VoIPiggy: Implementation and evaluation of a mechanism to boost voice capacity in 802.11 WLANs

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***Abstract*—Supporting voice traffic in existing WLANs results extremely inefficient, given the large overheads of the protocol operation and the need to prioritize this traffic over, e.g., bulky transfers. In this paper we propose a simple scheme to improve the efficiency of WLANs when voice traffic is present. The mechanism is based on piggybacking voice frames over the acknowledgments, which reduces both frame overheads and time spent in contentions. We evaluate its performance in a large-scale testbed consisting on 33 commercial off-the-shelf devices. The experimental results show dramatic performance improvements in both voice-only and mixed voice-and-data scenarios.**

I. Introduction

IEEE 802.11 [1] is one of the most commonly used wireless technologies. It is being commoditized for voice communication, with the proliferation of smart phones with voice applications, e.g., Viber and Skype. Given the short length of voice frames, the legacy DCF operation results extremely inefficient and the voice quality is highly vulnerable to data traffic. On the one hand, the inefficiency issue is not solved by introducing higher data rates, since they do not change the protocol overhead and do not significantly reduce the fraction of time wasted due to the 802.11 backoff mechanism. On the other hand, voice quality vulnerability can be reduced by means of EDCA prioritization mechanisms [2]. As a trade-off, the performance of data frame has to be reduced to sustain a decent level of quality for voice traffic.

The impact of protocol overhead on VoIP has been extensively researched in the literature. The authors of [3] and [4] have investigated on the number of VoIP calls that can be supported in a WLAN with different 802.11 versions and different audio codecs. The degradation of voice performance in presence of low-priority data traffic has been analytically tackled in [5]. In that work, the authors propose an ACK skipping policy that optimizes the performance of voice frames. Other papers also discuss the importance of the MAC parameter settings on the voice performance, e.g., [6] and [7].

The literature also provides simulation results and experimental studies based on commercial off-the-shelf (COTS) devices to measure the capacity of WLANs when voice traffic is present. For instance, the authors of [6] show that appropriate MAC tuning can improve capacity by 20% to 40%. Experiments reported in [8] confirm that commercial devices need non-trivial prioritization mechanisms in order to guarantee the quality of voice. Experiments in [4] show how voice conversations dramatically impair the performance of UDP data traffic since they reduce the available bandwidth.

Motivated by the limited efficiency of the standard operation of 802.11 with voice traffic, we propose a simple mechanism to dramatically reduce the overhead of the MAC operation, which also results in a reduction of contention. Our proposal, named *VoIPiggy*, consists in piggybacking voice frames over MAC acknowledgments (ACKs). Our approach allows VoIP traffic to be served with lower delay and jitter, and, by embedding a significant part of voice frames into MAC ACKs, it reduces the average number of nodes contending for the channel, which eventually improves overall system performance. We implement VoIPiggy on COTS devices, and validate its operation against the legacy 802.11 operation.

The main contributions of this paper can be summarized as follows: (i) We propose a mechanism, VoIPiggy, to improve the general performance of WLANs when voice traffic is present; (ii) We describe the implementation of VoIPiggy using COTS devices; and (iii) we present an extensive performance evaluation in a large testbed of 33 nodes. These experiments show that VoIPiggy practically doubles the capacity of a WLAN in terms of voice calls.

The rest of the paper is organized as follows. Section II presents the rationale behind our proposal. In Section III we introduce the design of VoIPiggy and compare it to the legacy MAC. Section IV describes the implementation details of VoIPiggy over a COTS-based platform. In Section V we present an extensive performance evaluation. Finally, Section VI summarizes our main results and exposes the directions of our future work.

II. Motivation

The standard operation of 802.11 introduces a large overhead for the case of voice traffic, given its small frame size. To quantify it, let us consider the exchange of two voice frames between the Access Point (AP) and one station (STA). Neglecting the impact of the backoff operation for simplicity,

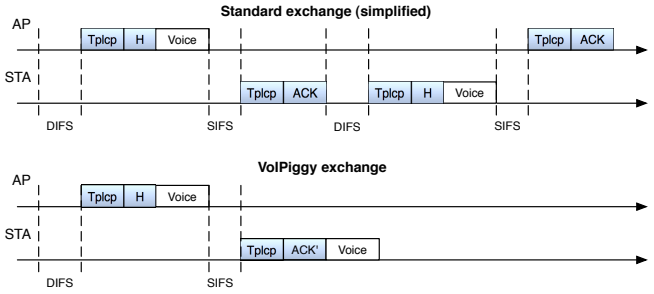


Fig. 1: Simplified frame exchange (top) and VoIPiggy proposal (bottom).

