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# A Study on the Enhanced Congestion Control Mechanism for Multimedia Traffic in Sensor Networks

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***Abstract***

*Modern network sensors can gather multimedia data as well as scalar data. And these sensors compose WMSN. Wireless multimedia sensor network is sensitive to latency. Also, it transfers mass multimedia data of various forms. This thesis proposes a routing technique based on traffic priority in order to improve the multimedia data transfer by minimizing latency. In addition, it proposes a congestion control mechanism that uses packet service time, packet inter-arrival time, buffer usage, etc. In this thesis, we verified the reduction of packet latency by the priority of a packet as a consequence of performance analysis through simulation method. Also, we confirmed that the proposed mechanism maintained a reliable network state by preventing packet loss due to network overload.*

***Keywords:*** *Priority routing based Congestion Control, Wireless Multimedia Sensor Networks, Congestion Control, Packet Service Time, Packet Inter-arrival Time, Congestion Detection and Priority Routing*

## 1. Introduction

In recent years, wireless sensors have become even smaller and lighter. Also, their performance has improved significantly. Those sensors are able to collect not only such numerical data as temperature, humidity, *etc.* but also multimedia data such as image, voice, video, *etc.* With the development of these sensors, things became able to communicate with each other. Also, there have been a lot of studies on the application thereof to a variety of fields including intelligent logistics management, health care, disaster control, *etc.* [1]. In the case of multimedia data, it is required to transmit information in real-time. In addition, multimedia data is very sensitive to latency. In a wireless sensor network, a large amount of sensing data are generated instantly and flowed into a network when an event takes place. As a result, one may encounter a congestion situation. When congestion takes place, packet transmission gets delayed or packets get lost. On this account, the study on the algorithm to prevent and control a congestion having an adverse effect on transmission of multimedia data is being conducted vigorously. Moreover, it is required to develop a mechanism to rout differentially depending on the traffic priority in a congestion situation. This thesis proposes an algorithm to detect and control a congestion precisely using packet service time, packet inter-arrival time and buffer usage. Furthermore, it proposes a traffic control technique to present data transmission latency differentially depending on the traffic priority. This thesis is organized as follows. Chapter 2 analyzes the existing congestion techniques in a

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wireless sensor network and Chapter 3 proposes a congestion control method and traffic control technique. Chapter 4 analyzes the results of performance evaluation and simulation as to the proposed mechanism and Chapter 5 makes conclusions.

## 2. Related Studies

Various studies on congestion control techniques at WSN (wireless sensor network) have been conducted. PCCP (Priority-based Congestion Control Protocol) is the protocol to guarantee congestion control and transmission rate depending on the priority assigned to the sensor node [2]. Currently, one detects congestion through defining a value reflecting the congestion state of each sensor node and using this defined value. The information about a detected congestion is piggybacked to the data header using the broadcasting feature [3] of a wireless network through ICN method so that the information can be transmitted to a neighboring node.

CODA (Congestion Detection and Avoidance) finds and controls a congestion situation based on the wireless channel share and the current buffer share [4]. There are two methods in this technique. First one is open-loop hop-by-hop backpressure. One node broadcasts backpressure signals toward the source if detects congestion. And nodes that receive backpressure signals can throttle their sending rates or drop packets based on the local congestion policy. Second one is closed-loop multi-source regulation. In this method, the sink node receiving particular packets transmits ACK packet to the source node and the source node controls the transmission rate under the assumption that any case in which the number of ACK packets do not reach the threshold value for a certain period of time is congestion [5].

ESRT (Event-to-sink Reliable Transport) [6] is the technique to detect and control congestion by adjusting the reporting rate as comparing the specific reliability of a sink node and the event reliability of an actual source node when a source node sends sensing information to a sink node. If source node's buffer overflowed, the node sets CN bit to 1 and transfer it to sink node in order to congestion notification. The sink node instructs the source node to reduce transfer rate through broadcasting message.

STCP (Sensor Transmission Control Protocol) provides the technique (Congestion Detection and Avoidance Mechanism) to detect and avoid a varying congestion. In other words, it is the transport layer protocol to support a variety of applications. A sink node performs a majority of functions and supports a continuous data flow and event based flow in the same network [7]. A source node generates a session by sending an initial value containing transmission rate, flow type and required level of reliability before transmitting packets. Since then, a sink node requests for re-transmission by sending NACK signal as to those packets that do not arrive within an expected time through measuring the arrival time of packets in the case of continuous data flow. In the case of event-based flow, it stores the packets received from a source node in the buffer until receiving ACK signal based on the required level of reliability and then deletes the responded packets from the buffer when receiving ACK signal. A sink node set congestion bit at ACK signal packet and then informs the congested path to nodes. Those source nodes receiving this information will select other paths or adjust the transmission speed accordingly [8].

