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An Efficient Bandwidth Management Scheme for the Real-Time Traffic on Multiple IEEE 802.11 WLANs

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Abstract

During the past years we have paid great attention to the wireless communication technology. As both speed and capacity of wireless media such as WLAN (Wireless Local Area Network) increase, so does the demand for supporting time-sensitive high-bandwidth applications. In such a real-time application, the message has a hard real-time constraint that it should be transmitted within a bounded delay and the loss of hard real-time message may jeopardize the correctness of the execution result or system itself. After all, a real-time message stream needs the guarantee from the underlying network that its time constraints are always met. In this thesis, we design and analyze an efficient bandwidth management scheme for the real-time traffic on multiple IEEE 802.11 WLANs.

This thesis consists of four main parts: bandwidth management scheme, multicast error control scheme, handoff scheme, and dual channel network management scheme. In the first part, we begin by proposing and analyzing bandwidth allocation scheme. Bandwidth allocation scheme generates efficient round robin polling schedule represented as a capacity vector by directly considering the deferred beacon problem caused by the intervention of non-real-time messages.

The second part of this thesis proposes and analyzes the performance of an

error control scheme for multicast streaming over IEEE 802.11 WLAN. This scheme makes all packets include the field indicating the total number of packets of the message to which they belong, the receiver nodes report errors in a best-effort manner through contention period, and finally access point retransmits the requested packets through the overallocated slot that is indispensably brought by a real-time QoS (Quality of Service) guarantee.

The third part proposes and analyzes a fast handoff scheme that exploits the overallocated bandwidth inevitably generated to guarantee the QoS requirement of real-time multimedia stream on the IEEE 802.11 WLAN. By using the reserved but not used network time as well as making the priority of the probe frame higher than any other data frames, AP (Access Point) and station can exchange RTS (Request To Send) / CTS (Clear To Send) to negotiate when to send probe message, enabling AP immediately respond to the probe request with probe response message in the CFP (Contention Free Period).

The last part addresses the issues on a message scheduling, bandwidth reclaiming, and error control scheme for the distributed hard real-time communication on dual WLANs. First, by making the superframe of one network precede that of the other by half, the dual network architecture can minimize the effect of deferred beacon and reduce the worst case waiting time by half. Second, the resource reclaiming scheme reassigns unused slot time to non-real-time traffic by extending the CP (Contention Period) without violating the hard real-time guarantee, resulting in enhancement of the timeliness of real-time message transmissions and the network throughput by minimizing the bandwidth waste. To improve the probability of reclaimed slot, it rearranges the polling order according to the degree of overallocation. Third, the even partition of transmission time makes it possible to inherit the optimality of EDF (Earliest Deadline First) scheduling scheme on the dual channels based on fixed-size time slots synchronized across the two channels. Also, slot rearrangement scheme maximizes the number of switchable pairs by enforcing the coordinator to select an error-free channel, so the allocation can be switched dynamically between the channels according to the current channel status, enhancing the reliability of timely message delivery.

The key idea of bandwidth allocation scheme is to decide the polling schedule for CFP in the form of capacity vector for the given network parameters and time constraints, after analyzing the effect of deferred beacon to the hard real-time message stream. The experiment demonstrates that the proposed bandwidth management scheme improves the schedulability by up to 18% for the experimental real-time stream sets. Error control scheme can not only efficiently utilize the network bandwidth by reusing the wasted bandwidth but also reduce the deadline miss ratio by 23% compared with the ordinary error control scheme without affecting other streams. The proposed handoff scheme focuses on the effect of the amount of overallocation and average number of simultaneous requests, and the result shows that the proposed scheme reduces the AP scan time maximally by 16%. Also, dual channel network scheme improves the schedulability by 36% for real-time messages and allocates 9% more bandwidth to non-real-time messages by enhancing achievable throughput, compared with the network whose bandwidth is just doubled. Bandwidth reclaiming scheme is able to reclaim up to 52.3% of bandwidth waste and also provides stable throughput throughout the utilization from 0.5 to 0.8. Using the even partition of transmission time, the rearrangement scheme can generate 70% of switchable pairs even when the utilization reaches the saturation point, improving the ratio of successful transmission by up to 12.5% when the packet error rate exceeds 0.4.

Keywords: IEEE 802.11 WLAN, Real-time communication, Message Scheduling, Bandwidth management, Bandwidth allocation, Error control, Handoff scheme, Dual channel network, Multiple network, Deferred beacon problem, Real-time message, Non-real-time messages

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Chapter 1

Introduction

1.1 Motivation

Wireless communication technology is gaining a wide-spread acceptance for distributed systems and applications in recent years [Carley et al. 2003]. In the mean time, the ongoing wireless protocols and standards offer the possibility of data transmission with very high bit rate, the convenience of having a secure and inexpensive connection without the need of cabling[Cassinis et al. 2005]. Also, many robot systems exploit WLAN as their inter-robot communication infrastructure [Cassinis et al. 2005][Potgieter et al. 2002]. First of all, the great success of IEEE 802.11 technology for WLANs is creating new opportunities for the deployment of advanced multimedia services.

One of the promising application areas of wireless technology is a wireless sensor network, where the periodically sampled data are delivered to the appropriate node within a reasonable deadline to produce meaningful data [Li et al. 2004]. In the past few years, smart sensor devices have matured to the point that it is now feasible to deploy a large, distributed network of such sensors [Madden et al. 2003]. Sensor networks are differentiated from other wireless, battery-powered environments in that they consist of tens or hundreds of autonomous nodes that operate without human interactions for a long time. The sensor network makes possible a new range of application such as environmental monitoring and control. Wireless sensor/actuator networks allow scientists to monitor remote environmental conditions such as temperature, pressure, and chemical presence. Such monitoring network can be used to detect forest fires and alert authorities, or even extinguish the fire. Besides, the application of sensor networks is being expanded to industrial, military, agricultural, and other various areas. Timely sensing and corresponding analysis is also of utmost importance in most ubiquitous applications [Rehman et al. 2004]. Message flows exchanged in a sensor network are mainly periodic and need guaranteed delay for a computing node to make a meaningful and timely decision [Choi and Shin, 2000].

In a real-time system, the current system context is perceived by one or more computing nodes to decide the appropriate control action [Muskinja, 2003]. The set of data sampled at each sensor constitutes a system context while the control decision generates a commands set to actuators. The correctness of the control decision depends not only the accuracy of the control logic but also the the timeliness of sampled data. The data are exchanged as a form of the message packet via the communication medium and the wireless network is the most promising technology for control system. Communication between the sensors and sinks requires a network [Carley et al. 2003]. Since sensor applications preclude the use of wired networks, wireless networks are commonly used in those applications. Wireless networks are inherently broadcast media, and all nodes in the network share one common communication media. Therefore, a method for resolving contention when multiple nodes require access to the medium is necessary. This is the purpose of a MAC (Medium Access Control) protocol [Gast, 2002].

The IEEE 802.11 MAC, a contention based medium access protocol, has been successfully deployed in WLAN and has also been implemented in many wireless testbeds and simulation packages for wireless multihop networks [IEEE, 1999]. The MAC protocol defines how and when nodes may access the medium. It must ensure that nodes share the medium in such a way that application requirements are met. Many real-time scheduling and fair packet scheduling algorithms have been developed for wired networks, however, it is not clear how well these algorithms work for wireless sensor networks where channels are subject to unpredictable location-dependent and time-varying bursty errors [Adamou et al. 2001], which make a real-time traffic application fail to send or receive some of its real-time packets. The wireless link is highly variable even over short distances due to the statistical distribution of path loss and the physical properties of propagation environment. In the presence of such unpredictable errors, real-time traffic applications cannot fully utilize the channel bandwidth assigned to them.

Also, it is widely recognized that the MAC in multihop ad hoc sensor networks is not only inefficient but also the network suffers from quality degradation and instability of the connection originated from the ad hoc mobility. A wireless ad hoc network is a self-configuring network of mobile routers and associated hosts connected by wireless links, forming an arbitrary topology. The routers are free to move randomly and organize themselves arbitrarily. Thus, the network's wireless topology may change rapidly and unpredictably. Such a network may operate in a stand-alone mode, or may be connected to the larger Internet. Minimal configuration and quick deployment make ad hoc networks suitable for emergency situations. Each sensor station has wireless communication capability and some level of intelligence for signal processing and networking of the data.

The real-time traffic will be a critical part of the wireless communication, when the traffic is transmitted to receivers using mobile devices such as PDA, telematics, and so on, with lower network and end-system costs. Unlike the general data traffic, real-time traffic is delay-sensitive and somewhat tolerant to packet loss through the use of error concealment technique. However, mobile clients suffer from quality degradation resulted from frequent handoff since each cell may cover just a small area, i.e., rooms or sections of a highway. Frequent handoffs and disconnections incur disruptions and instability of the connection between mobile host and server, even in the middle of an application session. To the worse, according to the handoff procedure defined in WLAN standard, the network connection as perceived by the application may be affected by the jittery and unpredictable handoff latencies. Such problem is particularly serious for the fast moving device. After all, the use of WLANs for the transport of the real-time traffic accompanies some problems resulted from the strict delay constraints and the inherent unpredictability of the wireless link [Majumdar et al. 2002].

Hence the time-sensitive message of real-time system has a hard real-time constraint that it should be transmitted within a bounded delay as long as there is no network error. Otherwise, the data is considered to be lost, and the loss of hard real-time message may jeopardize the correct execution of real-time control logic or system itself [Carley et al. 2003]. Accordingly, a real-time message stream needs the guarantee from the underlying network that its time constraints are always met in advance of the system operation or connection setup.

1.2 Objectives

As the objective of this thesis, the four main categories of an efficient bandwidth management scheme for the real-time traffics on multiple IEEE 802.11 WLANs include bandwidth management scheme, multicast error control scheme, handoff scheme, and dual channel network management scheme.

A. Bandwidth management scheme

Hard real-time guarantee depends strongly on the underlying polling policy, that is, how polling interval starts, how PC (Point Coordinator) picks the node to poll next and how long a node transmits for each poll. As an example, weighted round-robin scheme makes the coordinator poll each node one by one, and the polled node transmits its message for a time duration, or weight, predefined according to its traffic characteristics. While this scheme makes the guarantee mechanism simple and efficient, it suffers from poor utilization due to polling overhead as well as *deferred beacon problem* [Lindgren et al. 2001]. Deferred beacon problem means a situation that a non-real-time message puts off the initiation of some periods behind the scheduled start time. Though the maximum amount of deferment is bounded, it seriously degrades the schedulability of real-time messages.

In addition, hard real-time guarantee indispensably increases the number of unused slots as it depends on the worst case behavior. Real-time message scheduling scheme on WLAN standard should take such factors into account. The probability of beacon deferment depends on the arrival time distribution of the non-real-time message. This probability gets higher as more non-real-time traffic flows in the network, since a non-real-time message longer than remaining CP (Contention Period) time occupies the network more likely. Regardless of how often the beacon is delayed, any deferment can do harm to the hard real-time message that cannot tolerate any deadline miss unless it is caused by the network error.

Several MAC protocols have been proposed to support the hard real-time communication over a wireless channel, but they cannot be easily applied to the IEEE 802.11 WLAN standard, as they ignored the CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) part defined as mandatory in the WLAN standard, or just aimed to enhance the ratio of timely delivery for soft multimedia applications [Mao et al. 2005]. To enhance the schedulability and network utilization of IEEE 802.11 WLAN, we are to propose a bandwidth allocation scheme that decides the capacity vector for the round robin style polling mechanism. It cannot only make the guarantee scheme much simpler, but also cope with deferred beacon problem efficiently.

B. Error control scheme

WLAN needs to efficiently deal with network error during the delivery of sensory data. First, it is desirable to skip polling a node whose channel is not in normal condition. Second, once the packet transmission fails, it should be retried in a best-effort manner within its deadline. Third, due to the bursty nature of wireless network error, not all packets can be recovered by retransmission. In that case, the retransmission should carefully consider the priority of a packet, and network should try to enhance the successful retransmission of higher priority packets. The priority of packet can be decided by the importance of data the packet carries, or by the degree of QoS (Quality of Service) degradation the flow experienced [Adamou et al. 2001]. The bandwidth for the error control cannot be reserved respectively for each flow during PCF (Point Coordination Function), for it will result in the great waste of network bandwidth and inefficiency. In addition, it is impossible for AP (Access Point) to know which node wants to be polled for retransmission on a specific time. Hence, it is natural that error control is performed on DCF (Distributed Coordination Function), making sensor nodes to share the bandwidth.

Meanwhile, lost packets constitute one of the main causes of quality degradation, while the wireless channel error is characterized as bursty and location dependent [Shah et al. 2005]. Due to the delay constraints, the number of retransmission that can be used is limited and usually small. In case of multiple clients, each client will have different channel conditions, processing capabilities, and only limited feedback channel capabilities. The error control scheme for multicast is not aiming at recovering all lost packets but recovering as many packets as possible. Most importantly, the error control procedure should not affect the other guaranteed traffics in WLAN.

The real-time guarantee inevitably generates overallocation during PCF as the time constraints of message stream are usually described with the maximum value of message size. Retransmission via this overbooked bandwidth does not interfere the transmission of other guaranteed messages. In the other hand, if we let error report containing retransmission request be delivered via the DCF period in a best-effort manner, the entire error control procedure can be carried out without any influence to other real-time messages. Finally, though the variable message size makes it hard to decide when to report the error list, the receivers can determine the completion of message transmission by counting $Beacon$ frame AP generates periodically in WLAN. Based on the requirements described previously, this thesis will propose and analyze the performance of an error control scheme for multicast stream on IEEE 802.11 WLAN. We focus on the video streaming scenario in the last mile network, namely, between AP and the mobile devices.

C. Handoff scheme

Wireless LAN STA (STAtion) is the most basic component of the wireless network [Shin et al. 2004] and it means any device that contains the functionality of the 802.11 protocol. BSS (Basic Service Set) is the basic building block of an 802.11 WLAN, and each BSS consists of any number of stations. As a SSID (Service Set IDentifier) is a unique identifier that distinguishes one WLAN from the others, all APs and STAs attempting to join a specific WLAN must have the same SSID. A WLAN handoff is performed at the MAC layer when a mobile STA moves beyond the radio range of the current AP and enters another BSS [Mishra et al. 2003]. During the handoff, management frames are exchanged between the STA and the AP. The handoff procedure essentially requires the transfer of STA information such as authentication, authorization, and accounting information, from the old AP to the new AP.

Previous research results show that the probe phase overwhelms the total handoff latency while the variation in the probe-wait time also causes the large variations in the overall handoff latency. For the client, the service is ceased during the handoff. Because the STA must scan the channel to which an AP may belong for the maximum scanning period, and it must repeat iteratively for all channels, the probe time occupies the biggest part of the handoff latency. Thus any handoff scheme built upon the techniques/heuristics that either cache or deduce AP information without having to actually perform a complete active scanning definitely should cope with the dominating cost of the scanning process.

To solve such a problem by reducing AP scanning delay of handoff latency at MAC layer, this thesis proposes and analyzes a fast handoff scheme that exploits the overallocated bandwidth necessarily accompanied in providing QoS guarantee to real-time multimedia stream. Using the reserved but not used network time, AP and STA can exchange RTS (Request To Send)/CTS (Clear To Send) to negotiate when to send a probe message, making AP immediately respond to the *probe request* with matching *probe response* message in the next period. In addition, by making the priority of the probe frame higher than any other data frames, collision between *probe response* messages and ordinary data frames can be eliminated, and dynamic adjustment of the channel scanning time further improves AP scanning time. With the reduced AP scanning time, a seamless handoff process is performed, minimizing the deadline miss ratio.

D. Dual channel network schemes

In order to overcome poor schedulability, we are to propose a network architecture and corresponding network access scheme based on dual wireless LANs.

As the wireless channel supports multiple frequency bands, it is not unusual for a group of components belonging to a common control loop to be linked by two or more networks. The dual link system can reduce the worst case waiting time for each node to be blocked until it can send its own message, improving the guarantee ratio for real-time messages as well as allocating more network bandwidth to non-real-time messages. In addition, this thesis also designs a resource reclaiming scheme that improves network utilization by reallocating the network bandwidth unused by the real-time messages to non-real-time one.

In addition, the wireless network has an advantage that it can be easily duplicated, or a cell is able to operate dual channels. The dual channel networks have doubled network bandwidth, so intuitively such a network should be able to accommodate twice as much real-time traffic as a single network. However, real-time scheduling for dual or multiple resource system is known to be an NP-hard problem [Carley et al. 2003], while the uniprocessor system has optimal scheduling solutions such as RM (Rate Monotonic) for static scheduling as well as EDF (Earliest Deadline First) for dynamic scheduling. Applying RM or EDF method to multiple resource system is not optimal in scheduling preemptable jobs due to its work conserving nature [Liu, 2000]. However, the network transmission has no data dependency between each message, so the optimality of EDF scheme can be preserved also for the dual networks by evenly partitioning the traffic requirements rather than the message streams themselves.

Moreover, the dual channels can efficiently cope with network errors without violating the time constraints of messages, if the transmission order is rearranged so that the same stream is not scheduled on the time slots that cobegin at both channels. If the simultaneous slots are allocated to different stations, the allocation can be switched between the channels when the coordinator cannot reach the station via the originally scheduled channel. With these assertions, we are to propose and analyze the performance of a bandwidth allocation scheme for real-time sensor messages on the dual channel wireless networks, aiming at keeping the optimality of EDF scheduling scheme as well as maximizing the capability of coping with wireless channel errors [IEEE, 1999].

1.3 Organization

The rest of this thesis is organized as follows:

Chapter 2 introduces the background of this thesis, including the research issues and the related works focusing on real-time communications on the wireless medium.

Chapter 3 proposes the bandwidth allocation scheme with the description on network and message model for IEEE 802.11 WLAN based on the round-robin polling policy, and discusses the performance measurement result.

Chapter 4 proposes the multicast error control scheme with the communication architecture for time-sensitive sensor traffic, describing channel estimation, error report and retransmission scheme and demonstrates the simulation result.

Chapter 5 proposes the fast handoff scheme in IEEE 802.11 WLAN based on proposed bandwidth management scheme with the description on the handoff procedure, and discusses the performance measurement results.

Chapter 6 proposes the dual channel network architecture and its corresponding scheduling scheme, reclaiming scheme, and error control scheme, and discusses the performance measurement result.

Chapter 7 finally concludes this thesis and summarizes the result obtained from this thesis with some concluding remarks.

Chapter 2

Background

2.1 Research issues

In this section, we introduce various research issues on real-time traffics on IEEE 802.11 WLANs.

2.1.1 Sensor network

Sensor networks can be considered distributed computing platforms with many severe constraints including limited CPU speed, memory size, power, and bandwidth. Individual nodes in sensor networks are typically unreliable and the network topology dynamically changes, possibly frequently. Sensor networks can also be considered a form of ad hoc network. However, here also many constraints in sensor networks are different or more severe. Sensor networks also differ because of their tight interaction with the physical environment via sensors and actuators. Due to all of these differences many solutions developed for general distributed computing platforms and for ad hoc networks cannot be applied to sensor networks. Many new and exciting research challenges exist.

Many future applications will rely on an embedded sensor network. A sensor network is a general term that covers many variations in composition and deployment. A typical sensor network consists of a large number of nodes deployed in the environment being sensed and controlled. In many cases, each node of the sensor network consists of sensors and wireless communication. Memory, power and computational capacities are typically limited. In other sensor networks, nodes may also contain actuators. Often sensor nodes are densely deployed, are prone to failures, and the topology of the network can dynamically change. Sensor networks may consist of all homogeneous nodes or exhibit a heterogeneous structure where some nodes are much more powerful than others or contain different sets of resources. Regardless of the variant of a sensor network, it is necessary to support real-time communication and coordination.

A sensor network is subject to a great deal of uncertainty from many quarters. First, the sensor network is deployed in an environment with uncontrollable aspects. Second, the wireless communication is subject to many physical errors and missing messages due to radio interfaces of many types. Third, individual nodes are not reliable. Fourth, sensors may not all be calibrated properly. Fifth, the connectivity and routing structures are changing dynamically. Sixth, new nodes may be added or old nodes removed from the sensor network. This implies that the sum total of resource capacity is not fixed. Seven, power availability at each node can vary significantly even when initially deployed. Eight, nodes may be physically moved or be controlled to do so under their own power, thereby re-structuring the topology. And so on.

