Dynamic Behavior of Bandwidth Control Management in Mobile Ad-Hoc Network

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***Abstract*— Quality of Service (QoS) support in Mobile Ad-hoc NETworks (MANETs) is a challenging task. The major challenge in ad hoc networks lies in adapting multicast communication to environments, where mobility is unlimited and failures are frequent. Such problems increase the delays and decrease the throughput. To meet these challenges, authors have proposed bandwidth control management (BWCM) model to improve the QoS performance [9] by minimized end-to-end delay.**

In addition to end-to-end delay, an algorithm for end-to-end bandwidth calculation and allocation has been proposed in the paper. The system performance in various QoS traffic flows and mobility environments have been examined through ns2 simulator.

***Index Terms*— ns2, quality-of-service (QoS), DSDV, MANET.**

I. Introduction

A collection of nodes that communicate with each other by forming a multihop radio network and maintaining connectivity in a decentralized manner is called an ad hoc network. There is no static infrastructure for the network, such as a server or a base station. These types of networks have many advantages, such as self-reconfiguration and adaptability to highly variable mobile characteristics like the transmission conditions, propagation channel distribution characteristics and power level. They are useful in many situations such as military applications, conferences, lectures, emergency search, rescue operations and law enforcement. The idea of such networking is to support robust and efficient operation in mobile wireless networks by incorporating routing functionality into mobile nodes. Quality of Service (QoS) support in mobile ad-hoc networks (MANETs) is a challenging task. The model used in this paper supports both real time UDP traffic and best-effort UDP and TCP traffic [9]. A time division multiple access (TDMA) scheme is generally used in the wireless extension for bandwidth reservation for the mobile host to host similar to mobile and base station connections.

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In this paper, different QoS traffic flows in the network has been considered to evaluate the performance for proposed algorithm of BWCM model. The algorithm includes a set of mechanisms: control management that calculates the BW, co-ordination that provides allocation of the bandwidth, temporary resource reservation process that released the connection link or bandwidth after complete the communication. Different QoS traffic flows in the network to evaluate the performance of BWCM model has been considered. The paper is organized as follows. Section II presents the routing protocol, derived from destination sequenced distance vector (DSDV) [1], which contains the features of bandwidth calculation and reservation. In Section III, bandwidth reservation has been analyzed. In Section IV, there are some simulation environment and methodology to be done to demonstrate the efficiency of the proposed algorithm. Section V concludes the paper.

II. Routing Protocol & Related Work

Destination Sequenced Distance Vector (DSDV) [1] is a Proactive routing protocol that solves the major problem associated with the Distance Vector routing of wired. The DSDV protocol requires each mobile station to advertise, to each of its current neighbours, its own routing table. The entries in this list may change fairly dynamically over time, so the advertisement must be made often enough to ensure that every mobile computer can almost always locate every other mobile computer. In addition, each mobile computer agrees to relay data packets to other computers upon request. At all instants, the DSDV protocol guarantees loop-free paths to each destination. Kumar et al. [2] improves the quality of QoS parameter for MANET. C. Gomathy et al. [3] has been design a fuzzy based priority scheduler to determine the priority of the packets. Improve the end-to-end QoS target in MANET. L. Khoukhi et. Al [4], have proposed a flexible QoS routing protocol (AQOPC) based on multi-service classes and multi-path schemes. It provides information about the state of bandwidth, end-to-end delay and hop count in the network. AQOPC performs an accurate admission control and a good use of network resources by calculating multiple paths and generating the needed service classes to support different QoS user requirements. In [5], a core-extraction distributed ad hoc routing (CEDAR) algorithm is proposed that uses core extraction, link state propagation, and route computation to support QOS in wireless ad hoc networks. In [6], the authors have addressed the problem of supporting real-time communications in a multihop mobile network using QoS routing that permits bandwidth calculation and slotreservation. This protocol can be applied to two main scenarios: multimedia ad hoc wireless networks and multihop extensions of wireless ATM networks. The ad hoc QoS on-demand routing (AQOR) is discussed in [7], which integrates signaling functions for resource reservation and QoS maintenance at per-flow granularity. A link-state QoS routing protocol for ad hoc networks (QOLSR) was proposed in [8] with the aim of implementing QoS functionality while dealing with limited available resources in a dynamic environment.

