Improving the Multimedia traffic Performance over Wireless LAN

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Abstract—**In order to grapple with the continuously increasing demand on the multimedia traffic over high speed wireless LAN, we need a QoS-oriented MAC for wireless multimedia system. In this thesis, we will propose a efficient centralized scheduled access MAC approach in IEEE 802.11 to satisfy multimedia traffic. There are two main algorithms to achieve the goal, queue management algorithm for downlink and uplink to satisfy video traffic's high bandwidth requirement, polling list management method enhance the efficiency of the polling scheme for transmitting voice traffic. The result shows that the centralized scheduled access MAC approach we propose has excellent efficient and compelling performance.**

***Index Terms*—Wireless LAN, QoS, Multimedia traffic, DCF, PCF.**

I. INTRODUCTION

In recent years, the development of the wireless local network is very mature, there are many products have developed on the ISM bands 900MHz and 2.4GHz currently, and these products have broadly applied in many areas and used in many places. Today WLANs are becoming more widely recognized as a general-purpose connectivity alternative for a broad range of business customers. As the interest in broadband multimedia communications involving digital audio and video grows, a number of researchers have been looking into ways of providing QoS guarantees in wired point-to-point WLANs and LANs. Then a WLANs or a cell with quality-of-service guarantees for various types of traffic is considered. In wireless multimedia networks, mobile stations will be capable of generating a heterogeneous traffic mix and therefore it is critical to devise an efficient bandwidth allocation scheme to satisfy the quality of service requirements of each traffic class. Until now, there are many researches on voice transmission schemes over 802.11 to support real - time service. Successfully delivering voice over wireless LANs (VoWLAN) presents a tremendous opportunity; however, implementing the products is not as straightforward a task as it may first appear. Four possible VoWLAN schemes exist. They are based on Distributed Coordination Function (DCF) [2], Point Coordination Function (PCF) [3][4], Priority Queuing and Black-burst scheme. In the paper [8], performance comparison of these schemes is shown.

For the issues addressed above, we use the popular wireless local network IEEE 802.11 as the research object, to address a serial method and algorithm to make the standard to satisfy the multimedia traffic (multi QoS).

II. OBJECTIVES AND METHODOLOGY

2.1 Problem Definition

There are not many researches looking at multimedia or other traffic needing QoS using IEEE 802.11's PCF. Dr. Baldwin did investigate extensions to DCF to better support real-time traffic. They have some new interest in QoS support using LAN features, including IEEE802.1p/q and IEEE 802.11 PCF. But, we have yet to start work on this problem. They think it is promising.

Fundamentally, application-level QoS is an end-to-end problem but upper layer protocols must rely on QoS-related services and functions of lower level protocols.

The PCF of the MAC protocol defined in the 802.11 standard uses a polling scheme to distribute the channel access permission among contending stations wishing to transmit delay sensitive data. However the standard does not define the method of management the polling list and detail implementation of the PCF function. In paper [2] [3] [4], the round robin scheme was used as the scheduling algorithm to manage the polling list. These papers were mainly focused on evaluating the performance of the contention free access control mechanism of the 802.11 WLAN in an integrated Voice/Data environment. But the fact had been proved that their performance is not good.

Since all stations operations operating during the CFP are packetized voice users, they all have the same QoS requirements, and therefore priority polling mechanisms are not proposed.

2.2 Methodology delineation

2.2.1 Remote Deficit Round-robin scheme

A simple algorithm without any timestamp computation called Deficit Round Robin (DRR) [9] was proposed by Shreedhar et al. This scheme is a simple modification of the round robin service

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discipline. In the DRR scheme, each queue I waiting for service has a state variable called *deficit counter (DCi)*. If the length of the next packet in the queue I is less than the value of *DCi* then queue I is allowed to send out its packet. It is clear that the scheduler has a priori knowledge of the next packet in the queue as the queues are localized to the scheduler. After the transmission if there are no more packets in queue *i*, the corresponding state variable *DCi* is reset to zero. If there are more packets waiting in queue I, *DCi* decremented by Li, the length of the packet transmitted. After that the scheduler continues to check whether *DCi* is still greater than the length of the next packet, if so, the above procedure is repeated. If the *DCi* value is not sufficient to transmit the packet, then the scheduler moves to the next station in the list stopping service to queue i. At the start of subsequent rounds, *DCi* is incremented by a specific service share (quantum). The quantum values are used to specify the limit that the queues will be serviced in each visit. Thus DC keeps track of deficits in each round.

