A Survey of QoS parameters through reactive routing in MANETs

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Abstract

Traditional internet QoS protocols like RSVP cannot be easily migrated to the wireless environment due to the error-prone nature of wireless links and the high mobility of mobile devices. The Internet Engineering Task Force (IETF), the body responsible for guiding the evolution of the Internet, provides the definition as given below: A mobile ad hoc network (MANET) is an autonomous system of mobile routers (and associated hosts) connected by wireless links. The routers are free to move randomly and organize themselves arbitrarily; thus, the network's wireless topology may change rapidly and unpredictably. Such a network may operate in a stand-alone fashion, or may be connected to the larger Internet. QoS is usually defined as 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 is expected to guarantee a set of measurable pre-specified service attributes to the users in terms of end-to-end performance, such as delay, bandwidth, probability of packet loss, delay variance (jitter), etc.

Keywords: Mobile Ad Hoc Networks(MANETs), Quality of Service(QoS), Ad Hoc On Demand Distance Vector (AODV), Route Request(RREQ), Route Reply (RREP).

1. Introduction

In the field of computer networking and other packet-switched telecommunication networks, the traffic engineering term quality of service (QoS) refers to resource reservation control mechanisms rather than the achieved service quality. QoS is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow. For example, a required bit rate, delay, jitter, packet dropping probability and/or bit error rate may be guaranteed. Quality of service guarantees are important if the network capacity is insufficient, especially for real-time streaming multimedia applications such as voice over IP, online games and IP-TV, since these often require fixed bit rate and are delay sensitive, and in networks where the capacity is a limited resource, for example in cellular data communication. Providing suitable quality of service (QOS) support for the delivery of real-time audio, video and data in mobile ad hoc networks presents a number of significant technical challenges. QoS is sometimes used as a quality measure, with many alternative definitions, rather than referring to the ability to reserve resources. Quality of service [13] sometimes refers to the level of quality of service, i.e. the guaranteed service quality. High QoS is often confused with a high level of performance or achieved service quality, for example high bit rate, low latency and low bit error probability. The research on QoS support in MANETs spans over all the layers in the network: 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 above 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. The functionality of QoS signaling is determined by the QoS model. QoS routing is part of the network layer and searches for a path with enough resources but does not reserve resources [14]. QoS MAC 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, and provides resource reservation. Many things can happen to packets as they travel from origin to destination, resulting in the following problems as seen from the point of view of the sender and receiver [15]:

1.1. Low throughput

Due to varying load from other users sharing the same network resources, the bit rate (the maximum throughput) that can be provided to a certain data stream may be too low for real time multimedia services if all data streams get the same scheduling priority.

1.2. Dropped packets

The routers might fail to deliver (drop) some packets if their data is corrupted or they arrive when their buffers are already full.

1.3. Errors

Sometimes packets are corrupted due to bit errors caused by noise and interference, especially in wireless communications and long copper wires.

1.4. Latency

It might take a long time for each packet to reach its destination, because it gets held up in long queues, or takes a less direct route to avoid congestion. This is different from throughput, as the delay can build up over time, even if the throughput is almost normal.

1.5. Jitter

A packet's delay varies with its position in the queues of the routers along the path between source and destination and this position can vary unpredictably [17]. This variation in delay is known as jitter and can seriously affect the quality of streaming audio and/or video.

1.6. Out-of-order delivery

When a collection of related packets is routed through a network, different packets may take different routes, each resulting in a different delay. The result is that the packets arrive in a different order than they were sent. This problem requires special additional protocols responsible for rearranging out-of-order packets to an isochronous state once they reach their destination.

