Interference based Call Admission Control for Wireless Ad Hoc Networks

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Abstract

In this paper, we propose a call admission control mechanism for Wireless Ad hoc Networks called interference-based call admission control (iCAC). iCAC is unique in that it does not treat interference uniformly instead classifies interference based on estimates of the position of the interfering nodes. iCAC relies on two novel techniques: (1) estimation of position of the interfering nodes (2) fair allocation using bandwidth acquisition and rate control. By incorporating these techniques, iCAC is able to increase the estimated available bandwidth substantially without overloading the network. We compare iCAC with Perceptive Admission Control (PAC) [2] and IEEE 802.11 without admission control. Simulation results show that iCAC is able to admit substantially more requests than PAC, achieves more than 80% of the throughput of IEEE 802.11 and at the same time maintains very low packet loss rate and average delay comparable to PAC.

1 Introduction

Wireless Ad hoc Network is an infrastructure-less, adaptive and self-organizing network. Such a network can support both best effort and constant bit rate applications. In any case, in order to obtain good network performance, some form of admission control coupled with rate control can be desirable since traffic imbalance in any part of the network can lead to localized congestion, resulting in excessive packet loss and high packet delay. Majority of existing unfairness and performance problems in 802.11 networks can also be attributed to many interfering nodes transmitting at very high rate.

Generally, call admission control should ensure that in accepting a new flow, performance of on-going flows will not be affected. A good admission control scheme is one which admits as many requests as possible without compromising the performance of existing requests. An overly conservative scheme maintains good performance by admitting far too few flows, while an overly optimistic scheme allows all requests to be admitted without any regard to the performance of existing flows.

Call admission control (CAC) has been extensively studied for wired networks. While admission control solutions have been proposed for wireless ad hoc network, most of them are solutions for wired network ported for ad hoc networks. Inherent features of wireless ad hoc networks such as multiple access interference from nodes in the transmission and sensing range, change in topology, existence of multiple hops, etc make CAC in wireless ad hoc networks a rather difficult task.

In this paper, we present an interference-based call admission control (iCAC) protocol which provides admission control for flows in a single-channel (IEEE 802.11 [4] based) ad hoc network, which uses features such as radio state and noise levels at active nodes for computation of available resources. iCAC is unique in that it does not treat interference uniformly but instead classifies interference based on estimates of the position of the interfering nodes. iCAC relies on two techniques: (1) estimation of position of the interfering nodes, (2) fair allocation using bandwidth acquisition and rate control. Nodes continually update their knowledge of local resources and carry out re-computations whenever necessary. iCAC is performed *by all nodes* locally and the admissible rate for each flow can change with the arrival (or departure) of flows within the neighborhood. Our simulation results show that iCAC is able to increase the estimated available bandwidth substantially compared to previous work while maintaining low average delay and packet loss rate.

The rest of the paper is organized as follows: Section 2 presents the related works. Section 3 describes position estimation of interfering nodes. Section 4 describes our fairness notion, and explain how rate control is used to achieve fairness. Section 5 presents the iCAC algorithm. Section 6 describes the overall admission control process and provides the evaluation of the various call admission control schemes. Finally, in Section 7, we conclude our work and discuss future extensions.

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2 Related Works

In this section, we present related work which considers resource management through call admission control for wireless ad hoc networks.

CAC for wireless networks have been studied extensively. Typically, call admission is carried out by having a measure of how much resource has been used and how much resource is available for a new user. The key concept in these admission control algorithms is how resource availability or network utilization is measured. The various proposals for measurement usually involve one or more of the following parameters: available bandwidth, throughput, transmission delay, queue length or load at the node, collisions, power control, and signal to interference ratio.

A common approach to estimate the available bandwidth measurement is to measure the channel busy time [1, 2]. Let T be the sampling time-window, T_{idle} be the duration for which radio is in idle state in the last time-window T, T_{busy} be the duration for which radio is in busy state in the last time-window T and W be the maximum bandwidth. ABW, the available bandwidth can be computed as follow.

$$ABW = \frac{(T - T_{busy})}{T} * W \tag{1}$$

The authors in [6] measured the throughput of transmitting packet as: $Th = S/(t_r - t_s)$, where S is the size of the packet, t_s is the time-stamp at which the packet is ready at the MAC layer, and t_r is the time-stamp that an ACK has been received. They claim that the time duration $t_r - t_s$ includes the channel busy and contention time. They maintain separate throughput estimates for each and every neighbor. This throughput measurement is assumed to reflect the available bandwidth for the new flow. For this technique, it is important to make the throughput measurement independent of size of packet by normalizing the packet sizes.

Proposal by Sun et. al. [3] considers both the load at each node and predicted delay values to measure the network utilization, and used this information to carry out admission control mechanism. Each node maintains a neighbor set, which has the load information about each neighbor. Load information is in terms of number of service flows, and is also associated with *confirmed, pending, and unknown* states. When a request for new flow comes, based on this information of all neighbors, a node will predict the delay value. There have been proposals of constraint based routing which consider the load at each node. Further, SWAN [5] also considers load at each node for admission control decision.