The downlink deficit counter is valid only for the downlink queues set up at the PC. In a downlink queuing system the scheduler has access to the complete state of the queues. However in wireless networks the scheduler does not have direct access to the state of these remote queues. Therefore none of the above FQ scheme can be directly used to schedule the uplink traffic in wireless network.

Usually a network carries traffic in both uplink and downlink decider. Therefore the scheduler operating at the centralized coordinator (PC) must schedule both uplink and downlink. It is possible for an active downlink flow to exist while the corresponding uplink flow is inactive. Therefore the scheduler will have to schedule uplink and downlink flows wither simultaneously or independently.

This part like a flow control mechanism for different source's bandwidth requirement. For example, the video source requires more bandwidth than voice and data. To satisfy the QoS of multimedia, we must consider the characteristics of multimedia traffic. The PCF have the characteristic of supporting real-time, but lack of flow control mechanism.

2.2.2 Polling List management

If the PC supports use of the CFP for inbound frame transfer as well as for frame delivery, The PC shall maintain a "polling list", for use in selecting STAs that are eligible to receive CF-Polls during CFPs. The form of contention-free support provided by the PC is identified in the Capability Information field of Beacon, Association Response, Reassociation Response, and Probe Response management frames, which are sent from APs.

The polling list is used to force the polling of CF-Pollable STAs, whether or not the PC has pending traffic to transmit to those STAs. A minium set of polling list maintenance techniques are required to ensure interoperability of arbitrary CF-Pollable STAs in BSSs controlled by arbitrary access points with active PCs. APs may also implement additional polling list maintenance techniques that are outside the scope of this standard [1].

We propose a polling list management for voice flow to allocate unused bandwidth to currently active voice users. The polling scheme will mark the voice stations that are not active either transmitting or receiving voice frames on the OFF state to the un-active list. Thus we can save the time to poll the voice station when they can no data to transmit during the silence (OFF) state, thereby giving more bandwidth to data station. In the other word, we can decrease the overhead of polling scheme to enhance the efficiency of CFP.

III. SIMULATION MODEL

This model represents an infrastructure network which characterizes a signal BSS with a AP. The infrastructure network operates with asynchronous data users in the CP and packetized voice and video terminals operating in the CFP.

Several assumptions have been made to reduce the complexity of simulation models.

3.1 Traffic Models

Voice traffic

The voice source is modeled using an ON/OFF process. Here the voice source is modeled as a two Markov chain with a “talk state” and a “silent sate”. When the source is in talking state it periodically generates fixed size voice packets. We selected the CF repetition period identical to the inter-arrival time of these periodic voice packets.

The Inter-national Telecommunications Union (ITU) has developed numerous speech coding standards including the G.726 adaptive differential pulse code modulation (ADPCM) standard which uses a 32kbps rate and the G.711 PCM standard uses a 64kbps rate.

Video traffic

A video traffic source generates frames at a constant rate over its active period. The length of the video frame can be very large compared to the maximum length of the Mac Protocol Data Unit (MPDU) defined in the standard. These packets are segmented into constant size packets and sent to the MAC layer as a packet burst.

Data traffic

The presence of asynchronous traffic in an 802.11 based network results in transmission during the contention based period. The transmission during the contention period may delay the start time of CF cycle. Since this is common in practical situation, we use asynchronous data terminals to generated asynchronous traffic. The data traffic is generated using multiple stations. Both packet length distribution and data frame inter arrival time distribution form a exponential distribution.

3.2 System parameters

PHY medium capacity	10Mbps
CFP repetition interval	20ms
CFP_MaxDuration	15ms
Beacon interval	3ms
MaxMPDU	743bytes
Header length	32bytes
Preamble length	24bytes
SIFS	30μ sec
PIFS	60μ sec
DIFS	90μ sec

Figure 3.1 Table of system simulation parameters

3.3 Boundary conditions

During the Contention Free period of the Super frame the stations that are part of a voice conversation and video conference are polled according to the polling list. The downlink we use deficit round-robin algorithm, and the uplink, we use remote deficit round-robin algorithm to implement the scheduling algorithm.

