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INSTITUT NATIONAL DE RECHERCHE EN INFORMATIQUE ET EN AUTOMATIQUE

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Abstract: In this paper, we propose an accurate method to evaluate the available bandwidth in IEEE 802.11-based ad hoc networks. We improve the method described in [12]. We evaluate our solution on different scenarios. We also provide a comparison with different QoS routing protocols, BRuIT, AAC and QoS-AODV, based on different available bandwidth estimation techniques.

Key-words: Available bandwidth estimation, *ad hoc* networks, IEEE 802.11, Quality of Service

* Inria Ares / Citi, Insa de Lyon – 69621 Villeurbanne Cedex – France
† GET/ENST; LTCI-UMR CNRS 5141 – 37-39, rue Darreau – 75014 Paris – France
‡ Inria Ares / Citi, Insa de Lyon – 69621 Villeurbanne Cedex – France
§ Inria Ares / Citi, Insa de Lyon – 69621 Villeurbanne Cedex – France

Unité de recherche INRIA Rhône-Alpes
655, avenue de l'Europe, 38334 Montbonnot Saint Ismier (France)
Téléphone : +33 4 76 61 52 00 —Télécopie +33 4 76 61 52 52

Amélioration de la précision dans l'évaluation de la bande passante pour les réseaux ad hoc basés sur 802.11

Résumé : Dans ce rapport de recherche, nous proposons une nouvelle méthode pour évaluer la bande passante résiduelle dans un réseau ad hoc 802.11. Pour cela, nous proposons des améliorations par rapport à la première estimation faite dans [12]. Nous comparons par la suite, les performances de notre solution avec différents protocoles de qualité de service tels que BRuIT, AAC et QoS-AODV, qui sont basés sur d'autres techniques d'évaluation de la bande passante résiduelle

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

1 Introduction

The quality of service issues in ad hoc networks are now extensively studied and more and more QoS protocols are proposed. Many protocols concern QoS routing. The goal of such a routing is either to provide the best routes in function of some parameters (like bandwidth, delay, packet loss, etc.) or to find routes that will offer guarantees on some of these parameters. Many QoS routing protocols have been proposed so far and many of them consider the bandwidth parameter.

To design an efficient QoS routing, it is very important to get very accurate information on the used and available bandwidth. Such estimation is not so easy to compute in multihop wireless networks. The nodes share the medium and their perception of the used bandwidth or the available bandwidth can be very different from one mobile to another. Therefore, before introducing a new flow in the network, each mobile needs to precisely determine the available bandwidth that is offered to it, but it also needs to know the available bandwidth available to the nodes with which it may share the medium in order to not penalize them.

Different solutions, previously proposed, address this problem. They can be classified into two main categories: the intrusive techniques that send probe packets for the estimation and the passive techniques that are based on a local computation of the available bandwidth and sometimes on a sparse exchange of this information. In this article, we claim that the proposed solutions are not satisfactory and that it is possible to improve the accuracy in the available bandwidth estimation for ad hoc networks. Indeed, the intrusive solutions consume too much bandwidth while the passive solutions mainly compute the available bandwidth with the formula "capacity minus used bandwidth", which is not sufficient.

Our solution is based on the IEEE 802.11 technology as it is widely used in wireless local networks and multihop wireless networks. However, our method could also be applied on other technologies as soon as they are based on a CSMA/CA approach, simply by using different values for the parameters we use in our solution.

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 maximum throughput a flow can achieve between two neighbor nodes. In [12], we have proposed a method to evaluate the available bandwidth in ad hoc networks. If this method enhances the accuracy of the estimation, it is still unsatisfactory in some configurations. In this article, we improve this method. We also provide an evaluation of the technique by comparing it to several passive estimation solutions.

Section 2 presents related work. Section 3 describes our available bandwidth estimation mechanism. Then, Section 4 evaluates and compares our solution with some other techniques.

2 Related work

Available bandwidth estimation techniques can be divided in two major approaches:

- We call *intrusive approaches* techniques that are based on end-to-end probe packets to estimate the available bandwidth along a path.
- We call *passive approaches* techniques that use local information on the used bandwidth (like for instance the channel usage computed from the sensing of the radio medium) and that may exchange this information via local broadcasts. Usually these local broadcasts are performed using the *Hello* messages that are used in many routing protocols to discover local topology. If these exchanges are not too frequent, we consider that the technique is not intrusive.

2.1 Intrusive bandwidth estimation techniques

Many active bandwidth estimation techniques have been proposed for wired networks. A detailed survey of the different techniques is proposed in [9]. The Self-Loading Periodic Streams (SLoPS) technique measures the end-to-end available bandwidth by sending packets of equal size and by measuring the one-way delays of these probing packets. The source increases the rate of the probing packets; as soon as there is a variation in the delay, one can say that the path is saturated and that the measured point, just before the variation, corresponds to the available bandwidth. The Trains of Packet Pairs (TOPP [8]) technique is based on the same principle. The main difference between these two methods concerns the rate increasing function: TOPP increases linearly the rate whereas SLoPS uses a binary search.

Based on TOPP method, DietTOPP [1] has been developed for a wireless environment.

The main idea of [4] is that a probe packets delay higher than the maximum theoretical delay can characterize the channel utilization. The authors propose a method to compute the medium utilization from the delays and then to derive the available bandwidth from the utilization.

