ISSN (online): 2250-141X
Vol. 2 Issue 3, June 2012

Provisioning of QoS Routing based on Bandwidth estimation in Mobile Ad-Hoc Network

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***Abstract*— Most routing protocol focus on obtaining a workable route without considering network traffic condition for a mobile ad-hoc network (MANET). Therefore, the quality of service (QoS) is not easily achieved by the real time or multimedia applications. Providing quality-of-service (QoS) in wireless *ad-hoc* networks is an intrinsically complex task due to node mobility, distributed channel access, and fading radio signal effects. Proposed mechanism makes the resource consumption more efficient by minimizing the unnecessary signaling and stopping the session that cannot meet the demanded QoS requirement. Mechanism describes the QoS extension for Ad-hoc on-demand distance vector (AODV) routing. Our scheme does not modify the MAC protocol, but judge the effect of phenomena such as medium contention, channel fading and interference, which influence the available bandwidth, on it. Based on this phenomena the available bandwidth is estimated of a wireless host to each of its neighbors [3]. QoS AODV with bandwidth estimation shows a significant improvement in performance matrix, such as end-to-end delay statistics, available bandwidth, and probability of packet loss and so on. An implementation and simulation study in NS-2 for above algorithm is for improvement in throughput, overhead, delivery ratio and delay over the standard AODV for high work load scenario.**

KEY WORDS: *Ad hoc on-demand distance vector (AODV), Mobile Ad Hoc networks (MANET), Quality of service (QoS), routing algorithm, bandwidth estimation, performance matrix.*

1. Introduction

The introduction of real-time audio, video and data services into wireless networks presents a number of technical obstacles to overcome. Traditional QoS protocols cannot be easily migrated to the wireless environment due to the error-prone nature of wireless links and the high mobility of mobile devices. This is especially true for Mobile Ad Hoc Networks (MANETs) where every node moves arbitrarily causing the multi-hop network topology to change randomly and at unpredictable times. Such a network may operate in a stand-alone fashion, or may be connected to the larger Internet. Each host must forward traffic unrelated to its own use, and therefore be a router. The primary challenge in building a MANET is equipping each device to continuously maintain the information required to properly route traffic.

Quality of Service (QoS) is a set of service requirements that needs to be met by the network while transporting a packet stream from a source to its destination. The network needs are governed by the service requirements of end user applications in terms

ISSN (online): 2250-141X
Vol. 2 Issue 3, June 2012

of end-to-end performance, such as delay, bandwidth, probability of packet loss, etc. Power consumption is another QoS attribute which is more specific to MANETs.

QoS models specify an architecture in which some kinds of services could be provided. It is the system goal that has to be implemented. *QoS Adaptation* hides all environment-related features from awareness of the multimedia-application and provides an interface for applications to interact with QoS control. Above the network layer *QoS signaling* acts as a control center in QoS support. *QoS MA C* protocols are essential components in QoS for MANETs. QoS supporting components at upper layers, such as QoS signaling or QoS routing assume the existence of a MAC protocol, which solves the problems of medium contention, supports reliable communication [8].

2. Brief literature review

The network attempts to deliver all traffic as soon as possible within the limits of its abilities, but without guarantees related to throughput, delay or packet loss. Although best effort will remain adequate for most applications, QoS support is required to satisfy the growing need for multimedia over IP, like video streaming or IP telephony. The existing QoS models can be classified into two types according to their fundamental operation; the Integrated Services (IntServ) framework provides explicit reservations end-to-end and the Differentiated Services (DiffServ) architecture offers hop-by-hop differentiated treatment of packets.

The IntServ model merges the advantages of two different paradigms: datagram networks and circuit switched networks. It can provide a circuit-switched service in packet switched networks. The Resource Reservation Protocol (RSVP) was designed as the primary signaling protocol to setup and maintain the virtual connection. Based on these mechanisms, IntServ provides quantitative QoS for every flow [7].

