Chapter 1
Introduction

1.1 Introduction

Recently, the demand to efficiently interconnect devices in small environments or Personal Area Networks (PAN) leads to a new and specialized wireless standard. Bluetooth is the first WPAN technology. However, due to its low transmission rate of less than 1Mbps, the capability to support burst data or real-time multimedia traffic is limited. The requirements of high quality video/audio applications and huge data transmission have pushed the wireless technologies to support higher data rates, such as 802.15.3.

Wireless Personal Area Networks (WPAN) [1-4] studied by the IEEE 802.15 Working Group enables short range wireless communications, and is designed for high-speed, low-power, low-cost multimedia-capable portable consumer electronic devices. This standard provides extremely high data rates within a small coverage range and mechanisms to provide QoS controls. In addition, this standard is designed to provide simple ad-hoc connectivity that allows the devices automatically form a communication network and exchange information between each other.

According to the arising transmission rates, the growing demand for multimedia applications have brought new challenges in wireless systems. Because of the real-time property of multimedia communication, all real-time applications should be supported at a certain rate constantly in order to meet quality of service (QoS) constraints. Moreover, there are various traffics arrived at the same time. Thus, MAC layer which is in charge of resource allocation would becomes critical in communication systems. A considerable number of studies have been conducted on MAC scheduling algorithms. However, traditional methods such as fixed channel resource allocation, first in first out (FIFO), weighted fair queue (WFQ) [9-10], earliest deadline first (EDF) [11-13], and etc, might not be sufficient for high quality real-time service QoS requirements due to various traffic characteristics and the imprecisely priority setting for different services. The objective of this thesis is to provide a method to solve mixed traffic QoS issue with modified MAC protocols. Simulation results show that the proposed algorithm can not only guarantee a low error rate for high priority traffic, but also work better than other algorithms on low priority traffic.

1.2 Thesis Organization

The remaining sections are organized as follows: Chapter 2 is the Overview of IEEE 802.15.3 MAC protocol and the introduction of common traffic. In chapter 3, detailed description of both existing scheduling methods and QoS problems of transmitting real-time traffic are presented. Chapter 4 introduces our proposed MAC scheduling algorithm and modified MAC protocol. The simulation environment and results are shown in chapter 5. Finally, the conclusion and future work are included in chapter 6.

Chapter 2
Overview of IEEE 802.15.3 MAC Protocol and Traffic Specified by WINNER

IEEE Std 802.15.3 (WPAN) was designed to enable wireless connectivity of high-speed, low-power, low-cost, multimedia-capable portable consumer electronic devices. This standard provides data rates from 11 to 55 Mb/s at distances within 10m while maintaining quality of service (QoS) for the data streams. In addition, this standard is designed to provide simple, ad-hoc connectivity that allows the devices to automatically form networks and exchange information without the direct intervention of the user.

Because of high data rate provided by 802.15.3, it is possible that multimedia applications and non-real time traffic are served at the same time. In order to satisfy this traffic's requirement, we should realize the characteristics of them first. WINNER (Wireless World Initiative New Radio) is a global organization whose main objective is to develop a single new ubiquitous radio access system concept whose parameters can be scaled or adapted to a comprehensive range of mobile communication environments from short range to wide area, thus WINNER identified lots of traffic models and quantified those traffic's requirements.

In this chapter, as a beginning we will introduce the 802.15.3 MAC architecture and channel time management mechanism in standard, then we will describe the traffic types and their characteristics specified by WINNER.

2.1 The 802.15.3 Piconet Architecture

802.15.3 is based on a centralized and connection-oriented ad-hoc networking topology. This wireless ad-hoc data communications system which allows a number of independent data devices (DEVs) to communicate with each other is called piconet. A piconet is distinguished from other types of data networks because communications are normally confined to a small area around person or object that typically covers at least 10m in all directions and envelops the person or a thing whether stationary or in motion. This is in contrast to local area network (LAN), metropolitan area network (MAN), and wide area network (WAN), each of which covers a successively larger geographic area, such as a single building or a campus or that would interconnect facilities in different parts of a country or of the world.

An 802.15.3 piconet consists of several components, as shown in Figure 2-1. The basic component is the DEV. Any DEV can be a PNC (piconet coordinator) which always manages the quality of service requirements, power save modes, access control, piconet organization and channel time allocation etc.

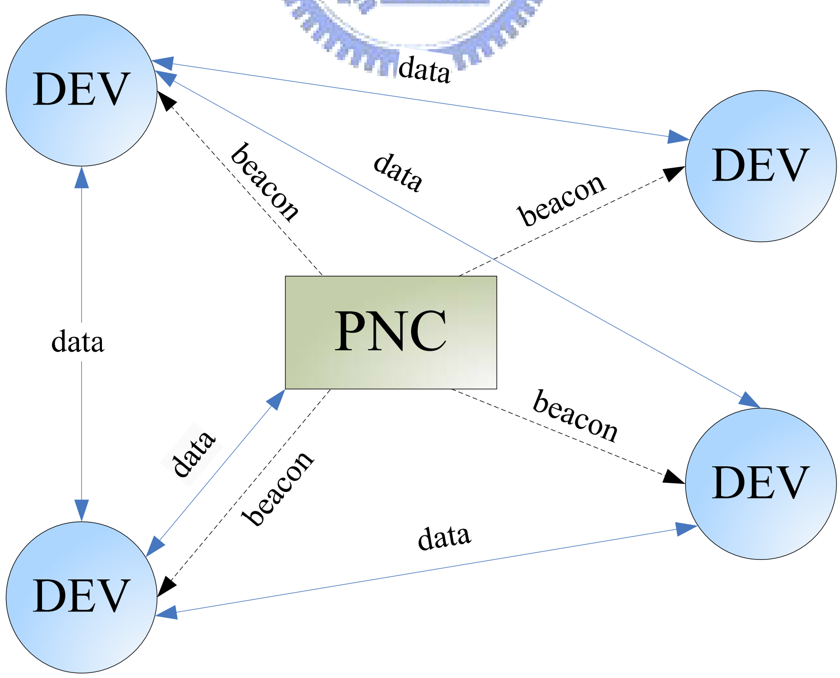

Figure 2-1 802.15.3 piconet elements

The 802.15.3 standard allows a DEV to request the formation of a subsidiary piconet. The original piconet is referred to as the parent piconet. The subsidiary piconet is referred to as either a child or neighbor piconet, depending on the method the DEV used to associate with the parent PNC. Child and neighbor piconets are also referred to as dependent piconets since they rely on the parent PNC to allocate channel time for the operation of the dependent piconet. An independent piconet is a piconet that does not have any dependent piconets.

2.2 The 802.15.3 Superframe Structure

Timing in the 802.15.3 piconet is based on the superframe. To handle large multimedia applications in the near future, a TDMA-based superframe structure is adopted in the IEEE 802.15.3 standard, which is illustrated in Fig. 2-2.

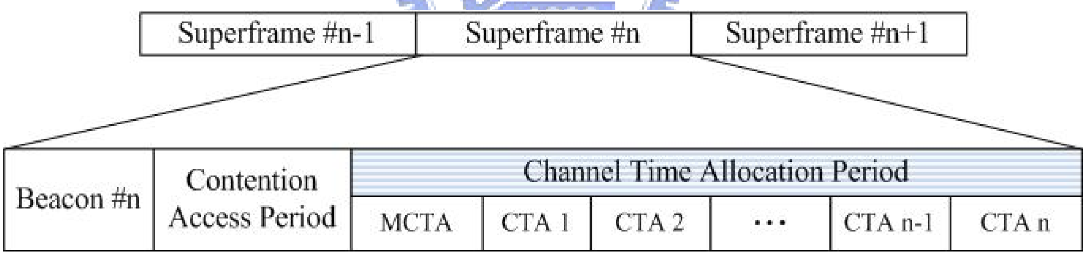

Figure 2-2 A superframe structure in IEEE 802.15.3 MAC

The superframe is composed of three major parts:

- **The beacon:** The beacon frame is transmitted by the PNC at the beginning of each superframe, which is used to provide all DEVs in the same piconet timing allocations and specific management information.
- **The contention access period (CAP)**: This period is used to transmit short, non-QoS data (asynchronous data) frames and command frames. The length of

the CAP is determined by the PNC and communicated to the DEVs in the piconet via the beacon. The basic medium access mechanism during the CAP is carrier sense multiple access with collision avoidance (CSMA/CA). To minimize collisions, a transmitting DEV is required to first sense whether the medium is idle for a random length of time, called "backoff interframe space" (BIFS). Only if the medium is idle after BIFS, the DEV starts its transmission. This process of waiting before transmission is termed "backoff." The backoff count is randomly selected from range [0,BW], where BW means backoff window chosen from the value set of [70µs, 150µs, 310µs, 630µs]. For the first transmission attempt of a frame, the BW value is set to the minimum number 7. If collision occurs, the BW value should be increased to the next larger value until reaching the maximum value 63. The DEV shall maintain a counter for backoff count which is decremented only when the medium is idle. Whenever the channel is busy, the backoff counter shall be suspended. The channel shall be determined to be idle for the duration of a BIFS period before the backoff slot countdown is resumed. When the backoff counter reaches zero, the DEV may transmit a frame.

- **The channel time allocation period (CTAP):** which is TDMA-based and contention free. The CTAP is composed of channel time allocations (CTAs) and management CTAs (MCTAs), which are both assigned by PNC through a beacon frame. A DEV with assigned directed CTA is guaranteed that no other DEVs will compete for the channel during the indicated time duration of the CTA. During one CTA period, only one DEV can transmit commands, isochronous streams and asynchronous data to one target DEV without collision. MCTAs is used for exchange specific information between the DEVs and the PNC , which are a type of CTA.

2.3 *The 802.15.3 Channel Time Management*

All data in the 802.15.3 piconet is exchanged in a peer-to-peer manner. In this section, we will introduce two major types of channel time management - isochronous stream management and asynchronous channel time reservation.

2.3.1 Isochronous stream management

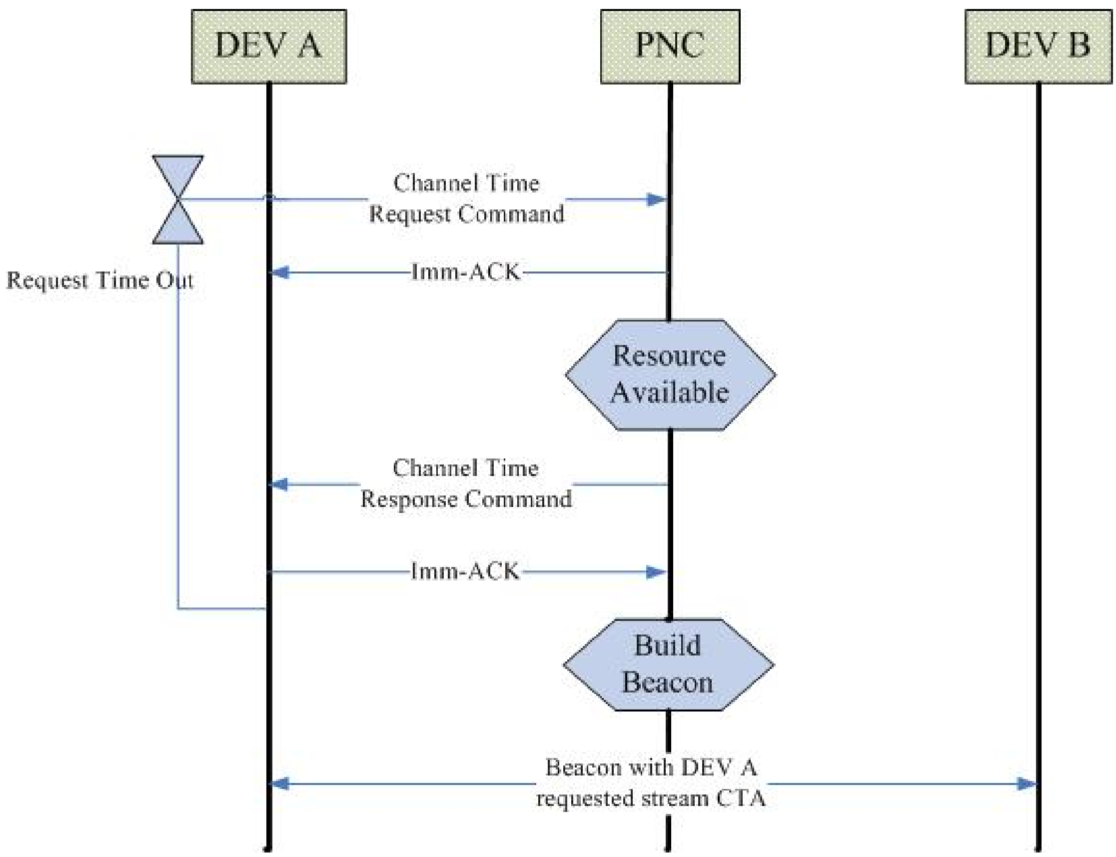

Figure 2-3 Isochronous Channel Time Request Procedure

If the DEV needs channel time on a regular basis, it makes a request from the PNC for an isochronous channel time. If the resources are available, the PNC allocates a CTA time for the DEV. The DEV requests for amount of channel time for transmission and the PNC calculates whether the remaining resource called "Time

Unit" (TU) is available. The TU represents the time length of the transmission time of a fragmentation frame, including ACKs. The DEV needs to inform PNC the TU length and the number of TU that are required for this transmission when sending the channel request command. According to the information, PNC can check whether the unallocated TUs in the superframe are sufficient to support the request. If the unallocated TUs are not enough, the channel time request will be dropped. Figure 2-3 illustrates the flows of successfully establishing a DEV A to DEV B stream in a piconet. The channel time request command should contain the desired number of TUs, the length of used TU, and the frequency that PNC should assign the CTA. In the figure, the Imm-ACK means the "Immediate Acknowledgement" policy, which provides an ACK process that each frame is individually ACKed following the reception of each frame. If the requirements for the data change, then the DEV is able to request a change to the allocation. The source DEV, destination DEV, or the PNC can decide to terminate the stream.

Figure. 2-4 shows an example of the channel time being requested for a CTA while Imm-ACKs are used. Here the SIFS means short interframe space, which is the duration that the destination DEV shall wait before starting transmitting the Imm-ACK frame after the end of each transmission.

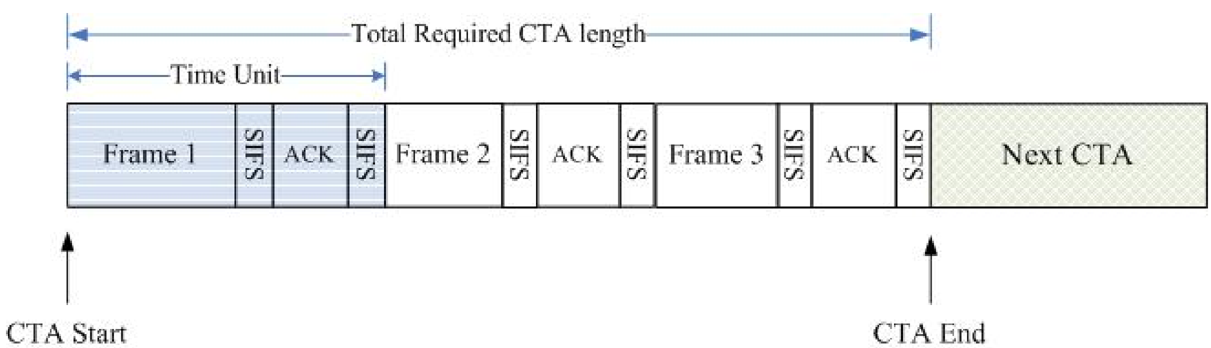

Figure 2-4 Time Unit with Imm-Ack

For regular CTAs, the PNC is able to change their position within the superframe. The CTA which its location can be moved within the superframe on a superframe-by-superframe basis is called dynamic CTA. This allows the PNC has the flexibility to rearrange CTA assignments in order to optimize the utilization of the assignments. The PNC moves a dynamic CTA by simply changing the CTA parameters in the beacon. Dynamic CTAs may be used for both asynchronous and isochronous streams.