III. Bandwidth Reservation

Multimedia applications such as digital audio and video have more stringent QoS requirements than traditional applications. For a network to deliver QoS guarantees, it must reserve and control resources. A major challenge in multihop, multimedia networks is the ability to account for resources so that bandwidth reservations can be placed on them. To support QoS for real-time applications, we need to know not only the minimal delay path to the destination, but also the available bandwidth on it. In the model [9], includes a temporary resource reservation process, co-ordination, and control management is name as BWCM controller. The controller is to determine whether the available resources in a network can meet the requirements of a new flow while maintaining bandwidth levels for existing flows, co-ordination among the packets. Accordingly, the decision is performed on the acceptance or rejection of a flow. In the controller, the source node has a final decision to accept or reject the user QoS requirements based on the feedback information of the network. This feedback measure is the packet delay measured by the MAC layer, which is calculated by the difference between the time of receiving an acknowledge packet (from the next-hop) and the time of sending a packet to the MAC layer (from the upper layer). This allows the controller to measure the local available bandwidth at each node in the network. The measured available bandwidth is then used by the controller to decide if the flow can be admitted for a particular service. The real-time traffic measured by the BWCM controller is in terms of bits per second. The estimation of the end-to-end available bandwidth is performed by sending a request from source node toward the destination. For that purpose, an UDP control packet is exploited by using an additional field "BW" that contains initially the value of the requested bandwidth. At each intermediate node, a comparison is performed between the value of BW and the available bandwidth of the current node. The value of the field BW is updated if it is bigger than the value BW_{avai} of the current node. When the destination receives the UDP control packet, BW represents the minimum bandwidth available along the path, and it is copied from UDP to a newly generated short replay message. The latter packet is transmitted back to the source node and at the same time the temporary resource reservation process (TRP) is performed. Additional fields are used during TRP mechanism, which are stored in each intermediary node in order to specify the temporary reservation status of the node, the status duration and the flow identifier. The first field is set to value of the reserved bandwidth and the status duration is set to a certain value "T". T indicates the period of time within which the temporary reservation is performed. Note that even when the temporary reservation is performed by a flow, other flows can also exploit the available resources of the node. The reserved bandwidth is released just after the expiration of T duration. The evaluation of the right status duration to be set at a particular node is explained in the following. The computation of the right status duration needs to take into account the number of hops between the source and the particular node, and also the delays between the intermediate nodes. Real and non-real time applications such as digital audio and video have much more stringent QoS requirements than traditional applications. For a network to deliver QoS guarantees, it must reserve and control resources. A major challenge in multihop, multimedia networks is the ability to account for resources so that bandwidth reservations can be placed on them. In cellular (single hop) networks, such accountability is made easily by the fact that all stations learn of each other's requirements, either directly or through a control station. However, this solution cannot be extended to the multihop wireless environment. To support QoS for real and non-time applications, we need to know not only the minimal delay path to the destination, but also the available bandwidth on it. A BWCM mode should be accepted only if there is enough available bandwidth and omitting signal-to-interference ratio, packet loss rate, etc.. This is because bandwidth guarantee is one of the most critical requirements for real-time applications. "BW" in time slotted network systems is measured in terms of the amount of "free" slots. The goal of the QoS routing algorithm is to find a shortest path such that the available bandwidth on the path is above the minimal requirement. To compute the "BW" constrained shortest path, we not only have to know the available bandwidth on each link along the path, but we also have to determine the scheduling of free slots. Though some algorithms were proposed to solve this QoS routing problem, they unfortunately may only work in some special environments [10].

A. *Algorithm & Bandwidth Calculation*

The transmission time scale is organized in frames, each containing a fixed number of time slots. The entire network is synchronized on a frame and slot basis. The frame/slot synchronization mechanism can be implemented with techniques similar to those employed in the wired networks and modified to operate in a wireless mobile environment. The entire network is synchronized on a frame and slot basis. Propagation delays will cause imprecision in slot synchronization. However, slot guard times (fractions of a microsecond) will amply absorb propagation delay effects (in microseconds). Each frame is divided into two phases, namely, the control and the data phase as shown in Fig. 1. The size of each slot in the control phase is much smaller than the one in the data phase. The call admission control is used to perform all the control functions, such as temporary resource reservation, co-ordination, control management, routing protocols AODV, etc. Each node takes turns to broadcast its information to all of its neighbors in a predefined slot, such that the network control functions can be performed distributive. We assume the information can be heard by all of its adjacent nodes. In a noisy environment, where the information may not always be heard perfectly at the adjacent nodes, an acknowledgment scheme is performed in which each node has to ACK for the last information in its control slot. By exploiting this approach, there may be one frame delay for the data transmission after issuing the data slot reservation.

