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A Dynamic Link Adaptation for Multimedia Quality-Based Communications in IEEE_802.11 Wireless Networks

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1. Introduction

Assuming that the IEEE 802.11 Wireless Local Area Networks (WLANs) are based on a radio/infrared link, they are more sensitive to the channel variations and connection ruptures. Therefore the support for multimedia applications over such WLANs becomes non-convenient due to the compliance failure in term of link rate and transmission delay performance. Voice and broadband video mobile transmissions (which normally have strict bounded transmission delay or minimum link rate requirement) entail the design of various solutions covering different research aspects like service differentiation enhancement (Rebai et al., 2009), handoff scheme sharpening (Rebai, 2009a, 2009b, 2010) and physical rate adjustment. The core of this chapter focuses on the last facet concerning the link adaptation and the Quality of Service (QoS) requirements essential for successful multimedia communications over Wi-Fi networks. In fact, the efficiency of rate control diagrams is linked to the fast response for channel variation. The 802.11 physical layers provide multiple transmission rates (different modulation and coding schemes). The original 802.11 standard operates at 1 and 2 Mbps (IEEE Std. 802.11, 1999). Three high-speed versions were added to the original version. The 802.11b supports four physical rates up to 11 Mbps (IEEE Std. 802.11b, 1999). The 802.11a provides eight physical rates up to 54 Mbps (IEEE Std. 802.11a, 1999). The last 802.11g version, maintains 12 physical rates up to 54 Mbps at the 2.4 GHz band (IEEE Std. 802.11g, 2003). As a result, Mobile Stations (MSs) are able to select the appropriate link rate depending on the required QoS and instantaneous channel conditions to enhance the overall system performance. Hence, the implemented link adaptation algorithm symbolizes a vital fraction to achieve highest transmission capability in WLANs. "When to decrease and when to increase the transmission rate?" are the two fundamental matters that we will be faced to when designing a new physical-rate control mechanism. Many research works focus on tuning channel estimation schemes to better detect when the channel condition was improved enough to accommodate a higher rate, and then adapt their transmission rate accordingly (Habetha & de No, 2000; Qiao et al., 2002). However, those techniques usually entail modifications on the current 802.11 standard. In (del Prado Pavon & Choi, 2003), authors presented a motivating rate adaptation algorithm based on channel estimation without any standard adjustment. However, this scheme supposes that all the transmission failures are due to channel errors and not due to multi-user collisions.

Another way to perform link control is based on local Acknowledgment (Ack) information for the transmitter station (Qiao & Choi, 2005). Consequently, two new techniques (Chevillat et al., 2003; Kamerman & Monteban, 1997) where accepted by the standard due to their efficiency and implementation simplicity. In fact, the source node tries to increase its transmission rate after successive successful Ack responses, and therefore they do not involve any change for the 802.11 standard. Moreover and as it was demonstrated by (Sangman et al., 2011) a fine and excellent physical link adjustment will carry out a quality-aware and robust routing for mobile multihop ad hoc networks. A good study (Galtier, 2011) was recently addressed regarding the adaptative rate issues in the WLAN Environment and highlighted the high correlation between the Congestion Window (CW) of the system, and the rate at which packets are emitted. The given analytical approach opens the floor and shows that the different mechanisms that have been implemented in the MAC systems of WLAN cards have strong correlations with other transmission parameters and therefore have to be redesigned with at least a global understanding of channel access problems (backoff and collisions) and rate adaptation questions.

In this chapter we propose a new dynamic time-based link adaptation mechanism, called MAARF (Modified Adaptive Auto Rate Fallback). Beside the transmission frame results, the new model implements a Round Trip Time (RTT) technique to select adequately an instantaneous link rate. This proposed model is evaluated with most recent techniques adopted by the IEEE 802.11 standard: ARF (Auto Rate Fallback) and AARF (Adaptive ARF) schemes. Thus, we are able to achieve a high performance WLAN transmission. Consequently, we can extend this approach in various Wi-Fi modes to support multimedia applications like voice and video tasks.

The rest of the chapter is organized as follows. Section 2 offers a literature survey on related link-adjustment algorithms and the actual used ones. Section 3 is dedicated to the new proposed MAARF method and its implementation details. Simulation results will be given in Section 4 to illustrate the link quality improvement of multimedia transmissions over Wi-Fi networks and to compare its performance with previous published results (Kamerman & Monteban, 1997; Lacage et al.,2004). We show how the proposed model outperforms previous approaches (ARF and AARF) because of its new time-based decision capability in addition to Ack count feature.

2. Review of the current rate-control approaches

First we recall that the standard IEEE802.11 (IEEE Std. 802.11a, 1999; IEEE Std. 802.11b, 1999; IEEE Std. 802.11g, 2003) includes various versions a/b/g and allows the use of multiple physical rates (from 1Mbps to 54Mbps for the 802.11g). Therefore several studies have been made to develop mechanisms which lead to adapt transmission attempts with the best physical available rate depending on the estimated channel condition to avoid transmission failures with Wi-Fi connections. The most important issues that should be taken into account and are responsible for the design of a reliable rate adaptation mechanism are:

- The channel condition variation due to a packet transmission error which results to multiple retransmissions or even a transmission disconnection.
- The channel sensitivity against interferences (Angrisani et al., 2011) due to disturbing incidences, additive random noises, electromagnetic noises, the Doppler effect, an accidental barrier or natural phenomena.

- The packet emissions latency which affects the autonomy of mobile stations in case of transmission error (since the communication duration is extended and the energy consumption becomes more important).
- The MSs Mobility leads to a distances change, hence, to an appropriate mobility management protocol over Wi-Fi connections.

Depending on the instantaneous channel quality, a rate adjustment will be always needed to achieve better communication performance with respect for multimedia QoS requirements.

Since WLAN systems that use the IEEE802.11g (IEEE Std. 802.11g, 2003) physical layer offer multiple data rates ranging from 1 to 54 Mb/s, the link adaptation can be seen as a process of switching or a dynamic choosing mechanism between different physical data rates corresponding to the instantaneous channel state. In other words, it aims to select the 'ideal' physical rate matching the actual channel condition. The best throughput can be larger or smaller than the current used one. The adequate rate will be chosen according to the instantaneous medium conditions. There are two criteria to properly evaluate this adaptation/adjustment: the first is the channel quality estimation; secondly is the adequate rate selection.

The estimation practice involves a measurement of the instantaneous channel states variation within a specific time to be able to predict the matching quality. This creates a large choice of indicator parameters on the medium condition that may include the observed Signal to Noise Ratio (SNR), the Bit Error Rate (BER), and the Received Signal Strength Indicator (RSSI). Those various physical parameters express instantaneous measurements operated by the 802.11 PHY card after completion of the last transmission.

Regarding the rate selection formula, it entails a first-class exploitation of channel condition indicators to better predict the medium state and then fit/adjust the suitable physical rate for the next communication. Consequently, this process will reduce packets' retransmissions and the loss rate. Bad channel-quality estimation would result in performance degradation. Thus, inaccurate assessments resulting from a bad choice of medium state indicators give rise to inappropriate judgments on the instantaneous conditions and cause deterioration on the observed performance. Therefore, this estimation is essential to better support multimedia services and maximize performance and the radio channel utilization.

Accordingly, during packets transmission, the corresponding MS may increase or decrease the value of its physical rate based on two different approaches:

a. With the help of accurate channel estimation, the MS will know precisely when the medium conditions are improved to accommodate a higher data rate, and then adapt its transmission rate accordingly. However, those techniques (Habetha & de No, 2000; Qiao et al., 2002) require efforts to implement incompatible changes on the 802.11 standard. Another research work (del Prado Pavon & Choi, 2003) have presented a very interesting data rate adapting plan based on RSSI measurements and the number of transmitted frames for an efficient channel assessment without any modification on the standard. On the other hand, this plan operates under the assumption that all transmission failures are due to channel errors. Thus, it will not work efficiently in a multi-user environment where multiple transmissions may fail due to collisions and not only to the channel quality.

b. The alternative way for the link adaptation is to carry out decisions based exclusively on the information returned by the receiver. In 802.11 WLANs, an acknowledgment (ACK) is sent by the receiver after the successful data recovery. Only after receiving an ACK frame the transmitter announces a successful transmission attempt. On the other hand, if an ACK is either incorrect or not received, the sender presumes a data transmission failure and reduces its actual data rate down to the next available physical rate-slot. In addition, the transmitter can increase its transmission rate after assuming a channel condition enhancement by receiving a specific number of consecutive positive ACKs. These approaches (Qiao & Choi, 2005; Chevillat et al., 2003) do not require changes on the actual Fi-Wi standard and are easy to deploy with existing IEEE 802.11 network cards.

Various additional techniques have been proposed in the literature to sharpen the accuracy of the rate adaptation process and improve the performance of IEEE 802.11 WLANs. The authors in (Pang et al., 2005) underlined the importance of MAC-layer loss differentiation to more efficiently utilize the physical link. In fact, since IEEE 802.11 WLANs do not take into account the loss of frames due to collisions, they have proposed an automatic rate fallback algorithm that can differentiate between the two types of losses (link errors and collisions over the wireless link). Moreover it has been shown in (Krishnan & Zakhor, 2010) that an estimate of the collision probability can be useful to improve the link adaptation in 802.11 networks, and then to increase significantly the overall throughput by up to a factor of five.

In (Xin et al., 2010) the authors presented a practical traffic-aware active link rate adaptation scheme via power control without degrading the serving rate of existing links. Their basic idea consists to firstly run an ACK based information exchange to estimate the upper power bound of the link under adaptation. Then by continuously monitoring the queue length in the MAC layer, it would be easy to know whether the traffic demand can be met or not. If not, the emitting power will be increased with respect to the estimated power upper-bound and will switch to a higher modulation scheme. A similar strategy was presented in (Junwhan & Jaedoo, 2006) that provides two decisions to estimate the link condition and to manage both the transmission rate and power.

Several research works (Haratcherev et al., 2005; Shun-Te et al., 2007; Chiapin & Tsungnan, 2008) have implemented a cross-layer link adaptation (CLLA) scheme based on different factors as: the number of successful transmissions, the number of transmission failures, and the channel information from the physical layer to determine actual conditions and therefore to adjust suitably transmission parameters for subsequent medium accesses. As well in (Chen et al., 2010) a proper-designed cross application-MAC layer broadcast mechanism has been addressed, in which reliability is provided by the application layer when broadcasting error corrections and next link rate adaptations (resulting from the MAC layer).

Another approach (Jianhua et al., 2006) has been developed where both packet collisions and packet corruptions are analytically modeled with the proposed algorithm. The models can provide insights into the dynamics of the link adaptation algorithms and configuration of algorithms parameters. On the other hand, in (An-Chih et al., 2009) the authors presented a joint adaptation of link rate and contention window by firstly considering if a proper backoff window has been reached. Specifically, if the medium congestion level can be reduced by imposing a larger backoff window on transmissions, then there may be no need to decrease the link rate, given that the Signal to Interference-plus-Noise Ratio (SINR) can be sustained.