2. Related Work

To provide quality of service (QoS) through minimizing interference is particularly challenging for mobile ad hoc networks (MANETs) due to frequent movement and formation of dynamic connections. The research identifies the channel interference and the collision problems in MANETs, and proposes a new algorithm (called the distributed channel assignment control) to efficiently solve the identified problems. The channels in the proposed algorithm are dynamically negotiated to enable multiple communications in different channels in the same region. This channel control algorithm can extend channel connectivity and reduce interference. It is assumed that if four channels are provided in IEEE 802.11 standards, it is common for more than one non-interfering channel to be available. Channel assignment and power control based in these systems are considered. The proposed channel assignment control algorithm can also reduce channel interference and maintain channel connectivity in dense environments. To evaluate the proposed algorithm, an analytical model is used: it investigates the possible success ratio versus number of nodes and how many channels are needed to support QoS. The channel bandwidth is determined by the number and size of the control packets. Thus, after an extensive study of the exposed node problem, the channel assignment control using power control is suggested as a potential solution. The objective of the proposed algorithm is to provide an upper bound for the number of channels required to provide a collision-free communication environment. Because every node in MANETs has the same mobility, it is important to compute the number of channels required to guarantee channel connectivity. To solve this problem, a new algorithm, named a distributed channel assignment control, is proposed that focuses on performance enhancements related to QoS and mathematical analysis techniques for the channel bandwidth. This novel algorithm uses channel assignment control with a power control to reduce the negative effects induced by the quasi-exposed node problem, and then the channels are adaptively negotiated to allow communication in the interference region. The proposed algorithm has been evaluated via extensive simulations, and the results show that it can successfully guarantee QoS by maintaining good throughput, reducing control message overhead, and enhancing delay. The issues surrounding the provision of reliable QoS and consistent reviews of the existing dominant schemes in MANETs [12] have been provided. After investigating the existing schemes, a quasi-exposed node problem has been designed, a novel distributed channel assignment control algorithm has been proposed, and a mathematical analysis technique has been provided. Through simulations and mathematical analyses, it was illustrated that a novel solution could guarantee reliable QoS when interference exists in MANETs. To solve the problems addressed, a channel assignment with power control capability has been added to the algorithm. The proposed algorithm locally assigns channels to nodes in such a way that interference and connection loss among nodes are avoided; thus, the channel availability is maximized. The proposed algorithm is particularly useful in avoiding interference and path failures in high mobility environments since the number of messages involved was reduced. The analytical boundary used for the worst case estimation for bandwidth, total number of channels, and control packets [8]. It was therefore concluded that the relationship between channel utilization and bandwidth utilization becomes equal at all traffic densities [1] and communication drop ratio is shown in Fig. 1.

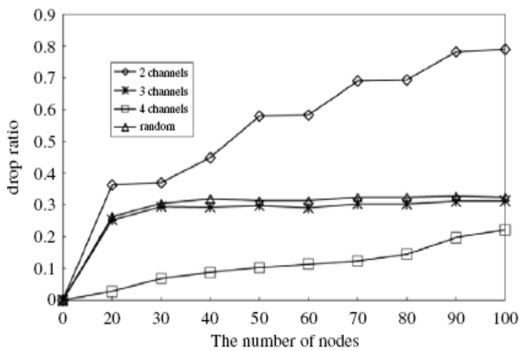

Figure 1. Communication Drop Ratio

To achieve efficient fault tolerance, FDCB is augmented so that the cluster-head has complete "cluster-state" knowledge. The cluster-head has connectivity awareness for all cluster nodes. This awareness includes knowledge of all QoS connections currently supported by each cluster member, each member's resource availability, and the cluster topology. With this scheme, when cluster node i leaves the cluster, due to mobility or failure, and the QoS paths supported by i are broken, the cluster-head has all information required to begin a renegotiating to reestablish the connection with minimal delay if possible. The cluster-head collects this knowledge via two processes: communication with the other clusters via clustered FSR and local clustered information exchange. These processes ensure, with high probability and low overhead, that knowledge of the systems' state is maintained both to repair existing paths and to initiate new ones[2]. When a source node wants to send data to a group of destinations, an efficient communication procedure is done between cell leaders to provide the source with all the nodes interested in this multicast session and their positions. Now the source will be able to divide the group members into manageable sub-groups and choose a coordinator for each subgroup to start the multicast [11] session. The QoS requirements that have been taken into consideration here are bandwidth and delay.