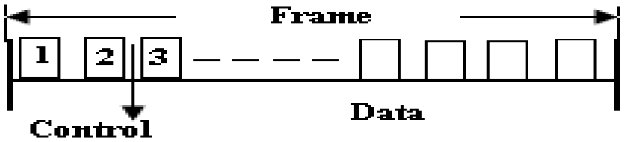
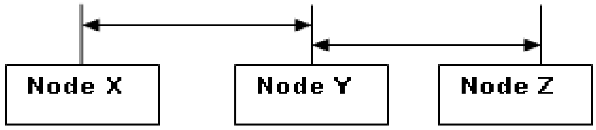


Fig. 1. Frame Structure



In Node X, the next node to destination Z is Y.
(Route [Z].next=Y)

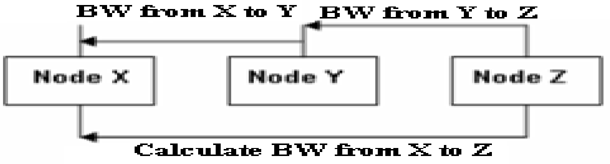


Fig. 2. Calculation of End-to-End BW

Ideally, at the end of the control phase, each node has learned the channel reservation status of the data phase. This information will help one to schedule free slots, verify the failure of reserved slots, and drop expired real-time packets. Because only adjacent node may hear the reservation information, the free slots recorded at each node can be different. It defines the set of the common free slots between two adjacent nodes to be the link bandwidth. As shown in fig. 2, in which X intends to compute the bandwidth to Z. Assume that the next hop is Y. If **Y** can compute the available bandwidth to Z, then X can use this information and the link bandwidth to Y to compute the bandwidth to Z. It is define the end-to-end bandwidth (path bandwidth) between two nodes. If two nodes are adjacent, the path bandwidth is the link bandwidth. Consider the example in fig. 2, and assume that one hop distance is between Z and Y. Assume the link bandwidth of both (Z, Y) and (Y, X) are the different as in fig. 3. If X uses slots 1, 2, 3 to send the packet to Y, then Y can use 4, 5 slot to forward packet to Z. This is because Y cannot be in transmitting and listening Z, denoted as path_BW (X, Z), can be {1, 2, 3}, and its size is three. In this case, five free slots can only contribute two slots for path bandwidth. If there are only three free slots on both links, then the size of path bandwidth is [3/2] =1. Similarly, four free slots can contribute two slots for path bandwidth is [4/2] =2.

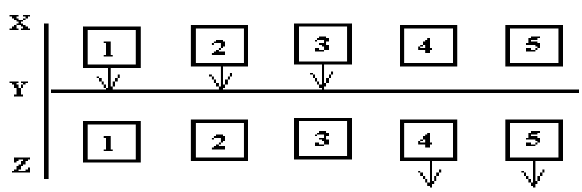


Fig. 3. Containing Case

The detail of BW calculation algorithm is given below. The algorithm maintains the routing table (two alternative routes in the algorithm, i.e., next2 and next3); next2 has larger bandwidth than next3. The "next" in the algorithm means the primary route. It is notable that the primary route is shortest, but is not necessary to have the largest bandwidth. When a host generates a new call, it uses the algorithm to construct the path. In the algorithm, the route that satisfies the QoS requirement in order to precedence next, next2, and next3. The chosen route will be the primary route. That is, the next entry in the routing table may be changed depending on the requirement. After choosing the primary route, the source node will send out a call setup message to next. When receiving the message, the next node will run the protocol in the algorithm to reserve bandwidth for the new call. When a topological change destroys the primary route, node will try to rebuild a new path immediately, using either next2 or next3. Thus, a new route from the breakpoint will be established by sending call setup message node-by-node to the destination.