In the rest of this section we decide to provide details and discuss only the two main currently-implemented techniques.

2.1 Auto Rate Fallback (ARF)

Auto Rate Fallback (Kamerman & Monteban, 1997) was the first rate-control algorithm published and quickly adopted/integrated with the Wi-Fi standard. It was designed to optimize the physical rate adjustment of the second WLANs generation (specifically for 802.11a/g versions which allow multi-hop physical rates). The ARF technique is based simply on the number of ACKs received by the transmitting MS to determine the next rate for the next frame transmission. This method does not rely on hidden out-layer information, such as a physical channel quality measurement like (i.e. the SNR value). Thus, it was easy to implement and fully compatible with the 802.11 standard. In fact, after a fixed number of successful transfers equal to 10 or the expiration of a timer T initially launched the ARF increments the actual physical transmission rate from R_i to a higher rate R_{i+1} among those allocated by the standard. In other words, the ARF mechanism decides to increase the data rate when it determines that channel conditions have been improved. Unlike other algorithms reported in the literature, the ARF detection is not based on Physical layer measurements upon a frame delivery. Basically it simply considers the medium status improvement by counting the number of consecutive successful transmissions made or the timer (T) timeout. This timer is defined as the maximum waiting delay which will be launched by the MS each time it switches between the given data rates. Once this timer ends without any rate swap the MS will be testing a higher available rate. This practical implementation is considered as second alternative to adapt the best transmission rate since it covers the case that medium conditions are excellent and favorable to adopt a higher rate and the counter of consecutive successful transmissions will never reach the desired value (10) due to other failures. This case is very common in such wireless networks where a transmission failure is not only due to an inadequate rate.

In addition, the next transmission must be completed successfully immediately after a rate increase otherwise the rate will be reduced instantaneously (back to the old smaller value), and the timer T will be reset. In fact, the mechanism has estimated that the new adopted rate is not adequate for next network transmissions.

Also after any two consecutive failures the algorithm automatically reduces its actual rate until it reaches again a number of 10 consecutive ACKs or the expiration of timer T. In this way, ARF detects the deterioration of the channel quality based on two consecutive failed transmissions, and chooses to back out to the previous rate. Figure 1 summarizes the operation of the ARF and shows the corresponding flow diagram.

While ARF increases the transmission rate at a fixed frequency (each 10 consecutive ACKs) to achieve a higher system throughput, this model has two main drawbacks:

- Firstly, this process can be costly since a transmission failure (produced by an unsuitable rate increasing decision made by the ARF mechanism) reduces the overall throughput. Specifically, for a steady channel status (stable characteristics) ARF will try periodically to switch to a higher rate by default which leads to unnecessary frame transmission failure and reduces the algorithm efficiency.

- Secondly, ARF is unable to stabilize the rate variations. In fact if the channel conditions deteriorate suddenly, the ARF mechanism will be unable to respond fast to these changes and to suit the current state. It will carry out numerous transmission failures so that it reaches the desired rate value. Therefore, this algorithm cannot cope with rapid medium status changes.

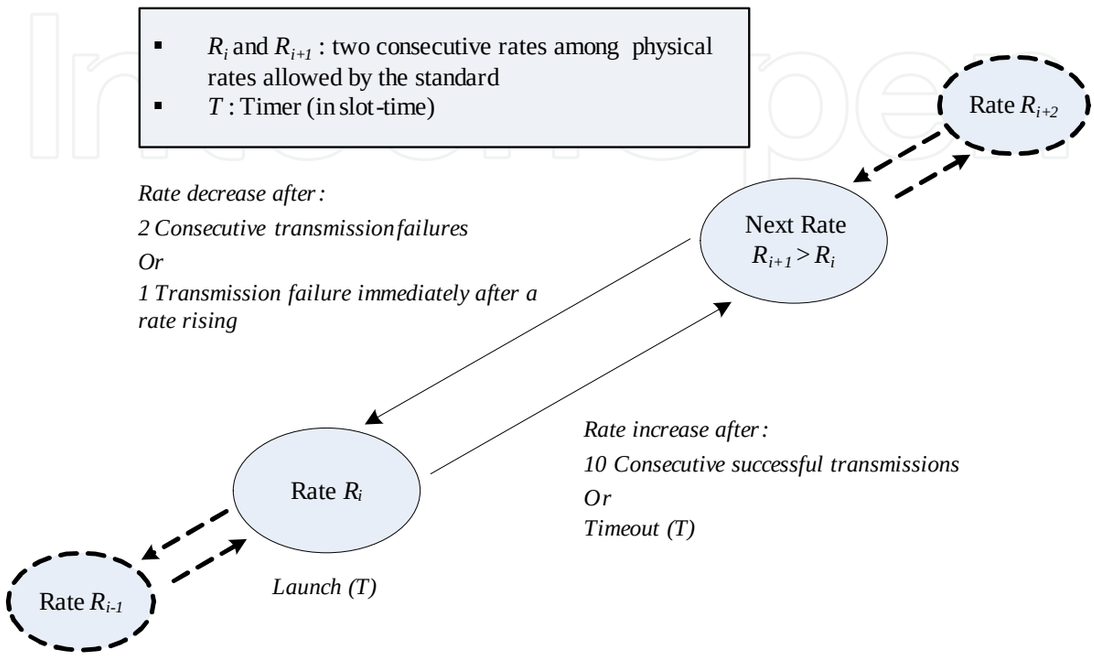


Fig. 1. The ARF flow diagram

2.2 Adaptive Auto Rate Fallback (AARF)

To overcome the given shortcomings, a new approach called Adaptive Auto Rate Fallback (AARF), was proposed (Lacage et al., 2004). It is based on the communication history and aims to reduce unnecessary rate variations caused by a misinterpretation of the channel state. Thus, this method controls the time-making process by using the Binary Exponential Backoff (BEB) technique (the same used by the CSMA/CD and CSMA/CA access mechanisms).

Therefore, when a packet transmission fails just after a rate increase, a lower rate is chosen for next transmission attempts. In addition, the number of consecutive successful transmissions n required for the next rate-switching decision will be multiplied by two (with a limit of n_{max} = 50). Similar to the old version in a rate decrease caused by two consecutive frames transmission errors, this value is reset to n_{min} = 10. The flow diagram in Figure 2 briefly explains the operation of AARF.

Consequently, this new version dynamically controls the number of positive ACKs needed for the rate control. Thus, AARF overcomes the old ARF version in case of a long steady channel conditions by eliminating needless and continuous rate-rising attempts. However it keeps the same disadvantage of the old implementation in case of rapid changes produced on the channel state.

Figure 3 illustrates the behavior of both ARF and AARF approaches for a time period equal to 0.4s needed for 230 data frames. Various physical rates were adopted in this experiment (1, 2, 5.5 and 11Mbps for 802.11b). During this experimentation, we set channel conditions supporting the use of the physical rate R_3 (5.5Mbps) for data transmission. We note that the period between two successive attempts is increased using the AARF technique while the ARF mechanism is trying regularly to increment the current rate to a higher value each ten successive successful transmissions. For example, within the time interval [0.2s, 0.25s] AARF doesn't create any unnecessary rate-switching effort, while ARF carries out three attempts. Likewise, the AARF algorithm considerably has reduced the number of produced errors due to bad decisions (3/4 of errors were removed compared to those generated by the ARF mechanism).

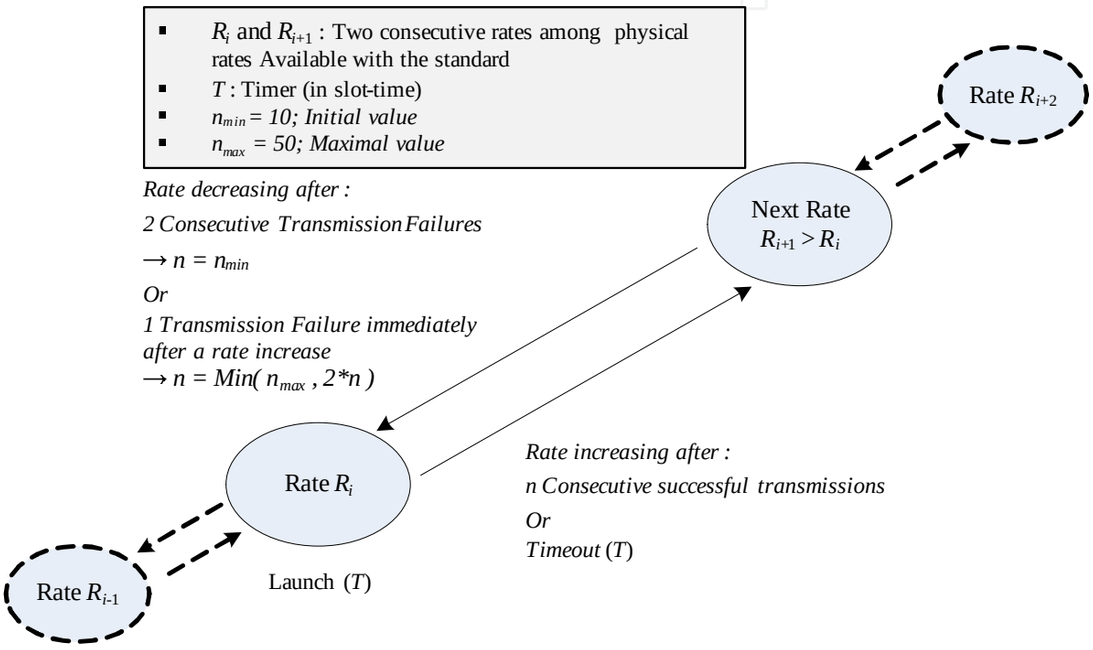


Fig. 2. The AARF flow diagram

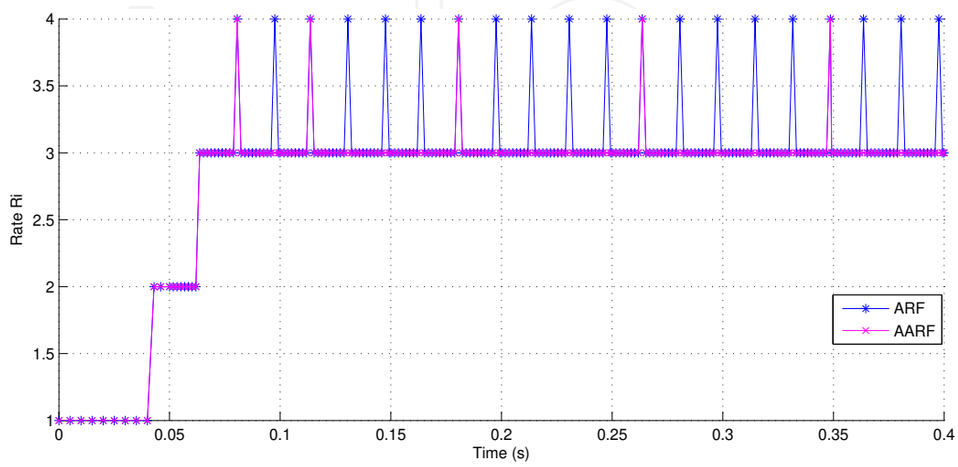


Fig. 3. ARF and AARF performance evaluation

2.3 Discussion

We have shown in the above study that the actual rate selection algorithms ARF and AARF do not conduct to an accurate decision when the channel is relatively noisy. Despite the given transmission enhancements both models still need improvement and refinement since they cannot react instantly to sudden changes of the channel state. In addition, an interval of time is needed to reach the maximum throughput in case of 'ideal' medium condition. Thus, these mechanisms do not represent optimal solutions for the physical link adaptation in noisy and 'ideal' environments.

