An Integrated QoS Support Architecture for Wireless Home Network Based on IEEE 802.11 Wireless LAN

Sung-Hwa Hong* , Byoung-Kug Kim* , Doo-Seop Eom*★

Abstract

In this paper, to support a QoS level appropriate to the user in Wireless Home Network based Wireless LAN, we propose a QoS support architecture which includes Wired Network and Wireless Network. Actually, an important problem to support QoS in Wireless Home Network is approached not only on a MAC level in Wireless LAN but also on a integrated method to combine Network layer with Datalink layer. By applying the integrated QoS support method, it is possible to provide QoS support architecture using a Wireless LAN terminal with a minimum changing, and the proposed scheme has advantage of QoS support method, which is more superior than a existing scheme to support QoS in MAC level of Wireless LAN. Simulations results show that overall performance of the proposed scheme can be improved.

Key words: Home Network, QoS, WLAN, Multimedia Service

I. Introduction

The increasing reliance on information available on Internet, and the rapid growth of the wireless subscriber population build a need for the Internet users to maintain communicationsnh as they move from one place to another, and the popularity of portable computers combined with the growth of wireless networks and services has led to many efforts to make mobile computing an every reality. To achieve this goal, it is necessary for designers to collect information about the behavior of mobile hosts (portable computers) and the characteristics of the networks they use. Therefore, solutions have been proposed to support mobility in the future IP based wireless networks [1],[2].

Although the Internet offers access to information sources worldwide, typically we do not expect benefit from that access until we arrive at some familiar point – whether home, office, or school. However, wireless computing is becoming increasingly important due to the rise of notebook computers, palmtop computers, electronic organizers, and smart phones, as well as the desire for having continuous network connectivity to the Internet regardless of the physical location of the node.

In today's world, wireless communication is the fastest growing field in the telecommunications industry. Last 20 years, the mobile voice services have already transformed both business and personal interactions, but the full potential of mobile communications won't be realized until the capability extends to data. Mobile computing will represent the true wireless communications revolution, by empowering user with anywhere, anytime, anyway access to the Internet.

Wireless networks usually consist of a wired, packet-switched, backbone network and one or more wireless hops connecting WMN (Wireless Mobile Node) to wired part. The wireless part is organized into geographically-defined cell controlled by AP (Access Point) for each of these cells. These APs are on the wired network and provide a gateway for communication between the wireless infrastructure and the backbone interconnect. These wireless networks give the computer the ability of moving from one domain to another easily. But this

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causes the problem how to serve a seamless network connection to a mobile node regardless of its attachment to some domain [3],[4].

The latest advances in the Internet technologies, the dropping of PC price, and the proliferation of smart devices in the house, have dramatically increased the need for an efficient home network. A home network connects several computers and networked appliances within a house using Ethernet or wireless technology, and then connects them all to the Internet through the home gateway. Caching at the home Internet gateway is an effective solution to improve the performance of the home network.

Home networking contains the issues of connecting multimedia devices, facilitating storage, and viewing digital media on these devices. Commonly, distribution of entertainment content between devices is done via dedicated wired interfaces. However, wireless communication would offer the end user enhanced benefits of easy connectivity and mobility in the home. The use of a wireless- (rather than a wired) link for streaming audio and video introduces new challenges for the design of a home-networking system. The main challenge of a wireless entertainment system is to provide wire-equivalent quality in the presence of interference throughout the home environment. Due to the isochronous character of streaming traffic accompanied with the substantial bandwidth need for video, the requirements imposed on the wireless channel are much higher than for traditional data applications. This paper discusses these requirements for streaming applications in terms of bandwidth, latency and transmission errors. Also, the paper explores solutions to meet these requirements. These solutions are explicitly searched within the well-established 802.11 technology [1].

There are at least two kinds of applications that are expected to take advantage of the success of the IEEE 802.11 based WLAN technology, namely AV transmission in home networks and Voice over IP. Both applications require for very tight Quality of Service (QoS) support. Since the legacy 802.11 standards [1] was liable to provide adequate QoS for these applications, a new task group (TCie) in IEEE 802.11 was initiated in September 1999 to enhance the 802.11 MAC layer.