If a DEV misses a beacon, it is unable to use the allocation for a regular CTA. To avoid lost throughput due to missed beacons, DEVs are allowed to request a special type of CTA called pseudo-static CTA. If the DEV is allocated a pseudo-static CTA, it is allowed to use the CTA for up to mMaxLost-Beacons missed beacons. The PNC is allowed to move the locations of these CTAs, but needs to maintain the time for the old allocation for mMaxLost-Beacons superframes to avoid collisions. Pseudo-static CTAs shall be allocated only for isochronous streams.

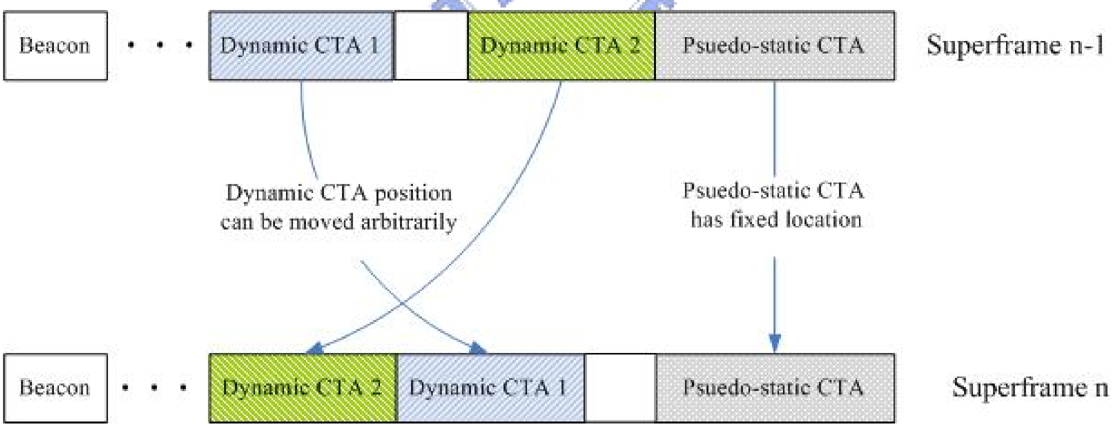

Figure 2-5 Dynamic CTA and Pseudo-static CTA mechanism

2.3.2 Asynchronous channel time reservation

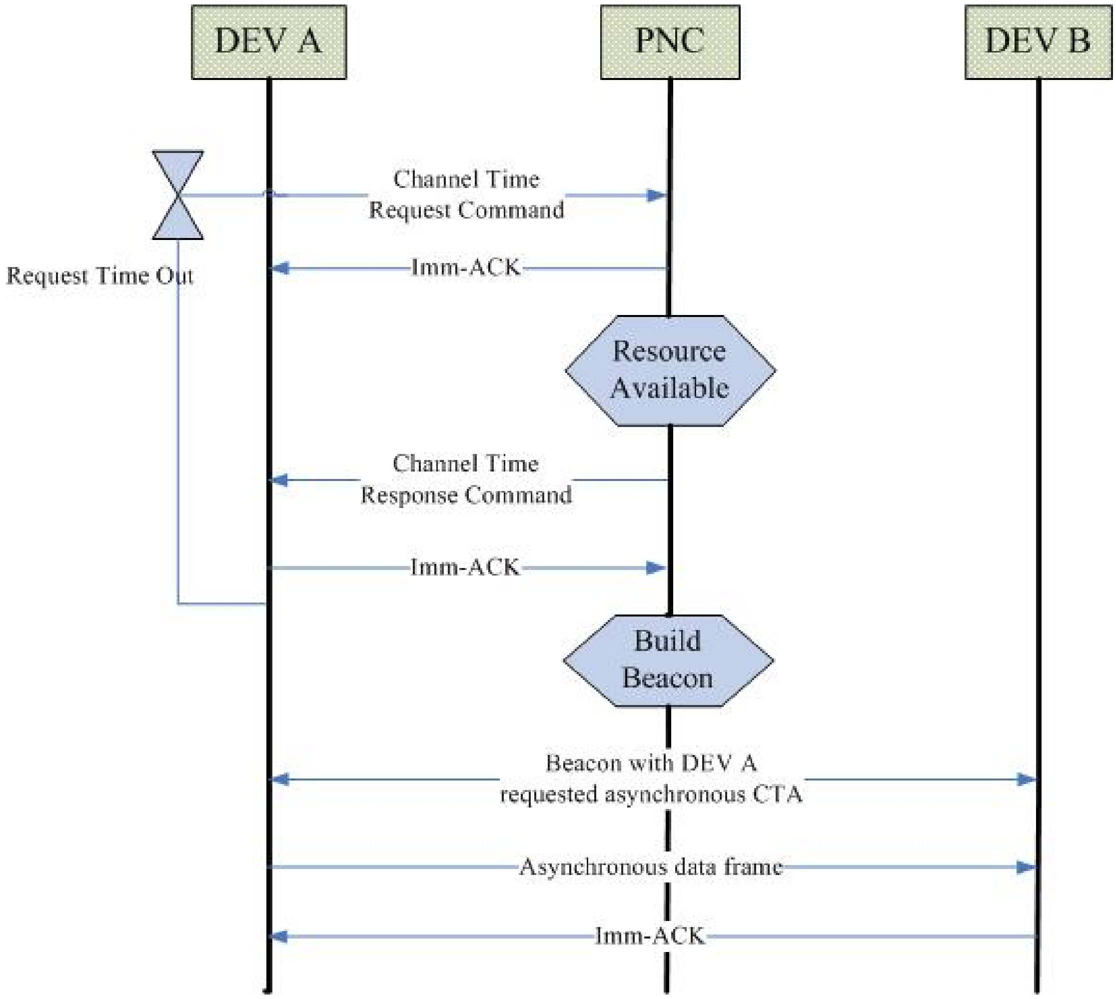

Figure 2-6 Asynchronous Channel Time Request Procedure

Asynchronous allocation is slightly different from isochronous stream. Rather than requesting recurring channel time, an asynchronous channel time request is a request for a total amount of time to transfer its data. The PNC then schedules the channel time for this request if the resource is available. If the DEV needs to transmit another asynchronous data frame, it has to send a new request again. What merits attention is that there is no absolute guarantee of the length of the delay between the time of the request and the reception of a beacon containing the requested CTA. If the DEV does not get its requested CTA in the beacon until the data frame's time out interval expires, transmission time out occurs and this frame will be dumped. Unlike an isochronous allocation, only the source DEV or PNC are allowed to terminate an asynchronous allocation. Figure 2-6 shows an example of successfully reserving the channel time for the exchange of asynchronous data between DEV-A and DEV-B in a piconet.

2.4 *Introduction to Traffic Specified by WINNER*

WINNER (Wireless World Initiative New Radio) [18] is a consortium of 41 partners coordinated by Siemens working towards enhancing the performance of mobile communication systems. One of its objectives is to identify and analyze challenging user and usage scenarios to derive requirements for the WINNER radio interface. For this reason, WINNER has provided various Traffic models to specify their parameters and characteristics. In this section, we will introduce two major traffic types – beast effort traffic and real-time service.

2.4.1 Best effort traffic

A "Best Effort" service is one which does not provide full reliability. It usually performs some error control (e.g. discarding all frames which may have been corrupted) and may also provided some (limited) retransmission. The delivered data is not however guaranteed. A best effort service, normally requires reliability to be provided by a higher layer protocol. Next we will introduce several types of best effort traffic.

2.4.1.1 TCP model

The Transmission Control Protocol (TCP) is one of the core protocols of the Internet protocol suite. Using TCP, applications on networked hosts can create connections to one another, over which they can exchange data in packets. The protocol guarantees reliable and in-order delivery of data from sender to receiver. TCP also distinguishes data for multiple connections by concurrent applications (e.g. Web server and e-mail server) running on the same host.

The TCP connection set-up and release protocols use a three-way handshake mechanism. The amount of outstanding data that can be sent without receiving an acknowledgement (ACK) is determined by the minimum of the congestion window size and the receiver window size. After the connection establishment is complete, the transfer of data starts in slow-start mode with an initial congestion window size of 1 segment. The congestion window increases by one segment for each ACK packets received by the sender. This results in an exponential growth of the congestion window. This process is illustrated in Figure 2-7.

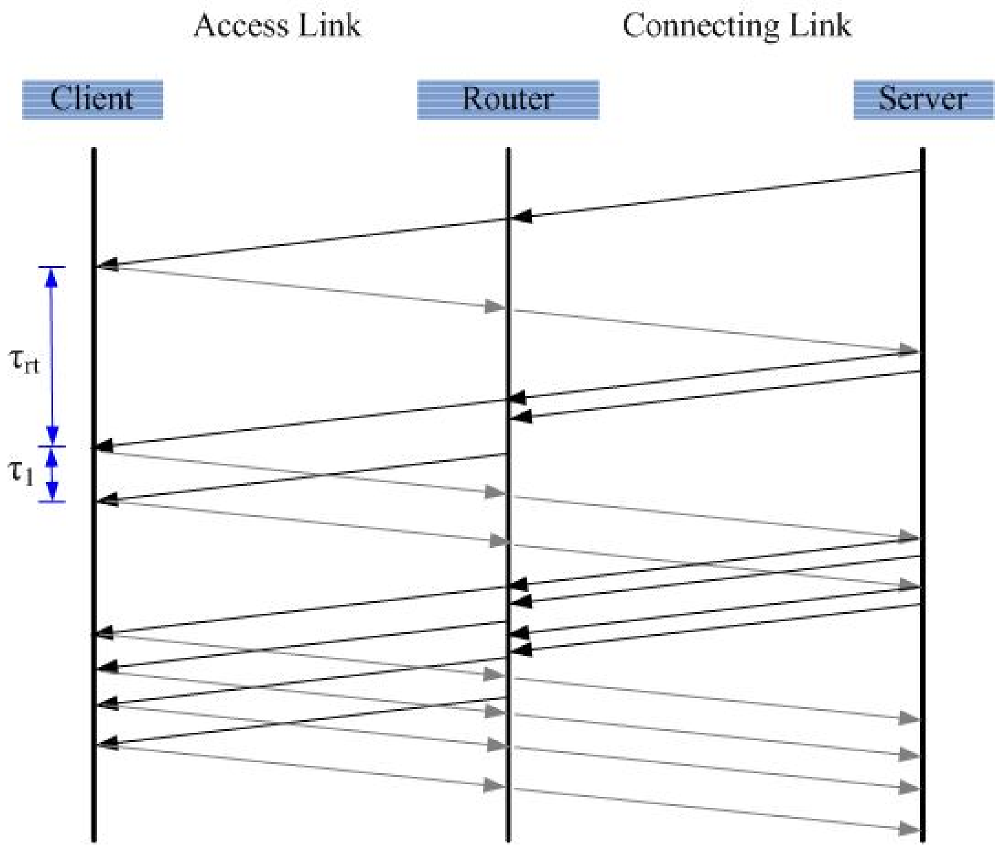

Figure 2-7 TCP flow during slow start

τ_1 is the transmission time of a TCP data segment over the access link from the base station router to the client, and τ_{rt} is the sum of the time taken by an ACK packet to travel from the client to the server and the time taken by a TCP data segment to travel from the server to the base station router. The value of τ_1 is usually less than 2ms, but the mean value of τ_{rt} is 200ms. It's obvious that the variance of arrival interval of traffic which is using TCP protocol will be very large. Thus we can distinguish this type of traffic from real-time traffic. We'll discuss it in chapter 4.

2.4.1.2 HTTP model

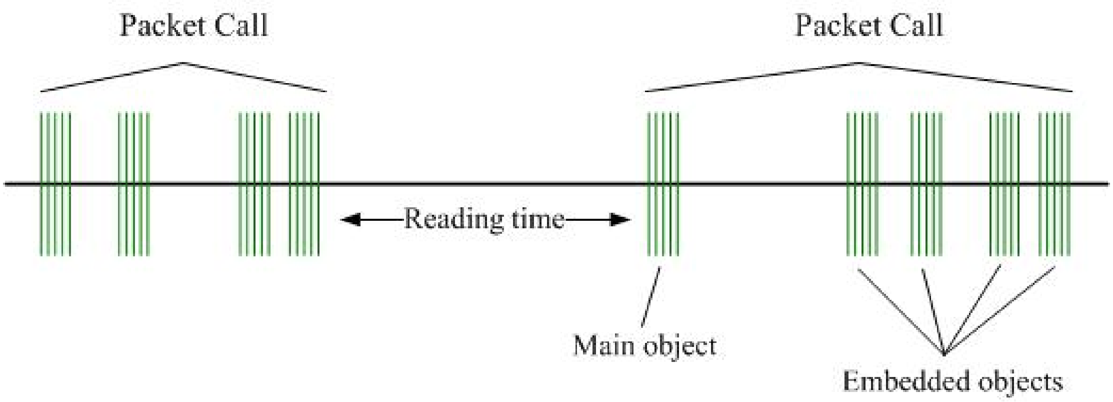

Figure 2-8 Contents in a packet call

Figure 2-8 shows the packet trace of a typical web browsing packet call. The packet call is divided into ON/OFF periods representing web-page downloads and the intermediate reading times. These ON and OFF periods are a result of human or machine interaction where the packet call represents a user's request for information and the reading time identifies the time required to digest the web-page. As an example, consider a typical web-page from the Yahoo Taiwan in Figure 2-9. This web-page is constructed from many individually referenced objects. A web-browser will begin serving a user's request by fetching the initial HTML page using an HTTP GET request. After receiving the page, the web-browser will parse the HTML page for additional references to embedded image files such as the graphics on the tops and sides of the page as well as the stylized buttons. The retrieval of the initial page and each of the constituent objects is represented by ON period within the packet call while the parsing time and protocol overhead are represented by the OFF periods within a packet call. For simplicity, the term "page" will be used in this paper to refer to each packet call ON period. As a rule-of-thumb, a page represents an individual HTTP request explicitly initiated by the user. The initial HTML page is referred to as the "main object" and the each of the constituent objects referenced from the main object are referred to as an "embedded object".