All these techniques are *active* as they use end-to-end probe packets to characterize the channel. When every node in an ad hoc network needs to perform such an evaluation for several destinations, the number of probe packets introduced in the network can be important. Therefore, such techniques have mainly two drawbacks: they consume much bandwidth and they have an impact on the on-going traffic they measure.

2.2 Passive bandwidth estimation techniques

In [6], each mobile estimates the available bandwidth by computing the channel utilization ratio and using a smoothing constant. The channel utilization ratio is deduced from a permanent monitoring of the channel status (idle or busy). The QoS routing protocol designed in this article is only based on the available bandwidth at each node and does not consider the possible distant interfering nodes.

Chaudet and Guérin Lassous have proposed a bandwidth reservation protocol, called Bandwidth Reservation under InTerferences (BRuIT) [3], that takes into account the notion of carrier sensing area in the available bandwidth estimation. Indeed, with CSMA protocols (like in IEEE 802.11), two nodes within carrier sensing range share the medium and thus the bandwidth, even if they cannot directly communicate. Therefore, each node needs not only to know the channel occupancy in its communication range, but also in its carrier sensing range. BRuIT attempts to compute the channel usage in the carrier sensing area. In BRuIT, the carrier sensing area is approximated by the two-hop neighborhood. Each node provides information about the total bandwidth it uses to route flows and about its neighbors and their usage of the bandwidth, by periodically broadcasting a Hello message containing this information. Then, each node can compute the bandwidth usage in its two-hop neighborhood and can then approximate the used bandwidth and derive the available bandwidth in its carrier sensing area. This last computed value is then added to the Hello messages. Thus, each node knows the available bandwidth of its two-hop neighbors. The main drawback of this method is that the two-hop neighborhood may not correspond exactly to the carrier sensing area.

In [13], Yaling and Kravets have proposed the Contention Aware Admission Control Protocol (CACP). The goal is, like in BRuIT, to determine the available bandwidth of the nodes in the carrier sensing area. First, each node computes the local idle channel time fraction by a permanent monitoring of the radio medium. Then, the authors propose three different techniques: to use, like in BRuIT, Hello messages to broadcast this information over the two-hop neighborhood; to increase the transmission power of nodes such that all the nodes in the carrier sensing area could be reached; or to reduce the sensitivity of the mobiles in order that each node takes into account the bandwidth used in its carrier sensing area. Thus, each node deduces, without an exchange of messages, the available bandwidth of the nodes in its carrier sensing area. The authors raise the intra-flow contention problem, also studied in [5]. As wireless links are not isolated, multiple links on the route of a flow may contend for bandwidth at a single location. Therefore, it is important to count the number of nodes that contend for bandwidth on each link along a path.

QoS-AODV [11] is a per node available bandwidth estimation. To estimate the available bandwidth, the authors propose a metric called BWER (Bandwidth Efficiency Ratio) that computes the ratio between the number of transmitted and received packets. To collect the neighbors' available bandwidth, Hello messages are periodically broadcasted in the one-hop vicinity. Then, the available bandwidth of a node is considered as being the minimum of the available bandwidth between the one-hop neighbors and current node.

In the protocol AAC [10], each node estimates its local used bandwidth by simply adding the size of sent and sensed packets over a fixed period of time. The packet size is computed by estimating the medium occupancy time. Therefore this method considers data sent in the carrier sensing area. Finally, the link available bandwidth is the minimum available bandwidth of all nodes belonging to the carrier sensing areas of the sender and the receiver. AAC also estimates the contention count of nodes along a QoS path to solve intra-flow contention problem.

2.3 Motivation

The active techniques are not satisfactory as they consume bandwidth and reduce the capacity of the network. They can also affect the on-going flows.

The passive techniques, presented previously, do also not seem satisfactory, as they are based on estimations computed on the nodes and they do not provide an accurate evaluation of the available bandwidth on the links. However, accurate estimations on the links are very important to find adapted paths with a QoS routing.

To provide an accurate estimation on the links, some considerations need to be taken into account. First, for a communication to take place, the medium has to be simultaneously available on both sender and receiver's sides. Thus, the available bandwidth on a link can largely vary depending on the synchronization of the receiver and sender medium idle periods. Therefore, knowing the overlapping of the emitter and receiver silent periods may improve the available bandwidth estimation on the link. Such knowledge is not computed in the passive evaluations presented in the previous section. Moreover, collisions that can happen on a link need to be taken into account in the evaluation. For example, let us consider the scenario depicted on Figure 1. This configuration, initially presented in [2], is a well-known unfair scenario.

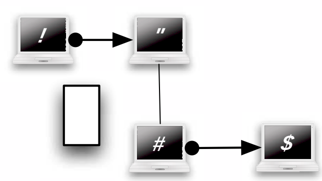


Figure 1: An unfair scenario

Let's consider that there is an ongoing flow on the link (C,D) of fixed rate and that one needs to estimate the available bandwidth on the link (A,B). In this case, the evaluations provided by BRuIT, CACP or AAC are the same and are represented by the "estimated available bandwidth" on Figure 2 in function of the flow throughput of the link (C,D). Actually, in these protocols, the available bandwidth on the link (A,B) corresponds to the available bandwidth computed by node B (equivalent to the one computed by node C) and is equal to the capacity of the radio medium (in the simulations done on NS2, the used modulation is at 2 Mb/s, thus the capacity corresponds to 1.6 Mb/s in practice) minus the bandwidth consumed by the flow on the link (C,D). The second curve of this figure represents the real available bandwidth on the link (A,B) that corresponds to the maximum throughput that can be actually transmitted on this link (and obtained by simulation). We notice that as the throughput of the link (C,D) is increasing, the real available bandwidth on the link (A,B) becomes lower than the estimated available bandwidth. This difference finds an explanation in collisions that happen at the receiver side B. These collisions greatly