Diffserv was designed to overcome the difficulty of implementing and deploying IntServ and RSVP in the Internet backbone and differs in the kind of service it provides. While IntServ provides per-flow guarantees, Differentiated Services (DiffServ) follows the philosophy of mapping multiple flows into a few service levels. At the boundary of the network, traffic entering a network is classified, conditioned and assigned to different behavior aggregates by marking a special DS (Differentiated Services) field in the IP packet header (TOS field in IPv4 or CLASS field in IPv6). Within the core of the network, packets are forwarded according to the per-hop behavior (PHB) associated with the DSCP (Differentiated Service Code Point). This eliminates the need to keep any flow state information elsewhere in the network.

IntServ over DiffServ provides a reservation-based QoS architecture with feedback signaling. It uses RSVP to signal resource needs but uses DiffServ as the technology to do the actual resource sharing among flows [8].

MANET routing protocol controls how nodes decide which way to route packets between computing devices in a mobile ad-hoc network .In *ad hoc networks*, the basic idea is that a new node may announce its presence and should listen for announcements broadcast by its neighbours. Each node learns about nodes nearby and how to reach them, and may announce that it, too, can reach them. Three proposed protocols are Reactive protocols (AODV), Proactive protocols – (OLSR), Hybrid protocol – (ZRP) have been accepted. Two of these are AODV uses the principals from *Distance Vector* routing and OLSR uses principals from *Link State* routing. A third approach, which combines the strengths of proactive and reactive schemes, is called a

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hybrid protocol.

Flexible QoS Model for MANET (FQMM) is designed to support QoS by mixing intserv with diffserv in smart way. The drawback of intserv & diffserv remains in FQMM [7]. In-band-signaling (INSIGNIA) and stateless wireless adhoc network (SWAN) are proposed for distributed and stateless distributed approaches respectively [2].

Therefore, we propose a QoS-aware routing protocol, which is based on residual bandwidth estimation during route set up. Our QoS-aware routing protocol is built off AODV, in which the routing table is used to forward packets, "Hello" messages are used to detect broken routes and "Error" messages are used to inform upstream hosts about a broken route.

3. Qos aware routing

QoS is an agreement to provide guaranteed services, such as bandwidth, delay, and packet delivery ratio to users. Supporting more than one QoS constraint makes the QoS routing problem complete [9]. Therefore, here consider only the bandwidth constraint when studying QoS-aware routing for supporting real-time video or audio transmission. Propose a QoS-aware routing protocol provides feedback about the available bandwidth to the application (feedback scheme). This require knowledge of the end-to- end bandwidth available along the route from the source to the destination.

Thus, bandwidth estimation is the key to supporting QoS. Work focuses on exploring the ways to estimate the available bandwidth, incorporating a QoS-aware scheme into the route discovery procedure and providing feedback to the application.

3.1 AODV protocol Overview

The protocol consists of two phases:

i) Route Discovery

ii) Route Maintenance.

In *routs discovery* when a source has data to transmit to an unknown destination, it broadcasts a Route Request (RREQ) for that destination. At each intermediate node, when a RREQ is received a route to the source is created. If the receiving node has not received this RREQ before, is not the destination and does not have a current route to the destination, it rebroadcasts the RREQ.

If the receiving node is the destination it generates a Route Reply (RREP). The RREP is unicast in a hop-by-hop fashion to the source. As the RREP propagates, each intermediate node creates a route to the destination. When the source receives the RREP, it records the route to the destination and can begin sending data. If multiple RREPs are received by the source, the route with the shortest hop count is chosen.

As data flows from the source to the destination, each node along the route updates the timers associated with the routes to the source and destination, maintaining the routes in the routing table. If a route is not used for some period of time, a node cannot be sure whether the route is still valid; consequently, the node removes the route from its routing table.

If data is flowing and a link break is detected, a Route Error (RERR) is sent to the source of the data in a hop-by-hop fashion. As the RERR propagates towards the source, each intermediate node invalidates routes to any unreachable destinations. When the source of the data receives the RERR, it invalidates the route and reinitiates route discovery if necessary. The following fig. 1 summarizes the action of an AODV routing protocol, when processing an incoming message. HELLO messages are excluded from the diagram for brevity.

