Data Management Routing Protocol in MANET

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Abstract

Providing quality of service (QOS) in a mobile ad hoc network is a challenging task due to its peculiar characteristics. This paper aims at presenting a routing protocol which identifies data volume to be sent, based on the data volume the route selected. If large volume of data wants to sent then the routing protocol consider the multiple node-disjoint paths are examined for satisfying QOS in terms of end-to-end delay and window-based measurements of channel estimation is performed. If small volume of data want to sent the routing protocol consider end to end delay only. In this paper analyze the data volume and end-to-end delay along the paths taking into account the IEEE 802.11 contention delays and outstanding capacity. Data Management routing scheme is proposed. To study the performance of routing System has carried out simulation. The results show that the proposed protocol performs better in terms of QOS satisfaction ratio and the throughput as compared to an existing protocol.

Keywords

mobile ad hoc networks; quality-of-service; capacity estimation; admission control; channel utilization, delay; contention delay; contention area; re-sequencing delay; node-disjoint paths, data volume.

1. INTRODUCTION

Ad-hoc networks are autonomous, nature-prepared, wireless, and mobile networks. They do not require setting up any fixed infrastructure such as access points, as the nodes organize themselves automatically to transfer data packets and manage dynamic topology due to mobility.

Many of the current contributions in the ad-hoc networking community assume that the underlying wireless technology is the IEEE 802.11 standard due to the broad availability of interface cards and simulation models. This standard provides an ad hoc mode, allowing mobiles to communicate directly. As the communication range is limited by regulations, a distributed routing protocol is required to allow long distance communications. However, this standard has not been targeted specially for multi-hop ad hoc operation,and it is therefore not perfectly suited to this type of networks.

Nowadays, several applications generate multimedia data flows or rely on the proper and efficient transmission of sensitive control traffic. These applications may benefit from a quality of service (QOS) support in the network. That is why this domain has been extensively studied and more and more QOS solutions are proposed for ad hoc networks. However, the term QOS is unclear and gathers several concepts. Some protocols intend to offer strong guarantees to the applications on the transmission characteristics, for instance bandwidth, end to end delay, packet loss, or network load. Other solutions, which seem more suited to a mobile environment, only select the best route among all possible choices regarding the same criteria.

In both cases, an accurate evaluation of the capabilities end to end delay of the routes is necessary. Most of the current QOS proposals leave this problem aside, relying on the assumption that the link layer protocols are able to perform such an evaluation. However, they are not. The resource evaluation problem is far from being trivial as it must take into account several phenomena related to the wireless environment but also dependent on less measurable parameters such as the node mobility. Throughout this paper, we will focus on one of the fundamental resources: throughput and delay. Estimating the outstanding bandwidth at a given time and in a given part of the network is tricky because, in a wireless network, the medium is shared between close nodes. Consequently, computing the available bandwidth between two neighbor nodes necessitates an accurate identification of all potential contenders at the emitter’s side, of all potential scramblers at the receiver’s side, and a proper evaluation of their impact. Information about nodes’ utilization of the shared resource should, therefore, be gathered and composed to derive the amount of free resources. Both tasks are usually difficult to realize and they become even harder in sparse networks, as two nodes may share the medium without being able to directly exchange information. In this paper, we present a new method to evaluate the available bandwidth and end to end delay in ad hoc networks based on the IEEE 802.11 MAC layer. This method uses the nodes’ carrier sense capability combined to other techniques such as collision prediction to perform this estimation.

2. RELATED WORKES

Devising a routing protocol for an ad hoc network with a provision of quality of service (QOS) is a challenging task due to their inherent characteristics. A lot of work is directed towards identification of multiple paths from the point of view of fault tolerance and load sharing [8]. Recently, the research is directed towards the use of multiple paths for the purpose of QOS provisioning in case of mobile ad hoc networks [9] [10]. In an ad hoc

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network, multiple paths between a given source and a destination may help in the provision of QOS in the following situations [15].

Nowadays, several routing protocols are available like AOMDV, QARMD and Attentive routing protocol. All these protocols do not consider the data volume that has to be sent. But the proposed protocol considers the data volume.