II. An Integrated QoS Support Architecture for Wireless Home Network

Basically, the DCF is a Carrier Sense Multiple Access with Collision Avoidance mechanism (CSMA/CA). In order to overcome the collision problem, the 802.11 uses a Collision Avoidance (CA) mechanism coupled with a Positive Acknowledge scheme. A station supposed to transmit performs a backoff procedure before starting a transmission. It has to keep sensing the channel for an additional random time after detecting the channel as being idle for a minimum duration called DCF Inter-frame Space (DIFS). Only if the channel remains idle for this additional random time period is the station allowed to initiate the transmission. The duration of this random time is determined as a multiple of slot time. Each station maintains a so-called Contention Window (CW)), which is used to determine the number of slot times a station has to wait before transmission. Stations use the so-called "Virtual Carrier Sense mechanism" based on Network Allocation Vector (NAV) together with the Physical Carrier Sense when sensing the medium. The PCF was designed to support some very basic form of QoS and maximize the utilization of the media. Relying on the point coordinator (PC), the PCF enables polled stations to transmit without contending for the channel.

The medium access control (MAC) protocol is one of the main factors which determine the performance of wireless LANs. A good MAC protocol for wireless local area networks (LANs) should provide an efficient mechanism to share limited spectrum resources along with simplicity of implementation.

The most popular contention-based wireless MAC protocol, CSMA-CA, becomes the basis of the MAC protocol for the IEEE 802.11 standard. However, it is observed that if the number of active users increase, the throughput performance of the IEEE 802.11 MAC protocol degrades significantly because of the excessively high collision rate.

To increase the throughput performance of a distributed contention-based MAC protocol, an efficient resolution algorithm is necessary to reduce the overheads (such as packet collisions and idle slots) in each contention cycle. To this end, many novel collision resolution algorithms have been proposed. For example, improved backoff algorithms are proposed to adjust the increasing and decreasing factor of the contention window size and randomly chosen backoff values, or to adjust dynamically the proper contention window size at each station based on the estimation of the number of active station.

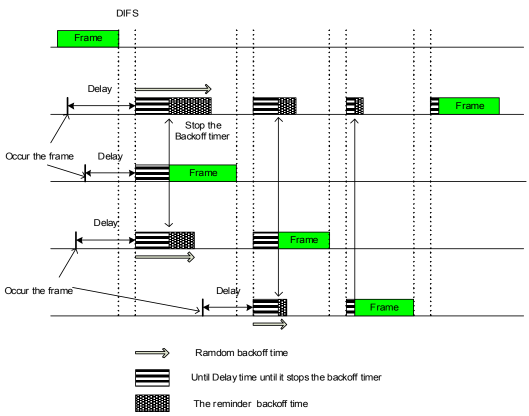

Fig. 1. Back-off mechanism

As we mentioned before, the most popular contention-based MAC protocol is the CSMA/CA, which is widely used in the IEEE 802.11 LAN's. The basic operations of the CSMA/CA algorithm are shown in Fig. 1. A packet transmission cycle is accomplished with a successful transmission of a packet by a source station and an acknowledgment (ACK) from the destination station. General operations of the IEEE 802.11 MAC protocol are as follows (since the RTS-CTS mechanism is optional [5], we only consider the distributed coordination function (DCF) without the RTS-CTS handshaking for simplicity). If a station has a packet to transmit, it will check the medium status by using the carrier sensing mechanism. If the medium is idle, the transmission may proceed. If the medium is determined to be busy, the station will defer until the medium is determined to be idle for a distributed coordination function inter-frame space (DIFS) and the backoff procedure will be invoked. The station will set its backoff timer to a random backoff time based on the current contention window size (CW).

$$\text{Backoff Time (BT)} = B * \text{aSlotTime} \quad (1)$$

where B is the backoff timer which is a randomly chosen integer from a uniform distribution over the interval between zero and the current contention window size CW (B=uniform[0,CW]), and aSlotTime is the length of a unit time slot.