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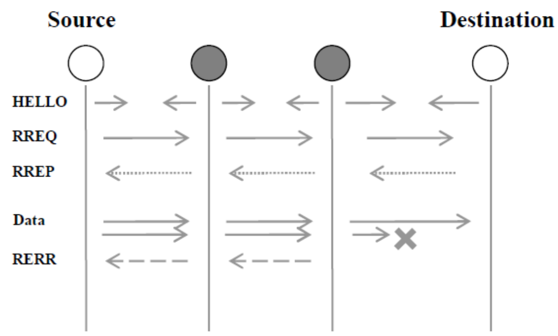

Fig. 1 AODV protocol review

The second phase i.e. *route maintenance* is performed by the source node and can be subdivided into: i) source node moves: source node initiates a new route discovery process, ii) destination or an intermediate node moves: a route error message (RERR) is sent to the source node. Intermediate nodes receiving a RERR update their routing table by setting the distance of the destination to infinity. If the source node receives a RERR it will initiate a new route discovery. To prevent global broadcast messages AODV introduces a local connectivity management. This is done by periodical exchanges of so called HELLO messages, which are small RREP packets containing a node's address and additional information

3.2. QoS routing for AODV

To offer bandwidth-guaranteed QoS, the available end-to-end bandwidth along a route from the source to the destination must be known. The end-to-end throughput is a concave parameter, which is determined by the bottleneck bandwidth of the intermediate hosts in the route. Therefore, estimating the end-to-end throughput can be simplified into finding the minimal residual bandwidth available among the hosts in that route. However, how to calculate the bandwidth using the IEEE 802.11 MAC is still a challenging problem, because the bandwidth is shared among neighboring hosts, and an individual host has no knowledge about other neighboring hosts' traffic status.

I. AVAILABLE BANDWIDTH ESTIMATION

Fig 2 shows the stages in the transmission of a single packet using the IEEE 802.11 DCF MAC protocol.

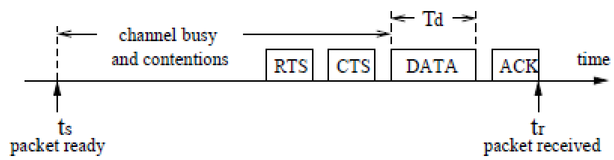

Fig. 2 IEEE 802.11 packet transmission

Throughput can be measure by transmitting a packet as-

$$TP = \frac{S}{t_r - t_s} \qquad \text{------ (1)}$$

Where,

TP is Throughput,

S is size of the packet,

t_r is the time the ACK received,

t_s is the time the packer is ready for transmission.

The time interval $t_r - t_s$ includes the channel busy and contention time. Separate throughput estimates should be kept to different neighbors because the channel conditions may be very different to each one.[13]

This link layer measurement mechanism captures the effect of contention time on available bandwidth. If contention is high, $t_r - t_s$ will increase and the throughput TP will decrease. This mechanism also captures the effect of fading and interference errors because if these errors affect the RTS or DATA packets, they have to be re-transmitted. This increases $t_r - t_s$ and correspondingly decreases available bandwidth. Our available bandwidth measurement mechanism thus takes into account the phenomena causing it to decrease from the theoretical maximum

ISSN (online): 2250-141X
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channel capacity. It should be noted that the available bandwidth is measured using only s*uccessful* link layer transmissions of an ongoing data flow.

It is clear that the measured throughput of a packet depends on the size of a packet. Larger packet has higher measured throughput because it sends more data once it grabs the channel. To make the throughput measurement *independent* of packet size, we normalize the throughput of a packet to a pre-defined packet size. In Fig 2,

$$T_d = \frac{S}{BW_{ch}} \quad \text{----(2)}$$

Where T_d is the actual time for the channel to transmit the data packet and BW_{ch} is the channel's bit-rate. Here we assume the channel's bit-rate is a pre-defined value. The transmission times of two packets should differ only in their times to transmit the DATA packets. Therefore, we have:

$$(t_{r1} - t_{s1}) - \frac{S_1}{BW_{ch}} = (t_{r2} - t_{s2}) - \frac{S_2}{BW_{ch}} \quad \text{------(3)}$$

$$= \frac{S_2}{TP_2} - \frac{S_2}{BW_{ch}} \quad \text{-----(4)}$$

where S_1 is the actual data packet size, and S_2 *is* a pre-defined standard packet size. By Equation (4), we can calculate the normalized throughput TP_2 from the standard size packet.