The effects of two different MAC protocols - IEEE 802.11and IEEE802.11e with Ad Hoc On-Demand Distance Vector (AODV) and Dynamic Source Routing (DSR) [16] of routing algorithms with Slow start and Arithmetic Increase and Multiplicative Decrease (AIMD) mechanism of TCP. IEEE802.11 uses distributed coordination function (DCF) where IEEE802.11e uses enhanced distributed coordination function (EDCF). Results [9] show that in all instances, the QoS parameters 15-20% improvement in throughput, 40-45% improvement in bandwidth delay product, 10-15% improvement in delivery ratio, packet loss is reduced drastically to 40-50% in IEEE802.11e with AODV [7] algorithm in network layer and slow start mechanism in transport layer. Similarly Results shows, the QoS parameters 35-40% improvement in throughput, 25-30% improvement in bandwidth-delay product, 15-20% improvement in delivery ratio, packet loss is reduced drastically to 20-25% in IEEE802.11e with AODV algorithm in network layer and AIMD mechanism in transport layer [4].

In DSR, overhearing leads to bad situation because stale route concept is applied to all unconditional overheard nodes and wastes energy resource while transmitting, receiving, rebroadcast and unconditional overhearing. DSR broadcasts control packets which waste channel capacity because it generates redundant rebroadcasts. Hence AODV routing protocol is proposed. Overhearing improves routing efficiency but consumes energy. Overhearing is caused by the fact that when a unicast packet transaction is carried out in a node's immediate neighborhood, it does not have any mechanism for not to receive that packet. Energy [10] expenditure during overhearing is same as that during reception. Goal of this work is to develop an energy conserving mechanism to reduce energy expenditure due to overhearing. Overhearing [5] improves network performance since nodes collect more route information. Nodes in the range of source learn about path to destination. But overhearing increases traffic and consumes energy. The proposed algorithm controls the level of overhearing. It reduces energy consumption without affecting quality of route information.

3. Our Proposal

In this section, we present a high-level overview of our proposed solution. The idea of leveraging node mobility to improve QoS can potentially be applied to any QoS-aware routing scheme. To demonstrate this idea, we chose to work with our modification of the AODV routing protocol, called Q-AODV, that enhances AODV to be aware of resource requirements. In order to discover a route, AODV broadcasts a route request message (RREQ) throughout the network. When this message reaches the destination node, the destination sends a route reply (RREP) along the travelled path back to the source. AODV does not consider resource availability at nodes, and simply assumes that the best path is the one with the least number of hops. Q-AODV is a stateless protocol. Resource availability is only checked during flow admission. Resource availability can change as nodes move, and is therefore not guaranteed. For this reason, no resource reservations are made. We do not perform any flow policing and assume that flows are well-behaved. Flow monitoring and adjustment of flow rate is also left to the application. We extend the AODV Route Request (RREQ) packet to include the following fields: minimum bandwidth, maximum bandwidth and bottleneck bandwidth. The first two fields specify the bandwidth requirements of the application and are populated by the

source node. The AODV RREQ flooding procedure is then followed, with one modification: a node forwards the RREQ only if it has enough bandwidth available to satisfy the request. The bottleneck bandwidth field contains the least bandwidth encountered along the path, and is appropriately populated by nodes forwarding the RREQ. On receiving the RREQ, the destination replies with a Route Reply (RREP). The RREP is extended to include a field specifying the available bandwidth on the path (this is obtained from the bottleneck bandwidth field in the RREQ). The RREP is then forwarded back to the source as in unmodified AODV with no additional processing at intermediate nodes.
To ensure that sufficient bandwidth is available on the end-to-end path, the RREQ must traverse the complete route to the destination. For this reason, no intermediate node is allowed to reply to the RREQ; only the destination may respond. Also, traffic from different sources to the same destination may need to be forwarded along different paths based on session requirements and resource availability. For this reason, Q-AODV [6] discovers and maintains routes on a per-source-destination-pair basis rather than a per-destination basis as in AODV. In Q-AODV, unlike AODV, it is beneficial for the destination to reply to multiple RREQ instances, since the first RREQ instance to reach the destination need not have travelled along the most resourceful path; another path with higher bandwidth availability may be present. The source can then select the best path from among the RREPs received. After selecting the best path, the source informs the destination of its decision by sending a message along the path, so that the destination can use the same path for reverse communication.