Table 1
Algorithm for BW calculation

```
Bandwidth information is embedded in the routing table
for all host i do
{
if (i==sender)
{
link_BW=free_slots of sender;
route[sender].BW=link_BW;
}
else if (route[i].next=sender)
{
comm_BW=link_BW & route[i].BW of sender;
comm_BW_size=size(comm._BW);
diff_BW1=size(link_BW ^ comm_BW);
diff_BW2=size(route[i].BW of sender ^ comm_BW;

if(diff_BW1 <= diff_BW2)
{
route[i].BW_size=diff_BW1;
remain_BW_size=diff_BW2 – diff_BW1;
}
else
{
route[i].BW_size=diff_BW2;
remain_BW_size=diff_BW1 – diff_BW2;
}
if(remain_BW_size > 0 && comm._BW_size > 0)
{

if(comm._BW_size <= remain_BW_size)
route[i].BW_size=route[i].BW_size + comm._BW_size;

else
{
route[i].BW_size=route[i].BW_size + remain_BW_size;

comm._BW_size=floor((comm._BW_size – remain_BW_size)/2);

if(comm._BW_size > 0)
route[i].BW_size=route[i].BW_size + comm._BW_size;
            }
          }
        }
      }
```

Table 2
Routing table maintenance algorithm

```
     When receiving a routing table from a neighbor:
1    for all i
     {
2    if(i!=sender && sender routing table[i].next!=myself && my routing
     table[i].next!=sender routing table[i].next)
     {
3    if(sender!=my routing table[i].next3)
     {
4    Calculate BW to destination host i using the same algorithm in
     DSDV;
5    if(calculate_BW > BW from original next2)
     {
6    new next2=sender;
7    update next2 BW information;
       }
      }
8    if(sender!=my routing table[i].next && sender!=my routing
     table[i].next2)
     {
9    calculate BW to destination host i using the same algorithm in DSDV;

10   if(calculate_BW > BW from original next3)
     {
11   new next3=sender;
12   update next3 BW information;
          }
         }
        }
      }
```

In each time frame as shown in fig. 3, the data slot in the data phase is 5ms, and the control slot in control phase is 0.1ms. Assume there are 16 data slots in data phase. So the frame length is 20*0.1+16*5=82ms. Since the number of data slots is less than the number of nodes, nodes need to compete for these data slots. The source–destination pair of a call is randomly chosen, and their distance must be greater than one. Once a call request is accepted on a link, a data slots is reserved automatically for all the subsequent packets in the connection. The window is released when either the session is finished or the packet is received. There are three types of QoS for the offered traffic. QoS_1, QoS_2 and QoS_4 need one, two, and four data slots in each frame, respectively. The total simulation time is 10^6 ms. A new call is generated every cycle (82 ms). Each call duration is an exponential distribution with the mean value180s. The inter arrival time of packets within a QoS_1 session is an exponential distribution with 100 ms on average. Similarly, the mean values of the inter arrival time for QoS_2 and QoS_4 are 50 and 25ms, respectively. The maximal queuing delay of a data packet within a node is set to four frame lengths (328 ms).

IV. Simulation Environment and Methodology

Our simulation modeled a network of 20 mobile nodes placed randomly within 100x100meter area. That is, two nodes can hear each other if their distance is within the transmission range. Data rate is 11Mbps. Each simulation is run for 1200 seconds of simulation time. Multiple simulations run with different seed values are conducted for each scenario and collected data is averaged over those simulated results. A free space propagation model is used in our simulation. Data sessions with randomly selected sources and destinations were simulated. In this model, the channel quality may affect the packet transmission. That is, the noise in the channel may cause errors in packets. The channel quality specified by the bit error rate is uniform. Because the connection traffic is delay sensitive rather than error sensitive. The effect of mobility to the system performance has been emphasized. The traffic load is varied, by changing the number of data and the effect is evaluated with DSDV routing protocols. The following metrics are used in computing the performance:

(a) Average Throughput (b) Rerouting
(c) Average Delay

In the first analysis, it has been considered the effect of variable mobility on the rerouting due to a broken path. If any one of the links on the path is broken, the connection over the path needs to be rerouted. fig. 4 shows the simulation result. The curve QoS_i means QoS is uniform for all traffic flows. Hybrid QoS means different QoS traffic flows in the system. At the call setup, each source–destination pair can randomly determine its QoS type with the uniform distribution that will not be changed during the active period. Observe that the percentage of calls that need to be rerouted during their active period increases as the mobility is increasing. That is, high mobility causes paths to be broken frequently. When mobility is 20 m/s of 'QoS_1', about 54% of the connections need to be rerouted. When mobility is 20 m/s of 'Hybrid QoS' about 44% of the connection need to be rerouted. It is notable that the result is independent of the QoS of the traffic flows. This is because what we measure is the fraction of connections that have already received a QoS route and need to be rerouted during their active periods. The second analysis is to find the average throughput. In fig. 5**,** we can find that the throughput of each connection decreases as the mobility increases. High mobility makes frequent rerouting and thus results in more end-to-end transmission delay and more packet loss. In addition, observe that the high QoS connection has high throughput on average because of the high input rate. In addition, slot reservation makes the input packet flow have lower queueing delay to avoid the packet loss. The throughput of hybrid traffic is similar to QoS_2 traffic.

