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A Routing Protocol for Ad Hoc Wireless Networks Based on CDMA
نظامِ توجيهِ لشبكاتِ اللاسلكي الخاصّةِ باستخدام تقسيم الشفرات متعددة الدخول

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إن شبكات اللاسلكي الخاصة الـ(Ad hoc Network)عبارة عن تجمع من النقاط (Nodes) والتي تستطيع فيما بينها تكوين شبكة بدون أية بُنى تحتية. فكل نقطة تساهم في الشبكة كمضيف (Host) وكمرحل (Router) فهي لابد ان تكون قادرة على نقل حزم بيانات (Packets) النقاط الأخرى من نقطة لأخرى خلال الشبكة. وكذلك تكون قادرة على الاتصال بالنقاط الأخرى رغم تغير طبو غرافية الشبكة نتيجة لحركة تلك النقاط.

وفي مثل هذه الشبكات فانه من الأمور الهامة ضمان جودة الخدمة (رُكز هنا على سعة النطاق فقط) ولتحقيق ذلك فلقد طور نظام تحكم القبول (admission control scheme) والذي يستطيع ضمان سعة النطاق ليناسب تطبيقات الوسائط المتعددة والمباشرة (real-time) في الشبكات المتحركة متعددة القفزات. يوجد في هذا النظام حساب لسعة النطاق المطلوبة من المصدر للهدف حيث يُعلم المصدر بسعة النطاق المطلوبة للهدف في الشبكة لكي يتكمن من إتمام الاتصال المناسب للتطبيق

Abstract

A Mobile ad hoc network is a collection of wireless nodes, all of which may be mobile, laptop, personal digital assistants that dynamically create a network without using any infrastructure. Each host participating in the network acts both as a host and as a router. As mobile hosts are no longer just end systems; each node must be able to function as a router as well as to relay packets generated by other nodes. As the nodes move in and out with respect to each other, the resulting changing topology must somehow be able to communicate to all other nodes as appropriate.

In such networks, it is of a special importance to guarantee Quality of Service (QoS), such as packet loss rate and bandwidth, should be guaranteed. To accomplish this, an admission control scheme which guarantees bandwidth for real-time applications in multihop mobile ad hoc networks is developed. In this scheme there is end-to-end bandwidth calculation and bandwidth allocation. To perform this objective, the source is informed of the bandwidth and QoS available to any destination in the network to enable the establishment of QoS connections and the efficient support of real time applications in the network.

Simulation results showed distinct performance advantages of the introduced protocol over those reported previously.

1. Introduction

Routing protocols for traditional wired networks are designed to support tremendous numbers of nodes, but they assume that the relative positions of the nodes generally remain unchanged. On the contrary, in mobile ad hoc networks there may be fewer which to route, however the network topology changes may be drastic and frequent as the mobile nodes move.

Efficient routing in multihop mobile network requires that the routing protocol operates in an on-demand fashion. Moreover, the routing protocol must limit the number of nodes that should be informed of topology changes. The challenge in the design of this mobile network is to develop a dynamic routing protocol that can find routes between two communicating mobile hosts (nodes) and should be able to keep up with the high degree of node mobility.

Ad hoc mobile hosts communicate with each other using multihop wireless links. Each mobile host in the network also acts as a router, forwarding data packets to other nodes. This kind of network may be implemented over the wireless local area network or the cellular networks (e.g., GPRS, UMTS or IMT-2000). Often it is internetworking with a wired network, e.g., ATM or Internet, so that the mobile users can be connected to Internet multimedia. There were different forms which

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have appeared in the wireless extensions of the wired ATM [1]. They concentrated on the cellular architecture for wireless PCN (personal communication networks). In this system all nodes in the cell are connected to the base station (ATM switch) directly (one hop). A TDMA (time division multiple access) scheme is generally used in the wireless extension for bandwidth reservation in the mobile host to base station connections.

The problem to connect a multihop wireless network to wired backbone network is to guarantee QoS parameters (in wireless part) through all the path.

