802.11 MAC Protocol with Selective Error Detection for Speech Transmission*

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Abstract. The IEEE 802.11 standard currently does not offer support to exploit the unequal perceptual importance of multimedia bitstreams. All packets affected by channel errors, in fact, are simply discarded, irrespective of the position and percentage of corrupted bits. The objective of this paper is to investigate the effect of bit error tolerance in WLAN speech communications. More specifically, we introduce QoS support for sensitive multimedia transmissions by differentiating the scope of the standard MAC error detection step in order to discard multimedia packets only if errors are detected in the most perceptually sensitive bit class. Speech transmission using the GSM-AMR speech coding standard is simulated using a set of experimental bit-error traces collected in various channel conditions. Perceived speech quality, measured with the ITU-T P.862 (PESQ) algorithm, is consistently improved with respect to standard link layer technique. In other words, the results show that the negative effect of errors in the less perceptually important bits is clearly counterbalanced by the lower number of speech packets discarded because of retransmission limits. In fact, the number of received packets is consistently doubled throughout all the simulation conditions with quality gains that reach 0.4 points of the MOS scale in noisy scenarios.

1 Introduction

In recent years the IEEE 802.11 wireless standard has been adopted in an increasing number of places: many shopping malls, airports, train stations, universities, have now wireless infrastructures to provide people with tetherless access to the Internet. This emerging scenario is creating the basis of a new set of services based on the communication opportunities offered by ubiquitous network accessibility. In particular a great deal of interest is focusing on interactive voice communication applications over Wi-Fi links.

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However several challenges need to be addressed to provide successful speech services over a network originally intended for generic data traffic and characterized by potentially high error rates. While for data transfers, in fact, throughput is the main parameter for measuring the network performance, multimedia services depend on strict quality of service (QoS) requirements in terms of packet loss and delay. Moreover, in current data-oriented WLAN's, packet losses are also due to the need of data integrity during transfers, i.e., every hop discards all packets affected by channel errors, irrespective of the percentage of corrupted data. This approach does not exploit what modern multimedia compression algorithms offer, namely, a certain degree of error resilience, so that the decoder can still benefit from corrupted packets. As a consequence a new error tolerant extension to the MAC layer is advisable for multimedia transmission in wireless environments.

In this paper we analyze the advantages of modifying the link layer of IEEE 802.11 networks allowing partially corrupted speech packets to be forwarded (and not discarded) without requiring additional retransmissions. We find that this new functionality enables differentiated treatment of voice streams and can be tuned to meet speech QoS requirements of low delay and losses.

IEEE 802.11 Medium Access Control (MAC) layer [1] provides a checksum to prevent forwarding of erroneous frames: if a bit or more are corrupted the packet is discarded and the sender will retransmit the data until a maximum retransmission limit is reached. Since speech data bits are known to have different perceptual importance [2], they can be packed in sensitivity order and a checksum applied only to the most important subset. Partial checksum will prevent useful frames to be dropped and will reduce the number of retransmissions, thus reducing the network load and delay when the error probability is high.

Previous work addressed the problem of multimedia transmission over lossy networks suggesting to apply selective checksums to the UDP transport protocol [3]: if the packet is received with errors, it is delivered to the application only if the checksummed bits are correct, otherwise the packet is dropped. First proposed by Larzon in [4], for progressive coding schemes, PCM audio, and MPEG video, UDP-Lite has then been studied by Singh [5] and Reine [6]. UDP-Lite has been shown capable of providing less end-to-end delay, reduced jitter, higher throughput, less packet losses, and better video quality than plain UDP.

