IEEE 802.11e 無線網路中 VoIP 傳輸品質保證與允入問題

Pei-Yeh Wu and Yu-Chee Tseng
Department of Computer Science
National Chiao-Tung University, Hsin-Chu, 300, Taiwan
E-mail: phwu, yctseng@csie.nctu.edu.tw

Hewitt Lee
ZyXEL Communications Corp.
No. 6, Innovation Road II, Science Park, Hsinchu, 300, Taiwan
E-mail: hewitt.lee@zyxel.com.tw

Abstract
Supporting telephone services using wireless LAN as the access network is an emerging service. The SIP and IEEE 802.11e are perhaps the two most promising protocols to support such services. In this paper, we show how to integrate SIP and 802.11e to conduct call admission control and resource reservation to support VoIP's QoS in IEEE 802.11e WLANs. Besides, we also suggest some adjustments and MAC enhancements to 802.11e to facilitate VoIP traffics over WLANs.

摘要
在無線網路環境下來支援網路電話的運行是近年來崛起的新服務。要來支援這樣的服務，SIP 和 IEEE802.11e 是兩個很重要且有遠景的協定。在這個論文裡面，我們提出了一個整合SIP 和802.11e 的架構來做網路資源管理，以達到網路電話在無線網路下的品質要求。此外，我們也提出了一些在 IEEE802.11e 網路擷取層上的改善方法，期許可以讓無線網路環境下的網路電話服務可以運行的更加順暢 。

Keywords: Call admission control, IEEE 802.11e, quality of service (QoS), voice over IP (VoIP), wireless network

關鍵字： 網路電話服務、傳輸品質、無線網路、IEEE802.11e

1 Introduction
Recently, we have seen two major trends in the area of communications. First, IEEE 802.11 WLANs have been widely deployed in the world. Second, due to the growth of Internet bandwidth, real-time audio and video applications have become more mature and popular. The combined efect has made VoIP (voice over IP) over WLANs possible. For example, to support VoIP, new products have appeared, such as Wi-Fi phones and dual-mode cellular-WiFi phones (e.g., Cisco Wireless IP Phone 7920 [1] and Motorola MPx [2]).

The above observation has raised an interesting issue: how do Weans support QoS (Quality of Service) and CAC (Call Admission Control) for VoIP traffics. The *IEEE 802.11 Task Group E (802.11e)* [3] has been formed to expand the current 802.11 MAC protocol to support applications with QoS requirements. In addition, the *Session Initiation Protocol (SIP)* [4, 5] has been widely accepted as the signaling protocol for VoIP to handle the setup, modification, and teardown of VoIP sessions.

In this work, we consider the cross-layer protocol design problem to facilitate VoIP traffics over IEEE 802.11e WLANs. We show how 802.11e can cooperate with VoIP SIP signaling to conduct QoS and CAC over the wireless channel. We also propose enhancements to 802.11e MAC part to improve its performance in delivering VoIP traffics.

Several prior works have focused on improving VoIP traffics in WLAN environments. Reference [6] claims that admission control is critical to protect VoIP traffics because resource in a WLAN cell is limited. A *CUE (Channel Utilization Estimation)* is proposed to determine whether to accept a new call. Alternatively, giving priority to VoIP traffics helps improving performance too [7]. References [8, 9] point out that the bottleneck is at the access point (i.e., down link traffic). Hence, a *BC-PQ (Backoff Control and Priority Queue)* mechanism [8] is proposed to give priority to voice traffics over data traffics and assign zero backoff time to voice packets in access points. On the other hand, it is proposed to separate real-time and non-real-time packets into two queues in [9], and an AP always processes the real-time queue first whenever it is not empty. The number of concurrent VoIP sessions that can be supported in a WLAN is evaluated in [10, 11]. It is reported that besides the bandwidth limitation of the physical layer, the codec, packetizaion interval, and delay budget may all influence the number of VoIP sessions that can be supported. It is further pointed out that the selection of packetization interval has more impact than the selection of codec.

On the standardization track, the IEEE 802.11 working group R (802.11r) is currently developing fast roaming mechanisms. Reference [12] proposes a structure to integrate Mobile IP with SIP to assist

802.1D Priority	802.1D Designation	802.11e AC Index	802.11e AC Designation	Comment
1 (low)	BK	01	AC_BK	Background
2	-		AC_BK	Background
0	BE	00	AC_BE	Best Effort
3	EE		AC_BE	Best Effort
4	CL	10	AC_VI	Video
5	VI		AC_VI	Video
6	VO	11	AC_VO	Voice
7 (high)	NC		AC_VO	Voice

Figure 1: The mappings of 802.1D priorities to IEEE 802.11e ACs.

VoIP mobility, while [13] suggests using ad hoc-assisted handoff to meet the Qos requirement of VoIP during handover. The IEEE 802.11e aims at enhancing its MAC mechanism to support QoS. Several works [14, 15, 16, 17] have studied using 802.11e to improve multimedia transmission under WLAN. QoS schedulers based on HCCA of IEEE 802.11e are proposed in [14, 15]. Some works discuss how to ameliorate EDCA in IEEE 802.11e to facilitate multimedia transmission. Reference [16] extends the basic EDCA by using an adaptive fast backoff mechanism along with a window doubling mechanism at busy period. In [17], it is suggested that in EDCA each mobile station must conduct admission control on real-time traffic streams to protect existing real-time sessions, and that AP must adjust lower-priority Access Categories' contention windows, Arbitration Inter-frame Spacing (AIFS), and so forth, to protect real-time sessions from being collided by those that do not require admission control.