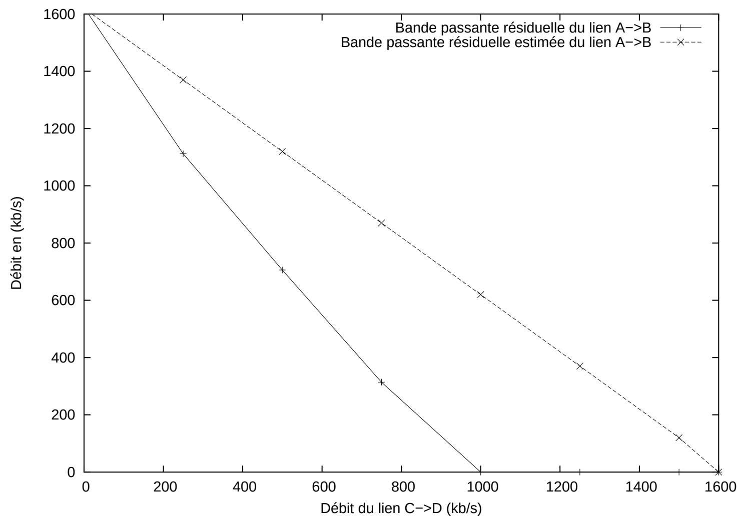


Figure 2: Available bandwidth in the unfair scenario

decrease the throughput of the link (A,B), and then need to be considered in the link's available bandwidth estimation. Therefore, it is essential to provide a mechanism to predict the time lost in collisions on the receiver side in order to refine the estimation.

To conclude this section, we see that there is still a need to improve the estimation techniques in order to provide accurate evaluations.

3 An accurate available bandwidth estimation

The purpose of this section is to present in details how we estimate the available bandwidth of a link based on the previous discussion. In our proposal, we combine three approaches:

- A time-based listening approach to estimate local and interfering used bandwidth by monitoring the channel utilization. This approach only allows an evaluation of the bandwidth available to a node.
- A probabilistic evaluation of the overlap of the silence periods of the two end-points of a link. Measuring this overlap is very difficult in practice as it requires time synchronization between the two peers and additional communication, that's why a probabilistic approach seems more suited.
- An estimation of the collision probability on links.

These two last estimates require to exchange bandwidth-related information between nodes. This exchange does not rely on dedicated probe packets but can be included in broadcasted packets like *Hello* messages found in many routing protocols. Therefore, the method we propose can be classified as passive and not intrusive.

In [12] we have proposed a method to evaluate a node's available bandwidth, which is then derived into a link's available bandwidth by a probabilistic estimation of the overlapping of the silence periods of both emitter and receiver. In the following, we quickly summarize the principles of this method.

3.1 Estimating a node's available bandwidth

During an observation interval Δ, each node can monitor the radio medium in its surroundings and measure the total amount of time that would be idle for emitting frames. Each node only considers the idle periods longer than IEEE 802.11's $DIFS$, as shorter periods do not allow medium access. Note that this method not only takes into account the bandwidth used in the transmission range of the nodes but in the whole carrier sensing area, as the medium is considered busy as soon as a signal above the carrier sensing threshold is received. Note also that such an estimation determines an upper bound on the amount of data that a node can transmit as the evaluation neither takes into account the IEEE 802.11 backoff period, nor the reception side of the transmission.

3.2 Estimating a link's available bandwidth

3.2.1 Taking into account the overlapping of silence periods

In [12], we have proposed a method to derive the available bandwidth on a link from the available bandwidth computed per node as described previously. In this method, we propose to use a probabilistic estimation of the overlapping of the silence periods. Such an estimation requires the exchange of the local available bandwidth between the neighboring nodes. This can be done via *Hello* messages. [12] concludes that this method improves the evaluation, but is still inaccurate in some configurations. In the following, we enhance the estimation based on the evaluation method of [12].

3.2.2 Taking into account collisions

The method presented in the previous section computes an estimation of the overlapping of the silence periods of both emitter and receiver in a probabilistic manner. Therefore, a flow emitting frames at a rate equal to this computed available bandwidth on a link cannot have the guarantee that this throughput will be achieved. It is possible that when a packet is emitted, the receiver available to receive it, leading to a collision. Such collisions lead to retransmissions and to an increase of the contention window, which reduces the real throughput of the flow.

For example, the method of [12] suffers from the same limitations as BRuIT, CACP or AAC on configurations like the one depicted by Figure 1. The difference between evaluated and real available bandwidths comes in this case from collisions at node B. Therefore, to provide an accurate estimation, we need to evaluate the collision probability at the receiver side of the link.

