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A Call Admission Control Scheme for Multimedia Support Over IEEE 802.11 Wireless LANs

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Abstract - These days there is an increasing interest for VoIP over wireless LANs. QoS support for real-time services like voice in the IEEE 802.11 WLAN is an important issue. Since IEEE 802.11 uses contention based MAC protocol – the distributed coordination function DCF, it is difficult to support the strict QoS requirements for voice in these networks. In this thesis a call admission scheme called "CAC" is proposed to achieve this goal, without changing the basic channel access mechanism of IEEE 802.11. CAC scheme regulates the arriving traffic in the wireless network to efficiently coordinate the medium among the contending traffic sources so that the network operates at optimal point, supporting the QoS requirements as well as providing better channel utilization. In this proposal, majority of available bandwidth is allocated to voice sources and remaining small amount is allocated for non real-time data traffic. It is expected that the proposed CAC scheme can well support strict QoS requirements, such as high throughput and low delay at the same time achieve a high channel utilization.

***Keywords** - IEEE 802.11, Wireless Local Area Networks (WLAN), Quality of Service (QoS), Medium Access Control (MAC).*

I. INTRODUCTION

A. Background

In recent years the mobile Internet has gained popularity and the IEEE 802.11 [1] WLANs have become widely accepted standard because of simple deployment and low cost. These years have also seen extensive growth in voice over Internet protocol (VoIP) applications across the globe. VoIP greatly reduces costs of long distance voice calls compared to voice calls over traditional circuit switched networks like PSTN, by delivering voice packets over the Internet. This system is fairly good over wired network, but in order to extend this technology over wireless domain, Quality of service (QoS) provisioning for real-time traffic is crucial. VoIP applications require WLANs to be able to support the strict QoS requirements of voice services. As per the ITU-T, G.114 recommendations, for real-time services the tolerable packet loss rate is 1 – 3% and the one way transmission delay is preferably less than 150 ms but should not be longer than 400 ms [3]. As devices grow smaller and more powerful, there is a general consensus that bandwidth demands on wireless networks will increase. We are already seeing a push to migrate the transmission of multimedia content to the wireless medium. When bandwidth hungry, delay sensitive media applications were first introduced to wired networks, the obvious and trivial solution was to supply more bandwidth as required. But in the wireless medium, bandwidth is not easily added. Strict limits on frequency use and strict constraints on power consumption mean that efficient protocols, and a clear understanding of these protocols, are crucial for provisioning of real-time applications like voice.

In IEEE 802.11 standard, to support real-time services like voice many challenges need to be addressed. The legacy IEEE 802.11 standard [1] MAC mechanism supports two access methods viz, the distributed coordination function (DCF) and a point coordination function (PCF). Though PCF is meant to support time bound services, it is an optional access method used only in infrastructure mode and not supported in all 802.11 based WLANs, whereas DCF is mandatory access method in all 802.11 based WLANs. DCF can well support non real-time data traffic but it introduces arbitrarily large delays and delay jitters, thus making it unsuitable for real-time applications where QoS requirements are stringent. In addition, unlike cellular networks where dedicated channels are assigned to voice traffic, voice packets in WLANs are multiplexed with data traffic. DCF leaves voice streams unprotected. When the best effort traffic load increases, the QoS of VoWLAN could be severely degraded. Thus it is a challenging job to provide QoS for voice traffic while maintaining as high throughput as possible for non real-time data traffic.