3.1. Performance Metrics

We measure the following performance metrics to evaluate the effectiveness and efficiency [14] of the protocol:

(a) Packet delivery fraction: This is the fraction of data packets sent by the server that are received by the client. The higher the packet delivery fraction, the more effective the protocol in reducing network congestion and the better the user experience in viewing/hearing the media stream.

(b) Number of data packets received: This is the total number of data packets received by the clients. It indicates the data throughput obtained. The packet delivery fraction alone is not enough to indicate the effectiveness of the protocol in delivering packets, since the number of packets sent with each routing protocol may be different (the QoS routing protocols do not admit a session unless a route with the required QoS is available). This metric therefore complements the packet delivery fraction in determining protocol effectiveness.

(c) Average end-to-end packet delay: This is the average end-to-end delay of packets that are received by the client. As this value decreases, the congestion in the network is reduced, and packets are more likely to be received in time for playout.

(d) Fraction of received packets with unacceptable end-to-end delay: Many multimedia applications, such as Internet telephony and live media broadcast, place an upper limit on acceptable end-to-end packet delay. For example, in typical voice applications, packets with end-to-end delay greater than 400 milliseconds arrive too late to be played out. Such packets waste the resources of the network since they traverse the path between the source and destination but are received too late to be used by the destination. This metric measures the fraction of received packets that are thus unacceptably delayed. A low value for this fraction indicates more efficient usage of network resources and better quality perceived by the user.

4. Simulation Results

The proposed strategy is implemented using NS2 simulator tool. Performance analysis of AODV is carried out by setting simulation time to 200s with a grid size of 1000×1000 m. Random way point mobility model with CBR (Constant Bit Rate) traffic is used to simulate nodes movement. This algorithm is tested with 30 nodes. The parameters used to measure the performance are number of packets received, delay, energy consumption and packet delivery ratio. Table 1 shows the simulation parameters. The results are compared with existing DSR protocol.

Table 1. Simulation Parameters

Parameter	Value
Transmission range	250 m
Carrier Sensing range	550 m
Packet Size	512 bytes
Channel Capacity	2 Mbps
Grid Size	1000×1000 m
No. of nodes	30
Mobility Speed	20 s
Simulation Time	200 s

Figure 2 shows the number of packets received by AODV and DSR method. Number of packets received in energy efficient AODV is more than DSR.

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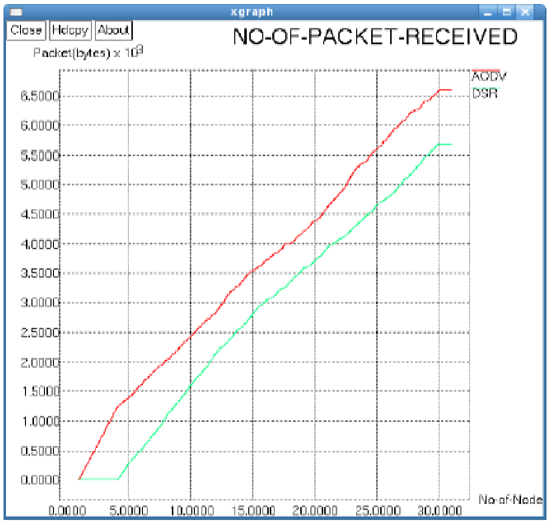

Figure 2. No. of Packets Received

The packet delivery ratio (PDR) is calculated as the ratio of the data packets delivered to the destination to those transmitted by the CBR traffic. The ability to deliver a high percentage of packets to a destination increases the overall utility of the system. PDR is high because most of the nodes are participated in packet transmission as shown in figure 3.