In this paper the bandwidth is considered as a measure of QoS. In time-slotted network systems the bandwidth is measured by the number of time slots. The aim of QoS routing is to find the shortest available path that achieves the required bandwidth. The protocol reported in [2] finds the shortest path and then checks the required bandwidth. If it happens that the bandwidth is unavailable the call will be rejected. On the other hand, the protocol reported in [3] finds the shortest path that provides the required bandwidth, i.e., the selected path may not be the shortest one. Moreover, it computes the bandwidth, determines and schedules the slots that will be used before the completion of the reservation process.

The protocol introduced here provides two advantages over that reported in [3]. That is, the system also computes the route that supports the required bandwidth and performs slot assignment during the reservation process. TDMA is used for bandwidth reservation and CDMA (code division multiple access) is also overlaid upon TDMA to improve the system performance.

2. QoS Definition and Bandwidth Calculation

In cellular single hop system, it may be easier to guarantee the QoS parameters than in multihop system. The QoS parameters should be propagated within the network in order to extend the ATM VC (virtual circuit) into the wireless network. The resources must be reserved and controlled to guarantee the QoS through the network. The difficulty in multihop multimedia networks is to account the resources so that bandwidth reservations can be performed efficiently. To support QoS for real-time applications, not only the minimal hop-distance path to the destination, but also the available bandwidth must be known. A VC can be accepted if not only it has enough available bandwidth, but also it must not disrupt the existing QoS VCs.

Here consideration is only on "bandwidth" as the QoS parameter (thus omitting Signal to Interference Ratio (SIR), packet loss rate, etc. This is because bandwidth guarantee is one of the most critical requirements for real time applications.

In TDMA (time-slotted) network systems the "Bandwidth" can be translated into the number of free slots. The objective of the QoS routing is to find the shortest path among all paths which satisfy the bandwidth condition.

A call admission control based on on-demand routing protocol for QoS support in multihop mobile networks is proposed. In this protocol, no routing information and no routing tables are exchanged periodically. For performance evaluation different QoS traffic flows in the network is considered.

In the protocol reported here , it is concentrated on finding a feasible bandwidth route that satisfied the bandwidth demand whether it satisfies the shortest path or not. However, the protocol finds the shortest path of the available feasible routes. Moreover no routing tables are used in the protocol introduced here.

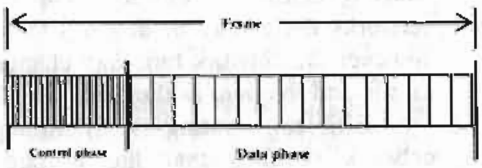

Figure 1: *"Frame structure."*

In this protocol, the transmission time scale is organized in frames. Each frame consists of fixed number of time slots as shown in *Fig.1*. The entire network is synchronized on a frame and slot basis. The frame/slot synchronization mechanism is not described here, but can be implemented in the mobile ad hoc networks with techniques similar to those employed in the wired networks [4]. Propagation delays in wireless ad hoc mobile environment will cause imprecision in slot synchronization. To avoid this problem slot guard times (fractions of a microsecond) are taken into consideration to absorb propagation delay effects. Referring back to *Fig.1*, it can be seen that all the control functions such as slot and frame synchronization, power measurement, code assignment, VC setup, slots request are performed in the control phase. The control phase uses pure TDMA with full power transmission in a common code. That is, 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 distributively. It is assumed that the information transmitted are heard by all 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 such schemes 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.

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 know free slots, verify the failure of reserved slots, and drop expired real-time packets.