Indeed, at a slow channel quality variation AARF is more suitable than ARF as it proceeds to the elimination of unnecessary rate increases. And thereafter, it decreases greatly the number of lost packets while relying on already rate exchanges made previously. However, this improvement is still insufficient since the decision criterion depends only on the nature of acknowledgments (ACKs), whereas this parameter no longer provides sufficient information about the instantaneous channel state. As result AARF need a high latency to reach the maximum throughput. In other words, a negative ACK (or lack of transmission success) is interpreted only by medium quality deterioration. However, this phenomenon may be caused by other networks anomalies (destination not reachable, collision occurred with another data frame, bad CRC, etc.).

It is also observed that when the competing number of stations is high, packet collisions can largely affected the performance of ARF and make ARF operate with the lowest data rate, even when no packet corruption occurs. This is in contrast to the existing assumption that packet collision will not affect the correct operation of ARF and can be ignored in the evaluation of ARF. Therefore, ARF and AARF can only passively react to the signal quality experienced at the receiver. In some occasions, we need to actively improve the signal quality in order to make the transmission rate to meet the traffic demand, even when the link length is a little large. This enhancement will optimize the overall performance and typically will demonstrate a practical effectiveness for multimedia transmissions over Wi-Fi WLANs.

Accordingly, in the next section we propose a new rate adaptation technique to improve the decision based on instantaneous channel conditions while respecting and still complying with the 802.11 standard. In addition, the new approach will be compared with those currently deployed. Simulation results will be also presented to demonstrate the enhancement of the proposed technique compared to those currently presented. Also parameters optimization of the new mechanism will be carried out to be then considered during next scenario simulations.

3. Proposed adaptive rate control technique

The main idea of the proposed method is to introduce a new channel status assessment parameter which cooperates with the number of ACKs to provide an efficient and accurate prediction of instantaneous channel conditions and subsequently to improve the actual rate adjustment mechanism. A logical way to cope with the slow accommodation characteristics of statistics-based feedback methods is to look for methods that use faster feedback, i.e., feedback that quickly provides up-to-date information about the channel status. Such a feedback — the RTT — has been theoretically discussed in (Rebai et al., 2008), but so far, to

our knowledge, it has not been used in a practical implementation. We use this RTT measurement in the proposed 802.11 radio to enhance multimedia performance, and also to provide feedback information about the channel conditions that the MAC layer requires. In this section, we first define the new parameter which will be required for the system design. Next, we will describe its implementation and the principle of operation.

3.1 Round Trip Time (RTT)

Reliable transport protocols such as Transport Control Protocol (TCP) (Tourrilhes, 2001) were initially designed to operate in 'traditional' and wired networks where packet losses are mainly due to congestion. However, wireless networks introduce additional sorts of errors caused by uncontrolled variations of the medium.

Face to the congestion problems, TCP responds to each loss by invoking congestion control algorithms such as *Slow Start*, *Congestion Avoidance* and *Fast Retransmission*. These techniques have been introduced in different versions of the TCP protocol (TCP Reno, TCP Tahoe, etc.).. These proactive algorithms consist to control the Congestion Window (CW) size based on observed errors. Another TCP-Vegas version has been proposed by (Kumar & Holtzman, 1998; Mocanu, 2004) and rapidly has been adopted by the TCP protocol since it includes an innovative solution designed for preventive systems. In fact, it performs a CW size adjustment based on a fine connection status estimation achieved by a simple measurement of the TCP segment transmission delay. This delay is called Round Trip Time (RTT) and represents (as illustrated in Figure 4) the time period between the instant of issuing a TCP segment by the source noted t_e and the reception time of the corresponding ACK noted t_r.

If the measured RTTs will have larger values, the TCP protocol infers network congestion, and reacts accordingly by reducing the congestion window size (symbolized by the number of sent frames and their size). If the values of observed RTTs become smaller, the protocol concludes an improvement on the medium conditions and that the network is not overloaded anymore. Therefore, it proceeds dynamically to increment the CW size, and thus a good operating performance will be achieved based on the new Vegas-version technique.

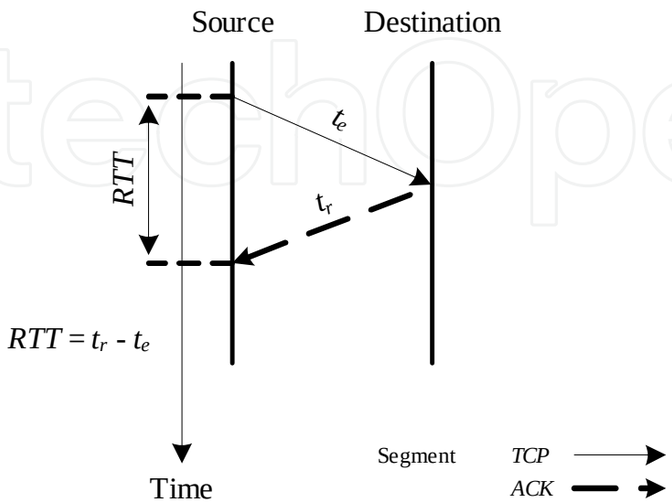


Fig. 4. The RTT delay computation

3.2 The RTT parameter integration

An interesting information and immediate channel observation will be deducted after each data frame transmission by means of RTT measurement and calculation. This feature represents the innovative part of the new control algorithm to adjust the data rate based on the channel capacity. A first integration attempt has been presented in (Rebai et al., 2008) and a rate adaptation design has been proposed. In this chapter, we implement an enhanced mechanism called *Modified Adaptive Auto Rate Fallback* (MAARF) which aims to predict the medium conditions and minimize the unnecessary loss of data. It chooses the appropriate rate value needed for the next transmission according to the measured RTT value. In fact, it performs a match between the observed value of RTT and the physical rate selection.

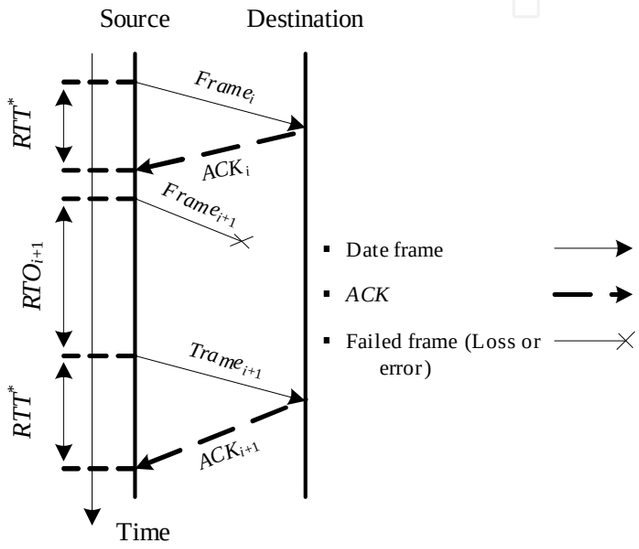


Fig. 5. The date frames transmissions

Furthermore we define two types of RTT. The first variety is the observed value directly measured from the channel following the frame sending and called instantaneous RTT denoted by RTT^*. The second, denoted by RTT_i, is a theoretical value computed based on the sending rate R and the data frame size. During a successful transmission of a frame i resulting the receipt of the associated ACK_i, a value of RTT^* is calculated. We introduce an associated recovery timer, called "Retransmission Time Out" and noted RTO_i, which will detect receipt/loss of a data frame. Based on this parameter, the transmitter detects the loss of a frame i in case of no receipt of the corresponding ACK_i till the expiration of the RTO_i timer. In this case, the last issued frame i will be retransmitted (see Figure 5).

Similarly, we introduce an interval defined by $[RTT_i^+, RTT_i^-]$ adjacent to the theoretical value RTT_i corresponding to the rate R_i. If the observed value of RTT^* belongs this interval (i.e. close enough to the theoretical expected value RTT_i) the channel conditions will be considered insignificant and do not require change of the current rate R_i for next transmissions. In other words, if the value of RTT^* belongs the interval $[RTT_i^+, RTT_i^-]$ the link quality is assumed stable and therefore it is suitable that the MS will transmit using the same current rate R_i since the measured RTT^* is considered close to the expected RTT_i value. Outside this window, channel changes are presumed:

- Improved, if the RTT^* value is less than RTT_i^+ since the corresponding ACK frame was received earlier than expected and therefore channel conditions had got better.
- Degraded, if the RTT^* value is greater than RTT_i^- since the received ACK frame was delayed and then we assume that the risk of data loss increases.

