Adaptive Resource Allocation for Traffic Flow Control in Hybrid Networks

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Abstract

Wireless network systems provide fast data transmission rates and various services to users of mobile devices such as smartphones and smart pads. Because many people use high-performance mobile devices, the use of real-time multimedia services is increasing rapidly. However, the preoccupation of resources by real-time traffic users is causing harm to other services—for example, frequent call interference, lowered service quality, and poor network performance. This paper suggests a resource allocation algorithm for effective traffic service support in a hybrid network. The main objective is to obtain an optimum value of data rates by comparing user requirements with the amount of resources that can be allocated. A new mechanism based on Adaptive-Quality of Service (QoS) and a monitoring system based on Queue-Aware are proposed. Adaptive-QoS supports effective resource control according to the type of traffic service, and the monitoring system based on Queue-Aware measures the amount of resources in order to calculate the maximum that can be allocated. We apply our algorithm to a test system and use Qualnet 4.5.1 to evaluate its performance.

Keywords: Resource allocation, traffic flow control, vertical handover, soft QoS, hard QoS

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1. Introduction

Recently, wireless networks have provided seamless communication and various multimedia services. At the same time, requirements for new wireless sytems are multiplying to keep pace with technological developments. The fast development of hardware and communication technologies has brought about the diversification of wireless communication services. In particular, real-time multimedia services are experiencing rapid progress. However, as more and more people come to use real-time multimedia services, the network system has begun to encounter problems. For example, excessive use of real-time services has led to system overloads, frequent call interference, and lowered service quality. Hence, recent discussions have focused on reducing such overload and providing an effective allocation of network resources.

In order to provide smooth and fast services to users, resources have to be provided in a way that satisfies user requirements in accordance with traffic types. However, when resources are provided to meet all user requirements, those users without priorities may not receive an acceptable level of service, i.e., if a number of users of real-time traffic preoccupy the resources, other traffic users can be affected. It can be difficult to keep up with user requirements whilst allocating resources fairly. In addition, the maximum amount of resources available for actual service must be calculated on the basis of the terminal's capacity, and the switching of traffic to other networks should be considered in order to maximize system performance and reduce overload.

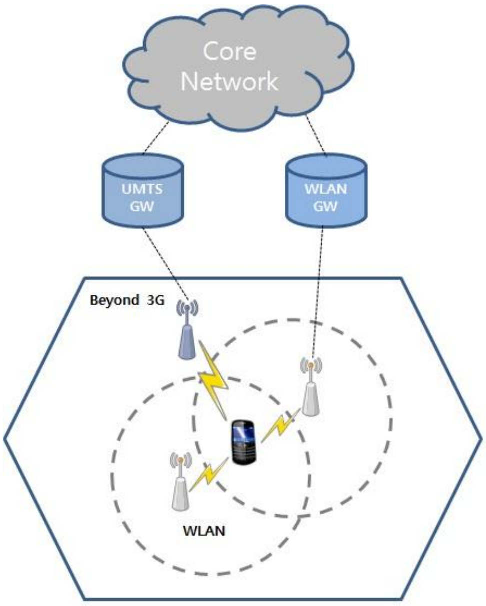


Fig. 1. Hybrid network architecture

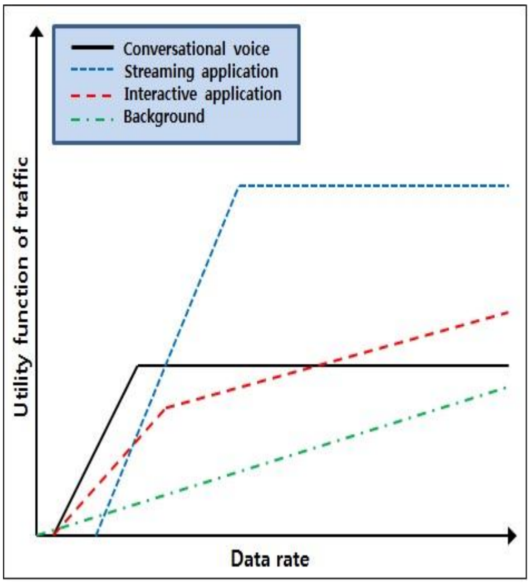


Fig. 2. Utility functions for traffic service

Fig. 1 shows a hybrid network structure. The terminal is placed in the area of overlap between a 3G network and a trusted WLAN. It is assumed that this terminal can move freely between 3G and the trusted WLAN. The relationship between the required transmission rate of various applications and its utility function can be seen in **Fig. 2** Each application program requires a different transmission rate, and the utility function value varies accordingly [1]. In the case of real-time applications, the utility function value changes dramatically as a greater transmission rate is allocated. As for non-real-time applications, the utility is seen to rise gradually. The most important goal of data-centered networks is to maximize the performance of the overall network system. Hence, methods to maximize the overall performance while meeting the requirements of each application should be sought. This is directly related to the problem of allocating optimal resources while maintaining the balance among users. In other words, there needs to be an effective distribution of resources between real-time and non-real-time traffic.