In this situation, if a node wants to send a small volume of data, then that node has no necessary to find the path for capacity, less delay and other terms because a small volume of data need short path with less capacity. The above situation protocol reduces the node work for capacity estimation and contention delay. The next situation where the resources of mobile nodes are limited, a single path may not be able to provide enough resources to satisfy the desired QOS. On the other hand, the resources along multiple paths may exist between the given pair of nodes, suffice for the QOS requirements of the application.[20]. The next situation where enough resources are available along each path to satisfy the QOS requirements, the traffic can be shared across multiple paths. In other words, packets are sent long to each path that is able to satisfy QOS requirements. This may help in achieving relatively larger throughput as compared to a single path. In this situation, if node wants to send a large volume of data, then that node need not want to find the shortest path with better outstanding capacity, less delay and other terms because large volume of data needs short path with high outstanding capacity. The above situation protocol effectively utilizes the resources of each node by estimating outstanding capacity and end to end delay [20].

Further, in case of single path routing, the failure of a path needs a new path to be rediscovered, which should also be capable of satisfying the QOS requirements. A new path may not be required to be rediscovered if there are multiple paths between the given source and the destination. Therefore, one would like to investigate the effect of multiple paths between a given source and a destination on the performance of the protocol that is aware of the QOS requirements of the application.

An issue that needs to be addressed while sending multiple packets along multiple paths is that packets may arrive at the destination out of order. Therefore, re-sequencing might be required at the destination which may further increase the overheads in terms of delays. The problem of re-sequencing in multipath routing is addressed using flow assignment in [11]. The issue of load balancing in multipath source routing for mobile ad hoc networks is addressed in [12], [13].

In this paper, we present a routing protocol that is aware of the QOS and may utilize multiple paths for traffic sharing. In our protocol, the source tries to identify multiple node-disjoint paths that are able to satisfy QOS requirements in terms of data volume, capacity estimation end-to-end delays for the real-time traffic and at the same time achieve high throughput while sending packets. The paths identified are selected for sending packets in such a manner so that contention delays along the paths and re-sequencing delays at the destination are alleviated.

2.1 Over View of IEEE 802.11

In a mobile ad hoc network, IEEE 802.11 is used as the Medium Access Control (MAC) protocol. IEEE 802.11 comes in different flavours, IEEE 802.11 with Distributed Coordination Function (DCF), is of particular interest in case of an ad hoc network because the mobile devices using DCF can be configured in ad hoc (or infrastructure less) mode. In the ad hoc mode, there is no need of a base station or an access point; and the devices themselves need to forward packets of one another towards their ultimate destinations. In IEEE 802.11, there is another coordination function called the Point Coordination Function (PCF), however, it works in infrastructure mode only, and therefore, it is not suitable for an ad-hoc networks. Note that IEEE 802.11 DCF is based on Carrier Sense Multiple Access (CSMA) with Collision Avoidance (CA).

In a CSMA/CA based network, if a node i wishes to transmit, it senses whether the channel is busy or idle. If it finds the channel busy, it waits until the channel becomes idle. If the node finds the channel idle for a certain amount of time called the Distributed Inter-frame Space (DIFS), it transmits the frame. The receiver node receives the fram and if it received the frame successfully, it waits for a time duration called Short Inter-frame Space (SIFS), and sends an Acknowledgment (ACK) to the sender. If there is a collision, the node generates a random back-off number between 0 and $CW - 1$, where CW is called the contention window. The contention window, CW, may vary from CWmin to CWmax, where CWmax = 2m CWmin. The node starts random back-off timer and counts down. During the countdown phase, if the channel becomes busy, the node freezes the back-off counter. When the channel is idle again, the node resumes the countdown of the back-off counter. When counter becomes zero, and the channel is idle for at least a DIFS time duration, it retransmits the frame. If again there is a collision, the node doubles its contention window and the back-off process is repeated until a specific number of retries. After the maximum number of retries either the frame is sent successfully or it is discarded. To reduce the probability of collision, there is an optional mechanism to reserve the channel using very small frames called Request-To-Send (RTS) and Clear-To-Send (CTS) [17]. If the medium remains idle, the node i tries to send a RTS control message and waits for a time duration called Short Inter-frame Space (SIFS). Every node which receives RTS will defer its transmission including the receiver. When the receiver receives RTS, it waits for an SIFS and sends a CTS control message to the sender node, which is i, then starts to transmit the data packets using CSMA/CA.