Measure of average collision ratio is another technique used to estimate the network utilization. Similar to bandwidth measurement, a sampling period T is maintained. The average collision ratio (R_c) is defined as the number of collisions occurred over the total number of transmissions (including retransmissions). Therefore, $R_c = N_c/N_t$, where N_c is the number of collisions, N_t is total number of transmissions. The collision ratio indicates how much the network is loaded. Typically thresholds are set for the collision ratios and the admission decision is taken depending on these threshold values.

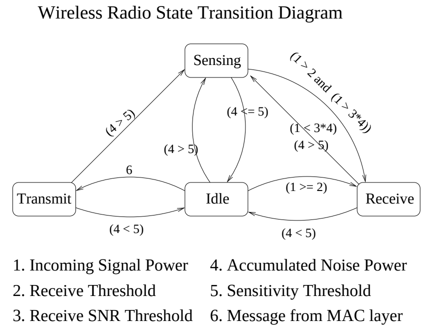


Figure 1. Radio State Transition

In this paper, we consider network utilization by estimating the bandwidth available by using the measure of busy times. However, we differ in the busy time definition from other approaches, and consider two versions of busy times, which will be explained in succeeding section. Along with busy time measurements, we also measure the noise values at both sender and receiver to estimate the position of interfering nodes, which would also help in available bandwidth estimation. This estimation of position of interfering nodes for available bandwidth measurement has not been used by any of the previous approaches. We also consider the aspect of fairness in admission control scheme, which is rarely considered by previous approaches. There have been various works ([8, 12, 11]) which consider interference and fairness in ad hoc networks. However, they achieve at the MAC layer usually modifying the scheduling mechanism.

3 Radio State and Bandwidth Model

3.1 Radio State Transition

Before explaining how the bandwidth estimation is performed, we need to explain the radio state transition, as shown in Figure 1, which is derived from the GloMoSim [7] simulator radio layer implementation.

The radio initially starts with the *Idle* state, from which it can either go to the *Receive*, *Transmit* or *Sensing* state. When a signal sent by any of the neighboring nodes arrives, it compares the signal power with the receive threshold (minimum power for the packet to be received). If the incoming signal power is greater than the receive threshold, it moves to the *Receive* state, or else it accumulates this power value as the noise value. If this accumulated noise value is greater than the radio sensitivity threshold (minimum power for a packet to be sensed), then the radio moves from the *Idle* state to the *Sensing* state. From the *Idle* state, if the node receives a message from upper layers to transmit, it moves to the *Transmit* state. Radio can change its state from *Sensing* to *Receive* if the incoming signal power is greater than both the receive threshold, and SNR threshold times the accumulated noise. However, the state changes back from *Sensing* to *Idle*, if the accumulated noise is less than the sensitivity threshold.

After the transmission, radio changes its state from *Transmit* to either *Idle* or *Sensing*, depending on whether the accumulated noise is lesser or greater than the sensitivity threshold, respectively. Similarly, radio state can change from *Receive* to either *Idle* or *Sensing*, under the same conditions.

A key observation in this radio transition model is that the power of interference and noise is calculated as sum of all the signals arriving at the radio, other than the one being received, and adding it with thermal noise. The resulting power is used as the base of SNR, which determines the probability of successful reception of the signal. The noise and interference model used can be mathematically written as [7]:

$$SINR = \frac{P_{incoming}}{(\sum P_{othersignals}) + (F * K * T * B)}$$

Where K is the Boltzman constant, T is the temperature, F is the noise factor of radio and B is the effective noise bandwidth.

As a result, it is possible for two flows within interference range to transmit at an aggregate throughput much higher than if the flows are within transmission range with very low probability of packet corruption due to noise.

3.2 Measuring Available Bandwidth

With this understanding on the radio state transition and signal reception model, we now describe the bandwidth measurement with sensing as idle (BSI).

As explained earlier, measuring the radio busy duration is the popular approach for measuring the available bandwidth. In our work, we also take the busy duration measurement approach, as given by equation 1 for bandwidth measurement.

All the earlier proposed schemes on bandwidth measurement using the measure of busy times consider *sensing state the same as receive state*. Therefore, *the sensing period is considered part of the busy period along with receive and transmit states*. However, such assumption is highly conservative as sensing state is not the same as receive state, and if more detailed classification is performed, the available bandwidth can be increased substantially.

In our work, we have two available bandwidth measurements. We called the available bandwidth measure using equation 1, and considering sensing, receive and transmit as busy states, as *BSB* or *Bandwidth considering Sensing as Busy*. In addition, we also consider BSI or *Bandwidth considering Sensing as Idle*. *We use the same equation to measure the BSI but the busy periods include only transmit and receive.* It should be noted that BSB provides the lower bound on available bandwidth, whereas BSI provides the upper bound. Depending on the other measurements to be presented, available bandwidth is somewhere in between these two values (BSB and BSI). It is to be noted that in previous approaches the available bandwidth is always BSB.