2.1 CDMA over TDMA and Hidden terminal problem

Consider the wireless network shown in *Fig.2* which uses TDMA for data transmission. A mobile host A requires to connect or transfer data to mobile host D. Assume all the slots in the TDMA frame are free and slots (1&2) are reserved to transmit data to mobile host B and slots (3&4) to forward packets to the next node in the route (mobile host C). The mobile host C may use slots (1&2) to forward packets to destination (mobile host D). Here a collision at B may occur (because B may receive packets from A and C simultaneously) [5]. CDMA can be used to solve this problem (all spreading codes are assumed to be orthogonal to each other). For example, for the same topology in *Fig.2*, *Fig.3* shows that when slots 3, 4 are used to transmit packets, one can assign a different code (say code 2) to another node to transmit in the same time on slots used before. That is, each transmitter- is assigned a code for data transmission. A spreading code can be reused if two nodes have a hop distance greater than two [5]. In *Fig.3*, it can be seen that the same slots (1 and 2) are used to send packets encoded by a different code without any collision at B.

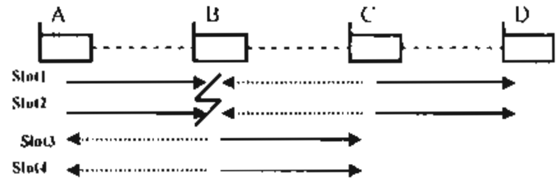

Figure 2: "Hidden terminal problem."[6]

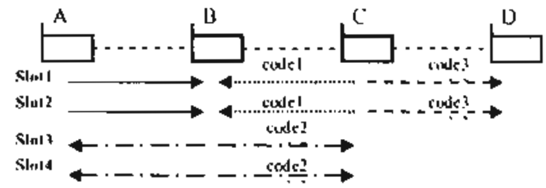

Figure 3: "CDMA over TDMA."

In simulation study , a TDMA is assumed first then CDMA is overlaid upon TDMA is considered and a comparison is carried out. As shown in *Fig.3*, ideal code assignment scheme [2] is assumed running in the lower layer of the system, and all spreading codes are completely orthogonal to each other. Thus, the hidden terminal problem shown in *Fig.2* is avoided.

Link bandwidth is the set of the common free slots between two adjacent nodes as the network is multihop and only adjacent nodes can hear the reservation information. To compute the bandwidth from C to A as in *Fig.4*

. consider the next hop is B. If B can compute the available bandwidth to A, then C can use this information and the "link bandwidth" to B to compute the bandwidth to A.

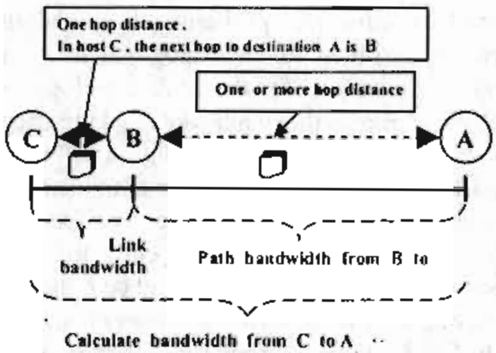

Figure 4: "Bandwidth information calculation overview."

Path bandwidth the end-to-end bandwidth between two nodes is the set of available slots between them. The two nodes may be adjacent or not. If they are adjacent, the path bandwidth is the link bandwidth . If C has free slots {2, 3, 5}, and B has free slots {0,2,3}, then the link bandwidth between C and B is {2, 3}. This means that only exploit slot 2 and slot 3 can be used for packet transmission from C to B. Thus, if a VC session needs more than two slots during a time frame, then it will be rejected. The link bandwidth may be estimated as follows:

link_BW(P,Q) = free_slot(P) ∩ free_slot(Q).

2.2 The on-demand routing with bandwidth constraint

2.2.1. Route Discovery

Like Ad-hoc On-demand Distance Vector routing (AODV) [1] and Dynamic Source Routing (DSR) [7], the protocol conforms to a pure on-demand rule. When a source node wants to communicate with another node for which it has no routing information, it floods a route request (RREQ) packet to its neighbors. If a route exists from the source to the destination, a route request (RREQ) will find (record) it. In the protocol reported here, all packets contain the following uniform fields:

<packet_type, source_addr, dest_addr, sequence#,route_list, data, TTL>

The definition of all packet types in the above field are shown in Table1 and explained as:. *<source_addr, sequence#>* is used to uniquely identify a packet. This *sequence#* is monotonically increasing, which can be used to supersede stale cached routes. *route_list* records the routing information. When any host receives a RREQ, it will perform the following operations:

1. If the pair *<source_addr, sequence#>* for this RREQ was seen recently, discard this redundant request packet and do not process it further.
2. Otherwise, if the address of this node appeared in the *route_list* in the RREQ, we drop this RREQ (do not rebroadcast) and do not process it any further.
3. Otherwise, (a) calculate the bandwidth from the source to this node to make sure that the required bandwidth is available through this path. The RREQ will be dropped if the result does not satisfy the QoS requirement, and do not process it any further. The state of the data slots is not modified at this time. (b) Decrement TTL by one. If TTL counts down to zero, we drop this RREQ and do not process it any further. TTL can limit the length of the delivery path. There may exist a very long path which satisfies the bandwidth requirement. However, this path will be difficult to maintain within a dynamic environment. In addition, unlimited packet flooding will deteriorate the network performance. The use of TTL can control the flooding traffic. (c) Append the address of this node to the *route_list* to track the route which the packet has traversed, and re-broadcast this request if this node is not the destination.

As expected, a destination node may receive more than one RREQ. Every RREQ packet indicates a unique feasible QoS path from the source to the destination. Thus, the destination node can keep more than one path. Multiple

Packet Type	Function
ROUTE_REQUEST (RREQ)	Send to discover route
ROUTE_REPLY (RREP)	Send to reserve route
RESERVE_FAIL	Nack for unsuccessful reservation
ROUTE_BROKEN	Nack for route broken
CLEAN_RREQ	Clean surplus RREQs
NO_ROUTE	Nack for finding no route
DATA	Use to transport datagram

Table1

connectivity between a source - destination pair can provide a more robust packet delivery. This is especially important in a multihop mobile network. In order to reduce the overhead of flooding, the destination node can broadcast a CLEAN_RREQ packet to clean RREQ packets that are still roaming around the network after receiving enough paths.

2.2.2 Route Reservation

When the destination node receives one RREQ packet from the source node, it returns a route reply (RREP) packet by unicasting back to the source following the route recorded in the route_list. As a RREQ travels from the source to the destination, it automatically sets up the reverse path from the destination back to the source. To set up a reverse path, a node records the address of the neighbor from which it received the copy of the RREQ. According to the information recorded within the RREQ, the destination can set up a bandwidth route and reserves resources (slots) hop-by-hop backward to the source.

Using the source routing algorithm, the fields <*route_list,* > is copied from RREQ to RREP. As the RREP traverses back to the source, each node along the path calculates and reserves slots. When the source receives a RREP, the end-to-end bandwidth reservation is successful, and the virtual circuit (VC) is established. Then, the source node can begin transmitting datagrams. It is to be noted that this establishment protocol for a VC connection from the source to the destination is a two-way handshaking. When a new call request arrives, the call admission control drives this establishment protocol. This new call will not be accepted until the reservation process is successfully completed

2.2.3 Unsuccessful Reservation

When the RREP travels back to the source, the reservation operation may not be successful. This may result from the fact that the slots which the system wants to reserve are occupied a little earlier by another VC or the route breaks and there are no slots that can be replaced. If this is the case, the route must be given up. The interrupted node sends a NACK (i.e., RESERVE_FAIL) back to the destination, and the destination re-starts the reservation process again along the next feasible path (note that in the route discovery process, each RREQ which arrives at the destination piggybacks a feasible bandwidth route). All nodes on the route from the interrupted node to the destination must free the reserved data slots when receiving RESERVE_FAIL. If there is no VC that can be set along all feasible bandwidth routes, the destination broadcasts another NACK (i.e., NO_ROUTE) to notify the source. Upon receiving NO_ROUTE, the source can either re-start the discovery process if it still requests a route to the destination, or reject the new call. In addition to NO_ROUTE arrival, if there is no response back to the source before the timeout occurs, the source can also re-perform the route discover operation.