B. Literature Survey

To model a QoS mechanism for voice over IEEE 802.11 WLAN, extensive study of articles in the literature dealing with the IEEE 802.11 MAC protocol, its performance evaluation based on different metrics, QoS requirements for real-time traffic, and various schemes suggesting how to overcome the inherent problems encountered with the IEEE 802.11 protocol to support real-time traffic was carried out. The IEEE 802.11 [1] standard for WLAN explains in detail the MAC and PHY mechanisms dealing with different timing and backoff procedures. References [4, 5] examine the performance of IEEE 802.11 protocol. Zhai et al have defined channel busyness ratio as a performance metric for IEEE 802.11 in [4]. It is shown that the throughput increases linearly with channel busyness ratio and the delay remains almost constant upto certain value of channel busyness ratio. When channel busyness ratio increases beyond this point, the throughput decays drastically and delay rises dramatically. This suggests that, this turning point is the optimal operating point of operation for IEEE 802.11. In [5], the discrete probability distribution for MAC layer service time is presented, by modeling the exponential backoff procedure as a Markov chain. Performance is evaluated for both the saturated and non-saturated states. It is shown that in the non-saturated case, the performance is dependent on the total traffic and not dependent on the number of transmitting stations whereas, in the saturated case the number of transmitting stations affects the performance significantly. The ITU-T recommendations defining the one-way transmission time limits, for real-time services are given in [3]. According to the recommendations, for real-time services the tolerable packet loss rate is 1 – 3% and the one way transmission delay is preferably less than 150 ms but should not be longer than 400 ms. Reference [2, 6, 7, 8, 9, 10] suggest different approaches to provide QoS guarantees for real-time applications over IEEE 802.11 WLANs. The enhanced standard named IEEE 802.11e [2], was proposed to enhance the performance of IEEE 802.11 WLANs for real-time traffic like, voice, video etc. It supports service differentiation or prioritized service. Enhanced Distributed Channel Access (EDCA) mechanism is defined, which supports four access categories (AC's). Each of the access categories achieve differentiated channel access by varying the inter-frame spaces and the initial (minimum) and maximum window sizes for back off procedures. In [6] a call admission and rate control scheme is proposed. It is shown that if the WLAN is operated in such a way that the channel busyness ratio is held below the optimal point, the QoS requirements of real-time flows can be met. Wasan Pattara-Atikom and Prashant Krishnamurthy et al. describe several proposed distributed mechanisms at the MAC layer for providing QoS support in [8]. The QoS mechanisms proposed use well known QoS techniques, based on resource allocation (e.g. priority assignment and fair scheduling), and map QoS metrics into some existing 802.11 MAC parameter, thus avoiding a redesign of the MAC protocol. A taxonomy of the mechanisms is provided and the essential concepts, problems and advantages of each mechanism is described.

In [9], Yu et al. proposed a dual queue strategy to enhance IEEE 802.11 for VoIP, which runs at the MAC layer and does not require modification of the existing hardware. Proposed scheme basically implements dual queues (each for real-time and non real-time traffic) on top of the 802.11 MAC controllers. In reality, these two queues can be implemented in the device driver of the 802.11 WLAN devices. Basically, RT (real time) and NRT (non real time) packets are classified and enqueued into one of the two queues. Then, a strict priority queuing is implemented to serve these two queues in order to give a priority to the RT packets; the NRT queue is never served as long as the RT queue is non-empty. However, it cannot provide QoS guarantee for VoIP flows since best effort traffic is not regulated based on the global traffic condition. Qiang Ni et al. [10] provide a survey of variety of proposals for QoS enhancements for 802.11 WLAN. Firstly the QoS limitations of 802.11 DCF and PCF are presented, then various approaches of QoS enhancement along with some schemes (like IEEE 802.11e standard) is provided.

II. THE VOIP TECHNOLOGY

VoIP an acronym for Voice over Internet Protocol, also called IP Telephony, Internet telephony, Broadband telephony, Broadband Phone and Voice over Broadband, is the routing of voice conversations over the Internet or through any other IP-based network. Voice over Internet Protocol is a in demand technology of recent years, that enables users to reduce costs of long distance voice calls compared to voice calls over traditional circuit switched networks like PSTN. In this, the voice is digitized and sent as packets, over the Internet rather than conventional circuit switched network like PSTN.

A. VoIP challenges

Because UDP does not provide a mechanism to ensure that data packets are delivered in sequential order, or provide Quality of Service guarantees, VoIP implementations face problems dealing with latency and jitter. The receiving node must restructure IP packets that may be out of order, delayed or missing, while ensuring that the audio stream maintains a proper time consistency. This functionality is usually accomplished by means of a jitter buffer.

Some broadband connections may have less than desirable quality. Where IP packets are lost or delayed at any point in the network between VoIP users, there will be a momentary drop-out of voice. This is more noticeable in highly congested networks and/or where there is long distances and/or interworking between end points. As per the ITU-T, G.114 [3] recommendations, for real-time services the tolerable packet loss rate is 1 – 3% and the one way transmission delay is preferably less than 150 ms but should not be longer than 400 ms. Hence VoIP applications require WLANs to be able to support the strict QoS requirements of voice services.

B. IEEE 802.11 WLAN standard

In recent years, wireless LANs (WLANs) have become popular to access the Internet on the go, and IEEE 802.11 has emerged as the de-facto standard for WLANs because of its low cost and ease of installation/operation. IEEE 802.11 supports two types of architectural modes viz; Infrastructure mode and Ad-hoc mode.

