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Abstract: By nature, IEEE 802.11-based mobile ad hoc networks do not dispose of any reliable mechanism for quality of service. Therefore, research in this field has received much attention these last years. However, the estimation of the available resources still represents one of the main issues when designing a QoS solution. In this paper, we propose an improved mechanism to estimate the available bandwidth in IEEE 802.11-based ad hoc networks. Through simulations, we compare the accuracy of the proposed estimation to the estimations performed by some state-of-the-art QoS protocols, like for instance, BRuIT, AAC and QoS-AODV.

Key-words: Available bandwidth estimation, *Ad hoc* networks, IEEE 802.11, Quality of service

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Estimation de la bande passante résiduelle dans les réseaux ad hoc basés sur IEEE 802.11

Résumé : De manière native, les réseaux ad hoc ne fournissent pas de mécanismes afin de garantir une certaine qualité de service. Par conséquent, les recherches dans ce domaine ont suscité un intérêt particulier ces dernières années. Cependant, l'estimation de la bande passante résiduelle reste toujours le point clé lorsqu'on désire mettre en place des solutions de qualité de service. Dans ce rapport, nous proposons une technique permettant d'estimer la bande passante résiduelle dans des réseaux ad hoc basés sur la norme IEEE 802.11. A travers les simulations, nous comparons notre solution avec d'autres techniques d'estimation comme BRuIT, AAC et QoS-AODV.

Mots-clés : Bande passante résiduelle, Réseaux *ad hoc*, IEEE 802.11, Qualité de service

Contents

1 Introduction

Ad hoc networks are autonomous, self-organized wireless and mobile networks. They do not require the presence of any fixed infrastructure such as access points. The nodes address themselves topology changes due to the mobility and to the arrival and departure of nodes from the network. Many of the current works in ad hoc networking assume that the underlying wireless technology is the IEEE 802.11 standard due to the availability of materials and simulation models. Nevertheless, this standard has not been designed for multi-hop ad hoc operation and it is therefore not perfectly suited to this type of networks. It provides an ad hoc mode allowing mobiles to communicate directly, but as the communication range is limited, a distributed routing protocol is required to allow long distance communications.

Today, several applications generate multimedia data or rely on the proper transmission of sensitive control traffic. These applications may benefit from a quality of service (QoS) support. Therefore, this field has been extensively studied and more and more QoS solutions are proposed for ad hoc networks. The term QoS is vague and gathers several concepts. Some protocols intend on offering strong guarantees to the applications regarding given parameters, for instance bandwidth, delay, packet loss, or network load. Some others, which seem more suited to a mobile environment, only intend on selecting the best route among all possible regarding similar criteria. In both cases, an accurate evaluation of the capabilities of the routes is required. Most of the current QoS proposals leave this problem aside by relying on the assumption that layer-2 protocols are able to perform this evaluation, but they are not. The resource evaluation problem is far from being trivial, as it must consider several phenomena related the wireless environment, but also to less measurable parameters such as the mobility of the nodes.

In this paper, we will focus on one of the fundamental resources, the bandwidth. Estimating the remaining bandwidth at a given time and in a given part of the network is tricky, as the medium is shared between close nodes in a wireless network. This implies that computation of the available bandwidth between two neighbor nodes requires identification of all the emitter's potential contenders and of all the receiver's potential jammers. These nodes' utilization of the shared resource should then be gathered and should be composed to derive the amount of free resources. This first means that a precise identification of all interfering nodes is required. Secondly, information on their shared bandwidth usage has to be gathered. Finally, the joint impact of this sum of traffics should be evaluated. These tasks are usually hard to realize and they get even harder in sparse networks, as two nodes may interact without being able to exchange information.

In this paper, we present a new method to evaluate the available bandwidth in ad hoc networks. It uses the nodes carrier sense capability combined to some other mechanisms like collision prediction. It provides an evaluation that represents an acceptable compromise between accuracy and measurement cost. This evaluation is designed for the IEEE 802.11 MAC-Layer but may easily be adapted to similar random medium access protocols.

Hereafter, we define the **available bandwidth** between two neighbor nodes as *the maximum throughput that can be transmitted between these two peers without disrupting any ongoing flow in the network.* This term should not be confused with the **link capacity** (also called **base bandwidth**) that designates *the maximum throughput a flow can achieve between two neighbor nodes*, even at the cost of other flows' level of service degradation. For performance evaluation, our proposed has been integrated into a simple reactive QoS protocol called ABE (Available Bandwidth Estimation).

As much literature on this topic is now available, we will consider in the remaining of this article that the IEEE 802.11 protocol is known by the reader. The rest of the paper is organized as follows: Section 2 presents related works. Section 3 introduces the available bandwidth estimation mechanism we propose, first from a single node's point of view and then from a link's perspective. Section 4 describes the basic protocol design and finally, NS-2 simulation results are presented in Section 5.

2 Related work

Available bandwidth evaluation has generated several contributions in the wired and wireless networks communities. They can be classified into two major categories:

- We call *active approaches* the techniques that rely on the emission of dedicated end-to-end probe packets to estimate the available bandwidth along a path.
- We call *passive approaches* the techniques that use only local information on the utilization of the bandwidth. The typical example of such approaches is a node monitoring the channel usage by sensing the radio medium. These mechanisms may exchange information via one-hop broadcasts. Usually, however, this exchange can be piggy-backed in the *Hello* messages used by many routing protocols to discover the local topology. As long as these exchanges are not too bandwidth consuming, we will call such a technique passive.

2.1 Active bandwidth estimation techniques

A detailed survey of the different techniques used in wired networks is proposed in [10]. Along with these strategies, the authors propose Self-Loading Periodic Streams (SLoPS). This technique measures the end-to-end available bandwidth by sending packets of equal size and measuring the one-way delays of these probing packets. The source then increases the probe packets emission rate linearly. As soon as a variation in this delay is noticed, the path is considered as saturated. The last measurement point before the variation indicates the path capacity which is assimilated to the available bandwidth. The Trains of Packet Pairs (TOPP [9]) uses a similar mechanism with a binary search instead of the linear increase.

Based on the TOPP method, the authors of DietTOPP [1] have evaluated the accuracy of this type of techniques in wireless networks. This article shows that the probe packets size as well as the volume of cross-traffic have a stronger impact on the measured bandwidth than in wired networks. Aside from the fact that this only measures the path capacity rather than the available bandwidth, this indicates that these techniques lead to inaccurate results in wireless networks.

The authors of [5] use the fact that when a probe packet's transmission delay gets higher than the theoretical maximum delay, the medium suffers from congestion. They propose a method to compute the medium utilization from these delays and then derive the available bandwidth from this channel use.

All the active techniques cited above present two major drawbacks. First, when many nodes in an ad hoc network need to perform such an evaluation for several destinations, the number of probe packets introduced in the network can be important and interact with the traffic and with other probes. Secondly, a mobile network can contain links of heterogeneous quality. An end-to-end evaluation technique may not be as reactive as a local technique when it comes to local reconstruction of routes. Any local technique should, however, be complemented with an appropriate measurements combination technique.

2.2 Passive bandwidth estimation techniques

When a node wishes to estimate the bandwidth available in its vicinity, the intuitive approach consists in monitoring the channel over a given time period and deducing from this observation the utilization ratio of the shared resource. [7] uses this technique and adds a smoothing factor to hide transient effects. The QoS routing protocol designed in this article is based on a simple estimation of the available bandwidth by each node and does not consider any interfering nodes.

QoS-AODV [12] also performs such per-node available bandwidth estimation. The evaluation mechanism constantly updates a value called BWER (Bandwidth Efficiency Ratio), which is the ratio between the number of transmitted and received packets. The available bandwith is simply obtained by multiplying the BWER value by the channel capacity. This figure is propagated in the one-hop neighborhood of each node through a local broadcast in Hello messages. The bandwidth available to a node is then inferred from these values as the minimum of the available bandwidths over a closed single-hop neighborhood.

In [3], Chaudet and Guérin Lassous have proposed a bandwidth reservation protocol called Bandwidth Reservation under InTerferences influence (BRuIT). This protocol's available bandwidth estimation mechanism takes into account the fact that, with the IEEE 802.11 standard, the carrier sense area is larger than the transmission area. In other words, emitters share the bandwidth with other nodes with whom they cannot communicate. Experimental studies have shown that this carrier sense radius is at least twice the communication radius. To address this issue, each node regularly broadcasts to all its immediate neighbors information about the total bandwidth it uses to route and emit flows (deduced from applications and routing information) and its estimated available bandwidth. It also transmits similar information from all its one-hop neighbors, resulting in a propagation of this information at one node's two-hops distance. Each node gathers the two-hops neighborhood knowledge to perform admission control. When the carrier sense radius is equal to twice the communication radius, the authors have shown that two hops communication represents the best compromise for estimation accuracy [4].