In this paper, we show how to support VoIP services over WLAN environments. Existing works are not designed for this purpose. In particular, we show how to integrate IEEE 802.11e with SIP to conduct call admission control over WLAN environments and support QoS for VoIP calls. We believe that cross-layer design is essential for maintaining the QoS of VoIP services. Moreover, we also show how to increase the number of VoIP sessions being supported under an AP with compromising QoS and how to improve the MAC mechanism of IEEE 802.11e to facilitate the transmission of VoIP traffic.

The rest of this paper is organized as follows. Some preliminaries are given in Section 2. The proposed QoS architecture for VoIP service is introduced in Section 3. Section 4 presents several MAC enhancements to IEEE 802.11e for VoIP services and Section 5 gives some simulation results. Finally, Section 6 draws our conclusions.

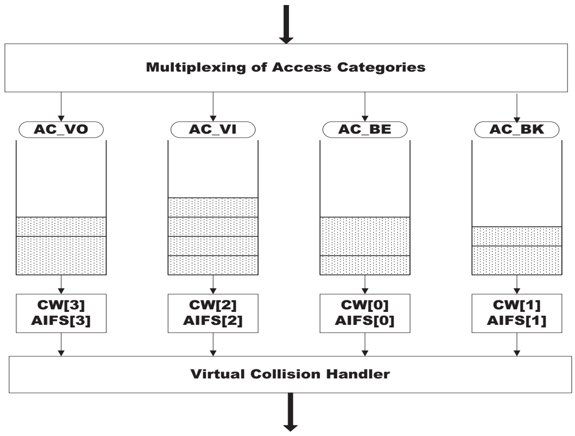


Figure 2: Management of access categories in EDCA.

2 Preliminaries

2.1 802.11e MAC Protocol

The current IEEE 802.11 MAC has no means of differentiating *TSs (traffic streams)* or sources. All packets are treated equally in both DCF and PCF. As a result, no consideration can be made for the service requirements of traffics. The IEEE 802.11 Working Group E has proposed a *HCF (Hybrid Coordination Function)* for both ad-hoc and infrastructure modes. Several enhancements are introduced in 802.11e. First, a concept called *TXOP (Transmission Opportunity)* is introduced, which is a period of time during which a QSTA (a station that supports 802.11e) can exclusively use the wireless medium. A TXOP is defined by a starting time and a maximum duration and it can be obtained by contention or by assignment from the HC (Hybrid Coordinator). Second, IEEE 802.11e supports traffic differentiation by giving traffic streams priorities. Third, it allows a TS to specify its traffic characteristic.

HCF supports two access methods, a contention-based mechanism called *Enhanced Distributed Channel Access (EDCA)* and a contention-free mechanism called *HCF Controlled Channel Access (HCCA)*. Since HCCA is enhanced from PCF and PCF is seldom implemented, we only discuss EDCA in the following.

EDCA of IEEE 802.11e

To differentiate services, IEEE 802.11e adopts the eight user priorities in 802.1D and maps then to four Access Categories (ACs) (refer to Fig. 1). EDCA supports these ACs by four separated queues in both QAP (an AP that supports 802.11e) and QSTA, as illustrated in Fig. 2. Each queue operates as an independent entity and conducts backoff as in the original IEEE 802.11 DCF. The *i*th AC, i=0..3, has its own arbitration inter-frame space (*AIFS*[*i*]), initial window size (*CWmin*[*i*]), and maximum contention window size (*CWmax*[*i*]). If multiple queues finish their backoff simultaneously, the *virtual collision handler* will choose the AC with the highest priority to send and the lower priority AC(s) will backoff as

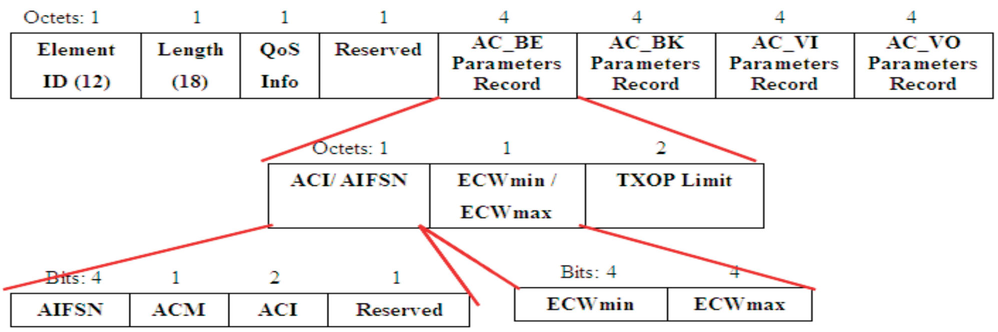


Figure 3: Structure of the IEEE 802.11e EDCA_Parameter_Set information element.