Besides providing an upper bound on the available bandwidth, BSI plays a very important role in providing some hints on the position of the interfering nodes and can be used to improve the available bandwidth estimate. For example, let MAX be the maximum bandwidth available. If $BSI = MAX$, and $BSB < MAX$ all the interfering nodes are outside the transmission range. On the other hand, if $BSB = BSI$ and the measuring node senses very little noise, all the interfering nodes are within the transmission range.

In addition, if BSI is y and BSB is x, where y and x are less than MAX, then we know that the bandwidth $y - x$ is consumed by the interfering nodes outside the transmission range. If an interfering node is within the transmission range, then the signal received from the interfering node will be greater than the $receive\_threshold$, therefore both BSI and BSB will be less than MAX. On the other hand, if $BSI < MAX$, then there should be some interfering node within its transmission range, whose signal is greater than the $receive\_threshold$.

Based on the estimation of the position of the interfering nodes (when a node identifies that there are interfering senders outside the transmission range and within the interference range), we also include power-control technique to get the count of number of interfering senders outside the transmission range. We take advantage of the power control capabilities of todays wireless technologies. We allow the sender to increase the transmission power level of a "probe-packet"(only a single packet is sent) to cover the interference range. This probe-packet is used to get the number of senders outside the transmission range and within the interference range. We use this count in our fair sharing technique, which is described in succeeding section.

4 Model For Bandwidth Sharing

The overall available bandwidth computation is based on the concept of fair (or equal) share. Fair share ensures that

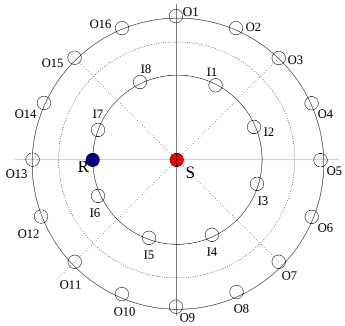


Figure 2. Topology

no admitted flow will be starved. In addition, a fair allocation has the advantage of encouraging better spatial reuse. Consider, for example, the simple case where 6 nodes (A-B-C-D-E-F) are arranged in a straight line. The distance between the nodes is such that neighbor nodes are within transmission range (e.g. A and B) and nodes one hop away are beyond transmission range but within interference range (e.g. A and C). Nodes two or more hops away (e.g. B and E) are beyond interference range. There are three active flows, flow 1 goes from A to B, flow 2 goes from C to D and flow 3 goes from E to F. Maximum bandwidth available is 2Mbps. If flow 2 is allocated 2Mbps by the admission control algorithm, then flows 1 and 3 will be starved. However, if a fair share of 1Mbps is given to all three flows, then the aggregated throughput is 3Mbps, a 50% increase. Of course, if flows 1 and 3 are allocated 2Mbps, the aggregated throughput will be 4Mbps but flow 2 will be starved.

In iCAC, each node of the route computes the fair share of the bandwidth available by estimating the number of active flows (*senders*) within its transmission or interference range, depending on the estimation of position of interfering nodes. If N flows are estimated to be within range, a flow i with weight r_i will receive a channel allocation of $C\frac{r_i}{\sum_{j\epsilon N} r_j}\delta t$ over some time window δt, where C is the channel capacity and N is the set of backlogged flows. The available bandwidth for a particular (multi-hop) flow is the minimum available over all hops taken.

iCAC admits as many flows as possible, as long as the allocation is greater than the minimum required bandwidth. That is, we define the following utility function:

$$Maximize\, F, such\, that,\ r_{f_i} >= MIN_{f_i}$$
$$and\, SUM(r_{f_i}) <= C, where\ f_i\ \in\ F.$$

where, F is the total number of flows admitted, $F \in N$, r_{f_i} is the rate allocated to flow f_i and MIN_{f_i} is the minimum rate required for flow f_i.

While the basic idea of fair sharing is simple, the implementation is not straight forward because in order to compute the correct fair share, the number of interfering sender and receivers, and their relative positions needs to be known. In our approach, using BSB, BSI and noise measurements, a node obtains the information of all the other contending nodes and depending upon the location of contending nodes, the node decides how sharing should be done. iCAC does not aim to provide any guarantees.

It is important to note that, in our scheme, when all the interfering flows (senders) are within the transmission range of either the sender or the receiver, we take advantage of the effectiveness of the RTS/CTS handshake mechanism. However, for all the other cases we need to have an estimate of number of existing flows, and based on this information we carry out the fair bandwidth sharing. The detailed description of the various cases considered are presented in the succeeding section.