Making the same observation, Yaling and Kravets have proposed in [14] the Contention Aware Admission Control Protocol (CACP). Each node first computes the local idle channel time fraction monitoring the radio medium. Then, the authors propose three different techniques to propagate this information to the greatest number of nodes within the carrier sense area. First, like in BRuIT, they propose to include the information in Hello messages to reach the two-hop neighborhood. Second, they propose to increase the nodes' transmission power. Finally, receiving nodes can also reduce their sensitivity in order to decode farther information. The authors, similarly to [6], also notice the existence of intra-flow contention. In other words, when a flow takes a multi-hop route, successive routers contend for resources in order to route frames belonging to the same flow. It is thus important to take into account at least the route length when performing admission control. Ideally the exact interactions between nodes along a path should be identified and considered.

Finally, the AAC protocol, proposed in [11], makes each node consider the set of potential contenders as a single node. It measures the activity period durations and considers that any such period can be seen as a frame emission of the corresponding length. With this mechanism, collisions and distant emissions are also considered when computing the medium occupancy. Based on this measurement, each node is able to evaluate its available bandwidth. It exchanges this information with its neighbors to compute the bandwidth on each link, i.e. for each pair of nodes. This value is defined as the minimum between the available bandwidths of both ends. AAC also takes into account the intra-flow contention problem mentioned before.

2.3 Motivation

The active techniques presented above do not yield to accurate results in a wireless ad hoc context. They do not consider the need for preserving existing flows service level when computing a path's capacity. They also introduce additional traffic in the network that may disturb the network operation and simultaneous measurements may interfere. Finally, in a mobile context, they require permanent re-evaluations of the paths' capacities and therefore do not facilitate local routes reconstruction.

The previously described passive techniques also lead, as further simulation results will show, to an inaccurate estimation. Indeed, they all tackle partially the problem, often reducing the evaluation to the sender's side of the links. Even though ensuring that the medium capacity is not overloaded anywhere in the network may be realized by only considering the emission volumes, accuracy could be improved by considering the synchronization or lack of synchronization of parallel emitters.

If parallel emitters are badly synchronized, repetitive collisions can happen on a link. For example, let us consider the scenario depicted on Figure 1. This configuration, initially presented in [2], is a well-known unfair scenario.

Let us consider that a constant bit-rate flow is present on the link (C,D), we would like to compute the available bandwidth on the link (A,B) in function of the (C,D) flow throughput. In this situation, the evaluations performed by BRuIT, CACP and AAC are identical and their value is represented on Figure 2(a) and 2(b) by the "estimated available bandwidth" curve. This graph depicts simulation results obtained using the NS-2 simulator with a 2 and 11 Mb/s medium capacity, corresponding respectively to a 1.6 and 5 Mb/s

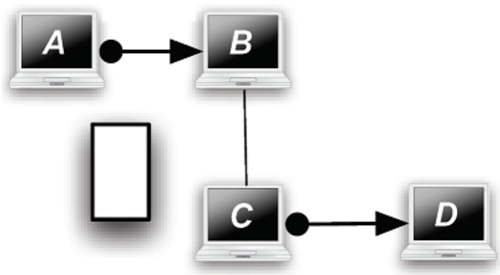
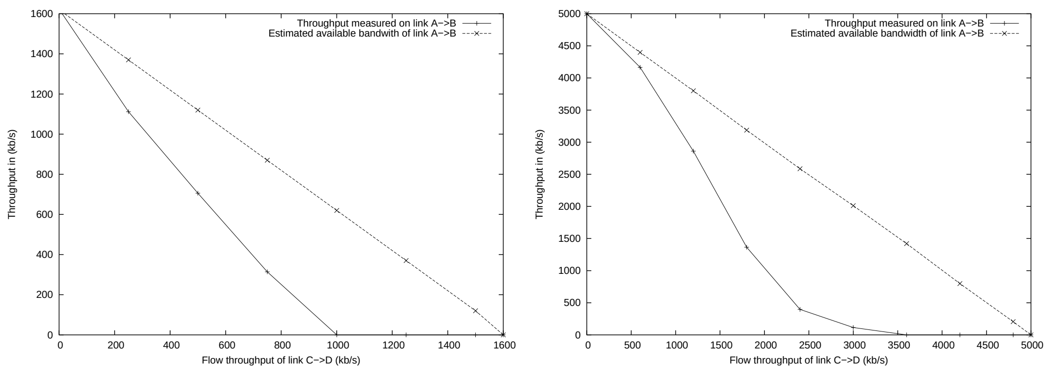


Figure 1: A typical unfair scenario in which asymmetric conditions perturbate node-based evaluations



(a) Capacity of 2Mb/s

(b) Capacity of 11Mb/s

Figure 2: Available bandwidth in the Fig.1 scenario (NS-2 simulation results)

application-layer achievable throughput. For all these protocols, the available bandwidth on the link (A,B) corresponds to the available bandwidth value computed by node B, which is equal to the value computed by node C. It is equal to the capacity of the radio medium minus the bandwidth consumed by the flow on the link (C,D).

The second curve of these figures represents the true available bandwidth on the link (A,B), which corresponds to the maximum throughput that can be actually transmitted on the link. We notice that increasing the throughput of the link (C,D) accentuates the difference between the estimation and the real available bandwidth on the link (A,B).

This difference can be explained by the presence of collisions happening at node B. These collisions lead to an important throughput decrease on the link (A,B), which none of the aforementioned estimators manages to predict. Therefore, it is essential, not only to consider the total amount of traffic emitted in each contention zone, but also to at least take into account collisions.

3 Available bandwidth estimation

Based on the previous literature study and considering how the IEEE 802.11 MAC protocol operates, we can point out a few phenomena that may have an impact on the available bandwidth estimation mechanism:

- Carrier sense mechanism prevents two close emitters from transmitting simultaneously. Therefore, an emitter shares the channel bandwidth with all these close emitters. The channel utilization has to be monitored to evaluate the capacity of a node to emit a given traffic volume.

- For a transmission to take place, both emitter and receiver need that no jamming occurs during the whole transmission. Therefore, the value of the available bandwidth on a link depends on both peers' respective channel utilization ratios but also on the idle periods synchronization. This synchronization needs to be evaluated.

- No collision detection is possible in a wireless environment. Therefore, whenever collisions happen, both colliding frames are completely emitted, maximizing the bandwidth loss. As shown by the scenario depicted on Figure 1, the collision probability needs to be estimated and integrated to the available bandwidth estimation.

- Finally, when collisions happen on unicast frames, the IEEE 802.11 protocol automatically retries to emit the same frame, drawing the backoff counter in a double-sized contention window. The time lost in additional overhead may also have an impact on the available bandwidth and has to be evaluated.

In this section, we examine in turn all four points listed above and describe how we consider these phenomena. Each point could, in theory, be evaluated by measuring some local metrics and exchanging information with close and farther neighbors. However, in this article, we are looking for a lightweight and local mechanism to avoid consuming too much resources for network management. Improving the locality of the estimation also allows a faster reactivity to mobility.

3.1 Carrier sense mechanism: estimating a node's emission capabilities

Whenever a node needs to send a frame, it first needs to contend for medium access and it cannot emit its frame until the medium is free. Therefore a potential sender needs to evaluate the load of the medium, i.e. the proportion of time the medium is idle to determine the chance he has to successfully gain access.

Let us consider a node s in the network during an observation interval of Δ seconds. We use the following notations:

- $T_{idle}(s)$ is the total idle time during which node s neither emits any frame nor senses the medium busy. It is the total time during which both physical and virtual carrier sense mechanisms report an idle state. This includes periods during which no frame is ready to be emitted as well as periods of deferral (backoff time and inter-frame spacing).
- B_s is the available bandwidth of node s, i.e. the maximum throughput it can emit without degrading close flow's rate.
- C_{max} is the capacity of the medium.

During an observation interval Δ, each node may monitor the radio medium in its surroundings and measure the total amount of time that is idle for emitting frames. Each node only considers the idle periods longer than or equal to IEEE 802.11's $DIFS$, as shorter periods do not allow any backoff decrementation or medium access.

As this monitoring neither takes into account the IEEE 802.11 variable overhead, nor the reception side of the transmission, the available bandwidth we can compute this way at node s is imprecise. Above this threshold, collision probability increases quickly. Some frames may still be correctly transferred, though, due to a favorable scheduling of transmissions or to capture effects. Under this threshold, a scheduling between different contending emitters that prevents two simultaneous emissions exists. We therefore consider that this value is an upper bound of the available bandwidth we are seeking:

$$B_s \leq \frac{T_{idle}(s)}{\Delta} \cdot C_{max} \tag{1}$$

This figure can then be rescaled to take into account the fixed overhead (headers, acknowledgments, ...) introduced by the MAC protocol. The medium is considered busy as soon as a signal above the carrier sensing threshold is received. Therefore, this method not only takes into account the bandwidth used in the transmission range of the nodes but also in the whole carrier sensing area.