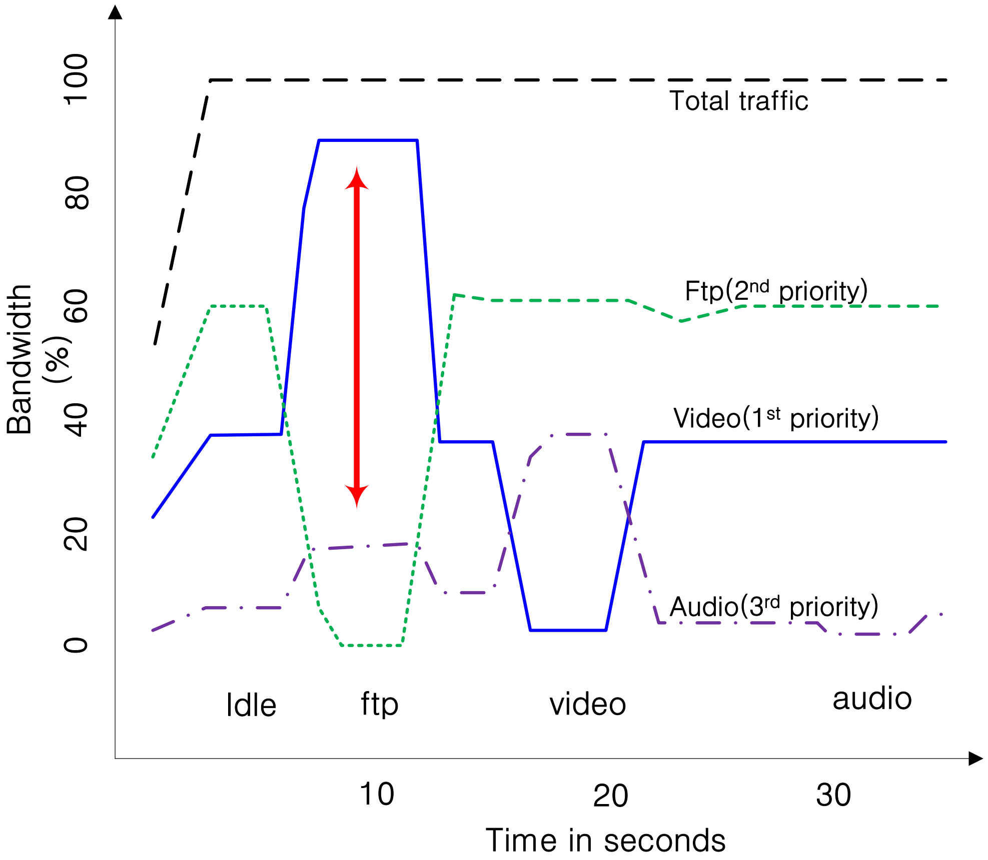


Fig. 3. The result of sharing bandwidth with priority

Fig. 3 shows that high priority traffic occupies resources and affects other, lower priority, traffic. In turn, it can be seen that traffic with the next highest priority occupies resources once the highest-priority traffic has finished transferring. We can see that an effective resource control system is needed to ensure the fair distribution of resources [2].

Hence, this paper suggests a resource distribution mechanism that can maximize the performance of the network system while meeting user requirements as much as possible. In particular, we suggest an Adaptive Resource Allocation Scheme that guarantees the minimum bandwidth of Hard-Quality of Service (QoS) and takes into account the flexibility of Soft-QoS.

This paper is structured as follows. Section 2 discusses some related work, and Section 3 explicates the algorithm suggested in this work. Section 4 contains an account of simulations to test our algorithm and a performance evaluation. Finally, we present our conclusions in Section 5.

2. Related Work

2.1 Hard-QoS

QoS can generally be divided into Hard-QoS and Soft-QoS. Hard-QoS refers to a guaranteed QoS that is only available when the specific amount of resource required for the traffic can be allocated. That is, the service is only provided when the minimum resource is available to support it. This can be explained by **Fig. 4** and **Fig. 5**.

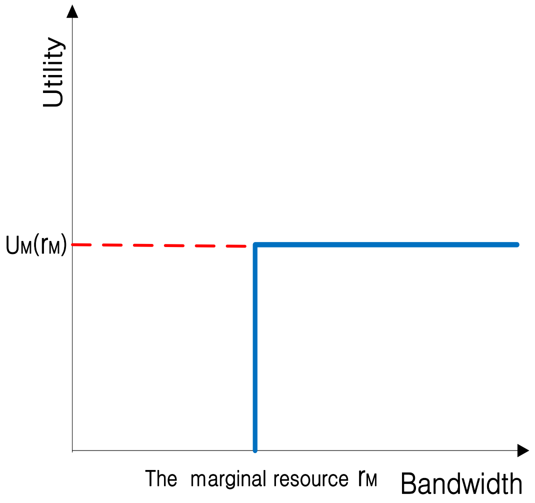


Fig. 4. The utility function for Hard-QoS

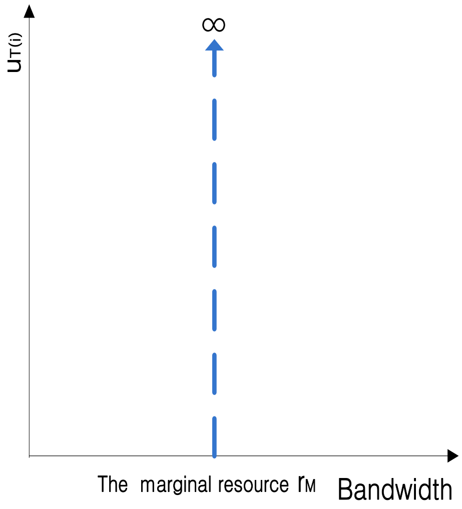


Fig. 5. The marginal utility function

Hard-QoS can be explained using the Unit-Step utility function. Hard-QoS traffic can only operate normally when it is allocated more than the marginal resource r_M. If the allocated resources are less than r_M, the utility function value cannot be obtained. In constrast, once the resource level reaches r_M, the utility function value can be obtained constantly. The utility function for user *i* can be defined as follows.

$$U_i(r_i) = U_{T(i)}(r_i \cdot q_i) \tag{1}$$

$U_{T(i)}(\cdot)$ denotes the utility function of user *i* according to different traffic types. r_i is the amount of resources to be allocated to user *i*, and q_i indicates the channel status of user *i*; this takes a value between 0 and 1. This utility function describes the Unit-Step utility function *U*step(r). The utility function for user *i* with Hard-QoS is described by:

$$U_{T(i)}(r) = U_{M_i} \times f_u(q_i r - r_{M_i}), \tag{2}$$

where $f_u(\cdot)$ is a unit-step function and M_i is the type of QoS traffic. By differentiating the utility function, we can show that its value with respect to resources is constant. The differentiated function is called the marginal utility function. By differentiating formula (2), we obtain the marginal utility function:

$$u_{T(i)} = \frac{dU_T(i)(r_i \cdot q_i)}{dr_i} = q_i \cdot u_{T(i)}(r_i \cdot q_i) \tag{3}$$

This formula can also be thought of as u_{step}(r), the derivative of *U*step(r). The Unit-Step utility function actually has the following conditions:

$$u_{step}(r) = \begin{cases} u_{step}(r_M) = \infty, r = r_M \\ u_{step}(r) = 0, otherwise \end{cases} \tag{4}$$

The value of the function u_{step}(r) is determined by whether the minimum amount r_M required by Hard-QoS traffic is obtained. If the allocated resource r exceeds r_M, the value of u_{step}(r) becomes infinite; otherwise, it is 0. Thus, Hard-QoS can be expressed using the above formulas.

2.2. Soft-QoS

Unlike Hard-QoS, Soft-QoS enables service when the minimum bandwidth for normal operation is obtained. The details of the resource requirement for Soft-QoS traffic can be explained using the sigmoid utility function.