After a DIFS idle time, the station performs the backoff procedure with the carrier sensing mechanism by determining whether there is any activity during each backoff slot. If the medium is determined to be idle during a particular backoff slot, then the backoff procedure will decrement its backoff time by a slot time (BTnew=BTold − aSlotTime). If the medium is determined to be busy at any time during a backoff slot with a nonzero backoff timer, then the backoff procedure is suspended. That is, if a station is deferring its packet transmission, then it will freeze the value of the backoff timer and the contention window size until next contention period. After the medium is determined to be idle for DIFS period, the backoff procedure is resumed. Transmission will begin whenever the backoff timer reaches zero. After a source station transmits a packet to a destination station, if the source station receives an ACK without errors after a short inter-frame space (SIFS) idle period, the transmission is concluded to be successfully completed. If the transmission is successfully completed, the CW for the source station will be reset to the initial (minimum) value minCW. If the transmission is not successfully completed (i.e., the source station does not receive the ACK after SIFS), the CW size will be increased (e.g., CW=2(n+5)−1, retry counter n=0,...,5), beginning with the initial value minCW, up to the maximum value maxCW (e.g., minCW=31 and maxCW=1023). This process is called the binary exponential backoff (BEB), which is intended to resolve collisions. More detailed operations can be found in [7].

As the following Fig. 2 and Fig. 3, the packet classifier plays a role of differentiating multimedia packet from general data packet. For such differentiation, transmission layer protocol of the packet is described as below. It categorizes UDP packet into real time or multimedia packet and TCP packet into general data packet. Considering that at present, most UDP-based applications are real time or multimedia applications with other few temporary packets including control packet, this packet classification is not optimal, however, it will be a practicable alternative unless other packet classification is available; even if UDP packet instead of multimedia or real time packet is categorized into multimedia packet and given priority, the number is few so it is negligible.

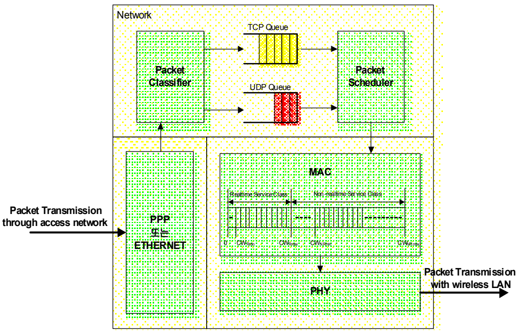

Fig. 2. A modification home gateway server

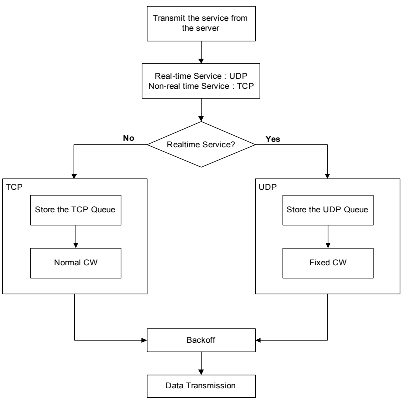

Fig. 3. A proposed algorithm

The packet categorized by the packet classifier is saved in a separate queue according to TCP packet or UDP packet. The basic concept of the method proposed in this study is to make multimedia packet serviced with top priority all the time through priority queuing, priority given upon medium access etc. In general, offering of service with top priority to the packet of one group among packets of two groups is not a desirable approach since packets of the other group may not be serviced. However, as assumed in this study, it does not cause any serious trouble once a traffic volume is limited to be given service with top priority through proper CAC, because extra resources to be allotted to traffic of the other group always remain in this situation. Although transmission of packets not being given service with top priority is increasingly delayed in this case, how fast a set of packets composing a file is transmitted – that is, throughput – instead of delayed transmission of individual packets is a more significant performance evaluation factor in most data services. Also, throughput is a factor closely related with rather bandwidth than delayed transmission. Therefore a conclusion may be deduced that even if not priority queuing but another queuing mode is used in multimedia traffic, as long as such queuing mode offers the same packet loss rate as that of priority queuing to multimedia traffic, other consequences to traffic throughput may little differ from those of priority queuing. Simplicity is the merit of priority queuing. The packet scheduler performs this as shown in Fig. 2 and Fig.3, which transmits packet, if any, in UDP queue or otherwise packet in TCP queue. Of course, if packet is absent in both queues, it must wait until the packet comes back.