Obviously, the raw throughput depends on the packet size; larger packet size leads to higher measured throughput. The normalized throughput, on the other hand, does not depend on the data packet size. Hence, we use the normalized throughput to represent the bandwidth of a wireless link, to filter out the noise introduced by the measured raw throughput from packets of different sizes.

We measure the bandwidth of a link in discrete time intervals by averaging the throughputs of the recent packets in the past time window and use it to estimate the bandwidth in the current time window. This estimation may not be accurate because the channel condition may have changed.

We measure and normalize the throughput for every 2 seconds using the average of packet throughputs in the past time window. Results show 15% error due to environment condition, channel errors due to physical object. Thus conclusion can be made that using average throughput of past packets to estimate current bandwidth is feasible and robust.

This algorithm is implemented in C++ and available to OTCL through an OTCL linkage that is implemented using tclcl. The whole thing together makes NS, which is a OO extended TCL interpreter with network simulator libraries.

II. Establishing a route with QOS parameter

The proposed scheme for QoS-aware routing protocol is based on bandwidth estimation during route set up. QoS-aware routing protocol is built off AODV, in which the routing table is used to forward packets, "Hello" messages are used to detect broken routes and "Error" messages are used to inform upstream hosts about a broken route.

Here when host want to send the data it has to listen to the channel and estimate the available bandwidth based on the ratio of free and busy times ("Listen" bandwidth estimation).

The main idea is to establish AODV routing with the QoS parameters. Performance matrix as a extension is added to the route message during route discovery. In order to handle QoS extension some changes are necessary in routing tables.

Four new fields to be added for QoS-AODV or performance matrix include:

i) No. of packet receiver ii) throughput iii) Bandwidth iv) Delay.

A node may be the destination of different sessions with different level of QoS, the

ISSN (online): 2250-141X
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routing tables should be maintaining per session and updated according to the respective value [1].

To estimate the available bandwidth, each host can listen to the channel to track the traffic state and determine how much free bandwidth it has available every second. Hosts are allowed to access the wireless channel when the media is free. The media can be free if no hosts are transmitting packets.

The IEEE 802.11 MAC utilizes both a physical carrier sense and a virtual carrier sense [via the network allocation vector (NAV)], which can be used to determine the free and busy times. The MAC detects that the channel is free when the following three requirements are met:

- NAV's value is less than the current time;
- receive state is idle;
- send state is idle.

The MAC claims that the channel is busy when one of following occurs:

- NAV sets a new value;
- receive state changes from idle to any other state;
- send state changes from idle to any other state.[3]

A host estimates its available bandwidth for new data transmissions as the channel bandwidth times the ratio of free time to overall time, divided by a weight factor. The weight factor is introduced due to the nature of IEEE 802.11. The DIFS, SIFS, and back-off scheme represent overhead, which must be accounted for in each data transmission. This overhead makes it impossible in a distributed MAC competition scheme to fully use the available bandwidth for data transmission.

Minimum bandwidth is a field which indicate requested amount of bandwidth for a specific route. When an intermediate host receives the RREQ packet, it first calculates its residual bandwidth. The host compares its residual bandwidth with the requested bandwidth.

If its residual bandwidth is greater than the requested bandwidth, it forwards this RREQ. Otherwise, it discards this RREQ. The host compares its residual bandwidth with the min-bandwidth field in the RREQ. If its residual bandwidth is greater than the min-bandwidth, it forwards the RREQ. Otherwise, it updates the min-bandwidth value using its residual bandwidth.

When the destination host receives the RREQ, it also check as described above. However, after completing this checking procedure, we can't say that the network can offer the min-bandwidth indicated in the RREQ packet. We can't put this kind of potential interference into consideration while estimating the residual bandwidth during the route discovery procedure. Therefore, one final check procedure is required before sending the RREP packet back to the source host. We directly use the relation of the end-to-end throughput with the no. of hops and the bottleneck bandwidth in the route as follows:

$$\text{totalBW}=\frac{oldBW+cur\ BW}{No.of\ hop}$$

This equation offers the upper bound of the available bandwidth. Finally, the destination host sends the RREP with a modified min-bandwidth to the source host. Once intermediate hosts receive the RREP, they enable the route and also record the min-bandwidth in their routing table.