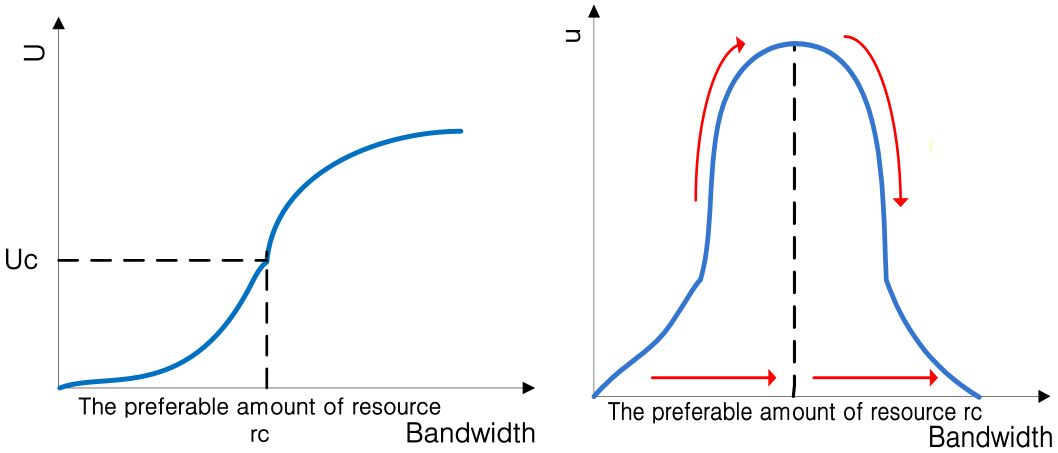


Fig. 6. The sigmoid utility function for soft QoS traffic

The basic formula for the sigmoid function is as follows:

$$y = \varsigma(x) = \frac{1}{1+e^{-x}} \tag{5}$$

The characteristics of the sigmoid utility function can be seen in **Fig. 6**, which shows *U*(r) > 0 and *u*(r) > 0 for all r. The resource level *rc* refers to the resource amount preferred by Soft-QoS traffic, and *Uc* signifies the utility value of the allocated resource *rc*. On the basis of the critical point *rc*, the graph is expressed in the form of an ‘S’. By differentiating the sigmoid utility function, we observe that the maximum utility value is attained at *rc*.

$$u'(r) = \begin{cases} u'(r) > 0, r < rc \\ u'(r) = 0, r = rc \\ u'(r) < 0, otherwise \end{cases} \tag{6}$$

It can be seen that as the resource amount r approaches *rc*, the marginal utility increases dramatically. The function then varies gradually and smoothly towards *rc*. However, when the allocated resource exceeds this critical point, the marginal utility begins to decrease, indicating that the allocation of further resources may not be helpful for operation. Soft-QoS has the merit of flexibility, as it provides service even when the optimum resource value is not available. It can flexibly control the resource amount preferred by the traffic.

2.3. Resource Allocation Studies

Many studies into Hard- and Soft-QoS have been carried out to solve the problem of resource distribution. In [3], a scheduling algorithm was suggested to allocate resources by distinguishing traffic sensitive to delay from the Best-Effort (BE) traffic. This balanced the distribution of bandwidth between a Delay-Sensitive (DS) flow and the BE flow. In [4], a structure that can dynamically schedule the channel rate of the physical layer of Wideband Code Division Multiple Access(WCDMA) was suggested for detecting the change in the transmission rate of the traffic. This structure used the capability of the WCDMA physical layer to reduce the calculation complexity of the Link Layer. Such a scheme solves the problem of flow allocation in multi-hop wireless networks, maximizing the overall network performance by forcing the cooperation of the physical layer, MAC, and the transport layer [5].

A scheduling algorithm that considers the channel and queue status was suggested by [6]. In this architecture, the transmission rate is computed by quantifying the data accumulated in the queue and the amount transferred, thus enabling the calculation of the resources that can actually be allocated. The research proposed a solution to the resource distribution problem using Hard-QoS—resources are distributed to real-time traffic in order of priority by distinguishing real-time traffic from non-real-time traffic. The remaining resources are then allocated to non-real-time traffic [7]. In contrast, [8] suggests a Soft-QoS-based resource allocation system. Their algorithm enables as much service as possible and maximizes network performance by allocating the minimum resource necessary for actual service.

Although various studies have been carried out, we are yet to find a method that fully satisfies user requirements while distributing resources effectively. In particular, the problem of certain traffic occupying most of the resources not only affects the system performance, but also causes low service quality for other users. Therefore, in the next section, we propose a resource distribution mechanism that can maximize the performance of the network system while still meeting the user requirements as much as possible.

3. Proposed Method

As noted in the previous section, the preoccupation of resources by high-priority real-time traffic causes poor system performance. Hence, we propose a new scheduling method for effective resource distribution among all network traffic. We propose an Adaptive-QoS scheduling method that combines the merits of Hard-QoS and Soft-QoS, Queue-Aware based

Monitoring scheme, Adaptive-QoS allocation scheme, and Resource Control scheme. The Queue-Aware-based Monitoring scheme determines the maximum amount of resources that user can be assigned to each user during specific timeslot. By comparing the maximum value and the minimum requirement of service, the Resource Control scheme adjusts the amount of resource that will be assigned. After the processes for calculating the resource of users, the Adaptive-QoS allocation Scheme allocates the optimized amount of resource to users by priority. The system model and the basic algorithm for overall resource distribution scheduling are explained in **Fig. 7**, **Fig. 8**, and **Fig. 9**.

3.1. System Model

The marginal utility function is considered to secure the minimum amount of resource for traffic service. Suppose that there are n users servced by a base station. Let r_i denote the amount of resource to be allocated to user i and r_{total} denote the total amount of radio resource available at the base station. Main objective is to get optimal utility value for the allocated resource and to maximize $\sum_{i=1}^{n} U_i(r_i)$, subject to $\sum_{i=1}^{n} r_i \le r_{total}$ and $\forall r_i \ge 0$.

The basic definitions for the utility function are as follows:

Definition 3.1: A resource allocation $\Re^* = \{r_1, r_2, \cdots, r_n\}$ for n users is an optimal allocation if for all feasible allocations $\Re_a = \{r_1', r_2', \cdots, r_n'\}$, $U(\Re^*) \ge U(\Re_a)$, where $U(\Re^*) = \sum_{i=1}^{n} U_i(r_i)$ and $U(\Re_a) = \sum_{j=1}^{n} U_j(r_j')$.

Definition 3.2: An adaptive-QoS utility function $U_{Adpt}(r)$ refers to a utility function whose $u_{Adpt}(r_M) = \infty$ if $r = r_M$ and $\alpha = 0$, $u_{Adpt}(r_M) \prec u_{Adpt}(r) \le u_{Adpt}(r_{\max})$ if $r_M \le r \le r_{\max}$ and $\alpha = 1$, and $u_{Adpt}(r) = 0$, otherwise, where $u_{Adpt}(r) = \dfrac{dU_{Adpt}(r)}{dr}$ and α is a control variable for Soft-QoS and Hard-QoS.

Definition 3.3: $\Re = \{r_1, r_2, \cdots, r_n\}$ is a full allocation if $\sum_{\forall r_i \in \Re}^{n} r_i = r_{total}$.