III. THE IEEE 802.11 MAC

The IEEE 802.11 MAC sub-layer defines two medium access coordination functions, the basic Distributed Coordination Function (DCF) and the optional Point Coordination Function (PCF) [1]. 802.11 can operate both in contention-based DCF mode and contention-free PCF mode, and supports two types of transmissions: asynchronous and synchronous. Asynchronous transmission is provided by DCF whose implementation is mandatory in all 802.11 STAs. Synchronous service is provided by PCF that basically implements a polling-based access. Unlike DCF, the implementation of PCF is not mandatory. The reason is that the hardware implementation of PCF is thought to be too complex at that time. Furthermore, PCF itself relies on the asynchronous service provided by DCF. As specified in the standard, a group of STAs coordinated by DCF or PCF is formally called a basic service set (BSS). The area covered by the BSS is known as the basic service area (BSA), which is similar to a cell in a cellular mobile network. There are two different modes to configure an 802.11 wireless network: ad-hoc mode and infrastructure mode. In ad-hoc mode, the mobile STAs can directly communicate with each other to form an Independent BSS (IBSS) without connectivity to any wired backbone. In infrastructure mode, the mobile STAs can communicate with the wired backbone through the bridge of access point (AP). Note that the DCF can be used both in ad-hoc and infrastructure modes, while PCF is only used in infrastructure mode.

A. DCF: Distributed Coordination Function

DCF is a distributed medium access scheme based on carrier sense multiple access with collision avoidance (CSMA/CA) protocol. In this mode, an STA must sense the medium before initiating a packet transmission. Two carrier sensing mechanisms are possible: PHY carrier sensing at air interface and virtual carrier sensing at PHY MAC layer. PHY carrier sensing detects the presence of other STAs by analyzing all detected packets and channel activity via relative signal strength from other STAs. Virtual carrier sensing can be used by an STA to inform all other STAs in the same BSS how long the channel will be reserved for its frame transmission. On this purpose, the sender can set a duration field in the MAC header of data frames, or in the Request To Send (RTS) and Clear To Send (CTS) control frames. Then, other STAs can update their local timers of network allocation vectors (NAVs) to indicate this duration. As shown in Figure 1. if a packet arrives at an empty queue and the medium has been found idle for an interval of time longer than a Distributed Inter Frame Space (DIFS), the source STA can transmit the packet immediately [1]. Meanwhile, other STAs defer their transmission while adjusting their NAVs, and then the backoff process starts. In this process, the STA computes a random time interval, called Backoff_timer, selected from the contention window (CW): Backoff_timer = rand [0, CW] * slot time, where $CW_{min} < CW < CW_{max}$ and slot time depends on the PHY layer type. The backoff timer is decreased only when the medium is idle; it is frozen when another STA is transmitting. Each time the medium becomes idle, the STA waits for a DIFS and continuously decrements the backoff timer. As soon as the backoff timer expires, the STA is authorized to access the medium. Obviously, a collision occurs if two or more STAs start transmission simultaneously. Unlike a wired network, collision detection in a wireless environment is impossible due to significant difference between transmitted and received power levels. Hence, a positive acknowledgement is used to notify the sender that the transmitted frame has been successfully received, see Figure1. the acknowledgement is not received, the sender assumes that the transmitted frame was collided, so it schedules a retransmission and enters the backoff process again. To reduce the probability of collisions, after each unsuccessful transmission attempt, the CW is doubled until a predefined maximum value CW_{max} is reached. After each successful transmission, the CW is reset to a fixed minimum value CW_{min}.