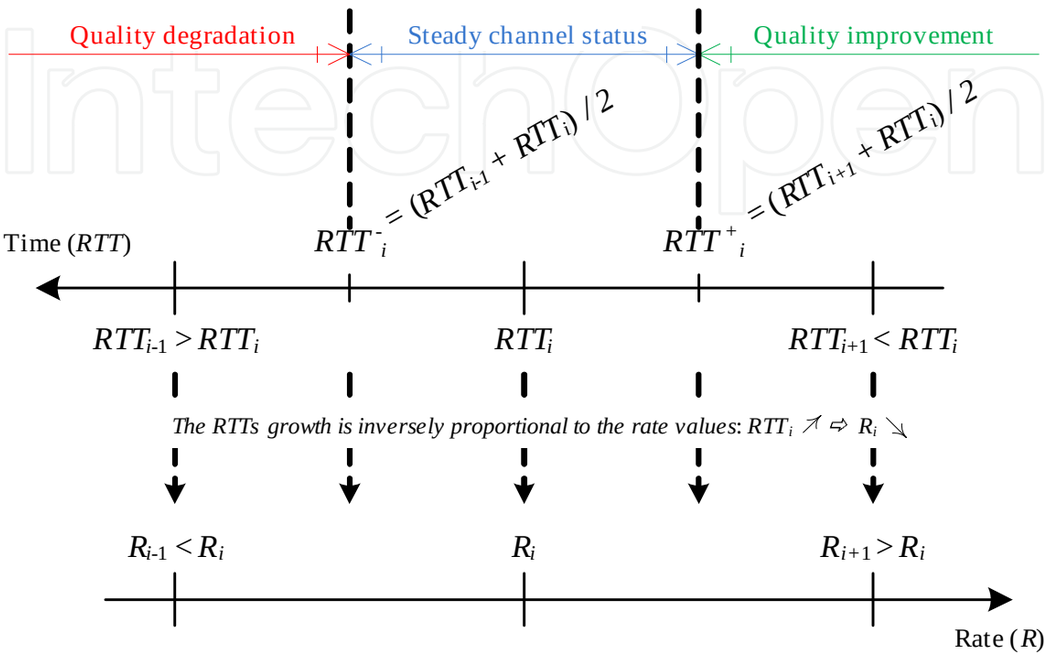


Fig. 6. The algorithm parameters set

In both cases, the MS must then change its emission rate and adapt it according to these instantaneous channel state interpretations. We point that $RTT_i^+ < RTT_i^-$ based on the statement $RTT_{i+1} < RTT_i < RTT_{i-1}$ since $R_{i+1} > R_i > R_{i-1}$. The calculation of the parameters values RTT_i^+ and RTT_i^- is associated to the RTT_i value as defined in the Equation 1 and 2.

$$RTT_i^- = (RTT_{i-1} + RTT_i)/2 \quad (1)$$

$$RTT_i^+ = (RTT_{i+1} + RTT_i)/2 \quad (2)$$

As stated in Figure 6 the RTT_i^- value will be the middle of the interval [RTT_{i-1} , RTT_i], and similarly, RTT_i^+ will have the center value of the interval [RTT_{i+1} , RTT_i].

3.3 Modified Adaptive Auto Rate Fallback (MAARF): Principle of operation

Subsequent to each successful frame transmissions, we compare the variation between instantaneous RTT^* and theoretical RTT_i values. More specifically, we test if the value of RTT^* has exceeded RTT_i^+ and RTT_i^- bounds or no. However, the according rate adjustment decision will be taken after several observations of RTT^* samples:

- As the number of consecutive successful transmissions did not reach a value of n (required as in AARF algorithm for the next rate-switching decision – it is initially initialized to n_{min}) we perform the following tests:
  - If the observed RTT^* value is less than RTT_i^+ ($RTT^* < RTT_i^+$) during h successive transmissions and the maximum speed (54 Mbps) is not yet acquired, then the instant RTT^* is considered smaller than the RTT_i value and rather close to the RTT_{i+1} one. Subsequently, the MAARF technique switches to a higher bit rate R_{i+1} starting the next attempt (since an improvement of the channel characteristics was interpreted).
  - If the value of RTT^* is greater than RTT_i^- ($RTT^* > RTT_i^-$) for the last g transmission attempts and the lower rate (6 Mbps) is not yet reached, this implies that the instantaneous RTT^* value is larger than the expected RTT_i and relatively close to RTT_{i-1}. Thus, MAARF detect an early deterioration of the link quality and therefore we reduce the current rate R_{i-1} for future communications.
  - If the value of RTT^* remains between the two theoretical bounds [RTT_i^+, RTT_i^-] (i.e. $RTT_i^+ < RTT^* < RTT_i^-$) then the rate will be kept and stay invariant R_i (MAARF assumes a steady state for subsequent network transmissions).
- Similar to the AARF algorithm, when the number of consecutive successful transmissions reaches the desired value (which can be at any given time 10, 20, 40 or 50), we switch to a higher throughput without consideration of the observed RTT^* values.

Analogically, when a transmission fails (no acknowledgment received within the RTO_i value) MAARF modifies values of the decision-making parameters (n, h and g) as follows:

- If a transmission error is occurred just after a rate increase, it will be then decremented. In addition, as shown in Equation 3 the number of successful transmissions n that should be attained for the next rise will be doubled with a limit value equal to n_{max}.

$$n = Min\left(2 * n\,;\, n_{\max}\right) \tag{3}$$

- If two consecutive errors are detected the MAARF mechanism reduces the current rate, while resetting the value of successful transmissions to the minimum one ($n = n_{min}$) for the next rising attempt.
- The same backoff control technique used for the parameter n adaptation is employed as well for the parameters h and g adjustment. In fact, these two variables will be dynamically adapted and will vary between the upper and lower limits to maintain a rigorous decision to increment/decrement the data rate.
  - When a transmission error occurs just after a rate increase decision caused by an interpretation of the RTT^* value, the current rate will be reduced and the h value (as shown in Equation 4) will be multiplied by two as the upper limit h_{max} is not reached.

$$h = Min\left(2 * h\,;\, h_{\max}\right) \tag{4}$$

  - In other words, during successful transmissions the condition ($RTT^* < RTT_i^+$) should verified using the new value of h to be able to increment the rate.

- Likewise, if a transmission error was detected immediately after a rate decrease decision based on comparisons between RTT^* and RTT_i^- values. Then the rate will be raised to its old value and the responsible g parameter (see Equation 5) will be doubled if its value does not reach g_{max}.

$$g = Min\left(2 * g\,;\, g_{\max}\right) \tag{5}$$

- This means that the condition $RTT^* > RTT_i^-$ should be established using the new value of g during subsequent transmissions to be able to decrease again the rate.
- Both of the above parameters will also be reset identically to the parameter n after two consecutive transmission failures as follows:

$$h = h_{min}\,;\, g = g_{min}$$

3.4 The MAARF setting

In this section we detail the new parameters designed for the MAARF algorithm. The IEEE 802.11 standard defines within its 802.11a and 802.11g versions, different physical rate values which can reach 54Mbps. Thus, we setup:

- R_i: the current data rate that varies from the following shown values {6, 9, 18, 12, 24, 36, 48, 54}Mbps.
- RTT^*: is the observed value when sending a frame (measured from the transmission channel after receiving the corresponding ACK).
- RTT_i: the theoretical time computed between the frame sending time to the ACK receipt instant. It reflects the channel occupation and does not include the waiting time to access the medium by the transmitter. It is given by Equation 6 as follows:

$$RTT_i = t_{em.Frame} + t_{propag} + t_{treat.Receiver} + SIFS + t_{em.ACK} + t_{propag} + t_{treat.Emitter} \tag{6}$$

- with, t_{em} is the emission time (Data Frame or ACK), t_{propag} is the propagation time over the transmission medium and t_{treat} is the treatment time of each received frame

In practice, this value will be represented only by the data frame transmission delay as shown in Equation 7. This approximation is made because of the negligibility of the other delays compared to the chosen value.

$$RTT_i \approx t_{em.Frame} = \frac{Frame.size}{R_i} \tag{7}$$

- RTO_i (Retransmission Time Out): is a recovery controlling timer after a frame loss. Its value is assigned based on RTT_i (see Equation 8).

$$RTO_i = 2 * RTT_i \tag{8}$$

- RTT_i^+ and RTT_i^- : the two decisional parameters (the RTT_i borders) which their values are chosen for each used rate R_i as defined in Equations 1 and 2.

- h: is the rate-increase responsible variable and it belongs to the interval $[h_{min}, h_{max}] = [4, 16]$.
- g: is the rate-decrement responsible variable and it belongs to the interval $[g_{min}, g_{max}] = [2, 8]$.
- n: is the already used parameter by the AARF technique. It represents the number of successive successful transmissions and belongs to the interval $[n_{min}, n_{max}] = [10, 50]$.

Finally, we illustrate the detailed MAARF functioning in Figure 7.

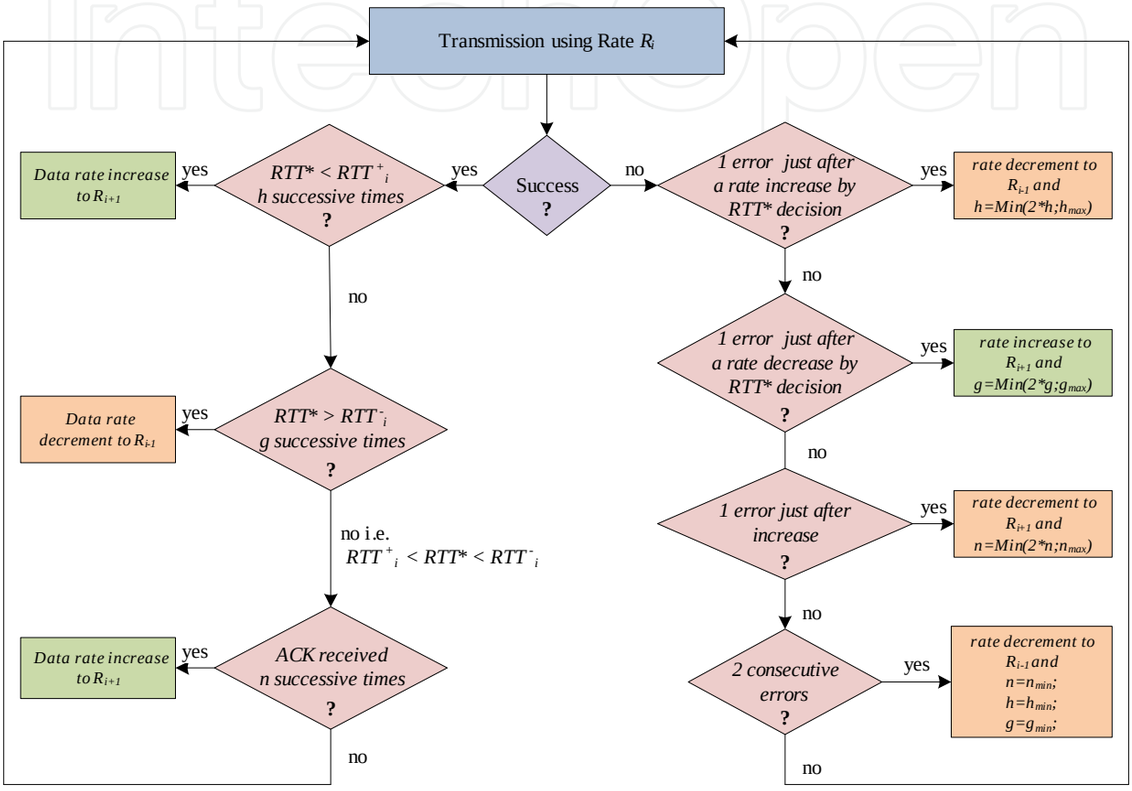


Fig. 7. The Transition diagram of the new MAARF algorithm

4. Results and performance evaluation

The algorithms were implemented using the C language on a Unix based operating-system environment (gcc/terminal MAC) to be then easily integrated into the network simulator.