By definition, an adaptive-QoS utility function is a discrete function or sigmoid function according to an amount of resource r. Fig.8 plots the marginal utility function for the adaptive-QoS utility function shown in Fig.7. The basic formula for the utility function is as follows:

$$U_{Adpt}(r_i) = \alpha\left(\frac{1}{1+e^{-\beta(x-r_M)}}\right) + U_{T(i)}(x) \tag{7}$$

,where $U_{T(i)}(\cdot)$ is the unit-step function in Hard-QoS and β is the rate of increase. By the change of β, the slope of graph is changed. By combining Hard-QoS with Soft-QoS, the value of the utility function is estimated according to the allocated resource.

Depending on r_i for the allocated resource, the equation (7) can be calculated as two types of Hard-QoS and Soft-QoS differently.

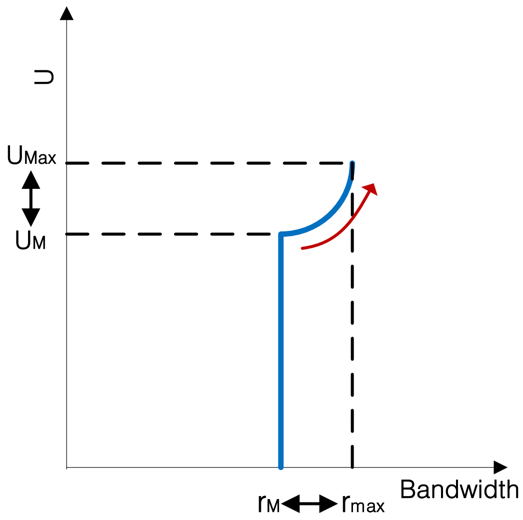


Fig. 7. The utility function for the proposed scheme

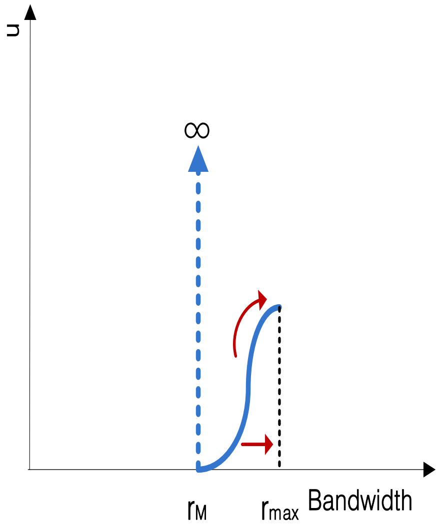


Fig. 8. The marginal utility function

In case of satisfying the minimum amount of resource r_M, the utility function is calculated as

$$U_{Adpt}(r_i) = U_{T(i)}(r_i) = U_{M_i} \times f_u(q_i r_i - r_{M_i}), \tag{8}$$

where $f_u(\cdot)$ is a unit-step function and M_i is the type of QoS traffic.

From (8), the equation has a constant utility function value as for Hard-QoS if $(q_i r_i - r_{M_i}) \geq 0$.

By differentiating equation (7), this can be found that service is only possible when the minimum amount of resource r_M is obtained. Consequently, U_{M_i} is utiliy function value of the minimum amount of resource. This guarantees the minimum quality of service, and provides an opportunity to move to another network if it is not satisfactory.

Once r_M is obtained, the amount of resource can be allocated flexibly up to the maximum obtainable value of r_{max}. Depending on the amount of resource r, $U_{Adpt}(r_i)$ changes smoothly.

The allocated resource can be expanded up to r_{max} depending on the network state. The equation is as follows:

$$U_{Adpt}(r_i) = \left(\frac{1}{1 + e^{-\beta(r_i - r_M)}} \right) + U_{M_i} \times f_u(q_i r_i - r_{M_i}) \tag{9}$$

If resource r_i satisfies the condition $r_M \prec r_i \leq r_{\max}$, , the flexible utility function is obtained from equation (9). Because the value of $U_{M_i} \times f_u(q_i r_i - r_{M_i})$ is a constant, $U_{Adpt}(r_i)$ increases smoothly according to increasing the value of r_i. The rate of increase in utility in **Fig. 7** is determined by the rate of change of β. If the amount of resource r_i increases, **Fig. 8** shows that the maximum range for increase is determined by r_{max}.

3.2. Queue-Aware-based Monitoring Scheme

This scheme calculates the maximum amount of resource that can be allocated to a certain service flow. It presupposes the following: It assumes that the network is OFDM-based. Because of sharing channels on OFDM, total bandwidth B is divided by k subcarriers, $k = \{1, 2, ..., k\}$. In addition, each subcarrier bandwidth is $\Delta f = B / K$ and each time slot is T_s. It also assumes that the achievable data transmission rate per Hz for user i on subcarrier k during time slot n, *Ci*[k, n], can be known at the base station by estimating the channel state information via pilot signals and feeding the information back to the base station. The *Ci*[k, n] are decided by the current channel signal-to-noise ratio (SNR) and the required bit-error-rate(BER). The related formula is as follows:

$$C_i[k,n] = \log_2(1 + \beta_{\rho_i}[k,n]) \tag{10}$$

,where $\beta = -1.5/\ln(5BER)$.

From (10), the transmission rate $r_i[n]$ for user *i* in *timeslot n* is calculated by

$$r_i[n] = \sum_{k \in D_i^{(n)}} C_i[k,n]\Delta f \tag{11}$$

where $D_i^{(n)}$ are the set of subcarrier indices.

In order to calculate available queue length during next timeslot, this scheme uses average arrival rate, current queue length and the transmission rate. First, the average arrival rate is calculated by

$$\lambda_i = \frac{1}{T_s} \lim_{n \to \infty} \frac{A_i[n]}{n} \tag{12}$$

where $A_i[n]$ is the total amount of data received during (0, nT_s].

But we use a Exponential Moving Average to find the queue length of user *i* within the specific period. With the characteristics of a wireless environment in mind, this is set to find the average over a particular period of time, rather than the average over the whole time period. This is to give more credibility to the data. The Exponential Moving Average is calculated as follows:

$$S_t = \omega \times Y_{t-1} + (1-\omega) \times S_{t-1} \tag{13}$$

where S_t is estimated value and Y_t is observed value.

S_{t-1} is replaced with the accumulated value for the previous time.

$$S_t = \omega \times Y_{t-1} + (1-\omega)\{\omega Y_{t-2} + (1-\omega) S_{t-2}\} \tag{14}$$

Finally, the final weighted sum of above equation can be found.

$$S_t = \omega \sum_{i=1}^{t-1} (1-\omega)^{i-1} Y_{t-i} + (1-\omega)^{t-1} S_1 \tag{15}$$

By using the above formula, $A_i[n]$ can be found during the specific period. The equation is as follows:

$$\lambda_t = \omega \sum_{i=1}^{t-1} (1-\omega)^{i-1} A_{t-i} + (1-\omega)^{t-1} \lambda_1 \tag{16}$$

In the same way, the data in the queue $Q_i[n]$ can be calculated by

$$S_t = \omega \sum_{i=1}^{t-1} (1-\omega)^{i-1} Q_{t-i} + (1-\omega)^{t-1} S_1 \tag{17}$$

$$Q_i = \lim_{N->\infty} \frac{\sum_{n=0}^{N-1} Q_i[n]}{N} \tag{18}$$

By taking into account the amount of data $r_i[n]$ transferred during *timeslot n*, the amount of data received $A_i[n]$, and the data in the queue $Q_i[n]$, the amount of resource that can be allocated in the following timeslot can be computed.

