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An Efficient Back-off Mechanism for Simulation Study of VoWLAN Capacity Improvement using OPNET

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Abstract — **Integration of delivering VoIP services over WLAN access networks (VoWLAN) increased the demand to support multiple of active simultaneous calls. Thus improving VoWLAN capacity has been addressed. The researchers are continuously contributing in improving VoWLAN capacity, (i.e. increasing the number of VoIP simultaneous calls), for various WLAN architectures. Although a lot of modifications have been developed for IEEE 802.11 standards toward improving its performance with real-time applications, all issued versions of this standard have a unique characteristic for infrastructure-based WLAN. That is MAC controls the access mechanisms for shared WLAN channels via distributed coordination function (DCF) entity. This paper aims to improve VoWLAN capacity by means of backoff time minimization for WLAN base station (i.e. Access Point – AP). AP MAC processing time (i.e. backoff overhead time) has significant effect on the downlink delay that all stations' packets suffer from. Weighting factor α is adopted for AP backoff time to minimize this downlink delay. This weighting factor points to the number of active VoIP clients during certain period. The new AP back-off time is the ratio of a uniformly distributed backoff random number to such weighting factor α. VoIP calls generated using adaptive multi-rate voice codec AMR (12 Kbps). OPNET simulator has been used to validate proposed mechanism. Obtained results show notable improvement for VoWLAN capacity (i.e. 25% increment in simultaneous VoIP calls).**

Keywords — **Backoff Time, DCF Access Mechanism, AMR Voice Codec, Voip, WLAN Capacity, WLAN Downlink Delay, OPNET.**

I. Introduction

Offering voice communication service as an Internet service (i.e. multimedia network application) mandates quantifying the Internet attributes to fulfil a required quality of service (QoS) level that is already offered through Public Switched Telephone Network (PSTN) [1]. Deployment of wireless network as an access network for multimedia applications grows dramatically. In particular, Wireless Local Area Networks (WLANs) have been elected widely as a last mile access network for offering voice communication service in many incorporations and campuses. From networking perspective it follows TCP/IP protocol stack except data link layer and physical layer. IEEE 802.11 standard and its amendments [2] describe the specifications of both data link layer, for medium access control mechanism (MAC), and physical layer (PHY) that responsible for injecting stream of bits into wireless medium. Providing channel access service for network layer is one of the crucial services that MAC entity provides. Unfortunately, providing voice communication service over a wireless channel is constrained with the voice QoS requirements (i.e. Delay falls in the 150 – 400 msec range, jitter ranges from 30 – 75 msec, and loss within 1% range) [1].

Accomplishing VoIP over WLAN (VoWLAN) transmission with acceptable quality level mandates attack the objections of providing such service – i.e. reliability, scalability, availability, range, signalling, security, and QoS – to be inline with PSTN service characteristics [3]. This article addresses the scalability issue via improving the VoWLAN capacity. This means increasing the number of simultaneous calls in single WLAN Basic Service Set (BSS) provided that all ongoing calls have the same level of voice quality. Although variety of studies has addressed the capacity improvement problem from different views of TCP/IP entities [4] – [19], this article presents minimization of downlink delay in WLAN BSS toward achieving such capacity improvement. Station-load based back-off time computation for WLAN base station (i.e. Access Point AP) is proposed to minimize the downlink delay that all BSS stations suffer.

The rest of the paper is organized as follow: related knowledge is given in section II. Proposed station load based backoff computation mechanism is demonstrated in section III. An experimental evaluation is described in section IV. Finally, conclusion and directions for future work are presented in section IV.

