Guaranteed Non Dropping Mechanism Deploying IEEE802.11e Standard

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Abstract

Recently, many people enjoy accessing wireless network to conveniently collect their required information or happily surf within the Internet worldwide by exploiting wireless mobility, flexibility and availability. However, the more sessions accessing the channel, the more packet dropping rate. In this paper, we propose a polling access control scheme which in its PCF mode deploys non-preemptive priority to efficiently transfer voice traffic which characterized by packet rate of voice source and the maximum tolerable jitter (packet delay variation) is forecasted. We also record the scheduling results in a queue, with which AP (Access Point) can poll and then enable mobile users to communicate with their opposite sites. This occurrence solves the problem that some voice packets do not suit QoS in IEEE 802.11e standard with multi-polling. While no voice packet can be transmitted, the scheme changes to DCF mode to transfer data packets. Furthermore we simulate and analyze the performance of the scheme in a WLAN environment. The experimental results show that our mechanism significantly improves packet dropping rate.

Keywords：IEEE802.11e, VoWLAN, QoS, MultiPolling, PCF

1 Introduction

Wireless LAN technology, as growing by leaps and bounds, is now rapidly becoming a crucial part of computer networks. It significantly attracts interests both in academic and industry communities. The finalization of the IEEE 802.11 wireless LAN standard has emerged wireless technology from the world of proprietary implementation to become an open solution for providing mobility as well as essential network services where wire line installations proved impractical. Now companies and organizations are investing in wireless network development at a higher rate to take advantage of mobile and real-time access to information. Net access through hotspots at airports, hotels, and coffee shops, via the high-speed wireless Internet access service known as WiFi, is also rapidly becoming common.

According to the estimate of Killen & Associate[1], the international phone calls using IP Internet for transferring VON (Voice on the Net) in 1997 were less than one percent of people in the world, but would dramatically increase to almost 50% in 2005. eTForecasts pointed out the number of wireless networks was 265 Million in 2003, and would rise to 714 Million by 2006. Therefore, we can predict that Internet phone calls will grow enormously, and a large number of the calls will be through wireless networks.

IEEE 802.11 is designed for best effort service only. The lack of a built-in mechanism to support real-time services makes it very difficult to guarantee quality-of-service(QoS) for throughput-sensitive and delay-sensitive multimedia applications. Therefore, modifying current 802.11 standards is necessary. Although the 802.11e standard is to provide QoS support to WLAN applications, how to choose right MAC parameters and QoS mechanism to improve QoS in 802.11 networks still remains unsolved. Besides, the process of creating a well-defined standard might be too slow for us to wait for it to be ratified. Hence, a guaranteed QoS in 802.11 Wireless LANs is still a challenge and needs further study.

2 Background

2.1 Components of WLAN Hardware Architecture

The IEEE 802.11 WLAN is composed of the following components, as shown in Fig. 2.1.

(1) Station (STA)：Any equipment with IEEE 802.11 MAC layer and physical layer interfaces.

(2) Basic Service Area (BSA)：A geometrical area that contains WLAN basic architecture. The size of a BSA is determined by the environment and efficiency of the particular wireless station which can detect the signal or access the channel.

(3) Basic Sercice Set (BSS)：Collection of all stations in a BSA.

(4) Distribution System (DS)：Usually a DS consists of LAN networks that connect several BSAs together.

(5) Access Point (AP)： A station able to access DS and STAs. Usually a BSA has only one AP.

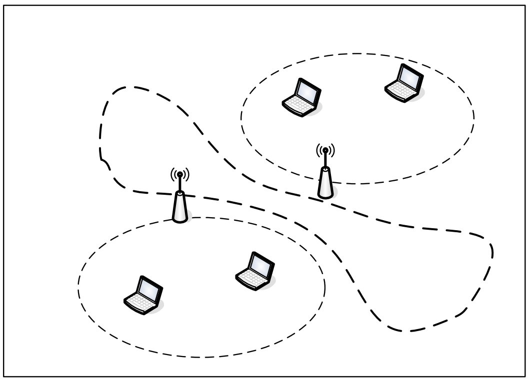

Fig. 2.1 Infrastructure of WLAN

2.2 IEEE 802.11 Topology

Components of the IEEE 802.11 mechanism interact with each other to enable wireless LAN station mobility transparent to higher protocol layers, such as the LLC. A station is any device containing functionality of the 802.11 protocol, such as the MAC layer and the PHY layer, and an interface to a wireless medium. BSS provides a coverage area where stations of the BSS are fully connected. A station can freely move within the BSS, but can not directly communicate with other stations if it leaves the BSS