Hence, finally, the formula calculating the amount of resource that can be allocated during a timeslot is as follows:

$$Q_i[n+1] = Q_i[n] - r_i[n]T_s + A_i[n] \tag{19}$$

The above formula gives the queue length $Q_i[n+1]$ of user *i* at time $(n+1)T_s$.

3.3. Adaptive-QoS Allocation Scheme

Resources must be allocated in order of priority to allow reasonable resource distribution to users. First of all, we assume the data of user *n* is saved in the queue. This scheme considers the utility function value for the allocated amount of resource as standard point. Let $r_{residue}$ denote the remainder of the system resource, r_{total} denote the total amount of resource, and r_{opt} denote the optimized and preferred amount of resource. The resource allocation algorithm is as follows:

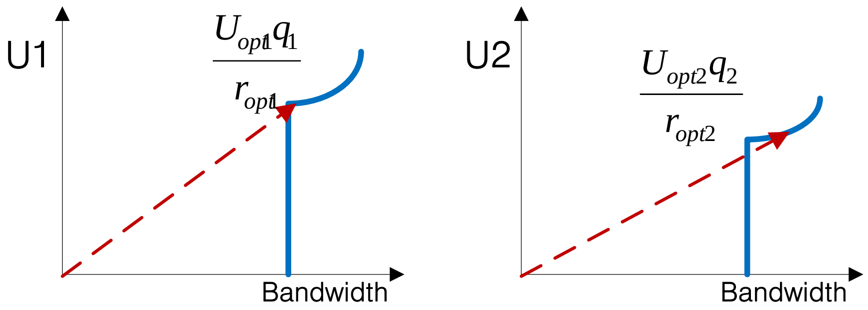
1) $r_{residue} \leftarrow r_{total}$. $r_i \leftarrow 0, i = 1,2,\cdots,n$

2) Sort users according to priority of QoS of 3GPP for each traffic group.

3) After ordering each group, place in descending order using $\frac{U_{opt_i} q_i}{r_{opt_i}}$, which takes into account the utility function value for each bandwidth.

4) Repeat 5).

5) Start resource allocation. If $r_{residue} > \frac{r_{opt_i}}{q_i}$, $r_i = \frac{r_{opt_i}}{q_i}, r_{residue} = r_{residue} - r_i$



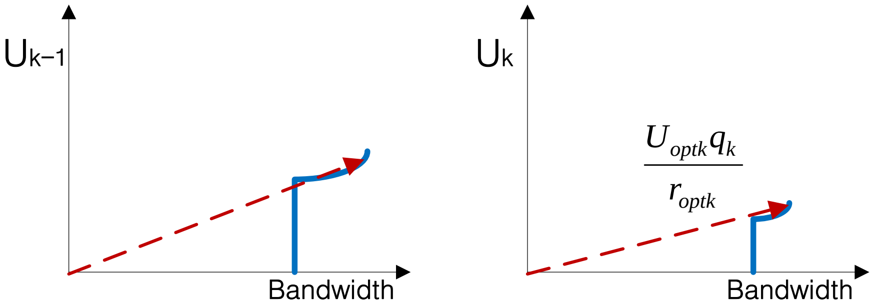


Fig. 9. Allocation ordering of k users in the Adaptive-QoS allocation

3.4. Resource Control Scheme

The network system is the most important element for resource distribution and traffic service. In order to provide the minimum bandwidth to support traffic service, we consider the marginal utility value r_M. The minimum resource required for each type of traffic is given in **Table 1**[11]. The proposed scheduling algorithm analyzes the capacity of the network and also considers the processing power of user devices. Hence, it aims to provide the resource that can actually be allocated. This algorithm is illustrated in **Fig. 10**.

Table 1. Preferred bandwith of traffic services in 3GPP

Medium	Type of Services	Application	Degree of symmetry	Data rate	Delay (ms)
Audio	Conversational service	Conversational voice	Two-way	4–25 kb/s	<150 preferred <400 limit
Audio	Interative service	Voice messaging	Primarily one-way	4-13 kb/s	< 1 sec for playback < 2 sec for record
Video	Streaming service	Movie clips, surveillance, real-time video	Primarily one-way	32–384 kb/s	<150 preferred <400 limit Lip-synch : <100
Data	Conversational service	Telemetry	Two-way	<28.8 kb/s	<250
Data	Conversational service	Realtime games	Two-way	<60 kb/s	<75 preferred
Data	Conversational service	Telnet	Two-way	<1 KB	<250

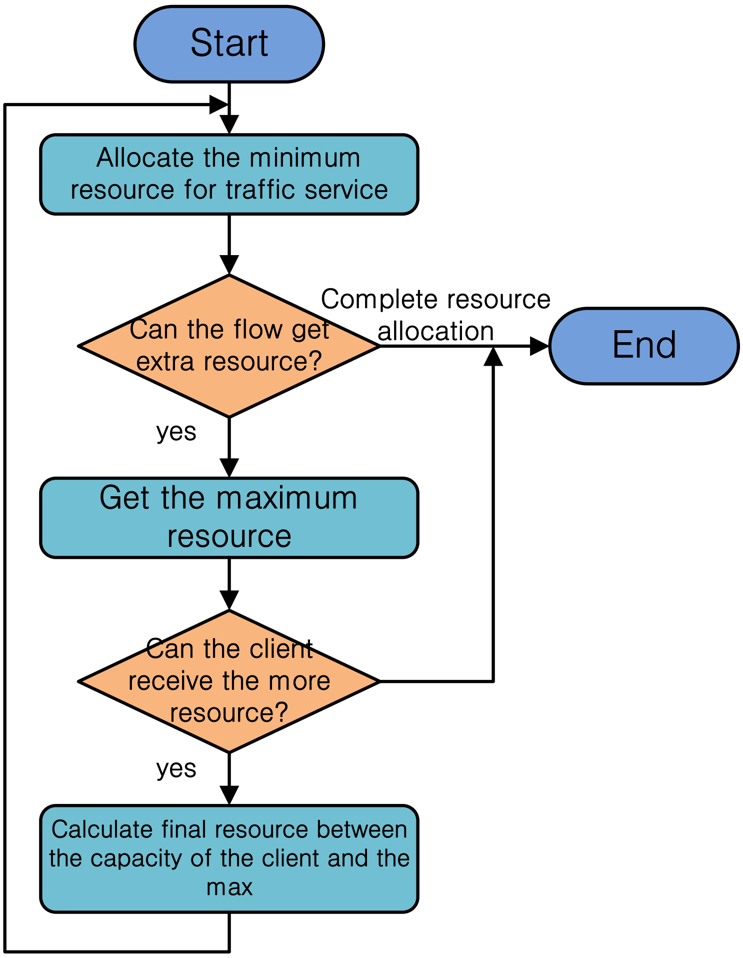


Fig. 10. Operations during resource controll phase

Ci is the amount of queued data that can be stored in a device, and this must have a greater value than the minimum r_M. Thus, service becomes impossible when $Ci < r_M$, and the resource allocation step is postponed for some period of time. If service is not possible after three attempts, the applicable service is stopped. After this, the maximum resource amount is computed using a Queue-Aware-based monitoring scheme. At this point, the current resource amount cannot exceed the maximum resource. The final resource amount that can actually be used is calculated as the difference between the device capacity and the maximum resource.