II. Related Knowledge

A. Preliminaries

Each VoIP client (i.e. WLAN stations/users), who would like to be served by WLAN access, is served by base station (i.e. access point AP) that covers a certain geographic area and relay/route data to another network or BSS. This coverage area that consists of AP and VoIP clients is called WLAN Basic service set (BSS). Each WLAN BSS has unique identification (BSS ID) that distinguishes it among the other BSSs. In this paper VoIP client or AP may be called BSS member. Once VoIP client authorized and authenticated to join certain BSS, it follows the infrastructure-based WLAN operation mode. In this mode AP and WLAN stations in BSS contend to access the wireless channel to transmit data stream. Distributed coordination function (DCF) identifies the channel access mechanism for all BSS members. It deploys the listen before talk concept (LBT). Each BSS member should senses the wireless medium prior transmitting intended data. Sensing the medium occurs for a certain time period called distributed inter-frame space period (DIFS). If channel state is busy BSS member will enter backoff state and wait for a random period, backoff time interval, before sensing channel again. For successful transmission the sending BSS member wait the acknowledgement from receiver for short interframe space time (SIFS). After

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successful transmission it enters *random backoff time* while sensing the medium to send the consecutive higher layer packets. Station enters back-off time whether frame transmitted successfully or not. However, the latter case back-off time selected uniformly from range of integers with upper bound of doubled contention window size (CW) of previous one until achieving maximum size. The former case backoff time selected from range of integers with upper bound of minimum CW. This protocol is called carrier sense multiple access with collision avoidance (CSMA/CA) detailed description can be found in [2].

B. Related Work

VoWLAN capacity improvement was exhaustively studied in literature. There are three salient improvement schemes. One proposes extending the coverage range through AP placement optimization and spatial distribution for the users in BSS [4] – [7] or via multi-hop communication scheme [8]. Second scheme presents to AP scheduling for user stations in the BSS through polling them in cyclic manner. This mechanism is alternative to DCF and it is called point coordination function (PCF) [9], [10], and [17]. Third one formulates the problem as minimization of the downlink delay with DCF mechanism [11] – [16], [18], and [19]. However, authors in third scheme built their hypothesis based on IEEE 802.11e mechanism. Although working with legacy DCF seems obsolete, all of IEEE 802.11 family has a DCF operation mode as a common operation mode.

In [11] the principle of transmission opportunity time slot (TXOP) in IEEE 802.11e mechanism was borrowed. The aggrementation concept has been proposed. The origin of this concept came from combining *aggre*gation of packets at AP and frag*mentation* of aggregated packets. This mechanism allows minimizing back-off time at AP through fragmentation of aggregated packets. The fragmentation of large aggregated MAC service data unit (MSDU) allows entering back-off only at the beginning of MSDU transmission; the time space between consecutive fragment packets is short interframe space (SIFS). However this mechanism mandates MAC layer perform addressing translation procedures for aggregated packets that destined to multiple stations. The processing delay of this addressing procedure has been overlooked in this work. In [12] the access priority mechanism, which presented in IEEE 802.11e standard, was borrowed to prioritize access point over the other stations of the basic service set (BSS). The priority governed according to the weight of AP queue size to the average weight of stations queue size. Thus the AP has more opportunity than the other stations to access the medium. Although this mechanism made balance between downlink delay and uplink delay toward capacity improvement, there is challenge of accurate interpretation of the average station queue size. On the other hand, the VoWLAN capacity of state-of-the-art codecs has been overlooked. In [13] AP has been prioritized over stations through adapting different DIFS and selecting small contention window size toward minimizing downlink delay. However, it lacks accurate estimation of the contention window weighting factor. In [14] authors proposed capacity enhancement mechanism in IEEE 802.11e. Their approach depends on centralized adaption of stations CW according to network condition (i.e. retransmission occurrence). On the other hand, TXOP adaption for AP only controls the uplink and downlink traffic. This scheme has increased G.711 codec capacity by 25% at most. Nevertheless, the adaption of CW depends on estimated value of retransmission rather than live measurement which may cause overestimation or underestimation problem. Furthermore, the process of counting ACK frames at AP to adapt its TXOP cause significant increase in AP media processing time. In [15] virtual contention free channel access mechanism was proposed. The aggregation mechanism presented in [11] has been deployed but with scheduling the stations for uplink transmission. The schedule was based on the service request of the members of the multicast group (i.e. aggregated packets), and the round-trip delay budget; hence the schedule is Round-Robin. The clients use this schedule and order their uplink transmission accordingly. Although analysis of this approach gave 300% improvement, its implementation is so complex that it can be realized. As stated before, the addressing for multicast transmission at AP should be taken into consideration while AP processing time being estimated. In [16] hybrid enhancement scheme has been proposed with IEEE 802.11e. It deploys contention free period (CFP) for voice transmission to avoid ACKs and retransmission delay. On the other hand, for contention period (CP) the voice packets are prioritized over data packets. Primarily, this work proposes capacity improvement through deploying ON/OFF voice sources, controlling CW and arbitrary interframe space (AIFS) to give priority of voice over data. Finally, it deploys header reduction for MAC and PHY layers. Nonetheless, proposed approach has not discussed the capacity improvement through minimizing downlink delay. In [18] authors carried out intensive analysis for optimizing TXOP in IEEE 802.11e toward improving VoWLAN capacity. However, investigating its efficiency with using generalized voice codec (e.g. AMR) has been overlooked. In [19] capacity improvement was accomplished through giving priority for stations that have low codec rate. The priority mechanism done in a sense analogous to that presented in IEEE 802.11e. AP is responsible for managing the priority among contending stations. Although it is brilliant scheme, it mandates continuous monitoring for the WLAN network status.