As we will use *Hello* messages to exchange information on the available bandwidth per node, we can compute a collision probability based on these *Hello* messages, denoted by p_{Hello}:

$$p_{Hello} = \frac{\text{number of collided } \textit{Hello} \text{ packets}}{\text{number of expected } \textit{Hello} \text{ packets}} \tag{1}$$

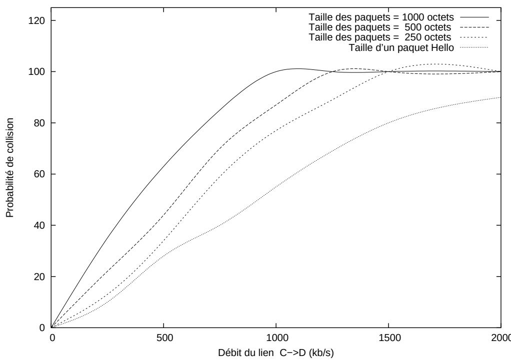
To compute these values, we consider that each node periodically sends a *Hello* message and that the period is identical and known. As soon as a node receives a *Hello* message from one neighbor, it can deduce how many *Hello* packets it should receive from this neighbor during the next measurement period. This value corresponds to the "number of expected *Hello* packets". The "number of collided *Hello* packets" correspond to this expected value minus the number of *Hello* packets actually received during the considered measurement period.

Considering the number of frames that should have been received in a time interval may mix congestion-related effects with collision-related losses. However, when a sender does succeed in sending as many *Hello* packets as it should due to an overloaded medium, the links associated to this sender will have a low available bandwidth and the inaccuracy in the computation of p_{Hello} will not have a strong impact on the evaluation. Moreover, the introduced error will imply an underestimation on the link, which is preferable to an overestimation. Another strategy could be to rely on sequence numbers of *Hello* packets to infer the collided packets, nevertheless, it does not give any indication when a hello packet is not received during a consequent time. These two strategies will be compared in future work and could be used side by side.

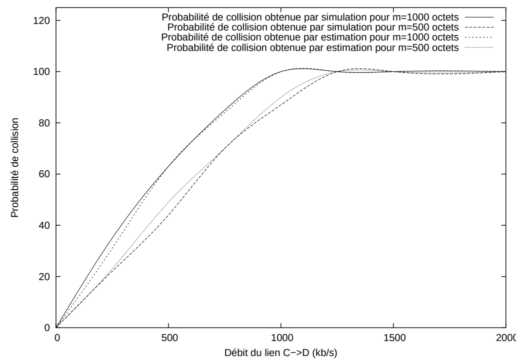
Hello packets have small and constant size. In consequence, p_{Hello} does not reflect the collision probability that may be applied to larger data packets. To address this issue, we extend the measurement by computing the Lagrange interpolating polynomial, $f(m)$, fitting the data so that: $p_m = f(m) \cdot p_{hello}$, p_m being the collision probability on packets of m bits.

When the RTS/CTS mechanism is used, the collision probability depends on the size of the RTS frame but not on size m, because if the RTS frame fits into the silence interval of C, the medium will be reserved for the time to transmit the packet of size m. The *Hello* messages we use only contain the idle time fraction computed by the nodes. Therefore, the size of *Hello* messages is of the same order as the size of RTS packets. Thus we consider that the collision probability in case of the use of RTS/CTS usage is equal to p_{Hello} and no interpolation is needed.

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



(a) Collision probability on B by simulations for different packet size



(b) Collision probability estimated on B for different packet size

Figure 3: Collision probability on B

Figure 3(b) represents the estimated collision probability at node B, using $f(m)$, for different packet sizes. It shows that the collision probability estimated is almost equal to collision probability measured by simulations for packet sizes of 1000 bytes and 500 bytes.

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 side. We have presented here a method to evaluate the collision probability that combines an on-line approach to evaluate the impact of the distribution of the medium occupancy at the receiver side via p_{Hello} and an off-line approach to take into account the packet sizes.

3.2.3 Taking into account the backoff

When a node experiences a collision, it doubles its contention window. Until now, we have considered the time wasted in collisions but not the additional backoff time introduced by this mechanism.

First, let's consider that there is no collision. Then the backoff follows a uniform law in the interval $[0; CW_{min} - 1]$ so the backoff can be approximated by the value $\frac{CW_{min}-1}{2}$. We will use the following notations:

- K is the proportion of extra time introduced by the backoff without collision.
- $backoff$ is the average value for the backoff time equal to $\frac{CW_{min}-1}{2}$.
- $DIFS$ ($SIFS$ resp.) is the DIFS (SIFS resp.) time defined by IEEE 802.11.
- $T(m)$ is the time separating the emission of two consecutive packets. This delay depends on the sending rate and of the packet size m.

The proportion of extra time K introduced by the waiting process is given by the formula:

$$K = \frac{DIFS + backoff}{T(m)} \tag{2}$$

The K value decreases as the packet size increases. In some cases (with small packet size for instance), not taking into account the K factor in the bandwidth estimation leads to an inaccurate estimation.

When collisions happen, exponential backoff mechanism is triggered. After each unsuccessful transmission, the contention window is doubled up to a maximum value called CW_{max}, therefore the average backoff increases above $\frac{CW_{min}-1}{2}$.

However, it can be shown that the time spent in backoff time when the contention window increases can be neglected compared to the time spent in collision.

Let's use the following notations:

- $T_{backoff}$ is the total time spent in backoff wait during the measurement period when the contention window is larger than CW_{min}.
- $T_{collision}$ is the total time wasted due to collision.
- $\alpha = \frac{T_{backoff}}{T_{collision} + T_{backoff}}$ is the proportion of extra time introduced by the backoff due to collisions.