2.2 Bandwidth Estimation

Bandwidth or more strictly speaking capacity estimation is a fundamental component in the provision of quality-of-service (QOS) in mobile ad hoc networks (MANETs). However, accurate capacity estimation can be difficult; because each host has only imprecise knowledge of the network status; thus an effective estimation scheme is highly desirable. Many previously proposed schemes [1-4] adopt 'Listen'-based estimation techniques derived from IEEE 802.11 MAC/PHY specification. The 'Listen' scheme requires each node to listen to the channel and estimate the local residual capacity based on the measurement of the local channel utilization. Given the local channel utilization (u (t)) and the maximum achievable channel capacity (Cmax), the local residual capacity (Cres) is estimated using the following equation

$$C_{\text{res}} = (1 - u(t)).C_{\max}$$

Where 0 <= u (t) <= □1 is a measure of the channel utilization. A simple and direct technique for determining channel utilization is to measure channel busy time at nodes within its carrier sensing range. A typical activity graph is shown in Figure 1. At any specific instant in time, a link is either transmitting a packet or it is idle, so the channel activity of a link can only be either 0 or 1. Thus, some meaningful measurement of the channel activity requires node to keep track of the busy channel periods over a time window (w) which is the time interval of interest. Consequently, the channel utilization (u (t)) for a time period (t-□w, t) is given by the area under the channel activity function (f (t)) curve [5]

$$u(t) = \frac{1}{w} \int_{t-w}^{t} f(t).dt$$

The accuracy of the available capacity estimation depends on the value of window (w). When the window is too small, the measure will not reflect accurately overall channel activity, i.e. if the channel was only busy or free momentarily. While, if the window is large then the utilization measure will contain historic but possibly redundant traffic during the route selection process thus reducing the overall network performance. To operate effectively, the window over which the integration is done must be chosen to balance these conflicting constraints. The choice of window size for appropriate capacity estimation is a major impediment to such window-based schemes and it is this aspect that is addressed in the next section. A new scheme for effective capacity estimation is presented in the next section where an additional function is introduced to mitigate the limitation of the previous approaches.

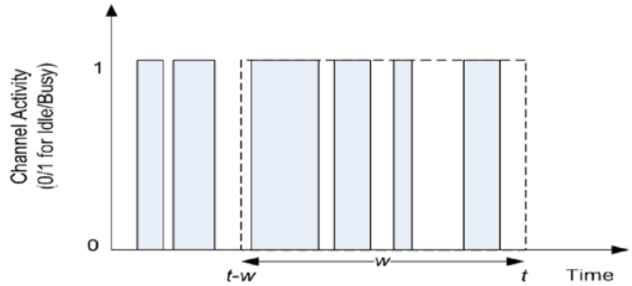


Fig.1 Example of channel activity in an IEEE 802.11

The accuracy of the available capacity estimation depends on the value of window (w). When the window is too small, the measure will not reflect accurately overall channel activity, i.e. if the channel was only busy or free momentarily. While, if the window is large then the utilization measure will contain historic but possibly redundant traffic during the route selection process thus reducing the overall network performance. To operate effectively, the window over which the integration is done must be chosen to balance these conflicting constraints. The choice of window size for appropriate capacity estimation is a major impediment to such window-based schemes and it is this aspect that is addressed in the next section. A new scheme for effective capacity estimation is presented in the next section where an additional function is introduced to mitigate the limitation of the previous approaches.

A simple available capacity estimation scheme with a lower reliance upon window size is proposed whereby the channel activity function (f (t)) is multiplied by a weighting function (g (t)) as described below

$$u(t) = \frac{1}{w} \int_{-w}^{0} f(t).g(t).dt$$

Where g (t) makes use of the raised cosine filter characteristics [6] widely used in digital communication system for pulse shaping and Δt is the excess (absolute) time The normalized excess time

$$\tau = \frac{\Delta t}{w/2}$$

2.3 End - To -End Delay Analysis

In our protocol, the source tries to estimate the average end- to-end delay based on per-hop delays. As mentioned earlier, IEEE 802.11 which is based on CSMA/CA is generally used as a MAC protocol in ad hoc networks. Hence, the contention delay at each node plays an important role and contributes to the major part of end-to-end delay. In this paper, we focus on contention delays along a path and try to minimize it. The actual procedure of how to minimize the contention delay shall be described later in this paper. The per-hop delays at each node, say node i, can be divided into three components which are as follows:

- T_s : Successful transmission time.
- T_c : Time consumed during collision.
- T_{bf} : Average back-off time at node **i**.