## 3. Proposal of Congestion Control Algorithm

This thesis proposes a technique to control congestion in accordance with the packet priority in order to improve the processing rate through traffic control while minimizing the cross-sectional latency. This allows for reliable transmission at WMSN that has a variety of traffic models. Also, it allows users to utilize a network flexibly by assigning quality level to each traffic having a different property and differing the path setting to a destination node depending on the assigned quality level. In addition, it is able to improve the condition of a network by determining the conditions of a congestion based on the proposed algorithm and adjusting the packet transmission rate for a congestion situation.

### 3.1. Priority Marking in Accordance with Traffic Characteristics

At a sensor node, the classifier classifies the quality level in accordance with the features of relevant packets and marks each packet with one of the three colors (green, yellow and red) and transmits them to a sink node. At WMSN environment exists four kinds of traffics with different characteristics from each other. First, the periodic monitoring traffic is the traffic to monitor and send the surrounding environment information such as temperature, humidity, intensity of illumination, *etc.* periodically sot a sink node. Second, the event driven traffic is the traffic that should promptly transmit any intrusive act to a sink node when it detects an intrusion or risky state within the detectable area of a sensor node. Third, the multimedia traffic is the traffic that continuously transmits video and audio information. It consists of key frame data requiring a high quality of service and the other auxiliary frame date requiring a low quality of service. Lastly, the query-based traffic is the response data to a query of a sink node; thus, it assigns a level of service quality at a sink node. Each node is able to assign a quality level of packets to be transmitted depending on the traffic pattern. The quality level is classified into the following three grades: green, yellow and red. It is included in the priority field of a packet. Green refers to the highest importance, whereas red refers to the lowest importance. There is a difference in the method to set a path depending on the quality level. A packet with green grade assigns the optimal path whenever possible. A red packet places higher priority on setting a path having a higher residual energy level than latency, whereas a yellow packet selects a path having a moderate grade.

### 3.2. Congestion Control Technique Based on Traffic Priority

The congestion control technique is classified into congestion detection, congestion notification and congestion control. To detect a congestion situation, packet service time of PCCP and inter-arrival time between packets are utilized. Packet service time refers to the time during which a packet arrives in Mac layer and is transmitted completely. Inter-arrival time refers to the time during which a packet arrives in a node and a subsequent packet arrives in the same node. In this thesis, $t_{serv}$ hereinafter refers to packet service time, whereas $t_{arr}$ hereinafter refers to inter-arrival time. The proposed technique makes calculations using packet wait time in order to detect a congestion situation more precisely. $t_{serv}$ value can be obtained after processing a specific packet. If the value of $t_{serv}$ cannot be renewed since packets were not processed due to a drastic increase in the amount of influx of packets, then the detection of a congestion situation may become delayed. To prevent this, it is required to use $t_p$ for packet wait time. $t_p$ is the value to be renewed continuously each time a node receives a packet. Thus, it represents how long a packet is processed sequentially. In the case of processing packets, they will be initialized. In the case where nodes receive other packets while these nodes are not processing the previously received packets, the time value will be accumulated. The following is the formula to obtain $t_s$ and $t_a$, which are the values to determine congestion.

$$t^{avg} = \left((t^{i-4} \times 0.6) + (t^{i-3} \times 0.8) + (t^{i-2} \times 1.2) + (t^{i-1} \times 1.4)\right)/4$$

$$t_s = \left(1 - w_{serv}\right)t_{serv}^{avg} + w_{serv} \times t_p^i \qquad (1)$$

$$t_a = \left(1 - w_{arr}\right)t_{arr}^{avg} + w_{arr} \times t_{arr}^i \qquad (2)$$