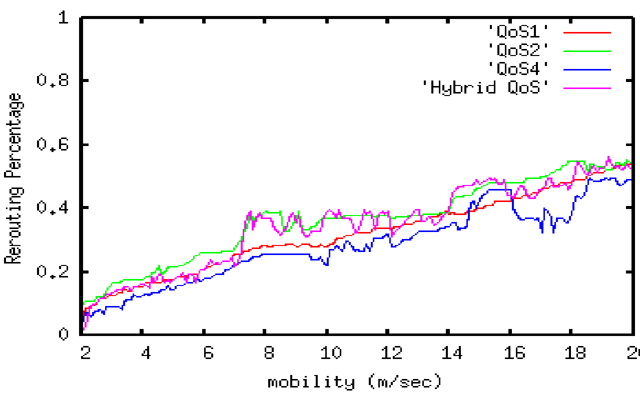


Fig. 4. The percentage of calls to be rerouted.

The average hop delay is shown in fig. 6. Since path length is not the same for all packets due to rerouting, in this paper we show the average hop delay instead of end-to-end delay for those packets that can reach the destinations. The hop delay is computed from the end-to-end delay divided by the path length. This can reduce the end-to-end delay. In addition, hop-by-hop slot reservation can also limit the queuing delay within a host. The stable delay can be observed. There are two factors to limit the delay within a host. First, for a host, the input rate is always less than the processing rate. The mean input rate of QoS$_1$ is 1 packet/100 ms, and the mean processing rate is 1 packet/82 ms For example, the connection has a stable hop delay of about 92ms, which is close to the frame length (82 ms). QoS$_2$ and QoS$_4$ also have stable delays of about 69 and 55ms, respectively. When the more slots are allocated per frame, a packet has a higher probability to be transmitted sooner. So the delay is lower. Mobility only makes the delay increase slightly. The following set of simulation is to assess the improvement introduced by the "standby" routing feature. This feature is of critical importance when stations are mobile. The total simulation time is 10^8ms. A new call is generated every two cycles (2*82 ms). If no data packet is sent over the reserved slots for ten cycles (10*82 ms), the reserved slots will be released. There are four simulation to be done. In the first one, evaluate the successful probability of constructing a connection through each route by exploiting the path bandwidth information and slot allocation algorithm under the condition of mobility. Each node is considered to run the algorithm in table 2 to set up a new call. fig. 7 of 20 m/sec, for example, 96% of calls that use next (primary route) at the source node can set up the QoS connection successfully, and 4% will fail because of outdated bandwidth information. Because of mobility, the path bandwidth information is changed dynamically. Fig. 7 presents the effect of the "possible" outdated bandwidth information on the primary route (next) and the standby routes (next2 and next3). From the simulation results, it can observe that no matter which route is selected at the source, we still have high probability (for example, 96% for next, 82% for next2, and 67% for next3 at a mobility of 20m/sec) to construct a connection successfully. That is, the effect of mobility on the route selection that establishes a connection is not too strong. For a given connection of a call, it may be constructed by a different route at the source. According to our algorithm for constructing a QoS path (Table 2), in fig. 8 near 35–55% of connection's are setup through the primary route (i.e. next) under different mobility. Similarly, 41–59% of connection's are through next2 under different mobility. From the simulation result, it is found that the standby route is particularly useful. The primary route is the shortest path calculated by the DSDV algorithm. However, if all source–destination pairs only consider the shortest path, there will be some hot spots that lack enough bandwidth. Once a call request is passing through those nodes, it will be rejected. Thus, this is the reason why there are only 35–55% of connection's that can pass through the primary route. There is only about 12% of connection's using next3. This means that if next and next2 do not have enough bandwidth, there is a small probability for next 3 to have enough bandwidth. Actually at this time, the system is saturated. When a link of a connection is broken, the new connection can be constructed from the "breakpoint" as shown in fig. 11, if there exists enough bandwidth in a standby route. In case of no bandwidth in any standby route to stop traffic flow from the upstream node to intermediate node that has a QoS route to the destination. In the worst case, a new connection will be reconstructed from the source node. All reserved slots by the old connection will be released hop-by-hop. fig. 9 shows the probability of finding a feasible alternate route at the breakpoint according to the current bandwidth information before the new call setup begins. At a mobility of 20m/sec for example, there is a probability of 0.33 for next2 (0.23 for next3), which has enough bandwidth to the destination at the breakpoint. Therefore, the mobility does not affect the probability. fig. 10 shows the probability of a successful call setup given a route (i.e., either next2 or next3) at the breakpoint to the destination. The next2 path can have a probability of more than 0.95 to set up a new connection in low mobility. In high mobility, the probability is still more than 0.85. Observe that in high mobility, there is a lesser chance of a successful call setup. This is because when the system is saturated, the node speed does not cause an intermediate node between the breakpoint to the destination to see another good neighbor who has enough bandwidth. Combine the results in figs. 9 and 10. From this simulation results, there is very low probability (about 0.24) of having another QoS route at the breakpoint. If we consider the set of