We conducted various tests using the following configuration:

- The number of sent frames is 100 frames (approximately 0.5 seconds).
- The size of each data frame is equal to the 802.11g minimum frame size (=1200 bytes).
- An initial data rate of R_i is 6Mbps (up to 54Mbps).
- Failure of an ACK return reflects a transmission failure: packet loss, RTO expired or error detected by the CRC.
- A returned ACK by the receiver indicates a successful transmission only if it is received before the RTO expiration.
- The current value of RTT (RTT^*) is read/measured after each ACK reception.

Several scenarios have been considered to evaluate the performance of the proposed algorithm compared to the other versions (ARF and AARF).

4.1 Optimization of algorithm parameters

This first experiment is designed to study and optimize the decision-making parameters of the new algorithm: h (counting the number of successive times in which $RTT^* < RTT_i^+$) and g (reflecting the number of consecutive times that $RTT^* > RTT_i^-$). We discuss the values of h_{min} and g_{min}. In Figure 8, we show the implementation results of different MAARF algorithm configurations for various parameters values. These results express the chosen physical rate for each transmitted frame in the network.

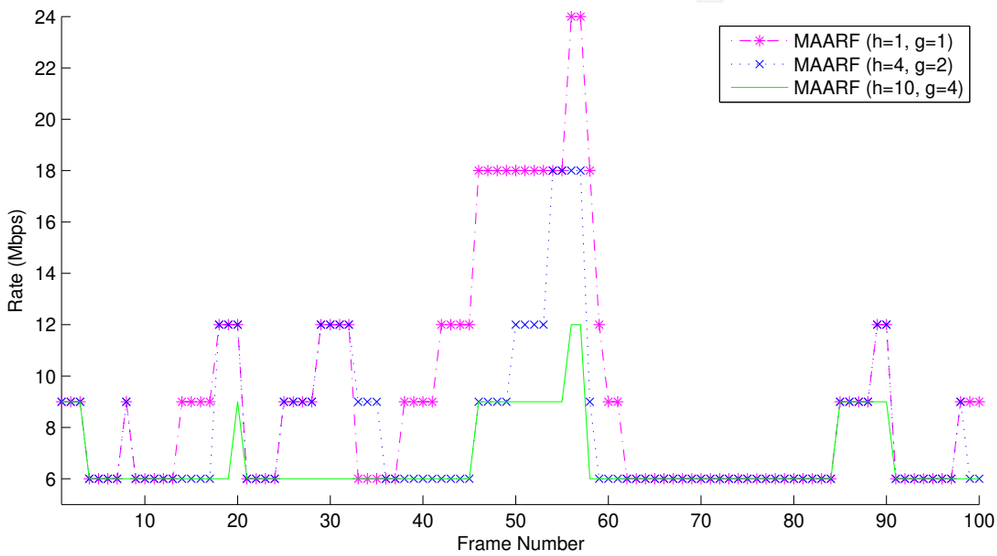


Fig. 8. Rate adaptation for different MAARF configurations

We note that by choosing low values of g and h (h_{min} = g_{min} = 1), MAARF makes quick decisions to increment and decrement the physical rate. In fact, it becomes sensitive for channel variations and adapts sinusoidal regime. On the other hand, by choosing large initial values of the g and h parameters (h_{min}=10 and g_{min}=4) the algorithm does not respond effectively to significant quality deviations and reacts as AARF. Therefore, we point out that the best initial and rigorous values of g and h with which MAARF gives the best results are respectively 4 and 2.

4.2 Test regimes

4.2.1 Unbalanced channel state

We compare now the new scheme against the AARF technique (currently used by the 802.11 WLANs) during unstable channel conditions (random improvement/degradation of the medium state). In Figure 9, we present the corresponding results graph and we clearly notice an efficient reaction of the MAARF technique against channel changes. In fact, the new algorithm detects faster the medium availability by adjusting its physical rate value starting from the 4th frame, while AARF reacts only from the 10th frame. We also note the

MAARF ability to dynamically respond against medium interferences dissimilar to the AARF mechanism. In addition, we conclude a significant improvement that has been reached (about 26%) regarding the mean value of recorded rates. In fact, we measure 6.1Mbps and 7.6Mbps respectively for the AARF and MAARF techniques.

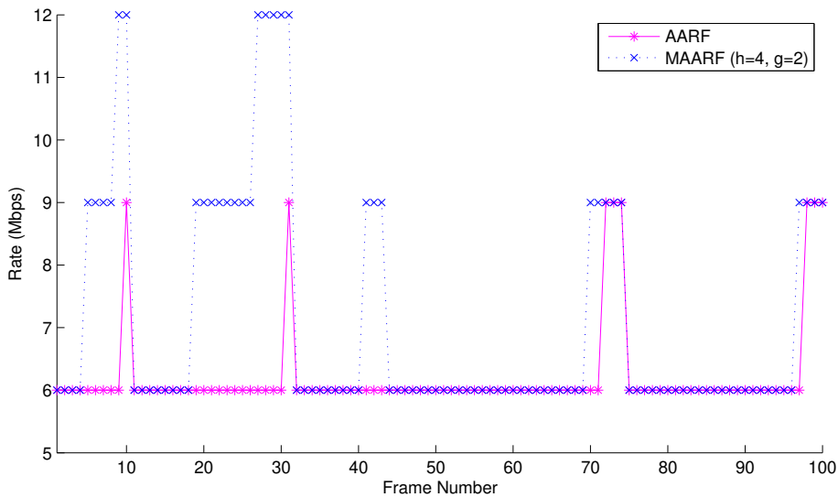


Fig. 9. Rate adaptation in transitory regime

4.2.2 Steady channel state

We assume in this case that only positive acknowledgments will be returned to the transmitter following the frame sending (packet transmissions without losses). Thus RTT^* values recorded from the medium will be close to those of the theoretical corresponding RTT_i values.

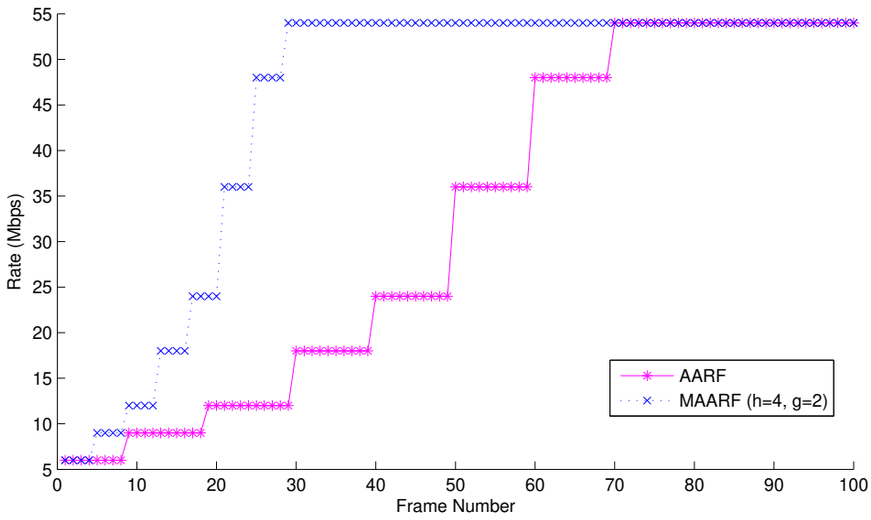


Fig. 10. Rate adaptation within steady regime

This first scenario shows a huge variation in terms of selected physical data rate and the overall mean value between the two techniques. In fact, a clear throughput enhancement is

obtained (41%) since we traced as a mean rate value during the simulation time, 32.04Mbps for the AARF and 45.48Mbps for the new algorithm.

According to the results in Figure 10, the maximum data rate (54Mbps) is reached earlier by the new algorithm as it detects the channel condition improvement (from frame No. 28) and thus takes advantage of the large possible rate values. However the AARF technique reaches the maximum rate later on (only from the frame No. 70). This was caused by the fact that AARF is required to wait at least 10 positive acknowledgments at each rate hop.

4.2.3 Mobile environment situation

A fourth simulation on the rate adaptation is conducted within a variable channel regime. In the case of 802.11 WLAN the medium quality variations are very fast and totally random. This is reflected by intervals where the channel conditions improve rapidly, separated by those where the medium state deteriorates suddenly. We note from Figure 11 that the new technique adapts the same rate as obtained by AARF; however MAARF is more agile and predictive of the medium communication conditions for the data rate rise decision. When transmission errors take place, both methods pass at a lower rate almost at the same time.

The average rate value obtained for both AARF and MAARF mechanisms is equal, respectively, to 6.72Mbps and 7.89Mbps. As a result, an improvement of 17% was reached due to the responsive capability and the fast adaptability of the new link control mechanism.

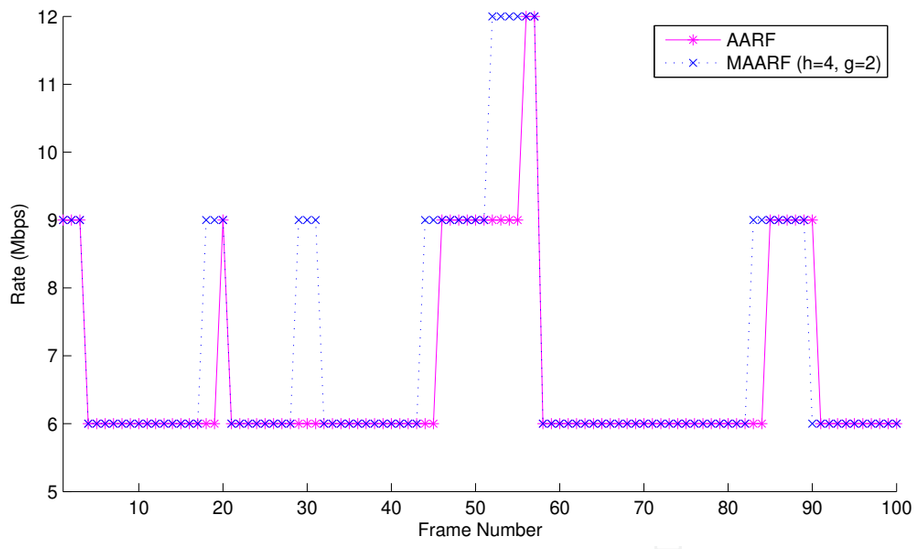


Fig. 11. Rate adaptation for instantaneous and unpredictable channel conditions

4.3 Network simulations under NS-2 platform

The results obtained during the new MAARF algorithm implementation have shown that it is possible:

- To estimate the channel conditions through the observed *RTT** values.
- To detect/avoid packet losses before they happen.
- To take the necessary decisions faster than current mechanisms.

We have revealed in this study that it is no longer necessary to wait 10 or more consecutive ACKs to adjust the theoretical rate as it was deployed by classical algorithms. We compare in Figures 12 and 13 the results obtained by applying the new MAARF mechanism and the current AARF for the same channel conditions. These results are reflecting, respectively, the computed rate mean value and the number of frames errors depending on the transmitted packets number.