III. Algorithm and the simulation results

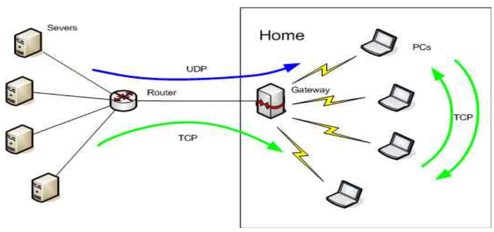

Fig. 4. The simulation model

Consumer electronics including PC in the home are wholly connected with each other or wirelessly with home gateway, and communicate with the outside through home gateway. Home gateway is connected with servers on Internet with access network offered by ISP(Internet Service Provider), but in simulation, to simplify this, it was modeled to be directly connected with individual server via router. Consumer electronics are provided with real time multimedia service using UDP from the server or non-real-time data service using TCP or communicate with each other by using TCP. In simulation, the verified NS2 was used, which is being broadly used as network simulator. Each node used RED(Random Early Detection) algorithm offered from NS2 for buffer control, and RED parameter adopted the default of NS2. MAC used both 802.11 module offered from NS2 and based on this, a modified module. Packet loss only occurs when it is abandoned by the buffer through butter control algorithm or when it is lost by the collision in MAC. The use of a separate queue from TCP packet was conceptually described earlier to give queuing priority to UDP packet, but it was implemented to use the physically same buffer, which intends to keep delayed transmission by queuing at a certain level by making the whole queue length constant although a ratio between UDP packet and TCP packet being on standby at the queue is changed. A transmission error arising out of wireless channel is not influenced by MAC or queuing algorithm, thus it was not allowed for. 500 seconds were spent on the whole simulation, and the outputs for the first 50 seconds were thrown away and those for 450 seconds thereafter were picked out. The major parameters commonly used in simulation are summarized in Table 1.

Table 1. Design summary

Bandwidth in Router-Gateway	100Mbps	Delaytime in Router-Gateway	2ms
Badwidth in Server-Router	10Mbps	Delaytime in Server-Router	2ms
Bandwidth in wireless link	10Mbpsz		

The first simulation is to see how the performance of UDP and TCP is changed according to the number of UDP connections. Each UDP connection transmits packet at a fixed transmission rate, and performance evaluation factors are packet loss rate and transmission delay. This reflects the characteristics of multimedia traffic and those of QoS as requested by multimedia service. Packet loss rate is not significant in TCP because it performs congestion control of automatically sensing network congestion and controlling transmission rate as well as error control through retransmission and in many cases, rather time taken to fully transmit a file than transmission delay of individual packet is important, thus throughput is regarded as performance measure. The number of external TCP connections(server-PC) and internal TCP connections(PC-PC) is each 5, and that of UDP connections is variable as shown in the graph. Each UDP source is CBR(Constant Bit Rate) source having 100kbps of packet rate. UDP packet loss rate by change in the number of UDP connections is shown in Figure 5.

The graph indicated in None displays loss rate when no priority is given to UDP packet. In the graph indicated in MAC, priority is given only in CW size, which is pertinent to the proposed mode. In the graph indicated in queue, priority is given to UDP just upon packet forwarding in gateway. In the graph indicated in Both, priority is given in both CW and packet forwarding, which is pertinent to the proposed mode. The aforesaid indications are applied to all graphs hereafter, as well. As easily expected, packet loss rate is lower if priority is given to UDP packet in any form compared to if it does not. If just priority in queuing is given, loss rate is low at a similar level to the case that both priorities are given when the number of UDP connections is low. On the other hand, as the number of UDP connections is increased, loss rate is sharply increased to become a situation to which no priority is given. This may be related with the result shown in Figure 6. Figure 6 shows the throughput of external TCP with change in the number of UDP connections; in two cases(None, Queue) where priority in MAC is not given to UDP, the throughput of external TCP is sharply decreased with increase in the number of UDP to be approximately 0. This implies that in these cases, as the number of UDP connections increases, the UDP packet occupies all buffers in the gateway. Then the priority held by UDP packet to TCP packet in queue becomes insignificant, so both the algorithms(None, Queue) are in the same condition. However, if queuing priority is given to UDP, UDP packet is always at the head of queue and if the number of UDP packets is low, UDP packet is hardly abandoned and consequently loss rate becomes very low, but if the number of UDP connections is very low, there is little difference from using both the priorities. However, delay by connection in MAC becomes different, so entire transmission delay becomes different as shown in Figure 8.