5 Computing Available Bandwidth

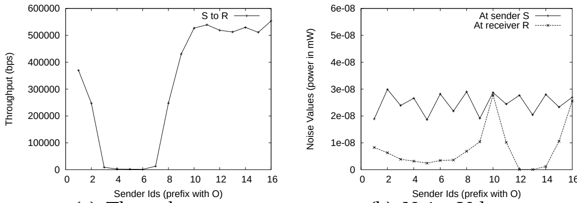
In this section, we will describe how iCAC estimates the number of active flows based on the values of BSB, BSI and noise measurements at sender and receiver. To simplify notations, we will refer to the set of all interfering flows as I_f, and the sender and receiver node as S and R respectively. In general, estimation and coordination is possible only when a sender is present. We identify positions of senders in I_f with respect to S and R, whereby estimation and coordination are effective, into the following categories:

- **Case 1:** All senders of I_f within transmission range of S.
- **Case 2:** All senders of I_f are beyond the transmission range and within the interference range of S.
  - **Case 2A:**All senders of I_f are within transmission range of R
  - **Case 2B:**Some sender(s) beyond the transmission range of R.
- **Case 3:** Senders are both inside and outside the transmission range of S (but within interference range).

It can be seen that the above listed cases are complete (it covers all possible cases of existence of interfering flows) and also that the effectiveness of RTS/CTS is limited to Cases 1 and 2A, whereas for cases 2B and 3 it is not effective.

5.1 Measurement of Noise Level at Sender and Receiver

In this section, we present noise measurement results that will be used in the following section for estimation. Two



(a) Throughput (b) Noise Values

Figure 3. Different Interfering pairs

sets of nodes are placed in a circular manner around S. The nodes that are within transmission and interference range are label as I_i ($1 \leq i \leq 8$) and O_i ($1 \leq i \leq 16$) respectively in figure 2.

Let the noise level at the sender node be denoted by NS and the noise level measured at the receiver node be NR. At any given time we consider only one interfering pair, and also any node with ID O_i transmits to its neighbor O_{i+1}. Figure 3(a) shows the throughput obtained for the pair (S, R), and figure 3(b) shows the noise values at Sender S and Receiver R. The x-axis shows the interfering sender id.

It can be seen that when the interfering nodes are farthest from the receiver, throughput is the lowest. In cases where the interfering nodes are out of the carrier sense range of the receiver R (O_3 to O_7), the throughput degrades significantly to almost zero. This is a very interesting case. The reason behind this behavior is that the control packet (CTS), that is sent by receiver R, experiences collision at sender S. This loss is due to the accumulated noise at the sender S. Though S sent the RTS first, as the interfering nodes are outside the transmission range of the sender, they will not be able to decode the RTS packet, and will not set their NAV. Further, as the interfering nodes are outside the sensing range of the receiver R, they will continue to send the CTS packets, which will be lost at the sender S. The reason for such low throughput is that since R does not sense the signal from the interference pair at all, the packet R sent has a high probability of collision during the reception by S.

For the positions (O_1, O_2, O_8 and O_9), R can sense the interference and can restrain from sending CTS, when the channel is busy. Hence, the S to R transmission can achieve higher throughput. Whereas, for positions O_{11} to O_{15}, R is within transmission range of the interfering pair and the RTS/CTS protocol works correctly, thus achieving the highest throughput.

Figure 3(b) shows the noise level at the sender and receiver, for different interfering pairs. When the noise level is strong enough or almost zero, the throughput is high as accesses are coordinated. However, when the noise level is below the detection threshold, as in the case of O_3 to O_7, interference is not taken into account and accesses are completely uncoordinated, resulting in very low throughput.

5.2 Case 1: All senders of I_f within transmission range of S

In this case, the senders of the interfering flows are within the transmission range of the sender S. The positions of the interfering receivers do not matter. It should be obvious that this case can be clearly identified by two conditions. First, the noise values at sender S is zero (because the interfering senders are within the transmission range). Second, both the bandwidth measurements should be equal and greater than 0 $BSI = BSB > 0$. Further, this is the case where the RTS/CTS scheme is most effective.

However, there are still two potentially interesting cases, where the I_f may be outside the transmission range of R (I_2, I_3), or inside the transmission range of R $(I_1, I_4, I_5, I_6 \ and \ I_8)$. In these two cases, BSB and BSI at sender will almost be equal. However, in the former case NR will be greater than NS. We carried out a detailed a study of both cases (results are not provided due to space constraint) and found that irrespective of I_f inside or outside the transmission range of receiver R, as long as it is within the transmission range of sender S, S and I_f can share the bandwidth equally. We allow new flow, which will share the bandwidth with other flows, through IEEE 802.11 RTS/CTS mechanism.

The available bandwidth estimation is achieved by continuously monitoring the number of neighbors who are senders, and setting the available bandwidth to $\frac{maximum-bandwidth}{number-of-senders+1}$.

5.3 Case 2: All senders of I_f are beyond the transmission range and within the interference range of S

5.3.1 Case 2A: All senders of I_f are within transmission range of R and outside the transmission range of S.

When the senders of I_f are beyond transmission range of S but within transmission range of R, the flow from S to R is at the mercy of the senders in I_f.