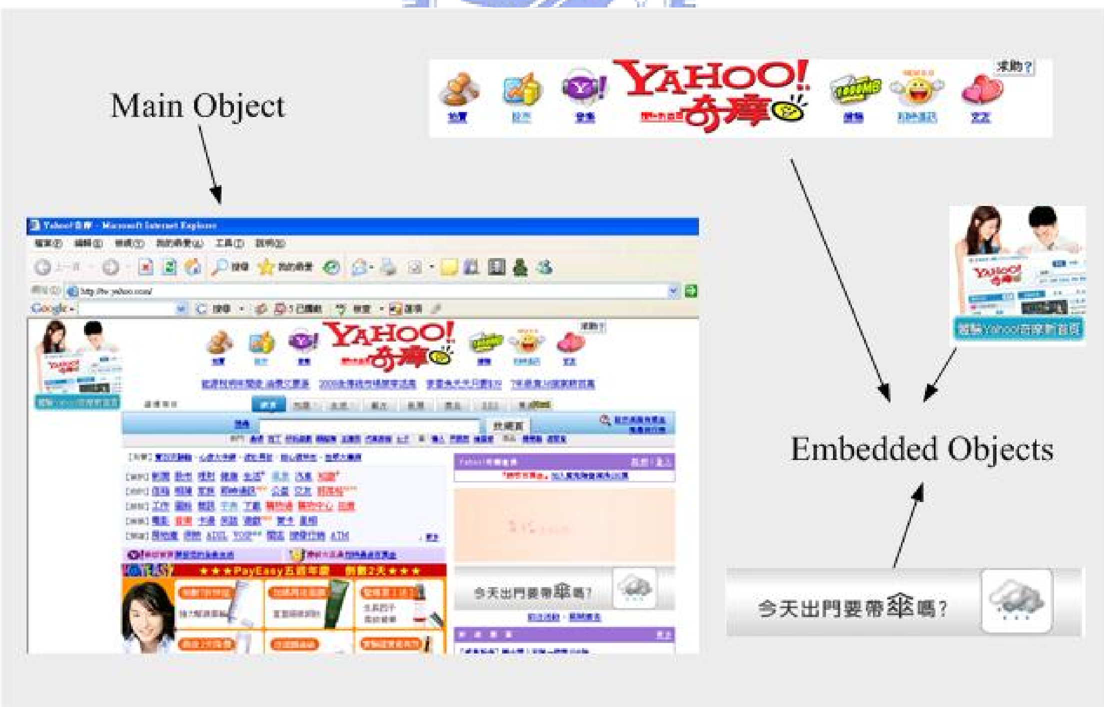

Figure 2-9　　A typical web page and its content

In HTTP/1.0, a distinct TCP connection is used for each of the main and embedded objects downloaded in a web page. Most of the popular browser clients download the embedded objects using multiple simultaneous TCP connections; this is known as HTTP/1.0-burst mode transfer. The maximum number of such simultaneous TCP connections, N, is configurable; most browsers use a maximum of 4 simultaneous TCP connections. If there are more than N embedded objects, a new TCP connection is initiated when an existing connection is closed. The effects of slow-start and congestion control overhead of TCP occur on a per object basis.

In HTTP/1.1, persistent TCP connections are used to download the objects, which are located at the same server and the objects are transferred serially over a single TCP connection; this is known as HTTP/1.1-persistent mode transfer. The TCP overhead of slow-start and congestion control occur only once per persistent connection.

2.4.1.3 FTP model

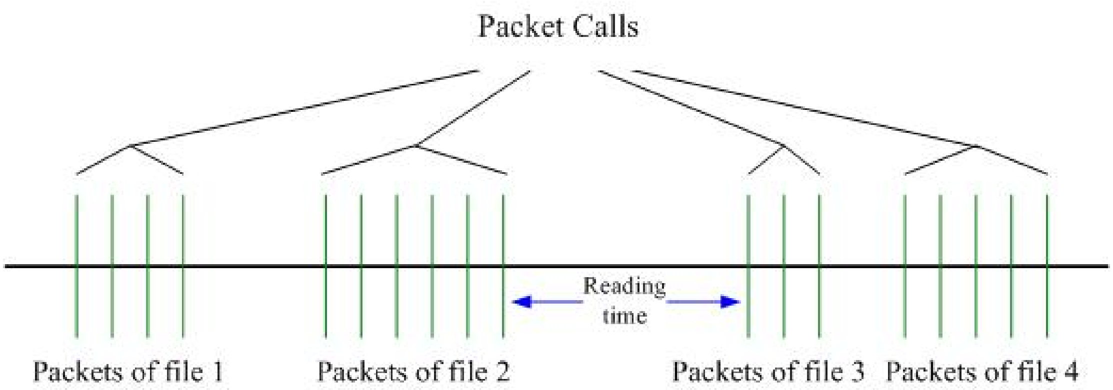

Figure 2-10 Packet trace in a FTP session

In FTP applications, a session consists of a sequence of file transfers, separated by reading times. The two main parameters of an FTP session are the size of a file to be transferred and reading time, which is the time interval between end of download of the previous file and the user request for the next file.

The underlying transport protocol for FTP is TCP. The packet trace of an FTP session is shown in Figure 2-10.

2.4.2 Real-time service

A real-time service provides its clients with the ability to specify their performance requirements, which include various types of delay bounds, throughput bounds, and reliability bounds, and to obtain guarantees about the satisfaction of those requirements.

2.4.2.1 VoIP model

Voice over internet protocol (VoIP), is a technology that allows you to make telephone calls using a broadband Internet connection instead of a regular (or analog) phone line. VoIP converts the voice signal from your telephone into a digital signal that travels over the Internet. If you are calling a regular phone number, the signal is then converted back at the other end. VoIP can allow you to make a call directly from a computer, a special VoIP phone, or a traditional phone using an adapter. Compression techniques(Codecs) were developed allowing a reduction in the required bandwidth while preserving voice quality. Within WINNER we consider 3 different rates of conversational voice traffic, corresponding to 8, 32 and 64 kbit/s constant bit rate (during talkspurt).

Figure 2-11 shows that voice activities can be considered as alternating between two states: talkspurt and silent. Data are generated during talkspurt only, and no data is transmitted during silence, thereby making statistical multiplex gain possible. Paul T. Brady discovered that both talkspurt and silence periods of digitized voice are exponentially distributed. The commonly accepted model for a speaker in a voice call is a continuous-time, discrete-state Markov chain. The holding time in each state is assumed to be exponentially distributed with mean 1/λ and 1/μ, respectively.

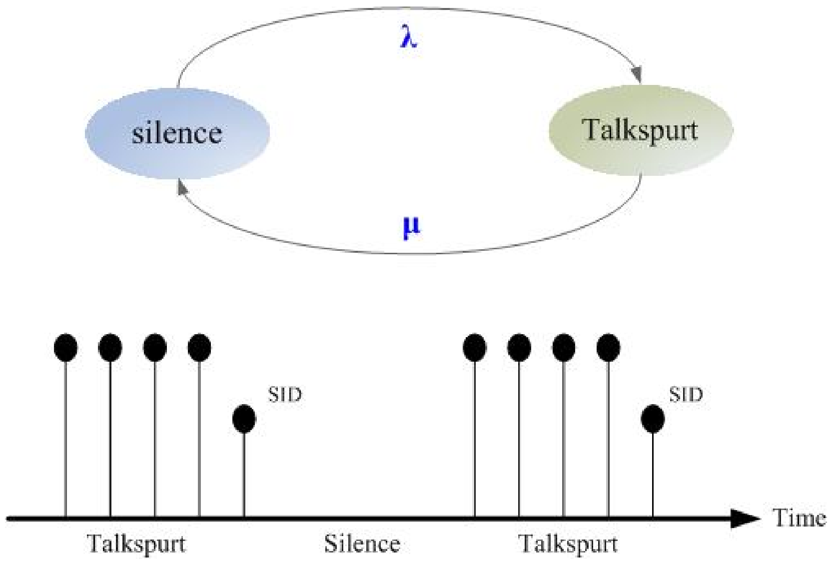

Figure 2-11 VOIP single speaker model

2.4.2.2 MPEG model

MPEG (Moving Pictures Experts Groups) is a coding method for moving pictures, audio, and related data. For different application, there are various coding format in MPEG family: MPEG1, MPEG2, MPEG4 and MPEG7. MPEG is most commonly used video compressing technology today.

An MPEG encoder generates three types of frames: Intra-coded (I), Predictive (P), and Bi-directional (B) frames.

- I frame：Intra-frame encoding is based solely on information within a single frame. Furthermore, the I-frame encoding and decoding process may begin as soon as the first 16 lines of a frame are received.
- P frame：Predictive encoding uses a previous frame and encodes only the differences between that frame and the current frame to be encoded.
- B frame：Bi-directionally encoded frames use both a previous (I or P) frame and a future (I or P) frame, forming a best match interpolation between

those two frames and the current frame and encoding the resulting differences.

The P frames contain less information than I frames. Generally P frame's average size is about a quarter of the I frame. As compared to I & P frame, B frame has the smallest frame size. The frame size is about one half of the P frame and therefore it is about an eighth of the I frames.

When compressing a video sequence, typical MPEG encoders use a pre-defined group of pictures (GOP). The parameter of GOP(N,M), N represents number of frames from one I frame to next I frame, and M stands for number of frames from one I frame to next P frame. For example, when GOP(6,2) is used, the encoded video frame sequence will be IBPBPBIBPBPBIB…

In WINNER standard, various display sizes and resolution are considered. Table 2-1 shows that frame sizes of different resolutions. f_F represents frame arrival rate, μ represents average frame size, and σ is variance of frame size. For Common Intermediate Format (CIF), Quarter Common Intermediate Format (QCIF) and High-definition television (HDTV) formats, it is necessary to scale the parameters down or up to different resolutions. CIF (352 x 288) is often applied to laptop video streaming, while QCIF is more suitable for the small display of mobile phones. The frame rates are different for these resolutions, too. For CIF 15 Hz and 30 Hz are common, while for QCIF 10 Hz and 15 Hz are used. HDTV is the standard for High Definition Television. In current practice, HDTV uses 1280 x 720 pixels displays in Progressive Scan Mode (720p) (with a frame rate of 24, 30 or 60 Hz) or 1920 x 1080 pixels displays in Interlace Mode (1080i), but with a reduction of the number of frames per second to 24 Hz. By table 2-1 and formula below, we can calculate the bit rate R_U .

$$R_U = \frac{x * y \text{ [pixels]}}{\text{[frame]}} * f_F \left[\frac{\text{frames}}{\text{sec}}\right] * S_P \left[\frac{\text{bits}}{\text{pixel}}\right]$$

Where x*y is the resolution of the video, S_p is the number of bits per pixel.

Table 2-1 Frame sizes of different resolutions [Kbits]

f_F [Hz]		HDTV(1080i) (1920*1080 pixels)			HDTV(720p) (1920*720 pixels)			VGA, NTSC (940*480 pixels)			CIF (352*288 pixels)			QCIF (176*144 pixels)		
		I	**P**	**B**	**I**	**P**	**B**	**I**	**P**	**B**	**I**	**P**	**B**	**I**	**P**	**B**
60	μ				920	230	115									
	б				365.4	148	64.5									
30	μ				438	110	55	146	60	18.9	73.8	8.1	2.2			
	б				174	70.7	30.8	58	38.6	10.6	29.6	5.1	1.2			
24	μ	832.2	208.1	104	365	91.3	46									
	б	330.6	134	58	145	58.7	25.8									
15	μ										45.0	6.8	3.2	6.2	0.8	0.5
	б										18.1	4.3	1.8	2.5	0.55	0.3
10	μ													7.7	0.5	0.15
	б													3.1	0.3	0.08

Chapter 3
QoS Issue and Existing Works

We will have the discussion on the QoS topics in this chapter. Section 3.1 is a brief introduction to the QoS. Section 3.2 introduces the general resource scheduling method. Some related works and the QoS problems are described in the section 3.2.

3.1 Introduction to QoS

The QoS is a concept which is general but difficult to make comprehensive explanation. The statement which is apt to let people understand is “Quality of Service, which refers to control mechanisms that can provide different priority to different users or data flows, or guarantee a certain level of performance to a data flow in accordance with requests from the application program. It is a general term that incorporates bandwidth, latency, and jitter to describe a network's ability to customize the treatment of specific classes of data.”

To ensure the QoS requirements, many ideas such as service differentiation and resource management have been proposed. The main concept of the service differentiation is to adjust the probability of obtaining the medium to transmit via assigning different traffic with different priorities. The policy of assigning the priority is on the basis of the following criteria: customer payment, traffic types, traffic demand, etc. Note that the priority does not provide any QoS guarantee actually. It only guarantees that the traffic with higher priority can acquire the resource more easily than the lower priority traffics.

There are many studies for resource management. Among those, the most are focusing on call admission control [25-26] and bandwidth allocation schemes [27]. The purpose of call admission control is to decide whether to accept or reject the new coming users according to different criteria. If the system accepts the new user will cause intolerable influence to the current serving applications, the new user should be rejected. On the other hand, bandwidth reservation control mechanisms are to make sure current resources are sufficient for exist traffics.

In order to provide QoS, wireless network systems usually have some coordinators, like base station in 802.16 or PNC in 802.15.3 network, which are responsible for managing service differentiation and resource management. Next we will introduce some existing channel resource scheduling method used by network coordinators.

3.2 General Resource Scheduling Method

Nowadays, a considerable number of studies have been conducted on resource scheduling methods. Next two kinds of those methods will be introduced: One is **fixed resource allocation,** and another is **dynamic resource allocation.**

Fixed resource allocation is a scheduling method that the coordinator will allocate fixed and equal length transmission time to each DEV in the network. Obviously, it is easiest method because the scheduler doesn't need to adjust resource allocation result frequently. Besides, this mechanism can achieve better fairness since every DEV gets the equal channel time allocations (CTAs) and allow suddenly arrived data to get CTAs without request in advance. However, this algorithm has several drawbacks. First, because source DEV will still be allocated CTAs even if it has no data to send, it results in the waste of resources. In other words, the bandwidth will be engaged without necessary. Second, the coordinator doesn't concern about user's QoS requirement at all. In short, this method is simple but not suitable for most real-time service.

Dynamic resource allocation is a method in opposition to fixed CTA scheduling. In dynamic resource allocation, the coordinator is able to allocate different resources for each DEV according to users requirements, system loading, or channel condition. In many cases, first in first out (FIFO), weighted fair queue(WFQ), and earliest deadline first(EDF) are used.

This FIFO improves the problem of resources waste because coordinator allocates resources only when DEV has transmission requirement. However, the data that comes earlier doesn't mean it is more urgent. The deadline of each transmission is not monitored. In brief, when considering user's QoS satisfaction, FIFO apparently lacks the concept of priority.

WFQ is a widely used scheduling method [10], which is designed for QoS supporting. This method employs the queue status and priority assignment to determine the allocation weighting which represents the allocating priority of each link. By providing allocation weighting to each link, the bandwidth allocation among each link considers not only the user request but also the weighting assigned to packet queues. However, in order to calculate the weighting precisely for different services, the scheduler, such as the PNC, must estimate the required data rate of each traffic flow and obtain detail information of all waiting queues in the whole wireless network. Therefore the implementation of WFQ is complicated.

The policy of EDF is always allocating the packet which is closest to its deadline. The objective of this method is to transmit each packet before deadline expired. It is especially applicative for scheduling real-time traffic [12] which has strictly timing constraint. In [13], the authors has proved that EDF is the optimal scheduling policy for real-time service when channel is always in "Good" state. Since 802.15.3 DEVs are basically fixed, it seems that the EDF is suitable for real-time transmission in 802.15.3 system. However, it is still not perfect. We will show that EDF might not be suitable for some applications in next section and chapter 4.