3.2 Estimating a link's available bandwidth: a first approach

This part has been first presented in [13]. For a better legibility, we describe the main ideas of this article, as our solution starts from this approach. In the previous section, we have evaluated an upper bound of the available bandwidth a node could use to emit frames. The reception part of the transmission also requires that the medium is free during the transmission, thus the previous bound should also be considered at the receiver's side.

3.2.1 Idle periods synchronization

Let us simply consider a radio link composed of two neighbor nodes s and r. To be able to use combinatorial tools, we consider that time is discrete. We introduce the following additional notations:

- δ is the time discretization step.
- $\tau_m = \Delta/\delta$ is the number of time units in a measurement period.
- τ_s (resp. τ_r) is the number of time units during which the medium is available for node s (resp. r) in a measurement period, computed according to the constraints described above.
- B_s (resp. B_r) is the available bandwidth bound for node s (resp. r), measured with the method described previous section.
- $B_{(s,r)}$ is the true available bandwidth on the link (s, r), i.e. the real bandwidth that can be used without degrading close flows.
- $b_{(s,r)}$ is the estimated available bandwidth on the link(s, r)

If B_s is null or close to zero, s either never gains access to the medium or already emits frames at a rate that saturates the radio medium. Similarly, if the medium is always busy on the receiver's side, s's emissions systematically collide and the communication never succeeds. Trivially, we can state that $B_{(s,r)} \leq \min(B_s, B_r)$. However, simply considering that the available bandwidth on a link is the minimum of both values is not sufficient. If sending a flow with a throughput higher than $\min(B_s, B_r)$ necessarily provokes a medium saturation around s and/or r, considering this minimum value as the available bandwidth would lead to an over-estimation of the available bandwidth. Silence periods are desynchronized *a priori*.

Figures 3 and 4 represent the medium availability during time at the emitter and a receiver of a given transmission. In both situations, the bounds measured at each node by the previously described mechanism are the same. On Figure 3, the periods of medium availability of both peers never overlap and the available bandwidth on the link is null. In opposite, the equality holds in the situation depicted on Figure 4 where the periods of medium availability fully match.

In ad hoc networks, the nodes are unlikely to be synchronized. However, precisely evaluating this impact of this asynchronism requires the communication of the exact transmission

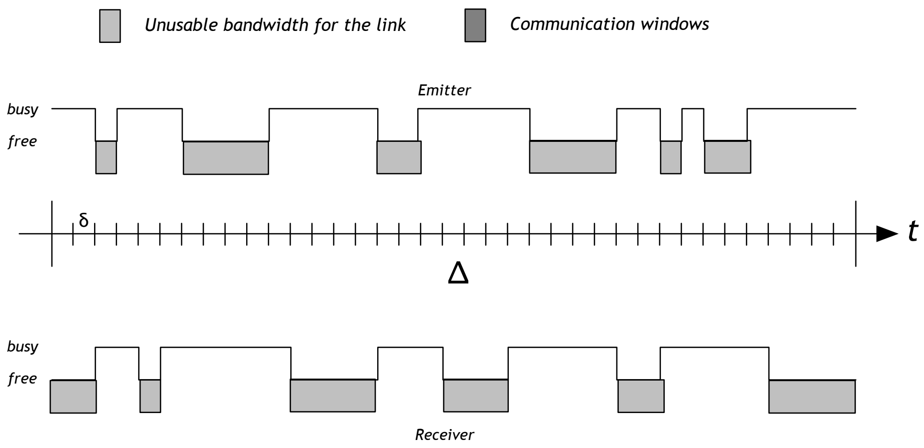


Figure 3: Worst case: medium idle periods of sender and receiver never overlap.

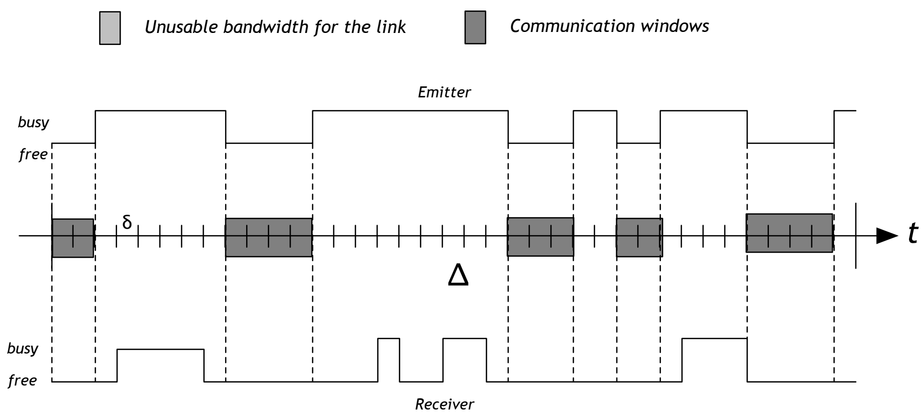


Figure 4: Best case: medium idle periods of sender and receiver always overlap.

patterns of both peers and a fine clock synchronization mechanism, which represents a huge overhead. Therefore, we proposed to use a probabilistic mechanism to estimate the effect of this phenomenon. Let us examine the requirements for a successful frame transmission.

First, for the communication to happen, the medium has to be free during at least $DIFS$ on the emitter's side so that this emitter gains access the medium. Once emission is started, the status of the medium at the emitter's side is irrelevant. On the receiver's side, the medium has to be free during the time required to transmit the whole data frame (T_{DATA}), otherwise a collision would happen. This value is not perfectly accurate, though. It makes the hypothesis that the level of signal that would provoke a collision is equal to the carrier sense threshold, regardless of the distance between emitter and receiver, for example. It also does not take into account propagation time.

Let us consider a uniform random distribution of the medium occupancy over the observation period, it is then possible to compute the expected delay $\mathrm{E}(l_{(r,s)})$ before nodes s and r sense the medium idle simultaneously. We denote by $p(i,j,k)$ the probability that:

- the first occurrence of such synchronization in a measurement interval occurs at time slot i;
- the sender has been idle for j time units before synchronization;
- the receiver has been idle for k time units before synchronization.

$$\text{Then, } p(i,j,k) = \frac{\binom{i}{j} \cdot \binom{i-j}{k} \cdot \binom{\tau_m - i - 1}{\tau_s - j - 1} \cdot \binom{\tau_m - i - 1}{\tau_r - k - 1}}{\binom{\tau_m}{\tau_s} \cdot \binom{\tau_m}{\tau_r}}.$$

From this expression we can compute the probability $\mathrm{P}(l_{(s,r)} = i)$ that the first synchronization occurs at a given time unit and the expected delay $\mathrm{E}(l_{(s,r)})$ before synchronization:

$$\mathrm{P}(l_{(s,r)} = i) = \sum_{j=max(0,\tau_s-(\tau_m-i))}^{\min(\tau_s-1,i-1)} \left(\sum_{k=max(0,\tau_r-(\tau_m-i))}^{\min(\tau_r-1,i-1-j)} p(i,j,k) \right) \quad (2)$$

$$\mathrm{E}(l_{(s,r)}) = \sum_{i=0}^{\min(\tau_m, 2.\tau_m-(\tau_s+\tau_r))} i.P(l_{(s,r)} = i) \quad (3)$$

Still considering a uniform random distribution of the medium occupancy, the available expected bandwidth $\mathrm{E}(b_{(s,r)})$ can be evaluated by expressing the probability that the medium is free simultaneously at the emitter's and receiver's side:

$$\mathrm{P}(b_{(s,r)} = i) = \frac{\binom{\tau_s}{i} . \binom{\tau_m - \tau_s}{\tau_r - i}}{\binom{\tau_m}{\tau_r}}$$

$$\mathrm{E}(b_{(s,r)}) = \sum_{i=0}^{min(\tau_s,\tau_r)} i.P(b_{(s,r)} = i) = \tau_s \times \tau_r$$

To illustrate the importance of this synchronization phenomenon, let us consider the scenario shown at Figure 5. Communications are represented by arrows and nodes in mutual carrier-sense range are linked with a dashed line. If no line joins two nodes, they are totally independent.

We performed simulation using the NS-2 simulator. When no medium access layer modification is performed, the simulated medium capacity is 2 and 11 Mb/s, resulting respectively in a 1.6 and 5 Mb/s application-layer throughput.