Sensor networks operate in the real world hence timing constraints are important. These systems have implicit time requirements such as when a user enters a room, he should be recognized within a very short time. The faster such a task is accomplished the better we consider the system. However, many sensor networks will also have explicit real-time requirements related to the environment. There may also be deadlines associated with end-to-end routing, for example, a sensitive pressure reading might have to periodically arrive at a monitor and actuation station on time, each time. Because of the large scale, non-determinism, noise, etc. it is extremely difficult to guarantee real-time properties.

A major requirement for sensor networks is to reliably aggregate and disseminate information within a time frame that allows the controllers to take necessary actions, even in the case of poor spatial distribution of sensor devices, wireless interference, and malicious destruction. Out-of-date information is of no use, for example, an object that was being tracked may no longer be in the vicinity when the information is received. This presents a key technical challenge in cooperative engagement - how to effectively coordinate and control sensors in real-time over an unreliable wireless ad hoc network.

2.1.2 IEEE 802.11 WLAN

Many different and sometimes competing design goals have to be taken into account for WLANs to ensure their commercial success.

- Global operation: WLAN products should sell in all countries, therefore, many national and international frequency regulations have to be considered.

- Low Power: Devices communicating via a WLAN are typically also wireless devices running on battery power. Hence, WLAN must implement special power saving modes and power management functions.

- License-free operation: LAN operators do not want to apply for a special license in order to be able to use the product. Thus, the equipment must operate in a license-free band, such as the 2.4 GHz ISM band.

- Bandwidth: Bandwidth is the one of the most scarce resource in wireless networks. The available bandwidth in wireless networks is far less than the wired links.

- Link Errors: Channel fading and interference cause link errors and these errors may sometimes be very severe.

- Robust transmission technology: Compared to wired counterparts, WLANs operate under difficult conditions. If they use radio transmission, many other electrical devices may interfere.

- Simplified spontaneous co-operation: To be useful in practice, WLANs should not require complicated setup routines but should operate spontaneously after power up. Otherwise these LANs would not be useful for supporting e.g., ad hoc meetings, etc.

- Easy to use: LANs should not require complex management but rather work on a plug-and-play basis.

- Protection of investment: A lot of money has already been invested into wired LANs. Hence new WLANs must protect this investment by being inter operable with the existing networks.

- Safety and security: Most important concern is of safety and security. WLANs should be safe to operate, especially regarding low radiation. Furthermore, no users should be able to read personal data during transmission i.e., encryption mechanism should be integrated. The network should also take into account user privacy.

- Transparency for application: Existing applications should continue to run over WLANs. The fact of wireless access and mobility should be hidden if not relevant.

The wireless LAN operates on both CP and CFP (Contention Free Period) phases alternately in BSS as shown in Figure 2.1. Each superframe consists of CFP and CP, which are mapped to PCF and DCF, respectively. It is mandatory that a superframe includes a CP of minimum length that allows at least one data packet delivery under DCF [IEEE, 1999]. PC node, typically AP, sequentially polls each station during CFP. Even in the *ad hoc* mode, it is possible to designate a specific node to play a role of PC in a target group. Only the polled node is given the right to transmit its message for a predefined time interval, and it always responds to a poll immediately whether it has a pending message or not.

The phase of network operation is managed by the exchange of control packets which have higher priority than other packets. The prioritized access is achieved by different length of IFS (InterFrameSpace) the node waits before it attempts to send its packet. The PC attempts to initiate CFP by broadcasting a $Beacon$ at

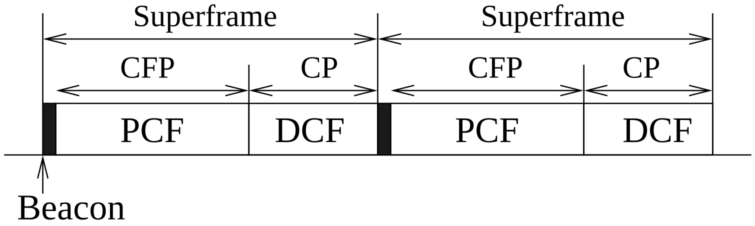


Figure 2.1: Time axis of wireless LAN

regular intervals derived from a network parameter of CFP repetition interval, *CFPRate*. The PC shall determine the CFPRate to use from the CFPRate parameter in the CF Parameter Set. The frames of different priorities have to wait different IFSs before they are transmitted. The SIFS (Short IFS) is used by immediate control frames, which always have the highest priority such as acknowledgment. The PIFS (Priority IFS) is used by real-time frames, while DIFS (DCF IFS), the longest IFS, is used by non-real-time frames [IEEE, 1999]. According to the network parameters configured in priori, the PC initiates CP and CFP by announcing a management frames such as *StartBeacon* and *EndBeacon* frames with PIFS, respectively. PC is responsible for generating beacons which generally contain a time information. Round robin is one of the popular polling policies for CFP, in which every node is polled once a polling round. A polling round may be completed within one superframe, or spreads over more than one superframe. In case the CFP terminates before all stations have been completely polled, the polling list is resumed at the next node in the ensuing CFP cycle.

The PC shall reside in the AP. It is an option for an AP to be able to become the PC. The operating characteristics of the PCF are such that all stations are able to operate properly in the presence of a BSS in which a PC is operating, and, if associated with a point-coordinated BSS, are able to receive all frames sent under

PCF control. It is also an option for a station to be able to respond to a CF-Poll (Contention-Free Poll) received from a PC. A station that is able to respond to CF-Polls is referred to as being CF-Pollable, and may request to be polled by an active PC. When polled by the PC, a CF-Pollable station may transmit only one MPDU (Message Protocol Data Unit), which can be to any destination, not just to the PC, and may piggyback the acknowledgment of a frame received from the PC using particular data frame subtypes for this transmission. If the data frame is not in turn acknowledged, the CF-Pollable station shall not retransmit the frame unless it is polled again by the PC, or it decides to retransmit during the CP. A PC may use contention-free frame transfer solely for delivery of frames to stations, and never to poll non-CF-Pollable stations. The PCF controls frame transfers during a CFP. The CFP shall alternate with a CP, when the DCF controls frame transfers. Combined CFP and CP is termed as one superframe. The CFPs shall occur at a defined repetition rate, which shall be synchronized with the beacon interval.

The PC shall send a CF-Poll to at least one station during each CFP when there are entries in the polling list. During each CFP, the PC shall issue polls to a subset of the stations on the polling list in order by ascending association id value. While time remains in the CFP, all CF frames have been delivered, and all stations on the polling list have been polled, the PC may generate one or more CF-Polls to any stations on the polling list. We have focused more on PCF mode than DCF mode. Standard specifies simple round robin scheduling. The polled node transmits its message for a predefined time interval, and it always responds to a poll immediately whether it has a pending message or not. The poll, transmission, and acknowledgment are atomic, namely, these steps must complete in their entirety to be successful. Senders expect acknowledgment for each transmitted

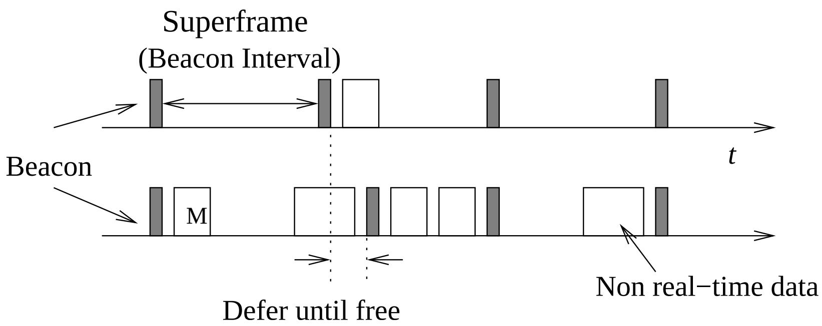


Figure 2.2: Deferred beacon problem

frame and are responsible for retrying the transmission. After all, error detection and recovery is up to the sender station, as positive acknowledgments are the only indication of success. If an acknowledgment is expected but does not arrive, the sender considers the transmission failed.

In case of an unsuccessful transmission, the station may retransmit the frame through either CP or CFP according to an control policy. As shown in Figure 2.2, the transmission of the beacon by the coordinator depends on whether the medium is idle at the time of TBTT (Target Beacon Transmission Time). Only after the medium is idle the coordinator will get the priority due to the shorter IFS. Thus the delivery of a beacon frame can get delayed if another packet is already occupying the network, invalidating the network schedule sophisticatedly determined for hard real-time messages. The systems in which the failure to observe timing constraints results in a fatal error, are called hard real-time systems [Mukherjee et al. 1993]. As can be inferred from Figure 2.2, the maximum amount of deferment coincides with the maximum length of a non-real-time packet.

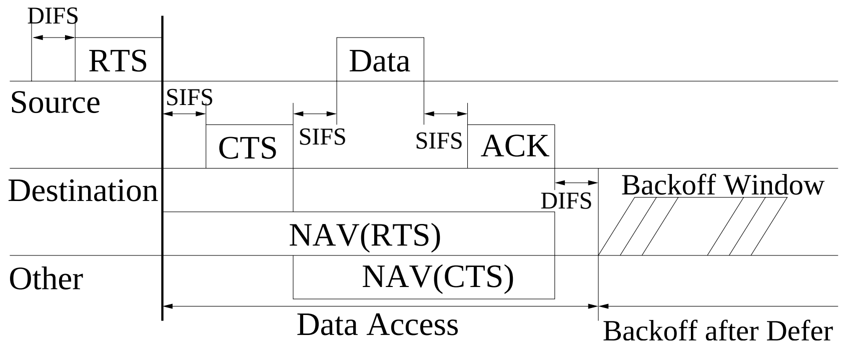


Figure 2.3: RTS/CTS frame exchange

In contrast, DCF is the basis of the standard CSMA/CA access mechanism and it uses the RTS /CTS clearing technique to further reduce the possibility of collisions, as shown in Figure 2.3. The DCF in IEEE 802.11 wireless LAN standard is the basic access method for 802.11. DCF functions as follows: Before initiating a transmission, a station senses the channel to determine whether or not another station is transmitting. If the medium is sensed idle for an specified time interval, called the DIFS, the station is allow to transmit. If the medium is sensed busy, the transmission is deferred until the ongoing transmission terminates. A slotted binary exponential backoff technique is used to arbitrate the access: a random backoff interval is uniformly chosen in [0, CW -1] and used to initialize the backoff timer, where CW is the maximum contention window. The backoff timer expires, the station attempts for transmission at the beginning of the next slot time.

Finally, if the data frame is successfully received, the receiver initiates the transmission of an acknowledgment frame is successfully received, the receiver initiates the transmission of an acknowledgment frame after a specified interval, called the SIFS, that is less than DIFS. The transmission of a frame is an atomic operation that cannot be interrupted by another packet. If an acknowledgment is not received, the data frame is presumed to be lost and a retransmission is scheduled. The value of CW is set to $CW_{min}(= 32)$ in the first transmission attempt, and is doubled at each retransmission up to a pre-determined value $CW_{max}(= 256)$. In addition to the physical channel sensing, virtual carrier sensing is achieved by using the NAV (Network Allocation Vector) fields in the packets. NAV indicates the duration of the current transmission. All nodes that hear the RTS or CTS message back off an amount of time indicated in NAV before sensing the channel again. Though PCF is optional, QoS guarantee cannot be provided without PCF. Moreover, previous researches based on the DCF can not avoid both the collision between the probe messages and normal data frames, and the probe delay according to the backoff time of DCF.

2.2 Related works

There exist a lot of works related to real-time communications. In this section, we introduce some well-known works on real-time communications on IEEE 802.11 WLANs classified by the issues which this thesis concerns.

2.2.1 MAC protocols

Several MAC protocols have been proposed to provide bounded delays for real-time messages while providing a reasonable performance for a non-real-time data over a wireless channel. However, these protocols are typically based on a frame-structured access comprised of a contention part and reservation part, demanding a time synchronization among the nodes on a network, so they make it impossible or unrealistic to apply to the IEEE 802.11 WLAN. In wireless sensor networks, the MAC performance has been predominantly measured in terms of bandwidth requirement, power consumption, contention mitigation, and support to maintain network connectivity. The latency incurred in message delivery has not been a metric to be optimized, but is likely to become increasing important as sensor networks are deployed in critical applications. Timeliness is perhaps the most difficult requirement to meet since it brings to the fore the tradeoff between power consumption, interference mitigation, and scheduling and routing efficiency. Existing MAC protocols for wireless networks can be classified into two categories: scheduling based and contention based.

A. Scheduling based MAC Protocols

In scheduling based MAC protocols, the time at which a node can transmit is determined by a scheduling algorithm, so that multiple nodes can transmit simultaneously without interference on the wireless channel. The time is usually divided into slots, and slots are further organized into frames. Within each frame, a node is assigned at least one slot to transmit. A scheduling algorithm usually finds the shortest possible frame so as to achieve high spatial reuse and thus high network utilization and low packet latency. A large amount of early work has been focused on TDMA (Time Division Multiple Access) scheduling [Li, 1991] [Stevens and Ammar, 1990] [Chlamtac and Kutten, 1985] [Cidon and Sidi, 1989] [Chlamtac and Lerner, 1987]. Most of the studies concentrated on devising fair conflict free algorithms that maximize the system throughput by using graph theory. Most of them are centralized and require global connectivity information. As a result, they cannot adapt adequately, and keep the optimality property, in highly dynamic environments such as topology change.

To resolve the above problem, Chlamtac et al. [Chlamtac and Farago, 1994] first proposed a topology-independent algorithm that depends only on global network parameters, i.e., the number of nodes and the maximum nodal degree. The algorithm ensures that for every node and for each of its neighbors, there is at least one slot assigned in each frame. The major problem associated with the above topology independent algorithms is that as nodes do not exchange their slot allocation functions, a sender node has no idea which slot is collision free, and has to transmit on all assigned slots within each frame, until its packet gets through. To deal with this problem, Shepard et al. [Shepard, 1996], Rozovsky et al. [Rozovsky and Kumar, 2001], and Bao et al. [Bao and Garcia-Luna-Aceves, 2001] proposed collision-free channel access schemes in which each node independently produces and publishes its schedule or transmission priority.

Choi and Shin suggested a unified protocol for real-time and non-real-time communications in wireless networks [Choi and Shin, 2000]. In their scheme, a BS (Base Station) polls a real-time mobile station according to the non-preemptable EDF (Earliest Deadline First) policy. The BS also polls the non-real-time message according to the modified round-robin scheme regardless of a standard CSMA/CA protocol to eliminate message collision. The retransmission of a damaged packet is considered as a normal non-real-time message. Additionally, to handle location-dependent, time-varying, and bursty channel errors, the channel state can be predicted via channel probing before the packet is transmitted. That is, before transmitting a downlink packet to the mobile, the BS transmits a probing control packet to that mobile, which then returns the control packet to the BS. With this estimation, it is possible to reduce the need for retransmission, since retransmission can be harmful in meeting deadlines.

Most works that conform to the IEEE standard are aiming at just enhancing the ratio of timely delivery for soft multimedia applications, rather than providing a hard real-time guarantee. As an example of pure bandwidth allocation scheme, DBASE (Distributed Bandwidth Allocation/Sharing/Extension) is a protocol that supports both synchronous and multimedia traffics over IEEE 802.11 *ad hoc* WLAN [Sheu and Sheu, 2001]. The basic concept is that each time a real-time station transmits a packet it will also declare and reserve the bandwidth demanded at the next CFP. Every station collects this information and then calculates its actual bandwidth at the next cycle. But it focuses on deciding whether to accept the respective real-time packet without considering a flow-level allocation or negotiation. This scheme can be ported to WLAN standard, but it does not provide a hard real-time guarantee as it does not directly consider the maximal bandwidth request. Moreover, unless a packet is correctly received by all member stations, they can have different schedules.

Adamou and his colleagues have addressed the scheduling problem of achieving fairness among real-time flows with deadline constraints as well as maximizing the throughput of all the real-time flows over a wireless LAN [Adamou et al. 2001]. They chose the scheduling objective of minimizing the maximum degree of the degraded QoS among all applications. Their scheduling policy includes EDF, GDF (Greatest Degradation First), EOG (EDF Or GDF), and LFF (Lagging Flows First). The BS performs the scheduling of real-time packet deliveries using a polling scheme. This scheme is built on the assumption that BS knows which station has messages to retransmit as well as their deadlines, and decides which one to poll among them according to the criteria described above. In addition to the retransmission technique, there exists a way the sender adjusts the transmission rate according to the currently available network bandwidth. Shah proposed a dynamic bandwidth management scheme in a single-hop ad hoc wireless network that is especially suitable for hot-spot networks, that is, a number of nodes in a small area share limited channel bandwidth [Shah et al. 2005]. As the available channel capacity changes and the traffic characteristics of various flows change, the bandwidth manager dynamically reallocates the channel access time to the individual flows.

By exploiting the periodic nature of sensor network traffic, Caccamo et al. realize collision free real time scheduling as follows [Caccamo et al. 2002]: FDM (Frequency Division Multiplexing) is used among adjacent cells to allow for concurrent communications in different cells. M. Caccamo and et. al have proposed a MAC that supports deterministic real-time scheduling via the implementation of TDMA, where the time axis is divided into fixed size slots. Implicit EDF is used inside each cell. There is a router cell located in the center in the center area of each cell. Router nodes are equipped with two transceivers so they can transmit and receive at the same time using two different frequency channels.

The key idea for conflict free real time scheduling is to replicate the EDF schedule at each node for packet transmission. If the schedules are kept identical, each node will know which one has the message with the shortest deadline and has the right to transmit next. In addition, when a node is listening to the channel, it is also able to know the completion of a node's transmission and thus update its scheduling queue for next round of communication. The scheduling table reserves the worst case message transmission time for each periodic message stream. If a node uses only part of the reserved frames, instead of transmitting its reserved periodic message early, other node may use the frames left by the node to send best effort aperiodic messages. This FRASH(FRAme SHaring) technique is designed to systematically and reliably exploit reserved but unused frames. Its peculiarity is to increase network utilization while preserving the hard real-time guarantee. This goal is achieved by reclaiming the slots reserved but not used by the hard real-time messages. Such frames can be reassigned to aperiodic server in order to improve the responsiveness of the aperiodic messages.

Referred as *implicit contention*, their scheme makes every station concurrently run the common real-time scheduling algorithm to determine which message can access the medium. Each message implicitly contends for the medium through the scheduling algorithm, for example with priorities, instead of explicitly on the medium. However, for this implicit contention, every node must schedule all messages in the network, making it difficult to scale to large networks and resulting in the complexity growing linearly with the number of messages. In addition, their scheme didn't consider the network error at all, just focusing on the scheduling policy. Such schemes cannot be ported to the IEEE 802.11 WLAN standard, as they ignored the mandatory CSMA/CA part of WLAN. In addition, the access right is given to a station not by the actual message arrival but just by the estimation. The discrepancy may invalidate the guaranteed schedule when non-real-time traffic coexists in the network.

Also, Caccamo et al. [Caccamo et al. 2005] presented a general CPU scheduling methodology for managing overruns in a real-time environment, where tasks may have different criticality, flexible timing constraints, shared resources, and variable execution times. The proposed method enhances the CBS (Constant Bandwidth Server) by providing two important extensions. First, it includes an efficient bandwidth sharing mechanism that reclaims the unused bandwidth to enhance task responsiveness. It is proven that the reclaiming mechanism does not violate the isolation property of the CBS and can be safely adopted to achieve temporal protection even when resource reservations are not precisely assigned. Second, the proposed method allows the CBS to work in the presence of shared resources. The enhancements achieved by the proposed approach turned out to be very effective with respect to classical CPU reservation schemes.

Sohrabi and Pottie [Sohrabi and Pottie, 1999] proposed an self-organization protocol for wireless sensor networks. Each node maintains a TDMA-like frame, called super-frame, in which the node schedules different time slots to communicate with its known neighbors. At each time slot, it only talks to one neighbor. To avoid interference between adjacent links, the protocol assigns different channels, i.e., FDMA(Frequency Division Multiple Access) or CDMA(Code Division Multiple Access), to potentially interfering links. Although the super-frame structure is similar to a TDMA frame, it does not prevent two interfering nodes from accessing the medium at the same time. The actual multiple access is accomplished by FDMA or CDMA. However, using TDMA protocol usually requires the nodes to form real communication clusters, like Bluetooth and LEACH (Low-Energy Adaptive Clustering Hierarchy) [Heinzelman et al. 2000]. Managing inter-cluster communication and interference is not an easy task. Moreover, when the number of nodes within a cluster changes, it is not easy for a TDMA protocol to dynamically change its frame length and time slot assignment. So its scalability is normally not as good as that of a contention-based protocol.