Rc: channel capacity
H: header length
PA: Preamble length
$Payload_{Voice}$: voice frame length
$Payload_{Video}$: video frame length = *Max MPDU*
DC_{voice} : voice flow's DC value
DC_{video} : video flow's DC value
m: number of video flow served in one *CFP*
n: number of voice flow served in one *CFP*
CFP_MAX: Maximum Contention Free Period

$$T_{Voice} = (PA + H + Payload_{Voice})/Rc$$

$$T_{Voice'} = T_{CF+Down} + SIFS + T_{CF+Up} + SIFS = 2[(PA + H + Payload_{Voice})/Rc + SIFS]$$

$$T_{video} = (PA + H + Payload_{Video})/Rc$$

$$T_{video}' = T_{CF+Down} + SIFS + T_{CF+Up} + SIFS = 2[(PA + H + Payload_{video})/Rc + SIFS$$

Total time for a video flow in one CFP

$$= (DC_{video}/Payload_{Video}) * T_{video'}$$

Total time for a voice flow in one CFP $= T_{Voice'}$

$$m * [(DC_{video}/Payload_{Video}) * T_{video'}] + n * T_{Voice'} <= CFP\_MAX \quad (1)$$

Figure 3.2 Boundary condition

From equations (1) we can calculate out the boundary video number *m* and boundary voice number *n*. At the first round simulation, we select the boundary number pair *(m,n)* as (4,16). Using these parameters we can see the simulation result about the frame mean delay and throughput, whether or not they beyond the maximum tolerate delay. If no, we will tune the number pair *(m,n)* until the mean delay beyond the maximum tolerate delay. Under the condition, number pair *(m,n)* have achieve the utmost. If we increase any number of number pair *(m,n),* the mean delay will beyond the maximum tolerate delay, in the other word, they will not satisfy the QoS of video and voice traffic mix.

PA represents the Preamble length in bits, H the overhead per frame in bits, Payload the Payload size of the data frame in bits, and *Rc* the transmission speed in bits per second (bps).T_{voice} and T_{video} are not under all circumstances the time required for a station to be polled and transmit its frame, but an upper bound of this time.

3.4 Result

The performance of our new scheme is better than original round-robin scheme and. In this simulation, we use the result calculated by boundary condition, 8 pair voice conversation and 4 video sources. Our new scheme has more data throughput, this mean that our new scheme has better efficiency. *Figure 3.3* reveal that our new scheme has lower video mean frame delay.

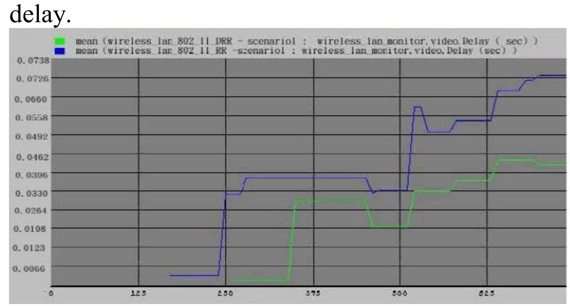

Figure 3.3 Video frame mean delay comparison

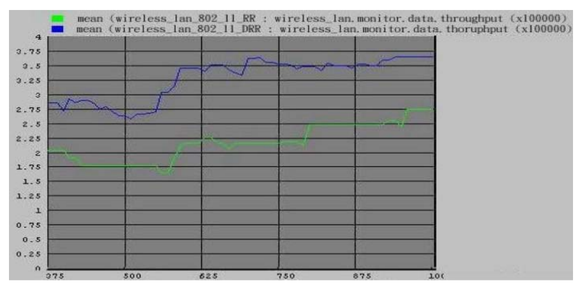

Figure 3.4 Data throughput comparison

The *Figure 3.5* show the detail information of the mean video frame delay and the improvement percent between our new scheme and the original round-robin.

	Voice = 0	Voice = 8	Voice = 16	Voice = 24	Voice = 32	Voice = 40
DRR Mean video frame delay (ms)	25.003	28.093	32.982	37.223	44.231	66.138
RR Mean video frame delay (ms)	26.223	37.277	64.384	82.489	102.122	130.251
Mean video frame delay Improvement	4 %	32.142 %	100 %	121.62%	127.27%	96.969%

Figure 3.5 The table of the mean video frame delay

IV. CONCLUSION AND FUTURE WORK

We design an efficient centralized Scheduled Access MAC approach in IEEE 802.11 to satisfy multimedia traffic mix. By simulation, we prove that our method has successfully decreased the mean delay of voice and video and increase the data throughput. By run different combination of video sources, we calculate the average of the voice and video mean frame delay and data throughput. According the upper data, we can find the most video sources it can be accommodated. It shows that, our new scheme can support more multimedia traffic mix, because it's better efficiency and have good characteristics for multimedia traffic mix.

At present, we have achieved the most basic and important goal, but there are some where still can be improved, and this are also the research stuff in the future. In our simulation, the video traffic model is constant bit rate. To conform the real world, we can use variable bit rate video traffic model, so we must design an efficient flow control for variable bit rate for video traffic.

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