Nodes C and D evaluate the available bandwidth on link (C, D) and this value evolves with the throughput of the (E, F) flow. The (A, B) flow constantly uses 50 % of the medium capacity (i.e. 800 kb/s at 2 Mb/s and 2500 kb/s at 11 Mb/s).

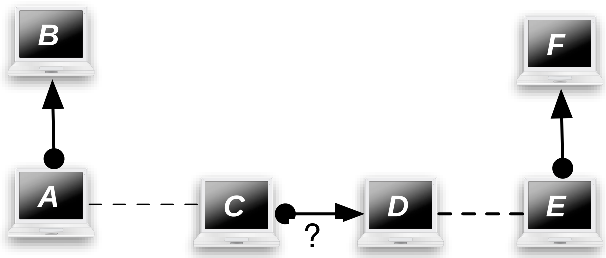
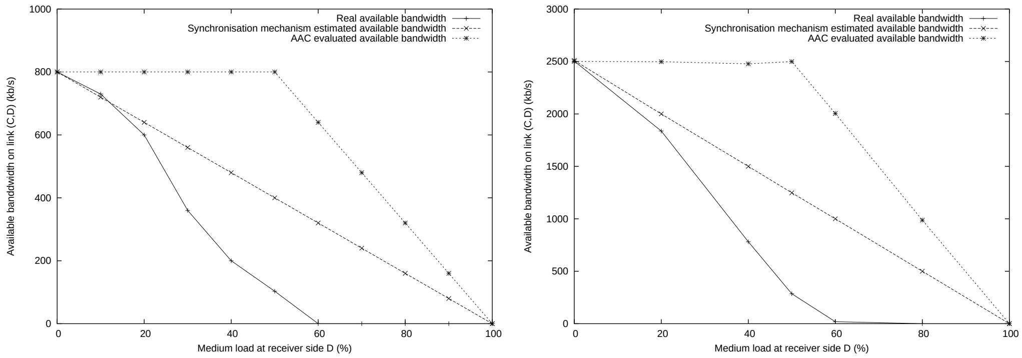


Figure 5: Scenario illustrating link synchronization phenomenon.

Figures 6(a) and 6(b) represent the real available bandwidth on link (C, D), measured by adding a flow between both nodes and evaluating its achieved throughput. It compares this value with the available bandwidth evaluated by the AAC protocol described above and the previously described mechanism.

AAC does not consider synchronization between the sender and the receiver. Hence, it over-estimates the real available bandwidth on this link. Considering this synchronization enhances the estimation but still leads to an over-estimation of the available bandwidth. Indeed, considering a uniform distribution of the silence periods is a strong approximation that does not always reflect the scenario details. Considering another type of idle periods distribution would not lead to better results in the general case, though. Moreover, with this estimation, collisions are not taken into account.

Therefore, if this mechanism achieves a better approximation, it still presents a certain inaccuracy. In the subsequent sections, we will try to refine this estimation by using other easily obtainable data.



(a) Capacity of 2 Mb/s

(b) Capacity of 11 Mb/s

Figure 6: Available bandwidth for the link synchronization scenario.

3.2.2 Taking into account collisions

The use of the previous probabilistic estimation still leads to a certain inaccuracy level. Indeed, there is a chance, even for a single frame that, when a packet is emitted, the medium is not idle at the receiver's side, provoking a collision. A typical example of such problem is the configuration depicted in Figure 1. In this case, protocols like BRuIT, CACP or AAC over-estimate the available bandwidth. The difference between the evaluated and real available bandwidths is due to repeated collisions at node B. This phenomenon has to be evaluated, but how to estimate collision probability without introducing explicit probe packets?

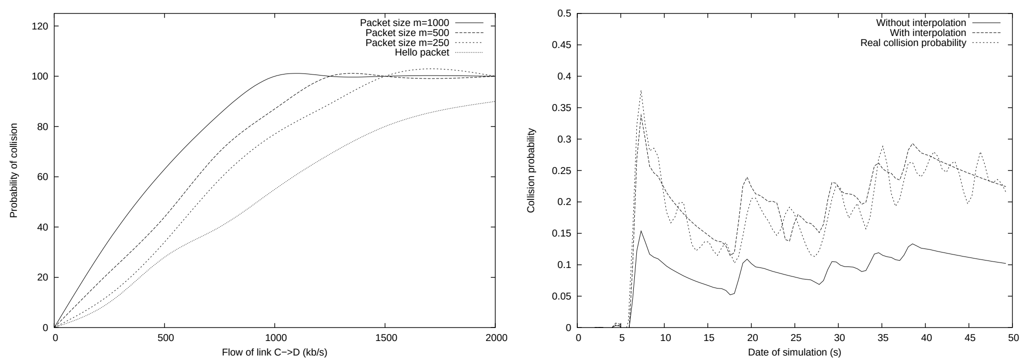
Emitters can estimate the collision probabilities towards certain receivers by counting the number of retransmitted frames. However, the evaluation mechanism should be active even when there is no data traffic, as we would like to predict the potential collisions before sending any data. In our solution, each node periodically transmits *Hello* messages to exchange connectivity and bandwidth-related information. Based on this regular sending, a receiver can compute a collision probability. This collision probability, p_{Hello}, can be expressed by:

$$p_{Hello} = \frac{\text{number of lost } Hello \text{ packets}}{\text{number of expected } Hello \text{ packets}} \tag{4}$$

As soon as a node receives a *Hello* message from one neighbor, it is able to deduce how many *Hello* packets it should receive from this neighbor during the next measurement period. This value corresponds to the "number of expected *Hello* packets". The "number of lost *Hello* packets" corresponds to this expected value minus the number of *Hello* packets actually received during the same time interval.

This estimation is still not completely accurate, though. Considering the number of frames that should have been received in a time interval may mix congestion-related effects with collision-related losses. However, if a sender does not succeed in sending as many *Hello* packets as it should due to an overloaded medium, it means that the links associated to this sender have a low available bandwidth and the inaccuracy in the computation of p_{Hello} does not have a strong impact on the evaluation. Moreover, the introduced error leads to an underestimation of the link's available bandwidth which is better than an overestimation regarding the accepted flows.

Hello packets have a small and constant size. In consequence, p_{Hello} does not reflect the collision probability that may affect larger data packets. To address this issue, we extend the measurement by computing the Lagrange interpolating polynomial fitting the data. If we denote by $f(m)$ this polynomial, the collision probability p_m for packets of m bits is approximated by $p_m = f(m) \cdot p_{hello}$.



(a) Figure 1 scenario: Collision probability on B by simulations for different packet sizes

(b) Precision of the interpolated collision probability on a random topology

Figure 7: Interpolated collision probability

Let us consider the scenario depicted on Figure 1. Figure 7(a) shows the results of simulations performed with NS-2 to obtain the collision probability on node B for different packet sizes (in bytes) and for *Hello* packets. From these measurements, we can deduce the interpolated polynomial corresponding to this situation: $f(m) = -5.65 \cdot 10^{-9}.m^3 + 11.27 \cdot 10^{-6}.m^2 - 5.58 \cdot 10^{-3}.m + 2.19$.

This Lagrange polynomial being computed on a particular scenario, it may not reflect the evolution of the probability in the general case. To illustrate the accuracy of this interpolated collision probability, we generated a random topology of ten nodes and five CBR connections (random source, random destination and random throughput composed of 1000 bytes frames). A specific link between nodes 0 and 1 is placed in the center of the topology. On this link, we measure three parameters:

- The collision probability without the interpolation mechanism p_{hello}.

- The collision probability when the interpolation mechanism p_m is triggered for frames of size m =1000 bytes.
- The real collision probability, computed offline with the simulation traces.

Figure 7(b) represents the three collisions probabilities described above for a frame size of 1000 bytes. As we can see, when the interpolated mechanism is not enabled, there is a clear underestimation of the real collision probability. However, when it is activated, the interpolated collision probability is very close to the real collision probability.

It is important to note that the collision probability depends on the packet size and on the distribution of the medium occupancy at the receiver's side. Up to here, the bandwidth evaluation method we propose combines passive measurements with piggybacking of the information in routing protocol messages.

It can be further enhanced, though. When a node experiences a collision, it doubles its contention window size. Until now, we have considered the proportion of bandwidth lost due to the collisions themselves, but not the additional overhead introduced by the backoff mechanism.

3.2.3 Taking into account the backoff

The time spent in the IEEE 802.11 binary exponential backoff procedure depends on the version of the protocol and on the amount of collisions on the link. It is independent of the frame size. Therefore, when transmitting small frames, ignoring the influence of this backoff introduces a high inaccuracy in the estimated available bandwidth.