B. Contention based MAC Protocols

The standardized IEEE 802.11 DCF is an example of the contention based protocol. It is widely used in ad hoc wireless networks because of its simplicity and robustness to the hidden terminal problem. Most of the distributed MAC Protocols are based on the carrier sensing and/or collision avoidance mechanism, and may employ additional signaling control messages to deal with hidden and exposed node problems. Such signaling messages may be delivered in two ways: in-band hand-shaking or out-of-band signaling [Stankovic et al. 2003]. BTMA(Busy-Tone Multiple Access) is a representative of the out-of-band signaling protocol. In BTMA, a node that hears an ongoing transmission transmits a busy tone, and any node that hears a busy tone does not initiate transmission. This eliminates the hidden nodes, but increases the number of exposed nodes.

Another class of medium access control uses in-band control packets like RTS and CTS to exchange the local view of channel status, so as to avoid potential collisions. MACA(Multiple access with collision avoidance) uses a three-way handshaking to solve the hidden node problem [Karn, 1990]. A node that has data to send transmits a short RTS packet. All nodes within one hop of the sending node hear the RTS and defer their transmission. The destination responds with a CTS packet. All nodes within one hop of the destination node hear the

CTS packet and also defer their transmission. On receiving the CTS the transmitting node assumes that the channel is acquired and initiates the data transmission. The hidden node problem is not completely solved by this scheme, but is avoided to a large extent. Several schemes have been proposed to enhance the RTS-CTS handshaking mechanism, the details of which can be found in [Fullmer and Garcia-Luna-Aveces, 1995] [Fullmer and Garcia-Luna-Aveces, 1997].

As the variations of RTS/CTS schemes, You et al. [You et al. 2003] investigated a new class of collision prevention MAC protocols, called CSMA/CP(CSMA with Collision Prevention), for wireless ad hoc networks. Carlson [Carlson et al. 2005] suggested DARE(Distributed end-to-end Allocation of time slots for REal-time traffic) protocol which allocate radio resources on the path between source and destination, and then compared the scheme with IEEE 802.11e standard for QoS support in WLAN-based ad hoc networks. Woo and Culler [Woo and Culler, 2001] examined different configurations of CSMA and proposed an adaptive rate control mechanism, whose main goal is to achieve fair bandwidth allocation to all nodes in a multihop network. They have used the motes and TinyOS platform to test and measure different MAC schemes. In comparison, this approach does not promote per-node fairness, and even trade it off for further energy savings.

In the non-academic effort, IETF has produced new drafts, EDCF (Enhanced DCF) and HCF (Hybrid Coordination Function) to replace the CSMA/CA based and centralized polling based access mechanism, respectively [Mangold et al. 2002]. No guarantees of service are provided, but EDCF establishes a probabilistic priority mechanism to allocate bandwidth based on traffic categories. According to HCF, a hybrid controller polls stations during a CFP. The polling grants a station a specific start time and a maximum duration to transmit messages.

As another example, in BB (Black Burst) contention scheme, stations sort their access rights by jamming the channel with pulses of energy before sending their packet [Sobrino and Krishakumar, 1999]. Since packets must be transmitted repeatedly in a constant interval, sending burst of energy for each packet will waste considerable network bandwidth. Moreover, the BB contention is not a regular scheme defined in IEEE 802.11 standard. In addition, a distributed fair scheduling is proposed for a wireless LAN by N. Vaidya, and et. al [Vaidya et al. 2000]. Its essential idea is to choose a backoff interval that is proportional to the finish tag of a packet to be transmitted. With fair scheduling, different flows share a bandwidth in proportion to their weights. However, this scheme cannot provide hard real-time guarantee due to unpredictable collisions between two or more packets that have same tag.

All the aforementioned contention-based MAC protocols are subject to the open challenge of providing a statistical bound on the real-time requirement. Due to the distributed and random back-off nature, contention-based MAC does not strict guarantee the priority order of packets from different nodes. For example, two high-priority packets may collide and cause each node to backoff, while a third node may send out a low-priority packet when the other two nodes are in the backoff phase. It is necessary to bound the probability of priority inversion in order to establish statistical end-to-end delay guarantees.

Existing wireless MAC protocols focus more on optimizing system throughput and do not adequately consider the requirements of sensor networks. The key challenge remains to provide predictable delay and/or prioritization guarantees while minimizing overhead packets and energy consumption [Monks et al. 2001] [Stemm and Katz, 1997] [Bennett et al. 1997] [Luo and Jha. 2001] [Ye et al. 2002]

[Songh and Raghavendra, 1998]. Recent studies have shown that radio communication is the dominate consumer of energy in sensor networks [Hill et al. 2000] The MAC is a broad research area, and many researchers have done research work in the new area of low power and wireless sensor networks [Sohrabi and Pottie, 1999] [Bennett et al. 1997] [Heinzelman et al. 2000] [Woo and Culler, 2001].

2.2.2 Error control scheme

Recent advances in wireless communication technology are making WLAN an appealing transmission media for parallel and distributed computing on networked computers [McKnight et al. 2004] [Clarke and Humphrey, 2002] [Lee et al. 2006a] [Lee et al. 2003b] [Lee et al. 2003]. The use of MPI (Message Passing Interface), PVM (Parallel Virtual Machine), or other variants can work in wireless environment because they are built on top of TCP/IP and therefore the physical medium does not impose any restriction [Macías and Suárez, 2002a] [Fagg et al. 2001] [Macías and Suárez, 2002b]. However, it is not clear how well this mechanism fits for wireless networks, since wireless channels are subject to unpredictable *location-dependent* and *bursty* errors.

Error control in wireless network has been intensively studied for both unicast and multicast. Most of the approaches use ARQ (Automatic Retransmission reQuest), FEC (Forward Error Correction), or a combination of both [Lu et al. 2005]. Pure ARQ-based schemes are less appropriate for the multicast case due to ACK explosions and the requirement to retransmit different packets to the respective users. For significant packet loss rates, each user will require frequent packet replacement, and different receivers are most likely to require different packets. To respond to requests by multiple users, we may have to resend a significant fraction of the original data even for small loss rates. For a packet network with dynamic bandwidth, a different class of the source coding technique called progressive coding is better suited. However, this method has the problem in deciding optimal coding parameter on wireless channel.

As a hybrid ARQ mechanism, Majumdar et al. have proposed a method that combines the reliability and fixed delay advantage of forward error control coding with bandwidth-conserving channel-adaptive properties of ARQ protocol [Majumdar et al. 2002]. Masala has also proposed a multicast scheme that aims at globally optimizing the parameters involved in a real-time video transmission, ranging from video encoding and packetization to the 802.11 MAC interface parameters [Masala et al. 2003]. These schemes are mainly built on top of DCF, so they didn't consider the effect of QoS reservation.

Lu et al. have proposed a timestamp-based content-aware adaptive retry mechanism where MAC dynamically determines whether to send or discard a packet by its retransmission deadline, which is assigned to each packet according to its temporal relationship and error propagation characteristics with respect to other video packets within the same group of pictures [Lu et al. 2005]. However, their scheme is too complex to be exploited in the WLAN standard, as it crosses the protocol layer boundaries.

Lee et al. proposed an error control scheme that dynamically adjusts the polling schedule so that more corrupted messages can be manipulated within their deadlines for real-time communication on IEEE 802.11 Wireless LAN. Monitoring the current traffic on CFP interval as well as inspecting whether a node is executing an error control procedure, AP terminates CFP, if possible, to extend the duration of the next CP. With the enlarged time, AP gives more polls to the node in need to increase the number of timely recovered messages [Lee et al. 2002].

Kang and Lee [Kang et al. 2006b], Lee et al. [Lee et al. 2005c] proposed and analyzed the performance of a prioritized error control scheme for time sensitive application on the wireless sensor network. As a modified version of IEEE 802.11 WLAN, the proposed scheme further divides DCF into H-DCF and L-DCF without changing PCF, aiming at maximizing the successful retransmission of a packet that carries critical data. While channel estimation eliminates the unnecessary polls to the sensor node currently unreachable during PCF, two DCF subperiods enable prioritized error recovery by making only the high priority packet be retransmitted via H-DCF. A good chop value, which distributes the retransmission to each period, can maximize recovered weight, or criticality, minimizing the possible degradation of network throughput.

2.2.3 Handoff scheme

A lot of works have been already carried out to reduce the handoff latency for the roaming client station. However, existing handoff schemes are not suitable for meeting requirements of real-time multimedia application due to its long and occasionally unbounded delay. The sequence of messages being exchanged during the handoff process can be categorized into three groups, namely, probe, authentication, and association. Accordingly, existing works are also classified by the delay element to reduce.

First, the researches to improve the probe delay are as follows: Kim et al. proposed a selective scanning algorithm using the neighbor graphs [Kim et al. 2004]. This approach forces changes in the network infrastructure and use of IAPP (Inter Access Point Protocol) though it narrows the search space with neighbor graphs. Moreover, this scheme does not consider the time amount required by the client to process the received *probe responses*. Shin et al. proposed a new handoff procedure which reduces the MAC layer handoff latency, in most cases, to a level where VoIP communication becomes seamless using both selective scanning algorithm and caching mechanism [Shin et al. 2004]. It needs just an insignificant modification in the client-side wireless card driver such as channel mask and improved cache dimensioning. According to the analysis result by Jain [Jain, 2003] and Mishra [Mishra et al. 2003], there are remarkable variations in handoff latencies with change in SSID and channel of APs, and probe delay is the major malicious factor to the total handoff performance. According to the analysis result by Jain, there are significant variations in handoff latencies with change in SSID and channel of APs. He addressed that probe delay is the major malicious factor to the total handoff performance. Mishra and et al. analyzed the handoff latencies by breaking down the whole process into a series of subphases to assess the rate of each phase to the handoff latency.

Second, to improve the authentication delay, Pack et al. proposed a fast Inter-AP handoff scheme using the predictive authentication method based on IEEE 802.1x model [Pack and Choi, 2002] [Pack and Choi, 2002b]. The IEEE 802.1x authentication delay is reduced by using the FHR (Frequent Handoff Region) selection algorithm that makes the candidate APs selected by the predictive algorithm, perform the pre-authentication, directly taking into account traffic patterns and user characteristics, which are collected and managed in the centralized system.

Third, to improve the association delay, Mishra et al. focused on reducing the reassociation delay [Mishra et al. 2004]. The reassociation delay is reduced by using a caching mechanism on the AP side. This caching mechanism is based on the IAPP protocol in order to exchange the client context information between neighboring APs. The cache in the AP is built by observing the information contained in an IAPP *Move-Notify* message or in the *reassociation request* sent to the AP by the client. By exchanging the client information with the old AP, the new AP prevents the client from sending its context information, resulting in the reduction of the reassociation delay.

2.2.4 Dual channel network

IEEE 802.11 MAC specification allows for two modes of operation: ad hoc and infrastructure modes [IEEE, 1999]. In ad hoc mode, two or more stations recognize each other through beacons and establish a peer-to-peer communication without any existing infrastructure, whereas in infrastructure mode there is a fixed entity called an AP that bridges all data between the mobile stations associated to it. Though PCF is optional, QoS guarantee cannot be provided without PCF. Applying the DCF of IEEE 802.11 in wireless networks leads to uncertainties. These uncertainties sum up over multiple hops, hence throughput and end-to-end delay can suffer from large variations. This is especially crucial for real-time applications with demanding QoS requirements. Recently, to support a certain level of QoS, the PCF-enabled schemes are increasingly being applied to the WLAN showing a reasonable throughput. For example, in smart-rooms and hot-spot networks, wireless access-enabled stations in a small area share the medium [Shah et al. 2005]. When 802.11 WLAN stations are close enough to form a direct connection without preplanning, this type of operation is referred to as an *ad hoc* network.

To support the multihop and mobile characteristics of wireless ad hoc networks, the rapid deployment of network and dynamic reconstruction after topology changes are efficiently implemented by clustering management. There are several algorithms used to divide the network into clusters. The most widely used clustering algorithms are LIDCA (Lowest IDentifier Clustering Algorithm) and HCCA (Highest Connectivity Clustering Algorithm) [Kai and Jiandong, 2000]. Nodes in the network have a unique identifier. LIDCA organizes the network based on this identifier, giving the role of a cluster-head to the node with the lowest ID in a neighborhood. The operation of HCCA is similar to LIDCA, but it divides the network according to the connectivity of each node, thus selects the nodes with highest connectivity - those with most neighbors - as cluster-heads. In both algorithms, every cluster is identified using the ID of its cluster-head. Upon deployment, nodes transmit their position in a single TDMA frame, with enough power to reach all the other nodes. In the ad hoc mode, this cluster-head node can designate to play a role of PC to schedule the transmission.

Two different key approaches can be followed in the implementation of a WLAN: an *infrastructure-based* approach and an *ad-hoc networking* one. The infrastructure-based architecture imposes the existence of a centralized controller for each cell, often referred to as AP. The AP is normally a gateway to the external networks, for example, wired backbone network, or other wireless networks. An

AP and associated mobile stations form a BSS (Basic Service Set) communicating on the unlicensed RF spectrum. Multiple APs form an ESS (Extended Service Set) that constructs the same wireless network. Whereas, an ad-hoc network is a peer-to-peer network formed by a set of stations within the range of each other that dynamically configure themselves to set up a temporary network. PCF access cannot be adopted in ad-hoc networks.

In ad hoc network, each cell is assumed to consist of a PC and multiple sensor stations, and each of them is capable of transmitting and receiving at the same time using two transceivers while the adjacent channels are separated by guard bands. Every station shares medium on the common frequency band and accesses according to the predefined MAC protocol. Each flow is ether an uplink or downlink, while PC coordinates the overall network operations. This thesis exploits the contention-free TDMA style access policy as in [Adamou et al. 2001, Carley et al. 2003, Choi and Shin, 2000], for the real-time guarantee, as well as the contention resolution via packet collisions consumes the precious communication energy.

Four wireless ad hoc routing protocols are as follows [Sesay et al. 2004]: DSDV (Destination-Sequenced Distance Vector) is a table-driven protocol, wherein each node maintains a routing table listing the ”next hop” for each reachable destination. Every node in DSDV periodically broadcasts its routing table with monotonically increasing even sequence number. TORA (Temporally Ordered Routing Algorithm) is an on-demand routing protocol design to provide loop-free and multiple routes to alleviate congestion and yet minimize communication overhead by localizing algorithmic reaction to topological changes when possible. Moreover, it is desirable to detect network partition and delete invalid routes. DSR (Dynamic

Sources Routing) is an on-demand routing protocol wherein the source determines the ordered list of nodes through which a packet must pass while traveling to its destination. The key advantage of source routing is that intermediate nodes do not need to maintain up-to-date routing information in order to route the packets they forward, since the packets themselves already contain all the routing decisions. This fact, coupled with the on-demand nature of the protocol, eliminates the need for the periodic route advertisement and neighbor detection packets present in other protocols. AODV (Ad hoc On-demand Distance Vector) is essentially a combination of DSR and DSDV. It borrows the basic on-demand mechanism of route Discovery and Route maintenance from DSR, plus the use of hop-by-hop routing, sequence number and periodic beacon from DSDV. Also, routing protocols for the wireless sensor network include SMECN (Small Minimum Energy Communication Network), SPIN (Sensor Protocol for Information via Negotiation), Directed Diffusion, LEACH (Low-Energy Adaptive Clustering Hierarchy), TinyOS Routing Protocols. These routing protocol issues are different problems and out of the scope of this thesis. We just assume that some existing routing protocol can be integrated to our framework.

The dual network architecture is analogous to the dual processor system, as both network and processor can be considered as an active resource. According to the IEEE 802.11 standards, at any instance a maximum of three channels can be used simultaneously as the channels overlap each other [Rangnekar et al. 2004]. In this system, each station has a transmitter and a receiver that can tune to either channel, giving the flexibility to transmit and receive on both channels. Jung and Flaviis investigated double rectangular path with 4-bridges fo solution of IEEE 802.11b/g(2.4GHz) and 802.11a (5.5GHz)[Jung and Flaviis, 2004]. Yu et al. [Yu et al. 2005] proposed a simple and viable approach to enhance the VoIP performance over the 802.11 WLAN by implementing two queues along with a strict priority queuing on top of the 802.11 MAC controller, e.g., in the device driver of the 802.11 cards. Traditionally, there have been two approaches for scheduling periodic tasks in dual or multiprocessors, namely, *partitioning* and *global scheduling* [Carley et al. 2003].

The partitioning scheme assigns each stream to a single network, on which messages are scheduled independently. The main advantage of partitioning approaches is that they reduce a multiprocessor scheduling problem to a set of uniprocessor ones. However, finding an optimal assignment to networks is a bin-packing problem, which is NP-hard in the strong sense. In global scheduling, all eligible tasks are stored in a single priority-ordered queue while the global scheduler selects for execution of the highest priority task from this queue. For example, CASP (Contiguous Algorithm for Single Priority) maintains an allocation vector A, where A_i represents the partial sum of slots currently allocated to channel i [Damodaran and Sivalingam, 2002]. For a given request, the scheduling algorithm allocates the request contiguously on the channel which has the least partial sum of allocation. Additionally, NCASP (Non-continuous Algorithm for Single Priority) defines an overflow amount Φ, and if an assignment makes A_i exceed Φ, it is split and then the overflown part is assigned to another resource. However, how to decide Φ brings another complex case-sensitive problem.

The multi-NIC approach has been discussed in some past work [Draves et al. 2004] [Raniwala and Chiueh, 2005]. They use multiple 802.11 NICs per node in an ad hoc network by assuming an identical channel assignment to all nodes - NIC-1 is assigned channel-1, NIC-2 to channel-2, and so on. This approach can only yield a factor 2 of improvement using 2 NICs. The Hyacinth architecture consists of a multi-channel WMN (Wireless Mesh Network) core, which is connected to a wired network through a set of wired connectivity gateways. Each WMN node has multiple interfaces, each operating at a distinct radio channel. A WMN node is equipped with a traffic aggregation device similar to an 802.11 access point that interacts with individual mobile stations. The multi-channel WMN relays mobile stations' aggregated data traffic to/from the wired network. The links between nodes denote direct communication over the channel indicated by the number on the link. If each node is equipped with 2 wireless NICs, the number of channels any node uses simultaneously cannot be more than 2.

Providing each node with multiple radios offers a promising avenue for improving the capacity. First, it enables nodes to transmit and receive simultaneously. Otherwise, with only one radio, the capacity of relay nodes is halved. Second, the network can utilize more of the radio spectrum. With two radios, a node may transmit on two channels simultaneously. Third, radios that operate on different frequency bands, for example, 802.11a at 5GHz and 802.11b/g at 2.4GHz, have different bandwidth, range, and fading characteristics. Using multiple heterogeneous radios offers tradeoffs that can improve robustness, connectivity, and performance. Finally, 802.11 radios are off-the-shelf commodity parts with rapidly diminishing prices. This makes it natural to consider the use of multiple inexpensive radios per node.

Chapter 3

Bandwidth management scheme

3.1 Network and message model

The IEEE 802.11 was developed as a MAC standard for WLAN [IEEE, 1999], and we exploited a architecture as shown in Figure 3.1. The standard consists of a basic DCF and an optional PCF for uplink channel access. In the other hand, for downlink transmission, AP distributes the multicast streams to the receivers as well as relays the other non-real-time traffic without any collision. The standard divides the time axis into the series of superframes, and each of them consists of CFP and CP as shown in Figure 3.2. In CFP, AP polls each stream once a polling round, providing a predictable access to each delay sensitive streams. On the contrary, during CP any station can send non-real-time message after contending the shared medium via CSMA/CA.