3.3 QoS issues and Related Works

As we described in section 3.2, the EDF method is suitable for scheduling real-time traffic. However, we find out three serious problems when schedulers employ the EDF method:

1. Problems of traffic identification：The resource scheduler, mostly in MAC layer, usually doesn’t have enough QoS information of each transmission. There are two reasons:

First, the QoS information is record in the header of IP layer which is the upper layer of MAC layer. Figure 3-1 shows the IPV4 specification. Size of the Differentiated Services Field (DS Field) [16] [17] which is defined to provide the information of traffic QoS by setting the codepoint (CP) is 6 bits. The description of different service codepoint is illustrated in Table 3-1. First bit is a flag of best effort traffic, which is used to declare whether this packet is belong to best effort traffic or not. Second bit to fifth bit are serving priority of different traffic. Sixth bit represents the Expedited Forwarding (EF) flag, which is used to build a low loss, low latency, low jitter, and assured bandwidth end-to-end service. Unfortunately, this standard is not widely adopted in practice. Moreover, the size of DS field is too small and the setting of CP is quite ambiguous. Thus the upper layer of MAC layer can not provide MAC layer enough information for precise resource scheduling.

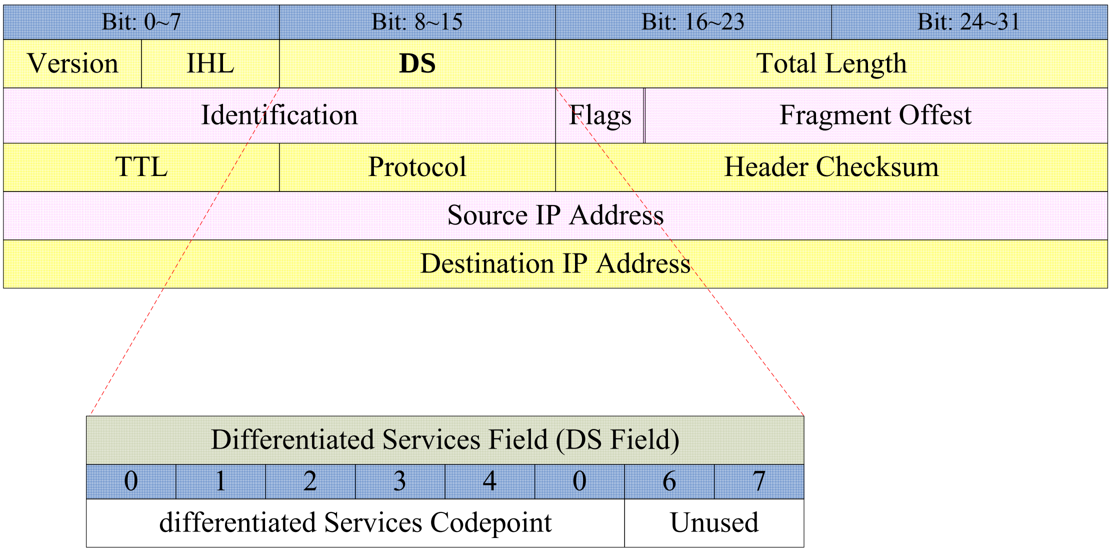

Figure 3-1 Differentiated Services Field in IP header

Table 3-1 Description of Differentiated Services Codepoint

differentiated Services Codepoint	
Bits	**Description**
0	Best effort
1	Class 1
2	Class 2
3	Class 3
4	Class 4
5	Express Forwarding
6	Stays the same (used for IP routing protocols)

Second, the scheduler in MAC layer usually doesn't decode upper layer's header in every packet due to working overhead or handling speed issues. According to the reasons described above, traffic identification is big issue of scheduling. Further discussion of this problem is in section 4.2.

2. "Real" QoS requirement: Traditionally, an application will be treated the same even the transmitted frames associated to that application might have different important levels or different sizes. For example, as the traffic introduction in chapter 2, the I , B, P frames of MPEG codec have different importance and sizes. That is to say, the QoS requirement of specific traffic might continually change, thus original QoS requirement defined by upper layer doesn't mean "real" requirement all the time. This problem is huge because although there is the perfect scheduling method, we still can't get best performance when the provided request doesn't represent the "real" need of specific traffic. We'll modify oversimplified EDF to solve this problem in section 4.1.

As today, only a few researches [23] [24] have made their efforts to solve the problem mentioned above. The author of [21] proposed a simple application-aware MAC scheme for the 802.15.3 in order to achieve a high quality video transmission of MPEG-4 stream. The main idea is let DEV informs PNC the maximum sizes of its I-frame, P-frame, and B-frame in the channel time requests before each creation of the isochronous stream and the PNC allocates a channel time for the DEV according to the predefined frame sequence. This method continuously changes QoS requirement and guarantees that transmitter has enough resources to transmit image frames. However, there is a huge drawback when realizing this scheme. Since the PNC always allocate the maximum length CTA in every superframe, resources may be occupied excessively. Thus it is important to find the right trade-off between user satisfaction and network efficiency.

[20] proposes a simple scheduling algorithm called MES (MCTAs at the end of Superframe) for MPEG-4 traffic in accordance with IEEE 802.15.3 standard. In this algorithm, piconet coordinator (PNC) allocates one Management Channel Time Allocation (MCTA) for each stream which is in the process of communication at the end of superframe. During the MCTA period, each transmitter should report current queue size belonging to the stream to PNC. PNC will allocate corresponding channel

time in the next superframe based on the queue size. The result shows better performance in QoS for multimedia streams compared to the existing schemes. However, the scheduler can't allocate resources whatever DEVs requested due to limited resource. This method does not concern about priority issues of different flows.

3. Scheduler information update: after the transmitter of network members have identified each traffic and computed their "real" QoS requirement, momentarily update those information of each stream for the scheduler of network is necessary but hard to achieve. We will aim at this problem and propose a modified MAC protocol of IEEE 802.15.3 in section 4.3.

In [19], the author propose a rate-adaptive Medium Access Control(MAC) protocol for HR WPAN. The RF transmission rate for the next transmission is selected by channel prediction based on the currently received frame and **informs** the sender about the changed rate using a **rate-adaptive acknowledgement** (RA-ACK) frame. By overhearing the RA-ACK frame, a piconet controller can efficiently allocate channel times. However, this method can only achieve better throughput but didn't concern about QoS issues.

Chapter 4
Priority-based EDF Algorithm and Proposed MAC Protocol

In this chapter, a new resource allocation scheme is proposed in chapter 4.1 to solve the QoS problems mentioned in the previous chapter. Section 4.2 introduces the traffic identifier which is the proposed mechanism in order to find the "real" requirement of each flow. The modified MAC protocol designed to implement our proposed algorithm is described in section 4.3.

4.1 Priority-Based EDF Algorithm

In section 3.2, we have mentioned that the EDF algorithm is suitable for scheduling real-time service. However, in section 3.3, we talk about several serious problems when schedulers apply the EDF method. According to our discussion, the EDF algorithm itself is a very good idea. The problem is the user requirements that the EDF scheduler follows are not "real" requirements all the time. Thus our proposed algorithm is to eliminate the problem caused by the traditional EDF method.

The characteristics of specific traffic must be analyzed if we want to find out the "real" requirement. Next we will use the most popular and generally used video and audio codec: MPEG series [14] [15] [21] to explain the existed problems and how we can solve them. As introduced in section 2.4.2.2, an MPEG encoder generates three types of frames: Intra-coded (I), Predictive (P), and Bi-directional (B) frames. When transmitting a video sequence, typical MPEG encoders use a pre-defined group of pictures (GOP). As shown in Figure 4-1, while the parameter of GOP(N,M) is set, N represents the distance between two I frames, and M stands for the distance between a

I frame and a P frame. f_F (Hz) is the image frame arrival frequency. In general, an I-frame is encoded with no reference to any past or future image frames, and a P-frame and a B-frame are encoded relative to the past or future reference image frames. In other words, I frame contains the whole information of GOP image frames, and P frames or B frames are related to I frame which is in same GOP. Thus when the I frame of certain GOP fails, the whole image frames in that GOP will not be reconstructed even if other frames are transmitted correctly. On the contrary, the failure of one P frame or one B frame represents just one damaged image frame. Thus when analyzing MPEG traffic, rather than focusing on frame error rate (FER) which represents the probability of MAC frames failed to transmit correctly, we will more focus on the job failure rate (JFR), that is the probability of image frames failed to reconstruct. The JFR is calculated as:

$$JFR = \frac{(F_I * N + F_{P|I_{correct}} + F_{B|I_{correct}})}{K} \tag{4-1}$$

where F_I is the number of I frames failed, N represents the distance between two I frames, $F_{P|Icorrect}$ and $F_{B|Icorret}$ are the number of failure P frames and B frames while I frame in same GOP is correct, and K is total number of image frames transmitted.

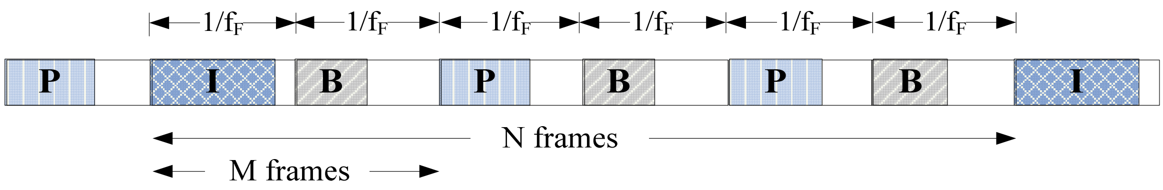

Figure 4-1 MPEG codec traffic flow structure (GOP=(6,2))

Generally speaking, if the EDF method is applied, MAC packet data unit (MPDU) fragmented from the same video frame will have the same delay bound (DB).

Even though there is only one packet that fails to deliver successfully before the deadline, the whole video frame will still fail. Traditionally, DB of MPEG video frame which is set by the layer above the MAC layer is defined below:

$$DB = \frac{1}{f_F} + T_{Buffer} \tag{4-2}$$

where f_F (Hz) is the image frame arrival frequency, T_{Buffer}(ms) is the average buffering time of the MPEG decoder (typically 500 ~ 1000 ms). This formula represents that the data of a image frame must be transmitted completely before the MPEG decoder starting to decode that frame.

The question now arises: The priority setting of EDF depends on the value of DB (the smaller DB, the higher priority), however, the DB of all frames in the same MPEG traffic flow is related to the frame inter-arrival time only. The **importance-level** and **size difference** of different frames is not considered. As introduced before, the information amount of image frames are different. One I frame is approximately four times larger than a P frame and two times larger than a B frame. Moreover, the importance of I frames is more than P frames and B frames. In fact, under the same streaming coding scheme, lager frame size represents more information with higher importance level. Thus the scheduler should give higher priority (lower DB) to lager frame in the same traffic flow. To reflect the important level and size difference of different frames, a **frame weight (FW)** is proposed as follows:

$$FW_i : FW_j = \frac{S_j}{R_j} : \frac{S_i}{R_i} \tag{4-3}$$

where S_i and S_j are frame sizes of the i frame and the j frame, R_i and R_j are the corresponding transmission rates. For example, for the same transmission rate, if FW_I is normalized to 1, FW_P and FW_B could approximately have values of 1/2 and 1/8. In short, the more important data will have higher probability to be served.

Table 4-1 Frame sizes of different resolutions [Kbits]

f_F [Hz]		HDTV(1080i) (1920*1080 pixels)			HDTV(720p) (1920*720 pixels)			VGA, NTSC (940*480 pixels)			CIF (352*288 pixels)			QCIF (176*144 pixels)		
		I	**P**	**B**	**I**	**P**	**B**	**I**	**P**	**B**	**I**	**P**	**B**	**I**	**P**	**B**
60	μ				920	230	115									
	б				365.4	148	64.5									
30	μ				438	110	55	146	60	18.9	73.8	8.1	2.2			
	б				174	70.7	30.8	58	38.6	10.6	29.6	5.1	1.2			
24	μ	832.2	208.1	104	365	91.3	46									
	б	330.6	134	58	145	58.7	25.8									
15	μ										45.0	6.8	3.2	6.2	0.8	0.5
	б										18.1	4.3	1.8	2.5	0.55	0.3
10	μ													7.7	0.5	0.15
	б													3.1	0.3	0.08

Another problem caused by applying EDF is the priority setting issue between different flows. Table 4-1 shows the frame sizes of different resolutions. Even if the frame arrival frequency (f_F) of different resolutions are the same, data rate of MPEG flows might has a gap from 2~1000 times. However, the traditional DB is only related to the f_F. Thus, in order to set appropriate priorities, another parameter, **traffic weight (TW),** is used to adjust DB of different traffic types. According to the queuing theory, to deal with the traffic which can be served quickly should improve the performance as a whole. For example, when the data rate of type-A traffic is *n* times larger than type-B's, serving one type-A traffic will lose the opportunity to serve n type-B traffics. Moreover, when there are limited resources left, it might be better to serve type-B traffic since we can not satisfy the requirement of type-A traffic anyhow. In short, the allocate opportunity should be directly proportional to the data rate, below is the formula:

$$TW_i \ : \ TW_j \ = \ \frac{D_i}{R_i} \ : \ \frac{D_j}{R_j} \tag{4-4}$$

Where *Di* and *Dj* are required data rate of traffic flows, *Ri* and *Rj* are the corresponding transmission rates.

In conclusion, first, the important level and size difference of different frames in the same traffic flow should be considered. Second, the priority of different traffic flows must be set. Thus the proposed priority-based EDF scheduling algorithm is defined as follows:

$$P^* = \arg \ \{ \min_{i \in (1,2.....k)} \ (\frac{1}{f_{F\ i}} * TW_i * \ FW_i \ + \ T_{Buffer}) \} \tag{4-5}$$

where *P** is a packet designed to serve next.

4.2 Traffic Identifier

In section 3.2, we have mentioned the problem of traffic identifying. In fact, the scheduler of MAC layer can only have limited QoS information of each flow. However, even if the QoS requirement is clearly recognized by the scheduler, it doesn't represent "real" requirement all the time due to the characteristics of each flow such as the variation of image frame size or the different importance between image frames. So in last section, we have introduced a priority-based EDF algorithm to prioritize what is the "real" requirement. In this section, the traffic identifier is brought up, which not only roughly identifies what type of traffic it is but also helps the scheduler find out the "real" requirement of each flow.

Figure 4.3 shows the flow chart of the proposed traffic identifier. When new connection is established, the identifier will profile current stream flow for a while. After gathering certain statistics, the first step is using the standard deviation of the arrival interval to distinguish real-time traffic from non-real-time traffic. As introduced in section 2.4, the characteristic of the real-time traffic is close to periodic. So its standard deviation of arrival interval should be small. On the other hand, non-real time traffic doesn't have the same characteristic. For example, HTTP traffic has long reading time, and FTP traffic which uses TCP transport protocol has a characteristic of slow start. Thus non-real time traffic has bigger standard deviation of the arrival interval, which helps us to classify the traffic types. Next, the traffic identifier will continuously analyze real-time traffic in order to apply our proposed algorithm. The current frame size can help identifier calculate frame weight (FW) and the average data rate is used to decide traffic weight (TW). By following the rules defined in last section, the delay priority which can reflect "real" requirement of current frame will be provided.