4. Performance Evaluation

This section presents the results of a simulation using the proposed algorithm. Qualnet 4.5.1 is used as the simulator. The simulation scenario and system parameters are given in **Fig. 11** and **Table 2**. The available bandwidth of the access point (AP) is limited to 100 kbytes in order to limit the communication between a mobile node and the applicable packet. The required amount of data for each packet relies on the recommendations of the 3GPP standard.

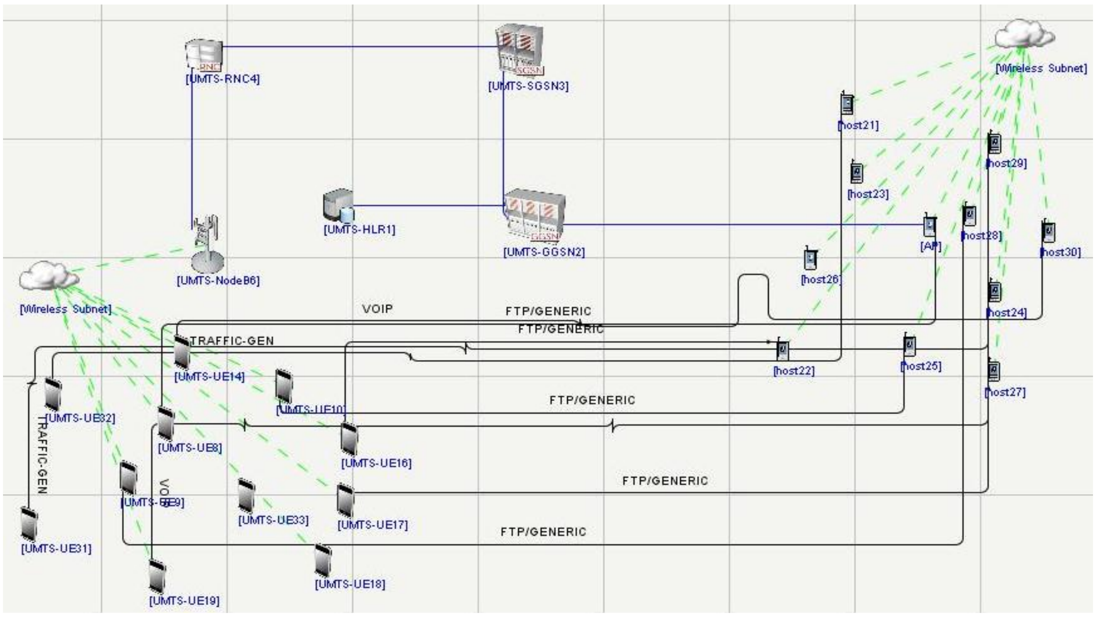


Fig. 11. Simulation scenario

Table 2. Simulation Parameters

Layer	Parameter	3GPP	WLAN
Physical	Channel frequency	2.4 GHz	2.8 GHz
	Radio Type	UMTS Layer1	802.11
	Noise Factor	10	
MAC	Protocol	UMTS Layer2	802.11
	Priority Input Queue Size	5000	
Network	Protocol	UMTS Layer3	IPv4
	Scheduler	Class based queue / -	
Traffic	FTP	Item to send	1000
		Item Size	512
	VoIP	Average Talking Time	20 s
		CODEC	G.711
	Traffic-Gen	Application	Video stream
		Traffic Type	Random

Fig. 12 shows the result of performance evaluation among the Hard-QoS, the Soft-QoS, and the Adaptive-QoS. We consider Hard-QoS as supporting a minimum of resources for services. We also chose Modified Largest Weighted Delay First (LWDF) of the Soft-QoS. Finally, by adjusting the number of real-time traffic, we simulated the Adaptive-QoS. According to increasing the users, we can see that the system throughput is improved gradually. Then the system throughput is maintained at a constant value when the number of users is more than a certain number. **Fig. 12** shows that the Adaptive-QoS performs good evaluation with increase

of users. We can assure that the performance of the Adaptive-QoS is improved according to increasing a number of real-time traffic users. The following result of the simulation is performance comparison for traffic service/The simulation limits are set to be identical to a mobile communication environment without QoS applied, and the following simulation results show the overall packet transfer volume and average transmission rate, as well as the transmission rate for each period of time. Each graph shows results centered around downloadable data, and the AP shows two kinds of data volume as upload and download are performed simultaneously.

In **Fig. 13 (a)**, we can see that the traffic volume is far below the recommended rate, due not only to video streams and VoIP in the fixed bandwidth, but also to data packet sharing. In particular, when video 8 connects with the AP after 20 s, the share of video 18 is seen to drop, whilst the share of FTP service does not differ greatly. The simulation result shows that it is difficult to expect enhanced traffic performance when real-time traffic increases as this occupies resources. Also, in **Fig. 14 (a)**, it can be seen that the recommended level is not met even by the real-time traffic, which indicates that satisfactory service is not being provided to users of real-time services.

When our proposed algorithm is applied to the same environment as the previous simulation, the improvement in results is obvious. In **Fig. 13 (b)**, it can be seen that there is a clear difference between the share of real-time traffic and general data packets. In **Fig. 14 (b)**, the real-time transfer volume meets the minimum amount required by the recommendation. As for FTP, because real-time service is not required, it takes a lower share compared to the real-time traffic. In contrast, real-time packets such as Video Streams and VoIP are seen to have their transfer rate secured within the boundaries of the required data volume. In particular, it can be observed that resource is allocated within these boundaries, and so does not affect the service of the existing real-time traffic, even when video 8 takes a share of bandwidth at around 20 s, thus satisfying the required data amount. Therefore, our results show that resources are effectively allocated to traffic users according to priority without harming other service users.