As simply mentioned in the above, Figure 6 displays change in the throughput of external TCP according to the number of UDP connections. If priority in MAC is given(MAC, Both), throughput is shown to be significantly high compared to if it does not. This result can be drawn from the fact that priority in MAC is advantageous in competition with internal TCP packet; as UDP having advantageous conditions to internal TCP reduces the transmission rate of internal TCP, UDP and external TCP occupy the position. Figure 7 shows the throughput of internal TCP according to the number of UDP connections, implying that the number of UDP connections little influences the performance of internal TCP as long as MAC priority is not given to UDP. In this case, internal TCP has nothing to do with kind of packet held in the queue of home gateway, so this condition can be readily expected.

Figure 8 shows the changes in average transmission delay according to the number of UDP connections. The changes in transmission priority of UDP only influence wireless transmission section, so attention should be paid only to this part to analyze the result, when the then delay depends upon delay in the queue of home gateway. Average delay in queue is firstly dependent upon how many packets stand by at the front when the packet comes in the queue and secondly upon how fast the packets get out of the queue. As long as queuing priority is not used, external TCP and UDP are in the same position and when the packet comes in the queue, the average number of packets standing by at the front becomes same as average length of the queue. It is estimated that the average length of the queue will be kept at a constant level by the congestion control of TCP and the buffer control algorithm of RED. Therefore unless both queuing priority and MAC priority are used, the first and second conditions have no relation with the number of UDP packets, resulting in delay irrespective of the number of UDP connections as shown in the graph. If MAC priority is used(MAC, Both), UDP fast gets away, so the more is UDP connections, the less is transmission delay. If just queuing priority is used, the first condition is little related with TCP packet and only influenced by UDP packet; therefore the more is the number of UDP connections, the more is transmission delay sharply.

Figure 9 and Figure 10 show the average transmission delay and loss rate of UDP packet according to the number of TCP connections, respectively. The number of TCP connections displayed on the horizontal axis is each number of external TCP and internal TCP in the graph. The number of UDP connections is two, and each connection has 400kbps of packet rate. The proposed mode shows that even change in the number of TCP connections relatively little influences UDP loss rate or transmission delay. If both multimedia service and web service are simultaneously used, for example, connection to a page composed of many pictures, each picture is downloaded at the same time. At this time, since several TCP connections simultaneously occur, it is important to make UDP traffic little influenced by this.

Putting the aforesaid simulation results all together, an offer of QoS in network layer and MAC layer in a combined manner through control of Priority Queuing and CW size shows better performance than that just in MAC layer through control of CW size as existing mode in terms of the average transmission delay and loss rate of UDP packet as used by real time service class, enhances the throughput of external TCP connections, and displays far better performance in fairness between external TCP connection and internal TCP connection. It is deemed that this result mostly originates from settling HOL blocking problem through Priority Queuing which occurs in existing mode.

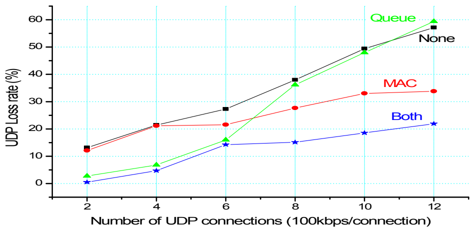

Fig.5 UDP packet loss rate by change in the number of UDP connections

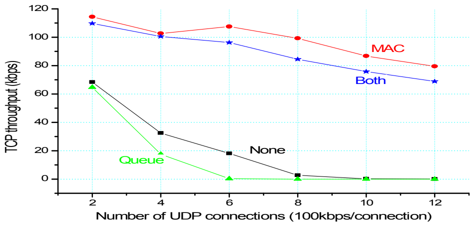

Fig.6 The throughput of external TCP according to the number of UDP connections

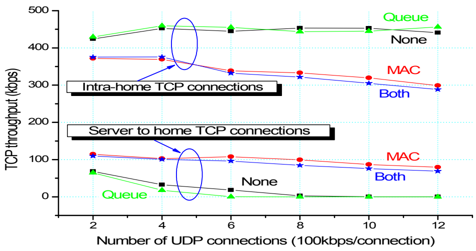

Fig.7 The throughput of internal TCP according to the number of UDP connections

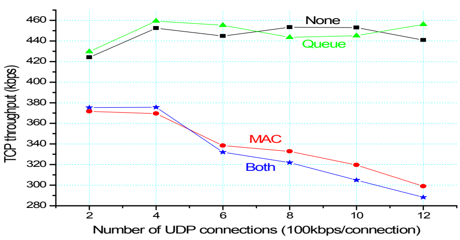

Fig.8 The changes in average transmission delay according to the number of UDP connections