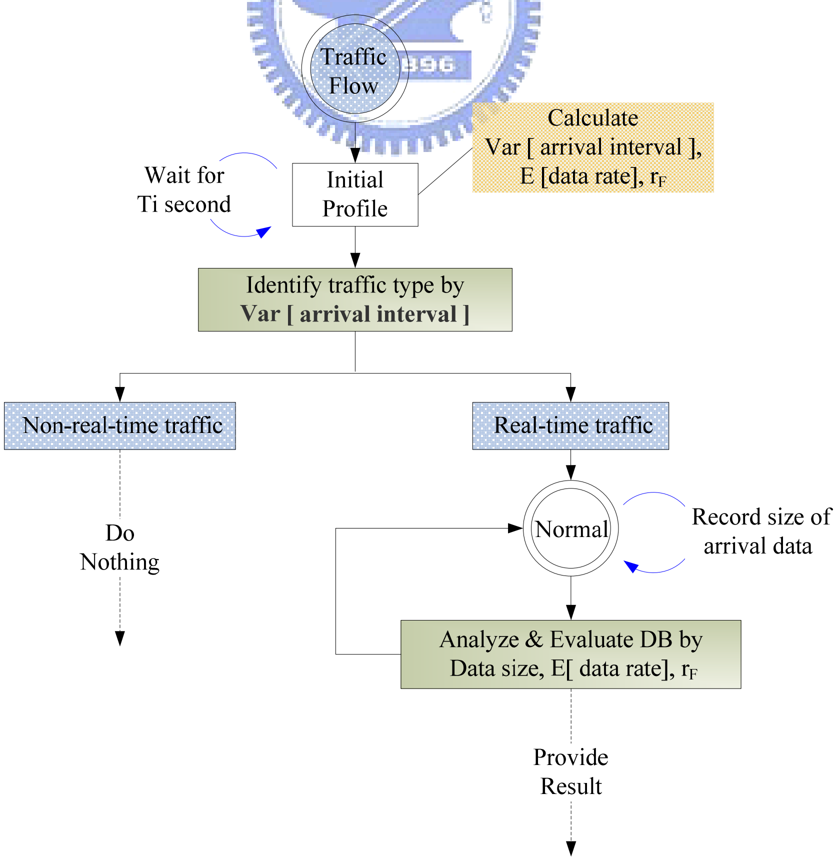

Figure 4-2 The flow chart of traffic identifier

4.3 *Proposed MAC protocol*

Now we have solved first two problems described in section 3.3. After the transmitter of network members have identified each traffic and computed their "real" QoS requirement, the remained question is how to momentarily inform the scheduler about the current status of each flow. In other words, PNC must have the ability to momentarily monitor the variation of piconet status.

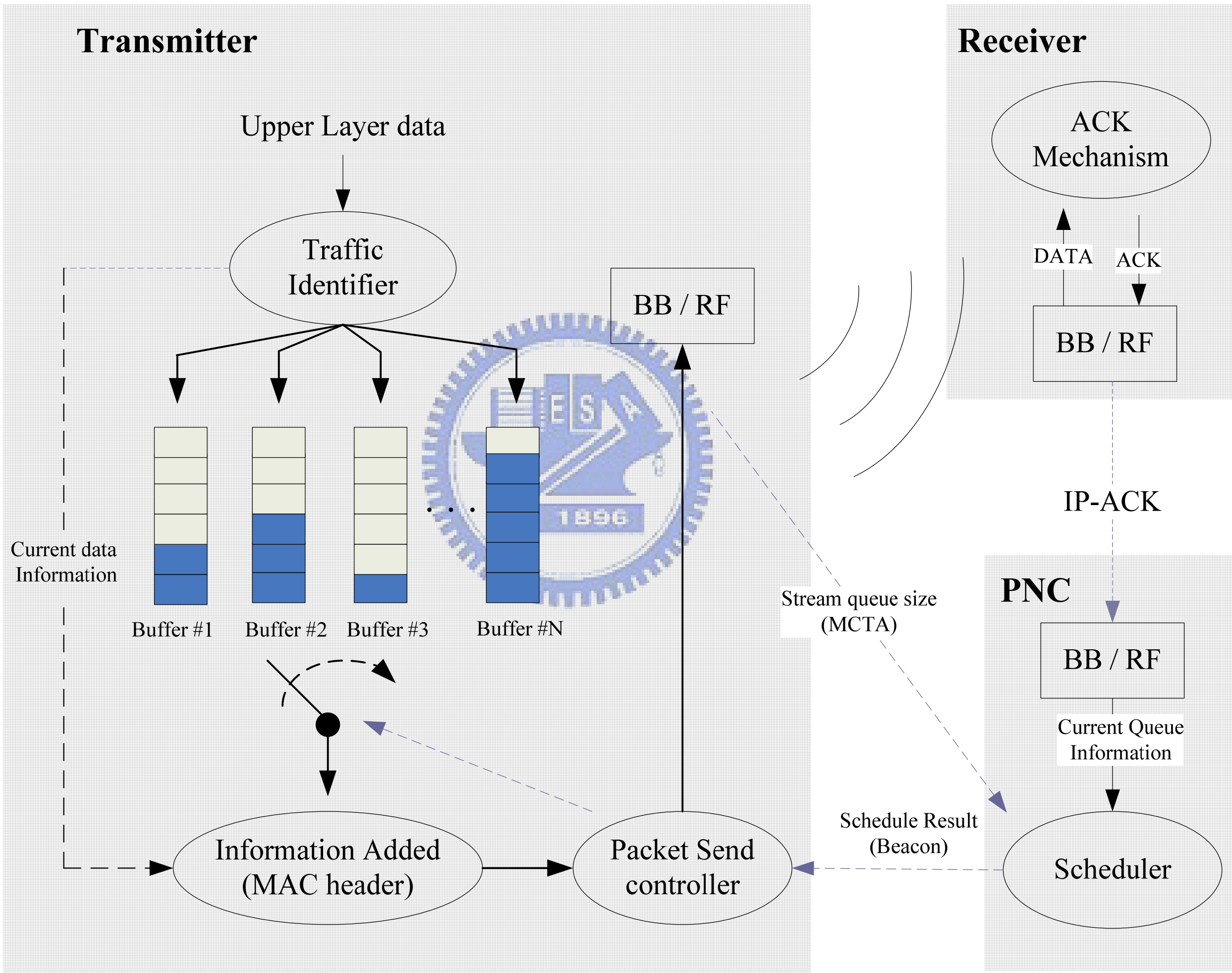

Figure 4-3 Reserved field in 802.15.3 MAC header

Fig. 4-3 is the proposed MAC protocol system architecture, the main goal is to make sure PNC can make correct scheduling decision based on proper information. The black lines are data flows and the dotted lines represent information flows. First, in the transmitter DEV side, when one streaming flow reaches the MAC layer, the traffic type and its corresponding delay bound must be recognized in traffic identifier. After analyzing current flow, the priority of each flow would be determined and the result will be put into 5 bits reserved subfield of the Frame Control field in the MAC header [19], as figure 4-4 shows. Table 4-2 is the description of traffic type field setting, value 11111 means that current flow is best effort traffic and value 00000 to value 11110 represents 31 levels priority setting which is provided by the traffic identifier. When the receiver has received a packet, it will return the Acknowledgement frame (ACK) to the transmitter. PNC can also get this ACK because they are in the same network. Figure 4-5 is the structure of ACK frame in 802.15.3 MAC standard, and ACK frame is composed of MAC header and transmission result reports. Due to the fact that we have already put information in MAC header, PNC is able to get information of current stream through the information provided ACK (IP-ACK). This mechanism is under a premise of conforming to the original specification. Finally, PNC uses current queuing information to allocate channel resources based on the proposed algorithm.

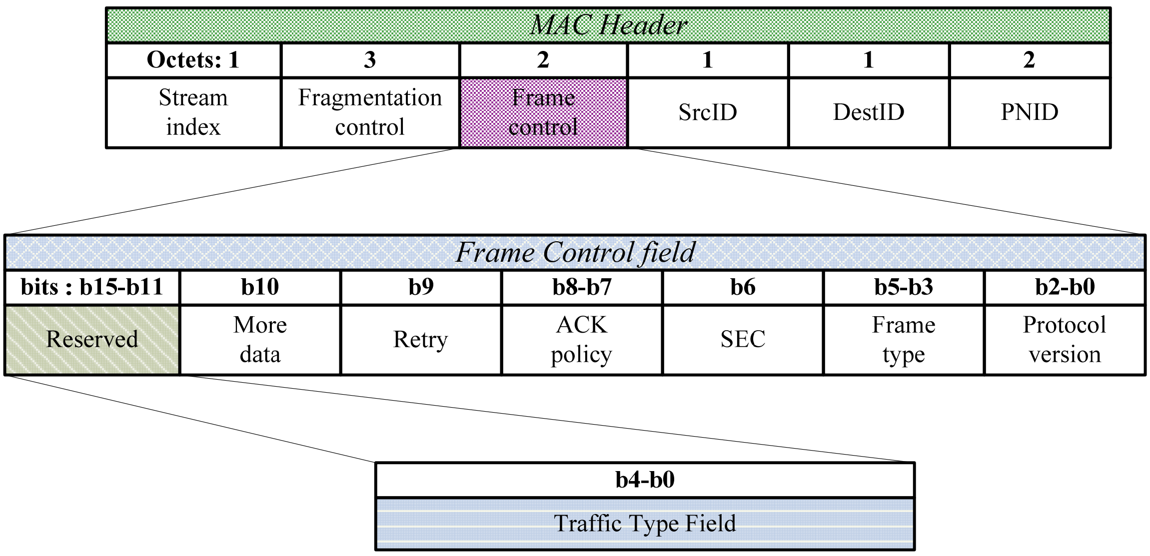

Figure 4-4 Reserved field in 802.15.3 MAC header

Table 4-2 Description of traffic type field setting

Traffic Type Field	
Value	**Description**
00000 ~ 11110	31 Levels Priority Setting
11111	Best Effort Traffic

Octets: 4	2	……	2	1	1	1	10
FCS	MPDU ID Block-n	……	MPDU ID Block-1	MPDUs ACKed	Max Frames	Max burst	MAC header

Figure 4-5 The structure of ACK frame in 802.15.3 MAC standard

Because it will be meaningless if these information is not received by PNC before scheduling, proposed channel time allocation period (CTAP) structure gives each transmitter DEV a fixed period of time (Dedicated CTA) to communicate with PNC, as shown in Figure 4-6. In this period, DEVs are allowed to transmit the first fragmented MPDU of new arrival data, and then IP-ACK will enable PNC to rapidly update stream priority information and make the best schedule toward the rest fragmented MPDUs. Next, PNC should have the ability to monitor current video stream queue size. In [20], the authors use MCTA for each stream which is in the process of communication at the end of the superframe. Each transmitter should report current queue size to PNC during MCTA. This method will be able to inform PNC about the change of queue size most instantaneously. Finally, below is our proposed CTAP structure.

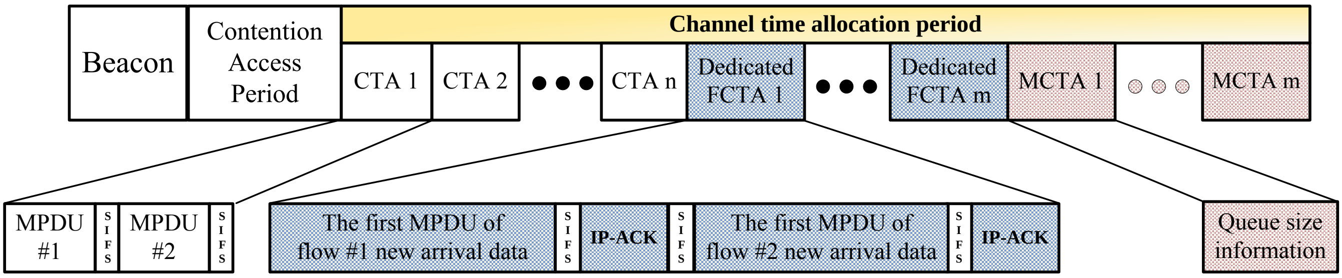

Figure 4-6 Proposed Channel time allocation period structure

Chapter 5
Simulation Results

This chapter describes the simulation results in order to examine the performance of the algorithm presented above. First, we introduce the simulation model and the simulation parameters. Then the performance metrics and achieved results are described.

5.1 Simulation Model

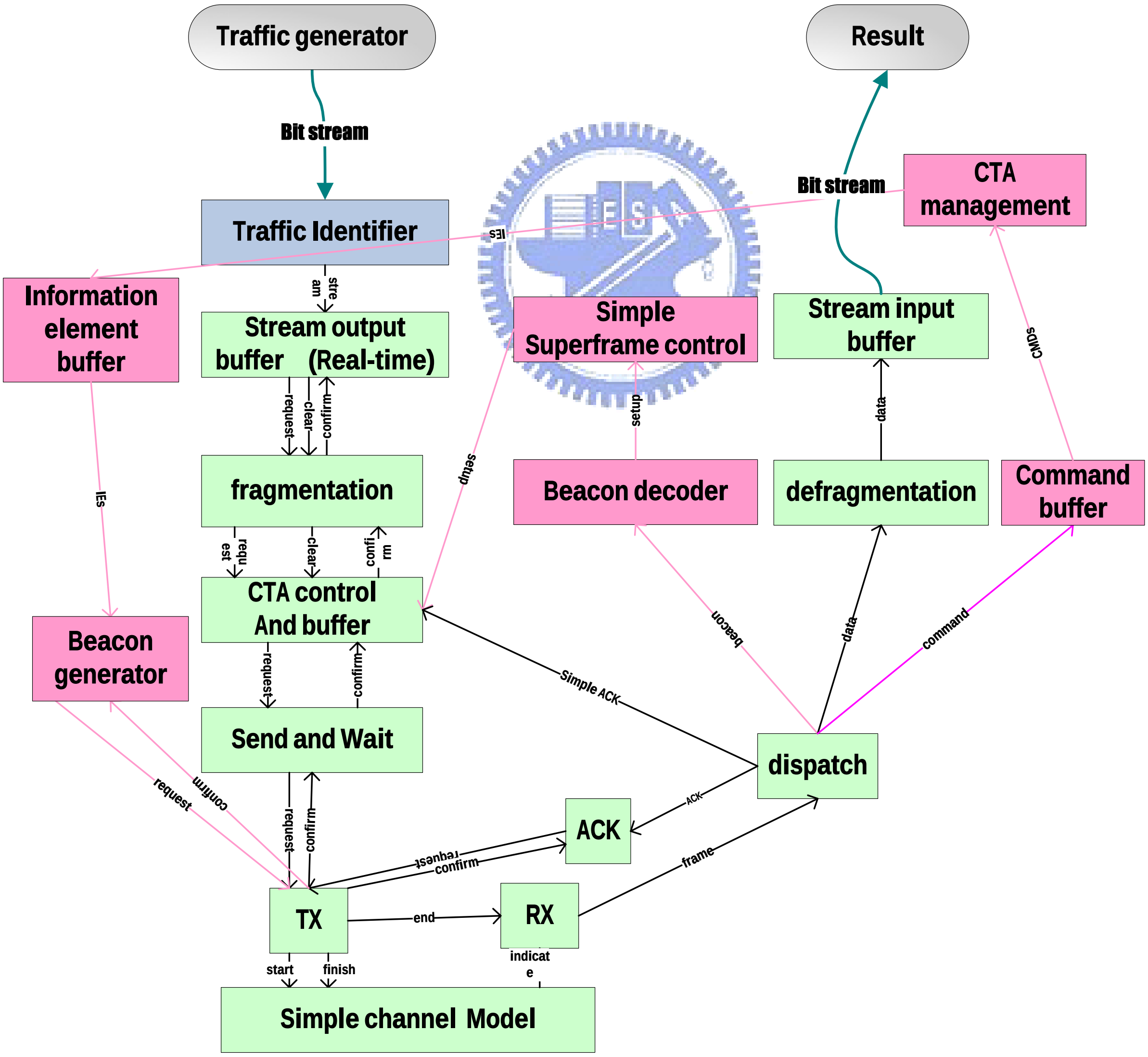

Figure 5-1 Simulation model

Figure. 5-1 shows the simulation model we applied. To evaluate the performance of the real-time service our simulation experiment, we use the traffic generator which is built referring to the specification defined by WINNER. In order to see how the proposed algorithm works toward various kinds of traffic and different types of frame, we choose three kinds of traffic which are shown in table 5-1 and compare three existing methods which are fixed channel time allocation (FCTA), FIFO and traditional EDF. As described in section 2.4.2.1, VoIP traffic which has highest serving priority has a strict requirement in delay. Type-1 and type-2 traffic represent different resolutions of MPEG traffic, which have the same arrival frequency but different data rates.