From figure (3(b)), we can see that there are two interesting cases where the noise level at the receiver R is zero or very low. The first case is where the receiver is out of the sensing range of the interfering pair, O_4, O_5 and O_6 in the figure 2. In this case, $N_S >> N_R > 0$. In the second case, the interfering pair is within the transmission range of the receiver and the RTS/CTS protocol will coordinate accesses $NS >> N_R = 0$ (O_{10} and O_{11} in the figure 2). This is the case where coordination is possible.

According to our BSI definition, we can see that this case can be identified by BSI = MAX (no interfering nodes

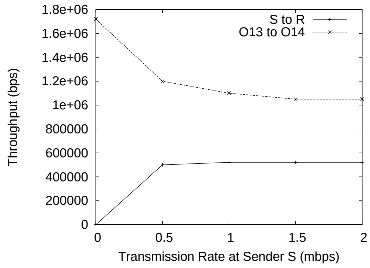


Figure 4. PDF with varying rate (Noise Values)

within transmission range of sender) and the noise level at the receiver R is zero (or very low). The case where only interfering receivers are present is not considered because coordination will not be effective.

As shown in the topology in figure 2, when O_{12}, O_{13} or O_{14} transmits, the noise at R will be null and S will be high. The throughput achieved by O_{13} and S with increasing rate at S is shown in figure 4. In the plot, O_{13} sends at maximum rate and transmission rate (sending rate) at S is varied. If both flows are allowed to send at the maximum rate, O_{13} will get a higher share of the bandwidth because R will receive RTS from both S and O_{12} and is therefore more restrained from replying to send CTS to S.

The fair share for this case is computed as setting the achievable bandwidth to (MAX/(number-of-senders in transmission range of R + 1)).

It it important to mention at this point that there is another scenario (all the interfering senders are beyond transmission range of sender S and their corresponding receivers within the transmission range of sender S) where RTS/CTS scheme is effective, and not necessary to carry out the probing. However, we found that this scenario is impossible to identify, as this scenario will not result in any unique parameter values (BSI, BSB or noise values). Therefore, we consider this scenario by probing, which is described in the succeeding section.

5.3.2 Case 2B: Some interference sender(s) are beyond the transmission range of R

This case considers the scenario where the interfering pairs are beyond the transmission range of sender S and there are some interfering sender(s) beyond the transmission range of R. The case is identified by $BSI = MAX$ and $NR > 0, NS > 0$. It should be noted that in the earlier two cases (Case-1 and Case-2A) we took advantage of effectiveness of RTS/CTS scheme. Whereas, in this and the following cases (Case-3), we consider scenarios where RTS/CTS scheme is not effective.

In this case, the sender sends out a probe packet (a very small packet with a single field) with increased transmission power, such that the packet reaches the nodes which are beyond the transmission range and within the interference range. We follow the technique proposed in [10] to determine the interference range to be used and corresponding transmission power.

Interfering nodes only beyond the transmission range of S node will respond to the probe packet, *if and only if it is an active sender*. This probing mechanism will result in sender S obtaining the information of the number of senders beyond transmission range and within the interference range (OSC). Complete implementation details of the probing mechanism are not provided due to space constraints. The “fair” allocation for S is computed as $\frac{MAX}{OSC+1}$.

5.4 Case 3: Nodes of I_f beyond and within the transmission range of S

In this case, the interfering flows are both beyond and within transmission range, making coordination, if not impossible, difficult. This case is identified when $BSI, BSB < MAX$ and $NS, NR > 0$. Even in this case, sender S carries out the probing technique, similar to Case-2B, to obtain the number of interfering senders beyond the transmission range of S or OSC. Further, we also have number of senders within the transmission range (SC). We achieve better sharing by setting the available bandwidth as $\frac{maximum-bandwidth}{SC+OSC+1}$. This setting ensures that no one flow takes the complete bandwidth share making other flows to suffer.

It is possible to trigger probe packets for all local bandwidth measurements. However, use of probe packets increases the complexity and consumes more energy. Hence, in cases where the number of interfering senders can be detected directly, for example, in Cases 1 and 2A, a simpler and efficient method is used.

5.5 Available Bandwidth Measurement Algorithm

In this section, we combine all the cases described before and present iCAC - an interference-based call admission control.

The admission control mechanism has four components: local bandwidth measurement, end-to-end bandwidth measurement, admission and rate-control, and bandwidth recomputation. Local bandwidth estimation is carried out by all nodes along the route by carrying out the algorithm explained in the preceding section. For end-to-end bandwidth estimation, the routing mechanism performs the task.

We modify just the Route-Reply (RREP) packet of DSR, with following fields- bottleneck bandwidth (BB) value, and noise-value (NV). The destination node, when it initiates the RREP message, adds its local bandwidth measurement into BB field and its noise-value into NV field, and sends to the next hop. The next hop when it receives this reply message, uses the NV value of the reply-packet to compute its local available bandwidth. It next checks if its local bandwidth value is lesser than the BB field of the packet, and if it is, it replaces the BB field value with its bandwidth value. Also, it replaces the NV value of the packet with its noise value. This process continues till the packet reaches the source node.