To evaluate the value of α, we consider a scenario in which mobiles are randomly positioned on a area of $1000\,m \times 1000\,m$. We increase both the network load and the number of nodes and compute the value of α. Results presented on Figure 4 are the average of 30 simulations.

We can notice that, in the worst case, the exra time induced by the backoff hardly represents more than 6 % of the collision-related overhead. Hence, the major part of this extra time is wasted by the colliding packets transmission. Moreover, when the network load increases, the value of α decreases until it becomes almost null. These evaluations have been performed for a 2 Mb/s data throughput to be consistent with subsequent simulations. Results for 11 Mb/s throughput are similar, the upper bound for α increasing to 11 %.

Therefore, it seems not necessary to take into account the exponential backoff time because the major part of wasted time is consumed by collisions.

To conclude, our method gives the following formula for the link available bandwidth beetween two neighbor nodes s and r:

$$\boxed{E_{final}\left(b_{(s,r)}\right) = (1-K)\cdot(1-p_m)\cdot E\left(b_{(s,r)}\right)} \tag{3}$$

where E($b_{(s,r)}$) is the expected available bandwidth on the link (s,r) computed with the method of [12].

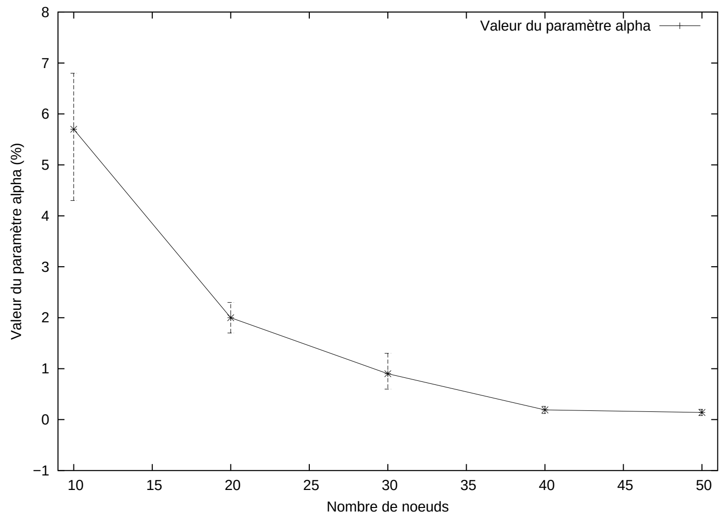


Figure 4: Proportion of extra time induced by the backoff

4 Simulations

In this section, we evaluate our estimation and compare it with some passive approaches described in Section 2. These solutions are included in more general protocols and especially in QoS routing protocols. We have considered BRuIT, QoS-AODV and AAC, available on the web[1]. As it is quite tricky to extract the available bandwidth estimation part from these protocols, we implemented a QoS routing protocol based on our bandwidth evaluation. The goal of this work is not to design a new QoS routing protocol with specific features, but rather to use a routing protocol having a behavior similar to BRuIT, QoS-AODV or AAC. With such a protocol, we can compare the performance of the different protocols, but we can also study the impact on our estimation technique on the bandwidth management in the network.

In Section 4.1, we quickly present the routing protocol we use, and noted henceforth ABE for *Accurate Bandwidth Estimation*. In Section 4.2, we present the performance results we obtained with ABE.

4.1 ABE

Available bandwidth estimation To derive a protocol from our solution, neighboring nodes will exchange the available bandwith computed locally via *Hello* messages. Every Δ seconds each node locally estimates its medium occupancy and includes this information in

[1]BRuIT: `http://citi.insa-lyon.fr/~iguerinl/QoS.html` – QoS-AODV and AAC: `http://www.ctr.kcl.ac.uk/members/ronan/default.asp`

a modified *Hello* packet. The accuracy of the bandwidth evaluation depends on the value of Δ, which is the time interval between two consecutive measurements. The larger Δ is, the stabler are the measurements. However, Δ should be small enough to allow fast reactions to load variation and to nodes mobility. We choose $\Delta = 1$ second, i.e. two times faster than OLSR's *Hello* messages period.

A node R that receives a *Hello* message from a node S estimates the available bandwidth of a link (S, R) using Equation 3. It is important to note that R only computes the available bandwidth on the link (S, R). This order in the knowledge is used in the routing process.

QoS routing The routing protocols used by BRuIT, QoS-AODV and AAC are all AODV-like. Therefore, we have kept the spirit of AODV for the routing part. We use the same broadcasted route request message (*RREQ*) to search for a route towards a specific destination. The source broadcasts the *RREQ* to its neighbors that retransmit it, and so on flooding the network. This message contains the address of the sender, the requirement at the application level, the destination address and a sequence number. Each intermediate mobile that receives a *RREQ* performs an admission control by simply checking whether the bandwidth asked for in the *RREQ* packet is inferior to the available bandwidth of the link on which the *RREQ* has been received. In case of success, the node add its own address to the route and forwards the *RREQ*, otherwise it discards it. When the destination receives a *RREQ*, it also checks resources availability. The destination then sends an unicast route reply (*RREP*) to the initiator of the request along the reverse path to ensure that mobiles along the reverse path are still reachable. The resources are then reserved and the new flow is then emitted.

The emissions of a node on a route having an impact on the available bandwidth at the following node on the route. In ABE, to consider these intra-flow contention problems, we use the relationship between the end-to-end throughput and the number of hops presented in [7].