The total delay at node i is the summation of these delays and can be written as follows.

Total delay

$$\Delta = Tc + Tbf + Ts$$

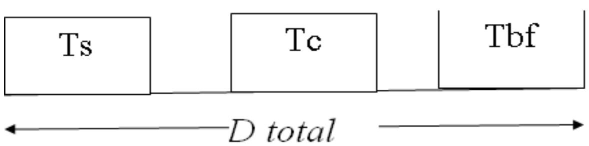


Fig.2 Total Delay

Successful Transmission

Tx = RTS + SIFS + CTS + SIFS + Packet load + SIFS + ACK + DIFS

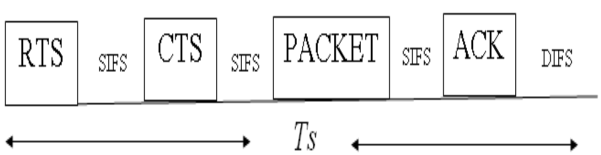


Fig.3 Successful Transmission

The Channel is busy due to collision

Fig.4 Channel is Busy

Ch busy = RTS + DIFS

- The overhead due to transmitting the data frame

T over = TACK + DIFS + 3SIFS + TRTS + TCTS

DIFS -> Distributed Interface Space

ACK -> Acknowledgment

RTS -> Request-To-Send

SIFS -> Short Interface Space

CTS -> Clear-To-Send

2.4. System Model and Assumptions for End To End Delay

Let there be an ad hoc network which can be represented by an undirected graph G = (V, E), where V is a set of vertices or nodes and E is a set of edges or links that connect the nodes where |V | = n, and |E| = m. We assume that all the links are bidirectional and each node has transmission range R. Further assume that each node employs IEEE 802.11 based on CSMA/CA as a MAC protocol, where RTS/CTS mechanism can optionally be used to alleviate the hidden terminal problem. Note that a node can transmit to another node in its transmission range. However, a node contends for the channel with all nodes which are in its carrier sense range. Let (x_i , y_i , z_i) be the location of node i, and (x_j , y_j , z_j) be the location of node j, then node i and node j are neighbors of each other if $(x_i - x_j)^2 + (y_i - y_j)^2 - (z_i - z_j)^2 \le R^2$, where R is the transmission radius of all nodes in the network. However, two nodes i and j may contend for the channel if $2(x_i - x_j)^2 + (y_i - y_j)^2 - (z_i - z_j)^2 \le Rcs$, where Rcs is called the carrier sense range. Generally, Rcs = αR, where α is constant and is usually assumed to be 2 (see Figure 1).

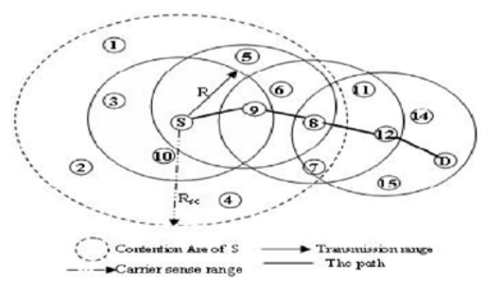


Fig.5 End to End delay System model

2.5 Throughput of A multi - Hop Network

In [18], an analysis throughput of IEEE 802.11 DCF is carried out under saturation conditions using Discrete-Time Markov Chain (DTMC). In [19], some new insights to the analysis of IEEE 802.11 DCF are presented using Fixed Point Theory. However, these analyses are for single hop wireless networks. In [15], the analysis of a multi-hop networks is presented by extending the throughput analysis for a single hop network to a multi-hop network. An expression for the throughput of a single hope network is given in [14]. In this paper, we extend the analysis of the throughput of a single hop network to a multi-hop network. For that purpose, let us start with the analysis presented in [14]. It has mentioned in [14] that the mean time between successive renewals, T, is as follows.

$$T=(1/\gamma\beta)+PsTs+PcTc\text{------}(1)$$

Where, β is the exponential back-off parameter, and ν is the number of nodes in the single hop network. The probability of collisions as a function of β is as follows [14].

$$Pc= 1-e^{-(\gamma-1)\pi\beta}\text{---------}(2)$$

The probability of transmitting the frame successfully as a function of β is given by [14].