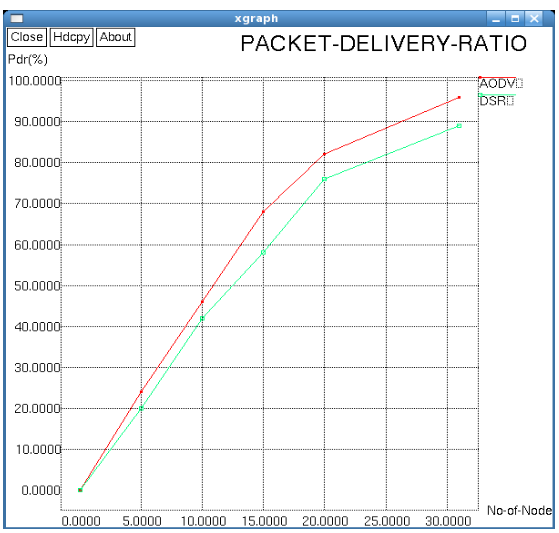

Figure 3. Packet Delivery Ratio (PDR)

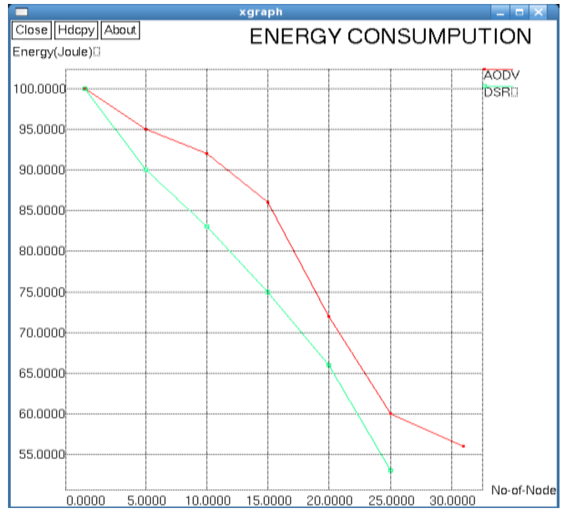

Figure 4. Energy Consumption

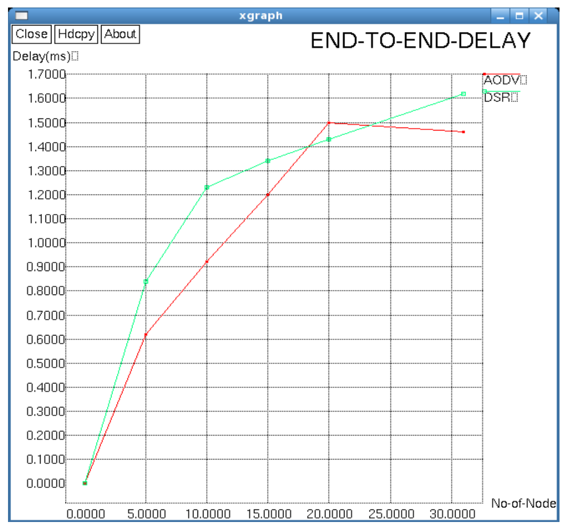

Figure 5. End-to-End Delay

Probability based overhearing method outperforms other algorithms with respect to energy consumption. Energy efficient AODV shows less energy consumption than DSR [16] as in Figure 4. This proposed approach with energy management still reduces the energy consumption. Effective energy management is obtained in proposed work. This is due to the variation of the transmit power between two nodes and also reduction in the number of overhearing nodes. This proposed approach with energy management still reduces the energy consumption. Figure 5 shows that the delay caused by energy efficient AODV is less than DSR.