In [20], Tay et. al. carried thorough WLAN throughput analysis to accomplish throughput enhancement. They proved that average backoff has inverse proportional to the number of active base stations in BSS. Eventually, the probability of collision derived in the terms of average backoff; Equation (1) demonstrates such relation. Soul of proposed work came from this claim.

$$p = 1-(1-1/Bk_{avg})^{N-1} \tag{1}$$

In [21], Estepa et. al. studied the capacity of wide area core network (WAN) for VoIP calls with adaptive multi-rate (AMR) voice codec. The voice quality and system capacity trade-off is addressed in this paper. It was

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required to study the impact of improving the voice source quality on the system capacity. The voice quality was improved through generating silence insertion descriptor (SID) packets within silence period of voice model. They compared the performance of the network with and without SID packets generation. It was found that there is subtle difference in WAN capacity with AMR-SID voice codecs. Accordingly, the cost of improving voice quality against system capacity was not so much. Thus, this paper extended this work for WLAN access network.

III. Station Load-Based Backoff Computation Mechanism

Our hypothesis believes in that AP throughput is the most convenient indication that points to the number of active stations in the BSS. As this number excess a certain level the throughput value will degrade drastically. On the other hand, all user stations, i.e. VoIP clients, in BSS suffer from downlink delay that is affected by the AP processing time. In this section we define a weighting factor α that contributes in minimizing the backoff time of the AP.

A. Basic Idea

The most prominent factor in AP processing time is the time required to execute MAC entity procedures. Furthermore, the backoff time contribute significantly in this MAC processing time. Accordingly, we propose WLAN downlink delay minimization through AP backoff time minimization. The backoff time computation at the AP is minimized by dividing the original backoff time by weighting factor α. This weighting factor α is computed according to the BSS status. In other word, BSS status indicates to the current active stations in this BSS at instant time.

It is clear that AP throughput measurement indicates quantitatively to density of stations in the BSS. On the other hand, the individual station load could be transmitted to AP to be used in the proposed Back-off computation mechanism. As shown in Figure 1. AP computes its backoff time based on the station load measurement that encapsulated within received packets from user stations in BSS. There are two functional procedures.

B. Procedures at WLAN User Station

1. Station load measurement (***ρ***) data filed is defined in MAC data header with 2 Bytes long as shown in Fig. 1.
2. Station load measured as total packet size (bits) that arrives from higher layer (i.e. network layer) during certain time interval. Equation (2) defines station load measurement:

$$\rho = \frac{n.p}{PI} \qquad (2)$$

   where ***n*** is the total number of packets that arrive from higher layer and are being served or will be served by MAC entity within a given time interval ***PI*** that is packetization interval, ***p*** is packet size in bits.