In recent years significant advances have been made in the area of real-time

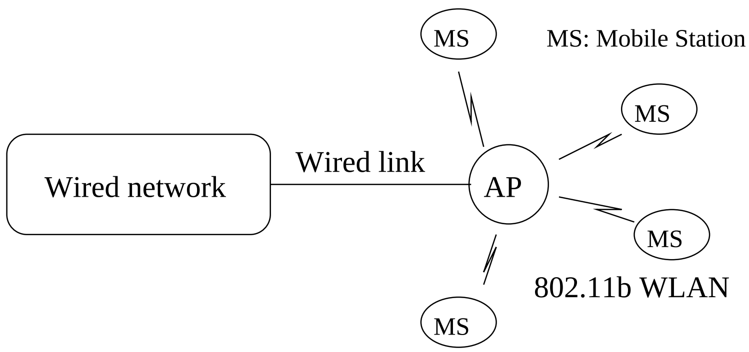


Figure 3.1: WLAN-based architecture

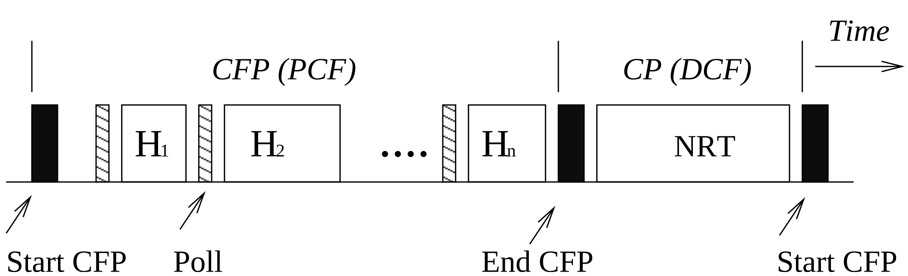


Figure 3.2: Polling procedure and capacity vector

communication scheduling. WLANs are emerging as an attractive alternative, or complementary, to wired LANs, because they enable us to set up and reconfigure LANs without incurring the cost of wiring [Crow et al. 1997]. It makes a sense that various real-time systems also adopt WLAN as their communication network. Real-time systems could benefit from component-based design, only if components can be assembled without violating compositionality on timing properties[Shin et al. 2004]. We mean by real-time that an information package has to be received by the recipient before a certain deadline, so the network should be able to meet the requirement of the time-constrained traffic [Malcolm and Zhao, 1995].

In real-time communication literature, the term real-time traffic typically means *isochronous* (or synchronous) traffic, consisting of message streams that are generated by their sources on a continuing basis and delivered to their respective destinations also on a continuing basis [Liu, 2000]. For example, a sensor node continuously reports the collected sensor data to a central high-performance server. The traffic characteristics such as period and message size are usually known before the system operation. That is, the stream set does not change, or the stream set remains unchanged for quite a long time as the connection establishment or relinquishment happens infrequently. So we can assume that the stream set is fixed in priori of bandwidth allocation. If the stream set is fixed, the network schedule is calculated offline as in the typical real-time system. The most important traffic characteristics that describe time constraint are its period and message size of each message stream. In addition, each node can send PC an association request message to get associated with the PC via CP period. This message contains a PCF related real-time traffic information, such as period and transmission time.

Every time the stream set changes, a new schedule, or bandwidth allocation procedure is performed and broadcasted to the participating nodes with the support of mechanisms such as mode change [Liu, 2000].

A node, currently inactive, can be activated by an upper layer query command that wants to monitor or process the data flow from the sensor node. The destination of message can be either within a cell or outside a cell, and the outbound messages are first sent to the router node such as AP and then forwarded to the final destination. A proper routing and reservation protocol can provide an end-to-end delay guarantee [Mao et al. 2005]. Internal messages are also relayed by the AP. The query may also specify sampling period and the precision level of needed data, hence, message length on a node. In case of a change in the active flow set, bandwidth is to be reallocated or network schedule mode is changed [Shah et al. 2005].

This thesis takes the general real-time message model which has n streams, namely, S_1, S_2, ..., S_n, and for each S_i, a message sized less than C_i arrives at the beginning of its period, P_i, and it must be transmitted by P_i. The period of stream, S_i, is denoted as P_i, and the maximum length of a message as C_i. The first message of each stream arrives at time 0. Each packet must be delivered to its destination within D_i units of time from its generation or arrival at the source, otherwise, the packet is considered to be lost. We assume that D_i is larger than P_i to give a sufficient margin to recover transmission failures. When S_i is polled, it can transmit up to H_i, and $\{H_i\}$ is named as *capacity vector*. The sampled data has its own weight and the weight should be mapped to the corresponding priority level.

As is the case of other works, we begin with an assumption that each stream has only one stream, and this assumption can be generalized with virtual station concept [Liu, 2000]. At the other extreme, the superframe time, F, may be a fixed network parameter that can hardly changeable. So we assume that the superframe time is also given in priori, focusing on the determination of capacity vector. It is desirable that the superframe time is a hyperperiod of the set and a message set can be made harmonic by reducing the period of some streams by at most half [Carley et al. 2003]. The H_i limits the maximum time amount for which S_i can send its message. A polling round may spread over more than one superframe, however, for simplicity, we assume that a polling round completes within a single superframe and the assumption will be eliminated later.

Ideally, PC attempts to initiate CFP every F interval if there is no deferred beacon. When its timer expires, PC broadcasts a beacon frame after waiting short IFS from the instant it detects the medium free. We just assume that some existing error control or QoS degrading schemes can be integrated into our framework [Adamou et al. 2001]. Otherwise, when experiencing packet losses above some specified threshold, the application undergoes degraded quality of service. Consequently, though there are many issues one needs to consider in wireless networks, we mainly focus on a significant performance issue, that is, timeliness [Caccamo et al. 2002].

3.2 Bandwidth allocation scheme

Allocation procedure determines capacity vector, $\{H_i\}$, for the given superframe time, F, and message stream set described as $\{S_i(P_i, C_i)\}$. Figure 3.2 illustrates that the slot size is not fixed, namely, $H_i \neq H_j$ for different i and j. It is natural that the more bandwidth is assigned to the stream with a higher utilization. At this figure, a message of size C_i is generated and buffered at regular intervals of P_i, and then transmitted by H_i every time the node receives poll from PC.

Let δ denote the total overhead of a superframe including polling latency, IFS, exchange of beacon frame, and the like, while D_{max} the maximum length of a non-real-time data packet. For a minimal requirement, F should be sufficiently large enough to make the polling overhead insignificant. In addition, if P_{min} is the smallest element of set $\{P_i\}$, F should be less than P_{min} so that every stream can meet at least one superframe within its period. For each superframe, not only the start of CFP can be deferred by up to D_{max}, but also at least a time amount as large as D_{max}, should be reserved for a data packet so as to keep compatibility with WLAN standard. After all, the requirement for the superframe time, F, can be summarized as follows:

$$\sum H_i + \delta + 2 \cdot D_{max} \leq F \leq P_{min} \tag{3.1}$$

The number of polls a stream meets is different period by period. Meeting hard real-time constraints for a station means that even in the period which has the smallest number of polls, the station can transmit message within its deadline.

Figure 3.3 analyzes the worst case available time for S_i. In this figure, a series of superframes are repeated in P_i and each period can start at any instant from the

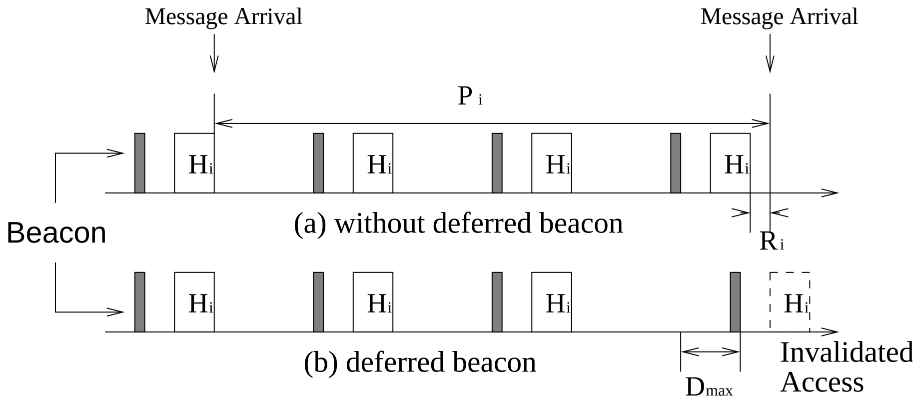


Figure 3.3: Worst case analysis

start of the superframe. Intuitively, the station can meet the smallest number of polls in the period which starts just after the end of its slot.

In this figure, R_i is the residual obtained by dividing P_i by F, namely, $R_i = P_i - \lfloor \frac{P_i}{F} \rfloor \cdot F$. Without the deferred beacon, the CFP starts at regular intervals of F. For S_i, the least bound of network access within P_i is $\lfloor \frac{P_i}{F} \rfloor$, as illustrated in Figure 3.3(a). On the contrary, if we consider the deferred beacon, the deferred start of the last superframe may invalidate one access when R_i is less than D_{max}, as shown in Figure 3.3(b). If R_i is greater than D_{max}, the number of available time slots is not affected by the delayed start of superframe. It doesn't matter whether the start of an intermediate superframe is deferred or not. In short, S_i can be affected by the deferred beacon in case R_i is less than D_{max} and the minimum value of available transmission time, X_i is calculated as Eq. (3.2). Namely,

$$\begin{aligned} X_i &= (\lfloor \tfrac{P_i}{F} \rfloor - 1) \cdot H_i \qquad && if (P_i - \lfloor \tfrac{P_i}{F} \rfloor \cdot F) \leq D_{max} \\ X_i &= \lfloor \tfrac{P_i}{F} \rfloor \cdot H_i && Otherwise \end{aligned} \tag{3.2}$$

For each message stream, X_i should be greater than or equal to C_i ($X_i \geq C_i$). By substituting Eq. (3.2) for this inequality, we can obtain the least bound of H_i that can meet the time constraint of S_i.

$$\begin{array}{ll} H_i = \frac{C_i}{(\lfloor \frac{P_i}{F} \rfloor - 1)} & if(P_i - \lfloor \frac{P_i}{F} \rfloor \cdot F) \leq D_{max} \\ H_i = \frac{C_i}{\lfloor \frac{P_i}{F} \rfloor} & Otherwise \end{array} \tag{3.3}$$

The allocation vector calculated by Eq. (3.3) is a feasible schedule if the vector meets Ineq. (3.1). By this, we can determine the length of CFP (T_{CFP}) and that of CP (T_{CP}) as follows:

$$T_{CFP} = \sum H_i + \delta, \quad T_{CP} = F - T_{CFP} \geq D_{max} \tag{3.4}$$

This calculation is easily fulfilled with simple arithmetic operations. In addition, the size of time slot is different for each stream, so the capacity allocation by Eq. (3.3) can expect a better network utilization compared to other schemes based on fixed size slots. Finally, as this allocation scheme generates a larger T_{CP} for the given F, the network can accommodate more non-real-time messages [Kang and Lee, 2004][Lee et al. 2005].

In case a polling round spreads over more than one superframe, say k superframes, each one can be marked as F_1, F_2, ..., F_k. The size of each superframe is F, while each includes its own CP duration and performs only a part of polling round. S_i receives poll once a $k \cdot F$ and the allocation formula can be modified by replacing F with $k \cdot F$ in Eq. (3.2). But the condition remains intact which checks whether a stream will be affected by a deferred beacon. After all, Eq. (3.2) can be rewritten as follows:

$$\begin{array}{ll} X_i = (\lfloor \frac{P_i}{k \cdot F} \rfloor - 1) \cdot H_i & if(P_i - \lfloor \frac{P_i}{F} \rfloor \cdot F) \leq D_{max} \\ X_i = \lfloor \frac{P_i}{k \cdot F} \rfloor \cdot H_i & Otherwise \end{array} \tag{3.5}$$

If a node has no pending message when it receives a poll, it responds with a null frame containing no payload. The rest of the polling in the superframe can be shifted ahead without violating the time constraint of their real-time messages if and only if all the subsequent nodes have their messages to send in that superframe. The CP in such a superframe, say *reclaimable superframe* can start earlier than those of other normal superframes. In addition, we can improve the probability of bandwidth reclaiming by rearranging the polling order.

3.3 Performance measurement result

A comparison with TDMA is inherently impossible because it requires so many assumptions to be decided on slot size, slot allocation scheme, and the way to cope with deferred beacon. Moreover, TDMA-based schemes excluded contention period, giving up compatibility with WLAN standard. In the mean time, other group of approaches guarantees meeting the time constraint for the hard real-time messages based on a pessimistic assumption that every stream suffers from the deferred beacon [Sheu and Sheu, 2001]. This is due to the fact that those schemes didn't accurately estimate the intervention of non-real-time message. So the experiment compares the schedulability of our scheme with this pessimistic guarantee mechanism via simulation using ns-2 event scheduler [Fall and Varadhan, 1997].

We have generated 2000 streams sets whose utilization has the value of 0.68 through 0.70, while the number of streams randomly distributes from 2 to 10. In the experiment, every time variable is aligned to the frame time. Accordingly, the period of each stream ranges from 5.0F to 10.0F, while its message length

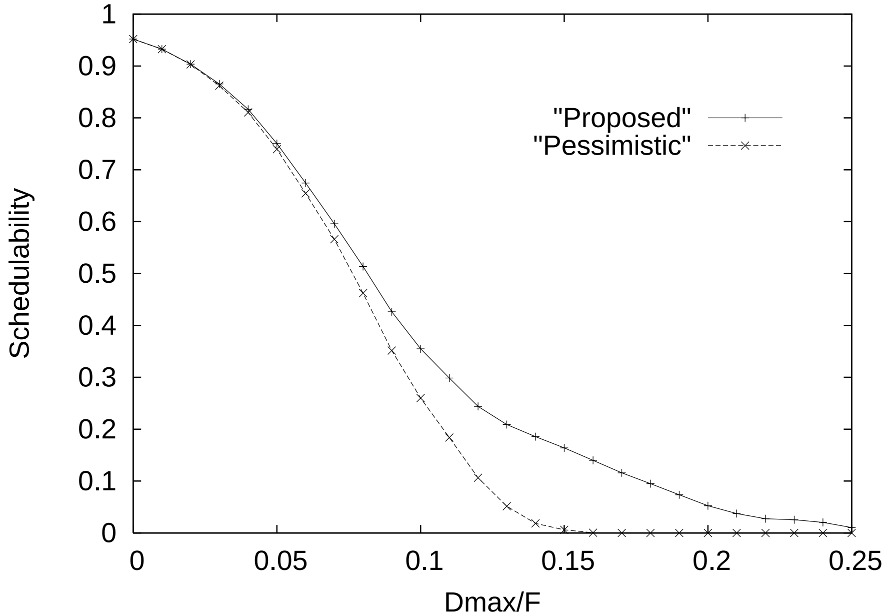


Figure 3.4: Measured guarantee ratio

from $0.3F$ to $3.0F$. Figure 3.4 plots schedulability of proposed and pessimistic schemes changing D_{max} from 0 to $0.25F$. At low D_{max}, both schemes show equal schedulability, however, the proposed scheme improves the guarantee ratio by 18 % at maximum. In addition, the proposed scheme generates the T_{CP} up to 5.3% larger than that generated by the pessimistic scheme, enabling more non-real-time messages to occupy the network.

Chapter 4

Error control scheme

4.1 Network model

Considering bursty and unpredictable nature of wireless channel errors, immediate retransmission does not seem to be appropriate, as the subsequent retransmissions to the original destination may fail repeatedly. To solve this problem, this thesis will propose and analyze the performance of an error control scheme for multicast video stream on IEEE 802.11 WLAN. We consider a wireless-cum-wired network scenario as shown in Figure 4.1. A fixed node is connected with an AP through a wired link which is overprovisioned so that no packets are dropped at its end [Bottigliengo et al. 2004]. This model enables us to concentrate on the behavior of inside WLAN part as it assumes ideal environment of no packet loss or jitter for outside WLAN part. Each cell is assumed to consist of an AP and multiple MSs (Mobile Stations), while each flow is either uplink (from MS to AP) or

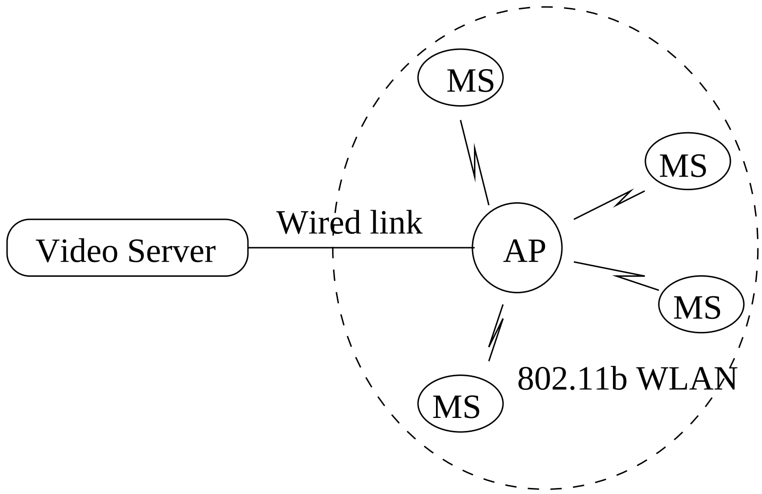


Figure 4.1: Multicast Network model

downlink (from AP to MS). Most flows are downlink, and the stream flow arrives periodically from the remote server outside the cell via reliable wired link.

The multicast on WLAN prevents the automatic ACK transmission from the receiver. The multicast stream is also an instance of message stream that flows from AP to WLAN. Appropriate time amount is allocated to each multicast stream, and the length of time interval is calculated according to the QoS requirement. The video stream traffic occupies the network as a form of streaming-specific packets. For example, H.264 standard decouples the coding aspect from the bit stream adaptation needed to transmit over a particular channel such as WLAN [Masala et al. 2003].

4.2 Error model and channel estimation

The traffic requires bounded delays, but is usually tolerant of some message loss. Hence, the requirement of these messages includes a guarantee from the system that they should be delivered within their deadlines as long as there is no network error [Arvind et al. 1991]. That is, hard real-time messages must be properly scheduled for transmission, and scheduling messages in a multiple-access network is the function of the MAC protocol [Malcolm and Zhao, 1995].

In the transmission control based real-time network, each node sends its message according to the round robin policy at run time while the central off-line scheduler determines how long a node can continue to access the network. This procedure, named as bandwidth allocation, is based on the traffic characteristics of the given message stream set. Though WLAN can schedule the message transmission according to this strategy, wireless transmissions are subject to interference from outside sources, absorption, scattering, and fading [Eckhardt and Steenkiste, 1998]. While the error correction code such as FEC is able to recover many damaged messages so reduce the probability of message discard, it can not deal with all of the transmission errors. Indeed this error correction mechanism seems to be promising, but information overhead and processing complexity are not negligible. More errors can be overcome by the additional error control procedure.

The networking community has explored a broad spectrum of solutions to deal with both wired and wireless error environments [Kurose et al. 1994]. For a real-time system, however, the traditional error recovery schemes, originally proposed for non-real-time data transmission, may introduce an unacceptably long delay. To the worse, the retransmitted message may interfere or prolong the delivery of other normal messages, resulting in the cascaded deadline miss. Hence, the error control scheme on real-time system should directly consider the time constraint of messages to eliminate a meaningless control step, or error control after deadline expiration.

WLANs may experience location-dependent channel errors, now that some MSs can correctly communicate with the AP, while at the same time others may suffer packet drops due to errors on the channel. In order to address these issues, an independent error model for each communicating pair of stations was introduced. Lost packets constitute one of the main causes of video quality degradation, while the wireless channel error is characterized as bursty and location dependent [Shah et al. 2005]. Due to the delay constraints, the number of retransmission that can be used is limited and usually small. In case of multiple clients, each client will have different channel conditions, processing capabilities, and only limited feedback channel capabilities. The error control scheme for multicast video is not aiming at recovering all lost packets but recovering as many packets as possible. Most importantly, the error control procedure should not affect the other guaranteed traffics in WLAN.

The real-time guarantee inevitably generates overallocation during PCF as the time constraints of message stream are usually described with the maximum value of message size. Retransmission via this overbooked bandwidth does not interfere the transmission of other guaranteed messages. In the other hand, if we let error report containing retransmission request be delivered via the DCF period in a best-effort manner, the entire error control procedure can be carried out without any influence to other real-time messages. Finally, though the variable message size makes it hard to decide when to report the error list, the receivers can determine the completion of message transmission by counting $Beacon$ frame AP generates periodically in WLAN. Based on the requirements described previously, this thesis will propose and analyze the performance of an error control scheme for multicast video stream on IEEE 802.11 WLAN. We focus on the video streaming scenario in the last mile network, namely, between AP and the mobile devices.