TABLE I: Total lengths of the frame exchanges of Fig. 1

Mode	R	R_c	$2L/R$	T_s	T_v
802.11b	1	1	1408	2968	2236
	11	2	128	785	415
802.11g	6	6	235	503	377
	54	24	26	200	111

the frame exchange will follow the one illustrated in upper part of Fig.1. According to the figure, the total time T_s required to perform this simplified two-frame exchange using DCF is given by:

$$T_s = 2\left(DIFS + 2T_{plcp} + \frac{H+L}{R} + SIFS + \frac{ACK}{R_c}\right), \quad (1)$$

where T_{plcp} represents the duration of the preamble, L is the payload length, H is the layer-2 header, ACK is the length of the acknowledgment and R and R_c are the transmission rates for data and control traffic, respectively.

In Table I we compute the value of T_s for different configurations of the physical layer, for a voice frame of 60 bytes transported over UDP. The results show that the total exchange T_s is significantly larger than the time actually devoted for payload transmissions $2L/R$, and it worsens as the transmission rate R increases. In particular, for 2 Mbps, T_s is more than twice the value of $2L/R$, while for 54 Mbps is almost 8 times larger.

This extremely simple analysis serves to illustrate that the basic access mechanism of the 802.11 standard, namely DCF (Distributed Coordination Function), is not well-suited to support voice applications. Given the short length of voice frames, DCF incurs a huge overhead, both in terms of backoff delay and MAC layers. In order to circumvent these limitations, we propose a simple modification to the MAC operation, named VoIPiggy, which piggybacks the voice frames over the MAC acknowledgments.

III. The VoIPiggy Mechanism

In this section we detail the operation of our proposal. First, we note that there are two sources of inefficiencies, inherent to the bi-directional nature of voice conversations: (i) A station (STA) sends an ACK frame immediately before its voice frame. Even neglecting the impact of the backoff, this basically doubles the introduced overhead (header and preamble) if a voice frame is immediately transmitted afterwards. Therefore, "merging" the upcoming voice frame with the precedent ACK frame seems an obvious choice to improve the efficiency; (ii) Furthermore, if a voice frame is sent in reply to a received voice frame a $SIFS$ time after the reception, it will not need to contend for channel access, thus preventing collisions.

Algorithm 1 STA operation

```
while true do
    while (Pck.out == null & !Rx) do
        Listen
    end while
    if queue.out! = empty then
        i = 1, T = 0
        if Pck.out == VoIP then
            while (T ≤ 25ms) do
                Listen
                if Rx(packet) == VoIP then
                    Send VoIP+ACK after SIFS
                    Remove packet from queue
                else
                    Legacy Tx
                end if
            end while
        else
            Legacy Tx
        end if
    else
        Legacy Rx
    end if
end while
```

Algorithm 2 AP operation

```
while true do
    i = 1
    while i ≤ MaxAttempts do
        Wait backoff and listen
        Send packet
        Wait for ACK or timeout
        if ACKreceived then
            Remove packet from queue
            if length(ACK)) > length(legacyACK) then
                Extract VoIP frame from the ACK
                Send VoIP frame to the upper layer
            end if
            Reset CW & Restart
        end if
        i++, CW* = 2
    end while
end while
```

These two observations motivate the design of our mechanism, whose operation is illustrated in the bottom part of Fig.1. As the figure shows, a $SIFS$ interval after the reception of the first voice frame, the STA sends in the same frame both the ACK and its voice frame towards the AP, which no further acknowledges its reception. As a result, the VoIPiggy mechanism saves airtime, and thus allocating a higher number of voice calls in the network.

In this way, we address the two sources of inefficiency identified above. Indeed, in this case the total time required for the two-frame exchange can be computed as:

$$T_v = DIFS + SIFS + 2T_{plcp} + \frac{H + ACK' + 2L}{R}, \quad (2)$$

where ACK' is the length of the modified acknowledgment header. As compared to the T_s values provided in Table I, T_v provides time savings between 55% and 75%.

As we have enlightened the voice frame from the STA to the AP is not acknowledged. We argue that this is not very critical, given that the main source for frame losses are collisions, and

this frame is protected from them as it is sent a $SIFS$ time after the medium was busy.

The use of VoIPiggy requires introducing changes in both the AP and the STA. For the case of the AP, although the first transmission follows the standard exchange, the node should be able to decode the subsequent frame from the STA, which includes both the acknowledgment for the frame (that triggers its removal from the transmission queue) and a new frame to be delivered to the upper layer. Furthermore, this frame shall not be acknowledged. Then, in this case we just modify the standard operation for the reception. We also prioritize voice traffic, thus we will set the minimum contention window of the AP to $CW_{\min} = 2$, which is the minimum allowed by the standard.