First, let us consider that there is no collision. Then the backoff is drawn according to a uniform law in the interval $[0; CW_{min} - 1]$, CW_{min} being determined by the MAC protocol specification. On a large observation window, the backoff can be approximated by its average value $\frac{CW_{min}-1}{2}$. When collisions happen, the exponential backoff mechanism is triggered. After each unsuccessful transmission, the contention window size is doubled up to a maximum value denoted by CW_{max}. In this situation, the average backoff value increases way above $\frac{CW_{min}-1}{2}$ and it seems necessary to model the time consumed by the exponential backoff process.

Let us consider that an arbitrary wireless link suffers from collisions with a probability p. For every frame, the transmission is successful at the first attempt with probability $(1-p)$. It succeeds at the second attempt with probability $p \cdot (1-p)$. After C unsuccessful attempts, C depending on the frame size, the IEEE 802.11 standard specifies that the frame should be dropped as depicted on Figure 8.

If we denote by X the random variable representing the number of attempts performed for the correct transmission of a given frame, we have:

$$P(X = k) = \begin{cases} p^k \cdot (1-p) & \text{if } k \leq C \\ p^k & \text{if } k = C+1 \\ 0 & \text{if } k \geq C+1 \end{cases}$$

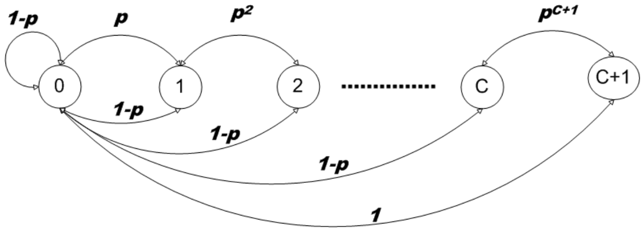


Figure 8: Retransmission mechanism for a collided packet

The average number of retransmissions n for a given frame can be expressed as follows:

$$n = \sum_{k=1}^{+\infty} k \cdot P(X = k) = \sum_{k=1}^{C+1} k \cdot P(X = k)$$

$$n = \sum_{k=1}^{C} k.p^k(1-p) + (C+1)p^{(C+1)}$$

The expected number of backoff slots decremented until the end of transmission attempts for a single frame is therefore:

$$\overline{backoff} = \sum_{k=1}^{+\infty} P(X = k) \cdot \frac{\min(CW_{max}; 2^{k-1} \cdot CW_{min}) - 1}{2}$$

To simplify the expression, let us suppose that $CW_{max} = 2^c \cdot CW_{min}$ with $c \leq C$:

$$\overline{backoff} = \left(\sum_{k=1}^{c} P(X = k) \cdot \frac{2^{k-1} \cdot CW_{min} - 1}{2} \right) + \left(\sum_{k=c+1}^{C} P(X = k) \cdot \frac{CW_{max} - 1}{2} \right)$$

$$\overline{backoff} = \left(\sum_{k=1}^{c} p^{k-1} \cdot (1-p) \cdot \frac{2^{k-1} \cdot CW_{min} - 1}{2} \right) + \left(\sum_{k=c+1}^{C} p^{k-1} \cdot (1-p) \cdot \frac{CW_{max} - 1}{2} \right)$$

$$\overline{backoff} = \frac{1-p}{2} \cdot \left(\frac{1 - (2 \cdot p)^c}{1 - 2 \cdot p} \cdot CW_{min} + \frac{p^c - p^C}{1-p} \right)$$

In this evaluation, the collision probability is independent of the sender's contention window size. Indeed, the collision probability reflects the probability that a frame, once emitted, suffers a collision. This is not exactly true in practice as a sender suffering from a collision will probably provoke a collision itself and trigger the collision avoidance mechanism at another emitter, modifying the collision probability for next frames. We neglect this effect, though, as the collision probability is regularly updated by the previously described mechanism.

Let us denote by K the proportion of bandwidth consumed by the backoff mechanism when collisions happen and by $T(m)$ is the time separating the emission of two consecutive frames. This delay essentially depends on the emission rate and on the frames size m. Then K can be expressed by:

$$K = \frac{DIFS + \overline{backoff}}{T(m)} \tag{5}$$

3.2.4 Available bandwidth computation

The different points mentioned above can be combined to estimate the available bandwidth on a wireless "link", i.e. between an emitter and a receiver. The whole mechanism is lightweight as it does not require much communication, but rather relies on perceptions the nodes have of their environment. To summarize, the available bandwidth between two neighbor nodes s and r can be estimated by the following formula:

$$\boxed{E_{final}\left(b_{(s,r)}\right) = (1-K)\cdot(1-p)\cdot E\left(b_{(s,r)}\right)} \tag{6}$$

where E($b_{(s,r)}$) is the available bandwidth on the link (s, r) evaluated by monitoring the radio channel and combining emitter and receiver's values in a probabilistic manner. p is the collision probability measured on *Hello* packets received and rescaled offline to the appropriate packet size. K is the proportion of bandwidth lost due to the backoff scheme, computed offline as stated in the previous paragraphs.

3.3 Estimating a link's available bandwidth: additional information

Let's consider again the example depicted on Figure 1. The method proposed in Section 3.2.2 allows us to estimate the collision probability at the receiver side B in function of the throughput of link (C, D). So, we can evaluate the available bandwidth on link (A, B).

Now let's consider the reverse problem. The new question is: what is the available bandwidth on link (C, D) if there is a flow on link (A, B)? In other words, what is the proportion of collisions that the emission of C to D will cause at the receiver B.

This estimation is crucial because if a flow is emitted on the link (A, B), we must quantify the maximum throughput on link (C, D) to avoid degrading the flow between A and B. This is equivalent to estimate the amount of collisions that will disrupt the existing flow if the new one is carried. We must determine with accuracy the available bandwidth of link (C, D) to evaluate its impact on link (A, B).

To address this issue, we perform a simulation where throughput of link (A, B) denoted by d_1 is initially equal to the channel capacity D_{max}. We increase gradually the throughput of link (C, D) from 0 to D_{max}. This increase will provoke collisions at the receiver B and degrade the throughput of link (A, B). By analyzing the simulation traces, we can gather information about the real throughput of link (A, B) denoted by d_{1ef} for each value of (C, D)'s throughput. Therefore, the degradation of the throughput of link (A, B) from D_{max} to d_{1ef} is caused by collisions on B due to emission from C towards D. Let's denote by p this collision probability which depends on the throughput of link (C, D).

We have the following relationship :

$$d_{1ef} = D_{max} \times (1 - p) \Longrightarrow p = 1 - \frac{d_{1ef}}{D_{max}} \tag{7}$$

In practice, the only unknown variable for node C is the effective throughput of link (A, B), d_{1ef}. However, all the packets sent from A to B which do not undergo collision are acknowledged. From these ACK packets, C is able to:

- determine whether it is in a configuration of hidden terminals if the destination node of the ACK packet is not in its vicinity;
- estimate the effective throughput d_{1ef} from the frequency of these ACK packets' emission. Actually, the frequency of the ACK packets' emission is equal to the associated data packets' frequency. By measuring the number of ACK packets received during a predefined interval and still considering a specific packet size, d_{1ef} can be evaluated. We can notice that information about packet size is not carried in ACK packets, then a first solution could be to modify these ACK packets in order to include this information. However, we did not consider this solution which needs to modify the IEEE 802.11;
- to deduce the collision probability p and determine the maximum throughput with which C will be able to send without degrading the flow between A and B.

Such an approach has some limitations. First, we need to decode properly ACK packets. If C and B are not in the same communication range but if C's transmissions cause collisions on B, C will not be able to take into account the throughput of flow between A and B in its evaluation. Second, we need to do some assumptions on the data packets size. In the designed protocol, we consider a fixed size of 1000 bytes.

In the following, we explain how to take into account these estimations, described in Sections 3.2 and 3.3, into a protocol version.

4 Basic protocol design

It is quite tricky, from an operational point of view, to evaluate the performance of the sole available bandwidth estimation part of an existing QoS protocol. Therefore, for comparison purposes, we integrated the previously described available bandwidth evaluation technique into a protocol, called ABE for *Available Bandwidth Estimation* and implemented it under NS-2. This simulator has been chosen because of the availability of other protocols models.

The protocol part, i.e. the setting up and maintenance of reservation, does not include any new or specific feature. It is similar to BRuIT, QoS-AODV or AAC, based on broadcasted route request messages, admission control at each intermediate node and explicit reservation by a unicast route reply message issued by the destination. With such a protocol, we can study the impact of our estimation technique on the bandwidth management in the network by comparing the performance of the different protocols.

4.1 ABE (Available bandwidth estimation)

In ABE, neighboring nodes exchange their available bandwidth computed locally via *Hello* messages. Every Δ seconds, each node locally estimates its medium occupancy ratio and includes this information in a *Hello* packet. These values are then converted into link evaluations using equation 6, as mentioned in the previous section.