3. Updating ***ρ*** field is executed as following:
   - At initialization ρ = application data rate (i.e. voice codec rate)
   - After first PI interval the measurement is updated with the current value of ρ as defined in (2).
   - For each PI interval ρ will be updated if and only if current measured ρ is greater than the last measured ρ.
4. Updated ρ value is encapsulated with MAC header to be sent to AP in the BSS.

C. Procedures at AP

1. AP average throughput measurement (*η*) is defined here as a ratio of number of aggregated frames to MAC processing time multiplied by average frame size.
2. Equation (3) defines achievable throughput

   $$\eta = (N.\ F / \tau) \qquad (3)$$

   where N is number of received frames within measurement interval (τ) and F is average frame size.
3. Measurement period (τ) = DIFS + SIFS
4. At initialization throughput assumed to be operating WLAN data rate.
5. In next MP η is updated with the newest measurement.
6. Updating throughput measurement occurs if and only if the next new measurement is greater than the last maximum measurement.

D. Implementation Details

The proposed backoff mechanism is implemented by OPNET Modeler 16.1 [22]. In the following, modification of back-off estimation in DCF IEEE 802.11 is described. On the other hand, the modification in voice generation model, to model AMR with Silence Insertion Descriptor (SID) packets that were generated during OFF period, is carried out too.

1) Modifications in WLAN Dispatch Process Model

In this parent process model, customization of interface statistic was done. That is AP throughput statistic measurement that will be measured by external process to update the throughput measurement in the child process to be an instant argument of <<wireless_lan_mac>> module in the AP node model.

2) Modifications in WLAN MAC Process Model

New state for back off computation for AP was defined. While invoking <<wlan_mac>> process model with transition to such state, the simulator will call the throughput-to-load ratio (α) function routine. Furthermore, station load measurement field in the WLAN data header structures was defined. New state variables as well as the function prototype were defined in header block of process model.

In function block new function, *<<static void wlan_ap_bkoff_ratio (double ap_thrpt, double sta_ld)>>* was created. It implements the transition executive in the process model while MAC transit from deferring packet state to backoff computation state provided that invocation is carried out by AP station only.

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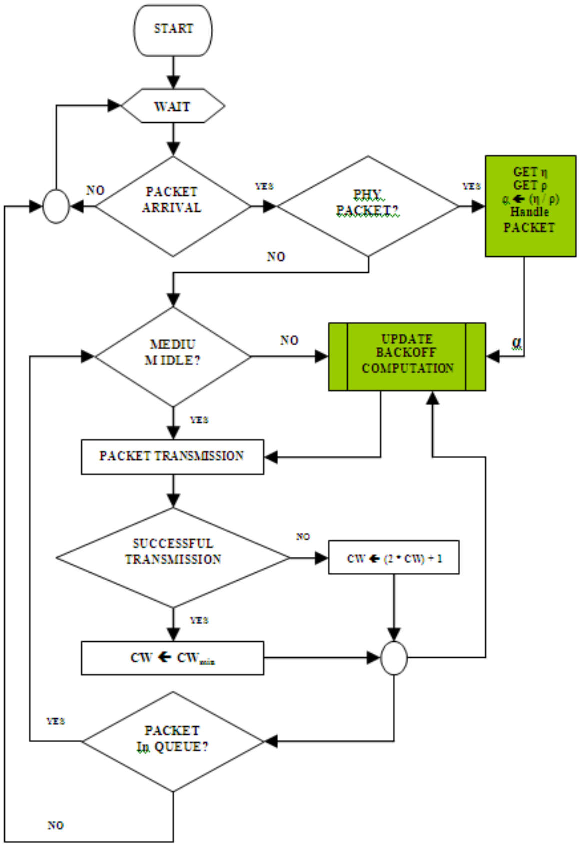

Fig.1. WLAN DCF access mechanism at AP

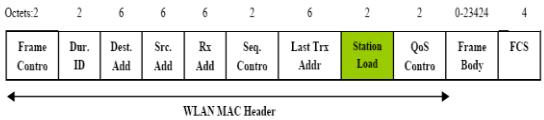

Fig.2. Modified WLAN MAC data header format

IV. Performance Evaluation

In this section we evaluate the impact of proposed backoff mechanism on VoWLAN capacity improvement. The downlink delay, packet loss, throughput, and uplink delay were adopted as performance metrics [11] – [16]. The system model was a BSS with AP and thirty VoIP clients. The simulations ran 10 times to collect the average values. The Modeler execution tool *op_run_sim* is configured with seed value 150. Seed value controls in the generation of random numbers within invoked process models. Obtained results were collected with average values out of 500 statistics per single run.