2.3 IEEE 802.11 MAC Protocol

IEEE 802.11 MAC provides two main access methods, Distributed Coordination Function (DCF) and Point Coordination Function (PCF). Coordination Function is a mechanism that coordinates when a station can start transmitting data. DCF is a basic access method, which primarily deploys Carrier-Sense Multiple Access/Collision Avoidance (CSMA/CA) to enable a station to send and receive non-synchronous data. CSMA/CA can be used in Ad Hoc and WLAN Infrastructure as well. PCF, a contention free method, enables stations to send and receive time bounded data, no packet collision may occur. However, PCF can only be employed in certain basic WLAN frameworks, e.g. WLANs containing AP.

2.4 IEEE 802.11e

IEEE 802.11e deploys Hybrid Coordination Function (HCF) as its medium access protocol. HCF in turn uses Contention-Based and Controlled Channel Accesses as its channel allocation strategies. The former is an Enhanced DCF (EDCF), and the latter an enhanced PCF.

2.5 Related Work

Several solutions have been proposed[2-7]. Some give voice packets a higher priority over data packets to shorten VoIP packets' waiting time. Others suggest transferring voice packets under the DCF contention mode with some special mechanism in order to meet the real-time requirement. But most compromise their service quality due to packet loss and delay.

Fig.2.2 shows an example of packet transmission during contention free period. When PCF starts, AP first detects whether the channel is free. If yes, AP waits a period, say PIFS (Point Inter Frame Spacing), and then sends a Beacon signal, indicating that the channel will switch from DCF to PCF mode. As no station has voice data to be transferred, AP sends a CF_End signal to terminate PCF and switches back to DCF mode.

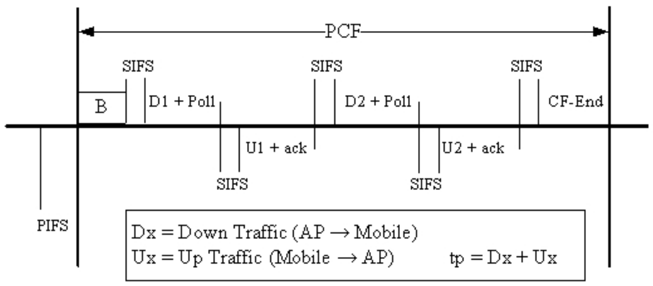

Fig. 2.2 The example of packet transmission during Contention-Free Period

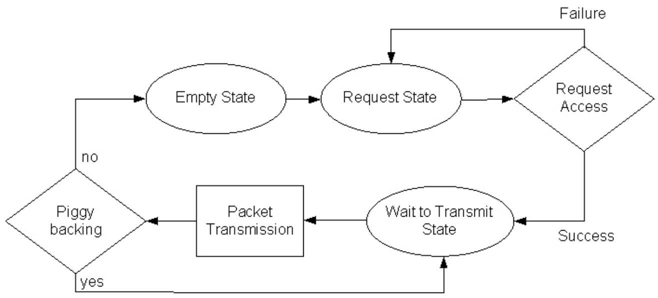

Fig. 2.3 Channel model of a real-time station

PCF's channel efficiency is usually poor owing to too many failed polls, particularly when most stations transfer no packets. Therefore, both of PCF and DCF are not suitable for voice transmission. In order to improve the quality of such transmission, we propose the following solution: build a Token Buffer within AP, dynamically assign priority to each voice packet according to its parameters, establish a transmission Polling List (PL) based on the parameters, and finally follow the PL to multi-poll stations.