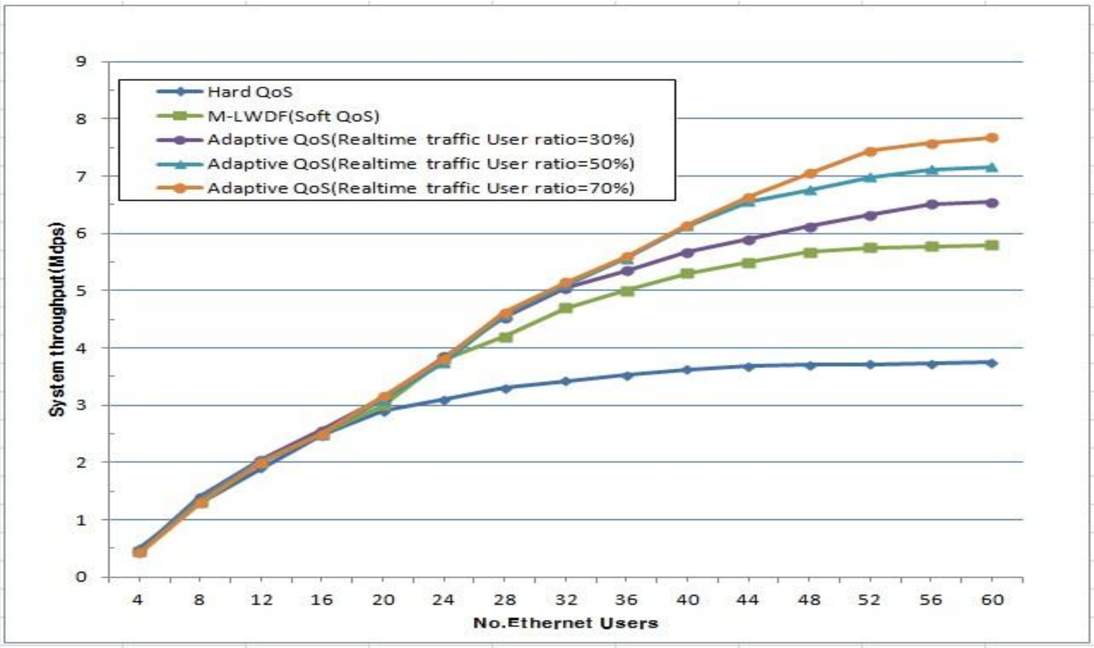
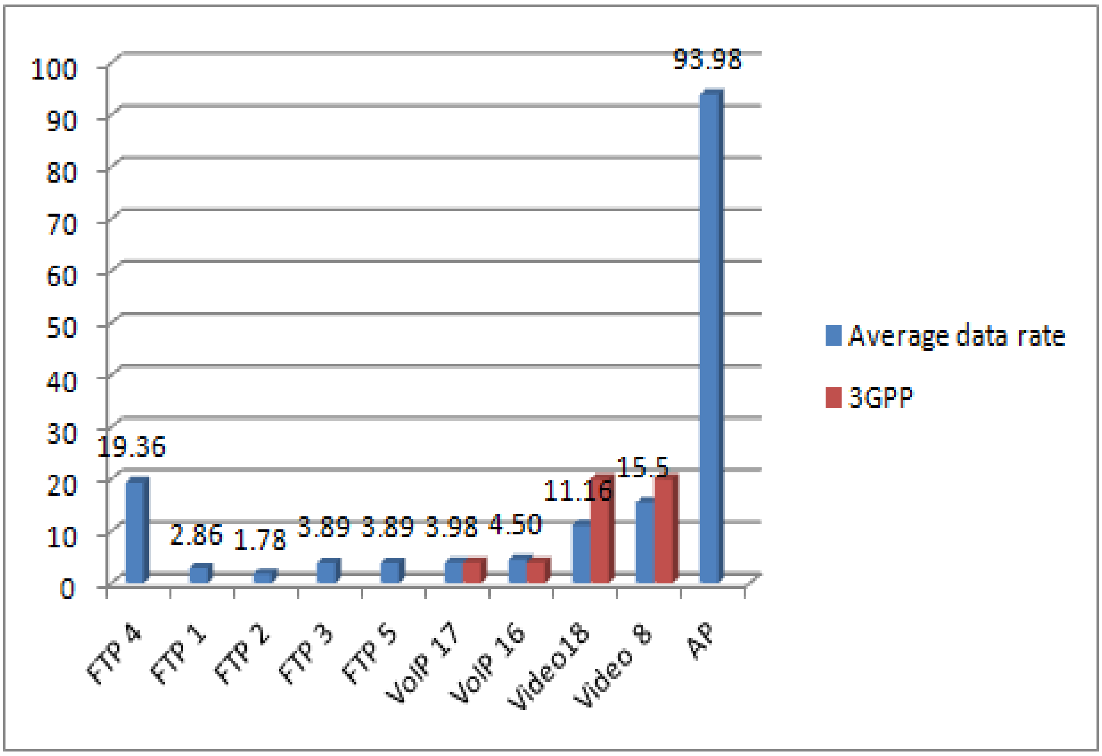
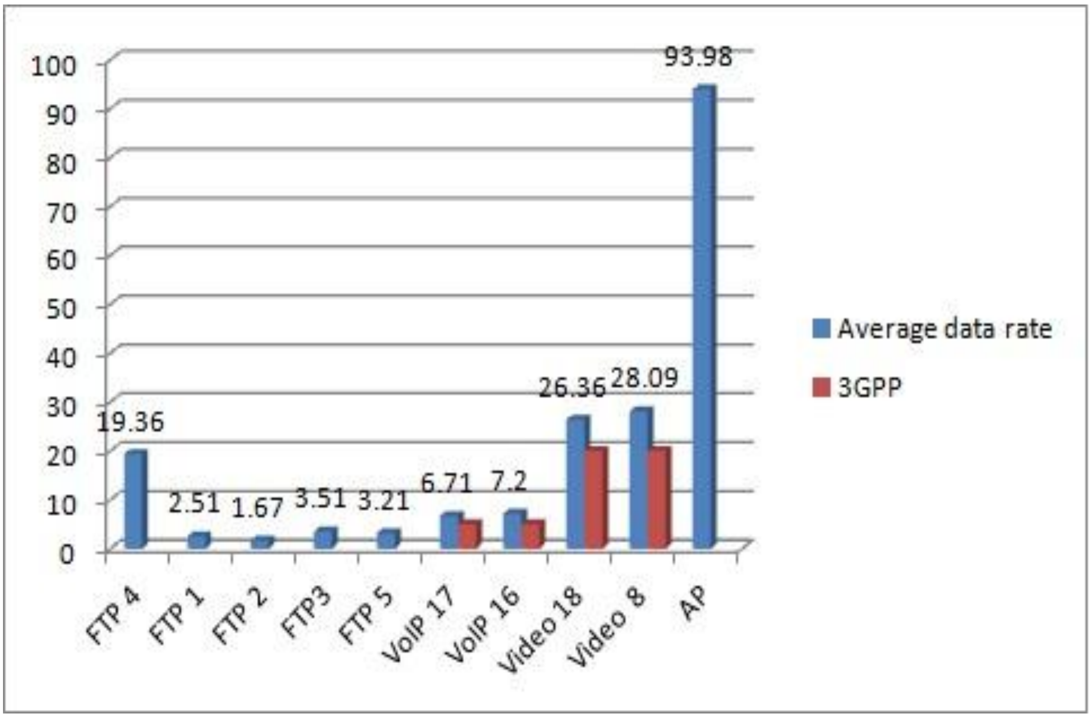