A. Evaluation of Capacity of AMR-12 With Modified Back-off

As shown from Fig.3,4. There is a significant improvement in downlink delay performance. However, it costs high uplink delay when number of stations excess twenty stations it jumps from tens of milliseconds to hundreds. That due to increasing the opportunity for AP to use the medium successfully, the stations should retry more than before to transmit their frames. Thus, AP retransmission rate has been minimized as depicted in Fig. 5. AP throughput pays off for such modification while number of BSS stations increases as shown in Fig. 6. Packet loss in WLAN takes place due to exceeding retry limit and/or buffer overflow. On the other hand, downlink delay has significant effect on retransmission attempt and buffer overflow. The effect on retransmission attempts is also demonstrated in Fig. 5. The latter, which occurs due to buffer overflow, is implicitly affected by delay. Thus while preparing MAC frame for transmission, the higher layer packets are arriving to the MAC queue. Hence, fast MAC transmission ***will not*** cause ***queue buffer overflow***. Accordingly, Fig. 7 emphasizes on that claim packet loss decreases as number of stations in BSS increase.

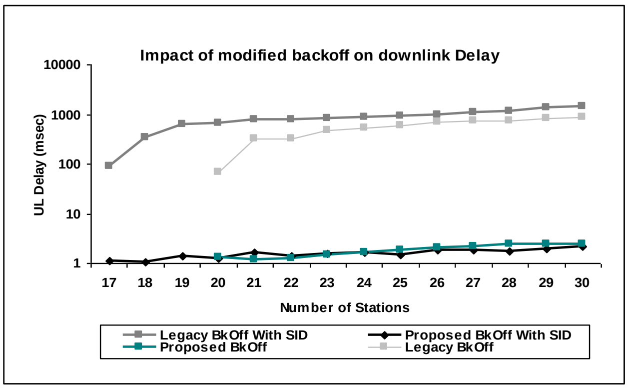

Fig.3. Effect of Modified Back-off on WLAN downlink delay of AMR-12 codec with and without SID

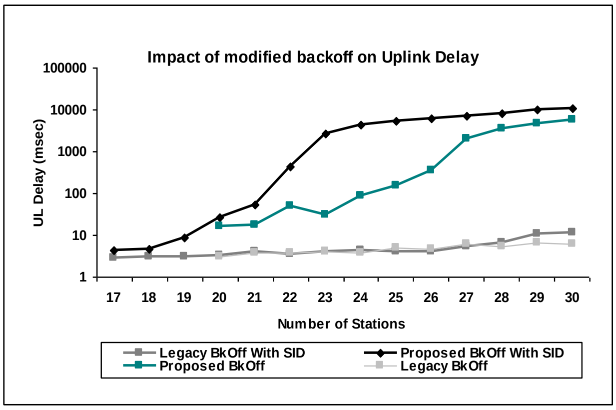

Fig.4. Effect of Modified Back-off on WLAN Uplink delay of AMR-12 codec with and without SID

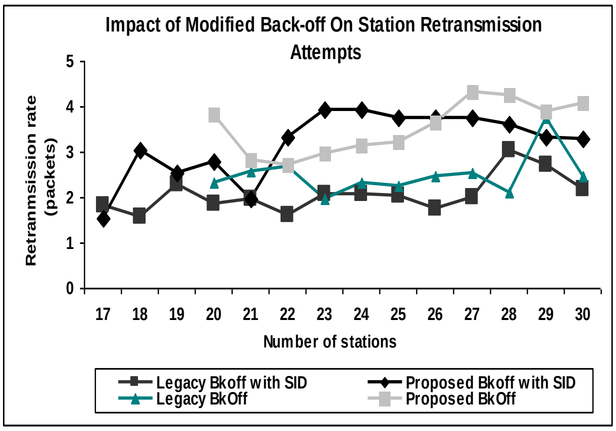

Fig.5. Effect of Modified Back-off on station retransmission attempts of AMR-12 codec with and without SID