Once a VC is established, the source begins sending datagrams in the data phase. At the end of the session, all reserved slots must be released. These free slots will be contended by all new connections. However, if the last packet is lost, the reserved slots that are released will not be known. This issue will be discussed in the next sub-section.

2.2.4. Connection Breakage

During the active period of a connection, a topological change may destroy a VC. The connection control must reroute or re-establish the VC over a new path. When a route is broken, the breakpoints send a special NACK (i.e., ROUTE_BROKEN) to the source and the destination. That is, once the next hop becomes unreachable, the breakpoint which is near the source sends an unsolicited NACK to the source, and the other breakpoint does to the destination. Each node along the path relays this ROUTE_BROKEN to its active neighbors and so on. Furthermore, they release all reserved slots for this connection and drop all data packets of this connection which are still waiting in the queue for transmission. Upon receiving the ROUTE_BROKEN, the source re-starts the discovery process to re-establish a VC over a new path, and the destination only drops the ROUTE_BROKEN (the main purpose of the travel of the ROUTE_BROKEN from the breakpoint to the destination is to release the reserved slots). This procedure is repeated until either the completion of data delivery or timeout. If timeout occurs, the source stops any data delivery for this session. If a link on the VC is broken before the completion of a session, the last data packet may be still on the way to the destination. This packet, thus, can not reach the destination, and is suspended within an intermediate node. In this situation, some resources are still occupied by this connection and can not be used by the others. In order to solve this problem, the timeout scheme is used for each reserved slot. If a reserved slot is not used to deliver data packets for a couple of data frames and timeout occurs, this slot is set free automatically. Such free slots will be fully utilized by the other new sessions.

3. Performance evaluation

The simulated environment consists of 20 nodes and moving uniformly in 1000×1000 ft^2 area. The nodes move at random directions, but uniform speed (the simulation program supports three types of movement –random model, deterministic model and semi-deterministic model). The direction of each node is random and changes every 1 sec. Radio transmission range is 400 ft. That is, two nodes can hear each other if their distance is within the transmission range. Data rate is 2 Mbit/s. In the simulation experiments, 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 in all experiments. Because the VC traffic is delay sensitive rather than error sensitive, packets therefore are not ACKed. A coding scheme is assumed running in the system to do the forward error correction.

The author built the simulation program using C#.NET language [8]. It supports a lot of options, it consists of four forms (windows). The fist one enables the user to choose which class he wants to perform his experiment on it. Each form of the remaining three forms presents a class of the three classes that the program supports.

In the experiments, more attention was paid to the effect of mobility upon the system performance. In each time frame (*Fig.1*), the data slot in the data phase is 5 ms, and the control slot in control phase is 0.1 ms. Channel overhead (e.g., code acquisition time, preamble, etc.) is factored into control/data packet length. It is assumed that there are 16 data slots in the data phase. So the frame length is 20 * 0.1+16 * 5 = 82 ms. The source-destination pair of a call is randomly chosen and their distance must be greater than one . The randomizer can not permit a call to be generated if this call existed in the last ten generated calls. When a call request is generated for a path, a transmission window(i.e. data slots) is reserved (on that link) automatically for all the subsequent packets in the connection. . The window is released when either the session is finished or the NACK packet (RESERVE_FAIL and ROUTE_BROKEN) is received. For a connection (call), the destination keeps three different bandwidth routes which are piggybacked within the first three RREQ's arriving at the destination. Simulation is carried out for three types of QoS for the offered traffic. QoS1, QoS2, and QoS4 which require one, two, and four data slots in each frame, respectively. For each simulation result, considering 10 different topologies and run 10000 frame time (i.e., 10000 * 82 ms) for each topology. The interarrival time of two calls is an exponential distribution with the mean value 10 cycles (820 ms). Each session length of QoS1, QoS2, and QoS4 is 100, 200, and 400 packets, respectively. Constant bit rate is assumed in all traffics. The interarrival time of packets within QoS1 is one cycle (82 ms). Similarly, the interarrival times for QoS2 and QoS4 are 41 ms and 21 ms, respectively.