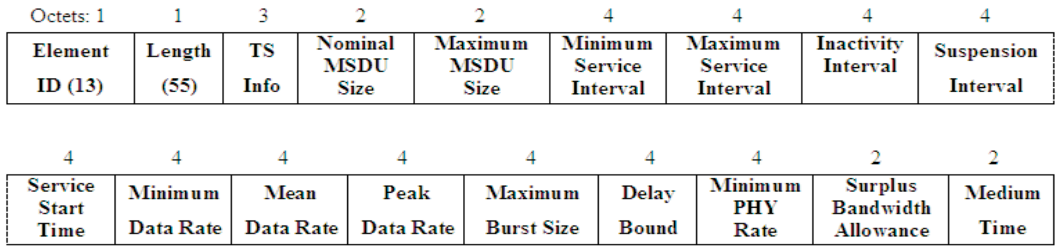


Figure 4: TSPEC information element of IEEE 802.11e.

experiencing an external collision.

The EDCA Parameter Set information element (Fig. 3) can be sent in beacon frames. It also contains the TXOP limit of each AC, which bounds the amount of burst transmission of a QSTA after it successfully contends the medium. If TXOP limit equals zero, a QSTA can transmit only one packet each time it gains the TXOP.

Admission Control in EDCA

A QAP uses the *ACM (admission control mandatory)* subfield advertised in EDCA Parameter Set to indicate whether admission control is required for each AC. A QSTA can send an *ADDTS (add traffic stream)* request to the QAP to request adding a new traffic stream by specifying its direction (uplink, downlink, bidirectional, or direct) and providing a *TSPEC (traffic specication)* information element as shown in Fig. 4. Some important fields in TSPEC are discussed below:

- Minimum Data Rate: the lowest data rate (in bits per second) to transport MSDUs.
- Mean Data Rate: the average data rate (in bits per second) to transport MSDUs.
- Peak Data Rate: the maximum allowable data rate (in bits per second) to transport MSDUs.
- Minimum PHY Rate: the desired minimum physical rate for this traffic stream.
- Medium Time: the amount of time admitted to a stream to access the medium. This field is not used in the ADDTS request frame, but will be set by the HC in the ADDTS response frame.

On receiving an ADDTS request, the QAP may decide to accept or reject it. In the former case, the QAP will calculate a *MT (Medium Time)* for this traffic stream per beacon interval and reply an ADDTS response containing this information; otherwise, an ADDTS response including rejection information is replied. For QAP and each QSTA, they keep the total MT and consumed MT of each AC. Only when the former is lager than the latter, can packets in the corresponding AC be transmitted. After each beacon interval, the consumed MT will be reset to zero. The QAP can identify a traffic stream by its TSID and direction. This information is available in the TS info field in TSPEC. In this paper, we will use bidirectional reservation for VoIP sessions.

2.2 SIP and SDP

SIP is a signaling protocol, which is considered as an attractive alternative to H.323 to support VoIP. SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions. It often cooperates with other protocols, such as *SDP (Session Description Protocol)*[1] to describe session characteristics and *RTP (Real-time Transport Protocol)*[2] to send traffic after call setup.

SIP is designed to keep signaling as simple as possible. Fig. 5 shows one example of call establishment. When a caller wants to make a VoIP connection with a callee, it sends an INVITE including the codecs that the caller supports in a SDP message body. Fig. 6(a) is an example, with G.726 (2), G.723 (4), and G.728 (15) as the selections (numbers in parentheses are payload types) and 123 the receive port. If the callee decides to accept the request, it replies a Ringing and an OK signals to the caller. The OK signal will contain the callee's choice of codec. In Fig. 6(b), the callee chooses 728 (15),

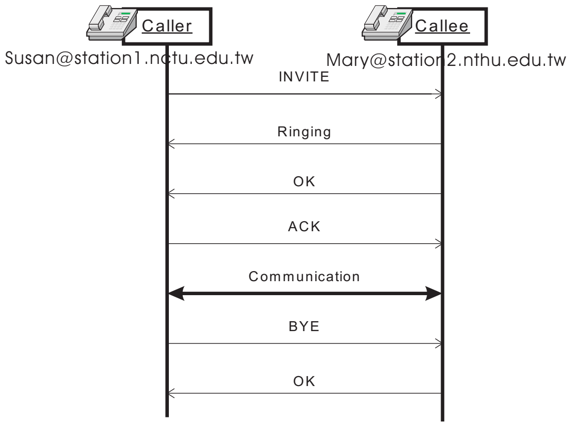


Figure 5: An example of SIP call setup and tear-down.

using the receive port of 888. A port number of 0 indicates a rejection.

1. SDP is specified in RFC 2327. It does not provide a means for transporting or advertising. RFC 3264 describes how SDP co-works with SIP.
2. RTP is often accompanied with *RTCP (RTP Control Protocol)* to provide transport services to support real-time applications.

3 The Proposed QoS Architecture for VoIP Services

We consider an IEEE 802.11e wireless network operating in the infrastructure mode to support VoIP pplications. We adopt SIP for call setup and management. We also assume that a VoIP session can dynamically adjust its *packetization interval (PI*) even during communication, where PI represents how frequently voice data should be encapsulated into packets. Our purpose is to guarantee high QoS for admitted VoIP sessions when the network load is not heavy, and to support as many VoIP onnections with acceptable QoS as possible when the network load is heavy. RFC 3312 (Integration of Resource Management and SIP) discusses how QoS can be made a precondition for sessions initiated by SIP. These preconditions require that participants reserve network resources before continuing. Inspired by this, we propose an architecture for IEEE 802.11e to incorporate with SIP to conduct resource reservation and admission control.