At the STA side, when a new frame from the AP is received the standard acknowledgment should be sent only if there are no pending voice packets towards the AP. Furthermore, to maximize the probability of piggybacking, incoming voice frames from the application layer will be queued until a timeout expires.

The operation of VoIPiggy is summarized in Algorithms 1 and 2, while its implementation is detailed next.

IV. Implementation Details

In this section we detail the implementation of the most relevant features of VoIPiggy. We describe the overall architecture, the development platform used for our implementation and the modifications to the Linux kernel and the device firmware to implement the VoIPiggy mechanism.

A. Linux 802.11 stack

The 802.11 network stack in Linux spans over three layers: (i) The *mac80211* framework that takes care of the operations related to 802.11 traffic; (ii) The *device driver*, which is a wrapper between the Linux internal 802.11 packet buffers and the physical device; (iii) The *device internal logic*, namely *firmware*, which controls time critical operations such as the ACK sending or the packet retransmission. These operations are offloaded within the hardware and hidden to both mac80211 and device drivers, due to the unpredictable latency and jitter that affect HW interfaces.

Since this is a proof-of-concept implementation, we opt to work at both the device driver and device internal logic levels leaving the *mac80211* framework unchanged.

B. The Broadcom chipset

We use the Broadcom chipset, which is based on a MAC processor (MP) that coordinates the data exchange among the different device blocks by running a binary firmware (FW) code. This code drives the transitions of the protocol state machine by reacting to external conditions, such as the arrival of a new frame or the expiration of programmable timers. A representative set of blocks that compose the chipset are presented as follows:

- **Tx FIFO queues**: driven by the DMA controller and deliver outgoing packets composed by the host kernel.
- **Tx Engine (TXE)**: The TXE prepares a frame for transmission adding the PLCP header and the CRC coefficient and waits for a transmission opportunity.
- **Rx Engine (RXE) and FIFO queue**: The RXE decodes the signal received from the air, checks the validity of the CRC coefficient and reports the received packet length.

The Broadcom chipset is supported in the Linux kernel by the open source driver, *b43*. The b43 driver passes each outgoing packet to the device together with a long block of data to setup the hardware on a per packet transmission basis, according to *mac80211* decisions. New private data can be passed to the device logic by extending this data structure.

The b43 driver loads the firmware at startup, therefore a different firmware can replace the original one. We use *OpenFWWF*, an "Open source FirmWare for WiFi networks" [9], enabling a very flexible customization of time critical operations. This platform has been used before to evaluate some Block Based Recovery (BBR) algorithms in [10] and [11], and more recently a TCP ACK-piggybacking MAC has been introduced by means of OpenFWWF [12].

C. Implementation

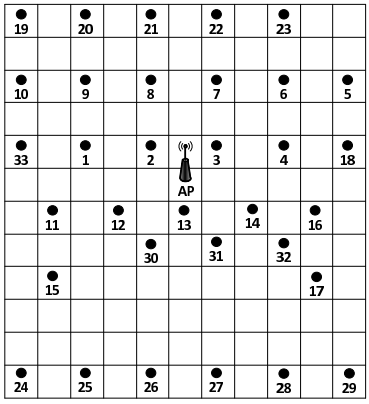
The VoIPiggy exchange described in Section III involves a legacy data frame from the Access Point followed by a Data+ACK frame sent by the corresponding station after a SIFS. For simplicity, we decide to implement our VoIPiggy reply by extending a legacy ACK. To accomplish this, we append the VoIP payload together with the IP and UDP headers skipping part of the data-type MAC header, as we are just interested in sending the payload, and the MAC header is already provided by the acknowledgment. Furthermore, the MAC address of the sending station is added between the legacy ACK frame and the appended IP packet, so that the AP can recognize it.

The legacy MAC operation is performed by the AP for every packet it transmits. Meanwhile, VoIPiggy mode is used by the station whenever it receives a VoIP packet incoming from the AP and the Head-of-Line (HOL) packet in its Tx FIFO queue is VoIP-data. In case the queue is empty or the HOL packet is not VoIP-type, the station will use the legacy operation.

1) Driver modifications: In order to develop our mechanism we modify the *b43* driver so that outgoing UDP traffic toward or from a specific port are marked as `PIGGYBACK_ENABLED` (PE).