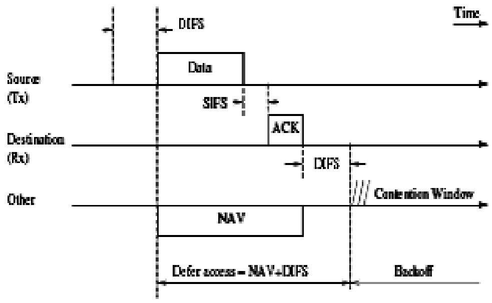

Fig.1. Basic DCF CSMA/CA

Hidden terminals are STAs that the receiver can hear but that cannot be detected by other senders. Consequently, the packets from different senders will collide at the same receiver. In order to solve the hidden terminal problem, an optional RTS/CTS scheme is introduced. The source sends a short RTS frame (20 bytes) before each data frame transmission, see Figure 2. and the receiver replies with a CTS frame (14 bytes) if it is ready to receive. After the source receives the CTS frame, it starts transmitting its frame. So, all other STAs hearing an RTS, a CTS or a data frame in the BSS can update their NAVs, and will not start transmissions before the updated NAV timers reach zero. Since a collision of a short RTS or CTS frame is less severe than a collision of data frame (up to 2346 bytes), the RTS/CTS scheme improves the performance of basic DCF scheme considerably in many cases. The overhead of sending RTS/CTS frames becomes considerable when data frame sizes are small, thus the channel is used sub- optimally. Moreover, an uncorrectable error in a larger frame leads to wasting more bandwidth and more transmission time as compared with an error in a smaller frame. So an optimization parameter of fragmentation threshold is used. That means, when data frame size exceeds this threshold, the data frame will be partitioned into several smaller MAC level frames.

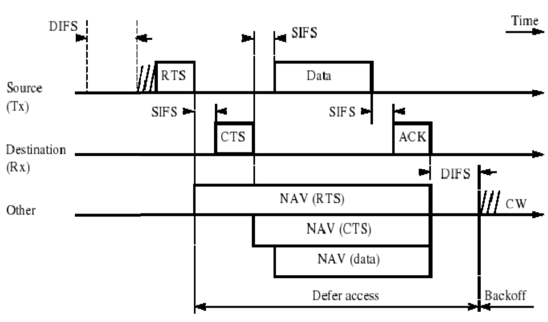

Fig. 2. RTS/CTS access scheme

IV. QOS PROVISIONING FOR VOICE OVER WLANs

A. *Quality of Service (QoS)*

QoS is the ability of a network element (e.g. an application, a host or a router) to provide some levels of assurance for consistent network data delivery. QoS refers to control mechanisms that can provide different priority to different users or data flows, or guarantee a certain level of performance to a data flow in accordance with requests from application program. There are several ways to characterize QoS in WLAN such as parameterized or prioritized QoS [11 and refs. therein]. Parameterized QoS is a strict QoS requirement that is expressed in terms of quantitative values, such as data rate, delay bound, and jitter bound. i.e. Network guarantees a set of QoS parameters for traffic e.g. ATM approach. In a Traffic Specification (TSPEC), these values are expected to be met within the MAC data service in the transfer of data frames between peer stations (STAs). Prioritized QoS is expressed in terms of relative delivery priority, which is to be used within the MAC data service in the transfer of data frames between peer STAs. i.e. Network treats traffic based on relative priority e.g. Diffserv approach. In prioritized QoS scheme, the values of QoS parameters such as data rate, delay bound, and jitter bound, may vary in the transfer of data frames, without the need to reserve the required resources by negotiating the TSPEC between the STA and the AP. In this paper our goal is to provide packet level parameterized QoS for voice flows over the IEEE 802.11 WLAN.

B. *QoS limitations of DCF*

DCF can only support best-effort services, not any QoS guarantees. Typically, time-bounded services such as Voice over IP, or audio/video conferencing require specified bandwidth, delay and jitter, but can tolerate some losses. However, in DCF mode, all the STAs in one BSS compete for the resources and channel with the same priorities. There is no differentiation mechanism to guarantee bandwidth, packet delay and jitter for high-priority STAs or multimedia flows. So, there is no way to guarantee the QoS requirements for high-priority audio and video traffic unless admission control is used.

V. PROPOSED SOLUTION: CAC

To provide QoS guarantees to the voice traffic over IEEE WLAN we propose a call admission control scheme called “CAC”. It is basically a measurement based admission control scheme, which provides packet level QoS for the voice traffic. The idea is to control the number of contending flows below network capacity so as to limit the collisions thereby reducing the delay.

A. CAC Overview

Our call admission control scheme can be summarized as under-

The CAC scheme determines when and how the packets are to be passed from the outgoing queue to the MAC layer to contend for the shared channel. The admission decision is based upon the availability of bandwidth resources required for the flows. It can be thought of as a control entity lying on top of the MAC sublayer protocol, a software upgrade approach hence no need to replace/upgrade existing hardware. Channel Busyness Ratio [4] is used as a measure of network status for traffic regulation, which can be obtained easily and can accurately represent the network utilization as discussed in the following section. CAC is able to provide statistical QoS guarantees for real-time traffic. Also it allows the non real-time traffic to utilize all the residual channel capacity left behind by the real-time traffic, without affecting their QoS level, thereby enabling the network to approach the theoretical maximal channel utilization. Each node keeps track of the channel busyness ratio locally to execute admission control, hence this scheme is distributed and well suited with the DCF mode of channel access.