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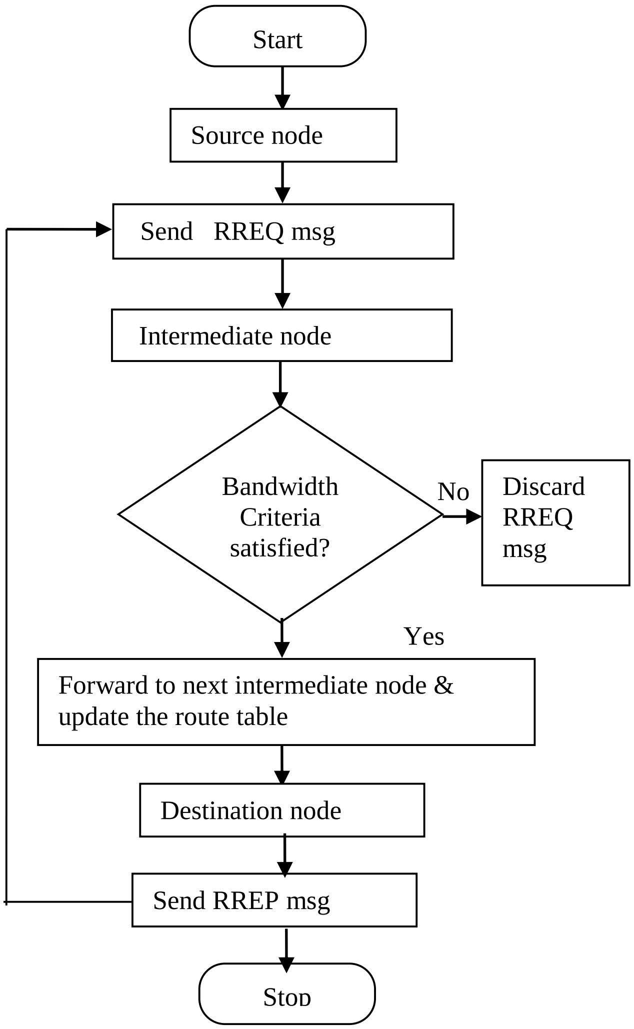

Fig.3. overview of QoS-AODV protocol

Using this method to estimate residual bandwidth is straightforward. However, using this approach, the host cannot release the bandwidth immediately when a route breaks, because it does not know how much bandwidth each node in the broken route consumes. "Listen" only counts the used bandwidth but does not distinguish the corresponding bandwidth cost for each flow. This greatly affects the accuracy of bandwidth estimation when a route is broken.

A simple overview of all this operation can be shown by flowchart as shown in fig. 3. Here these steps or operations all added in normal AODV protocol as shown in fig. 1 or we can say these are the extension added to normal AODV protocol.

In the MAC layer, ready-to-send (RTS), clear-to-send (CTS), and acknowledgment (ACK) packets consume bandwidth, the back-off scheme cannot fully use the entire bandwidth, and packets can collide, resulting in packet retransmissions. Furthermore, the routing protocol needs some overhead to maintain or discover the routes.

4. Simulation and result

To test the performance of our QoS-aware routing protocol, we will perform simulations using NS-2(version 2.34). We use the IEEE 802.11 MAC protocol in RTS/CTS/Data/ACK mode with a channel data rate of 2 Mbps. 25 mobile nodes are moving in 600 by 600 meter flat space. Attach CBR (Constant Bit Rate) application that generates constant packets through the TCP connection. CBR packet size is chosen to be 512 bytes, data rate is set to 1 Mbps. Duration of the scenarios is 10 seconds. Appropriate positions of the nodes are defined manually. During simulation 5 different source nodes want to sand data to five different destinations.