We adjust the *b43* driver to optionally force the transmission of the VoIP traffic at a given modulation coding scheme configured by the user, which makes possible a better assessment of the VoIPiggy efficiency. Finally, we add a hook in the receiver code to intercept long ACK frames from the device. The driver must transform them back into full featured data packets by moving the IP section, inserting the missing MAC parts and finally sending them to the *mac80211* module.

2) Firmware modifications: At the firmware level, the code to be run in the AP is modified so that it can recognize long acknowledgments (i.e., VoIPiggy replies) by checking the received packet length reported by the RXE. When such



(a) Testbed scenario

(b) Detailed view of a node

Fig. 2: Deployed testbed with 33 wireless nodes.

packets are received they are pushed directly to the host kernel. In addition, the AP is prevented from acknowledging VoIP packets if they were piggybacked.

At the station side, the firmware required the following changes: (i) Prevent STA from waiting for an ACK for the piggybacked voice frame and remove the HOL packet in the queue immediately after its transmission under VoIPiggy operation; (ii) Tuning the ACK transmission. Before transmitting the packet the firmware overwrites the first ten bytes to mimic a legacy ACK frame header. Then, our modification properly sets up the TXE when the PE flag is set, by using the values precomputed by the kernel, i.e., bytes to skip, packet length and timing to use to build the PLCP; (iii) Delay outgoing packets when PE flag is set, until a voice frame is received or a maximum threshold T is reached. In this way it is more likely that a voice frame from the AP will arrive and therefore the HOL frame can be piggybacked.

V. Experimental Evaluation

In this section we experimentally validate our VoIPiggy implementation in a real testbed deployment and compare the results with the legacy operation.

A. Testbed Description

Our testbed is composed of 33 Alix 2d2 devices from PC Engine,[1] as depicted in Fig. 2. These embedded devices are popular low-cost computers, equipped with a Geode LX800 AMD 500 MHz CPU, 256 MB DDR DRAM, 2 Mini-PCI sockets and a Compact Flash socket, to which we attached a 4 GB card with a Linux distribution. As a wireless interface we installed a Broadcom BCM94318MPG 802.11b/g MiniPCI card, while as software platform we installed Ubuntu 9.10 Linux (kernel 2.6.29), using the modified b43 WLAN driver described in Section IV.

One of the devices acts as AP, while the rest are stations associated to the AP, distributed as Fig. 2a shows. All nodes are equipped with a 5-dBi omnidirectional antenna and use a transmission power of 27 dBm. Stations are spaced a few meters from each other (squares in Fig. 2a represent 60 $cm \times$ 60 cm floor units), and the resulting link quality is excellent for all nodes to communicate with each other.

The deployment is set up under a raised floor (Fig. 2b), which protects devices from physical harm and provides radio shielding to some extent [13]. Configuration and control of the experiments are centralized in a single terminal, not shown in the figure.

For traffic generation we use *mgen*,[2] which supports the computation of relevant voice traffic metrics, such as delay, jitter and loss rate. In particular, latencies can be evaluated at the receiver side, by means of the timestamps inserted by *mgen* in all packets, provided that all nodes are synchronized. We run the *PTP daemon* [3] over the wired interfaces of the nodes, achieving synchronization with μs accuracy.

We emulate the voice behavior by running independent instances of the *mgen* traffic generation tool, each transmitting a 60-byte voice frame every $T_f = 20$ ms, following the behavior of the G.726 codec. In the case of data emulation, we use an instance of *mgen* run on a single station under saturation conditions with a packet length of 1472 bytes. After a calibration process, the timeout threshold during which stations wait for a voice packet from the AP is set to $T = 25$ $msec$.

B. Voice-only scenario

We start our performance evaluation with a scenario in which only voice traffic is present. In all considered cases, we will assume that a voice call is active between a station in the WLAN and a node outside the WLAN, which is translated into a "downlink" (DL) flow from the AP to the wireless station, and a corresponding "uplink" (UL) flow in the other direction.

We analyze the maximum number of flows that can be admitted in the WLAN. To this aim, we compute the obtained *mgen* throughput in each direction as a function of the number of voice flows n in the WLAN. The obtained results for the worst performing flows are depicted in Fig. 3 for the DL (top of the figure) and UL (bottom part of the figure) directions, for the standard DCF (denoted as *Legacy*) and for our VoIPiggy mechanisms, for two 802.11b modulation coding schemes, i.e., $R = 1$ Mbps and $R = 2$ Mbps.

The results show that the use of VoIPiggy is able to significantly increase the number of voice conversations supported in the WLAN. Indeed, while for the legacy case the maximum n values before losses become unacceptable are $n = 5$ for $R = 1$, and $n = 8$ for $R = 2$, for the case of VoIPiggy these values grow to $n = 8$ and $n = 13$, respectively. Results show that VoIPiggy almost doubles the capacity in a voice-only scenario.