The accuracy of the available bandwidth evaluation obviously depends on the value of Δ, which is equivalent to a sampling period. The larger Δ is, the more stable the measurements will be, hiding fast variations in the medium load. However, Δ should also be small enough to allow fast reactions to long-term load variation and to nodes mobility.

Hello-based techniques generate additional overhead depending on the *Hello* emission frequency. Ideally, the *Hello* packets frequency should be adapted to the nodes mobility and/or to the flows dynamics. In this protocol version of ABE, we chose, in order to have meaningful comparisons, to fix this value to $\Delta = 1$ second. Similarly, all compared protocols are tuned accordingly to emit one information frame each second.

4.2 ABE Admission control and QoS routing

As indicated previously, the ABE routing protocol is indeed a cross-layer routing protocol. The MAC layer estimates proactively and periodically the available bandwidth of the neighboring links and the routing layer is in charge of discovering QoS routes complying to the applications demands, basing their decisions on the MAC layer result. The routing of ABE is strongly inspired by AODV and consists in two major parts: route discovery and route maintenance.

4.2.1 Route discovery

The aim of the route discovery procedure is to find a route between the sender and the receiver that meets the constraints specified by the application level in term of bandwidth. Therefore, two flows with the same source and destination can follow different routes depending on the network state.

ABE performs on-demand route discovery like in AODV. When a source node has data to send, it broadcasts a route request (*RREQ*) to its neighbors. This broadcast undergoes a first admission control to ensure that throughput of existing flows which are in hidden nodes configuration with the emitter, will not be degraded. This verification is performed by the estimation provided in Section 3.3. The *RREQ* packet contains the address of the sender, the channel use, the requirements at the application level, the destination address and a sequence number.

Each mobile that receives such a *RREQ* performs a second admission control by simply comparing the bandwidth requirement carried in the *RREQ* packet and the estimated available bandwidth on the link it received the *RREQ* on. This verification ensures that the new flow to be emitted will not be degraded by existing flows. If this checking is positive, the node then cheks that it will not degrade hidden flows (estimation of Section 3.3). If it is not the case, it adds its own address to the route and forwards the *RREQ*; otherwise it silently discards the message. Finally, when the destination receives a first *RREQ*, it sends a unicast route reply (*RREP*) to the initiator of the request along the reverse path. The resources are then reserved and the new QoS flow can be sent.

4.2.2 Intra-flow contention problem

Simply comparing the bandwidth application requirement and available bandwidth on a link is not sufficient to decide on the network ability to convey a flow. Indeed, the intra-flow contention problem, described above, has to be considered when performing multi-hop admission control.

In [6], the authors compute a value called *contention count* (CC) of a node along a given path. This value is equal to the number of nodes on the multi-hop path that are located within the carrier-sensing range of the considered node. To compute the CC of each node, authors analyze the distribution of the signal power.

As in [8], in ABE, for simplicity reasons, we rather use a direct relationship between the end-to-end throughput and the number of hops. Hence, after consideration of the intra-flow contention on an intermediate node j which is located at K hops from the source and has received the RREQ from a node i, the available bandwidth considered for admission control, denoted by $B(i, j)$, is equal to:

$$B(i,j) = \frac{E_{final}(b_{(i,j)})}{min(K,4)} \tag{8}$$

where $\mathrm{E}_{final}(b_{(i,j)})$ is the available bandwidth of link (i, j) as computed by Equation 6.

4.2.3 Route maintenance

A route maintenance process is essential, especially in the case of mobility. However, the goal of this article is not to propose a new QoS routing protocol. Therefore, we implemented a simple detection and reaction mechanism. ABE detects a broken route by monitoring the *Hello* messages. If a node does not receive any *Hello* packet from a neighbor within a certain time interval equal to 3 s (the time to transmit 3 *Hello* packets), or if one of its link does not meet the reserved bandwidth any more, it sends a route error (*RERR*) to the source which subsequently rebuilds its route.

5 Simulations

In this section, we compare the accuracy of our estimator with other passive approaches by simulation, using network simulator 2 (NS-2.27)[1] and the IEEE 802.11 implementation provided with the simulator.

The parameters used for all scenarios are presented on Table 1. We compare the performances of our estimation technique through the ABE protocol described above with three admission control protocols available on the web and described in Section 2: QoS-AODV, AAC, and BRuIT[2].

Parameters	Values
HELLO interval	1 s
Packets size	1000 bytes
Medium capacity	2 or 11 Mb/s
Communication range	250 m
Carrier sensing range	550 m
Grid size	1000,m×1000 m
C (Number of retransmissions)	6

Table 1: General parameters for simulations

5.1 Admission control mechanism accuracy

To evaluate the different protocols and illustrate the effectiveness of ABE, we generate random topologies with random constant bit-rate flows (random source, random destination and random throughput with fixed 1000 bytes frames). For each of these protocols, similar scenarios (same number of nodes and same number of flows) lead to similar behaviors. Hereafter we only analyze two specific scenarios. For each scenario, the results presented here were obtained over 30 simulation runs with different random seeds.

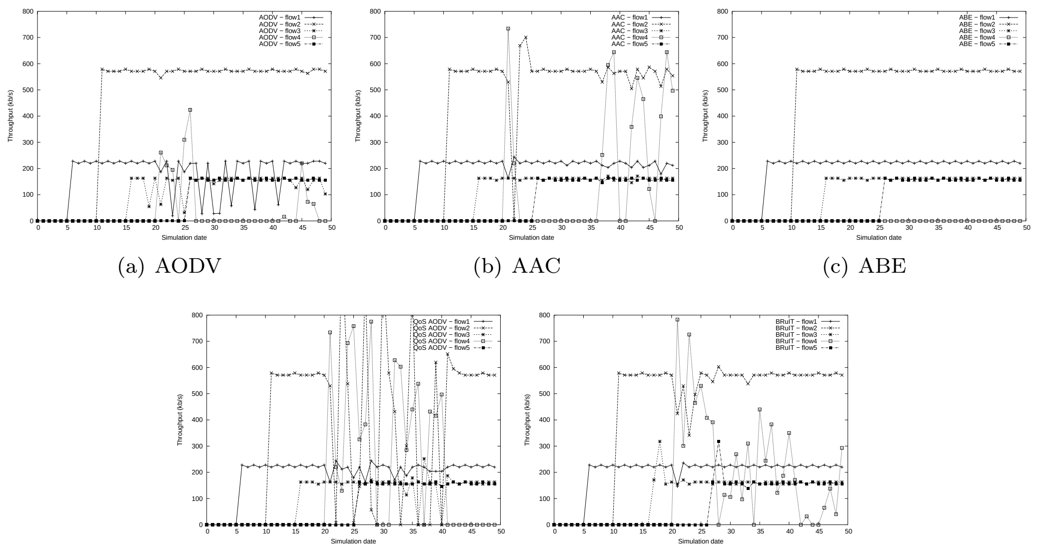
5.1.1 Single-hop communications

First of all, to illustrate the accuracy of ABE, let us examine a small static network involving 10 randomly positioned nodes. The channel capacity is set to 11 Mb/s and five one-hop CBR connections are established in the network. For each flow, the source is randomly chosen

[1] `http://www.isi.edu/nsnam/ns/`

[2] BRuIT: `http://perso.ens-lyon.fr/isabelle.guerin-lassous/QoS.html` – QoS-AODV and AAC: `http://www.ctr.kcl.ac.uk/members/ronan/default.asp`

and the destination is also randomly chosen among the source's neighbors. Nodes are not in the same contention area, but we only consider one-hop flows, leaving intra-flow contention problems aside for now. Each simulation lasts fifty seconds and one flow is started every five seconds. Figure 10 represents the evolution of the different flows throughputs in function of the simulation time for all 4 compared solutions.