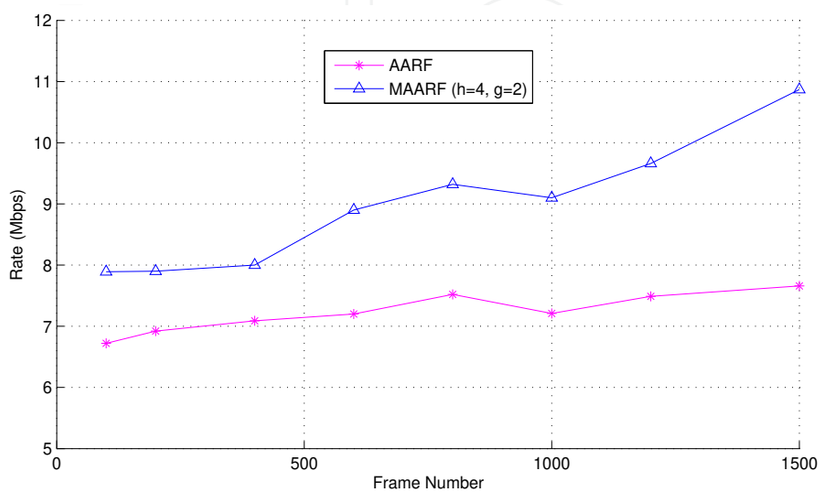


Fig. 12. Observed throughput for both AARF and MAARF mechanisms

Parameter	Value
WLAN version	802.11b
Radio propagation model	Two Ray Ground
Transmission range	250 meters
Number of Mobile Stations	2
Available physical rates	1Mps to 11Mbps
Routing protocol	Ad-hoc On demand Distance Vector
Slot Time	16 µs
SIFS Time	8 µs
DIFS Time	40 µs
Packet size	1000 Bytes
Traffic	CBR / UDP
Simulation time	80s (starting t=10s)
Simulation grid	745x745 meters

Table 1. NS-2 simulation parameters

We show based on the conducted experimentations, the improvements are distinguished and very clear in terms of the overall throughput and packet errors. We also confirm the MAARF algorithm performance by simulating the new technique on the Network Simulator NS-2 platform (Network Simulator-II, 1998). Figure 14 outlines various simulation arrangements performed on NS-2. We easily confirm the initial results by varying the bit error rate. The simulation parameters are summarized in Table 1.

First we note that the chosen traffic for the carried simulations was Constant Bit Rate (CBR) over the Transport-layer User Datagram Protocol (UDP). In fact, the CBR service category is

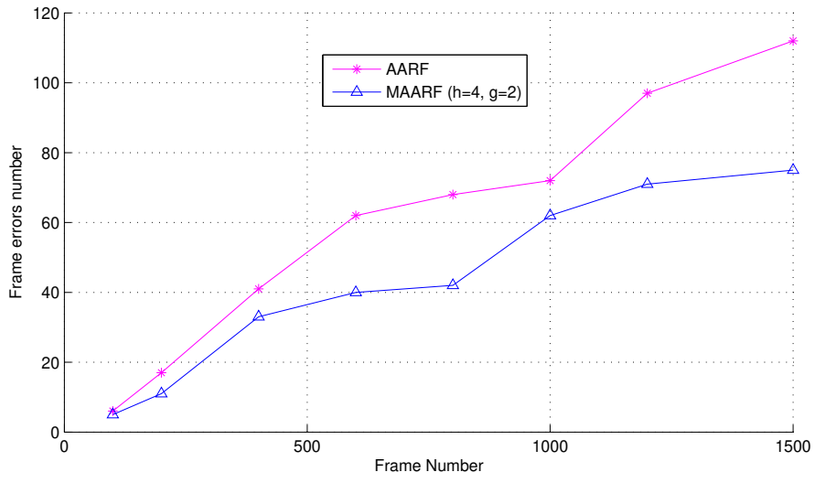
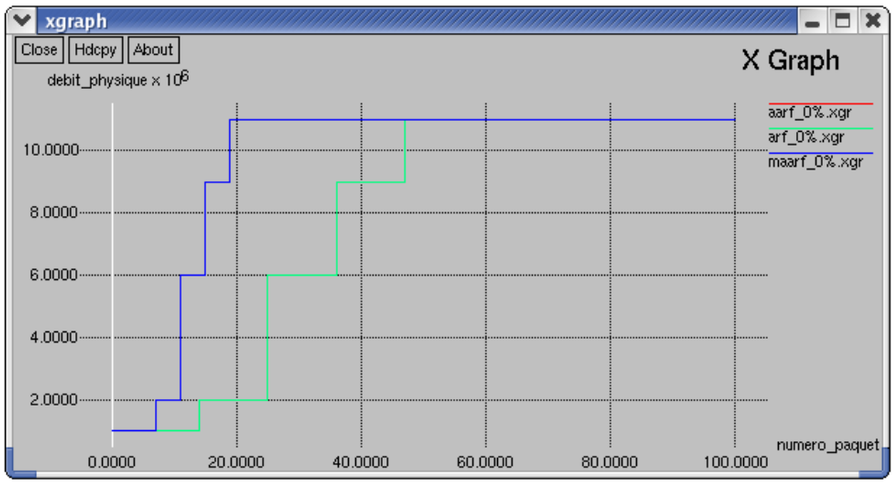


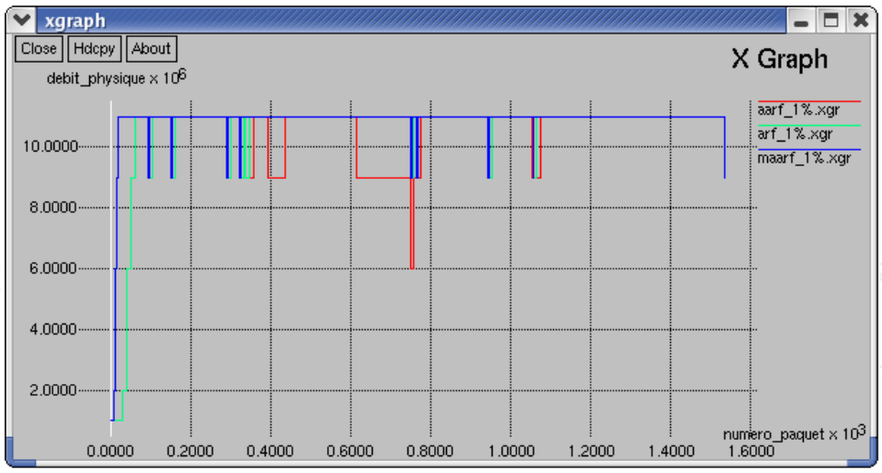
Fig. 13. Number of frame errors compared to the number of transmitted frames

used for connections that transport traffic at a constant bit rate, where there is an inherent reliance on time synchronization between the traffic source and destination. CBR is tailored for any type of data for which the end-systems require predictable response time and a static amount of bandwidth continuously available for the life-time of the connection. These applications include services such as video conferencing, telephony (voice services) or any type of on-demand service, such as interactive voice and audio.

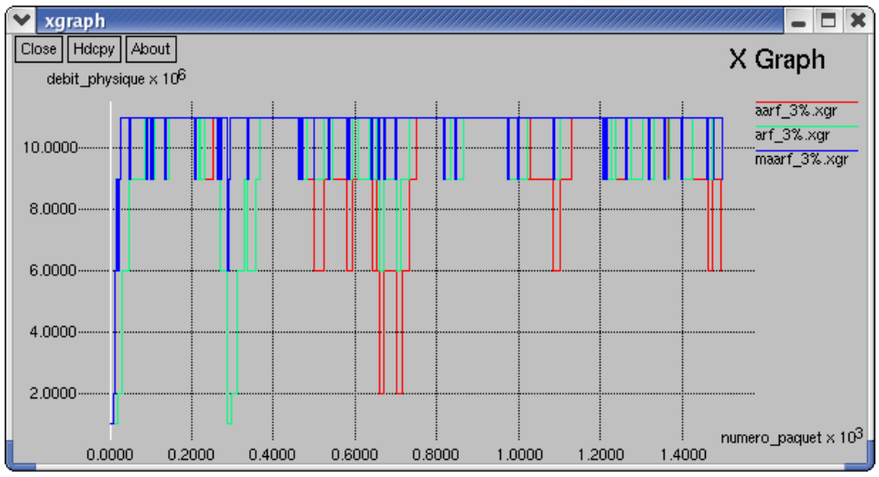
The obtained results verify the initial theoretical observations and validate the efficiency and adaptability of the new mechanism for both slow and rapid fluctuations of the transmission channel quality. In absence of errors (as shown in Figure 14.a.), MAARF reacts quickly and rises the higher allowed rate before the current techniques. While varying the Bit Error Rate (BER) value during simulation scenarios (Figures 14.b., 14.c. and 14.d.) the physical rate adjustment corresponding to MAARF is more suitable and faster than other tested algorithms. This outcome is clearly confirmed in Figures 14.e. and 14.f. by measuring the achieved throughput for two different BER values. Accordingly, we note an enhanced response from MAARF against the channel quality variations compared to the other two techniques.