According to the current IEEE 802.11 MAC standard UDP-Lite cannot be effectively employed on wireless networks because erroneous frames are dropped by the link layer before reaching the UDP layer. Also the recent 802.11e extension to the MAC standard [7], specifically proposed for multimedia applications, does not allow the adoption of partial checksum techniques. However the idea of an error tolerant 802.11 network has already been discussed in the literature. The performance of the UDP-Lite protocol is, in fact, evaluated for a WLAN scenario in [8] where the checksum coverage is limited to protocol headers, and the 802.11 MAC level error checking feature is completely disabled (thus disabling the retransmission mechanism too). [9] and [10] take a step forward with the idea of reflecting the UDP-Lite policy of sensitive and insensitive data in the

bytes: 2	2	6	6	6	2	0–2304	4
Frame Control	Duration/ ID	Address1	Address2	Address3	Sequence Control	Frame Body	FCS

← MAC Header →

Fig. 1. Frame format of an 802.11 data frame MPDU. Each data-type frame consists of a MAC Header, a variable length information Frame Body, and a Frame Check Sequence (FCS). The MAC overhead due to the MAC header and the FCS is 28 bytes in total.

MAC protocol. That permits to detect, discard, and retransmit heavily "perceptually" corrupted packets not only at the receiver-end, but also at every wireless hop. As stated in these papers, the link-layer implementation of selective error detection can be more effective especially in case of high end-to-end delay scenarios, where end-to-end retransmission would not be applicable. Hop-by-hop retransmission is, in fact, a prompter solution that enables the delivery of an "acceptable" packet with lower delay. The results obtained by means of network simulations in infrastructure [9] and in ad-hoc [10] scenarios show that error checking only the most sensitive part of the multimedia payload is very effective when speech transmission is considered: delay, packet loss and network load are significantly reduced. Substantial gains are also reported for video transmission, notably in [11], where the H.264 coder is considered.

In comparison with previous works, the original contribution of this paper is in the simulation and evaluation method. Instead of using a model to generate a realistic 802.11 bit-error behavior, we use experimental 802.11b error traces collected under various network conditions. Moreover our performance analysis is not limited to consider the overall packet loss rate, but the perceptual quality of GSM AMR-WB [12] coded speech transmission is measured using the ITU-T P.862 standard, i.e., the Perceptual Evaluation of Speech Quality (PESQ) [13]. Network simulations show that the speech distortion introduced by decoding partially corrupted packets is clearly lower than the distortion that would have been caused by their discarding.

The paper is organized as follows. In Section 2, we introduce the wireless Voice over IP scenario and we describe the selective bit error checking MAC protocol for wireless multimedia. Performance evaluation and quality results are presented in Section 3 followed by the conclusions in Section 4.

2 Selective Error Detection in 802.11 WLAN's

The wireless case is a quite challenging environment for packet communications. Hop-by-hop transmissions are affected by high error probability because of interference with others signal sources as well as fading. To overcome this problem the current 802.11 medium access control implementation forces a receiver station to discard every erroneous packet and a simple retransmission control policy is used to reduce end-to-end packet loss.

In this scenario Voice over IP applications can tolerate few losses ($< 3\%$); above that threshold, the communication quality becomes unacceptable. If many retransmissions are employed to reduce losses, the positive effects of receiving more packets is undone by the consequences of higher end-to-end delays that reduce the communication interactivity. Delays are influenced also by the shared nature of the wireless link, in fact, stations contend for a transmission opportunity for each packet transmitted, a severe problem in congested scenarios.

2.1 IEEE 802.11 MAC Protocol

The IEEE 802.11b physical layer describes a Direct Sequence Spread Spectrum (DSSS) system with an 11 Mbps bit-rate [1] operating in the industrial, scientific, and medical (ISM) band at 2.4 GHz. The fundamental transmission medium defined to support asynchronous data transfer on a best effort basis is called Distributed Coordination Function (DCF). It operates in a contention mode to provide fair access to the channel for all stations. If two or more stations initiate their transmission at the same time, a collision occurs. A collision avoidance mechanism based on a random backoff procedure is meant to reduce this problem.

Because of collisions and channel noise, each station must be informed about the success of its transmission. Therefore each transmitted MAC protocol data unit (MPDU) requires an acknowledgment (ACK). Acknowledgment are sent by the receiver upon successful and error free packet reception. To verify the data-unit integrity the frame format, shown in Fig. 1, provides a 4-byte Frame Check Sequence (FCS) field. The destination station compares the packet FCS with a new one computed over all the received MAC bits. Only if all the bits are correct the two FCS's match each other and the packet is positively acknowledged by sending an ACK frame back to the source station. If this ACK frame is not received right after the transmission the sending station may contend again for the channel to transmit the unacknowledged packet until a maximum retry limit is reached. For each retransmission the random backoff time increases because drawn from a double-sized time window (up to a maximum defined value), so that the probability of repeated collisions is reduced.