Table 5-1 Traffic parameters

	Resolutions	**Codec**	**R_F (Hz)**	**R_c (Mb/sec)**	**GOP**	**DB (ms)**
VOIP	VOIP		50	0.008		20
Type-1	HDTV (1028i) (1920*1080 pixels)	MPEG-2	24	13.2	(6,2)	500.042
Type-2	HDTV (720p) (1280*720 pixels)	MPEG-2	24	4.180	(6,2)	500.042

After traffic has been generated, the data will pass through the traffic identifier and the result of priority level will be added on its MAC header. Next, the classified data will be put into the stream output buffer. The CTA control block contains the function of controlling when the CTA belongs to this DEV should begin and finish, so that the DEV can transmit in the given duration without interfering with other DEV's transmission. To focus on the performance of the access mechanism only, the wireless channel is assumed to be ideal which means there is no distortion, noise, or other interference for data transmission. Once receiving a data from the channel, the decoder will identify which type of data it is. If it is a beacon, the decoder will send the timing information to CTA control. If it is a traffic data, it will be de-fragmented and put into the receive data buffer. If it is send back information and the receiver is PNC, this information will be sent to the CTA management block to help PNC to allocate and schedule the CTA.

The system parameters which are listed in Table 5-2 are set according to 802.15.3 standard. The simulation is run for 200 seconds.

Table 5-2 System parameters

Parameter	Value
TCP / MAC header	20 /10 byte
MPDU payload	1024 byte
Transmission rate	110 Mbps
Superframe duration	20 ms
CTAP duration	18 ms
Simulation time	200 sec
SIFS / BIFS	5 / 8.65 us

5.2 Simulation Scenario and Performance Metrics

There are two scenarios we use to create mixed traffic environment and to see how proposed algorithm affect the simulation result.

- Scenario 1: There are one VOIP and one type-2 traffic as a beginning. Then the type-1 traffic will be continuously added in the network.
- Scenario 2: There are one VOIP and one type-1 traffic as a beginning. Then the type-2 traffic will be continuously added in the network.

We will compare our proposed algorithm with fixed channel time allocation (FCTA), FIFO and traditional EDF. The performance metrics used in our simulation are defined as follows:

- ***Frame error rate (FER)***: The frame error ratio is defined as the number of

MAC packet data units (MPDUs) dropped divided by the total number of MPDUs. Since we assume the channel is error free, the failure of the packet transmission is caused only when the MPDUs exceed their delay bound.

- ***Job failure rate (JFR)***: The job failure ratio is defined as the probability of image frames failed to reconstructed**.** The image frame is successful transmitted only when no MPDUs fragmented by this frame are failed. JFR is stricter than FER, which represents true performance of streaming services.
- ***Number of error image frames***: This metric is defined as the total number of failure I frames and total number of failure B and P frames. Because our proposed algorithm has different priorities on different image frames in the same traffic flow, this metric will help us to understand how this policy works.

5.3 Simulation Results

Scenario 1 :

Fig. 5-2 shows the FER of VoIP traffic under mixed traffic scenario 1. The VoIP FER of FIFO declines rapidly due to its lack of priority concept. High constraint but small data amount traffic such as VoIP will easily be encumbered by other traffic. Traditional EDF method is similar to FIFO, which has its own limit. The FCTA performance is good because the resource got by VoIP is fixed. The VoIP traffic won't be encumbered by other traffic. However, the FCTA method will fail to serve the large overhead traffic such as type-1 and type-2 owning to lack of flexibility. We will prove below. As shown, our proposed algorithm seldom fails due to exceed delay bound, because the setting of TW makes sure VoIP traffic which has highest priority will be served first.

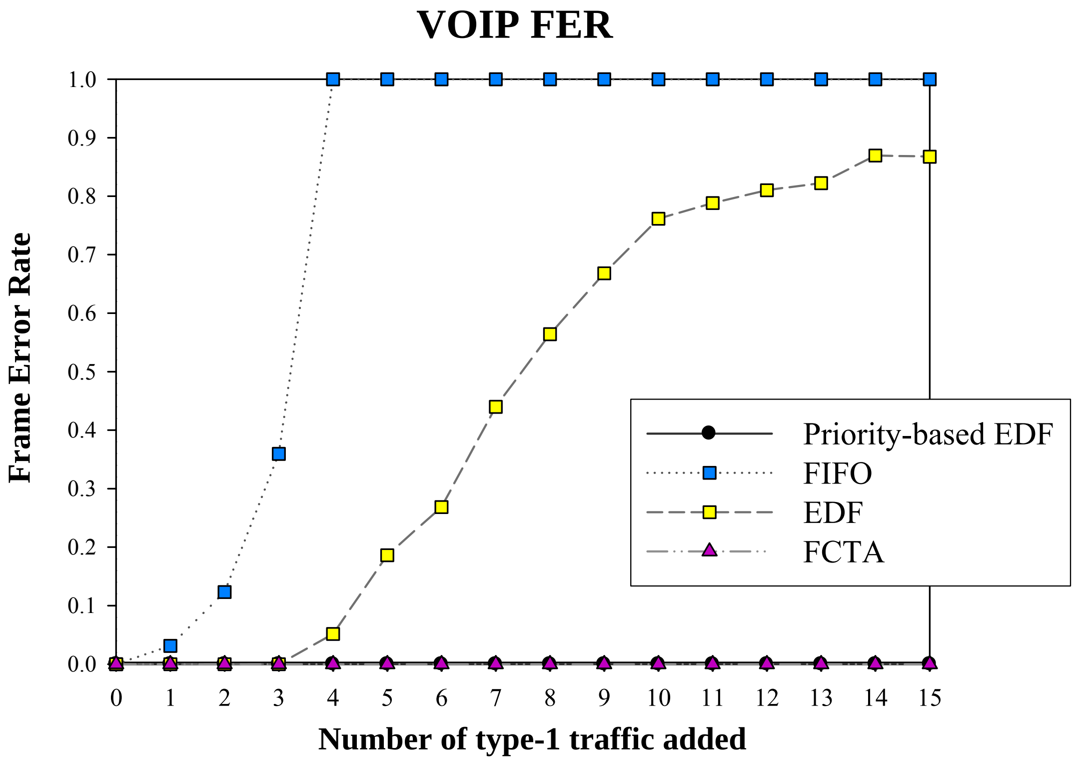

Figure 5-2 FER of VOIP traffic in scenario 1

Fig. 5-3 and Fig. 5-4 is the result of type-1 traffic FER and JFR verse type-1 flow count. Considering the operation range of MPEG, which means that the JFR is below 20%, priority-based EDF can support 4 type-1 traffic and other's can support most 3 type-1 traffic. As shown, the type-1 FER and JFR of FCTA increases with the increase of the traffic load. The reason is the resource each DEV can get is an inverse proportion to total DEV number. Thus FCTA method lacks flexibility to support this kind of high requirement and high loading traffic such as type-1. In spite of type-1 traffic has highest data rate which represents lowest priority in our proposed algorithm, the outcome of JFR is still relatively low to other methods. Comparing with traditional EDF, we can discover that there are no big differences between their FER performances. When four type-1 traffic flows are added, the FER of traditional EDF is 14.16% and the FER of proposed algorithm is 14.37%. Although traditional EDF has lower FER, the JFR of traditional EDF which is 30.61% is much higher than JFR of proposed algorithm which is 16.41%. The reason is shown in figure 5-5. We have discovered although error number of P and B frames under proposed algorithm is more than the result under traditional EDF scheme, the correct probability of I frames under proposed algorithm is much better. Due to the setting of FW which makes scheduler adjust scheduling decision according to importance level of different frame types, the JFR of proposed algorithm has significant improvement. To sum up, the error number of I frame dominates overall performance.

Fig. 5-6 and Fig. 5-7 are the result of type-2 traffic FER and JFR over increasing type-1 flow counts. It has a similar outcome with VoIP. Even though the data rate of type-2 is lower than type-1, which is easier to serve, if they are not given a higher priority just like the way we do in the proposed algorithm, it is predictable that it will be cumbered by the rest of traffic flows. Besides, type-2 FER and JFR of FCTA are low due to no interference by other traffic and lower QoS constraint of type-2 traffic.