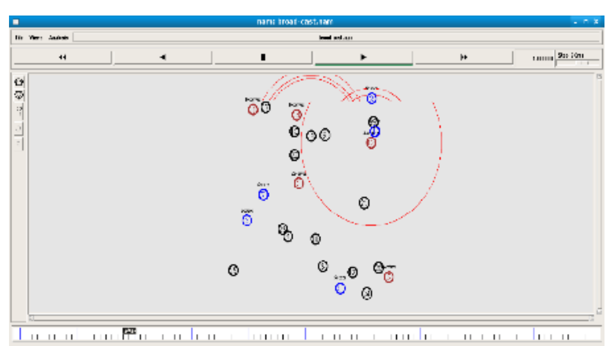

Fig 4. Nam window

Fig. 4 shows the nam window consists of 25 nodes. Blue nodes are sources and red nodes

ISSN (online): 2250-141X
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are destination.

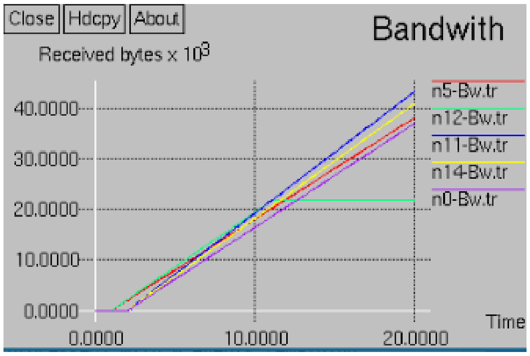

Fig. 5. No. of bytes received

All the above approaches do not consider that the supported bandwidth should be less than the bandwidth available during the route discovery, which is caused by the potential bandwidth sharing brought by the new routes.
Delay is perhaps the most important parameter to be considered for video traffic. Figure 6 shows the variation in average delay as we increase the number of nodes.

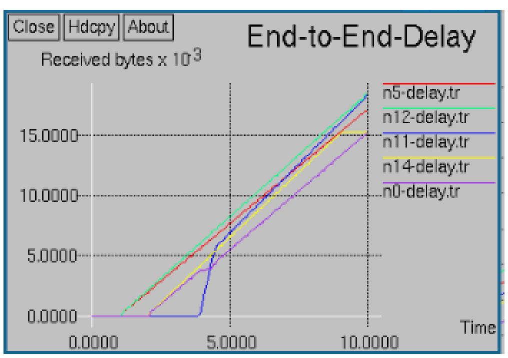

Fig. 6 End-to-End delay

Moreover, we also show the increase in delay due to node mobility. We can easily see from Figure 6 that the greater the mobility, the higher the delay and vice versa.

We also took power minimization issue into consideration and minimize power consumption. Simulation results showed that this strategy does minimize power consumption and it did not degrade multihop communication improvement.

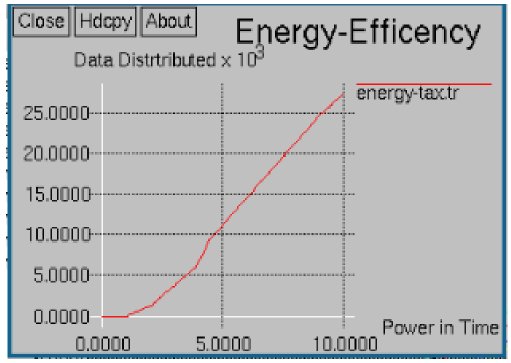

Fig 7 Energy consumption

5. Future work

We will compare two different methods of estimating bandwidth. Theoretically, the “Hello” bandwidth estimation using two-hop method performs better than the “Listen” bandwidth estimation method when releasing bandwidth immediately is important “Listen” performs better in term of packet delivery ratio. From the perspective of overhead, “Listen” does not add extra overhead. In our protocol, we have not incorporated any predictive way to foresee a route break, which causes performance degradation in mobile topologies. Therefore, some methods such as preemptive maintenance routing and route maintenance based on signal strength might help to reduce the transient time when the required QoS is not guaranteed due to a route break or network partition, so that the routing protocol can react much better to mobile topologies [3]. In a real scenario, shadowing will cause a node’s transmission range to vary, and it will not be the ideal circle that is assumed here. How to incorporate these no idealities into our protocol is the subject of our future research.
Furthermore, future study incorporates call admission control in MANET. Two approaches: distributed and local admission control has been added in future work. The

ISSN (online): 2250-141X
Vol. 2 Issue 3, June 2012

study shows that the local admission control (EWGTW) **is** more performing than approach based only on DPS in terms of admitted calls, total satellite utilization and signaling overhead [10].