2.2 Selective Error Detection

The adoption of selective error detection in the IEEE 802.11 MAC layer, provided that packet discarding can be avoided as long as errors do not affect important bits of the multimedia payload, can produce several positive effects on the network.

To illustrate the behavior of the proposed technique let us consider an n-bit long frame with m bits covered by the checksum. For simplicity's sake, only in the current section, we assume that the bit error probability is uniform and equal to p, and each bit is independent of the others. The packet loss rate (PLR) of a packet with m checksummed bit is expressed by the equation $PLR_m = [1-(1-p)^m]$, independently by its size n. For a full checksum scheme, m is equal to n. If selective error detection is implemented then $m < n$ and a partial checksum, that do not covers all data bits, is employed.

If a maximum of $N-1$ retransmissions are allowed then the packet loss rate, that is the probability of N unsuccessful transmissions, is $PLR_m(N) = (PLR_m)^N$. Finally, consider the expected number of transmissions T for each packet when maximum N attempts are allowed:

$$E\{T\} = \sum_{i=1}^{N-1} i(1-PLR_m){PLR_m}^{i-1} + N \cdot PLR_m^{N-1}$$
$$= \frac{1-PLR_m^N}{1-PLR_m} = \sum_{n=0}^{N-1} PLR_m^n. \tag{1}$$

As expected, the reduction of checksummed bits effectively decreases the packet loss and the total number of transmissions. Consequently end-to-end delay and network load are reduced with positive effects on the quality of service that can be achieved by all the transmissions in the network.

Among the effects of selective error detection we should also introduce the concept of *corrupted* packet that refers to a packet where errors occur only outside the checksum coverage. The packet corruption rate (PCR) is then defined as [11]

$$PCR_m(N) = (1-p)^m \cdot [1-(1-p)^{(n-m)}] \sum_{i=0}^{N-1} [1-(1-p)^m]^i. \tag{2}$$

For the case of $N = 1$ the equation simply becomes: $PLR_n(1) = PLR_m(1) + PCR_m(1)$. With no retransmissions, in fact, the sum of corrupted and lost packets for selective detection shall be the same as the number of lost packets if the whole data is checksummed. The complex effect of retransmissions will be further investigated in Section 3. Retransmissions play a fundamental role in favoring full or partial checksum techniques. The latter solution guarantees a lower PLR often accepting lightly corrupted packets, while the former presents more packet losses, but it ensures the integrity of the received ones.

Performance evaluation of the proposed technique should not only be limited to considering the packet loss rate, but should also include the effect of decoding partially corrupted packets. Hence, in the following, we drop the simplistic assumption of uniform bit error probability, and we derive bit error rate and location from actual transmission measurements to prove the effectiveness of the proposed technique in a real scenario.

3 Performance Evaluation for Speech Transmission

Selective error detection performance is evaluated for speech transmission over a wireless channel using the wideband GSM Adaptive Multi Rate (AMR) coder [12]. Different channel conditions and maximum number of retransmissions are considered. Objective speech quality measures are then employed to compare the perceived speech quality of the received streams.

Figure 2 shows the error detection coverage in a 802.11 data MPDU with speech payload. At the application level voice is encoded by the 23.85 kb/s GSM

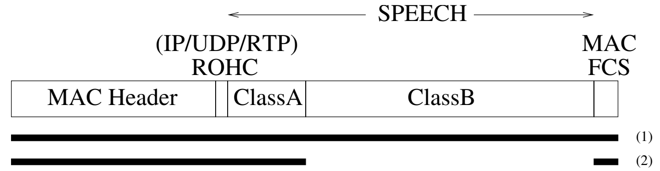


Fig. 2. Standard (1) and modified (2) MAC checksum coverage.