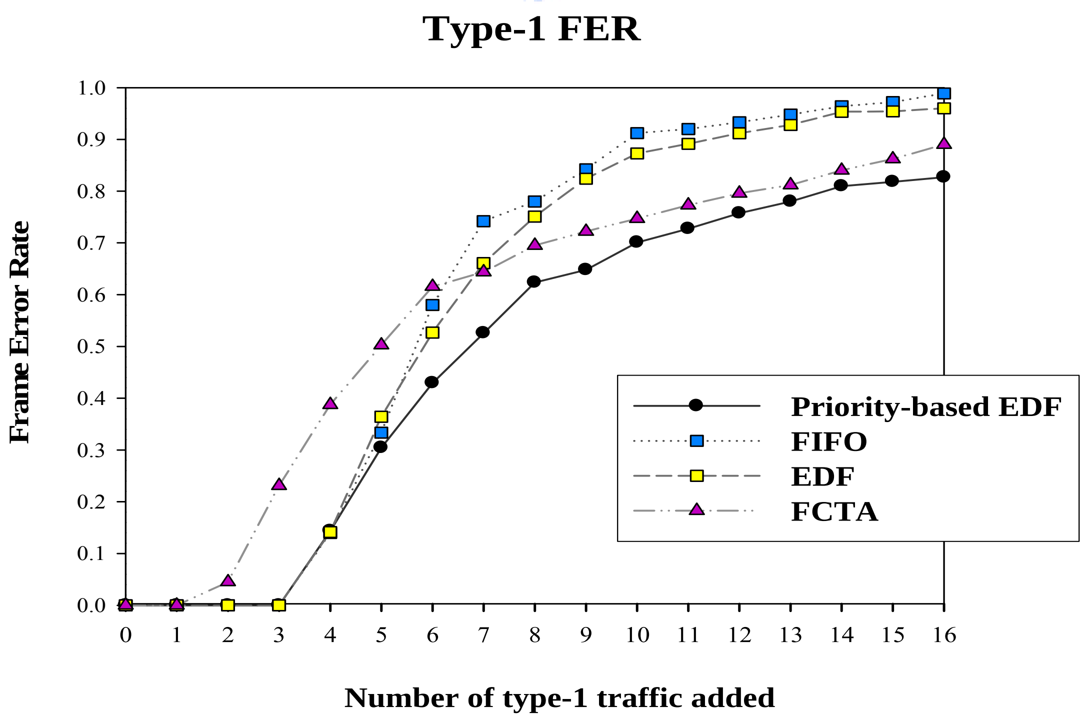

Figure 5-3 FER of type-1 traffic in scenario 1

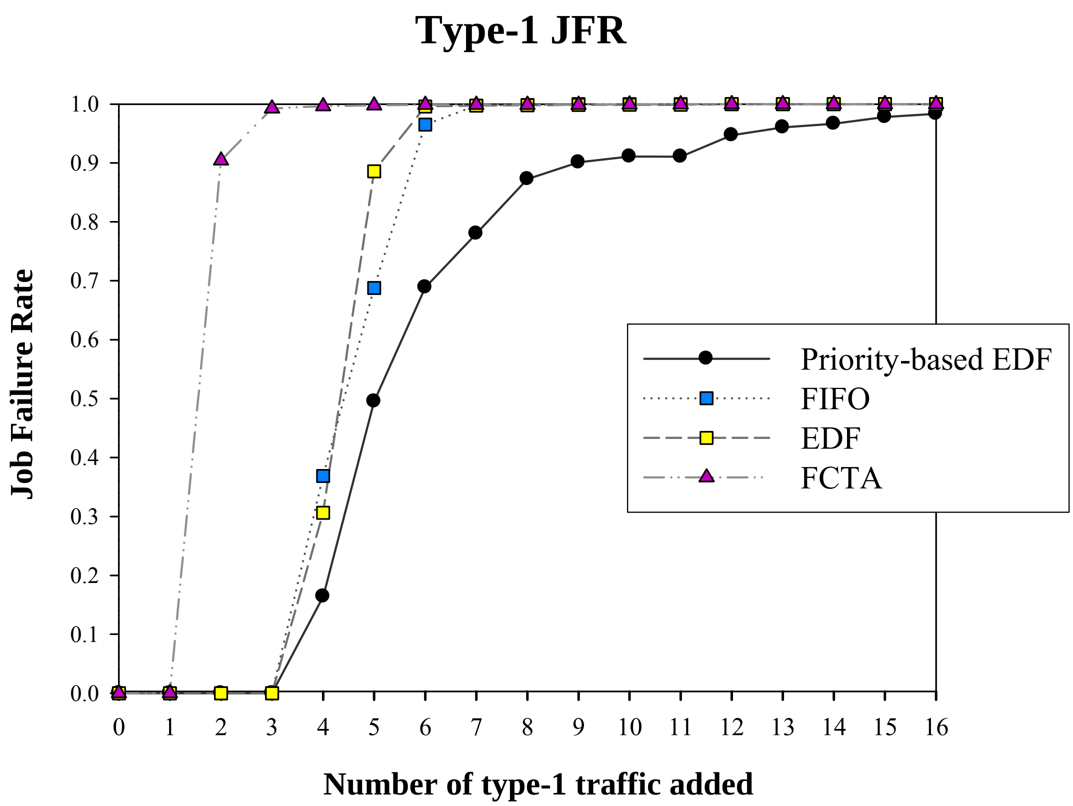

Figure 5-4 JFR of type-1 traffic in scenario 1

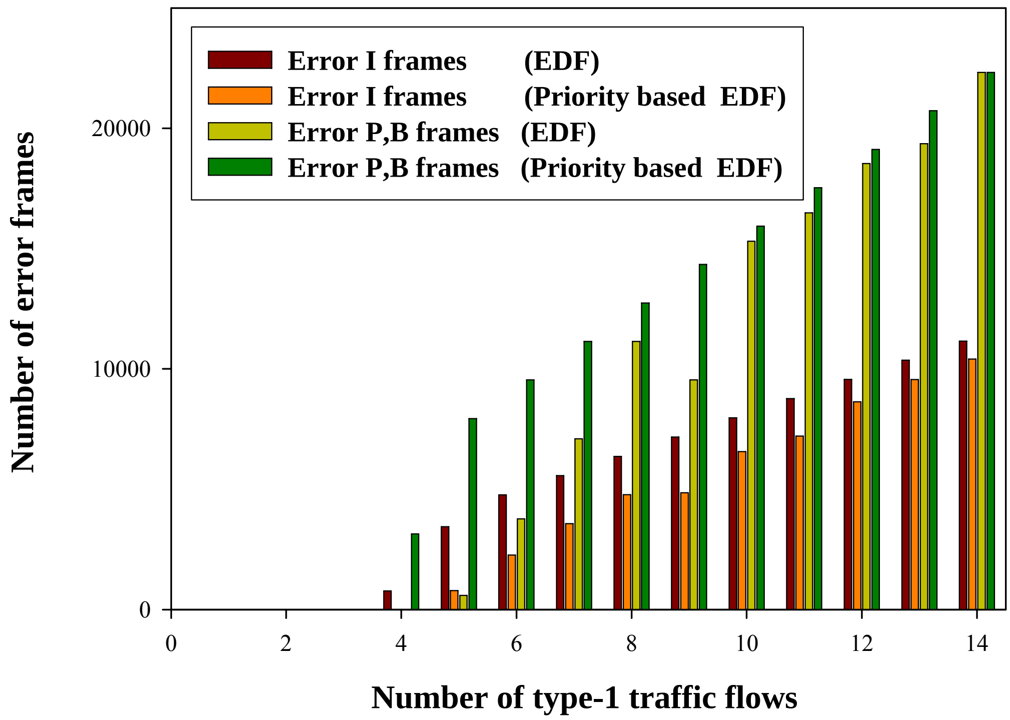

Figure 5-5 Number of error image frames in scenario 1

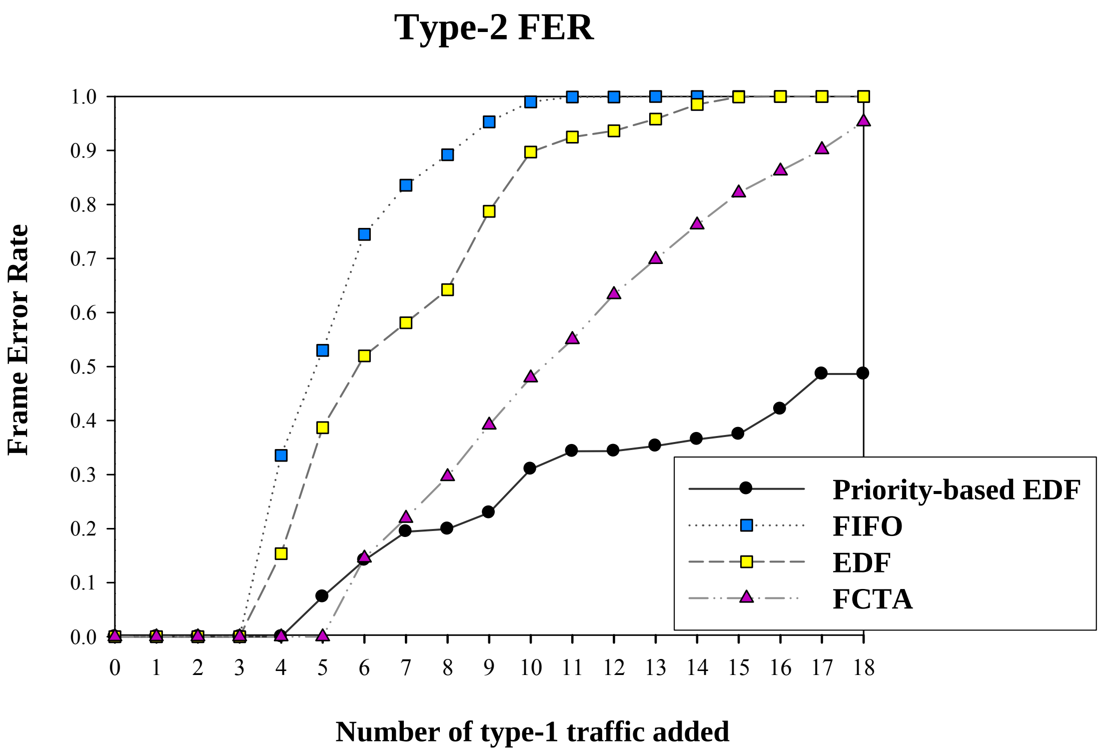

Figure 5-6 FER of type-2 traffic in scenario 1

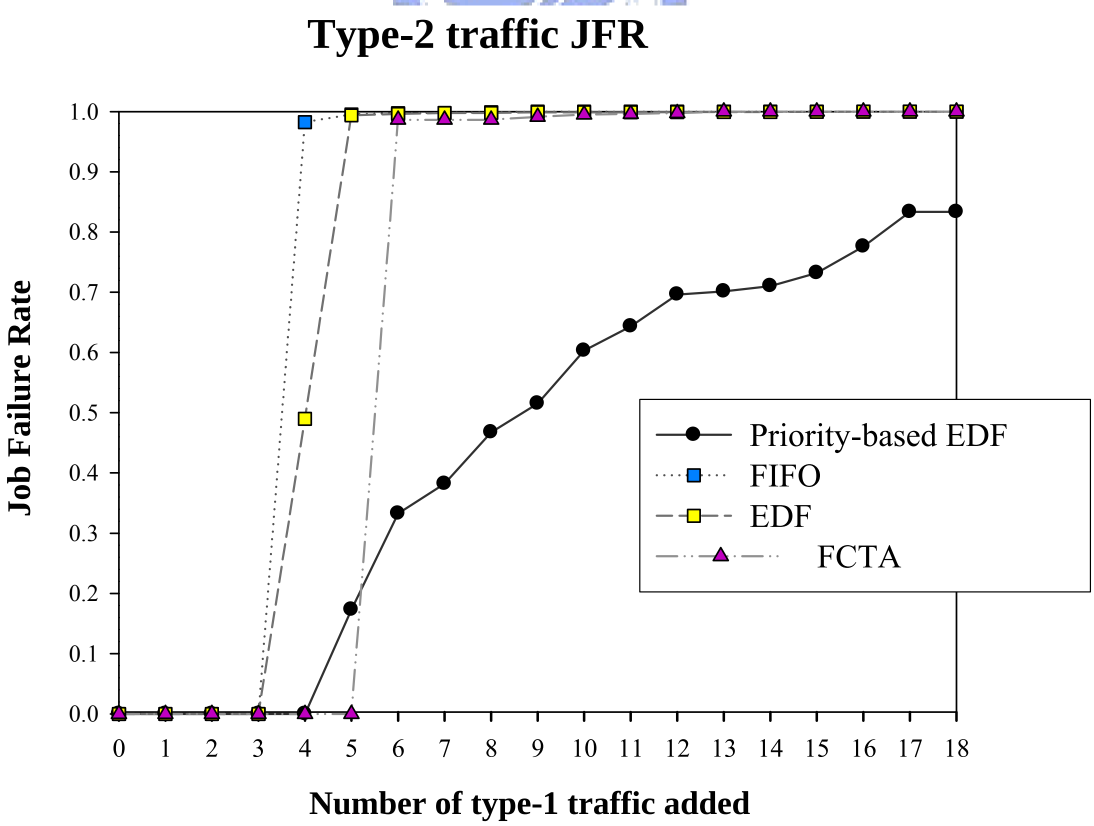

Figure 5-7 JFR of type-2 traffic in scenario 1

Scenario 2 :

Scenario 2 has a similar result with scenario 1's. Fig. 5-8 shows the FER of VoIP traffic under mixed traffic scenario 1. The performance of our proposed algorithm is still perfect because the setting of TW lets scheduler make sure VoIP traffic which has highest priority will be served first.

Figure 5-9, 5-10, 5-11 and 5-12 are the FER or JFR of type-1 and type-2 traffic under scenario 2. Considering the operation range of MPEG, priority-based EDF can support 7 type-1 traffic, which is largest number when we compare with other method. Just as the result of scenario 1, although the FER of the proposed algorithm is similar or even higher than other scheduling schemes, the JFR of the proposed algorithm is always better than others' due to the setting of TW and FW. In short, our proposed algorithm not only guarantee low JFR for high priority traffic, but also works not worse and even better than other algorithms on low priority traffic.