We also enhance the routing table information with the bottleneck bandwidth value, which is associated with each and every route it stores. Further, we include a neighbor management mechanism in DSR, which includes exchange of hello messages between nodes for every fixed duration (5secs). This message includes the status of the neighbor (whether the node is a sender or not), along with its ID.

The admission control mechanism decides if the required bandwidth is less than or equal to the bottleneck (minimum available) bandwidth of the route. If it is, then the call will be admitted, and if not, the call will be either blocked or the rate of the transmission is reduced. If the bottleneck bandwidth falls below some minimum value the call will be completely blocked.

Bandwidth re-computation is performed after a call is admitted, and is triggered on two conditions. First, when the number of senders among the neighboring nodes changes (increases or decrease), Second, when the noise value changes by certain fixed amount (increases or decreases). Note that in the current framework, a flow will not be given more than the minimum of its end-to-end fair share, which can under-utilize the network. However, we feel that fully utilizing all the bandwidth is too aggressive and ensuring that all bandwidth are assigned similar to the max-min assignment described in [8] requires multiple iteration and is too expensive in terms of messages required and admission control duration.

In the flowchart of our available bandwidth measurement algorithm as shown in figure 5, a sender node S on which this algorithm is run and a receiver node R is considered. In the flowchart, AB represents the available bandwidth, NS represents the noise value at the node Sender, whereas NR represents the noise value at the receiver. BSB and BSI are the same terms as explained in the previous sections. SC and OSC represents the number of senders within the transmission range and beyond the transmission range (within the interference range) of node S, respectively. m represents the number of senders with the transmission range of receiver R. Finally, MAX is the maximum available bandwidth. The detailed description of flow chart is excluded both due to space constraints, and majority of description is covered in preceding sections.

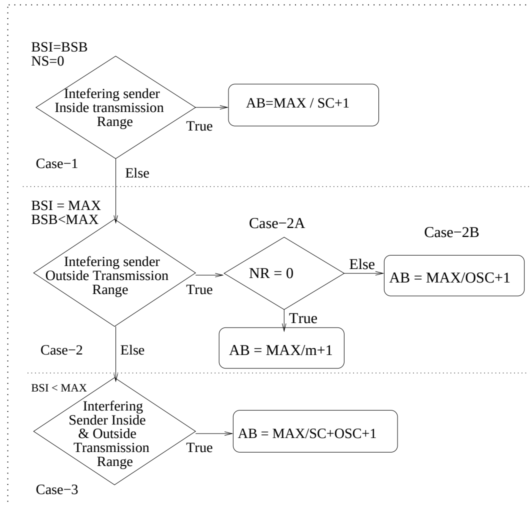


Figure 5. Flowchart of Available Bandwidth Measurement Algorithm

6 Evaluation of Admission Control Mechanism

In the simulation, we modified Dynamic Source Routing (DSR) [9] protocol to carry out end-to-end bandwidth estimation. DSR is used only because of the simplicity in implementation, and any reactive routing protocol can be used. Evaluations are performed for both single-hop and multi-hop scenarios.

We compare iCAC with Perceptive Admission Control (PAC) [2] and the IEEE legacy 802.11 [4] mechanism without any admission control process. PAC scheme depends on self estimation of the available bandwidth, which is performed by changing the range of the bandwidth measurement. This technique of enhancing the range and letting each node measure the bandwidth without the need for communicating with the other nodes. In PAC, the *Sensing Range* of the node is enhanced to the distance of $2 * RxR + RID$, where RxR is the Transmission/Reception Range and RID is the Receiver Interference Distance. RID is basically the distance between a receiver and any other sender, such that the corresponding receiver's ability to decode a packet from its sender is not affected. The authors of PAC believe that at any distance greater than $2 * RxR + RID$, two ongoing transmissions will not interfere with the packet receptions, and therefore a node can make decisions is its available bandwidth (by considering this large range) is sufficient to support new flow.

PAC is basically designed for single-hop networks, therefore in this section we consider topologies with single-hop ad hoc network. The evaluation of our scheme for multihops is provided in section 6.3.

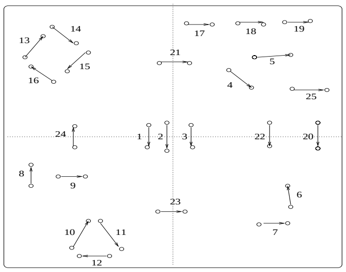


Figure 6. Simulated Topology for Fairness

All our evaluations are carried out on GloMoSim [7] simulator. All the parameters for PAC ($RxR = 250mts$, $RID = 440mts$, Sensing range = 940mts) are same as used in [2]. Each mobile host has a transmission range of 250 meters and shares a 2 Mbps radio channel with its neighbors. The simulation includes a two-ray ground reflection model and IEEE 802.11 MAC protocol. All the simulations are run for 200 seconds. A single hop network with 50 mobile nodes (25 pairs) is simulated. The network area is 2000m x 2000m. For all the pairs, the nodes are within the transmission range of each other, so that we can focus only on the effects of admission control scheme. Nodes move together, and this movement happens only 2-3 times. The link between the two nodes of a pair is always intact.