B. Design Metrics

In this section we define the design metrics used and discuss why and how channel busyness ratio be used to represent the network status of IEEE 802.11 WLAN.

Channel busyness ratio R_b: It can be defined as the ratio of time the channel is determined to be busy to the total time. Busy time represents both; periods of successful transmissions as well as collisions.

$$R_b = \frac{\text{Busy Time(successful transmissions + collisions)}}{\text{Total Time}}$$

Channel utilization ratio R_s: It is the ratio of successful transmission periods to the total time.

$$Rs = \frac{\text{successful transmission period}}{\text{Total time}}$$

It counts every period T_{suc} with a successful transmission, which includes time for RTS, CTS, DATA, ACK and all necessary inter frame spaces i.e. SIFS and DIFS. R_b can be easily calculated using the physical and virtual carrier sensing mechanism of IEEE 802.11 CSMA based MAC. The channel is determined to be busy when the measuring node is sending, receiving or its NAV [1] indicates channel is busy, otherwise channel is considered idle. From the results, of work conducted by Zhai et al. [4], it can be seen that there is an optimal point for IEEE 802.11 DCF, which corresponds to certain amount of arriving traffic. At this optimal point MAC protocol can satisfy the strict QoS requirements of real-time traffic and achieve maximal channel utilization. Figure 3 presents ns-2 simulation results of [4] that illustrate the performance of throughput, delay and delay variation as a function of channel busyness ratio when RTS/CTS is used. Every node initiates an identical UDP/CBR traffic flow to a randomly selected neighbor. Different points in the Figure 3. correspond to different sending rates of flows.

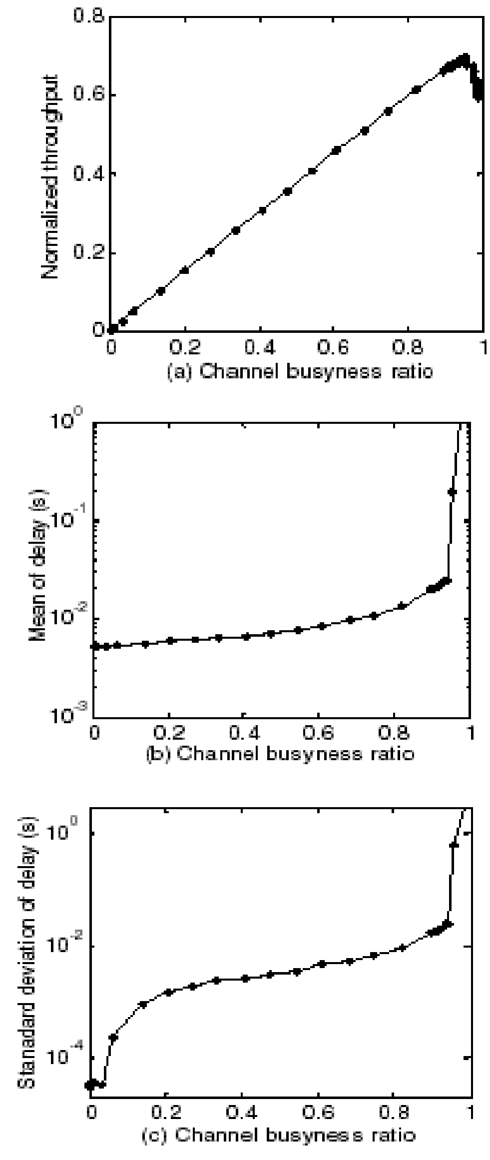

Fig. 3. Throughput and delay performance with 50 nodes, channel busyness ratio vs:a) normalized throughput, b) mean of delay (s), c)standard deviation of delay (s).

As can be seen from the graphs, there is a turning point in all the curves where the channel busyness ratio is about 0.95, before this point, as the input traffic increases the channel busyness ratio increases and the throughput keeps on increasing linearly with R_b, the delay (including queuing delay, channel contention time or back off time and transmission time) and delay variation only slightly increase and are small enough to support the real-time traffic. After this point, the throughput drops quickly, and the delay and delay variation increase rapidly.