We take the estimation method from Bottigliengo's work [Bottigliengo et al. 2004]. Therefore, given N MSs, there are also N independent error models. An 802.11 radio channel is modeled as a Gilbert channel [Bai and Atiquzzaman, 2003], where two states, $good$ and bad, linked with a Marcov chain, represent the state of channel during an 802.11 slot time. The channel condition is estimated as follows: The ACK/NAK is sent from the receiver to AN as soon as it receives a packet. If the AN does not receive an ACK/NAK within predefined time-out interval, the packet will be assumed to be lost. Then the state triggers to bad. AN sets the state to $good$ whenever it receives from the corresponding node, namely, a MAC-layer acknowledgment in response to a data frame, a CTS frame in response to a RTS frame, or any other error-free frame. The AN sets the state to bad after a transmission failure. Each bad channel has its own counter, and when a counter expires, the AN attempts to send a single data frame to check the channel status. The duration of timer is reset to its initial value upon a transmission from bad to $good$, and the value is doubled whenever the probing fails in bad state. The value of timer should be set small so as to quickly recover from short channel error period. This probing procedure is carried out via currently idle link, either uplink or downlink.

We denote the transition probability from state $good$ to state bad by p and the probability from state bad to state $good$ by q as shown in Figure 4.2. The pair of p and q representing a range of channel conditions, has been obtained by using

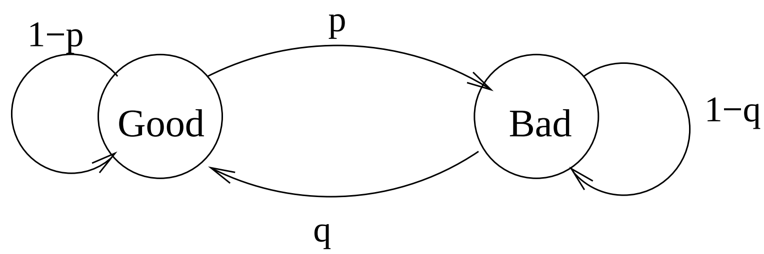


Figure 4.2: Error model

the trace-based channel estimation. e_g denotes the error probability at the $good$ state, while e_b denotes the error probability at the bad state. In addition, denote p_g and p_b as the steady-state probabilities that the channel state is $good$ and bad, respectively. It is straightforward to have $p_g = q/(p+q)$ and $p_b = p/(p+q)$. Consequently, the average error probability of the channel is $P = p_g * e_g + p_b * e_b = (p * e_b + q * e_g)/(p+q)$.

The packet is received correctly if the channel is in state $good$ for the whole duration of packet transmission, otherwise, it is received in error. In addition, according to WLAN standard, poll, transmission, and acknowledgment are atomic, namely, these steps must complete in their entirety to be successful. Senders expect acknowledgment for each transmitted frame and are responsible for retrying the transmission. After all, error detection and recovery is up to the sender station, as positive acknowledgments are the only indication of success. If an acknowledgment is expected but does not arrive, the sender considers the transmission failed.

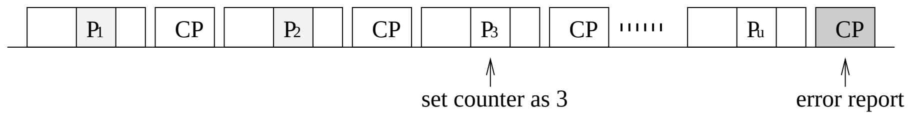


Figure 4.3: An example of error reporting

4.3 Error report and retransmission

A message is divided into packets of size H_i, hence, for a message to be correctly assembled at a receiver, the receiver should receive all packets successfully. If a packet experiences an error during transmission, additional explicit messages should be sent to request the retransmission of that packet. Per-packet response to the receiver increases overhead, but if the receiver reports the list of damaged packets after it estimates the end of delivery of one message, this waste can be eliminated. It is desirable to send error report via the contention period in order not to interfere the transmission of other stream. If the size of every message is fixed and known in advance, the receiver can easily decide the end of one message transmission. However, the size varies message by message in multimedia streams. Hence, we make all packets include a field specifying the number of packets of the message to which they belong so that the receiver can decide when to report error packet list as long as it receives at least one packet.

To construct the error report message, the receiver initializes the error packet list when it receives a packet arrives. If the sequence number of this packet is not 1 but k, the receiver appends the numbers of 1 through $(k-1)$ to the list. From then, the receiver appends each number of the erroneously received packets. As the receiver also hears $Beacon$ and a stream sends a packet per a polling round, it knows the end of transmission even if the last message is omitted. Figure 4.3

shows the example, where a message is transmitted with u packets and each of them has an information that specifies u as well as its sequence number and message identifier. At first, a packet numbered as 3 (not 1) arrives. Then the receiver initializes and adds 1 and 2 to error list as well as sets counter to 3. The counter increases each time the receiver hears $Beacon$ frame. When the counter reaches u, receiver sends error report back to the sender via contention period.

AP receives error report containing error packet list from the receivers of multicast via CP each time it completes a message transmission until it begins a new transmission. AP builds a retry list sorted by the frequency of appearance in the set of error reports. A new error list arrival reorders the list. Then AP resends the packet according to the order in the retry list when it meets an extra slot for the corresponding video stream. The retransmission scheme can be extended to consider the priority of packet given by the coding scheme performed at the upper layer [Lee et al. 2006b].

4.4 Performance measurement result

This section begins with the description of analytic model for the recovered error for the proposed scheme. The average extra bandwidth for a message, R_i, can be calculated as shown in Eq. (4.1).

$$R_i = (\frac{H_i}{F} - \frac{\bar{C}_i}{P_i}) \cdot P_i \tag{4.1}$$

Then the average number of extra access time, w, can be approximated as

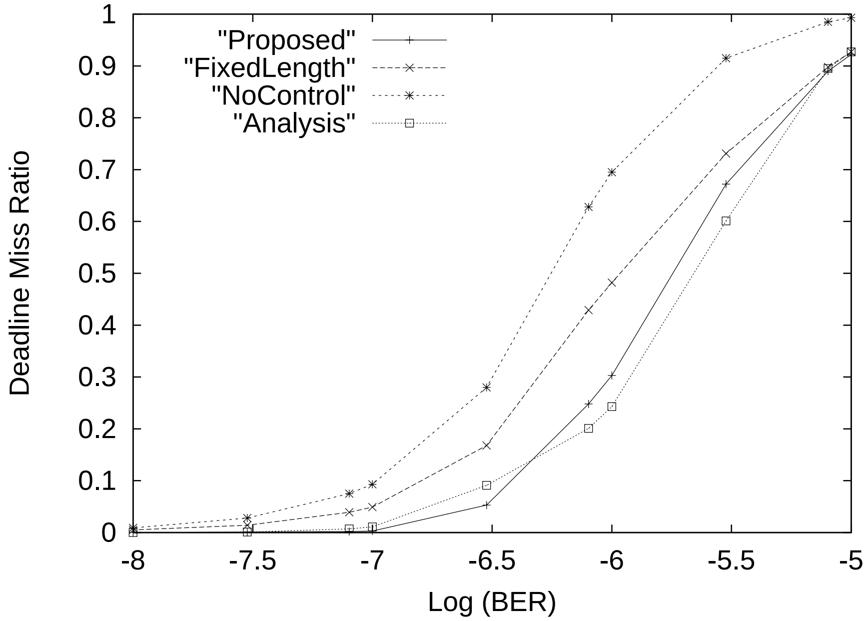


Figure 4.4: Deadline miss ratio vs. Bit error rate

R_i/H_i. If there are w extra frames for a message composed of k frames, the probability of successful transmission of a message, $T(k, w)$, is calculated recursively. Namely,

$$T(k,w) = \sum_{i=0}^{m} {}_kC_i \cdot \epsilon^i (1-\epsilon)^{k-i} \cdot T(i, w-i), T(0,w) = 1, T(k,0) = (1-\epsilon)^k \quad (4.2)$$

, where m is the smaller of k and w while ϵ denotes the frame error rate.

Now we will show the performance measurement results obtained via simulation using ns-2 event scheduler [Fall and Varadhan, 1997]. To begin with, we assume that there are 10 MSs in a cell, 3-5 video streams exist simultaneously. Each video stream has the same traffic requirement for simplicity such as bit rate, error characteristic, video packet size, and the number of receivers. Figure 4.4 and

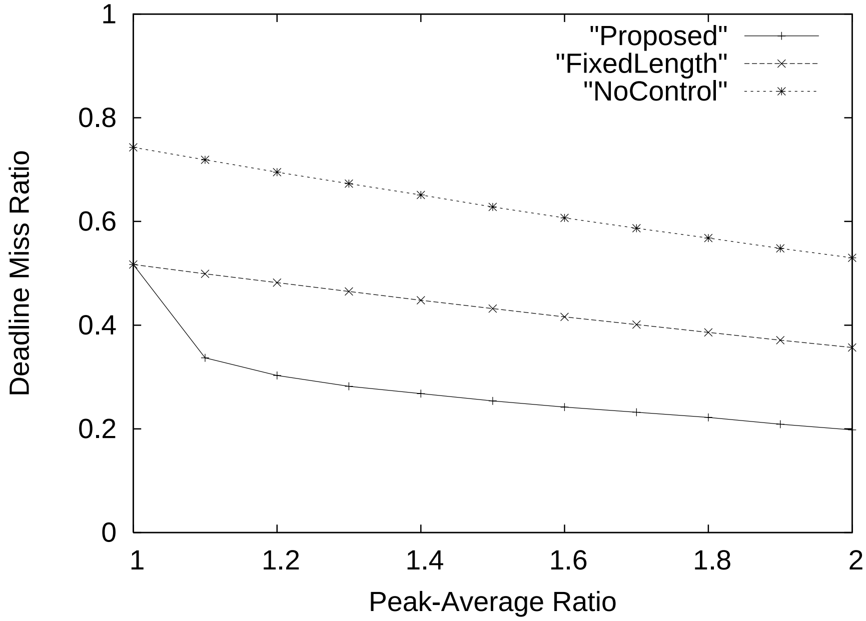


Figure 4.5: Deadline miss ratio vs. Peak-average ratio

Figure 4.5 plots the average deadline miss ratio observed at each receiver according to BER (Bit Error Rate) and peak-average ratio. In these figures, the number of messages not packets is counted. The curve denoted as fixed length is for the case where the receiver only knows the maximum length of message.

Figure 4.4 exhibits that the proposed scheme can reduce the deadline miss ratio by 23 % compared with fixed length scheme and by 48 % compared to no error control case, respectively, when BER is around 10^{-6}. In Figure 4.5, when the peak-average ratio of the message is 1 (all message sizes are equal), both proposed and fixed length schemes show same performance as expected. However, the proposed scheme improves the deadline miss ratio due to the efficient error report mechanism when the ratio increases over 1.0. Other results are omitted due to space limitation. After all, the simulation results demonstrate that overbooked bandwidth can be efficiently used for error recovery of multicast video stream.

Chapter 5

Handoff scheme

5.1 Handoff procedure

With the enormous market potential of telematics industry and the rapid technology progress, automotive telematics has become a hot R&D area in mobile computing and intelligent transportation system [Jung and Koh, 2004]. To support IP mobility is classical but still the most challenging problem [Helmy, 2000] in IP-based mobile/wireless networks, and the prime goal of existing schemes is to minimize handoff latency [Pack and Choi, 2003]. During the handoff procedure, there is a time period during which the MN is unable to send or receive any packets, due to both link switching delay and IP protocol operation [Kang et al. 2005].

According to the OSI model, the handoff procedure is performed at both level 2 and level 3. The layer 2 function is carried out by a MN which changes radio

AP. Whereas, level 3 handoff demands that MN acquire a new IP address, accompanying a series of registration messages, possibly resulting in an interruption of communication service. It is natural that the time of such interruption increases as the number of users increases. This will be very harmful for the real time applications or delay sensitive traffic.

The handoff process can be divided into two logical steps of discovery and re-association [Mishra et al. 2003]. The discovery process involves handoff initiation and scanning phases. As signal strength and signal-to-noise ratio from a station's current AP get weaker, STA loses connectivity and initiates a handoff. Then the client is not able to communicate with its current AP, so the client needs to find the other APs available. This scanning function is performed at a MAC layer, and the station can create the available AP list ordered by the received signal strength.

For the scanning phase, STA can perform scanning operation either in passive or active mode. In passive scan mode, using the information obtained from beacon frames, STA listens to each channel of the physical medium to try and to locate an AP. In the active mode (the wireless NICs do by default), as shown in Figure 5.1, STA broadcasts additional probe packets on each channel and receives responses from APs. Thus the STA actively probes for the APs, and the actual number of messages varies from 3 to 11. Figure 5.2 shows the sequence of messages typically observed during a handoff process. The handoff process starts with the first *probe request* and ends with a *reassociation response* from the new AP.

The probe function follows the IEEE 802.11 MAC active scanning function and the standard specifies a scanning procedure as follows [Mishra et al. 2003]

[Shin et al. 2004] :

1. Using CSMA/CA, acquire the access right to the medium.
2. Transmit a *probe request* containing the broadcast address as destination, SSID, and broadcast BSSID (Basic SSID).
3. Start a *ProbeTimer*.
4. If medium is not busy before the *ProbeTimer* reaches *MinChannelTime*, scan the next channel. Otherwise, process all received *probe responses*.
5. Move to next channel and repeat the above steps.

After all channels have been scanned, information received from *probe response* are scrutinized by STA to select a new AP. Once the STA decides to join a specific AP, authentication messages are exchanged between the STA and the selected AP, and after a successful authentication, the STA sends a reassociation request and expects a reassociation response back from the AP.

5.2 AP scanning procedure

Real-time guarantee is provided based on the worst case available transmission time, so a stream can meet extra slots in some periods. Moreover, as C_i is usually just the upper bound of message size in multimedia applications, some period, P_i, has message to send less than C_i. As a result, a node possibly has no pending message when it receives a poll. Though there have been plenty of bandwidth allocation schemes for the real-time message stream or sensor data stream, we

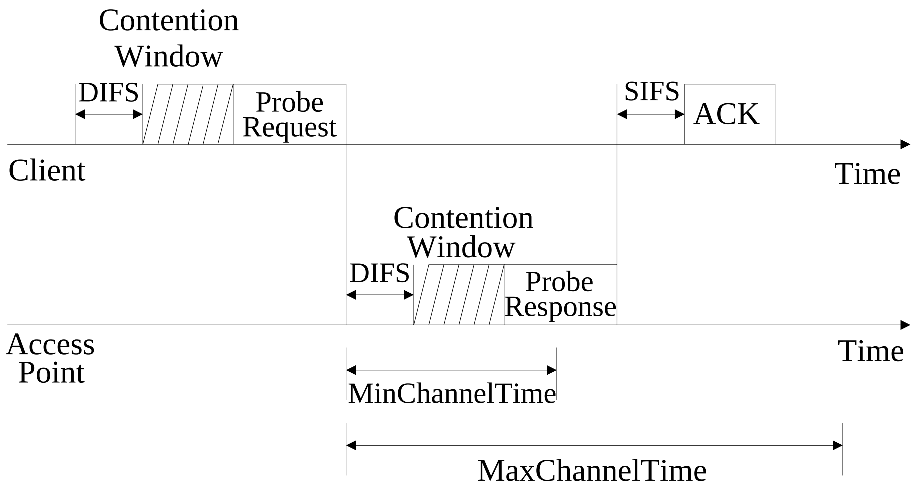


Figure 5.1: Active Scanning

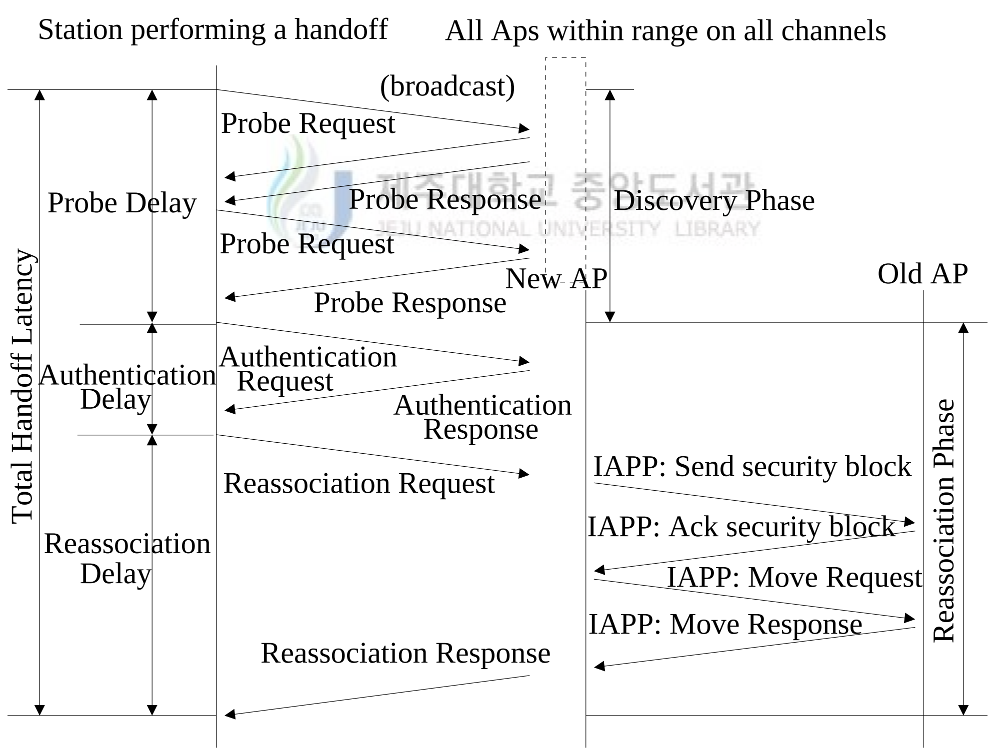


Figure 5.2: Handoff procedures

exploit the scheme of Chapter 3, as it completely conforms to WLAN standard [Kang and Lee, 2004]. For detailed description, refer to Chapter 3.

In general, it takes more time to perform a passive scanning than active scanning in collecting the necessary information. Thus current WLAN equipments use active scanning mode in order to reduce handoff delay. In the standard active scanning procedure, though the probe phase of one channel can be terminated before the *ProbeTimer* reaches *MaxChannelTime*, the STA has to wait during the *MaxChannelTime*. In addition, the *probe response* messages and other ordinary data frames contend for the shared channel, so they can collide with one another. To minimize the collision using active scanning mode, we will assign the higher priority to the *probe response* message, and also provide variable scanning time.

As shown in Figure 5.3 and Figure 5.4, the proposed AP scanning scheme is performed through the unused slots. AP can not only send RTS message to the STA which waits for *probe response* message, but also STA can respond with CTS message to the appropriate AP which will transmit the *probe response* message. Once the RTS/CTS messages are exchanged, the priority to send *probe response* message is assigned to the AP, and other APs can not receive the CTS message until the selected AP sends probe message to the STA.

The procedure of the proposed scheme using adaptive scanning time is as follows [Kang et al. 2006a]:

1. Using CSMA/CA, STA acquires the access right to the medium.

2. STA transmits a *probe request* frame containing the broadcast address as

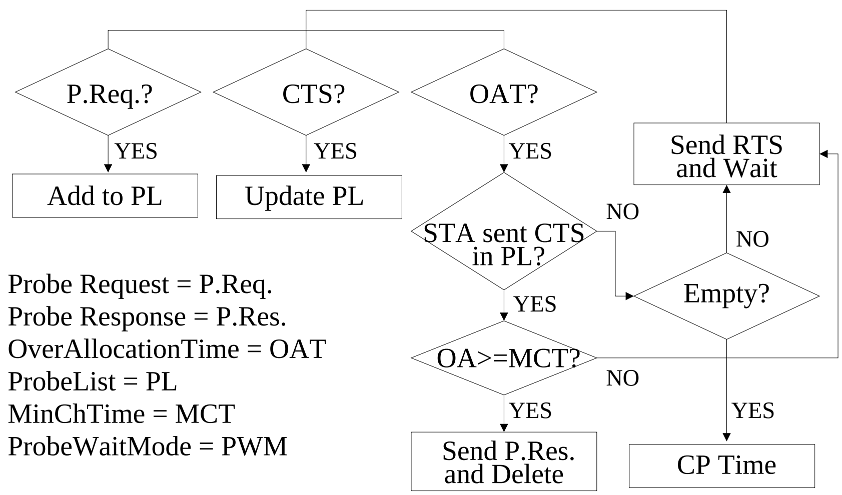


Figure 5.3: AP Process

destination, SSID, and broadcast BSSID, to all APs in the reachable channels. And AP is informed the existence of a STA that waits to join.