ABE detects a broken route by monitoring the *Hello* messages. If a node doesn't receive a *Hello* packet from a node within a certain time interval, it sends a route error (*RERR*) message to the source that tries to recompute a route. Moreover, when a mobile sees that the network conditions do not allow the data transmission with the desired throughput anymore, it also sends a route error to the source node.

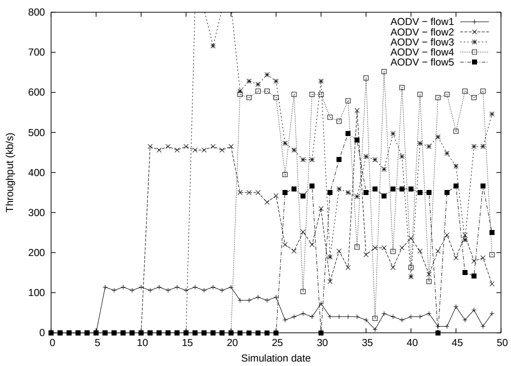
4.2 Performance evaluation

We have evaluated ABE using the NS-2 simulator (version 2.27). We have used the IEEE 802.11 implementation provided with the simulator. For all scenarios, the physical rate is 2 Mb/s, packet size is 1000 bytes and the radio transmission range is 250 m while the carrier-sensing range is 550 m. We compare the performance of ABE to AODV, QOS-AODV, AAC and BRuIT.

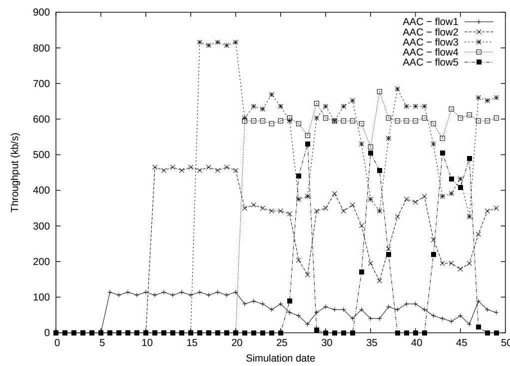
4.2.1 Random topologies

To evaluate the different protocols, we have generated random topologies with random flows (random source, random destination and random throughput). For any of these protocols, similar scenarios (same number of nodes and same number of flows), every lead to similar results. Therefore, for random topologies, we only present two specific scenarios. For each scenario, the results are the average of 30 simulation runs with different random seeds.

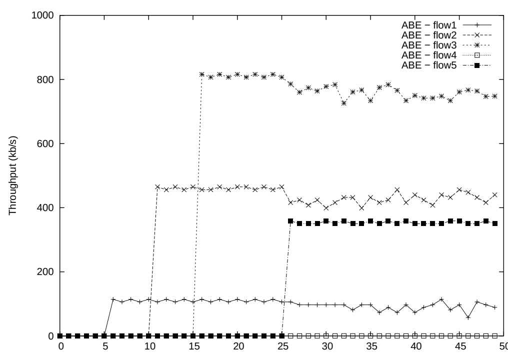
One-hop flows Let us first consider only one-hop flows. In these configurations, there will is no intra-flow contention. We consider a static network composed of 10 randomly positioned nodes and five CBR flows. The simulation duration is set to 50 seconds and, at the beginning, one flow starts every 5 seconds.



(a) AODV



(b) AAC



(c) ABE

Figure 5: One-hop flows throughputs with AODV, AAC and ABE

Figure 5(a) represents the simulation results for AODV, AAC and ABE. BRuIT and QoS-AODV exhibit a behavior similar to AAC in these scenarios and are therefore not represented here. Performing no admission control (AODV) leads to a congested network when new flows are added and achieved throughputs are degraded.

With ABE all flows except the fourth, that the admission control procedure has rejected, are admitted without any throughput degradation.

AAC overestimates the available bandwidth for the admission control of the fourth flow. Therefore, as soon as it is accepted the throughputs of the already existing flows begin to decrease. Finally the fifth flow is also accepted and the throughputs are decreases by 65 % for the fifth flow and up to 30 % for the three first ones, compared to ABE.

Multi-hop flows For this simulation, we consider a topology composed of 20 nodes randomly positioned. 7 CBR connections are established with random throughputs. A routing mechanism is necessary as the sender and the receiver of each flow may not be neighbors. Simulations last 50 seconds and the starting of flows is scheduled every 5 seconds from the start.

Figure 6(a) shows the throughput of the seven flows when AODV is used, i.e. no admission control is performed. The network becomes rapidly congested and routes are often broken, leading to a poor performance.

Figures 6(b) and 6(c) depict the flows throughputs with AAC and QOS-AODV. These figures show that these protocols admit more flows than they should leading to a congested network. AAC does not consider collisions while QOS-AODV neither considers collisions nor intra-flow contention problems. Therefore, AAC and QOS-AODV often overestimate the available bandwidth in the network. These results confirm that collisions and intra-flow contention are crucial factors for the available bandwidth estimation.