(a) AODV (b) AAC (c) ABE

(d) QoS-AODV (e) BRuIT

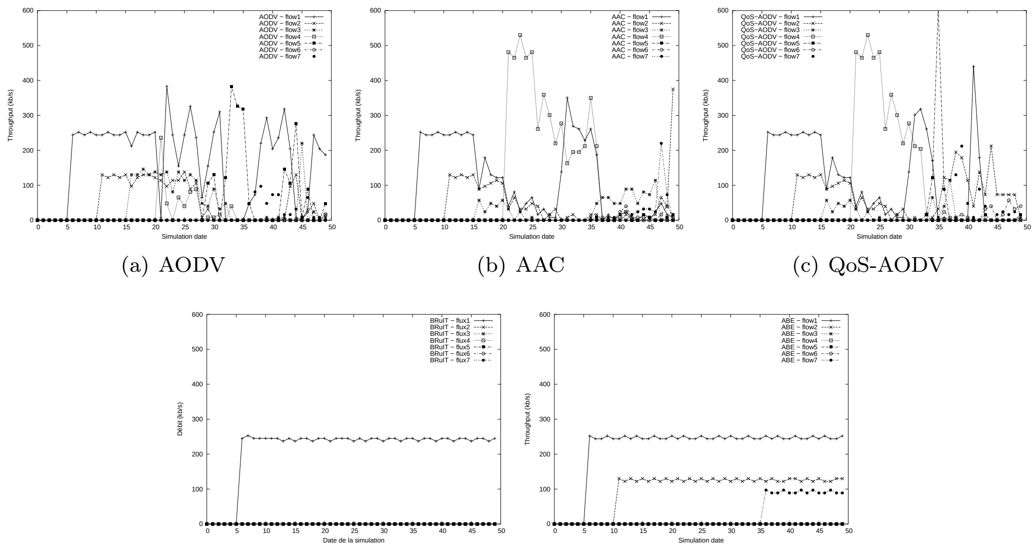
Figure 9: Throughput of each flow using AODV, AAC, QoS-AODV, BRuIT and ABE (11 Mb/s medium capacity)

Not performing any admission control (AODV) leads to a congested network as new flows are added, which indicates that the channel is not able to sustain this scenario's traffic. As a consequence, the throughputs achieved by the flows are reduced. AAC overestimates the available bandwidth during the admission control phase of the fourth flow. Therefore, as soon as it gets accepted, the throughputs of the already existing flows begin to decrease. Similar observations also apply to QoS-AODV and BRuIT. None of the these protocols takes into account the collision probability due to hidden nodes phenomenon, the influence of the increased contention window or the de-synchronization of emitter and receiver. Consequently, this lack of information leads to an upper estimation of the available bandwidth, which is particularly noticeable when the fourth flow requires admission control.

With ABE, all flows except the fourth one are admitted without any throughput degradation. The fourth flow is rejected by the admission control procedure. This scenario indicates that considering collision probability and synchronization between emitter and receiver's medium occupancy is necessary to improve the estimation of residual bandwidth and therefore to avoid false admission of QoS flows.

5.1.2 Multi-hop flows

Let us now consider multi-hop flows. In this scenario, 20 nodes are randomly positioned in the simulation square. Seven CBR connections are established with random throughputs composed of 1000 bytes frames. Each flow's source and destination are not direct neighbors. This requires to perform routing and admission control at each intermediate node and considering intra-flow contention. The channel capacity is now 2 Mb/s. We choose this value to show that our bandwidth estimation scheme works as well from 2 to 11 Mb/s. Simulations last fifty seconds and one flow is started every five seconds.



(a) AODV (b) AAC (c) QoS-AODV

(d) BRuIT (e) ABE

Figure 10: Throughput of multi-hop flows with AODV, AAC, QoS-AODV and ABE (2 Mb s medium capacity)

Figure 10(a) shows the throughput of the seven flows when no admission control is performed. Once again the network becomes congested and routes are often broken resulting in a decreased throughput for all flows.

Figures 10(b) and 10(c) represent the throughput of all flows when AAC and QoS-AODV are used. The behavior observed in the first simulations can still be noticed here. AAC does not consider the impact of collisions and QoS-AODV neither takes into account collisions, nor evaluates the intra-flow contention problem. Therefore, the available bandwidth is over-estimated by both protocols and more flows are accepted than the network is capable of sustaining.

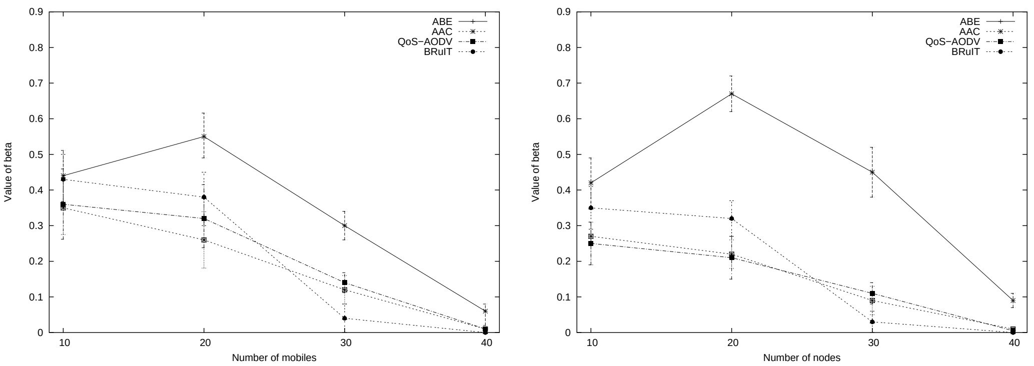
Figure 10(d) presents the results obtained with BRuIT. On the opposite, only the first flow is admitted. BRuIT considers the worst scheduling case and underestimates the available bandwidth, not taking into account the fact that distant emissions may be performed in parallel.

Finally, Figure 10(e) shows the throughput achieved by ABE which performs a more accurate admission control by admitting three flows out of seven. All the admitted flows meet their bandwidth requirements.

5.2 Accuracy of the estimation

Let us now investigate the general case. To reflect the accuracy of the available bandwidth estimation, we define a metric accounting for the number of *right admissions*. A right admission happens when the admission control protocol allows the routing of a flow and this flow's throughput is not degraded by more than 5 % when it gets transfered. The metric we represent hereafter is defined by the following expression:

$$\beta = \frac{\text{Number of } \textit{right admission}}{\text{Number of flows requesting QoS routes}}$$



(a) Carrier sensing range = 2 × Communication range (b) Carrier sensing range = Communication range

Figure 11: Acceptance rate with ABE, AAC, QoS-AODV and BRuIT

A falsely admitted flow either degrades the throughput of close flows or is not able to achieve its desired throughput. Hence, the value of β decreases. We measured the value of β by simulation on networks composed of 10 to 40 nodes, using an 11 Mb/s medium capacity. Each simulation lasts 100 seconds and five randomly chosen pairs of nodes try to establish CBR connections towards five random destinations. The throughput of each connection is uniformly drawn between 0 kb/s and 500 kb/s.

Figure 11(a) represents the value of β for ABE, AAC, QoS-AODV and BRuIT. As the density of the network increases, β decreases, which is expected as the available bandwidth per link decreases and lower quality routes are established. When the network is not too loaded, the acceptance rate of QoS-AODV and AAC is smaller than BRuIT and ABE. QoS-AODV and AAC cause more false admissions due to the overestimation of the available bandwidth and throughputs of close flows are degraded. However, when the network becomes loaded, BRuIT's acceptance rate decreases. Performing an under-estimation of the available bandwidth, it tends to accept less flows than what the network is able to convey. ABE exhibits good performance in every situation.

Carrier sensing area
Reducing the carrier sensing range until it becomes equal to the communication range (Figure 11(b)) decreases the value of β for protocols like AAC, AODV and BRuIT for a same number of nodes comparatively to Figure 11(a). Actually, reducing the carrier sensing range will create more hidden terminals configuration. Protocols like AAC, AODV and BRuIT don't integrate in their evaluation this kind of configuration contrary to ABE (see Section 3.3). This situation allows ABE to increase its acceptance rate β. The results show that reducing the carrier sensing range if hidden nodes configuration is taken into account in the available bandwidth evaluation like in ABE is an interesting approach.

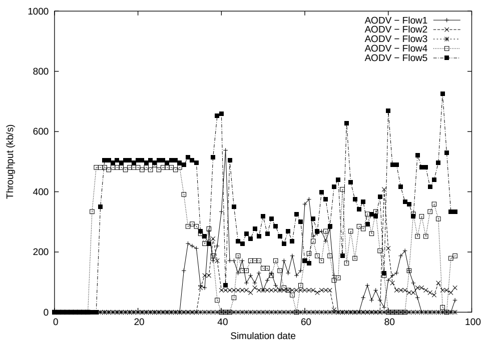
Packet size
We consider the previous scenario with the carrier sensing range equal to the communication range. This situation, as seen before, will create more hidden terminals configuration. In Section 3.3, to estimate throughput of emitter in hidden nodes configuration, we need to gather information about packet size. However, in IEEE 802.11 ACK packet, this information is not available. Consequently, we consider a mean packet size. In all the simulations presented above, we use a fixed packet size of 1000 bytes. If the packet size is smaller than this threshold of 1000 bytes, the throughput of hidden emitter is upper-estimated. Therefore smaller flows will be admitted but no existing flows is degraded. For larger packets with a MTU of 1500 bytes, the throughput of a hidden emitter is larger and the evaluation provided by ABE is over-estimated.

