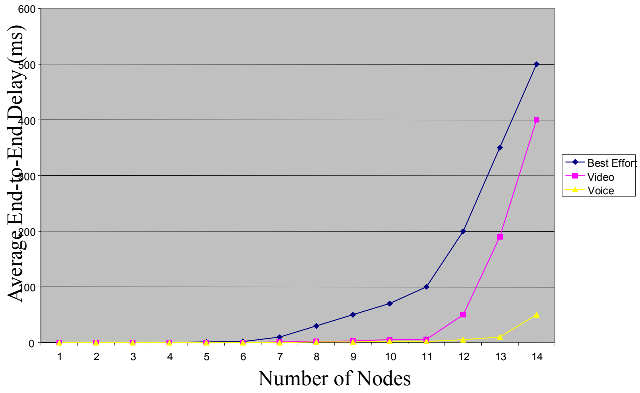

Figure 5. Effect of network load on the average end-to-end delay for different access categories using CW size and AIFSN values of IEEE 802.11e draft, 2003.

Table 5. Packet drop ratio vs. number of stations.

Number of Stations	Best Effort	Video	Voice
3	1.5%	0%	0%
6	7.67%	1.22%	0%
9	19.45%	5.49%	2.38%
12	61.55%	22.87%	9.21%
15	81.85%	44.32%	19.12%

7.2. Second Simulation Scenario

In this scenario, we consider two traffic types, video and best effort data. We have selected the video flow due to its sensitivity to delay and throughput, and the delay tolerability of the best effort data flow. Our aim in this scenario is to show the effect of the CWmin and CWmax sizes on the performance of higher priority traffic compared to the lower one. In this simulation, each traffic type includes one station, i.e. one video station, one best effort data station. The data rate of the video station is 6 Mbps and the data rate of the best effort data station is 1.5 Mbps.

We fix the CWmin and the CWmax of the data stream as 32 and 1023 respectively. Then we change the CWmin and the CWmax of the video stream to study their effect on the throughput. We can obtain the throughput of video and best effort data under all conditions.

Figure 6 shows the different throughput values of the video and best effort data traffics while changing the CWmax size and fixing the CWmin size to be 8. This will give a better idea of the effect of CWmax value in the simulation. The maximum throughput we get for the video flow is when the CWmax size is 16. The graph shows a slight decrease of the video flow throughput when we increase the CWmax size represented by the X-axis. It is observed that the throughput of the video flow increases at the cost of decreasing the throughput of the best effort data flow. Relatively, we conclude from this test that changing the value of the CWmax has small influence on the performance of EDCF. We compare the throughput value when CWmax is 16 with its value when CWmax is 1023, it changes a little for both, video and best effort data flows.

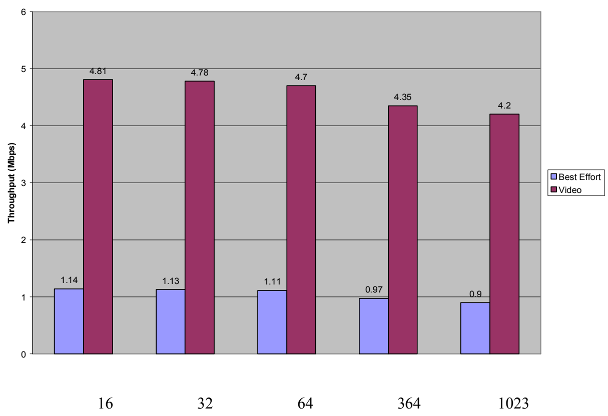

Figure 6. Effect of CWmax change on throughput of video and best effort data flows.

In comparison, Figure 7 shows the effect of changing the CWmin size while fixing the CWmax size to 32 on the throughput of the video and best effort data flows. We observe that when the value of the CWmin is 8, the throughput of the video flow achieves the saturation throughput of 5.91 Mbps. At the same point, the throughput of the best effort data flow is decreased to 0.17 Mbps.

We conclude that the CWmin size of the video flow has a greater influence on the performance of EDCF as well as on the overall performance of IEEE 802.11e WLAN. However, EDCF significantly improves the performance of high priority traffics, but this improvement is at the cost of decreasing the performance of the low priority traffics.

8. Conclusions

IEEE 802.11e introduces two enhanced MAC functions, the Enhanced Distributed Channel access and coordination Function (EDCF) and HCCA to provide sufficient throughput, bounded delay and high reliability for multimedia data flows such as VoIP and video streaming. This standard is not considered efficient when used for some applications that don't involve data and background in the transmission, but only voice and video.

With the help of our simulation patch development, we managed to show how to enhance the performance of a specific traffic in the network by varying some of the simulation parameters, which have great influence on the basic performance metrics of IEEE 802.11 WLAN protocol. Relatively, this is an indication to show that a patch has been successfully designed for the platform.

In the analysis of result part, we extracted results in order to compare IEEE 802.11e with our proposed patch against the previous protocols. The performance plots indicate the comparison. We observe that our version offers a lower packet loss and thus a higher throughput for the voice and video flows, which is very crucial for multimedia data transmission over IEEE 802.11e WLAN for QoS support.

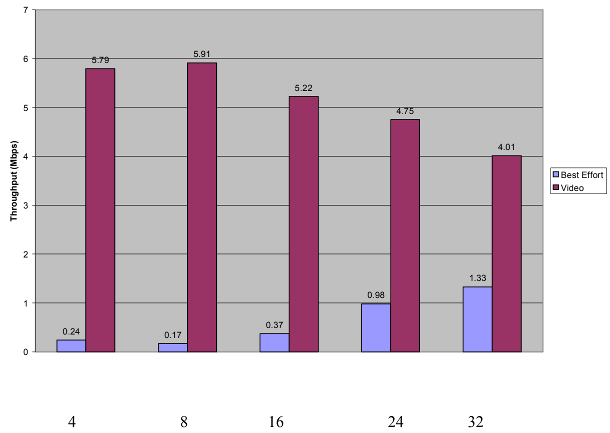

Figure 7. Effect of CWmin change on throughput of video and best effort data flows.

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