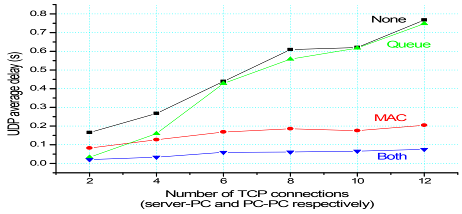

Fig.9 The average transmission delay of UDP packet according to the number of TCP connections

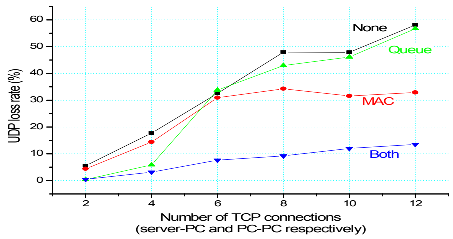

Fig.10 The loss rate of UDP packet according to the number of TCP connections.

IV. Conclusion

In this paper, In order to offer the real-time multimedia service in which the demand of users is growing rapidly in the wireless home network environment based on wireless-LAN, it proposed the QoS support architecture from the overall network which includes the wired-Internet and the wireless home network. The QoS support from the wired-Internet uses a DiffServ method basically and that it is assumed that it is possible. In this paper, unlike other research, we focus on QoS support with the wireless home network environment which is the part where it becomes problem. With researches of existing only MAC level of wireless LAN, it does not treat a problem differently and does not consider a network layer and a data link layer integrated in the method which problem it approaches, the QoS provision is possible in the minimum change of the existing wireless LAN terminal which will use this integrated QoS support method. A simulation is led and it is wireless LAN which is provable in a performance much more excellent than the existing methods of dealing with a QoS support problem. As for this kind of research result when considering the actuality that a capacity of Internet Backbone is much larger than traffic, it will be able to provide a real-time multimedia service to the home network users in the minimum change of the home gateway and the wireless LAN terminal in order, the place it is thought with the fact that it will be able to contribute a lot. Also, 4G mobile communication system based on all IP network focus element the wireless LAN which is the possibility of doing VoIP (Voice over IP) there is to implement and the application is possible with the reference data which is important.

References

[1] T,Nadagopal, S. Lu, and V. Bharghavan, *"A Unified Architecture for Design and Evaluation of Wireless Fair Queuing Algorithms,"* in Proceedings of ACM MOBICOM, Aug. 1999

[2] K.Kim, A. Ahmad, and K. Kim, *"A Wireless Multimedia LAN Architecture Using DCF with Shortened Contention Window for QoS Provisioning,"* IEEE Communication Letters, Vol. 7, No. 2, pp. 97 - 99, Feb. 2003

[3] A.Lin, and S. Lee, *"A Modified Distributed Coordination Function for Real-Time Traffic in IEEE 802.11 Wireless LAN,"* in Proceedings of the 17th Iternational Conference on Advanced Information Networking and Applications

[4] A.Veres, A. T. Campbell, M. Barry, and L. Sun, *"Supporting Service Differentiation in Wireless Packet Networks Using Distributed Control,"* IEEE JSAC, Vol 19., No. 10, pp. 2081 - 2093, Oct. 2001

[5] A.Banchs, and X. Perez, *"Providing Throughput Guarantees in IEEE 802. 11 Wireless LAN,"*, Wireless Communications and Networking Conference, 2002. WCNC2002. 2002 IEEE ,Volume: 1 pp 130-138, March 2002

[6] A,Ayyagagari, Y, Bernet, and T. Moore, *"IEEE 802.11 Quality of Service-White Paper,"* IEEE 802.11-00/028

[7] AImad and C. Castelluccia, *"Differentiation Mechanisms for IEEE 802.11,"* In Proceedings of INFOCOM, Apr. 2001

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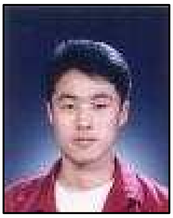

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