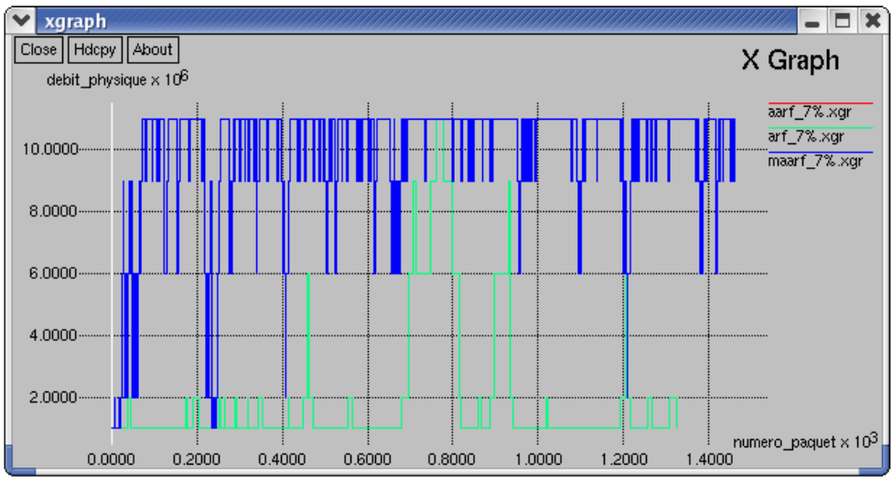
a. Physical rate tuning for ideal channel



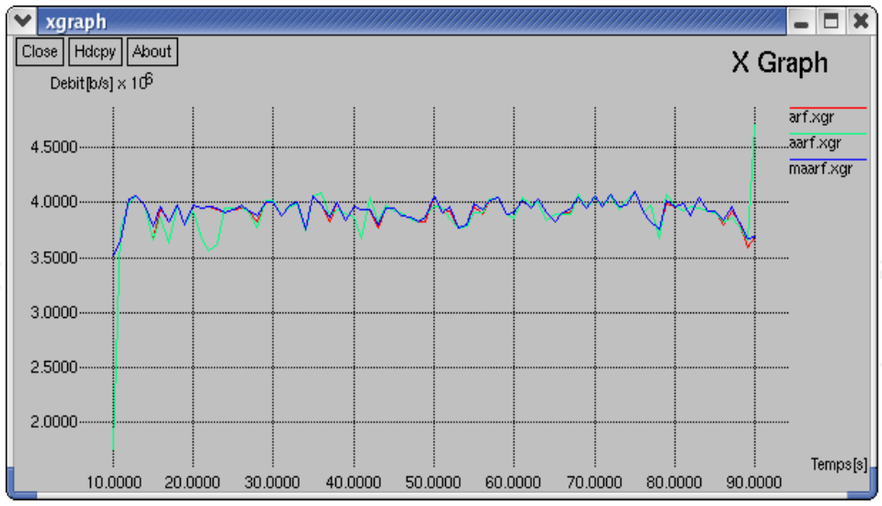
b. Physical rate tuning with BER=1%



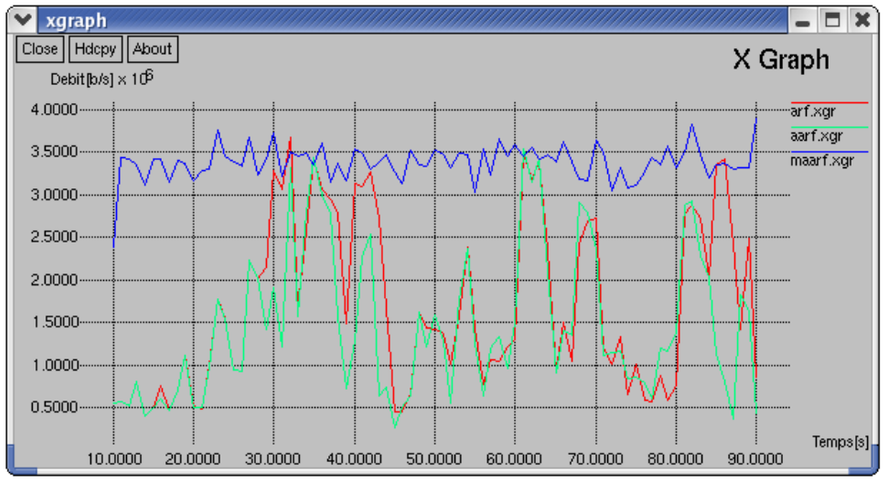
c. Physical rate tuning with BER=3%



d. Physical rate tuning with BER=7%



e. Measured throughput with BER=1%



f. Measured throughput with BER=5%

Fig. 14. Experimental NS-2 simulation results by varying the BER value

5. Conclusions

5.1 General remarks

The IEEE 802.11 standard defines several MAC-level parameters (settings) that are available for tuning at the side of the Wireless Network Interface Card (NIC). The most important parameter available for adjustment is the transmit rate. Each rate corresponds to a different modulation scheme with its own trade-off between data throughput and distance between the MSs. In order to decide which rate is optimal at each specific moment, a control algorithm needs information about the current link conditions. Since it is difficult to get these information directly, most of the MAC rate-control algorithms use statistics-based feedbacks, for example, ACK count. We have shown, through a deep study of the currently-used rate control mechanisms, that the main disadvantage of this indirect feedback is that it

is inherently slow, causing communication failures when the link conditions degrade rapidly (e.g., when the user moves fast). The short-term dropouts are normally handled by frame retransmissions. This is acceptable for download applications whose key requirement is a flagrant data throughput. However it leads to a significant increase in (average) packet delay and in the jitter due to the variations in the number of retransmissions. Streaming applications are very sensitive to long packet delays and high jitter, and less sensitive to the overall throughput (of course when this throughput is larger than the minimum value required by the application). Consequently, streaming applications achieve poor performances under a standard rate control (like ARF and AARF techniques).

Hence, we have proposed a new algorithm noted Modified Auto Rate Fallback (MAARF). This technique implements a new decisional variable called Round Trip Time (RTT) which complies and cooperates with the basic parameter (number of returned ACKs). This new parameter is designed to make a good estimate of the instantaneous channel quality (observation of the channel state after each transmitted frame), and choose the adequate rate accordingly.

Based on the simulation results we have shown a remarkable improvement in data throughput and physical rate control. In fact, the proposed MAARF mechanism provides higher values (about 17% to 58%) in comparison with those resulting from conventional algorithms. Table 2 presents an observed throughput summary of the MAARF scheme compared to existing algorithms (ARF and AARF) and gives an overview on the rate control enhancement for different BERs. The overall throughput observed within MAARF is much higher than other mechanisms when the BER reaches high values (12 times when the error rate is 10%).

In conclusion simulation experiments were performed on the new dynamic time-based link adaptation mechanism and the corresponding results have shown the quality improvement on the transmission link. The results also demonstrated that the proposed mechanism outperforms the basic solution in terms of providing support to both acknowledgment-based and time-based rate control decisions. Therefore MAARF meets the desired objectives by being able to reduce errors resulting from bad rate adjustment and then satisfy the transmission of multimedia applications in terms of required QoS.

Observed throughput in 10^6 Mbps				***Enhancement in %***
BER	***ARF***	***AARF***	***MAARF***	***MAARF / AARF***
0%	*4.0335*	*4.0339*	*4.0461*	*0.3%*
1%	*3.8930*	*3.8877*	*3.9225*	*0.8%*
3%	*3.5419*	*3.3612*	*3.6719*	*9%*
5%	*1.6992*	*1.4564*	*3.3826*	*>80%*
7%	*0.4914*	*0.4914*	*3.0195*	*>80%*
10%	*0.0774*	*0.7740*	*1.8273*	*>80%*

Table 2. Enhancement ratio of the MAARF technique

5.2 The MAARF working out performance for real-time video streaming

In the context of the Voice-over-IP and real-time video streaming (Video conferencing) which represent the most end-user demanded multimedia streaming applications to be supported over wireless connections, the proposed technique is supposed to enhance the video quality transmission by keeping the same compression degree, and so, avoiding the

cross layer design implemented by (Haratcherev et al., 2005) which involves an interaction between the application and MAC layers by assuming that the video encoder can also adapt to the link quality by changing the compression degree for example, and thus modifying the data rate. This so-called Coupled scheme is based on a cross-layer signaling by letting the rate control loops - of the MAC-layer and the Video coder - be mutually associated. In such way, the video coder will poll frequently for the current and predicted rate. For performance evaluation purpose, the same real experiment as performed in (Haratcherev et al., 2005) was carried out by streaming a video file between two laptops, both running Linux. The 802.11a cards used are based on the Atheros AR5000 chipset, and the card driver implements the discussed rate control algorithms. One of the laptops had a fixed position and the other one is following a predetermined track. The track consists of three parts: reaching from the room to a specific start position in the hallway, and waiting until certain time elapses (10s). Then keep moving with the laptop three times up and down the hallway (60s). Finally we back again into the room, where the fixed laptop lies (20s). We have evaluated four cases: ARF, AARF, MAARF and a Coupled version of the hybrid rate control (responsive MAC-adaptation using radio-SNR and packet loss statistics) as described in (Haratcherev et al., 2004). Each experiment took 90 seconds and we have compared the quality of the received videos using the Peek-Signal-to-Noise-Ratio (PSNR), a commonly used metric in video compression.

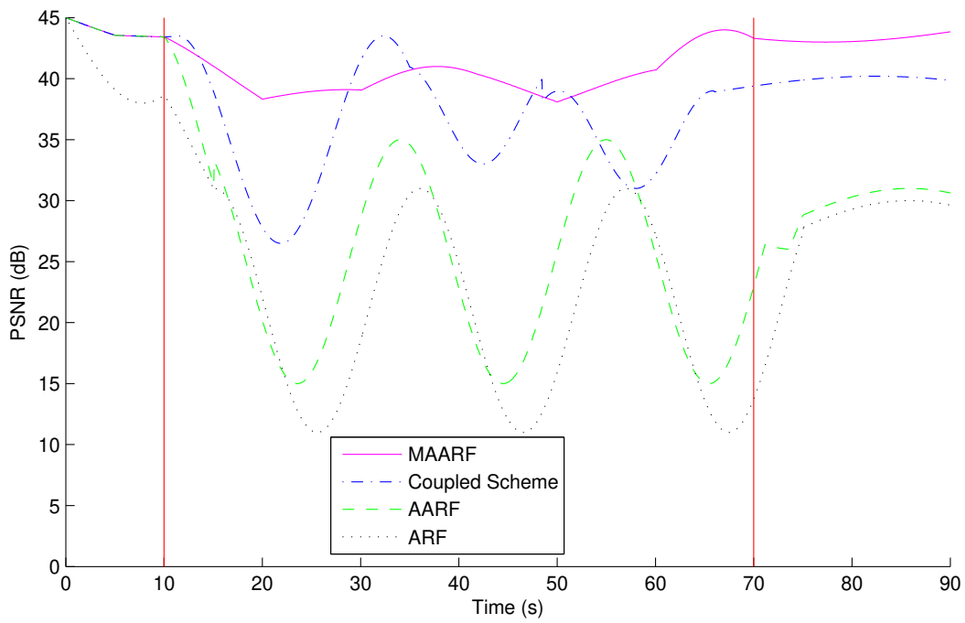


Fig. 15. PSNR measurements during the 90s real-time video streaming experiments

Algorithm	# Skipped Frames	# Lost Packets	PSNR	PSNR Middle
MAARF	*63*	*15*	*41.41*	*40.4*
Coupled Scheme	*50*	*13*	*37.9*	*36.23*
AARF	*182*	*16*	*28.19*	*25.32*
ARF	*205*	*21*	*24.95*	*21.74*

Table 3. Summary of algorithms results

In Figure 15 the quality (PSNR) is shown for the whole experiment (90s). In the left part (0–10s) the channel conditions are excellent since a high quality was fulfilled with all cases. The middle part is best described as having continuous variation. The right part has good channel conditions again. As we can notice, the MAARF case has a higher PSNR in almost all cases. Table 3 summarizes the number of skipped frames (by the encoder), the number of lost packets, the average PSNR and the average PSNR in the unstable period (between 10 and 70s). Although the lost packets counts are not the same in the four cases, it does not justify the difference in PSNR values. In the ARF and AARF cases, the total number of skipped frames is much higher, as we expected. Looking at the average PSNR in the 10–70s period, we conclude that MAARF outperforms the other techniques based on an enhanced channel assessment. As a result, the proposed video streaming experiments better illustrate the improvement made using MAARF by reducing the number of lost packets and skipped frames. Both effects resulted in a high visual quality opposed to the cases of ACK-based rate control algorithms. Meanwhile the Coupled cross-layer signaling scheme had also very encouraging results in terms of packets loss and skipped frames which let us consider a warning concept between the video encoder and the MAARF decision in the MAC layer. Therefore, a second version of the MAARF technique including a cross-layer signaling solution will be investigated in the next research step. This extended adaptation would be able to further avoid packet losses and especially to prevent skipped frames of the emerging voice/video streaming services. Of course, it would have an adverse effect on the visual quality of the decoded video however we will need to further carry out real streaming video experiments over a wireless 802.11 link between MSs to conclude if the cross-layer signaling between the MAC-layer and the video coder will lead to a visual quality increase in term of measured PSNR.