3. Start a *ProbeTimer*.

4. If medium is not busy before the *ProbeTimer* reaches *MinChannelTime*, STA scans the next channel. Otherwise, following steps are applied.

5. Using the unused slot, AP sends RTS to STA specifying the ID submitted in step 2.

6. STA responses with CTS if it still wants to receive the *probe response*.

7. AP sends *probe response* to STA if overallocated slot time amount is large enough to send the probe message. Otherwise, *probe response* is postponed until AP meets such a slot.

8. If the number of RTS messages is equal to the number of *probe responses*, and if SSID of being transmitted packet is equal to that of already received

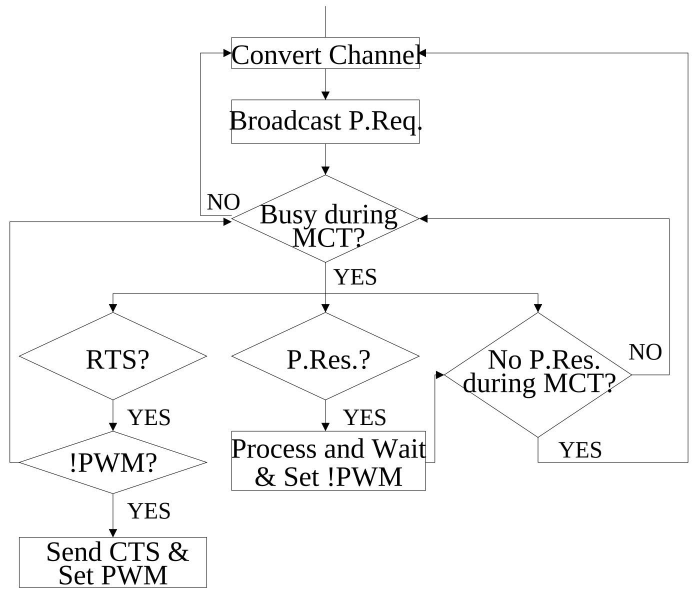


Figure 5.4: STA process

probe response during the *MinChannelTime*, that is, after sensing that the *probe response* doesn't arrive at STA any more, STA scans the next channel.

5.3 Performance measurement result

This section will show performance measurement results performed via simulation. The experiments are based on some assumptions to concentrate on the major performance parameters, namely, the amount of overallocation and the number of pending handoff request. Every stream has equal period and communication time, while each time variable is aligned to F and total number of streams is set to 5 for simplicity. However, these assumption can be easily generalized into the more realistic circumstances. Finally, we compared the handoff time of the proposed scheme with that of a scheme which just gives the precedence to the packets relevant to handoff procedure.

Figure 5.5 shows the effect of overallocated bandwidth to the AP scanning time. The probability of unused slot due to overallocation can be estimated as $(\frac{H_i}{F} - \frac{C_i}{P_i})$. The efficient usage of overallocated bandwidth can speed up the handoff time by 16 % when the overallocation value is 0.1. Figure 5.6 also plots the AP scanning time according to the number of simultaneous requests. The performance gap gets narrow when more handoff requests are submitted to the network as the proposed scheme can expect the improvement only if CFP has an overallocation larger than the handoff procedure. The total AP scanning time can be calculated by the sum of time needed to scan a used channel and time to scan on empty channel.

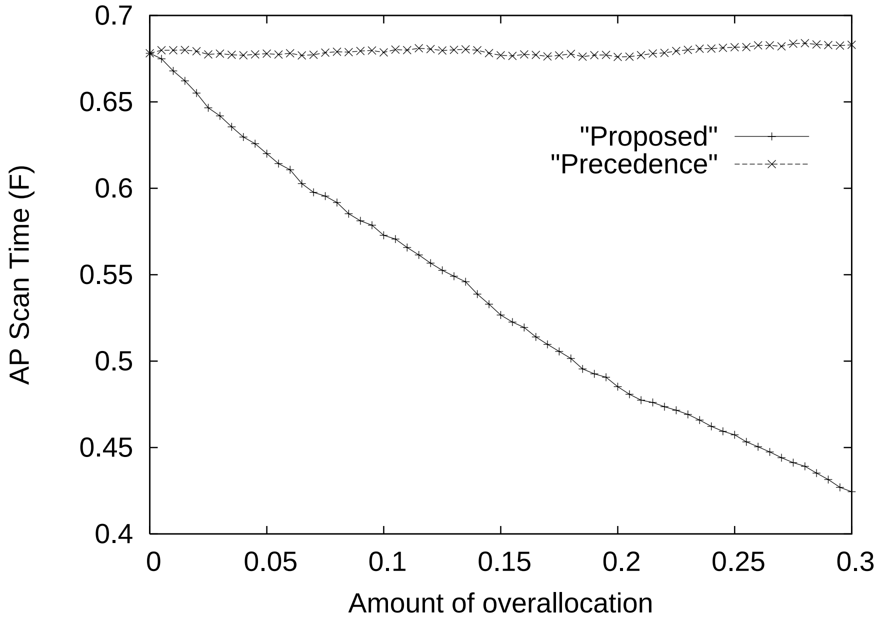


Figure 5.5: Scanning time vs. overallocation

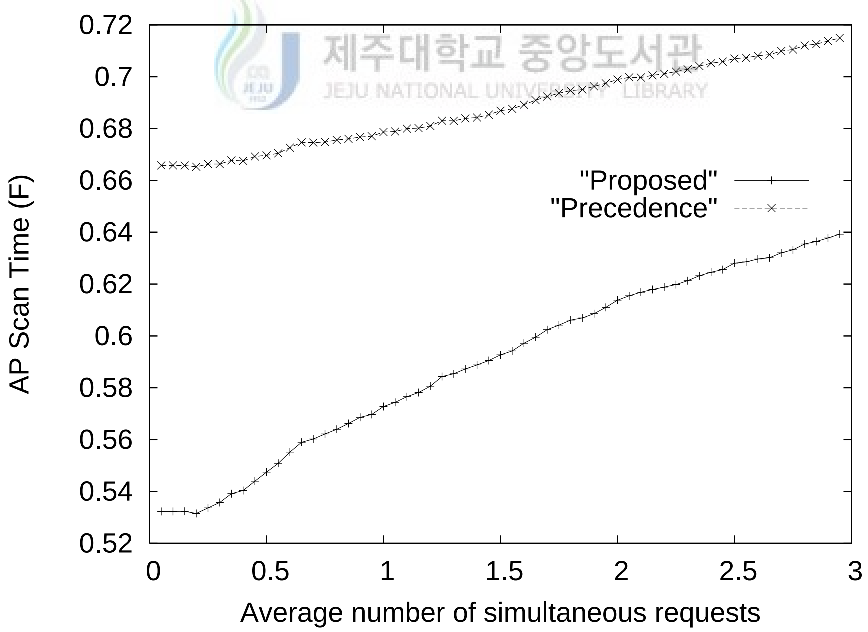


Figure 5.6: Scanning time vs. the number of requests

Chapter 6

Dual channel network schemes

6.1 Dual channel network using bandwidth allocation

6.1.1 Bandwidth allocation procedure

The poor utilization of the polling scheme stems from the fact that a message cannot be transmitted on demand, as there is no way for a node to request a immediate poll to the coordinator. The node should wait up to one superframe time in the worst case. However, dual network architecture can reduce this waiting time by half in round-round style network. Each wireless device has two transceivers, each of the two transceivers communicates on a different channel, and both transceivers can operate simultaneously. It means the environment where each node has two radio and can operate on each channel. Hence, the dual network architecture is

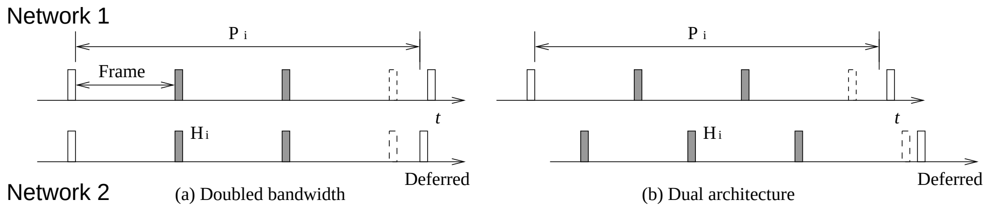


Figure 6.1: The time axis of dual wireless LAN

analogous to the dual processor system, as both network and processor can be considered as an active resource. The jobs or messages are scheduled on two equivalent processors or networks. Priority-driven scheduling on multiple resources induces scheduling anomaly that less tasks can meet their deadlines even with more resources. In addition, it is known that off-line scheduling for this environment is a NP-hard problem [Liu, 2000]. Contrary to priority-driven scheduling, the transmission control scheme proposed in Chapter 3 can easily calculate the feasible and efficient schedule for the dual or multiple resource systems.

It is desirable for the two networks to have a common coordinator that schedules and synchronizes them. A general non-PC node sends its message on any poll from either network. If we make a superframe progress simultaneously or randomly as shown in Figure 6.1(a), S_i may simultaneously lose one access on each network, two in total. Beacon deferment on one channel is independent of that on the other, so the amount of delay is different on each channel. For a period, a node may lose 0, 1, or 2 accesses, and the real-time scheduling system should consider the worst case access scenario. This case is analogous to the case when network bandwidth is just doubled. So X_i is formalized as shown in Eq. (6.1)

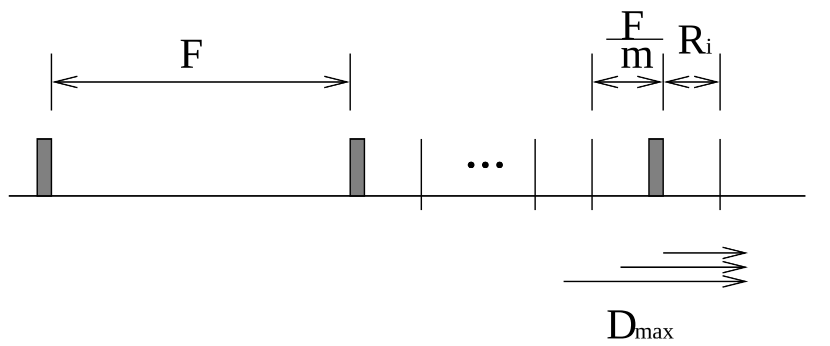


Figure 6.2: Access times in case of m networks

[Lee et al. 2005b].

$$
\begin{aligned}
X_i &= 2 \cdot \lfloor \tfrac{P_i}{F} \rfloor \cdot H_i & if(R_i \geq D_{max}) \\
X_i &= 2 \cdot (\lfloor \tfrac{P_i}{F} \rfloor - 1) \cdot H_i & otherwise
\end{aligned}
\tag{6.1}
$$

On the contrary, if we make one network precede the other by $\frac{F}{2}$ as illustrated in Figure 6.1(b), one loss can be saved, improving the worst case behavior. In addition, the first poll on the second network is not lost, either. As F is reduced by half, so X_i can be calculated by replacing F in Eq. (3.2) with $\frac{F}{2}$. However, though D_{max} is smaller than F, it can be larger than $\frac{F}{2}$. In that case, the second network also loses the last access due to deferred beacon. So X_i is formalized as in Eq. (6.2) [Lee and Kang, 2005d].

$$
\begin{aligned}
X_i &= \lfloor \tfrac{2 \cdot P_i}{F} \rfloor \cdot H_i & if(R_i > D_{max}) \\
X_i &= (\lfloor \tfrac{2 \cdot P_i}{F} \rfloor - 1) \cdot H_i & if(R_i \leq D_{max} \leq R_i + \tfrac{F}{2}) \\
X_i &= (\lfloor \tfrac{2 \cdot P_i}{F} \rfloor - 2) \cdot H_i & if((R_i + \tfrac{F}{2}) \leq D_{max})
\end{aligned}
\tag{6.2}
$$

In addition, Figure 6.2 illustrates how many polls may be lost according to the value of D_{max} when there are m networks.

The figure shows the last two superframes of the first network, with the access times of the other networks also marked in the time axis. The access times are evenly spaced by $\frac{F}{m}$. If R_i is greater than D_{max}, the least bound on the number of access times is not affected irrespective of deferred beacon. However, D_{max} is greater than R_i, one access can be lost if the start of CFP is delayed. One more access is lost when D_{max} exceeds $R_i + \frac{F}{m}$ on the $(m-1)$-th network. After all, each time D_{max} increases by $\frac{F}{m}$, one more access is lost. After all, X_i can be generalized into the case of more than 3 networks.

$$
\begin{array}{ll}
X_i = \lfloor \frac{m \cdot P_i}{F} \rfloor \cdot H_i & if(R_i \geq D_{max}) \\
X_i = (\lfloor \frac{m \cdot P_i}{F} \rfloor - 1 - \lfloor \frac{D_{max}}{\lfloor \frac{F}{m} \rfloor} \rfloor) \cdot H_i & otherwise
\end{array}
\tag{6.3}
$$

6.1.2 Performance measurement result

TDMA-base schemes can be competitors to the proposed scheme in the sense that they can provide hard real-time guarantee [Carley et al. 2003] [Caccamo et al. 2002] [Choi and Shin, 2000], but, the comparison with them is nearly impossible because it requires so many assumptions on slot size, superframe operation, way to handle deferred beacon, and so on. Moreover, TDMA-based schemes generally did not consider CP interval, and their guarantee mechanism assumes the ideal condition of no deferred beacon. We measured the performance of the proposed scheme in the view of schedulability and achievable throughput via simulation using ns-2 event scheduler [Fall and Varadhan, 1997]. In the experiments, every time variable is aligned to the superframe time, and an initiation of superframe is deferred by from 0 to D_{max} exponentially.

For the first experiment on schedulability, we have generated 2000 stream sets

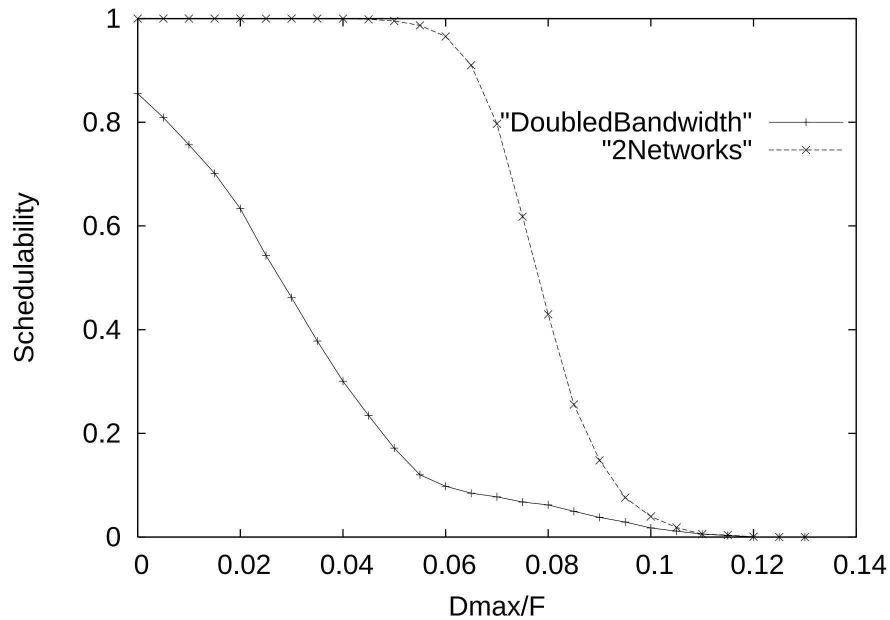


Figure 6.3: Schedulability vs. D_{max}

whose utilization ranges from 0.68 to 0.70 for each network. The number of streams in a set is chosen randomly between 5 and 15. The period of each stream ranges from $5.0F$ to $10.0F$, while its message length from $0.3F$ to $5.0F$. We measured the schedulability, the ratio of schedulable stream sets to all generated sets, changing D_{max} from $0.0F$ to $0.14F$. Figure 6.3 plots schedulability of proposed dual network architecture comparing with that of the network whose bandwidth is just doubled. The schedulability begins to drop abruptly at a certain point for each curve. The proposed scheme can schedule every stream set until D_{max} is $0.042F$ while that of doubled bandwidth network remains at 0.24. That is, the breakdown point shifts to the right, minimizing the protocol overhead of round-robin polling mechanism.

We define *achievable throughput* as the virtual throughput for a given stream

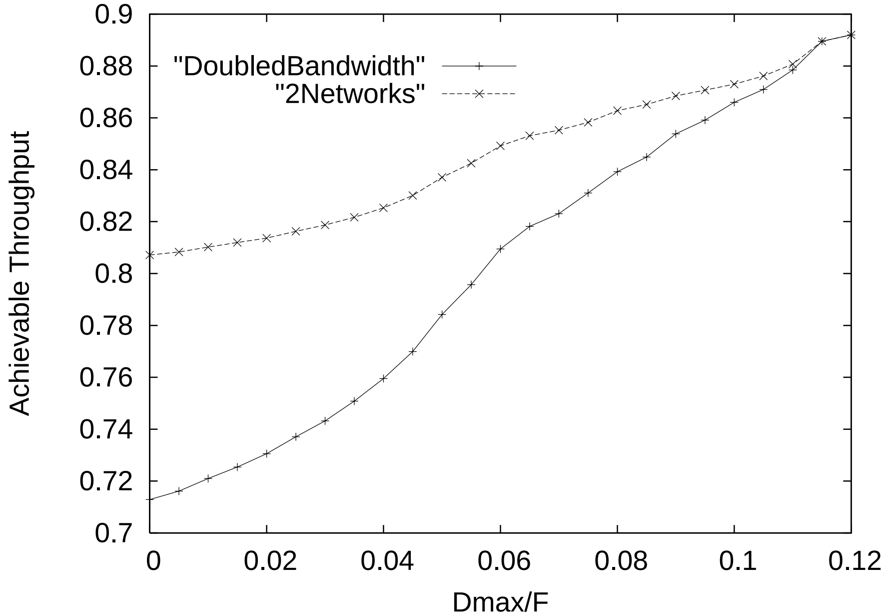


Figure 6.4: Achievable throughput vs. D_{max}

set assuming no collision even in CP. This can be calculated as the sum of utilization of real-time message stream set and ratio of average length of CP to F. Overallocation is the difference between the allocation, $\sum H_i$, and the actual requirement, U. After all, this parameter means how much overallocation is reduced to give more bandwidth to non-real-time messages. We address that our scheme provides tighter and smaller H_i's, as Eq. (6.2) is significantly weaker constraint than Eq. (6.1). According to the increase of D_{max}, more bandwidth is assigned to non-real-time traffic, because D_{max} is its minimum requirement of CP, as shown in Figure 6.4.

6.2 Bandwidth reclaiming scheme

6.2.1 Reclaiming procedure

Network management consists of two parts on real-time communications, namely, static bandwidth allocation and dynamic adaptation parts, respectively [Sheu and Sheu, 2001]. Based on the static information that do not change for a long time, for example, period and maximum transmission time of each stream, the bandwidth allocation procedure determines the network access schedule for the given set of active stations or streams to guarantee that their time constraints are always met as long as there is no network error. In the other hand, the dynamic change in network condition needs further management that can cope with such situations as non-real-time traffic load oscillation, channel status change, and so on. In WLAN, one of the most challenging problems is bandwidth reclaiming scheme that reassigns the network time reserved but left unused to another node [Caccamo et al. 2002]. Apparently, these two parts are closely related with each other and underlying MAC protocol.

The reclaiming scheme is very crucial to the network throughput, as hard real-time guarantee inevitably incurs overbooking, resulting from a pessimistic assumption that the stream has the worst case available time at each period. Moreover, to prevent starvation of stations that are not allowed to send during the CFP, the CP is at least long enough to transmit one maximum MAC protocol data unit, making it possible for a non-real-time message to put off the initiation of a PCF behind the expected start time [Sheu and Sheu, 2001]. This phenomenon called as a *deferred beacon problem* further deteriorates the worst case available time for a time-sensitive stream, increasing the amount of overallocation. In addition, even a station having the right to the network cannot sometimes send its message due to error-fluctuating channel status [Shah et al. 2005].