3.1 Call Establishment

Fig. 7 shows the proposed QoS architecture after integrating SIP with IEEE 802.11e. When a caller under QAP1 wants to establish a VoIP connection with a callee at QAP2, it can send an INVITE signal with a SDP message containing necessary codec information to the callee. QAP1 and QAP2, on receiving this INVITE signal (refer to boxes A and B in Fig. 7), will do pre-resource reservation and possibly filter out some codecs that they cannot support due to bandwidth constraints. When the callee receives this INVITE signal (refer to boxes C and D), it will exchange 802.11e ADDTS request and response with QAP2. These steps can prevent ghost rings[3]. After exchanging ADDTS messages, the callee can send Ringing and OK signals to the caller. The OK signal will contain the codec being selected by the callee. After receiving the OK signal, the caller will exchange ADDTS request and response with QAP1 (refer to boxes E and F). If these steps successfully go through, an ACK signal will be replied to the callee. In the following, we will explain the detail actions to be taken in boxes A, B, C, D, E, and F.

3A ghost happens when a user can not communicate with the other side as he/she picks up a ringing phone. The shortage of bandwidth is often a reason for ghost rings in VoIP applications.

A. Pre-resource Reservation at the Caller

A QAP has to broadcast the PHY rates that it can support in its beacon frames. When a QSTA is associated with a QAP, it also registers with the QAP its supported rates. In IEEE 802.11e, a QSTA can specify its minimum PHY rate when adding a new traffic stream. When the QSTA can transmit/receive at this rate, the requested QoS should be guaranteed; otherwise, the requested QoS is not necessarily guaranteed. To conduct pre-resource reservation, we propose that each QAP keeps a *Packet Size Table (PST)* as in Fig. 8, which contains the packet sizes when different codecs and packetization intervals (PI) are used. For example, in G.726 with a sampling rate of 32 kbps, if a packetization interval of 20 ms is used, then each packet is of size 154 bytes (which contains 80 bytes of voice payload, 40 bytes of IPv4/UDP/RTP/error-checking overhead, and 34 bytes of MAC/error-checking overhead). The payload sizes generated by different codecs can be inferred from [5]. Note that the calculation does not include the PLCP preamble and header, which are 24 bytes and must be sent at the lowest rate of 1 Mbps. Therefore, given a codec and its packetization information, QAP1 can compute a *medium time (MT)* that should be reserved for the traffic stream per beacon interval (BI):

MT = (total time needed per BI)
= (time to send one packet) * (no. packets per BI) * (surplus bandwidth allowance)
= [(PLCP preamble and header) + payload + *SIFS* + ACK] * (BI/PI) * (surplus bandwidth allowance)
= [(packet size) / (min PHY rate) + 2 * PLCP/Mbps + (ACK/min PHY rate) + *SIFS*] *(*BI/PI*) * (surplus bandwidth allowance) (1)

According to 802.11, SIFS is 10 μs, ACK packet is 14 bytes, and PLCP preamble and header are 24 bytes. The surplus bandwidth allowance is a value slightly larger than 1 to take into account the excess time for possible contentions and retransmissions (in statistical sense). In this work, we assume its value to

```
INVITE sip:Mary@station2.nthu.edu.tw SIP/2.0
From: Caller<sip:Susan@station1.nctu.edu.tw>; tag=abc123
To: Callee<sip:Mary@station2.nthu.edu.tw>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Disposition: session

v=0
o=Susan 123 001 IN IP4 station1.nctu.edu.tw
s=
c=IN IP4 staion1.nctu.edu.tw
t=0 0
m=audio 123 RTP/AVP 2 4 15
a=rtpmap 2 G726-32/8000
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
```

(a)

```
SIP/2.0 200 OK
From: Caller<sip:Susan@station1.nctu.edu.tw>; tag=abc123

To: Callee<sip: Mary@station2.nthu.edu.tw>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collee 456 001 IN IP4 station2.nthu.edu.tw
s=
c=IN IP4 station2.nthu.edu.tw
t=0 0
m=audio 0 RTP/AVP 2
m=audio 0 RTP/AVP 4
m=audio 888 RTP/AVP 15
```

(b)

Figure 6: An example of SIP with SDP message bodies: (a)INVITE siganl and (b)OK signal.

be 1.1. For example, when BI is 1 sec and min PHY rate is 11 Mbps, if we use G.726 with 32 kbps and PI of 20 msec, then MT = [154/11(*bytes/Mbps*) + 2 * 24(*bytes/Mbps*) + 14/11(*bytes/Mbps*) + 10μs] * (1000/20) * 1.1 = 28.39 ms.

For each codec in the INVITE signal, if its MT exceeds the remaining MT of QAP1, we will remove the codec from the codec list. In case that the remaining resource in QAP1 does not allow it to support any codec, QAP1 can drop the INVITE silently or reply a SIP response to the caller with a status code of 480, which means "temporarily not available". Also note that since voice communications are bi-directional, the AP should reserve 2 **MTmax*, where *MTmax* is the maximum time required by all codecs in the list.