A station ready to transfer real-time data has three different states: Empty, Request, and Wait to Transmit (WTT). The first two and the third respectively exist under the DCF and the PCF modes. When the Ethernet card transfer buffer of a station, say S, is initially empty, S is in the Empty state. When one or more packets are generated and placed in the buffer, S enters the Request state. As S already in the Request state, its state will remain unchanged. During Contention Period, S in Request state will request AP to compute if the request is acceptable. If yes, AP replies an ACK, otherwise ignores this request and S repeatedly request to send until accepting an ACK. S then enters the WTT state to wait for being polled by AP. Once polling S, AP reserves a time slot for S. S can send packets within the slot. If the transmitted packet is not the last one of underlying session, its PGBK bit = 1, and S remains in its WTT state. Otherwise, PGBK bit = 0 and S will return to the Empty state to wait for next session. Fig. 2.3 shows the state change of a station.

3 Improved Approach

For each real-time station S, we use two variables, r_c and δ, to represent its transmission characteristics. r_c is the packet transfer rate, and δ the maximum amount of jitter (i.e. packet delay variation) allowed for a specific packet. Transmitting voice data too fast and too slow both should be avoided. In other words, each packet of S should inherit r_c and δ from its voice source, i.e., S.

In the BSA of IEEE 802.11, our AP reserves some of its memory to create token buckets, each represents a real time session that connects two stations, say A and B, and is generated when A or B enters the WTT state. A packet with relatively smaller jitter has lower priority, telling AP to place its token to a lower priority bucket.

3.1 Theoretical Discussion

Assume there are n voice sources and their characteristic parameters are (r_{ci}, δ_i), $i = 1, 2, \cdots, n$. The maximum waiting time of a token T, from the time point T's corresponding packet P arrives at transfer buffer to P is delivered, is δ_i^*. According to theorem 1, each packet can be delivered within δ_i.

Theorem 1：

Let $\delta_1^* = 2 \cdot \mathrm{SIFS} + \mathrm{CFPoll} + \mathrm{t_p} + \mathrm{ACK}$, and

$$\delta_i^* = (2 \cdot \mathrm{SIFS} + \mathrm{CFPoll} + \mathrm{t_p} + \mathrm{ACK})$$

$$+ \sum_{k=1}^{i-1} \left\lceil \frac{\mathrm{r_{ck}}}{\mathrm{r_{ci}}} \right\rceil \cdot (2 \cdot \mathrm{SIFS} + \mathrm{CFPoll} + \mathrm{t_p} + \mathrm{ACK})$$

$i = 2, \cdots, n$, and t_p is the time needed to transmit and/or receive a packet.

If $\delta_i^* < \frac{1}{r_{ci}}$ and $\delta_i^* \le \delta_i$ then all voice packets of session I can be transmitted within their jitter constraints, $i = 2, \cdots, n$ [8].

Theorem 2:

Suppose n voice sources are scheduled in the given priority order. The average waiting time is minimized for voice packets if $r_{ci} \le r_{cj}$ for all i<j[8].

3.2 Proposed Scheme

The improvement is as follows:

1. As shown in Fig. 3.1, if accepting the request of a new voice source P in the previous DCF mode (i.e. a successful connection), AP will build a new token bucket in its buffer for P, and assign a priority to it based on P's tolerated jitter. AP scans token buckets according to their priorities. When a token T with parameter r_c appears, AP removes T, reserves a time slot, polls and requests T's station to immediately transfer voice packet.