$t_{serv}^{avg}$ means the mean value after multiplying the constant value so that a more recent value among those $t_{serv}$ values that were recently processed 4 times would have a greater influence, and $t_{arr}^{avg}$ represents the value that was calculated by $t_{arr}$ instead of $t_{serv}$. $t_s$ is calculated by multiplying the value $w$ (real number between 0 and 1) to $t_{serv}^{avg}$ and $t_p$, and $t_a$ is calculated by multiplying $w$ to $t_{arr}^{avg}$ value and the current $t_{arr}$ value. Even though the network state information so far has to be taken into consideration, the current network state is more important. On this account, the weighted value should be set higher toward $t_p$ and the current $t_{arr}$. Each time a packet is processed, $t_{serv}^{avg}$ value should be renewed. Also, each time a packet arrives, $t_s$ value and $t_a$ value should be renewed periodically through Formulas referred to (1) and (2). It is required to sub-divide the aforementioned case into the following two cases by comparing the obtained $t_s$ and $t_a$: the case where ts is greater and the case where ts is smaller. Then, it is required to create an algorithm to determine a congestion by checking the buffer share. It is assumed to be in a strong congestion state when $t_s$ is greater and the buffer share is high. When the buffer share is low, it is assumed to be in a weak congestion state where a congestion situation might become severe soon. When $t_s$ is smaller, it is defined to be in a well-provisioned state since it is not congested, where the share is at an appropriate level. $t_s$ is smaller than $t_a$; however, it is still required to increase the packet generation speed in consideration of the network efficiency in an under-provision state if the buffer usage is less than 10 percent.

The method of controlling congestion can be broadly divided into ECN (Explicit Congestion Notification) and ICN (Implicit Congestion Notification)[9]. The proposed algorithm transmits the information of a congestion situation to a neighboring node by using ICN method that has fewer overheads than ECN. Those nodes receiving a notice of congestion control congestions appropriately depending on the degree of congestion.

### 3.2. Congestion Control Algorithm

Figure 1 represents the congestion control algorithm, where $t_a$ and $t_s$ are equivalent to the values obtained by Formula (1) and Formula (2) and $q_i$ refers the buffer share of Node $i$. In the case of $t_s > t_a$, the arrival time of packets is faster than the processed time; thus, those packets should be accumulated in the buffer. If the buffer usage is 40 to 60 percent, then there may be a congestion situation. As a result, it will be required to stop the generation of packets temporarily after notifying that it is in a weak congestion state. If the buffer usage is higher than 60 percent, then it is required to reduce the influx speed of packets by increasing the packet generation cycle after notifying that it is in a strong congestion state. In the case of $t_s < t_a$ and the buffer usage of 10 percent or less, it is required to increase the influx speed of pacts after notifying that it is in an under-provisioned state. Any other case is deemed reliable; thus, it will be maintained after informing that it is in a well-provisioned state. It takes some time for the transition of each state. In other words, it takes some time for each state to be changed. Thus, it is required to manage a network efficiently by limiting the state transition for a certain period of time after receiving a notice.

 

**Algorithm** Congestion Control

**Input** : $t_a$, $t_s$, $q_i$

**Begin**

```
If( t_s > t_a && q_i > 0.6 )
    reduce packet creation period
else if( t_s > t_a && (q_i > 0.4 && q_i <=0.6) )
    stop packet creation for a few seconds
else if( t_s < t_a && q_i <=0.1)
    increase packet creation period
else
    continue packet creation
```

**End**

**Figure 1. Congestion Control Algorithm**

# 4. Analysis of Simulation Result

## 4.1. Configuration of Simulation

The network configuration for the simulation is based on the placement of 100 nodes with 10x10 as shown in Figure. 2. The nodes are consisted of source node, sink node, relay node and background traffic node. A source node is set as No. 99 node; thus, it generates packets periodically. A sink node has the arrival information of packets as a destination node of packets. A relay node transmits a packet received from a neighboring node to a destination through path setting. A background traffic node generates packets in accordance with the packet generation cycle in order to set network state. Each node is able to transmit and receive packets with the neighboring nodes on the adjacent top, bottom, left and right, and these nodes limit the transmission only to the left and right in order to send packets to a sink node. The nodes have a routing table that is in accordance with the priority of packets, and transmits bad packets on this routing table.