In the first experiment, the effect of variable mobility on incomplete connection ratio is considered. A call may not be completed due to the mobility. If the last packet can arrive at the destination, this session is defined to be complete. Otherwise, it is incomplete. The calls are classified into complete, incomplete, rejected (that cant not find any routes or the three routes failed to be reserved) and incomplete at end (this case occurs when the source finishes sending all the packets and the route is broken and the last packet can not arrive to the destination). Some data packets of a complete session may be lost because of the topological change. The VC of a connection may be re-established during the active period and the last packet finally can get to the destination. This is also considered as a complete session. In the simulation, only the source is allowed to re-perform the route discovery operation once. If the session still can not be completed, it will be incomplete. Fig.5, Fig.6 and Fig.7 illustrate the result.

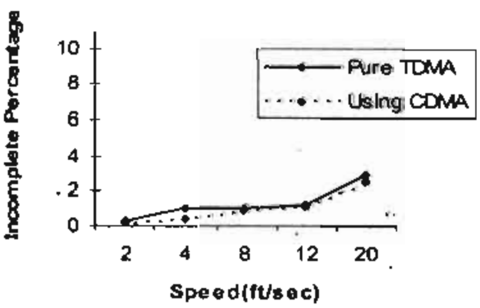

Figure 5: *Incomplete ratio of QoS1.*

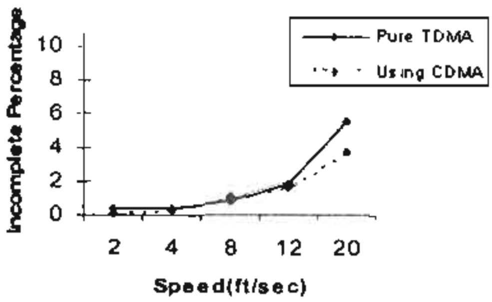

Figure 6: *Incomplete ratio of QoS2.*

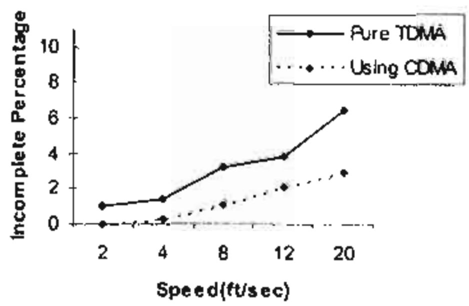

Figure 7: *Incomplete ratio of QoS4.*

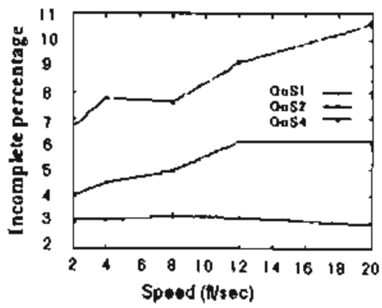

Figure 8: *Incomplete ratio of different QoS's*[3]

Observe that high mobility causes a path to be broken frequently. When mobility is 20 ft/s for example, QoS4, QoS2 and QoS1 respectively have 6.47% , 5.57% and 3.58% sessions which can not be completed. Using CDMA the results were improved to 2.98%, 3.78% , 2.35% respectively (that is the incomplete ratio decreases with 54%, 32% and 34% respectively). The bandwidth route of lower QoS traffic-as is to be expected, can be re-established easier. If results were compared with the results in [3] which are illustrated in Fig.8, one can find a great improvement in the results specially at low mobility. This improvement is due to the difference between the two protocols. The protocol reported here performs slot assignment and the route reservation in the same time but in [3] the slot assignment is performed during the route discovery (before route reservation) so the reservation operation may not be successful. This may result from the fact that the slots required to be reserved are occupied a little earlier by another VC and the system can not replace the selected slots before with other free slots.