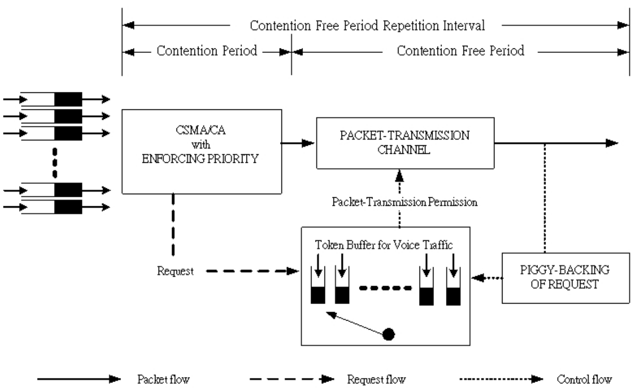

Fig. 3.1 Proposed packet transmit-permission policy

2. Based on the IEEE 802.11b protocol, under the PCF mode, the station when polled must wait a period of time SIFS before transferring its packet. Therefore, as piggyback indicates that underlying session does not terminate, AP produces a new token every $\frac{1}{r_c}$.

   However, AP needs SIFS + CFPoll to poll a station which requires SIFS + ACK to respond. Therefore, in the same connection, the time duration after removing T to producing the next token is $\frac{1}{r_c} - (2SIFS + CFPoll + t_p + ACK) \cdot$

3. When the underlying session is ready to close, the piggybacking bit = 0, i.e., End-of-file. AP removes the corresponding bucket.
4. When all buckets are temporarily empty, AP checks if there is enough time to run DCF mode before next token T arrives. If yes, it sends a CF-End frame to end CFP and enters CP mode, otherwise waits for T.

We also found that we can apply some basic principles to improve Multi Polling of IEEE 802.11e protocol without changing its framework.

Our improvement is as follows:

1. Our design is based on a hypothetically perfect environment. AP creates a PL (recall, polling list) in its buffer to arrange the order and relative time of packet transmission, according to the parameters (r_{ci}, δ_i) of packets having arrived at AP's transfer buffer. It finally broadcasts the PL to all stations of underlying BSA.

Theorem 3:

n voice sources with r_{ci} and δ_i, i=1,2,3,…,n, are given. There exists a cycle LCT = L.C.M. (The Least Common Multiple) $\left[\frac{1}{r_{c1}}, \frac{1}{r_{c2}}, \ldots\ldots\ldots\ldots, \frac{1}{r_{cn}}\right]$ within which the amount of transmitted packets is $\sum_{i=1}^{n} (r_{ci} \cdot LCT)$.

Proof:

The proof is trivial since based on LCM's definition, every LCM $\left[\frac{1}{r_{c1}}, \frac{1}{r_{c2}}, \ldots\ldots\ldots\ldots, \frac{1}{r_{cn}}\right]$ packet production sequence repeats. Q.E.D.

When no common factor exists among $\frac{1}{r_{c1}}, \frac{1}{r_{c2}}, \ldots\ldots\ldots\ldots, and\ \frac{1}{r_{cnc}}$ LCT will be the maximum, i.e., $(\frac{1}{r_{c1}} \cdot \frac{1}{r_{c2}} \cdot \ldots\ldots\ldots \cdot \frac{1}{r_{cn}}) = \prod_{i=1}^{n} \frac{1}{r_{ci}}$.

The number of packets generated in max (LCT) will be

$$(r_{c1} + r_{c2} + r_{c3} + \ldots\ldots + r_{cn})\ LCT$$

Theorem 3 depicts that L.C.M. $\left[\frac{1}{r_{cA}}, \frac{1}{r_{cB}}, \frac{1}{r_{cC}}, \ldots, \frac{1}{r_{cn}}\right]$ forms a cycle if n stations are now in underlying BSS. Theorem 1 shows that we can predict the transmission time point of the next generated packet for each session with the defined parameters as long as the transmitted packet is not the last one. We therefore record only one cycle in stead of representing entire schedule.

2. In a BSA, stations follow their current PL to transmit voice packets. Normally, AP and stations monitor if the sequence is correct or not. If any discrepancy or collision occurs, e.g., a session is newly established or disconnected, a station crashes or follows out-of-date PL, AP updates its PL if needed and again broadcasts it to all stations which will then follow the new PL. AP needs not poll stations one by one, thus significantly saving polling time.

4 Simulations

In the following, we evaluate the performance of the proposed scheme.

4.1 Simulation Environment

Any station in a BSS can directly communicate with other stations. The basic assumptions of our simulation environment are as follows.

Two types of traffic are considered.