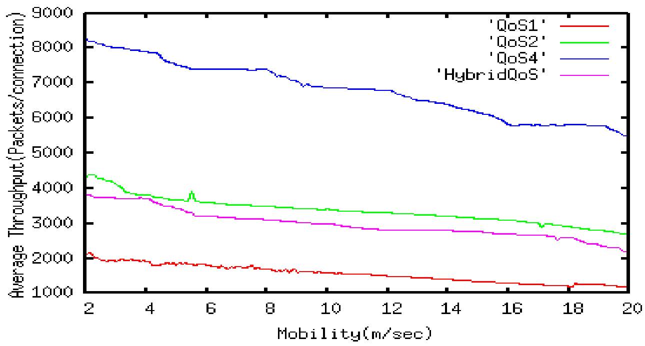


Fig. 5. Average throughput of different QoS's

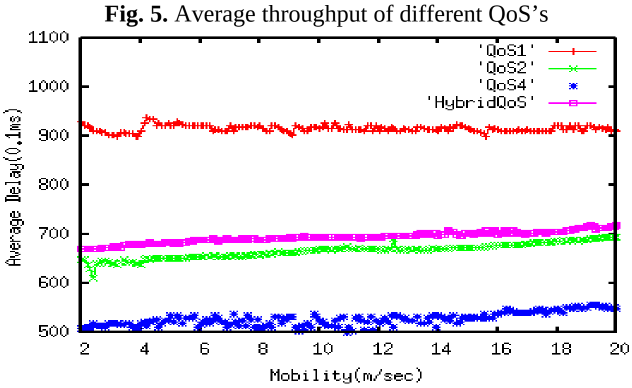


Fig. 6. Average hop delay

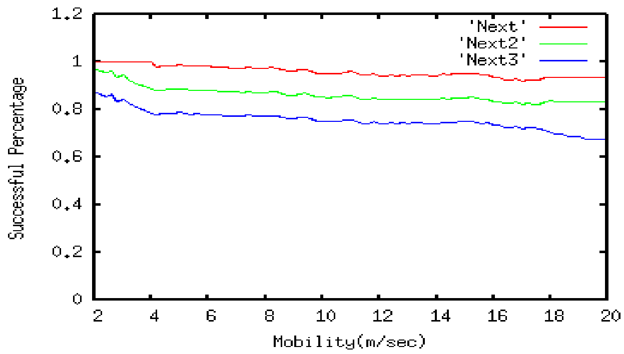


Fig. 7. The reliability of different routes for the QoS requirement

nodes from the source to the breakpoint along the path, the probability of having a QoS route at any one of these nodes will be much higher than the case of just considering the breakpoint. We must note that there is no extra communication cost to maintain the standby routes. In the last simulation analysis, we intend to assess how useful the bandwidth information is that is obtained from the bandwidth calculation algorithm presented in table 2. We can exploit this QoS indication to determine if a new call can be accepted or not. This information lets us foresee whether a connection can be established along a given route before the call setup begins. If we only use the DSDV algorithm and the reservation algorithm (Table 3), then a new call may be blocked in some intermediate node that is saturated. No source can construct a connection via the saturated node until one of the connections over the node ends its transmission, and the bandwidth becomes available. Periodically mobile nodes exchange bandwidth information. The data is propagated hop-by-hop and cannot reach all nodes immediately. From the simulation results, the call blocking rate of two systems that are running the same routing algorithm (i.e., DSDV) and the reservation algorithm in table 2 has been compared. In addition, we also consider the case in which bandwidth information shows that there is no bandwidth, but the new call still can be set up. That is, the current bandwidth is less