B. Pre-resource Reservation at the Callee

The calculation of medium time at the callee when receiving the INVITE signal is similar to the above discussion. QAP2 will also filter out those codecs that it cannot support from the INVITE siganl and reserve the maximum required bandwidth. The INVITE signal will then be forwarded to the callee if at least one codec can be supported.

C. ADDTS Request by the Callee

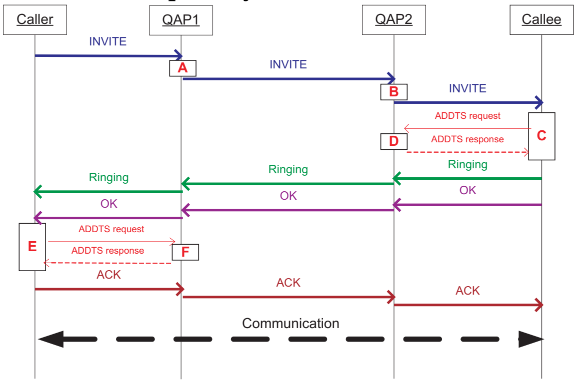


Figure 7: The proposed QoS architecture for SIP call establishment in 802.11e networks.

Codec	Data Rate (kbps)	Packitization Interval (ms)				
		5	10	20	30	40
G.711	64	114	154	234	314	394
G.726	16	84	94	114	134	154
	32	94	114	154	194	234
G.728	16	84	94	114	134	154
G.723.1	5.3				94	
	6.3				98	

Figure 8: The Packet Size Table, which gives the packet sizes (in bytes) when different codecs and packetization intervals are used.

After deciding the codec, the callee can send a bidirectional ADDTS request to QAP2 by including a TSPEC element. We suggest to convey VoIP service requirements with the following fields in TSPEC:

- Minimum Data Rate = the acceptable longest packetization interval of the corresponding codec.
- Mean Data Rate = the packetization interval selected by the callee.
- Maximum Data Rate = the acceptable shortest packetization interval.
- Medium Time = the codec selected by the callee.

With these information, QAP2 can do CAC as described in the following part D.

D. Call Admission Control in QAP2

According to the callee's ADDTS request and the Packet Size Table, QAP2 can compute the required medium time following the equation in step A. Note that with a bidirectional request, the same medium time should be applied to both uplink and downlink directions. In order to conduct call admission control, each QAP should keep the following variables:

- TXOPBudget[*ACi*]: The remaining bandwidth that can be allocated by *ACi*,i=0..3.
- TxAdDn[*ACi*][TSID]: The admitted

medium time for stream TSID of *ACi* in the downlink irection.

- TxAdUp[*ACi*][TSID]: The admitted medium time for stream TSID of *ACi* in the uplink irection.
- TxAdDn[*ACi*]: This value is set to ∑*TSID*TxAdDn[*ACi*][TSID], to record the overall resource allocated to *ACi* in the downlink direction.
- TxUsedDn[*ACi*]: The summation of used medium time of all downlink streams of *ACi*.

Initially, TXOPBudget[*ACi*] contains all the bandwidth (in terms of medium time) that is reserved for *ACi*. Whenever a new stream is added, the corresponding resource is subtract from TXOPBudget[*ACi*], and the resource is assigned to TxAdDn[*ACi*][TSID] and/or TxAdUp[*ACi*][TSID]. Also, each QSTA should keep the following variables:

- TxAdUp[*ACi*][TSID]: The admitted medium time for stream TSID of *ACi* in the uplink direction in this STA per BI.
- TxAdUp[*ACi*]: This value is set to ∑*TSID*TxAdUp[*ACi*][TSID], to record the overall resource allocated to *ACi* of this STA in the uplink direction.
- TxUsedUp[*ACi*]: The summation of used medium time of all uplink streams of *ACi*.

Resource reservation at QAP2 is done as follows. First, we compute the value of TXOPBudget[*ACi*]-2*MT. If the value is non-negative, there is sufficient resource to support this call and we can set

TXOPBudget[*ACi*] = TXOPBudget[*ACi*] – 2*MT;
TxAdDn[*ACi*][TSID] = MT;
TxAdUp[*ACi*][TSID] = MT;
TxAdDn[*ACi*] = TxAdDn[*ACi*] + TxAdDn[*ACi*][TSID] ;

Where MT is computed form Eq. (1) based on the information of codec, PI, min PHY rate, etc., provided by the TSPEC. If there is no sufficient resource, the QAP can choose the next larger PI (if possible), recompute a new MT, and repeat the above testing, until a satisfactory PI is found. Then QAP2 will reply an ADDTS response to the callee with the Mean Data Rate = PI and Medium Time = MT in the TSPEC. If there is no sufficient resource, then an ADDTS response is replied with Medium Time = 0.

At the callee's side, if an ADDTS response with a positive Medium Time is received, then the QSTA sets its TxAdUp[*ACi*][TSID]=Medium Time and retrieves the PI in the Mean Data Rate field and passes it to the upper layer VoIP application program. Otherwise, the call is considered rejected. In both cases, the callee should reply a response signal with the proper status code to the caller.