Thus, this turning point is the optimal point that we should select the network to operate, where the throughput is maximized and, delay and delay variation are small. When the WLAN operates at the optimal point, there is almost no possibility of collisions and $R_b \equiv R_s$. R_b is stable around 0.9 (without RTS/CTS) or 0.95 (with RTS/CTS) independent of packet size or number of users [5]. Let B_U denote the channel utilization corresponding to the optimal point. Since $R_b \equiv R_s$; $B_U \equiv$ 0.95 or 0.90 (depending upon whether RTS/CTS is used or not) independent of packet size or number of active nodes [6]. Therefore, the network status is known by keeping track of the channel busyness ratio and can be used to regulate the total input traffic to support QoS. CAC should maintain R_b close to B_U to guarantee both, a good QoS level and high aggregate throughput.

C. CAC Mechanism

1. The call admission control mechanism CAC admits or rejects new traffic and shall guarantee the QoS level of the admitted traffic flow.

2. A new traffic flow is admitted only if the requested resources are available. The AP or point coordinator of the WLAN takes the admission decision for each traffic flow. Out of the total available bandwidth utilization B_U, we reserve 75% of bandwidth for real-time voice traffic and remaining 25% for non real-time background data traffic (which can be adjusted depending upon traffic composition).

3. Let B_M denote the share of the bandwidth for real-time voice traffic hence, $B_M = 0.75 * B_U$. And let B_N denote the share of bandwidth for non real-time traffic hence, $B_N = 0.25 * B_U$. This ensures maximum channel resources for real-time voice traffic, at the same time non real-time traffic remains operational all the time since it is allotted with some part of channel resources.

4. We model the voice traffic as VBR (variable bit rate) and background data traffic as CBR (constant bit rate). Three parameters viz; R, R_{peak} and len are used to characterize the bandwidth requirements of the traffic flows, where R is the average rate, R_{peak} is the peak rate (both in bits/sec) and len is the average packet length in bits. For CBR traffic, $R = R_{peak}$ and for VBR, $R < R_{peak}$. To conduct admission control, these parameters of voice flows are converted into channel utilization parameter 'u' (i.e. the channel time a flow will occupy) as:

   $u = R / len * T_{suc}$ (1)

   And $u_{peak} = R_{peak} / len * T_{suc}$

5. Similarly for data flow, if 'v' denotes the channel utilization we can have

   $v = R / len * T_{suc}$ (2)

   Where, T_{suc} is the transmission time of one packet, including RTS, CTS, Data and ACK and all the necessary inter-frame spaces i.e. DIFS, SIFS [5]. Therefore,

   T_{suc} = Data + ACK + RTS + CTS + 3 * SIFS + DIFS (with RTS/CTS) (3a)

   T_{suc} = Data + ACK + SIFS + DIFS (withoutRTS/CTS). (3b)

   Thus (u, u_{peak}) specify voice flows' bandwidth requirement and (v) specifies data flows' bandwidth requirement.

6. At the coordinator/AP, the total bandwidth occupied by all admitted real-time flows is recorded in two parameters, called the aggregate (u, u_{peak}) denoted by (u_A, u_{peakA}) and the total bandwidth occupied by all admitted non real-time data flows is recorded as aggregate (v) denoted by (v_A). They are updated when a flow joins or leaves the network through the following procedure. When a node wants to establish a flow, it must convert the bandwidth requirement into the form of (u, u_{peak}) or (v), and send a request with this requirement, to the AP/coordinator. Upon receiving a request with these parameters, the AP/coordinator examines whether there are enough resources to accommodate the new flow i.e. whether the remainder quota of B_M &/or B_N can accommodate the new traffic flow by carrying out the following procedure:

7. **For real-time voice traffic:**

   If $(u_A + u) \leq B_M$ & $(u_{peakA} + u_{peak}) \leq B_U$, the AP issues connection admitted message, and updates the value of (u_A, u_{peakA}) with ($u_A + u$, $u_{peakA} + u_{peak}$)

   Otherwise AP issues connection-rejected message.

8. **For non real-time data traffic:**

   If $(v_A + v) \leq B_N$, AP issues a connection admitted message and updates (v_A) with ($v_A + v$) Otherwise AP issues connection-rejected message. When the flows end, the source nodes of the flow should send a "connection terminated" message to the AP/coordinator. The AP/coordinator respond with a "termination" confirmed message and updates (u_A, u_{peakA}) or (v_A) respectively.

VI. CONCLUSION

In this paper we have proposed a simple and effective call admission control scheme (CAC) to support QoS of real-time and streaming traffic in the 802.11 wireless LAN. Based on the novel use of the channel busyness ratio, which is shown to be able to characterize the network status, the scheme enables the network to work at the optimal point. Consequently, it statistically guarantees stringent QoS requirements of real-time services, while approaching the maximum channel utilization.

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