6.1 Single Hop Evaluation

We consider a traffic load with 25 flows. The source and destination nodes of each flow are within transmission range. We consider real-time UDP traffic with a packet size of 1460 bytes, with varying transmission rate (source sending rate). We consider three transmission rates 100kbps, 200kbps, 500kbps. The flow arrivals are 5 seconds apart. Therefore, after 125 seconds of simulation time all the sender-receivers pairs are active. It is to be noted that this evaluation part is similar to the one used in [2].

The results are summarized in the tables 1, 2, 3 and 4. The tables show the average end-to-end delay, number of calls admitted, number of packets delivered and packet losses respectively for the transmission rates of 100, 200 and 500kbps, for the three schemes- PAC, 802.11 and iCAC.

A good admission control scheme is one which admits as many requests as possible without compromising the performance of existing requests. A conservative scheme maintains good performance by admitting far too few requests and an optimistic scheme allows all requests to be admitted without any regard to the performance. We admit as many requests as possible according to our bandwidth sharing model described in section 4.

For a low load, all three schemes have similar performance, though iCAC has the lowest delay and loss compared to both PAC and IEEE 802.11. At a medium load, the performance gaps start to appear. iCAC admits slightly more requests than PAC, has low average delay and attains higher throughput. However, iCAC does have a small amount of loss. Compare to iCAC, the IEEE 802.11 scheme admits the same number of requests but has a much higher delay and packet loss.

In high load, iCAC admits almost twice as many requests as PAC (23 vs. 11) and only slightly less than IEEE 802.11 (23 vs. 25). In addition, in spite of the fact that the overall traffic load is much higher, by using a better local bandwidth estimation and rate control, iCAC is able to provide fairly low end-to-end delay, high throughput and low packet losses.

We carried out detailed simulations on the percentage ratio of number of times the probing technique is involved over total number of time the available bandwidth is computed. Due to space constraints we are not providing the plots. We found that as the flow-density increases, this percentage ratio increases. With 40 to 50 flows, the ratio is about to 70%.

Table 1. Average End-to-End Delay

	100Kbps	200Kbps	500Kbps
iCAC	7.8ms	8.5ms	0.206sec
PAC	8.0ms	11.5ms	0.105sec
IEEE 802.11	9ms	15ms	2.98sec

Table 2. Average Number of Calls Admitted

	100Kbps	200Kbps	500Kbps
iCAC	25	25	23
PAC	25	23	12
IEEE 802.11	25	25	25

Table 3. Average Number of Packet Delivered

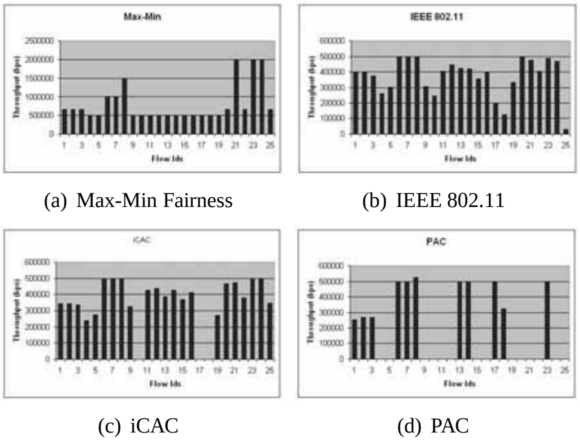
	100Kbps	200Kbps	500Kbps
iCAC	28905	56799	92476
PAC	28905	45633	45862
IEEE 802.11	28901	57696	108335

6.2 Fairness Evaluation (Single-Hop)

In this section, we evaluate how fair our call admission control algorithm is. For our evaluation, we also consider the fair allocation algorithm presented in [8]. This allocation algorithm computes the fair share allocation for every

Table 4. Average Number of Packet Losses

	100Kbps	200Kbps	500Kbps
iCAC	0	12	1021
PAC	0	0	346
IEEE 802.11	7	240	43290



(a) Max-Min Fairness (b) IEEE 802.11

(c) iCAC (d) PAC

Figure 7. Comparison of Flow Shares by various approaches

flow by considering the flow contention graph and building cliques out of the flow contention graph. The major drawback of this scheme is that, it considers only the transmission range of nodes in developing the flow contention graph and cliques, and does not consider the interferences and noises due to flows outside the transmission range, which would affect the transmission. Therefore, its estimation is highly optimistic and sometimes far from reality. However, we included it to see how our algorithm performs relative to this fair allocation.