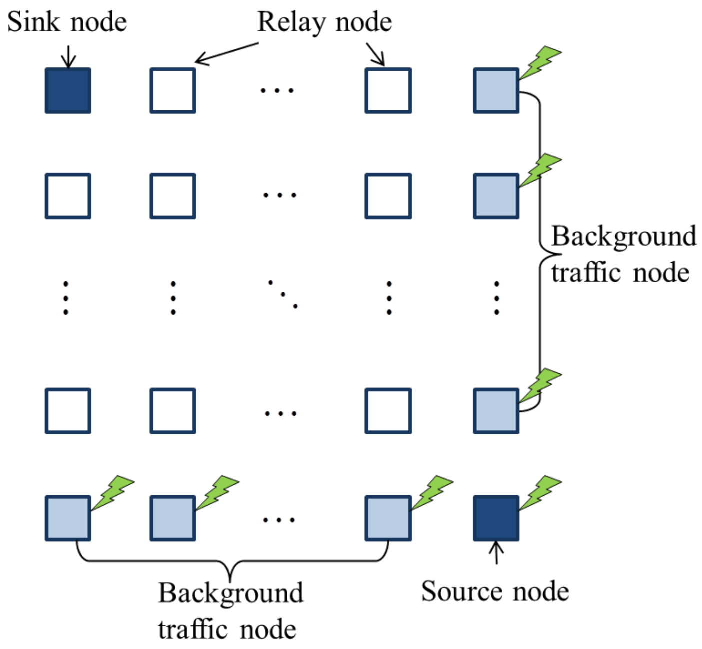


**Figure 2. Network Topology**

### 4.2. Environment of Simulation

The simulation was conducted in Visual Studio 2010 environment and the simulation program utilized C++ program. As for the link cost value between the nodes, a random value within the range (the unit of 0.02 between 1.0 and 2.0) and a difference in the transmission time was given depending on the link cost. When the link cost is 1.0, the transmission time becomes 0.96ms. When the link cost is 2.0, the transmission time becomes 1.92ms. The initial energy value of the nodes was set at 1000000 (1J). The consumed energy value was 185uj for transmission, 83uj for reception and 15uj for standby for collision. The sink node was set in a way that there would be no energy consumption. The parameters used in the simulation are as shown in Table I.

### 4.3. Analysis of Simulation Result

In this thesis, the performance was evaluated in order to verify the validity of priority based traffic control and congestion. A source node set priority on a random basis and transmits packets. Those packets having a mutually different priority are periodically generated and transmitted to a sink node. To set network state, the experiment was conducted by adjusting the packet generation cycle of background traffic nodes in the unit of 10 up to 10 to 110ms. It is assumed to be an under-provisioned network with the cycle of 10ms, which is translated into a very low degree of network load. With the cycle of 60ms, it is assumed to be a well-provisioned network. With the cycle of 110ms, it is assumed to be an over-provisioned network, which is translated into a very congested state. The packet generation cycle of a source node was fixed at 100ms, and the experiment was conducted for both of the following cases: the case of executing congestion control and the case of not executing congestion control. The below graphs show the results thereof.

 

**Table 1. Simulation Parameter**

| Parameter | Values |
|---|---|
| Field size | (0m, 0m)~(100m, 100m) |
| Sink node | (0m, 0m) |
| Source node | (99m, 99m) |
| Traffic node | (90m, 90m) ~(96m, 96m), (9m, 9m)~(69m, 69m) |
| Transmission range | 1m |
| Source packet generation cycle | 0.1s |
| Traffic packet generation cycle | 0.01s ~ 0.11s |
| Packet size | 16 bytes, 256 bytes |
| Initial node energy | 1J |
| Transmission energy consumption | 185Uj |
| Receiving energy consumption | 83uJ |
| Standby energy consumption | 15uJ |
| Simulation running time | 100ms |
| Transmit time | 0.96ms |
| Collision waiting time | 1.58ms |
| MAC protocol | IEEE 802.15.4 |

The graph created by summing the mean values of Figure 3 and Figure 4 is Figure 5. From Figure 3 and Figure 4, one can see that the priority based traffic control is valid since the latency of green packet is smaller than the latency of red or yellow packet. When the amount of incoming packets is large in the case of not executing congestion control, a network becomes paralyzed and consequently the latency of packets increases exponentially. When applying the congestion control algorithm of this thesis, this congestion situation is successfully controlled and reliably maintained. In addition, Figure. 6 represents packet loss resulting from network overload when congestion occurs. The packet generation cycle is fixed at a source node during the period of simulation; thus, it always generates 100 packets. In the case of not executing congestion control, packets will be lost when the congestion degree is 90 percent or higher. When the congestion degree is 100 percent, the packet loss rate will be 42 percent. However, the packet loss rate becomes 0 percent after applying the proposed mechanism.