(1) Pure data The arrival of data frames from a station's higher-layer to MAC sublayer is Possion. Frame length is assumed to be exponentially distributed with mean length 1024 octets.

(2) Voice traffic We use the mio8380's[9] built-in audio codec which is based on GSM610 format to generate voice traffic patterns. Frames of voice traffic that are not successfully transmitted within their maximum jitter constraint are assumed to be lost.

4.2 Simulation Results

The simulation is performed on different numbers of sessions. Some packets issued from different sessions but generated at the same time are scheduled to a sequence of time slots.

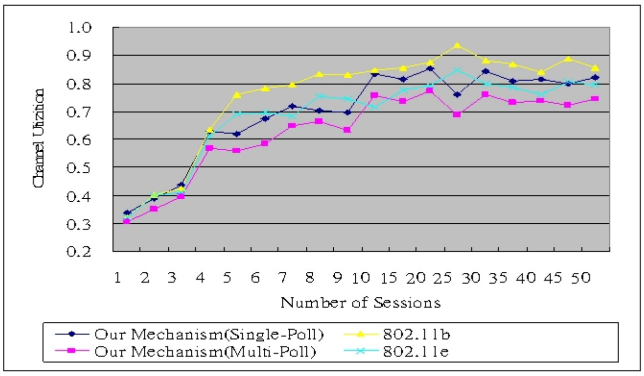

Fig 4.1 Channel Utilization Rate for all sessions

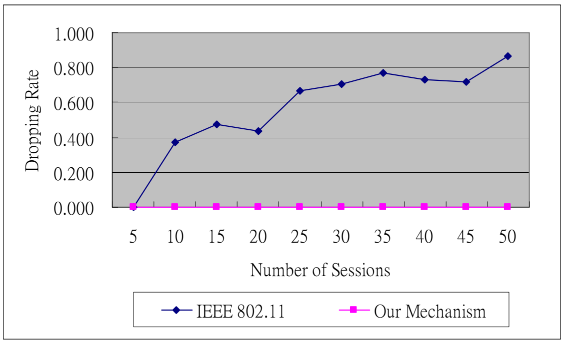

Fig 4.2 Dropping Rate for all sessions

Channel utilization of our mechanism may be sometimes not as high as that of the IEEE 802.11 standard. However, it can dynamically adjust the channel utilization via Theorem 1, to control the bandwidth usage.

Dropping rate is significantly less than the IEEE 802.11 standard. Since no matter in single-polling or multi-polling, our mechanism checks if a request is acceptable or not, in order to prevent dropping. The reason is that voice packets are delay and dropping sensitive.

5 Conclusion and Future Research

Based on previous simulation, our mechanism has significantly improved packet dropping rate. But owing to system checking if collision may occur or not, unaccepted stations hence repeatedly request AP resulting in connection delay.

Our mechanism is flexible, intelligent and able to control current network bandwidth using token buffer. The flexibility of controlling sessions ensures collision free. An analogy would be placing traffic lights at intersections merging into a highway to regulate vehicles entering the highway group by group, instead of all at once during rush hours. Another purpose is to control the total number of vehicles traveling on the highway to efficiently maintain a reasonable minimum driving speed. The actual bandwidth is not necessarily full, and therefore the packet transfer rates and tolerated jitters can be both fulfilled. Jitters of multiple sessions being simultaneously small will cause a jam, forcing token buffer to be full much earlier. However, the bandwidth is not fully occupied, theoretically allowing more tolerated jitters of large size. Session jitters of simultaneously large will also cause a jam. Same as small, our mechanism can accept more tolerated jitters of small size. In summary, the capacity of pairing the packet transfer rates and tolerated jitters is a tradeoff but critical importance.

We have found a way to group and manage all connections using tolerated jitter. Connections of the similar jitter value are classified into a single group. At any moment, the number of sessions coming from a group is limited, thus allowing more session to be established, and then maximize bandwidth utilization. Efforts are continuing, and we hope that further experiments and research will shed light onto this issue to improve the currently limited capacity of the token buffer.

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