Fig. 12. System throughput

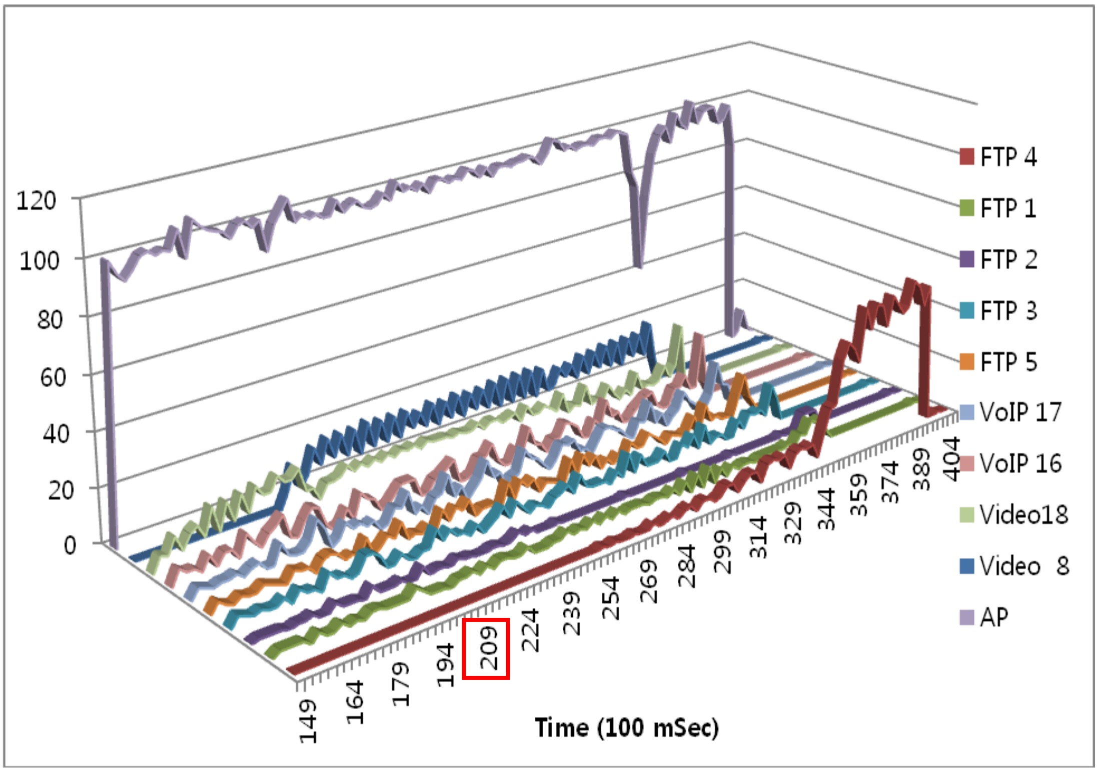


(a) Original

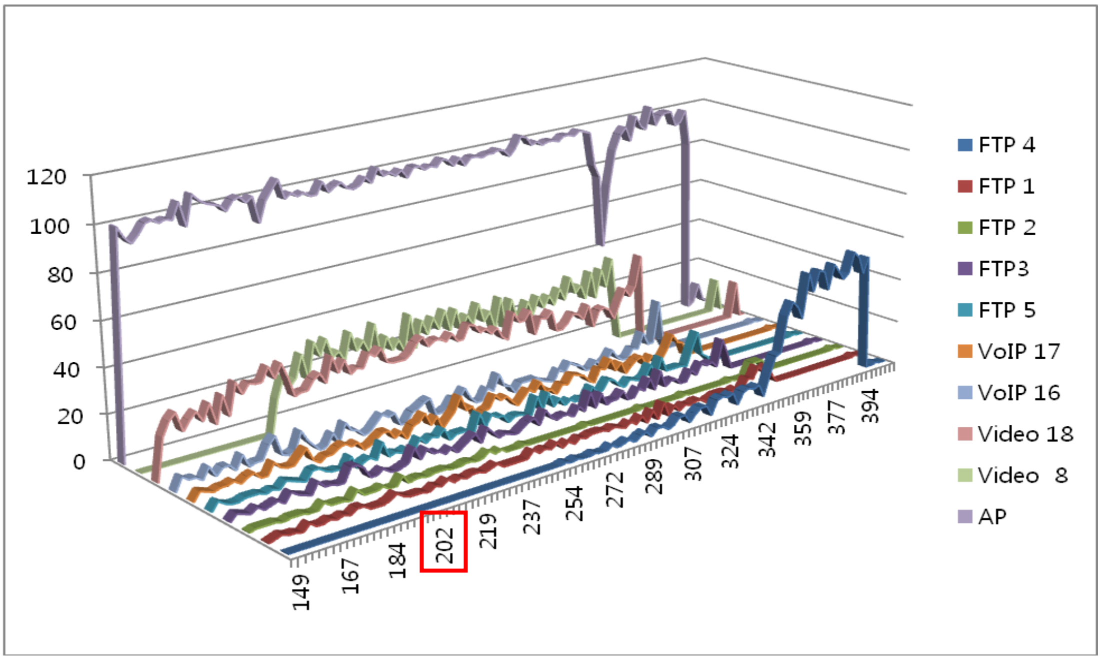


(b) Proposed

Fig. 13. Average data rate



(a) Original



(b) Proposed

Fig. 14. Throughput

5. Conclusion

In this paper, we analyzed the characteristics of real-time and non-real-time traffic with respect to the performance degradation suffered by certain traffic services when resources are occupied by high-priority users. We have proposed the Adaptive Resource Allocation algorithm to assign resources fluidly while simultaneously satisfying the minimum traffic requirements. Our algorithm has been designed to allocate the optimized resource by calculating the minimum necessary for service and the maximum that can be allocated. We applied the advantages of Hard-QoS, in order to protect other traffic users and guarantee the minimum requirement of traffic service, and Soft-QoS to dynamically allocate resources.

The performance of our algorithm was evaluated using a simulation model of the throughput and average data rate. It was demonstrated that the proposed algorithm achieves better throughput than the original algorithm. Each priority level of traffic service was achieved with an improvement in throughput without causing performance degradation to other service users.

In future work, we will extend the provision of service to users who are not satisfied with the level of resources by switching them to another network for a smoother service.

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