AMR-WB coder in frames of 477 bits. Speech encoder output bits are ordered according to their subjective importance and divided in two classes: Class A and Class B (as defined in the standard [14]). Class A (first 72 bits) contains the bits most sensitive to errors and any error in these bits typically results in a corrupted speech frame which should not be decoded without applying appropriate error concealment. Class B contains bits where increasing error rates gradually reduce the speech quality, but decoding of an erroneous speech frame is usually possible without annoying artifacts.

Real-time Transport Protocol (RTP) is used according to RFC 3267 specification for AMR encoded speech signals: ten control bits are present at the beginning of the speech payload and additional bits are added to the end as padding to make the payload byte aligned for a total of 61 bytes. For the RTP, UDP, and IP headers, Robust Header Compression is assumed that allows the 40-byte header to be compressed in 2 bytes. Then each data-type MPDU comprises a 24-byte header plus a 4 byte checksum. To support partial checksum in the modified MAC proposal an additional two-byte fixed-length field should be introduced to specify the number of covered bit, starting from the beginning of the MAC data unit.

3.1 Wireless Bit-Error Trace Collection

Characterizing the error behavior of the 802.11 channel is a fundamental issue that strongly influences wireless network simulations. While it is well known that wireless links have typically higher error rates than their wired counterparts, the detailed characteristics of wireless errors are not easy to reproduce in a computer simulation [15]. Lacking of a widely acknowledged model for simulating the wireless 802.11b bit-error behavior, we decided to use an experimental, trace-based approach.

The basic idea, for bit-error trace collection, has been to transmit a well-known packet stream over an 11 Mbps 802.11b wireless network using specially formatted UDP packets that include in the payload information for error detection such as a redundant sequence number and a repeated signature. For our modeling of packet errors, we decided to send sequences of 100-byte packets every 20 ms.

All the bit-error traces were collected at the receiver by modifying the wireless device drivers. More specifically, the receiver was a Linux box using a Prism2 wireless 802.11b PCMCIA-Card and Wlan-Ng (ver. 0.2.1-pre20) device drivers. When in monitoring mode the modified device drivers passed all packets to the

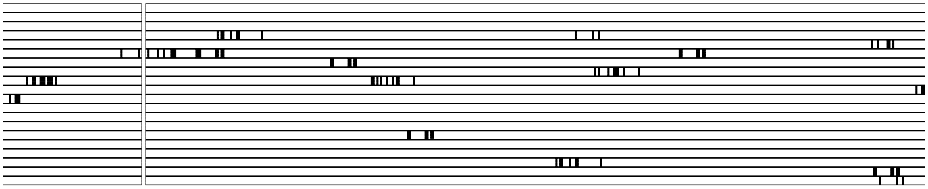


Fig. 3. Measured 802.11 error pattern mapped on transmitted speech data. Each line corresponds to a 23.85 kb/s GSM AMR-WB frame (the block on the left represents Class A bits). Erroneous bits are marked as black points.

upper network layers, thus the traces collected at the client by the network sniffer (ethereal ver. 0.10.0a, with libpcap 0.7.2) included successful (i.e., packets with no errors) and unsuccessful (i.e., packets failing the 802.11 standard MAC layer checksum) transmissions.

Due to our interest in analyzing bit errors inside corrupted packets, our traces provided bit-level information about all transmissions by bit-wise comparing sent and received packets. These bit-level traces were analyzed to study the wireless channel error characteristics and then used in the simulations to generate bit errors. Packet losses are simulated by using non overlapping windows on the bit error traces: any window with one or more errors (in the checksummed bits) is classified as a lost packet.

The bursty nature of the wireless channel is visible in Figure 3, where part of a trace is illustrated and bit errors are represented by black boxes. Error bursts clearly appear as bit sequences with errors at their ends but that allow few corrected bits within them. Bit-level errors are, in fact, a consequence of errors in decoding symbols of the Complementary Code Keying (CCK) modulation scheme used for the 11 Mbps data rate, where each symbol represents 8 bits of information. Thus, assuming a sequence of at least nine correct bits as a burst delimiter, the burst length distribution for an entire trace is as in Figure 4. The plot confirms what is already noticeable in Figure 3, the vast majority of bit error bursts last from 13 to 16 bits.