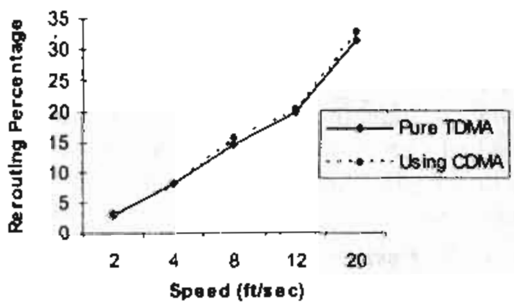

Figure 9: *Rerouting ratio of QoS 1*

In Fig.9, Fig.10 and Fig.11 it is noticed that the percentage of calls that were broken and be re-established is proportional to the mobility as expected. The general shape of curves here is approximately the same to the curves in [2] which are illustrated in Fig.12 but results may be different. This difference is due to the experiment parameters e.g. the call duration and the interarrival time of packets.

Note that in case of QoS4 the rerouting percentage using CDMA is greater than pure TDMA, this is due to the performance enhancement that makes some calls that were broken find other routes.

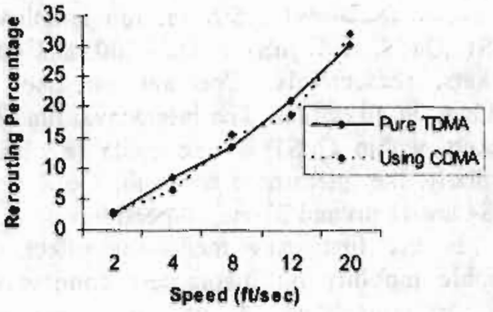

Figure 10: *Rerouting ratio of QoS2*

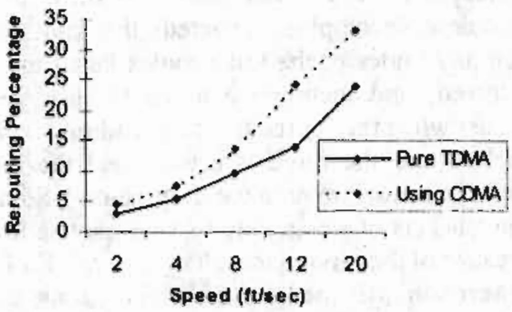

Figure 11: *Rerouting ratio of QoS4*

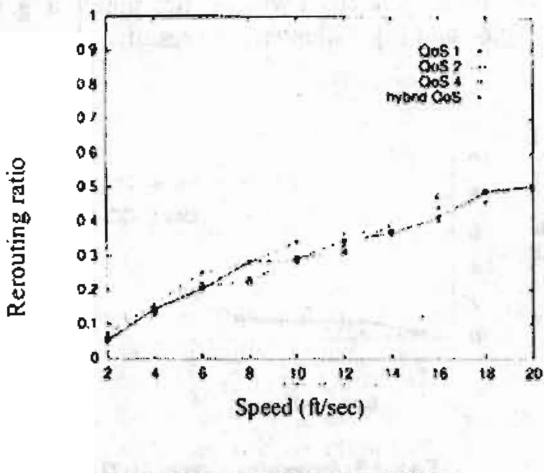

Figure 12: *Rerouting ratio of different QoS's*[2].

4. Conclusions

In summary, an admission control was presented over an on-demand routing protocol which is suitable for use with multihop mobile networks. This protocol is more powerful in the resource management than the works in [2,3]. Thus it can accept more calls in the network according to the simulation results. The route may not be the shortest in hop length. In the route reply process, performing slot assignment and the route reservation are made hop-by-hop backward from the destination to the source. This admission control can be applied to two important scenarios: multimedia ad-hoc wireless networks and multihop extension wireless ATM networks. In the performance experiments, traffic flows with different QoS types are considered. Finally, the protocol introduced here provides two advantages over that reported in [3]. That is the system also computes the route that support the required bandwidth and performs slot assignment during the reservation process. TDMA is used for bandwidth reservation and CDMA is also overlaid upon TDMA to improve the system performance.

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