We consider the similar simulation settings as previous sections. The simulation area is 2000m x 2000m with 50 nodes (25 pairs) randomly placed. Nodes are static and flow is single hop. Flow contention graph is developed for this topology and using algorithm 1 of [8], we compute the fair share for each flow. A total of 14 cliques with different degrees were formed. The allocation using this algorithm is shown in figure 7(a). For the other three algorithms, the traffic load per flow is 500 Kbps and the allocations for IEEE 802.11 without admission control, iCAC, PAC are shown in figure 7(b), 7(c) and 7(d), respectively. Table 5 summarizes the performance results (delay, packet loss, packets received, calls admitted) for IEEE 802.11, PAC and iCAC, for the scenario considered.

From figure 7(a), we see that the ideal max-min allocation without taking into account interference would accept all calls and at the same time provides at least 500 Kbps to all. However, once interference is taken into account, the bandwidth is much lower as indicated by figure 7(b), which shows the performance of IEEE 802.11 without admission control. Using IEEE 802.11, flow 25 gets very little bandwidth, average delay is more than 2 seconds and packet loss rate is more than 30%. With PAC, only 11 out of 25 requests are accepted. Out of these 9 requests, 4 of the requests have rates below 300 Kbps. Thus, the control is both too conservative and unfair. Finally, iCAC admits 22 out of 25 calls, and all requests admitted have more than 200 Kbps. The total throughput achieved is almost double that of PAC and within 83% of IEEE 802.11. Loss rate and delay are slightly higher than PAC but this is unavoidable since the throughput is much higher. Considering the topology in figure 6, and figure 7(c) we can see that whenever the flows are within the transmission range {for example, flows 1,2, 3 and flows 13,14, 15, 16 etc}, nodes tend to share the available bandwidth fairly.

Table 5. Fairness Evaluation of iCAC

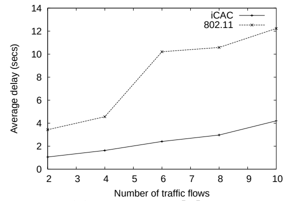
	Avg Delay (sec)	Total Pkt loss	Total Pkts Rcvd	No/of calls Admitted
iCAC	0.231	176	97833	22
PAC	0.136	20	47371	11
802.11	2.72	36115	118215	25

6.3 Multi-Hop Evaluation

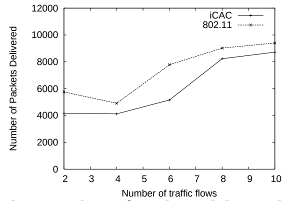
In this section we present the evaluation of our admission control scheme in multihop scenarios. The node capabilities are similar to the previous simulations. The simulation area is 1000m x 1000m, with 25 nodes. Random waypoint mobility model is used with a maximum speed of 4m/sec and with pause time of 50sec, which is a relatively slow moving scenario. We compare our admission control scheme with IEEE 802.11 without admission control scheme. We vary the number of traffic flows in the network from 2 to 10 flows. The source and destination are chosen randomly.

Figures 8(a), 8(b), 8(c) show the average delay, number of packets delivered and the number of packet loss with varying number of traffic flows. It can be seen from the figures that iCAC performs much better than the mechanism without admission control with respect to average delay and packet losses. However, the number of packets delivered are slightly lesser compared to IEEE 802.11 without CAC. The decrease in packets delivered is mainly due to flows that are blocked. This helps in reduction of delay and packet losses, which are crucial for real-time applications.

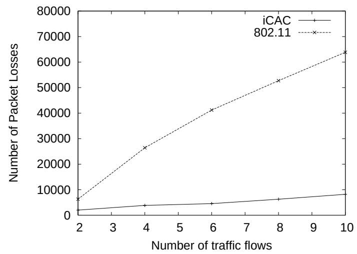
Similar to the single-hop evaluation, we also carried out a detailed simulation study on the percentage ratio of number of times the the probing technique is involved with respect to number of times the available bandwidth is computed. This ratio is approximately 65%.



(a) Average delay



(b) Number of packets delivered



(c) Number of packet losses

Figure 8. Performance of iCAC and IEEE 802.11 in Multihop Scenarios

7 Conclusion and Future Works

In this paper, we presented iCAC an interference based call admission control mechanism. iCAC is novel with respect to the bandwidth estimation technique it uses. Apart from traditional busy time measurements, where sensing state of a radio is considered as part of the busy period, we consider the busy time where sensing is taken as part of idle periods. Also, we consider the noise values at both sender and the receiver of the flow, and number of active senders around the node. These measurements help a node to estimate the position of the interfering nodes and accordingly estimate the available resource.

We enhanced DSR routing protocol to carry out end-to-end estimation, so that iCAC works for multihop networks with mobility. We further carried out a detailed evaluation of iCAC under both single and multihop scenarios. We also studied how fair iCAC performs when allocating the bandwidth share. We found that iCAC provides better performance in terms of delays and packet losses, which are crucial for real-time flows. iCAC even though cannot provide any fairness guarantees, provided fair shares considering the node positions.

Enhancing the fairness aspect of iCAC could be an interesting future work. Another important future work would be to study the performance iCAC across different mobility scenarios, to understand how multihop and mobility affects the measurements involved in iCAC.

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