It is desirable that as much unused bandwidth as possible should be reclaimed and allocated to another (real-time or non-real-time) stream to minimize bandwidth waste and improve the network throughput. To this end, this thesis is to propose and analyze an bandwidth reclaiming scheme for time-sensitive message transmission on WLAN that strictly obeys the IEEE 802.11 standard, assuming that PCF operates according to weighted round robin schedule. This scheme is aiming at maximizing the amount of reclaimed bandwidth by adapting the polling order as well as adjusting the length of CP without affecting the worst case behavior of each stream. The extended CP can improve the response time of messages that belong to error control, connection management, and normal non-time-constrained traffic.

A. Reclaiming test

Hard real-time guarantee is provided based on the worst case available transmission time as described in the previous section, so a stream can meet extra slots in some periods. Moreover, as C_i is just the upper bound of message size, some period has message to send less than C_i. So a node possibly have no pending message when it receives a poll and in that case it responds with a null frame containing no payload. Naturally, how to cope with this unused slot is critical to the network utilization. For example, FRASH reclaims such unused slots to reassign to a aperiodic server, aiming at improving the responsiveness of non-real-time messages. Whenever the transmission of the current dispatched message is over

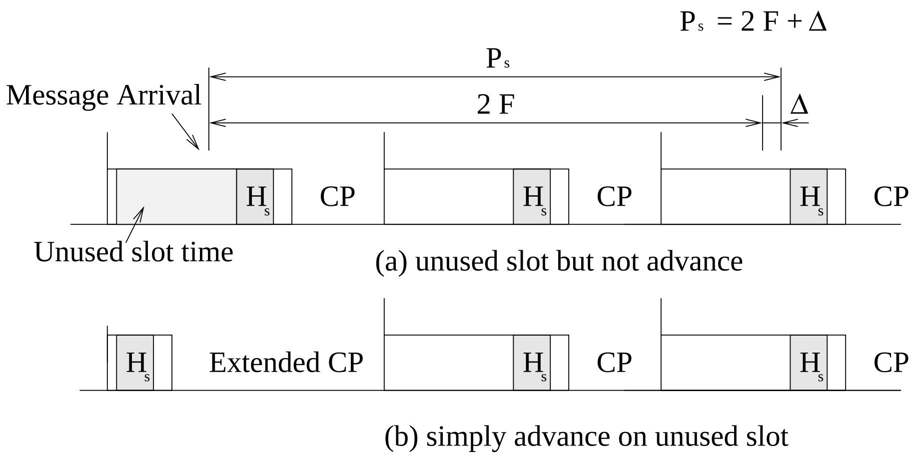


Figure 6.5: Bandwidth reclaiming

and it does not use all the reserved frames, its identifier is put in a field in the header of the last data packet of the current message. The identifier should be correctly received by all stations in the network to alleviate unnecessary contention on the slot. Furthermore, FRASH can perform properly only for TDMA protocols that operate on fixed size slots. The first step to reclaim the bandwidth is to determine whether to advance the rest of the polling or leave the slot unused.

To begin with, let's assume that if a slot is unused, AP simply shifts ahead the rest of subsequent polls. Figure 6.5 shows the example in which the predecessors of S_s generate unused slots. Figure 6.5(b) illustrates the unused slots are reclaimed, CFP terminates earlier than scheduled, and finally the CP is extended for more non-real-time message transmission. However, this method may deprive S_s of one access and the real-time guarantee can be broken. If we let $P_s = k \cdot F + \Delta$, where Δ is chosen from 0 to F, then the least bound of network access within P_s is k or $k-1$ according to whether $\Delta > D_{max}$ or not, as noted in Eq. (3.2). (The figure shows the case of $k = 2$). If AP simply moves ahead

H_s, S_s loses scheduled access as shown in Figure 6.5(b), in case the new arrival of message falls in between moved and original polling instants. As contrast, that access can be saved if the AP does not reclaim the unused bandwidth as shown in Figure 6.5(a).

In the contrary, the remaining part of polling schedule can be advanced within that superframe without violating the time constraint if all the subsequent streams have their messages to send, that is, if all of the remaining slots are not the first one of their period. As the PC can complete the polling schedule of that superframe, the length of CP can be extended to transmit more non-real-time messages and finally improve network utilization. In addition, we can increase the probability of bandwidth reclaiming by rearranging the polling order. The frequency of first slot gets higher as the stream has shorter period. Therefore, it is desirable for PC to poll first the stream with shorter period [Kang and Lee, 2006c].

The main idea of proposed scheme is that the rest of polling schedule can be advanced without violating the time constraint if all the subsequent streams have their messages to send, that is, if none of them are waiting for a new message arrival. As the PC can finish the polling schedule of that round earlier than the original CFP duration, CP can be extended to transmit more non-real-time messages. As the AP receives all the information on period and transmission time before bandwidth allocation, it can estimate the status of each stream, namely, whether its transmission buffer is empty or not [Caccamo et al. 2002].

B. Polling order rearrangement

Polling order is important not only in deciding a stream will be affected by a deferred beacon but also in improving the probability of reclaiming. Considering the reclaiming ratio, the stream which has a shorter period had better not have its predecessor, due to the fact that as a stream has empty buffer frequently, it makes it hard to reclaim the unused slot of its predecessor. After all, as the period is shorter, its stream is polled first. In addition, the more a stream generates unused slots, it would be better to put the stream in the latter place, as small number of successor increases the probability of being reclaimed. How much a stream generates unused slot depends on the amount of overallocated bandwidth. This amount is calculated by subtracting the actual bandwidth requirement, $\frac{C_i}{P_i}$, from allocated bandwidth, $\frac{H_i}{F}$. As a result, the overallocation, O_i, is calculated as in Eq. (6.4).

$$O_i = \frac{H_i}{F} - \frac{\bar{C}_i}{P_i} \tag{6.4}$$

With this information, the polling order should be decided such that the stream with larger O_i is placed at the rear of polling list. The two criteria may conflict, for example, a stream of short period have much overallocation. However, the more important of the two is the overallocation factor. Hence, we classify the streams into two groups according to the degree of overbooking. The first group members rarely generates unused slots and they are put into the front part of polling list after being sorted by their periods. In the other hand, the second group generates relatively more unused slots, so they had better be ordered by the degree of overallocation. Hence, more over allocated stream has fewer successors.

Finally, the stream which has higher error rate brings more unused slots, so it seems better to place such a stream to the latter part. However, the error dynamics, conforming to Guilbert model, are so unpredictable that the average behavior cannot provide meaningful criteria [Shah et al. 2005]. If we are to consider the error characteristics, the channel probing mechanism should be reinforced to the reclaiming scheme.

6.2.2 Performance measurement result

This section measures the performance of the proposed reclaiming scheme via simulation using NS-2 event scheduler [Fall and Varadhan, 1997]. The experiments focus on measuring the achievable throughput to demonstrate the effectiveness of reclaiming scheme. We define *achievable throughput* as the virtual throughput for a given stream set without any collision even in CP. This can be estimated as the sum of both utilization of real-time message streams and ratio of average length of CP to F.

Figure 6.6 plots achievable bandwidth according to the average number of streams on the superframe to evaluate the performance of reclaiming scheme. Without overallocation caused by the hard real-time guarantee, the only waste is polling overhead, but overallocation makes the throughput much less than ideal. However, the resource reclaiming scheme can narrow the gap between those two curves, that is, considerably relieves the problem of poor utilization of PCF operation, as shown in Figure 6.6. The amount of overallocation does not depend on the number of streams but how much F is harmonic with each P_i. For the experiment, 200 stream sets are generated for each number of streams ranging from 2 to 20 with utilization between 0.64 and 0.65. At last, it is certain that the improvement increases when the number of streams is small, and the 52.3 % of waste was recovered. As shown in the figure, the improvement gets smaller as the number of streams increases. This is due to the fact that the reclaimed portion gets smaller and the probability of reclaiming decreases.

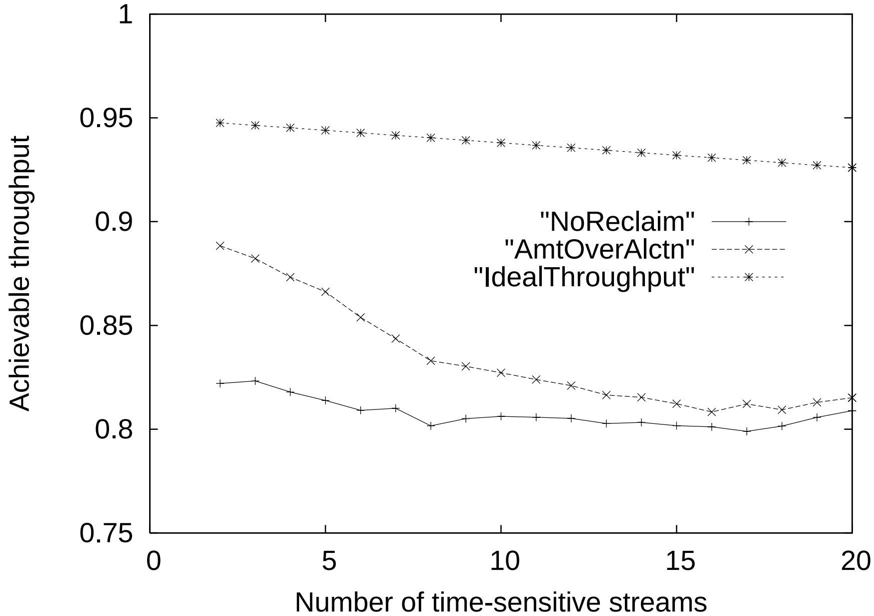


Figure 6.6: Throughput vs. the number of streams

Figure 6.7 plots the reclaimed throughput measured by changing the utilization of stream set from 0.5 to 0.8, while the number of streams randomly distributes from 2 to 10. If only a stream set has a feasible schedule, the throughput goes high as utilization increases. On the contrary, reclaimed scheme provides stable throughput throughout the given utilization range. When the utilization is from 0.5 to 0.65, about 75.3 % of waste was reclaimed.

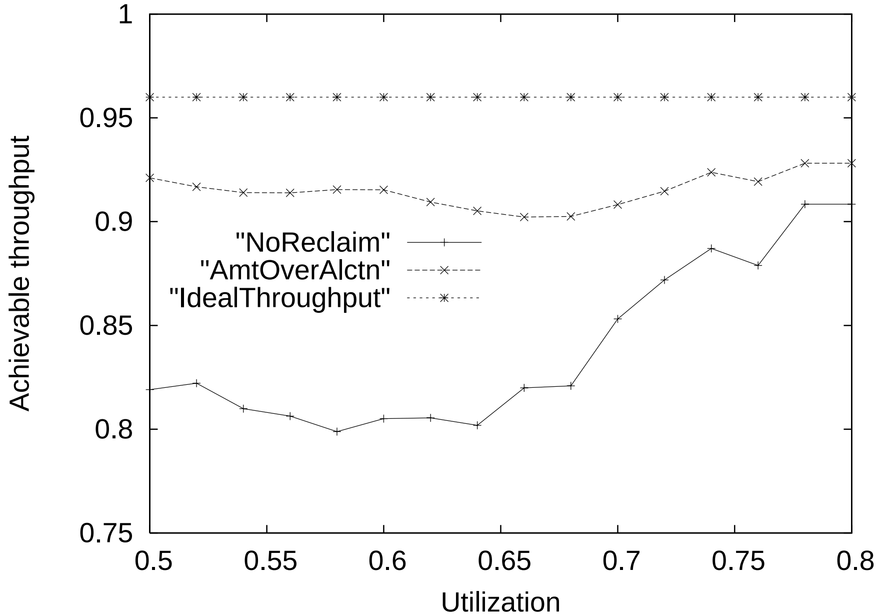


Figure 6.7: Throughput vs. utilization

6.3 Dual channel network based on fixed sized slots

6.3.1 Bandwidth allocation procedure

As the most prominent dynamic priority scheduling mechanism for the uniprocessor real-time system, EDF algorithm assigns priorities to individual jobs in the tasks according to their absolute deadlines. The schedulability of message streams can be tested by the following sufficient condition [Caccamo et al. 2002]:

$$\forall_i,\ 1 \le i \le n, \quad \sum_{k=1}^{i} \frac{C_k}{P_k} + \delta \le 1.0 \tag{6.5}$$

, which assumes that all the messages are sorted by increasing relative deadlines and that there are n streams, while P_i and C_i denote the period and maximum transmission time of stream S_i, respectively. The δ denotes the overhead term originated from the network management such as polling/probing overhead, beacon packet broadcast, interframe space, and so on.

Since the invocation behavior of a set of periodic tasks repeats itself once every T time units, where T, called the $planning\ cycle$ of the task set, is the least common multiple of the periods of all periodic tasks, we only need to consider all the task invocation in a planning cycle. Let $< f_i^1, f_i^2 >$ be the i-th slot assignments of channel 1 and channel 2. If f_i^1 and f_i^2 are allocated to different streams, say A and B, respectively, their transmission channels can be switched without violating their time constraints. We define a $switchable\ pair$ if f_i^1 and f_i^2 are allocated to different streams. The $< f_i^1, f_i^2 >$ is also a switchable pair if any one of f_i^1 and f_i^2 is left unassigned. The purpose of bandwidth allocation, or slot assignment is to maximize the number of switchable pairs, as it can overcome channel errors.

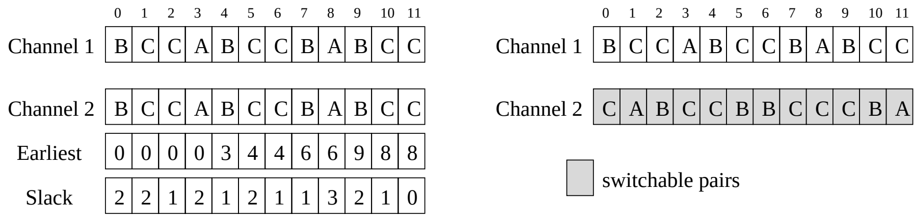
To inherit the optimality of EDF in a single resource system, the allocation scheme first partitions the given stream set into two identical sets so that each of them has the same period set but the transmission time of every stream is reduced by half. Namely,

$$\Theta : \{(P_i, C_i)\} \rightarrow \Theta_1 : \{(P_i, \frac{C_i}{2})\}, \Theta_2 : \{(P_i, \frac{C_i}{2})\} \tag{6.6}$$

Next, the schedule for Θ_1 and Θ_2 is determined by EDF policy, both schedules being identical. And then, the allocation in Θ_2 is rearranged to maximize the number of switchable pairs. When the allocation scheme generates the schedule of Θ_2, it also creates the list of range to which an allocation can be migrated. The earliest time of movement, E_t, is the arrival time of message that occupies slot t, while the amount of backward movement is marked as its laxity, L_t. The E_t and L_t of unassigned slot are set to 0 and T, respectively, as it can be relocated anywhere within the planning cycle. From the last slot, f_t^1 and f_t^2 are checked whether they are equal, namely, they are allocated to the same station. If so, the rearrangement procedure attempts to change f_t^2 as follows [Lee et al. 2006e]:

```
for slot i from E_t to t
  if (f_i^2 == f_t^2) continue; // same station
  if (L_i + i < t) continue; // cannot be deferred
  else exchange f_i^2 and f_t^2 and break;
```

[**Example 1**] This example shows the bandwidth allocation for the given stream set. The stream set consists of 3 streams, A(6,2), B(3,2), and C(4,4). Their utilization is 2.0, the length of planning cycle being 12. The schedule for both networks is identical as they follow the same EDF scheduling policy after partitioning the stream set into Θ_1 : {(6,1), (3,1), (4,2)} and Θ_2 : {(6,1), (3,1), (4,2)} as shown in Figure 6.8(a). The figure also shows that the earliest relocatable slot and slack time by which the allocation can be deferred. The rearrangement procedure begins from slot 11 backward to slot 0. As shown in Figure 6.8(a), f_{11}^1 and f_{11}^2 are both C, so it is desirable to relocate C in f_{11}^2. Among slots from 8 (decided by E_{11}) to 11, as f_8^2 is A and $L_8 + 8 \geq t$, f_8^2 and f_{11}^2 are exchanged, making $< f_{11}^1$, $f_{11}^2 >$ a switchable pair. This procedure will be repeated up to slot 0 and Figure 6.8(b) shows the final allocation. In this example, every slot pair turned into the switchable one.



(a) Step 1 : partition (b) Step 2 : rearrangement

Figure 6.8: Scheduling procedure

6.3.2 Runtime scheduling and error control

Before polling a station, the PC transmits a probing control packet to the scheduled station, which then returns the control packet to the PC. If the PC does not receive the probing control packet correctly from the station, the channel is estimated to be bad. Even though the probing indicates the channel is good, the ensuing transmission can fail if a state transits to the bad state during the transmission. For simplicity, this thesis disregards the overhead terms including channel probing, polling overhead, mandatory ACK/NAK, and so on, but the amount is bounded and constant for all slots, so it can be merged into transmission time.

Let's assume that PC is to start slot i which is originally allocated to A on channel 1 as well as B on channel 2, namely, $< A$, $B >$. PC first probes the channel status from itself to A and B on all two channels. Table 6. 1 shows the probing result and corresponding actions. As shown in row 1 (case No.1), PC can reach A on channel 1 and also B on channel 2, so PC polls each station as scheduled. In row 2, by switching polls between the two channels, PC can save the 2 transmissions that might fail on ordinary schedule. PC can reversely reach A only on channel 2 while B on channel 1. In row 3, all connections from PC are bad except the one to A through channel 2. If PC switch $< A$, $B >$ to $< B$,

No.	Ch1−A	Ch2−B	Ch1−B	Ch2−A	Ch1	Ch2	save
1	Good	Good	X	X	A	B	0
2	Bad	Bad	Good	Good	B	A	2
3	Bad	Bad	Bad	Good	−	A	1
4	Good	Bad	Good	Bad	A	−	0
5	Bad	Bad	Good	Bad	B	−	1
6	Bad	Good	Bad	Good	−	B	0
7	Bad	Bad	Bad	Bad	−	−	0

X : don't care

Table 6.1: Channel status and transmission

$A >$, only A can send on channel 2. In row 4, only A can send on channel 1 as scheduled. In row 7, all scheduled connections from PC are bad.

The polling table in Table 6.1 has some entries marked as '-', which means PC cannot poll A or B. To improve the network throughput, it seems better to poll another station. Let $S_i^{n,a}$ be channel status of allocated message a (A, B, or C) on channel n (1 or 2) at i-th slot time. According to the following procedure, some entries marked as '-', can be filled with switching slot. The next pending message whose arrival time, E_i, lies prior to the current time, and slot time lies prior to the deadline of current bad channel message, will be picked and tried [Kang and Lee, 2006d].

if (${S_t}^{1,a1}$ and ${S_t}^{2,a2}$ are *Good*) continue;
else if (${S_t}^{1,a2}$ and ${S_t}^{2,a1}$ are *Good*)
    exchange ${f_t}^1$ and ${f_t}^2$ and break;
else
    if (there is a good pair when ${f_t}^1$ is switched with ${f_t}^2$)
        allocate corresponding one good pair (channel-message);
    for slot i from $(t+1)$ to $(t+L_t)$ // bad time slot t
        if (${f_i}^n$ == f_t^n) continue; // same station
        if ($E_i^n > t$) continue; // cannot be advanced
        else exchange f_i^n and f_t^n and break;
    if ($S_t^{bn,ba}$ is still *Bad*) allocate failed message.

[**Example 2**] This example shows the runtime scheduling for the given stream set. As shown in Figure 6.9, the channel status and transmission are as follows: In slot time 0 (case No.1), PC can reach B on channel 1 and also C on channel 2, so PC can poll as scheduled. In slot time 1 (case No.2), C and A messages can meet time constraints by switching polls between the two channels. In slot time 3 and 10 (case No.3 and No.4), corresponding message can improve network throughput by exchanging messages between the two channels or two slots satisfying the condition.