5. Conclusion

Results show that the idea is indeed effective in mobile ad hoc networks with low to medium mobility, and can be accomplished with low overhead. In the future, we plan to examine how other QoS-aware routing protocols [14] can be extended to use this idea. We will also look at how this approach can be made to work in the presence of obstacles. With obstacles, some potential neighbors may be unable to directly receive the high power broadcast, and so a multi-hop mechanism may be necessary. In particular, we plan to remove the restriction imposed by the maximum transmission range and design a solution that enables a mobile node to look beyond a single hop to find a location with better connectivity. Finally, the idea of leveraging mobility to improve service can also be extended to intermediate nodes on a network path in some deployment scenarios. Our future work intends to be the comparison between reactive and proactive routing schemes to come out with a more reliable and secure routing protocol.

6. References

[1] Kyungjun Kim, "A distributed channel assignment control for QoS support in mobile ad hoc networks", Journal of Parallel and Distributed Computing 71 (2011) 335–342.

[2] Larry C. Llewellyn, Kenneth M. Hopkinson, Member, IEEE, and Scott R. Graham, Member, IEEE, "Distributed Fault-Tolerant Quality of Wireless Networks", IEEE TRANSACTIONS ON MOBILE COMPUTING, VOL. 10, NO. 2, FEBRUARY 2011.

[3] Mohammad M. Qabajeh, Aisha-Hassan A. Hashim, Othman Khalifa and Liana K. Qabajeh, "Position-based QoS Multicast Routing Protocol for Mobile Ad Hoc Networks", International Conference on Computer and Communication Engineering (ICCCE 2010), 11-13 May 2010, Kuala Lumpur, Malaysia.

[4] J.Premalatha and P.Balasubramanie (2010), "Enhancing Quality of Service in MANETS by Effective Routing", 978-1-4244-5137-1/10/2010 IEEE.

[5] Sanzgiri and Elizabeth M. Belding-Royer, "Leveraging Mobility to Improve Quality of Service in Mobile Networks".

[6] C. Zhu and S. CorsonC, "QoS Routing for Mobile Ad Hoc Networks", IEEE Infocom, 2002.

[7] C. Perkins and E. Royer, "Ad-hoc On-Demand Distance Vector Routing", IEEE WMCSA, 1999.

[8] Adib M.Monzer Habbal and Suhaidi Hassan, "Loss Detection and Recovery Techniques for TCP in Mobile Ad Hoc Network", SMIEEE in 2010 Second International Conference on Network Applications, Protocols and Services.

[9] K. Kang and H.-J. Suh, "Adaptive buffer control to minimize delay and guarantee service reliability", The Institution of Engineering and Technology (IET) Communications, 2011, Vol. 5, Issue 1, pp. 110–118.

[10] N.Sumathi and Dr. Antony Selvadoss Thanamani, "Evaluation of Energy Efficient Reactive Routing Protocols in QoS Enabled Routing for MANETS", IJCA (0975 – 8887) Volume 14– No.2, January 2011.

[11] Thomas J. Sapienza, "Optimizing Quality of Service of Wireless Mobile Ad-hoc Networks using Evolutionary Computation", CSIIRW '08, May 12-14, Oak Ridge, Tennessee, USA.

[12] K. Dridi, K. Djouani and B. Daachi, "Three Scheduling-levels algorithm for IEEE 802.11 QoS efficiency improvement", Proceedings of MoMM2008, Linz, Austria.

[13] Roch A. Guerin and Ariel Orda, "QoS Routing in Networks with Inaccurate Information: Theory and Algorithms", IEEE/ACM TRANSACTIONS ON NETWORKING, VOL. 7, NO. 3, JUNE 1999.

[14] http://www.scribd.com/doc/55865371/Improving-QoS-in-Mobile-Networks-Through-Effective-Routing

[15]http://www.en.wikipedia.org/wiki/Quality_of_service

[16] N.Sumathi and Dr. Antony Selvadoss Thanamani, "Evaluation of energy efficient reactive routing protocols in QOS enabled routing in MANETs", International Journal of Computer Applications (0975 – 8887), Volume 14– No.2, January2011.

[17] http://www.connection-telecom.com/news/142-voip-quality

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