References

[1] De Renesse R, Ghassemian M, *Friderikos V, Aghvami A.H "QoS enabled routing in mobile ad hoc networks" Dept. of Electr. Eng., King's Coll., London, UK 3G Mobile Communication Technologies, 2004. Fifth IEE International Conference on* Publication Date: 2004 On page(s): 678- 682

[2] Chenn-Jung Huang,1,4 Wei Kuang Lai,2 Yi-Ta Chuang,3 and Sheng-Yu Hsiao2 "A Dynamic Alternate Path QoS Enabled Routing Scheme in Mobile Ad hoc Networks" vol. 14, No. 1, march 2007, 1-16

[3] L. Chen and W. B. Heinzelman, "QoS-aware routing based on bandwidth estimation for mobile ad hoc networks," IEEE J. Selected Areas in Comm., vol.23, no. 3, pp. 561-572, Mar, 2005.

[4] Tung-Shih Su, Chih-Hung Lin, Wen-Shyong Hsieh "A Novel QoS-Aware Routing for Ad Hoc Networks"Department of Computer Science and Engineering National Sun Yat-Sen University

[5] Ian D. Chakeres and Elizabeth M. Belding-Royer, "AODV Routing Protocol Implementation Design" Dept. of Electrical & Computer Engineering. University of California, Santa Barbara, idc@engineering.ucsb.edu and Dept. of Computer Science University of California, Santa Barbara ebelding@cs.ucsb.edu

[6] Chengyong Liu, Kezhong Liu, Layuan Li; "Research of QoS-Aware Routing Protocol with Load Balancing for Mobile Ad Hoc Networks", Wireless Communications, Networking and Mobile Computing, 2008. WiCOM '08. 4th International Conference on Digital Object Identifier: 10.1109/WiCom.2008.647, 2008, Page(s): 1 – 5

[7] Hannan Xiao, Seah W.K.G., Lo A., Chua, K.C.; "A flexible quality of service model for mobile ad-hoc networks", Vehicular Technology Conference Proceedings, 2000. VTC 2000-Spring Tokyo. 2000 IEEE 51st Volume: 1, Digital Object Identifier: 10.1109/VETECS.2000.851496, 2000, Page(s): 445 - 449 vol.1

[8] Georgy Sklyarenko, "AODV Routing Protocol" Institut f¨ur Informatik, Freie Universit¨at Berlin, Takustr. 9, D-14195 Berlin, Germany

[9] Jianbo Xue and Prof. Dr. Gustavo Alonso, The master thesis on "Quality of Service for Mobile Ad Hoc Networks" Eidgenossische Technische Hochschule swiss federal institute of technology Zurich, March 2003.

[10] F.De Rango, M.Tropea, S.Marano; "Call Admission Control for Integrated Diff-Serv Terrestrial and Int-Serv Satellite Network", D.E.I.S. department, University of Calabria, Rende, Italy.

[11] Kevin Fal and Kannan Varadhan, The *ns* Manual, The VINT Project, A Collaboration between researchers at UC Berkeley, LBL, USC/ISI, and Xerox PARC. hkfall@ee.lbl.govi, hkannan@catarina.usc.edui, January 6, 2009.

[12] Dhurandher S K, Misra S, Obaidat M S, Bansal V, Singh P, Punia V; "An Energy-Efficient On-Demand Routing algorithm for Mobile Ad-Hoc Networks", Electronics, Circuits and Systems, 2008. ICECS 2008. 15th IEEE International Conference on Digital Object Identifier: 10.1109/ICECS.2008.4675014, 2008,

ISSN (online): 2250-141X
Vol. 2 Issue 3, June 2012

page(s): 958 – 961

[13] Samarth H. Shah, Kai Chen, Klara Nahrstedt. ”Available Bandwidth Estimation in IEEE 802.11-based Wireless Networks” Department of Computer Science University of Illinois at Urbana-Champaign.