F. Call Admission Control in QAP1

The action is similar to step D. If the caller receives a successful ADDTS response, it will send an ACK signal to the callee. Then, the voice communication can be started.

Because of the pre-resource reservation in steps A and B, a lot of potential ghost rings can be avoided. Also, voice quality can be guaranteed because of the CAC in steps D and F. Finally, we remark that although we assume that both the caller and the callee are under WLANs, the above procedure should work well if any side is not under a WLAN.

3.2 Resource Readjustment During Transmission

The above steps are for the setup of new calls. However, during transmissions, a stream may dynamically change its bandwidth requirement. In this subsection, we will introduce the steps to be taken alleviate such problems.

1. Estimation of Downlink PI by QAPs

We note that the PI selected by a codec is not conveyed via SIP signals to the codec at the other side. Therefore, although the resource reservation mentioned above in the uplink direction (from QSTA to QAP) is accurate, the MT reserved for the downlink direction is only an approximation. To solve this problem for each stream TSID, we require a QAP to observe packets from the other side for several beacon intervals and estimate the actual PI being used. After estimating the actual PI, the QAP should calculate the MT according to Eq. (1) for this stream and then update TxAdDn[AC VO][TSID] and TxAdDn[AC VO].

2. Adjustment for PHY Rate Change at QSTAs

When a traffic stream finds that its admitted medium time is not enough to send all of its packets because its physical rate drops below its specified min PHY rate, we suggest that the QSTA can send an update ADDTS request to its QAP with the min PHY rate field equal to its current PHY rate or below. The operations are similar to the above steps C and D. The QAP may respond in two ways: to allocate more medium time for the stream if it still has more resource available, or to suggest a longer PI to reduce the required medium time of the corresponding traffic stream. If the request succeeds, a new medium time will be replied; otherwise, the QAP will reply with the stream's original medium time. In the latter case, the call may suffer from lower quality.

3. Mechanisms to Support More VoIP Sessions

When a WLAN is very congested or when there are more new VoIP calls intending to join the WLAN, we may ask current calls to reduce their resource consumption. On finding such a situation, a QAP can send a beacon frame by carrying such a notification to its QSTAs. A QSTA may respond in two ways:

- The QSTA may change the PI of one of its streams by notifying the corresponding codec as well as sending a new ADDTS request to the QAP with a longer PI. The QAP should grant the ADDTS request.
- The QSTA may decide to ask one of its streams to change to a lighter-load codec. This can be achieved by the RE-INVITE or UPDATE signal of SIP.

4 MAC Enhancements for VoIP Traffics

In this section, we discuss several enhancements to improve the performance of 802.11 medium access to support VoIP traffics.

- Bookkeeping: Some control mechanisms are needed to achieve the CAC mentioned above. Only admitted VoIP sessions can drop packets to the AC VO queue. Besides, for each *ACi* that requires admission control, we must keep the medium time that it can use in TxAdUp[*ACi*] and TxAdDn[*ACi*] and the amounts of time that have been used in TxUsedUp[*ACi*] and TxUsedDn[*ACi*] in current beacon interval. This bookkeeping work must be done for every data frame being transmitted. Only when TxUsedDn[*ACi*] < TxAdDn[*ACi*] (resp., QAPs or TxUsedUp[*ACi*] < TxAdUp[*ACi*]) can the corresponding access category *ACi* contend for the medium in the QAP (resp., QSTA). Also, the value of TxUsed[*ACi*] in each station should be reset to zero at the end of each BI.
- Redirecting: To avoid the sudden congestion of the network, whenever a packet is generated by an admitted TS, the system can estimate whether the remaining medium time of the AC_VO queue (i.e., TxAdDn[AC_VO] - TxUsedDn[AC_VO] - (the amount of data buffered in the AC_VO queue) for QAP, or TxAdUp[AC_VO] - TxUsedUp[AC_VO] - (the amount of data buffered in the AC_VO queue) for QSTA) is enough to send this packet or not. If so, this frame will be dropped to the AC_VO queue as normal; otherwise, this packet will be placed to any AC that doesn't need admission control. Intuitively, this is to transfer the burst arrival of VoIP packets that can not be delivered in the current beacon interval to other best-effort queues, hoping to deliver them by their due dates. This mechanism may help improve the performance of VoIP traffics when the physical rate decreases or when there is a sudden increase of the collision probability in the network.
- Adjusting Access Parameters: When a station finds that its dropping rate is higher than a threshold, it can check the receive signal strength of its current QAP. If the signal quality is poor, it may consider switching to a new QAP of better signal quality. If the signal is good, then the cause might be an unexpected high contention from other *ACis*. In this case, the STA may ask the QAP to increase the CW and AIFS of other access categories. Afterward, when the network is not so highly congested, the QAP may decide to ask other STAs to return to their original CW and AIFS. This is similar to what is suggested in [17].
- Favoring Downlink: We observe that in many cases the performance bottleneck of the network is at the QAP. This is because the QAP is in charge of delivering packets for multiple streams. So, a higher priority should be given to QAP. In our design, we will facilitate its transmission as follows: whenever the QAP receives VoIP packets from a station *i*, it is allowed to immediately allocate a TXOP to transmit packets of station *i* in the AC VO queue after a SIFS. In this way, the QAP will have more change to transmit than QSTAs.