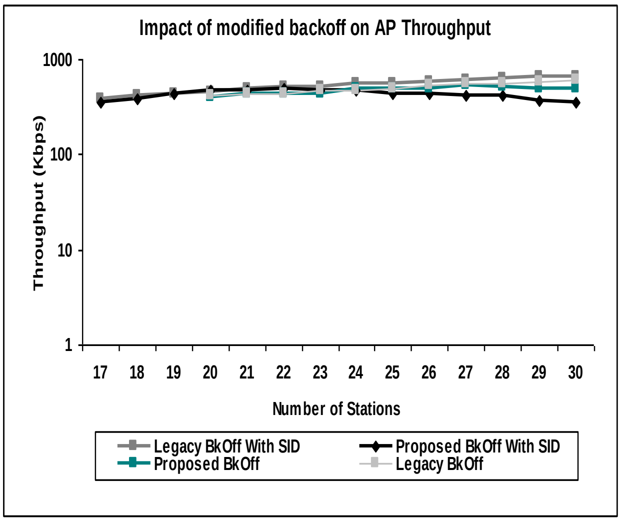

Fig.6. Effect of Modified Back-off on AP throughput of AMR-12 codec with and without SID

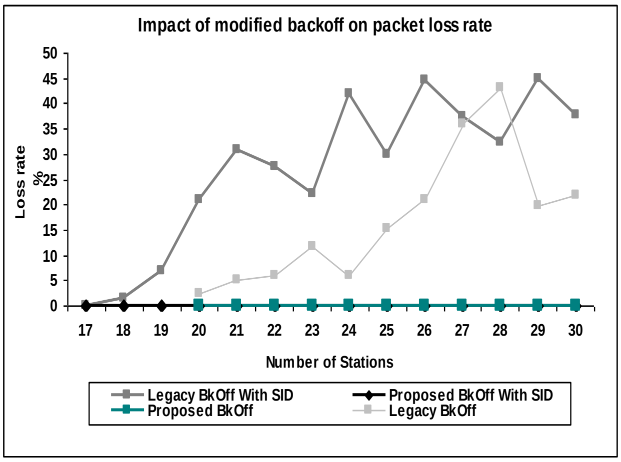

Fig.7. Effect of Modified Back-off on packet loss rate of AMR-12 codec with and without SID

Table I. Evaluation of proposed algorithm

Codec Type	Without α	With α	Capacity Improved by %
AMR (12 kbps)	20	25	25
AMR-SID (12 kbps)	17	21	23.529412

Table (I) summarizes the capacity improvement of VoWLAN with AMR-12 codec. It depicts the how weighting factor α contributed in AP backoff computation. Thus, downlink delay that user stations suffer from has been minimized. This minimization made the VoWLAN capacity constrained by another important metric that is uplink delay. Nevertheless, uplink delay of stations have been degraded drastically when BSS capacity exceeded 25 stations against 20 stations for AMR VoIP clients and 21 stations against 17 stations for AMR-SID VoIP clients. Thus, VoWLAN improvement claim holds.

V. Conclusions

This paper presented Station-load based AP back-off mechanism. It recomputes the AP backoff to minimize downlink delay. Weighting factor α was defined as a ratio of AP throughput to the maximum measured station load. This scheme improves VoWLAN capacity in a simple and efficient way. AMR-12 voice codec was deployed as a voice packets generator. VoWLAN capacity was improved by 25%. However, uplink delays as well as AP throughput pays off for such improvement. As WLAN standards advances rapidly, we are going to integrate the proposed mechanism with the state-of-the-art WLAN standards with co-existence of heterogeneous wireless systems. Offering heterogeneous applications that are running concurrently with multimedia application is also being investigated.

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