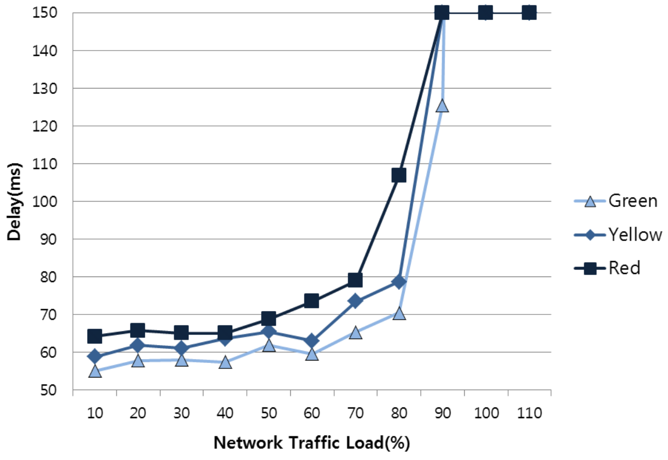


**Figure 3. Packet Delay Before Congestion Control**

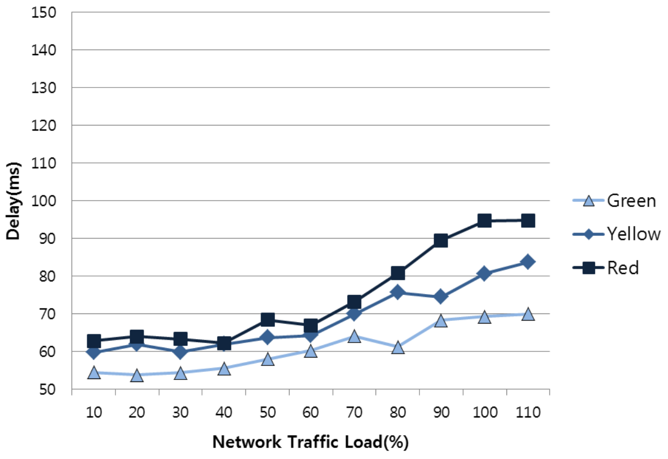


**Figure 4. Packet Delay After Congestion Control**

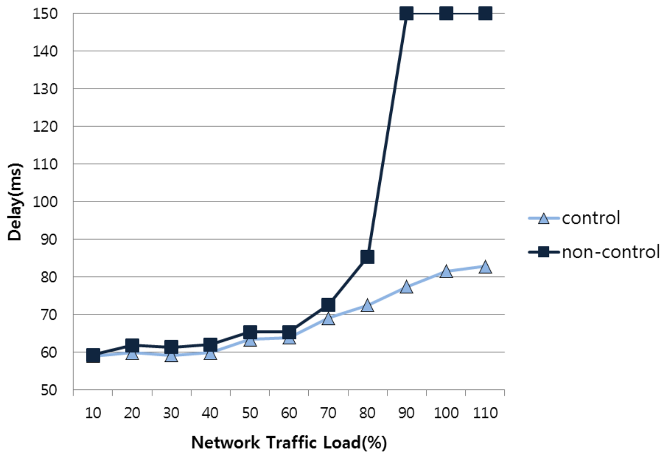


**Figure 5. Average Packet Delay as a Result of Congestion Control or Non-Control**

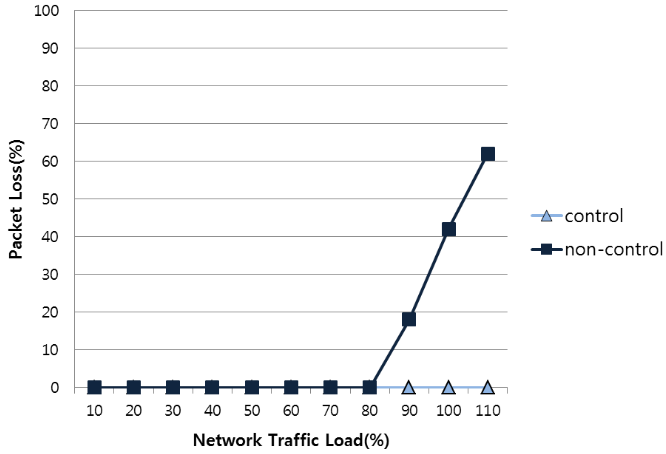


**Figure 6. Packet Loss as a Result of Congestion Control or Non-Control**

## 5. Conclusion

Multimedia information in a wireless multimedia sensor network requires transmission and calculation of a large amount of data unlike the conventional sensing information. Thus, it is difficult for the existing sensor networks to accept it. On this account, WMSNs requires a different technique from the conventional congestion control technique at WSN.

This thesis conducted a study on a technique to address a congestion at WMSNs by reducing a congestion through detecting, notifying and controlling it. The proposed technique utilized the multiplication of weighted values of packet inter-arrival time and packet service time in order to detect a congestion situation more precisely. It utilized packet wait time along with buffer share. In addition, it controlled traffics efficiently by using the quality level of packets. This thesis conducted an experiment depending on the presence of congestion control while fixing the traffic of the source node and varying the background traffic. As a result of analyzing the performance, the mean packet latency from 10 percent to 100 percent as for the degree of network load in the network leveraging the proposed mechanism of this thesis was reduced by approximately 72 percent than the network not leveraging the proposed mechanism. In addition, the packet loss rate was reduced from 42 percent to 0 percent.

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