3.2 Results

MPDU transmission is simulated between two wireless 802.11 hops. Packet loss and corruption are modeled using the collected bit error traces and retransmissions are scheduled based on the error detection technique under examination. Corrupted speech frames are decoded as is without further processing by the AMR-WB decoder, lost frames are concealed as defined in the standard [16]. Perceived quality results are expressed my means of the objective quality measure given by the PESQ-MOS. Speech samples have been taken from the NTT Multilingual Speech Database. We chose 24 sentence pairs spoken by 2 English speakers (male and female). 9600 packets are transmitted in the simulations for a total of 192 seconds of speech.

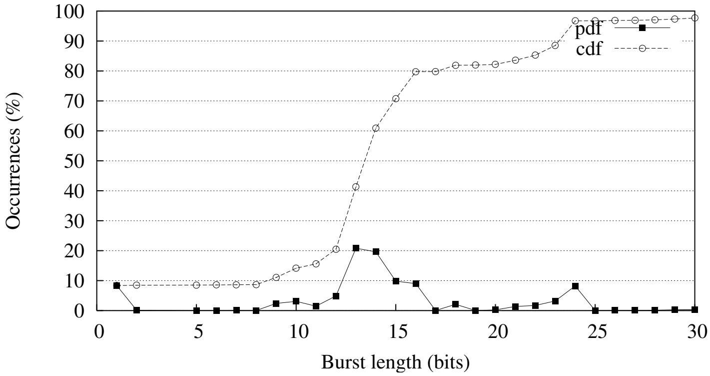
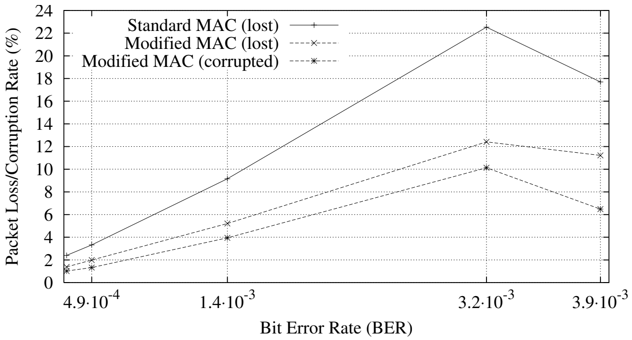


Fig. 4. Cumulative distribution and probability density function for burst error lengths in a trace. Mean burst length is 14.81 bits. An error burst begins and ends with an error and does not contain more than eight consecutive correct bits.

In the first simulation scenario two error detection strategies with different data coverage (see Figure 2) are tested without any link-level retransmission: the standard MAC checksum technique drops packets wherever a bit error occurs, while the modified MAC with partial checksum forwards packets only if errors occur in the perceptually least important part of the speech payload. Figure 5 presents the simulation results in terms of lost and corrupted packets *(top)* and objective speech quality *(bottom)* for different error traces. The percentage of lost frames is not always proportional to the bit-error rate because of the non-uniform nature of wireless errors, i.e., for the same BER, a lower number of packets is lost if bit errors have a higher burstiness. However, with selective error detection the percentage of lost frames is consistently lower. The PESQ score of the corresponding decoded speech also confirms that, for the scenarios under consideration, it is better to receive and decode partially corrupted packets than to lose them altogether. Improvements that range from 0.2 to 0.5 on the PESQ-MOS scale are clearly noticeable, proving modified MAC checksum particularly efficient at high error rates. This quality gain is motivated by the presence of a great number of bits only lightly sensitive to errors in the speech frame.

In the second simulation scenario, see Figure 6, a maximum of three retransmissions are allowed: the two techniques under analysis present different loss behavior *(top)* and quality *(middle)* , but also different number of retransmissions *(bottom)* given a particular BER. The modified MAC with partial checksum on speech data reduces the network load because in case of slightly corrupted packets it does not require a retransmission but forwards them to the next hop. Accepting a damaged packet proves to be a better solution than relaying on another uncertain transmission (especially at high bit error rates), provided that the most sensitive speech bits are correct and useful. The benefits are twofold: the perceived speech quality at the receiver is significantly increased, 0.3 on the MOS scale with 8% of packet losses, and the channel utilization due to retransmissions is roughly reduced by one third, with also positive effects on the end-to-end delays.