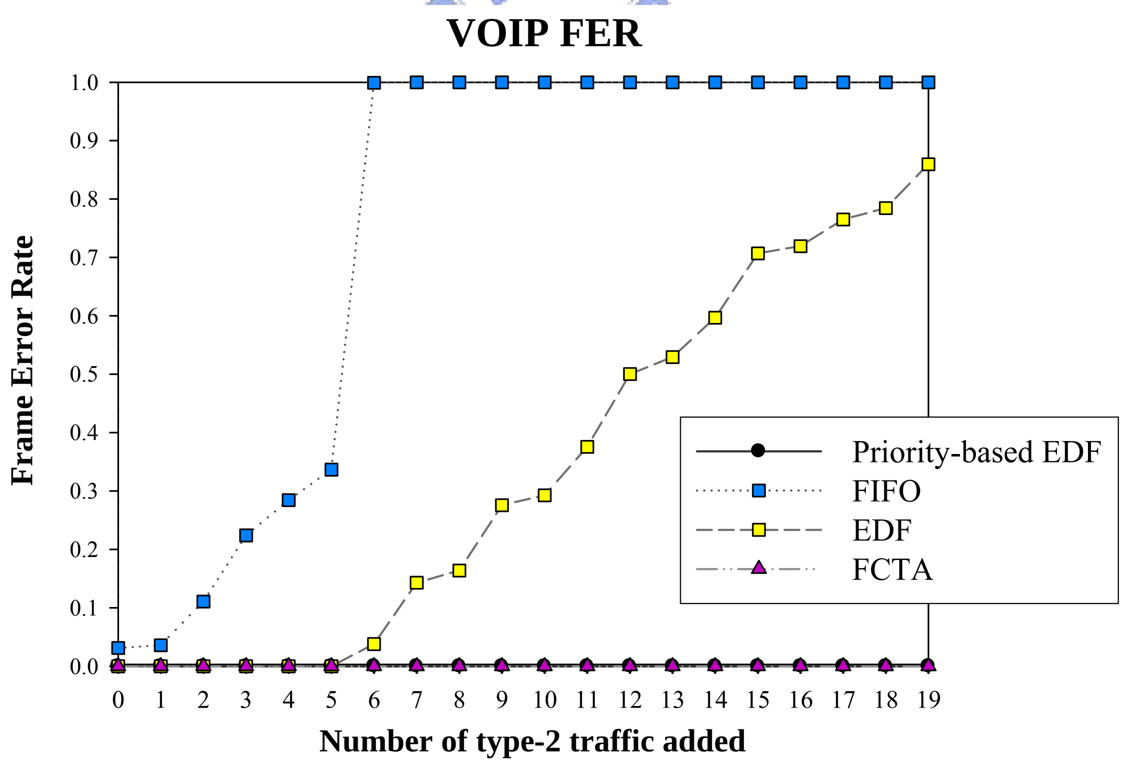

Figure 5-8　FER of VOIP traffic in scenario 2

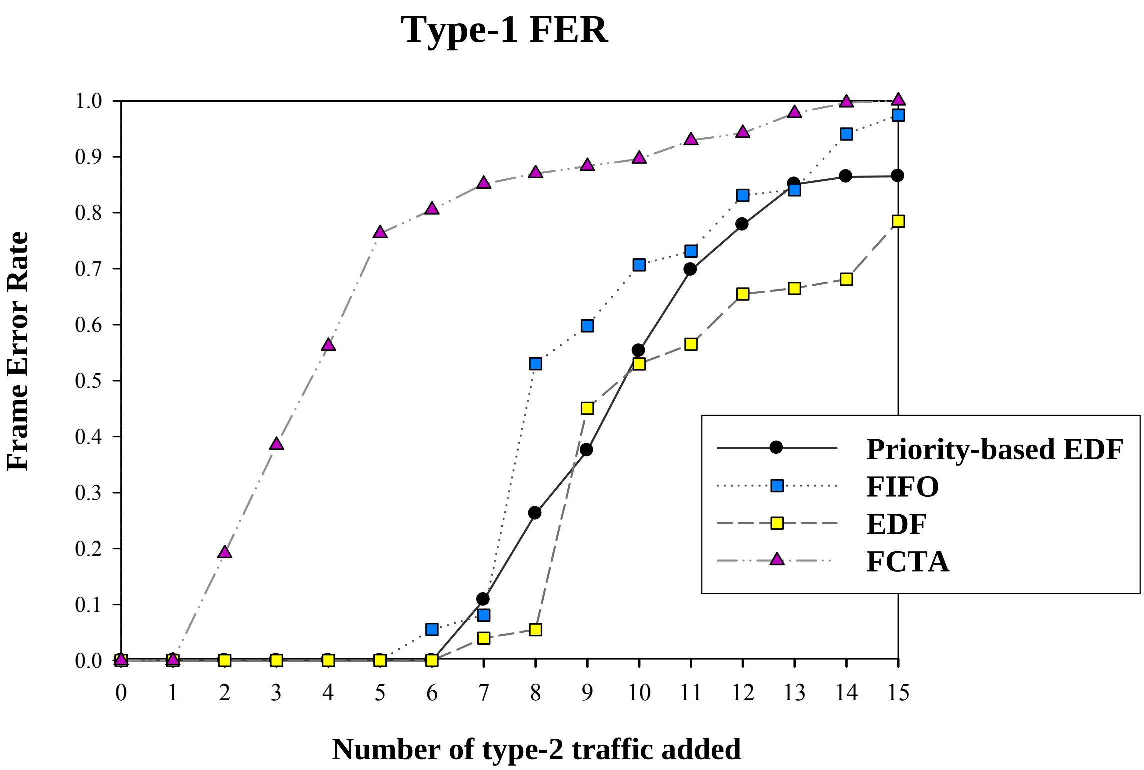

Figure 5-9 FER of type-1 traffic in scenario 2

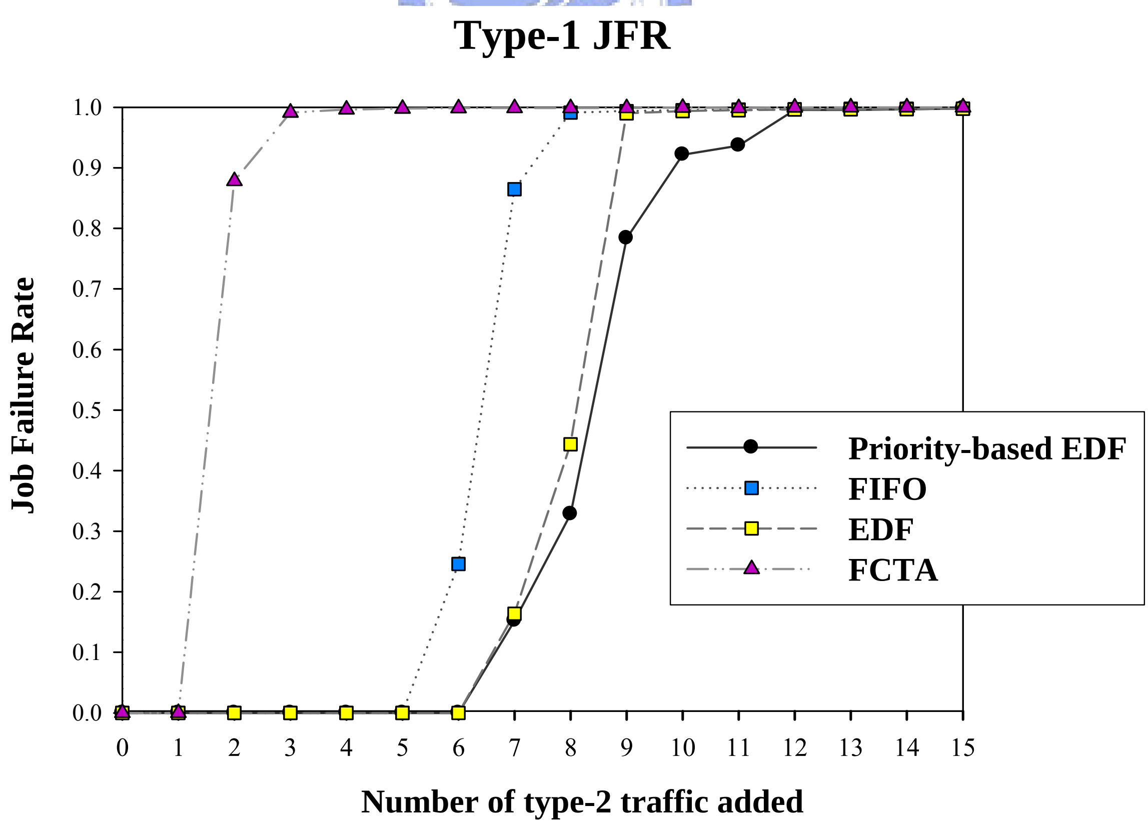

Figure 5-10 JFR of type-1 traffic in scenario 2

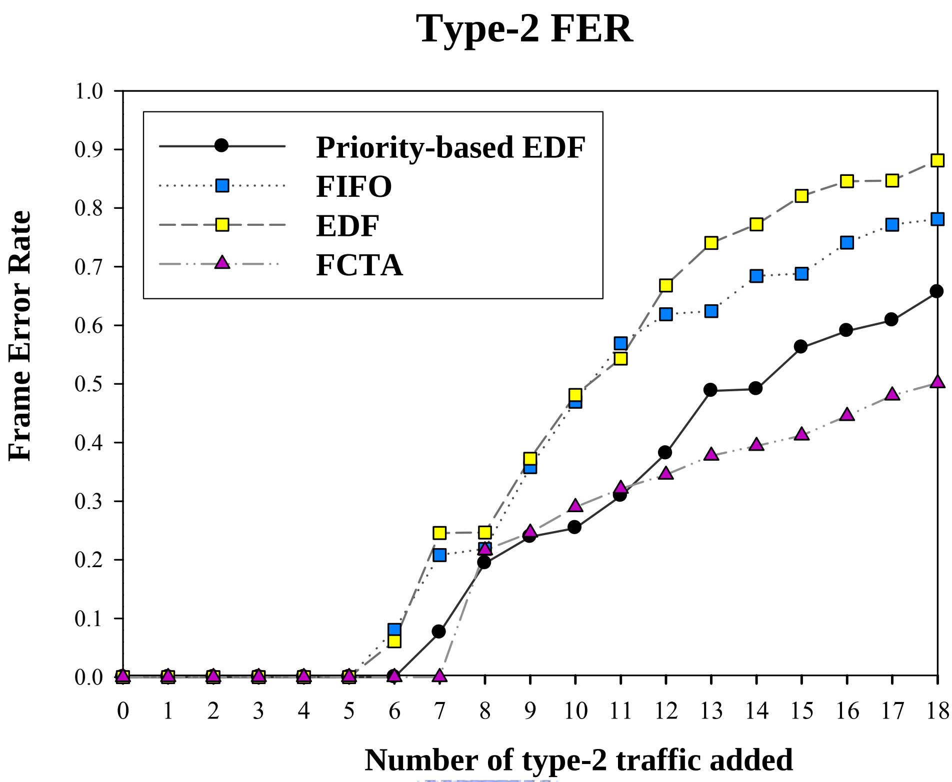

Figure 5-11 FER of type-2 traffic in scenario 2

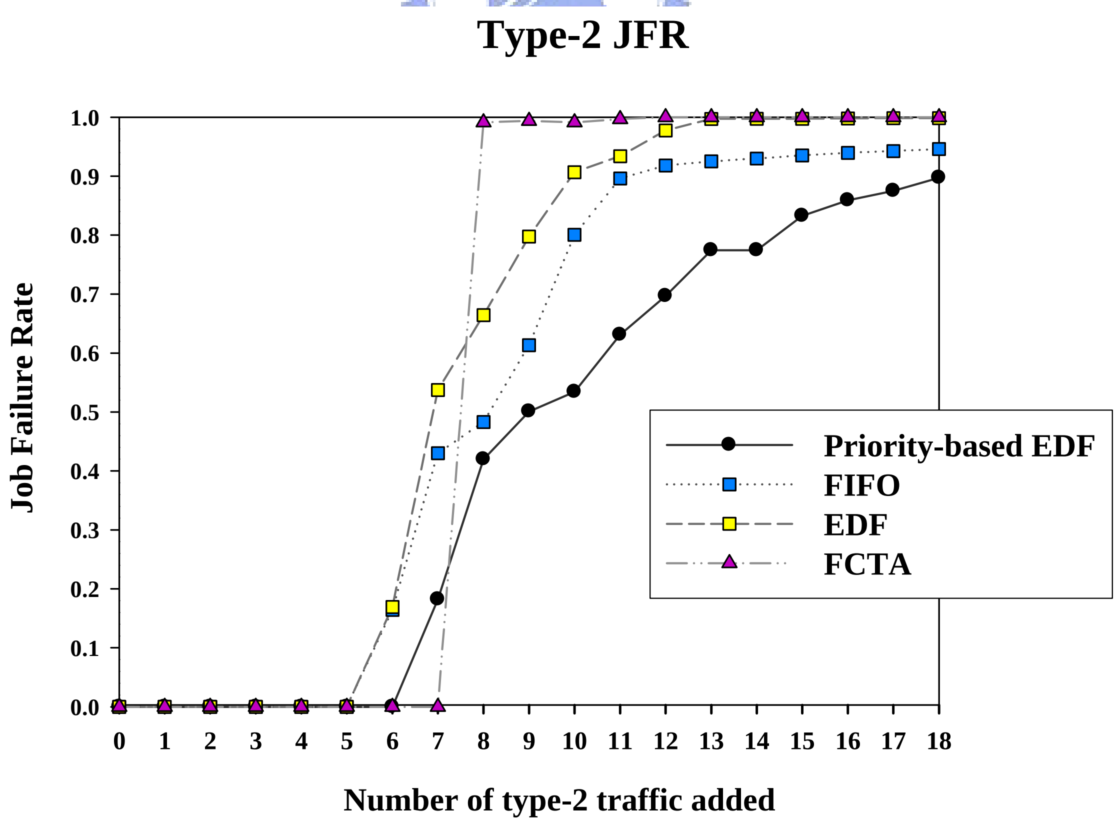

Figure 5-12 JFR of type-2 traffic in scenario 2

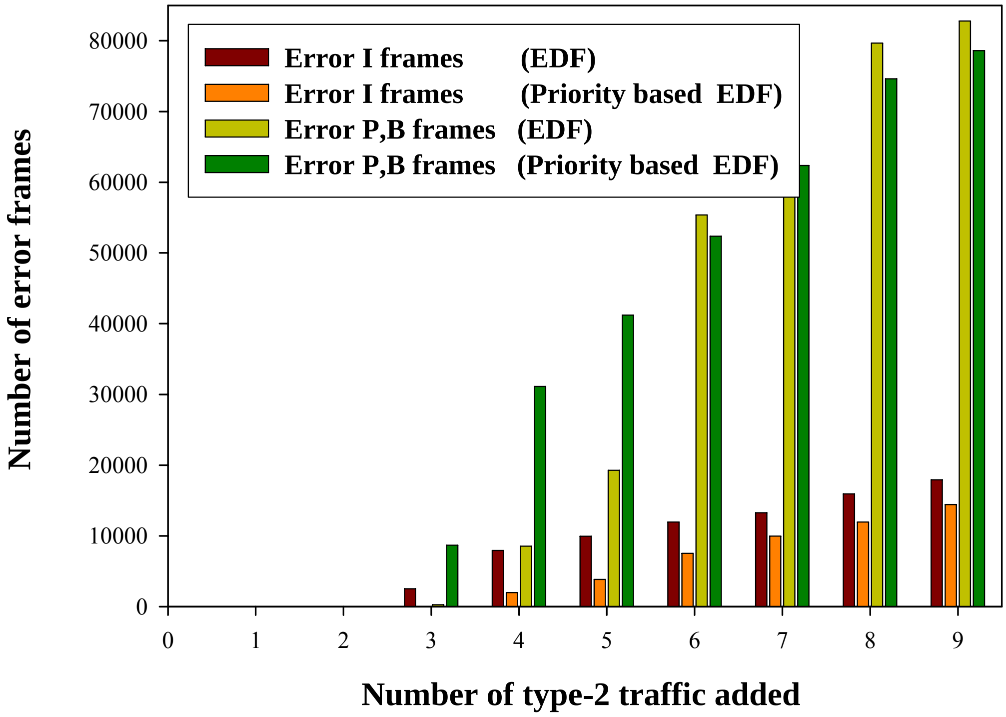

Figure 5-13 Number of error image frames in scenario 2

Chapter 6
Conclusion

In this thesis, some QoS issues of real-time service transmission were investigated. We also analyzed the advantages and drawbacks of the previous works which try to improve the MAC scheduling performance. To provide the best QoS, we proposed priority-based EDF scheduling algorithm which is based on the frame size and required transmission rate and modified MAC protocol with information gathering mechanism. Our approach begins with traffic identifying, under the premise of not to increase too much overhead and to accord with specification, we use IP-ACK as PNC information collecting tool. Then the proposed scheduling algorithm is applied for multimedia traffic after analysis. Simulation result shows when we add low priority traffic into system, proposed algorithm can not only maintain good performance of high priority traffic but also has best performance of low priority traffic. Because the priority setting according to various traffic's data rate and different frame types, we have proved that proposed algorithm not only guarantee low JFR for high priority traffic, but also works not worse and even better than other algorithms on low priority traffic. Most commendably, our proposed algorithm and modified MAC protocol are very simple and realistic, which are very adaptable for any central -controlled-based communication system.

Reference

[1] IEEE 802.15.3 Working Group, “Part 15.3: Wireless medium access control (MAC) and physical layer (PHY) specifications for high rate wireless personal area networks (WPAN),” IEEE Standard, Sept. 2003.

[2] A. Batra, J. Balakrishnan, A. Dabak, R. Gharpurey, P. Fontaine, J. Lin, J. M. Ho, S. Lee, “Physical Layer Submission to 802.15 task group 3a: Time-Frequency Interleaved Orthogonal Frequency Division Multiplexing,” Texas Instrument, Mar. 2003

[3] J. Foerster, E. Green, S. Somayazulu and D. Leeper, “Ultra wideband technology for short or medium range wireless communications,” in Intel technology journal, 2nd quarter, 2001

[4] M. Z. Win and R. A. Scholtz, “Impulse radio: How it works,” IEEE Commun. Lett., Feb. 1998, Vol. 2, pp. 36-38

[5] IEEE, “802.11e. Draft Supplement IEEE LAN/MAN Specific Requirements - Part 11: Wireless MAC and physical layer specifications: Medium Access Control (MAC) Enhancements for Quality of Service (QoS)”, Draft 3.0, May 2002

[6] X. Wang, Y. Ren, J. Zhao, Z. Guo, R. Yao “Comparison of IEEE 802.11e and IEEE 802.15.3 MAC,” Emerging Technologies: Frontiers of Mobile and Wireless Communication, 2004. Proceedings of the IEEE 6th Circuits and Systems Symposium, Vol.2, June 2004, pp.675-680

[7] IEEE, "IEEE Standard for Local and Metropolitan Area Networks Part 16: Air Interface for Fixed Broadband Wireless Access Systems", IEEE Std 802.16-2004 (Revision of IEEE Std 802.16-2001), 2004 Page(s):0_1 - 857

[8] IEEE, "IEEE Standard for Local and metropolitan area networks Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems Amendment 2: Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Bands and Corrigendum 1",IEEE Std 802.16e-2005 and IEEE Std 802.16-2004/Cor 1-2005 (Amendment and Corrigendum to IEEE Std 802.16-2004), 2006 Page(s):0_1 - 822

[9] Mong-Fong Horng, Wei-Tsong Lee, Kuan-Rong Lee and Yau-Hwang Kuo, "An Adaptive Approach to Weighted fair Queue with QoS Enhanced on IP Network", IEEE Catalogue,2001

[10] Chin-Chang Lil, "Proportional Delay Differentiation Service Based on Weighted Fair Queuing", IEEE Computer Communications, 2000

[11] Mehdi Kargahi , "Non-Preemptive Earliest-Deadline-First Scheduling Policy: A Performance Study", IEEE MASCOTS'05,2005.

[12] Yulong Fan and ChingYao Huang, "Real-Time Traffic Scheduling Algorithm in WLAN", IEEE 4GMF'2005, 2005

[13] Sanjay Shakkottai and R.Srikant, "Scheduling Real-Time Traffic With Deadlines over a Wireless Channel", Wireless Networks, 2002.

[14] Rahul Mangharam, Mustafa Demirhan, "Size Matters: Size-based Scheduling for MPEG-4 over Wireless Channels," SPIE Conference on Multimedia Computing and Networking 2004, Jan.2004

[15] F. Fitzek, M. Reisslein. "MPEG-4 & H.263 Video Traces for Network Performance Evaluation," IEEE Network, 15: 40-54,2001

[16] K. Nichols, S. Blake, F. Baker, D. Black, "Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers", RFC 2474, December 1998

[17] S. Blake, et al, "An Architecture for Differentiated Services" RFC 2475,

December 1998.

[18] IST-2003-507581 WINNER D1.2 “Intermediate requirements per scenario” ,02/03/2005

[19] Byung-Seo Kim, Yuguang Fang and Tan F.wong, ” Rate-Adaptive MAC protocol in High-Rate Personal Area Networks”, IEEE Communication Society, WCNC 2004

[20] Xin Liu, Qionghai Dai, Qiufeng Wu, “Scheduling Algorithms analysis for MPEG-4 traffic in UWB”,IEEE VTC2004,200XtremeSpectrum, “Trade-off analysis (802.11e versus 802.15.3 QoS mechanism) white paper,” July. 2002

[21] Frank H.P. Fitzek, Martin Reisslein, “MPEG-4 and H.263 Video Traces for Network Performance Evaluation”, IEEE Network Vol. 15, No. 6, pp 40-54, Dec. 2001 http://www-tkn.ee.tu-berlin.de/research/trace/trace.html

[22] S.H. Rhee, K. Chung, Y. Kim, W. Yoon, K.S. Chang, “An application-aware MAC scheme for IEEE 802.15.3 High-Rate WPAN,” WCNC 2004, IEEE Vol.2, Mar. 2004

[23] R. Mangharam, M. Demirhan, “Performance and simulation analysis of 802.15.3 QoS,” IEEE 802.15-02/297r1, Jul. 2002

[24] Kwan-Wu Chin, Darryn Lowe, “A simulation study of the IEEE 802.15.3 MAC,” Australian Telecommunications and Network Applications Conference (ATNAC), Sydney, Australia, Dec. 2004

[25] Jeffrey M. Capone, loannis Stavrakakis, “Determining the Call Admission Region for Real-lime Heterogeneous Applications in Wireless TDMA Networks”, IEEE Network March/April 1998, 1998

[26] Mahmoud Naghshineh, Mischa Schwartz, “Distributed Call Admission Control in Mobile/Wireless Networks”, IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, VOL. 14, NO. 4, MAY 1996

[27] Shigeki Shiokawa, Shuji Tasaka "Bandwidth Allocation with Enhanced Priority-Based Control among Multimedia Components in Mobile Multimedia Networks", IEEE PIMRC 2000, 2000