than the real bandwidth. We run 100 simulations (each of length 10 ms) with different initial topologies to compute the averages. The call generating rate is one call every two frames (i.e.2*82=164 ms). Thus, there are $(1/64)*10^6$=6098 calls generated during 10 ms. Observe that about 11% calls will be blocked if there is no bandwidth information. However, only from two to three calls of the 6098 calls will be blocked if the source node has the bandwidth information. This information lets the source node determine if a new call should be blocked. In addition, this information is seldom underestimated by our algorithm. That is, the reliability of the information is high. Thus, this knowledge enables more efficient call admission control.

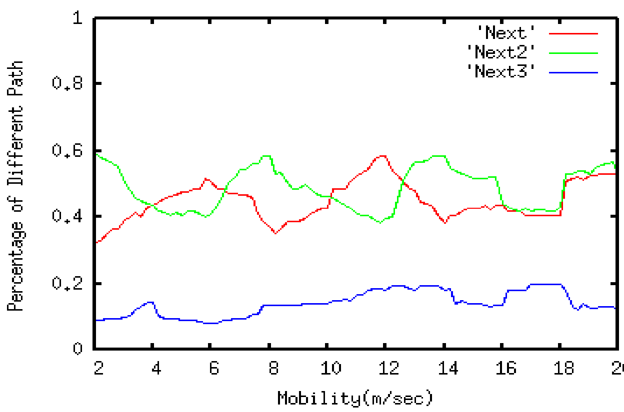


Fig. 8. Route selections at the source for a given connection

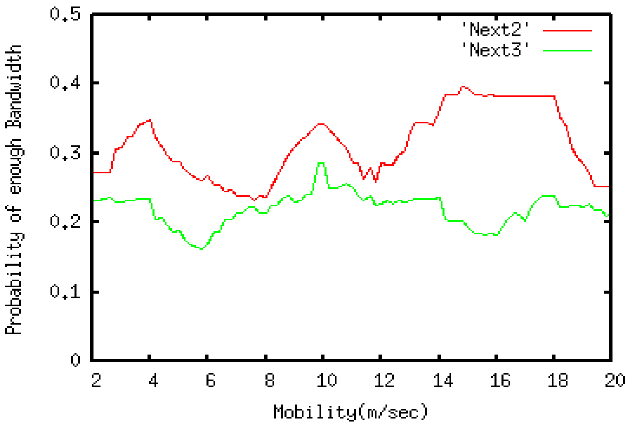


Fig. 9. Probability of the breakpoint having alternative routes

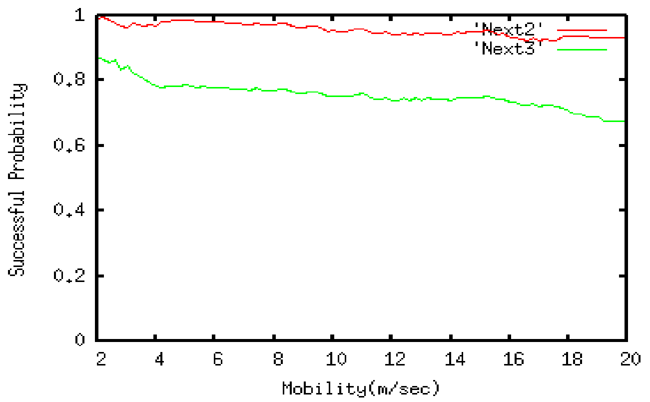


Fig. 10. The performance of the standby routing at the breakpoint

V. Conclusion

In this paper, a algorithm has been proposed that contains bandwidth calculation and slot reservation for mobile networks. That can be applied to multimedia ad hoc wireless networks. Specially, the bandwidth information can be used to assist in performing the handoff of a mobile host between two base stations. Traffic flows with different QoS types have been considered. In addition to, standby routing enhances the performance in the mobile environment. The proposed algorithm has been analyzed for different QoS services in term of no. of connections, average hop delay and average throughput.

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