5 Simulation Results

An event-driven simulator is developed to evaluate the performance of the proposed CAC and MAC enhancements. Unless otherwise stated, the following assumptions are made in our simulation. The network contains one QAP and multiple QSTAs. We set TXOP limit to zero for four ACs, which means that a QSTA can only transmit one packet in each successful contention. The communication channel is assumed to be error-free. No RTS/CTS is used. The beacon interval (BI) is set to 500 ms. There are two static QSTAs, each of which will generate background traffic (AC_BK) by a poisson process of rate 10 kbps. For other QSTAs, each has a VoIP session (AC_VO) using G.726 as the codec with PI equal to 20ms. For AC_VO, we set CWmin to 7, CWmax to 15, and IFSN to 2. For AC_BK, we set CWmin to 31, CWmax to 1023, and IFSN to 7.

A. Influence of Admission Control

In this scenario, we want to verify the importance of admission control. Voice calls already admitted to the system can only contend to transmit when TxUsedDn[AC_VO] (resp., TxUsedUp[AC_VO]) is smaller than TxAdDn[AC_VO] (resp., TxAdUp[AC_VO]). Initially, there are two QSTAs which will generate background traffic. We then add one QSTA with one voice call to the network every two seconds. QSTAs are assumed to be static and can transmit at a rate of 11 Mbps. The queue size is set to 50 for each AC.

With CAC, our scheme can accept up to 17 VoIP calls. The rest of the calls will be rejected. Without CAC, calls are all accepted to the system. Fig. 9 shows the goodputs of VoIP traffics with and

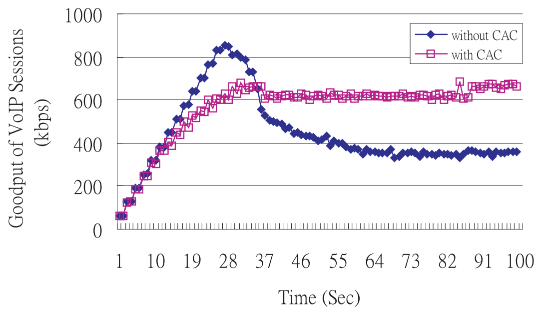


Figure 9: Impact of CAC: the total goodputs of VoIP calls with and without CAC.

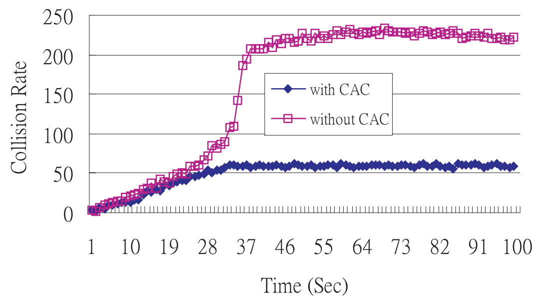


Figure 10: Impact of CAC: the collision rates per QSTA with and without CAC.

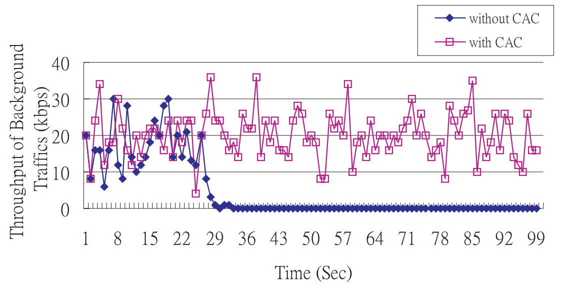


Figure 11: Impact of CAC: the throughputs of background traffics with and without CAC.

without CAC. As can be seen, without CAC the collision rate will increase rapidly, and the goodput will drop sharply after there are more than 14 calls. With CAC, the goodput can be maintained at a stable level after there are 17 calls. Before the 34th second, the goodput with CAC is worse than the no-CAC case because of the regulation and control overhead. Nevertheless, CAC regulation is helpful to control the collision probability, as Fig. 10 shows. In Fig. 11, we show that the throughputs of background traffics with and without CAC. With CAC, because the resource usage is under control, the best-effort traffics also have chance to go through. So we conclude that admission control is necessary especially when there are multiple VoIP streams.

B. Influence of Host Mobility

Host mobility will affect the transmission rates of stations. Furthermore, the decreasing of transmission rate may run short of previous reserved resource. In this scenario, we assume that QSTAs

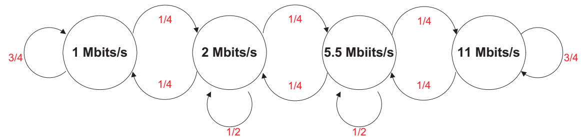


Figure 12: The state transition diagram of transmission rate when QSTAs are mobile.

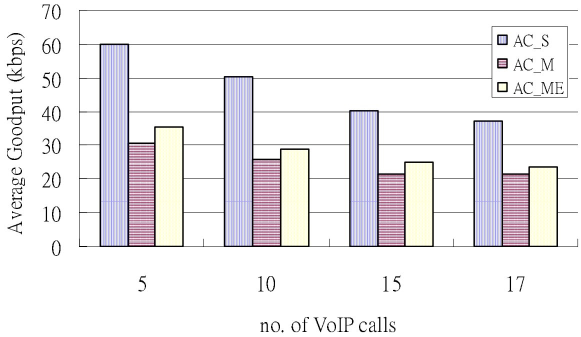


Figure13: Impact of host mobility: average goodputs per QSTA when there are 5, 10, 15 and 17 VoIP calls.