The slot length, say L, is as large as the basic unit of wireless data transmission and every traffic is also segmented to fit the slot size. Each invocation needs $\lceil \frac{C_i}{L} \rceil$ slots, so every packet should be received correctly to be reassembled at the receiver. When the failed packets are managed and allocated to reclaim the unused slot, we consider the probability that the entire message can be correctly received. The only one message is selected among the failed messages according to the

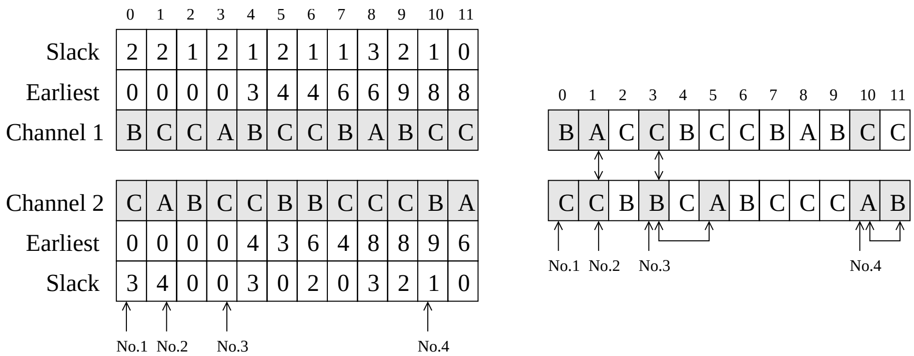


Figure 6.9: Runtime scheduling procedure

following rules: It has the higher probability of the success ratio by being allocated one more slot. If we let N be the number of successfully transmitted slots until now, the message whose $\frac{N}{\lceil \frac{C_i}{L} \rceil}$ value is the largest one, is selected.

6.3.3 Performance measurement result

This section measures the performance of the proposed scheme in terms of deadline meet ratio according to the packet error rate via simulation using ns-2 event scheduler [Fall and Varadhan, 1997]. We fixed the length of planning cycle to 24 as well as the number of streams to 3, and generated every possible stream sets whose utilization ranges from 0.2 to 2.0, to measure how many the rearrangement scheme can improve the utilization.

Figure 6.10 plots the measurement result sorted by the utilization of stream sets. Even when the utilization is 2.0, 17 out of 24 slots are rearranged to switchable pairs on average. As the utilization gets lower, the number of switchable pairs increases, since the number of unassigned slots also grows. Global EDF generates

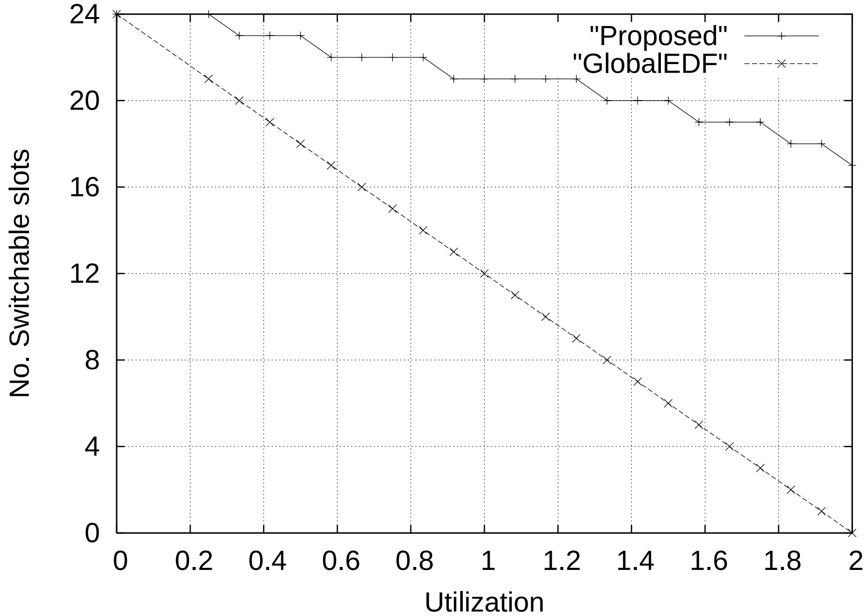


Figure 6.10: Number of switchable slots

the switchable pair only by unassigned slots, so the number of switchable pairs proportionally decreases as the utilization increases. The gap between the two schemes becomes large on the higher utilization.

Figure 6.11 shows the deadline meet ratio according to the packet error rate to demonstrate the effect of runtime slot rearrangement. The curve marked as *Switch* represents the result of switching pairs and the *Reclaim* curve shows the effect of reclaiming unused slots. The success ratio means the probability of timely delivery of real-time packet, so it is the probability that the channel remains good during whole slot time. The packet error rate is the function of packet length and bit error rate, and it ranges from 0.0 to 0.5, considering the error-prone wireless channel characteristics. As shown in Figure 6.11, the deadline meet ratio is improved by around 12.5 % when the packet error rate is over 0.4 and the number of

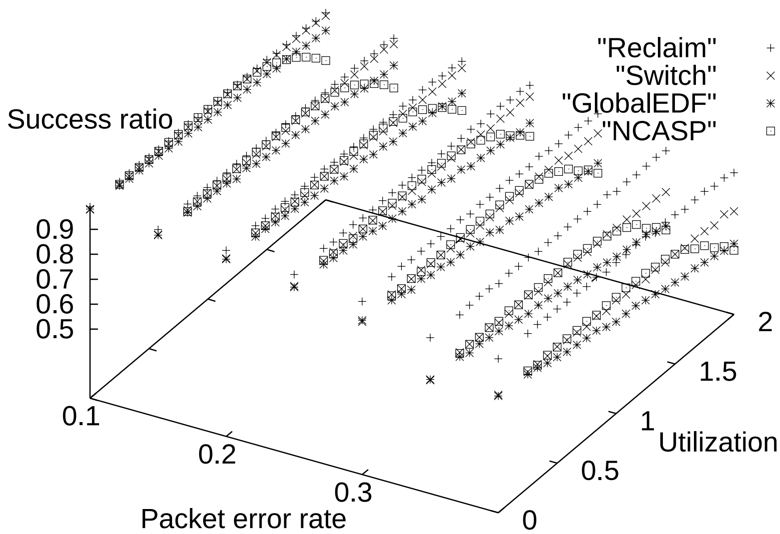


Figure 6.11: Success ratio based on packet error rate and utilization

switchable pairs is 24. The performance gap is extended according to the increase of packet error rate. Figure 6.12 shows the result of runtime scheduling and error control scheme based on the example described in section 4.

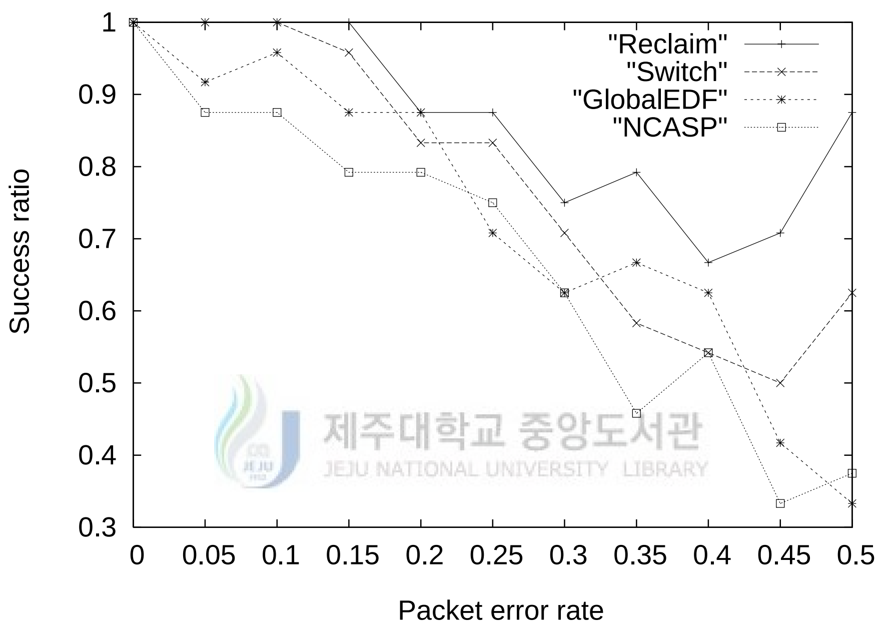


Figure 6.12: Success ratio based on packet error rate

Chapter 7

Conclusions

In this thesis, we have addressed the problems of designing bandwidth management scheme for the real-time traffic on multiple IEEE 802.11 WLANs, by determining capacity vector and using the overallocated bandwidth over the multiple networks. The followings summarize the main results obtained from the thesis:

A. Bandwidth management scheme

The deferred beacon problem is a serious hindrance to the real-time guarantee in the IEEE 802.11 WLAN standard. This thesis has proposed and analyzed an hard real-time guarantee scheme based on the round robin style polling mechanism on the wireless network that conforms to the standard. The key idea is to decide the polling schedule for CFP in the form of capacity vector for the given network parameters and time constraints, after analyzing the effect of deferred beacon to the hard real-time message stream. The simulation results show that the proposed scheme can not only enhance the schedulability of wireless network by up to 18 % but also give more bandwidth to the non-real-time traffic up to 5.3 %.

B. Error control scheme

We have proposed and analyzed the performance of an error control scheme for multicast video streaming over IEEE 802.11 WLAN. The proposed error control scheme makes the receiver node report errors in a best-effort manner via CP, and also makes AP retransmit the requested packets through the overallocated slot that is indispensably brought by QoS guarantee in CFP. Therefore, this scheme is able to eliminate the interference to the guaranteed stream transmission, while the message size contained in each frame enables the timely report of error list as long as at least a packet of message arrives at the receiver. A continuous trace of each video stream channel shows that the proposed scheme can not only efficiently utilize the network bandwidth, but also reduce the deadline miss ratio by 23 % compared with fixed length scheme and by 48 % compared with no error control case, respectively.

C. Handoff scheme

We have proposed and analyzed a fast handoff scheme that exploits the overallocated bandwidth inevitably generated to guarantee the QoS requirement of real-time multimedia stream on the IEEE 802.11 WLANs. Using the reserved but not used network time, AP and STA can exchange RTS/CTS to negotiate when to send probe message, making AP immediately respond to the probe request with probe response message in the next CFP. In addition, by making the priority of the probe frame higher than any other data frames, collision of probe response messages and ordinary data frames can be minimized. Simulation results show that the proposed scheme improves the AP scanning time according to the amount of overallocation and average number of simultaneous requests. The efficient usage of overallocated bandwidth can speed up the handoff time by 16 % when the overallocation value is 0.1.

D. Dual channel network scheme

- This thesis has proposed and evaluated a bandwidth allocation scheme that enhances the schedulability of hard real-time streams via dual wireless network architecture, strictly observing IEEE 802.11 WLAN standard. By exploiting the round-robin style polling mechanism, network schedule and bandwidth allocation become much simpler, and they can check efficiently whether a stream is affected by deferred beacons. Additionally, by making the start times of both networks different by $\frac{F}{2}$ under the control of common coordinator, the proposed scheme improves the least bound of network access time for each message stream. Simulation results show that the capacity vector decided by the proposed scheme schedules more stream sets by up to 36 % than the network whose bandwidth is just doubled, and that it can accommodate 9 % more non-real-time message to the network, for the given experiment parameters.

- We also have proposed and analyzed a bandwidth reclaiming scheme that can overcome poor utilization problem of PCF for real-time communication in WLAN. When an unused slot occurs, AP tests whether it can be reclaimed by checking all of its successors have messages to send. This test confirms that the other time-sensitive traffics are not affected by the early termination of polling round. The reclaimed bandwidth is reassigned to CP to improve the response time of connection management, error control, and other non-real-time messages. The simulation results show that the proposed scheme is able to reclaim up to 52.3 % of bandwidth waste when the number of streams is 2 and that it also provides stable throughput throughout the utilization from 0.5 to 0.8.

- This thesis has proposed and analyzed the performance of a real-time guarantee scheme for time-sensitive applications using message scheduling and error control on the dual wireless ad hoc sensor network. Basically, simultaneous use of dual channels increases the available bandwidth of the network and addresses the ever-increasing demand for higher bandwidth. After dividing the network time into fixed size slots synchronized across the two channels, polling schedule is determined according to the optimal EDF policy. In addition, slot rearrangement scheme makes it possible to maximize the number of switchable pairs by enabling the coordinator to select an error-free channel, and rearrange schedule order by using the reserved but not used channel, resulting in enhancing the capability of coping with instability of wireless channels. Simulation results show that the proposed scheme can generate 70 % of switchable pairs even when the utilization reaches the saturation point, improving the ratio of timely delivery by up to 12.5 % when the packet error rate exceeds 0.4.

Throughout this thesis, we have assumed that the deadline miss ratio and packet collisions are the major bottleneck on the performance of wireless sensor network, since wireless channels are subject to unpredictable *location-dependent* and *bursty* errors. Another system resource such as embedded system power should be effectively managed and also treated carefully in the wireless sensor network. The effective management of them remains to be solved in the future.

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초 록

최근 무선 통신 기술의 발전과 이에 따른 관심이 증폭되고 무선 근거리 통신망과 같은 무선 매체의 속도와 수용력이 증가됨에 따라 시간에 민감한 고대역폭 응용에 대한 요구도 늘어나고 있다. 마감시한 내에 메시지가 전송되어야 하는 경성 실시간 통신의 중요성은 날로 증대되고 있으나 경성 실시간 메시지의 손실은 시스템 자체 또는 실행 결과에 치명적인 오류를 초래하게 된다. 결국 실시간 스트림은 시간 제약조건을 만족시키는 하부 네트워크로부터의 보장을 필요로 하게 됨에 따라 본 논문에서는 다중 IEEE 802.11 무선 근거리 통신망에서의 실시간 트래픽을 위한 효율적인 대역폭 관리 기법을 제안하고 그 성능을 분석한다.

본 논문은 크게 네 개의 주제, 즉 대역폭 관리 기법, 멀티캐스트 오류 제어 기법, 핸드오프 기법 및 이중 네트워크 관리 기법으로 구성된다. 첫 번째 파트에서는 대역폭 할당 기법을 제안하고 분석한다. 대역폭 할당 기법은 비실시간 메시지의 간섭으로 인해 발생되는 비콘 연기 현상을 고려함으로써 용량 벡터로 표현되는 효율적인 라운드 로빈 폴링 스케쥴링 기법을 제시한다.

두 번째 파트에서는 무선 근거리 통신망에서의 멀티캐스트 스트리밍을 위한 오류 제어 기법을 제안하고 분석한다. 이 기법은 모든 패킷들이 자신이 속한 메시지의 패킷 번호를 나타내는 필드를 포함하게 하여, 수신 노드가 경쟁 주기를 통하여 최선 노력 (best-effort) 방식으로 오류를 보고하도록 한다. 그 결과 액세스 포인트는 QoS (Quality of Service) 보장을 위해 불가피하게 발생되는 과할당된 슬롯을 통하여 요청된 패킷을 재전송할 수 있다.

세 번째 파트에서는 실시간 멀티미디어 스트림의 QoS 요건을 보장해주기 위해 역시 과할당 대역폭을 활용하는 빠른 핸드오프 기법을 제안하고 분석한다. 사전에 예약되었으나 사용되지 않은 네트워크 시간을 활용하고 일반 데이터 프레임보다 프로브 프레임의 우선순위를 높임으로써, 액세스 포인트와 스테이션은 RTS (Request To Send) / CTS (Clear To Send) 메시지를 통해 프로브 메시지 전송 시간을 미리 정하여, 액세스 포인트가 비경쟁주기 동안 프로브 요청 메시지에 대해 즉시 응답해 줄 수 있도록 한다.

마지막 파트에서는 이중 무선 근거리 통신망에서의 분산 경성 실시간 통신을 위한 메시지 스케쥴링, 대역폭 환수 및 오류제어 기법에 초점을 둔다. 첫째, 한 네트워크의 수퍼프레임을 다른 네트워크보다 반주기만큼 선행하게 함으로써 이중 네트워크 구조는 비콘 연기 현상의 영향을 최소화시킬 수 있고 대기 시간을 반으로 줄일 수 있다. 둘째, 대역폭 환수 기법은 다른 실시간 메시지에 전혀 영향을 주지 않고 과할당 슬롯에 비실시간 트래픽을 할당하여 경쟁 주기를 확장함으로써, 실시간 QoS를 보장하기 위해 발생되는 대역폭 낭비를 최소화하여 네트워크 성능뿐만 아니라 실시간 메시지의 적시성을 향상시킬 수 있다. 또한 환수 확률을 높이기 위해서는 과할당 정도에 따라 폴링 순서를 재조정한다. 셋째, 전송시간을 균등하게 분할함으로써, 동기화된 고정 크기의 슬롯에 기반을 둔 이중 채널을 이용한 EDF (Earliest Deadline First) 스케쥴링 기법을 사용한다. 또한 조정자가 현재의 채널 상태에 따라 두 채널 사이에 메시지를 동적으로 교환할 수 있도록 하는 슬롯 재할당 기법을 제안하여 메시지 적시 전송의 확률을 높인다.

대역폭 할당 기법의 기본 아이디어는 경성 실시간 메시지 스트림에 대한 비콘 연기 현상의 영향을 분석한 후, 주어진 네트워크 인자 및 시간 제약조건에 대응되는 비경쟁주기 폴링 스케쥴을 용량 벡터의 형태로 결정하는 것이다. 실험

결과는 제안된 대역폭 관리 기법이 주어진 실시간 스트림 집합에 대하여 최대 18%까지 스케쥴 가능성을 향상시킬 수 있음을 보였다. 오류 제어 기법은 낭비되는 대역폭을 재활용함으로써 네트워크 대역폭을 효율적으로 활용할 수 있을 뿐만 아니라 다른 스트림에 전혀 영향을 주지 않으면서 기존의 오류 제어 기법과 비교하여 마감시한 만족도를 23% 향상시켰다. 제안된 핸드오프 기법은 과할당된 정도와 평균 핸드오프 동시 요청 수에 따른 액세스 포인트의 스캔 시간을 실험하였으며 그 결과 스캔 시간을 최대 16% 까지 최소화할 수 있음을 보였다. 또한 이중 채널 네트워크 기법은 단지 대역폭이 두 배인 네트워크와 비교하여 실시간 메시지 스케쥴 가능성을 36% 향상시키고 비실시간 메시지에게 9% 더 많은 대역폭을 할당함으로써 처리량을 향상시킬 수 있었으며, 대역폭 환수 기법은 대역폭 낭비를 최대 52.3% 까지 개선할 수 있음을 보였다. 전송시간을 균등하게 분할함으로써 재할당 기법은 사용율이 포화 상태에 이를 때에도 채널 사이에 전체 메시지의 70% 까지 교환이 가능했으며 패킷 오류율이 0.4 이상이면 종료시한 만족도 (deadline meet ratio)를 12.5% 까지 향상시켰다.

감사의 글

학과 1회 신입생으로 입학한 것이 엊그제 같은데 벌써 박사학위 논문이 결실을 맺게 되었습니다. 이렇게 빠르게 느껴지는 것은 학교생활이 저에게는 무척 소중하고 값진 것이었기 때문일 것입니다. 그 동안 저에게 많은 도움을 주신 분들에게 이 지면을 빌어 감사의 말씀을 드립니다.

먼저 이 논문이 완성되기까지 부족한 저에게 많은 관심과 격려로 오랜 시간 늘 변함없이 따뜻하게 지도해 주시며 저에게 귀감이 되어 주신 이정훈 교수님께 진심으로 감사 드립니다. 제게 가장 큰 발전과 발돋움의 기반을 만들어 주신 교수님의 은혜 잊지 않겠습니다. 그리고 제가 폭넓은 식견을 가질 수 있도록 도움을 주셨을 뿐만 아니라 든든한 후원을 베풀어 주셨던 박경린 교수님께도 감사 드립니다. 또한 학부시절부터 줄곧 깊은 관심과 배려로 보살펴 주셨던 김철수 교수님, 김익찬 교수님, 이봉규 교수님께 감사 드리며, 김철민 교수님을 비롯하여 바쁘신 중에도 미흡한 저의 논문 심사를 기꺼이 맡아 주셨던 심사위원분들께 감사 드리는 마음 금할 길 없습니다. 그 동안 저에게 진심 어린 충고와 격려를 해주셔서 고맙습니다.

늘 함께 하며 많은 것을 공유했던 ITRC 센터 연구원들과 학교생활 내내 지원을 아끼지 않아 주었던 학과 조교, 그리고 학과 후배들에게도 고마움을 전합니다. 더불어 제가 힘들 때마다 위로가 될 수 있었던 친구들과 작은세상 회원들을 비롯하여 항상 가까이에서 저를 지켜봐 주신 많은 분들께 고마움을 전하며 기쁨을 함께 나누고 싶습니다.

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앞으로도 이 모든 분들의 사랑과 기대에 어긋나지 않도록 하루하루 노력하는 사람이 되겠습니다. 감사합니다.

2006년 6월

강미경 올림