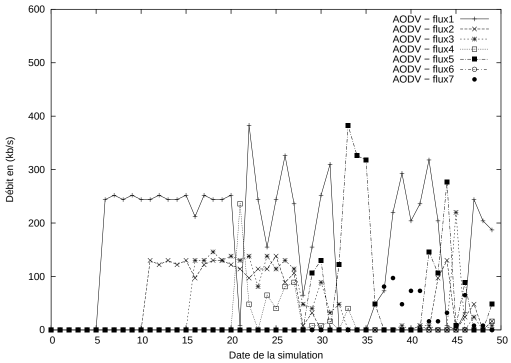
Figure 6(d) shows the throughput achieved by ABE which performs a more accurate admission control by admitting only three out of the seven flows. The admitted flows throughputs meet their bandwidth requirements. This scenario indicates that the estimation we perform does not over-estimate available bandwidth.

Finally Figure 6(e) presents the results obtained with BRuIT. Only the first flow is admitted, which means that BRuIT underestimates the available bandwidth. This is due to the fact that BRuIT is pessimistic and does not take into account the fact that some distant emissions can be performed in parallel.

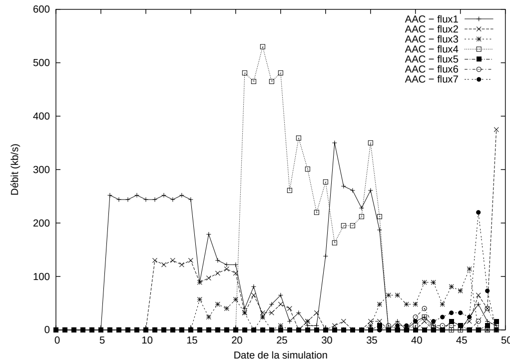
4.2.2 Overhead

Routes establishment and reconstructions require many messages exchanges (*RREQ*, *RREP* and *RERR*). We evaluated the overhead introduced by ABE, which gives an indication on the convergence speed of the routing mechanism and on its stability. We simulated networks of 10 to 50 nodes random networks, with 10 CBR flows. Throughputs of CBR sources are randomly distributed between 10 kb/s and 80 kb/s.

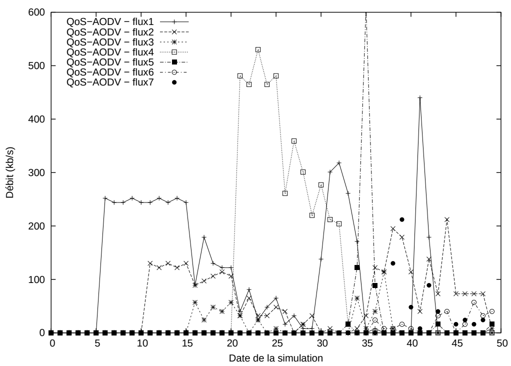
Figure 7 represents the total number of control messages exchanged for the routes setup and maintenance, as a function of the number of nodes in the network. AAC and QoS-