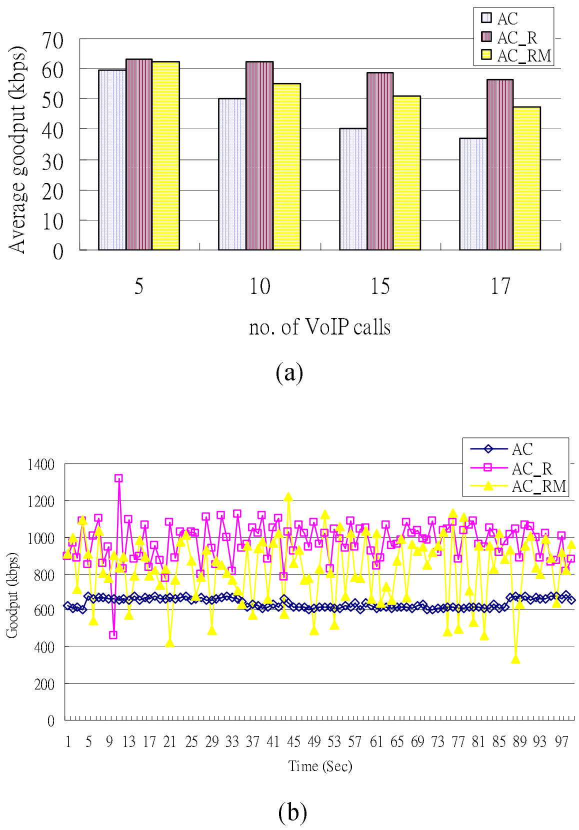
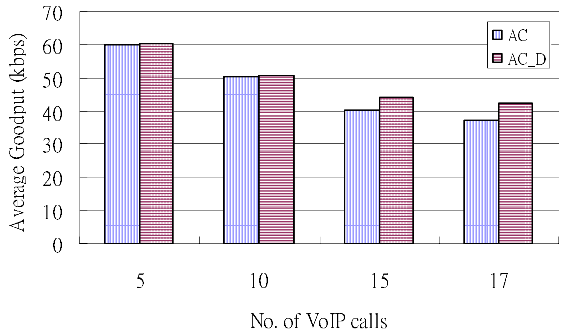


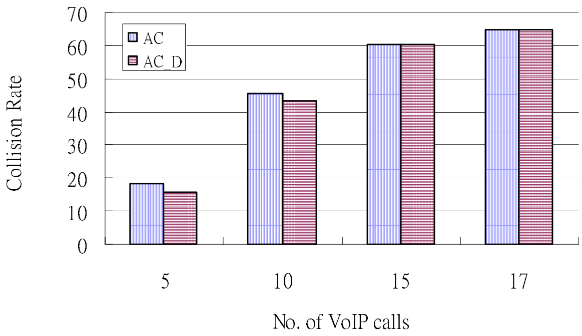
Figure 14: Impact of redirecting burst VoIP traffic to best-effort queues.

will move within the transmission range of the QAP and their transmission rates will change every second according to Fig. 12.

Our goal is to observe how our scheme can adapt to host mobility. Fig. 13 shows the average goodputs per QSTA when there are 5, 10, 15 and 17 VoIP calls in the network (with the existence of 2 background streams). AC_S stands for the static case



(a)



(b)

Figure 15: Impact of favoring downlink taffics: (a) average goodput and (b) collision rate.

where QSTAs always transmit at the rate of 11 Mbps. AC_M stands for the mobile case where hosts move according to Fig. 12 (the initial state is 11 Mbits/s) but no special treatment is taken. AC_ME means that host mobility is taken care of by our enhancement proposed in 3.2. Clearly, host mobility will decrease performance, but our enhancement can relieve the impact.

C. Influence of Redirecting Packets to Other Queues

In this experiment, we verify the effectiveness of transferring burst VoIP traffic to other queues (refer to Section 4). In Fig. 14(a) we show the average goodputs per QSTA without and with redirecting, which are denoted by AC and AC_R, respectively. AC_MR means that there is host mobility. Indeed, the performance is significantly improved. Fig. 14(b) shows the goodputs in the time domain in the same experiment when there are 5, 10, 15 and 17 calls.

D. Influence of Giving Priority to QAP

In this experiment, we verify the effectiveness of favoring downlink traffics (refer to Section 4). Fig. 15 compares the goodputs and collision rates without and with preference to downlink traffics, denoted by AC and AC_D, respectively. The enhancement does improve performance since the downlink traffics are delivered more smoothly.

6 Conclusions

In this paper, we have described several schemes to enhance the performance of VoIP calls by integrating the SIP and 802.11e. IEEE 802.11e is in the final stage to become a standard, so we choose to make good use of it to solve the QoS problem in WLANs. We believe that cross-layer protocol design is an important issue to facilitate special applications. In addition, we present several MAC enhancements to facilitate VoIP traffics under WLANs. Our simulation results show that these adjustments do help improve the network performance. For future work, it deserves to investigate the handoff problem where QoS is a concern.

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