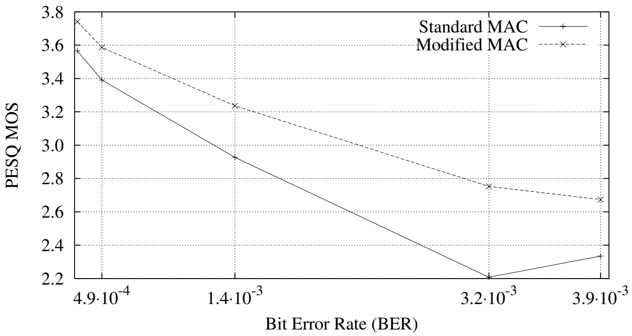


Fig. 5. Performance in terms of packet loss rate *(top)* and PESQ score *(bottom)* of IEEE 802.11 GSM AMR-WB speech transmission at 23.85 kb/s, with standard and modified MAC checksum, for different bit error rates. Link-level retransmissions are disabled.

4 Conclusions

A selective error detection technique to enhance the IEEE 802.11 link-layer effectiveness in supporting QoS sensitive speech communications has been presented. Since speech decoders can often deal with corrupted frames better than with lost ones, the standard 802.11 MAC-level error detection on the whole packet has been limited to the most perceptually sensitive bits of a speech frame. Retransmissions are then not required when bit errors occur outside the checksum coverage, on the assumption that they will result in only minor distortion. Full and partial checksum strategies have been simulated using 802.11b bit-error traces measured in different scenarios. Speech transmission shows a clear quality improvement if error detection is applied only to the most sensitive part of the payload. Furthermore, the proposed error detection technique also reduces end-to-end delays and network load, since successful packet delivery requires, on average, less retransmissions.

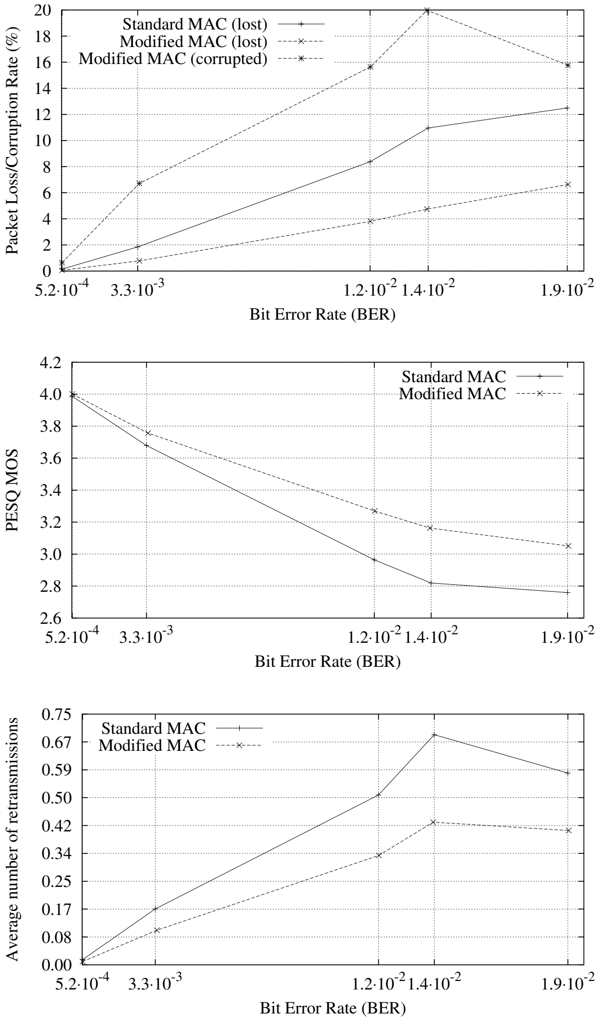


Fig. 6. Performance in terms of packet loss rate *(top)*, PESQ score *(middle)* and average number of retransmissions *(bottom)* of IEEE 802.11 GSM AMR-WB speech transmission at 23.85 kb/s, with standard and modified MAC checksum, for different bit error rates. The maximum number of link-level retransmissions is set to three.

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