6. References

Angrisani, l.; Napolitano, A. & Sona, A. (2011). VoIP over WLAN: What about the Presence of Radio Interference?, *VoIP Technologies, Shigeru Kashihara (Ed.)*, ISBN: 978-953-307-549-5, InTech, pp. 197-218, Feb. 2011

An-Chih, L.; Ting-Yu, L. & Ching-Yi, T. (2009). ARC: Joint Adaptation of Link Rate and Contention Window for IEEE 802.11 Multi-rate Wireless Networks, *The 6th Annual IEEE Communications Society Conference on Sensor, Mesh and Ad Hoc Communications and Networks SECON '09*, Rome, Italy, July 2009.

Chen, X.; Wan, Y. & Lu, J. (2010). Cross-layer link rate adaptation for high performance multimedia broadcast over WLANs, *in Proc. IEEE GLOBECOM Workshops GC-Workshops'10*, pp. 965-969, Miami, FL, USA, Dec. 2010

Chevillat, P.; Jelitto, J.; Barreto, A. N. & Truong, H. (2003). A Dynamic Link Adaptation Algorithm for IEEE 802.11a Wireless LANs, *in Proc. IEEE International Conference on Communication ICC'03*, Anchorage, AK, May 2003

Chiapin W. & Tsungnan L. (2008). A Cross-Layer Link Adaptation Algorithm for IEEE 802.11 WLAN with Multiple Nodes, *in Proc. IEEE Asia-Pacific Services Computing Conference APSCC '08*, pp. 1161-1167, Yilan, Taiwan, Dec. 2008

del Prado Pavon, J. & Choi, S. (2003). Link Adaptation Strategy for IEEE 802.11 WLAN via Received Signal Strength Measurement, *in Proc. IEEE International Conference on Communication ICC'03*, Anchorage, AK, May 2003

Galtier, J. (2011). Adaptative Rate Issues in the WLAN Environment, *Advances in Vehicular Networking Technologies, Miguel Almeida (Ed.)*, ISBN: 978-953-307-241-8, InTech, pp. 187-201, April 2011

Habetha, J. & de No, D. C. (2000). New Adaptive Modulation and Power Control Algorithms for HIPERLAN/2 Multihop Ad Hoc Networks, *in Proc. European Wireless (EW'2000)*, Dresden, Germany, Sept. 2000

Haratcherev, I.; Langendoen, K.; Lagendijk, I. & Sips, H. (2004). Hybrid Rate Control for IEEE 802.11. *In ACM International Workshop on Mobility Management and Wireless Access Protocols MobiWac'04*, pp. 10–18, Philadelphia, USA, Oct. 2004

Haratcherev, I.; Taal, J.; Langendoen, K.; Lagendijk, I. & Sips, H. (2005). Fast 802.11 link adaptation for real-time video streaming by cross-layer signaling, *In Proc. IEEE International Symposium on Circuits and Systems ISCAS'05*, pp. 3523-3526, Vol. 4, Kobe, May 2005

IEEE Standard 802.11 (1999), Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, *IEEE Standard 802.11*, 1999

IEEE Standard 802.11a (1999), Supplement to Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: High-speed Physical Layer in the 5 GHz Band, *IEEE Std. 802.11a-1999*, Sept. 1999

IEEE Standard 802.11b (1999), Supplement to Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: Higher-speed Physical Layer Extension in the 2.4 GHz Band, *IEEE Std. 802.11b-1999*, 1999

IEEE Standard 802.11g (2003), Supplement to Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: Further Higher Data Rate Extension in the 2.4 GHz Band, *IEEE Std. 802.11g-2003*, June 2003

Jianhua, H.; Kaleshi, D.; Munro, A. & McGeehan, J. (2006). Modeling Link Adaptation Algorithm for IEEE 802.11 Wireless LAN Networks, *The 3rd International Symposium on Wireless Communication Systems ISWCS '06*, pp. 500-504, Sep. 2006

Junwhan, K. & Jaedoo, H. (2006). Link Adaptation Strategy on Transmission Rate and Power Control in IEEE 802.11 WLANs, *in Proc. IEEE Vehicular Technology Conference VTC'06*, Montreal, QC, Canada, Sept. 2006

Kamerman, A. & Monteban, L. (1997). WaveLAN-II: A High-Performance Wireless LAN for the Unlicensed Band, *Bell Labs Technical Journal*, pp. 118–133, Summer 1997

Krishnan, M. N. & Zakhor, A. (2010). Throughput Improvement in 802.11 WLANs Using Collision Probability Estimates in Link Adaptation, *in Proc. IEEE Wireless Communications and Networking Conference WCNC'10*, Sydney, Australia, April 2010

Kumar, A. & Holtzman, J. (1998). Performance analysis of versions of TCP in WLAN, *Indian Academy of Sciences Proceedings in Engineering Sciences*, Sadhana, Feb. 1998

Lacage, M.; Manshaei, M. H. & Turletti, T. (2004). IEEE802.11 Rate Adaptation: A Practical Approach. *INRIA Research Report*, number 5208, May 2004

Mocanu, S. (2004). Performance Evaluation of TCP Reno and Vegas. *Technical report* CR256, Laboratoire d'Automatique de Grenoble, Département Télécom, July 2004

Network Simulator II (1998), ns-2, available from http://www.isi.edu/nsnam/ns/

Pang, Q.; Leung, V.C.M. & Liew, S.C. (2005). A rate adaptation algorithm for IEEE 802.11 WLANs based on MAC-layer loss differentiation, *in Proc. International Conference on Broadband Networks BroadNets'05*, pp. 659-667, Vol. 1, Boston, MA, USA, Oct. 2005

Qiao, D.; Choi, S. & Shin, K. G. (2002). Goodput Analysis and Link Adaptation for IEEE 802.11a Wireless LANs, *IEEE Transactions on Mobile Computing*, vol. 1, pp. 278–292, Oct. 2002

Qiao, D. & Choi, S. (2005). Fast-Responsive Link Adaptation for IEEE 802.11 WLANs, *in Proc. IEEE International Conference on Communication ICC'05*, Sept. 2005

Rebai, A. R.; Alnuweiri, H.; Hanafi, S. (2009). A novel prevent-scan Handoff technique for IEEE 802.11 WLANs, *in Proc. International Conference on Ultra Modern Telecommunications & Workshops ICUMT '09*, St. Petersburg, Russia, Oct. 2009

Rebai, A. R.; Fliss, M.; Jarboui, S. & Hanafi, S. (2008). A new Link Adaptation Scheme for IEEE 802.11 WLANs, *in Proc. IEEE New Technologies, Mobility and Security NTMS'08*, Nov. 2008

Rebai, A. R.; Haddar, B.; Hanafi, S. (2009). Prevent-scan: A novel MAC layer scheme for the IEEE 802.11 handoff, *In Proc. International Conference on Multimedia Computing and Systems ICMCS '09*, pp. 541-546, Ouarzazate, Morocco, April 2009

Rebai, A. R.; Hanafi, S.; Alnuweiri, H. (2009). A new inter-node priority access enhancement scheme for IEEE_802.11 WLANs, *in Proc. International Conference on Intelligent Transport Systems Telecommunications ITST'09*, pp. 520-525, Lille, France, Oct. 2009

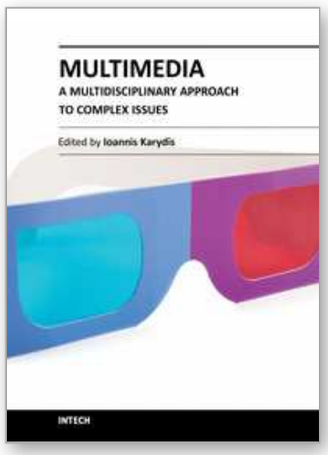
Rebai, A. R.; Rebai, M. F.; Alnuweiri, H.; Hanafi, S. (2010). An enhanced heuristic technique for AP selection in 802.11 handoff procedure, *In Proc. IEEE International Conference on Telecommunications ICT'10*, pp. 576-580, Doha, Qatar, April 2010

Sangman, M.; Moonsoo, K. & Ilyong, C. (2011). *Link Quality Aware Robust Routing for Mobile Multihop Ad Hoc Networks, Mobile Ad-Hoc Networks: Protocol Design, Xin Wang (Ed.)*, ISBN: 978-953-307-402-3, InTech, pp. 201-216, Jan. 2011

Shun-Te, W.; Wu, J.-L.C.; Chung-Ching, D. & Chun-Yen, H. (2007). An adaptation scheme for quick response to fluctuations in IEEE 802.11 link quality, *in Proc. IEEE Region 10 Conference TENCON'07*, Taipei, Oct. 2007

Tourrilhes, J. (2001). Fragment Adaptive Reduction: Coping with Various Interferers in Radio Unlicensed Bands", *in Proc. IEEE International Conference on Communication ICC'01*, Helsinki, Finland, July 2001

Xin, A.; Shengming, J. & Lin T. (2010). Traffic-aware active link rate adaptation via power control for multi-hop multi-rate 802.11 networks, *in Proc. IEEE International Conference on Communication Technology ICCT'10*, pp. 1255-1259, Nanjing, Nov. 2010



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The nowadays ubiquitous and effortless digital data capture and processing capabilities offered by the majority of devices, lead to an unprecedented penetration of multimedia content in our everyday life. To make the most of this phenomenon, the rapidly increasing volume and usage of digitised content requires constant re-evaluation and adaptation of multimedia methodologies, in order to meet the relentless change of requirements from both the user and system perspectives. Advances in Multimedia provides readers with an overview of the ever-growing field of multimedia by bringing together various research studies and surveys from different subfields that point out such important aspects. Some of the main topics that this book deals with include: multimedia management in peer-to-peer structures & wireless networks, security characteristics in multimedia, semantic gap bridging for multimedia content and novel multimedia applications.

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