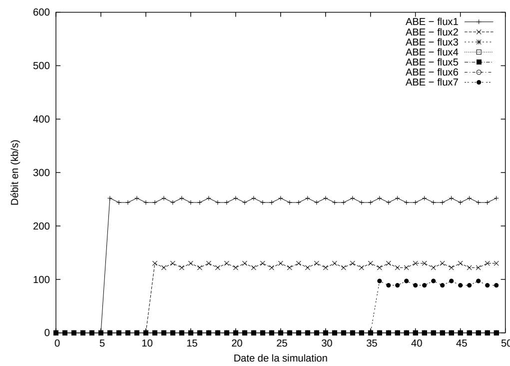
(a) AODV



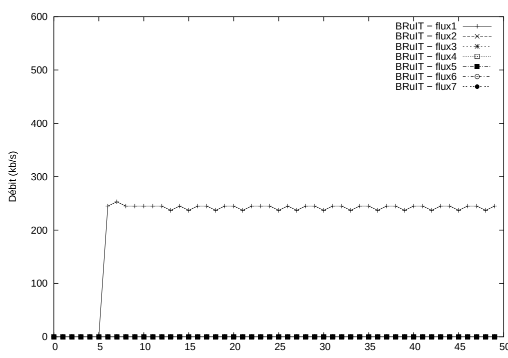
(b) AAC



(c) QOS-AODV



(d) ABE



(e) BRuIT

Figure 6: Multi-hop flows throughputs with AODV, AAC, QOS-AODV, ABE and BRuIT

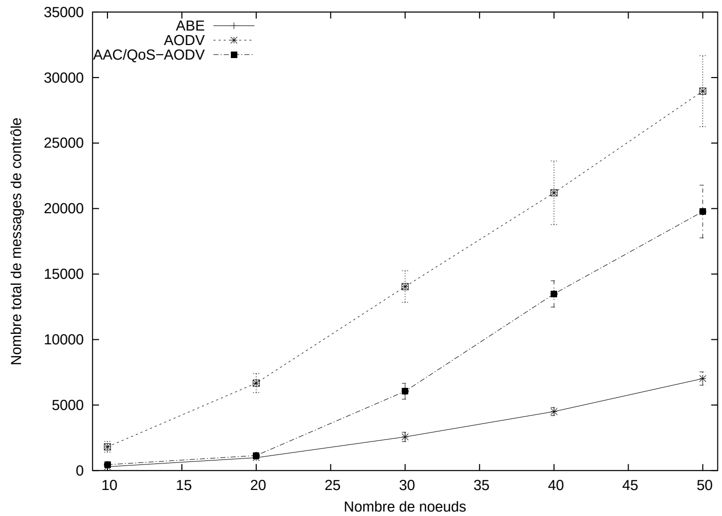


Figure 7: Routes establishment and maintenance-related messages overhead

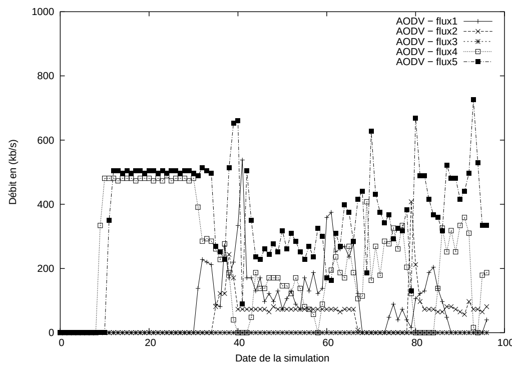
AODV generate similar overhead as they are based on a similar protocol. The overhead introduced by BRuIT is not represented on these figures as the tested implementation does not perform route reconstruction when traffics are not accepted or when routes are broken, leaving this task to the application. The results are therefore not comparable.

AAC and ABE generate less overhead than AODV as the admission control phase eliminates routes that do not have enough available bandwidth to carry a flow. Hence, the admission control reduces significantly the number of route request packets forwarded in the network. Moreover, when the network becomes congested, AODV suffers from a lot of routes breakage and performs route reconstruction. These added requests increase the overhead amount. ABE generates a slightly lower overhead than AAC because as AAC overestimates the available bandwidth, it forwards more route requests during the route discovery phase.

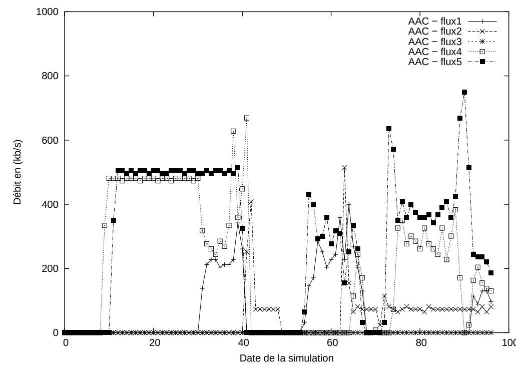
4.2.3 Mobility

It is hard to provide hard QoS guarantees when nodes are mobile. QoS violations can appear due to the changes in topology, which results either in route breakage or in decreased achievable throughputs. With ABE, both problems trigger a route error message sending to the source that then recomputes new routes. This search process may take a long time and generate extra message overhead.

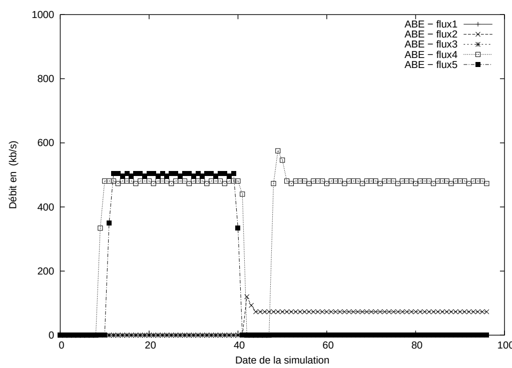
To investigate the effect of mobility on flows throughputs, we have performed simulations with 10 nodes randomly positioned. Five CBR traffic are generated with random throughputs and the starting dates of flows are spaced by two seconds. Nodes move according to a random way point mobility model with a maximum speed of 20 m/s and a pause time of 10 s. The simulations last 100 s.



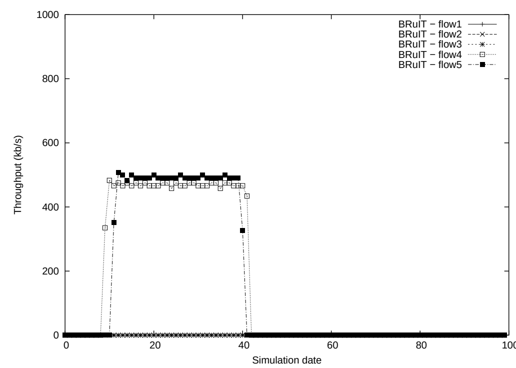
(a) AODV



(b) AAC



(c) ABE



(d) BRuIT

Figure 8: Throughput obtained by AODV, AAC, BRuIT and ABE in mobile scenarios

Figures 8(a) and 8(b) show that when nodes are mobile, both AAC and AODV lead to severe throughput degradations. With BRuIT (Figure 8(d)), the underestimation already pointed out before still holds and only two QoS flows are admitted. Moreover, BRuIT does not succeed in recomputing routes for flows 4 and 5. Figure 8(c) shows that despite mobility, ABE sends flows with their specified bandwidth requirements. In this scenario, 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 see that the introduction of pause times during which flows search for new routes with enough available bandwidth to meet their requirements allows some other flows to be routed.

5 Conclusion

In this article, we have presented an improved technique to compute the available bandwidth between two neighboring nodes and, by extension, along a path. The estimation leads to more accurate results than previous solutions by estimating the collision probability that packets will experience. Through simulations, we show the accuracy of our available bandwidth measurement that has been integrated to a reactive routing protocol for comparison purposes. Simulations show that flows are routed according to their throughput requirements even in the case of mobility.

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Unité de recherche INRIA Rhône-Alpes
655, avenue de l'Europe - 38334 Montbonnot Saint-Ismier (France)

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