C. Mixed voice-and-data scenario

Here we evaluate the maximum number of conversations in presence of a data flow and the data throughput performance.

Fig. 4 considers the legacy 802.11 MAC and the VoIPiggy, and depicts the data throughput achieved by the data flow for an increased number of voice conversations. For the

[1]PC Engines: http://www.pcengines.ch/

[2]The Multi-Generator Toolset: http://cs.itd.nrl.navy.mil/work/mgen/

[3]*Precision Time Protocol*: http://ptpd.sourceforge.net/

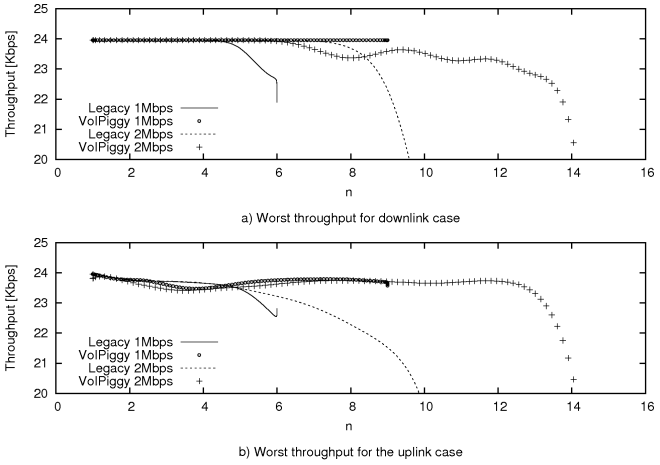


Fig. 3: Throughput delay for downlink and uplink cases (voice only).

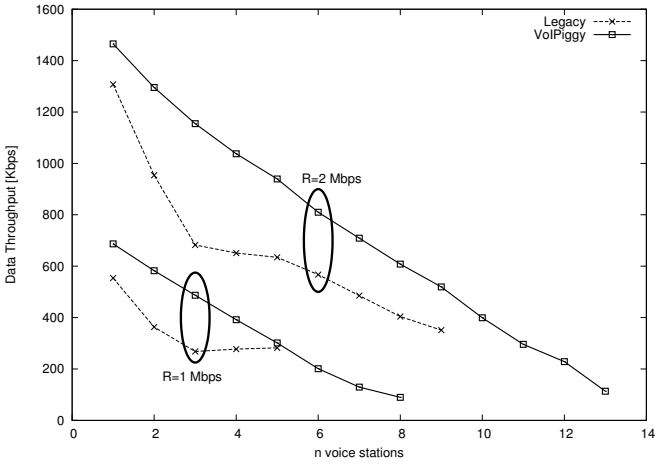


Fig. 4: Data throughput for data plus voice scenario.

legacy case, the system cannot support 2 voice flows with $R = 1$ Mbps, and no more than 3 flows with $R = 2$ Mbps. Therefore, legacy MAC turns out to be highly inefficient also in this scenario.

In the case of VoIPiggy, we conclude that: (i) The throughput of data decreases linearly with the number of voice flows; (ii) The maximum number of possible conversations with VoIPiggy is equal to 8 and 13, respectively with $R = 1$ Mbps and $R = 2$ Mbps; (iii) The data throughput, for a given value of n, is higher when using VoIPiggy as compared to DCF. This result shows that VoIPiggy increments the voice capacity of the WLAN and leaves more resources for data flows.

VI. Summary & Future Work

In this paper we have designed, implemented and evaluated VoIPiggy, a mechanism to dramatically improve the efficiency of MAC operation when voice traffic is present in 802.11 WLANs. In contrast to legacy operation, which spends large amounts of time in contention and overhead transmissions, VoIPiggy extends the control frames sent from the stations to the AP with user data, thus practically halving the time required to transmit voice frames.

We have described the modifications required by VoIPiggy, which are supported by existing COTS devices. To validate VoIPiggy and assess its effectiveness, we have deployed a large-scale testbed consisting of 33 devices. Through extensive performance evaluation we have demonstrated the performance improvements yielded by our mechanism, which provides a strong empirical support for the adoption of VoIPiggy.

As future work we envision the following tasks: (i) Assessing the performance when using real VoIP clients, i.e.: Skype, Ekiga or Viber. We have already carried out some preliminary tests with Ekiga, obtaining promising results; (ii) Implementing additional MAC enhancement mechanisms, e.g., enabling piggybacking at the AP; (iii) Testing VoIPiggy with higher modulation coding schemes.

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