Packet size / Number of nodes	**10**	**20**	**30**	**40**
1000 bytes	42%	67%	38%	7%
1500 bytes	37%	61%	35%	5%

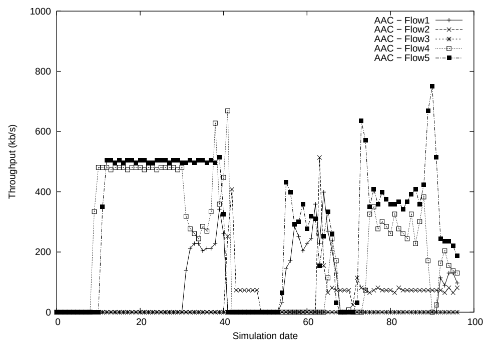
Table 2: Value of β for packet size of 1000 et 1500 bytes

Table 2 represents the value of β for packet size of 1000 and 1500 bytes in function of the number of nodes in the network. We notice that when packet size increases from 1000 to 1500 bytes, the value of β is reduced to almost 6%. For example, for 30 nodes in the network, β decrease from 38 to 35% of acceptance rate. Therefore, this simulation shows that the packet size used to approximate the throughput of hidden emitters will not have a strong impact to the acceptance rate. Therefore, it seems reasonable to consider a mean packet size of 1000 bytes.

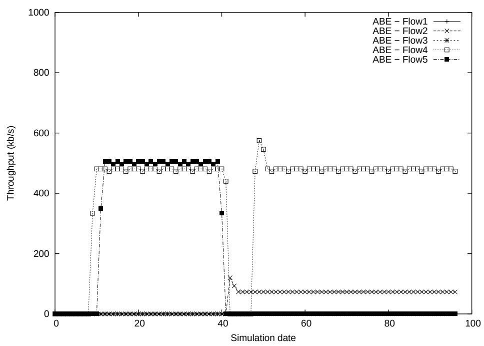
5.3 Mobile networks



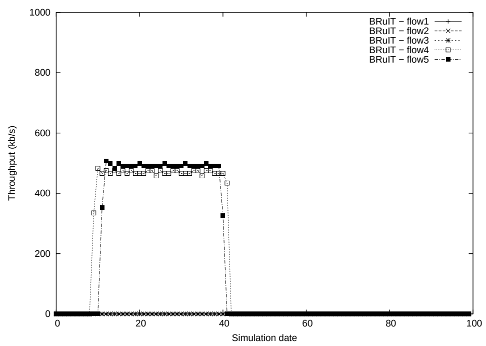
(a) AODV



(b) AAC



(c) ABE



(d) BRuIT

Figure 12: Throughput obtained by AODV, AAC, BRuIT and ABE in mobile networks

It is illusory to provide hard QoS guarantees when nodes are mobile. QoS violations appear due to the topology changes which result either in route breakage, or in unexpected variations of the available throughputs. With ABE, both problems require a whole new route discovery process. This search process may take a long time and generate extra message overhead.

To investigate the effect of mobility on flows throughput, we have performed simulations with 10 randomly positioned nodes. Five CBR traffic are generated with random throughputs and the starting dates of these flows are spaced by two seconds. Nodes move according to a random waypoint mobility model with a maximum speed of 20 m/s and a pause time of 10 s. We chose this simple mobility pattern as the goal here is not to investigate practical scenarios involving human-like mobility. EAch simulation lasts 100 s and the physical rate is of 2 Mb/s.

Figures 12(a) and 12(b) show that when nodes are mobile, both AAC and AODV lead to severe throughput degradations. With BRuIT (Figure 12(d)), the underestimation that was already pointed out before still holds and only two QoS flows are admitted. Figure 12(c) shows that despite the mobility, ABE forwards flows with their specified bandwidth requirements.

In this scenario, it is possible to notice that flow number 4 loses its route during 5 seconds, while flow number 5 does not find any alternate route after the date 40 s. We also noticed that introducing pause times during which flows search for new suitable routes leads to a greater success in the re-routing process.

6 Conclusions and future works

In this paper, we have presented a new technique to compute the available bandwidth between two neighbor nodes and by extension along a path. This method combines channel monitoring to estimate each node's medium occupancy including distant emissions, probabilistic combination of these values to account for synchronization between nodes, estimation of the collision probability between each couple of nodes and variable overhead's impact estimation. This mechanism only requires one-hop information communication and may be applied without generating a too high additional overhead.

This technique has been integrated into a reactive routing protocol for comparison purposes. We show the accuracy of the available bandwidth measurement through NS-2 simulations. These results show that single-hop flows and multi-hop flows are admitted more accurately, resulting in a better stability and overall performance. Results are encouraging in fixed networks as well as in mobile networks. From our point of view, these scenarios prove that the most difficult point when designing a QoS protocol is not the routing process, but the estimation of available resources through the network.

As future works, we plan to focus on two issues. First, in our current evaluation, we make no difference between the bandwidth consumed by QoS flows with the bandwidth consumed by best effort flows. Therefore, it may be possible that a node considers its available bandwidth on a link as almost null whereas the whole bandwidth is consumed by best effort flows. Decreasing the rate of these flows may lead to a higher acceptance rate. Differentiating flow types may also result in a better utilization of the network resources. In parallel, we are investigating the delay metric, as preliminary studies indicate that some parts of the approach described in this article may be used or converted to this other important parameter.

References

[1] Andreas Johnsson, Bob Melander, and Mats Björkman. Bandwidth Measurement in Wireless Network. Technical report, Mälardalen University, Sweden, March 2005.

[2] Vaduvur Bharghavan, Alan J. Demers, Scott Shenker, and Lixia Zhang. MACAW: A Media Access Protocol for Wireless LAN's. In *SIGCOMM*, pages 212–225, 1994.

[3] Claude Chaudet, Isabelle Guérin Lassous. BRuIT - Bandwidth Reservation under InTerferences influence. In *In Proceedings of European Wireless 2002 (EW2002)*, Florence, Italy, Feb 2002.

[4] Claude Chaudet, Isabelle Guérin Lassous. Evaluation of the BRuIT protocol . In *In Proceedings of the IEEE 61st Semiannual Vehicular Technology Conference (VTC Spring)*, Stockholm, Sweden., May 2005.

[5] Frank Y. Li, Mariann Haugea, Andreas Hafslund, Oivind Kure and Pal Spilling. Estimating Residual Bandwidth in 802.11-based Ad Hoc Networks: An empirical Approach. In *Proceedings of The Seventh International Symposium on Wireless Personal Multimedia Communications (WPMC 2004)*, Abano Terme, Italy, September 2004.

[6] K. Sanzgiri, I. D. Chakeres, and E. M. Belding-Royer. Determining Intra-Flow Contention Along Multihop Paths in Wireless Networks. In *In proceedings of BroadNet*, San Jose, California (USA), October 2004.

[7] Kaixin Xu, Ken Tang, Rajive Bagrodia, Mario Gerla and Michael Bereschinsky. Adaptive Bandwidth Management and QoS Provisioning in Large Scale Ad Hoc Networks. In *Proceedings of MilCom*, Boston, USA, October 2003.

[8] L. Chen and W. Heinzelman. QoS-aware Routing Based on Bandwidth Estimation for Mobile Ad Hoc Networks. *IEEE Journal on Selected Areas of Communication*, 3, 2005.

[9] Bob Melander, Mats Bjorkman, and Per Gunningberg. A new end-to-end probing analysis method for estimating bandwith bottlenecks. In *Proceedings of the Fifth Global Internet Symposiumin held in conjunction with Globecom 2000*, San Francisco, USA, November 2000.

[10] Ravi Prasad, Margaret Murray, Constantinos Dovrolis, and K. Claffy. Bandwidth estimation: metrics, measurement techniques, and tools. *IEEE Network*, 17(6):27–35, November 2003.

[11] R. de Renesse, M. Ghassemian, V. Friderikos, A.H Aghvami. Adaptive Admission Control for Ad Hoc and Sensor Networks Providing Quality of Service,. Technical report, King College London, May 2005.

[12] Ronan de Renesse, Mona Ghassemian, Vasilis Friderikos, A. Hamid Aghvami. QoS Enabled Routing in Mobile Ad Hoc Networks. In *IEE 3G 2004*, 2004.

[13] C. Sarr, C. Chaudet, G. Chelius, and I. Guérin Lassous. A node-based available bandwidth evaluation in IEEE 802.11 ad hoc networks. *International Journal of Parallel, Emergent and Distributed Systems*, 21(6), 2006.

[14] Yaling Yang and Robin Kravets. Contention Aware Admission Control for Ad Hoc Networks. *IEEE Transactions on Mobile Computing*, 4:363–377, 2005.

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