$$Ps = 1 - Pc = e^{-(\gamma-1)\pi\beta}.\text{------}(3)$$

Using 2&3, we can write 1

$$\overline{T} = \frac{1}{\nu\beta} + e^{-(\nu-1)\beta\pi}T_s + T_c(1-e^{-(\nu-1)\beta\pi})$$

Nβ Let us denote the network throughput by Θ. According to the renewal and reward theorem [7], the network throughput is given by

$$\Theta(\beta) = \frac{PsL}{T}$$

The expression for throughput of a single hop network can be written as follows.

$$\Theta(\beta) = \frac{e^{-(\nu-1)\beta\pi}L}{\frac{1}{\nu\beta} + e^{-(\nu-1)\beta\pi}T_s + T_c(1-e^{-(\nu-1)\beta\pi})}$$

To extend the analysis of the throughput of a single hop network to a multi-hop network, we use the notion of transmission range and the carrier sense range. Note that a node can receive the transmission of another node if it lies in its transmission range. In a single-hop network, a node contends with ν − 1 nodes, where ν is the number of neighbors of the node, and ν=nρπR2. Here ρ is called the node density and ρ = A, where n is the total number of nodes in the network and A is the area deployment. In case of a network where there are multiple hops, h, between a given source and a destination, a node contends with, νcs − 1, number of neighbors, where νcs is the

$$\beta = \frac{1}{2\nu_{cs}T_c}\sqrt{1+\frac{4T_c\nu_{cs}}{\pi(\nu_{cs}-1)}} - 1$$

1. Route Maintenance

If a node senses a link failure, it informs upstream nodes along all those paths whose part the failed link was by unicasting a Route Error (RERR) message, one for each failed path. Every node that receives an RERR message marks the path invalid and unicasts the RERR upstream. Eventually, the RERR arrives at the source. When the source receives an RERR message it marks the failed path invalid. The source then retransmits the data through alternate valid paths that are not yet failed and that satisfy QOS requirements of the application. If there is no such path, then the source initiates a new route discovery.

3. DATA MANAGEMENT ROUTING PROTOCOL SYSTEM MODEL

In DATA MANAGEMENT ROUTING module provide the routing the packet based on the outstanding band width, data volume and end to end delay so, it will reduce the traffic, effective utilization of the channel& resources and increasing the through put. DATA MANAGEMENT ROUTING is a highly adaptive, distributed routing algorithm based on the principle of link-reversal. It provides multiple loop-free paths from source to destination. The key design concept of DATA MANAGEMENT ROUTING is to localize control messages to a small set of nodes in the neighborhood of the topological changes. If we want to send the data from Node A to Node E means, there are two paths are available one is ACHE and AFGHE. The routing protocol choose the path based on the end to end delay and capacity estimation

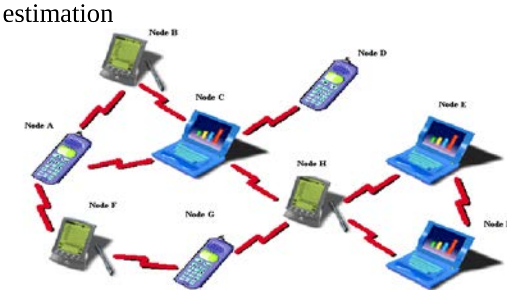


Fig.6 MANET Architecture Design

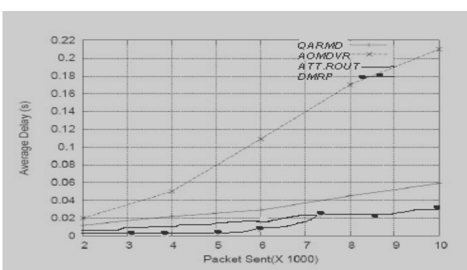
Data management Routing Algorithm
INPUT : PATH,NUMBER_OF_HOP,AVAILABLE_BANDWIDTH_IN_PATH,DATA_VOLUME
OUTPUT : MX,I

1. **If(data_volume > small volume)**
2. **for i ←0, i ≤n, i ←i + 1 do**
3. **data rate[i] ← available b.w in path[i]/no.of.hop[i]**
4. **end for**
5. **mx=maxvalue (data rate [0], data rate [1]…data rate[no of path])**
6. **for i ←0, i ≤n, i ←i + 1 do**
7. **If data rate[i] eq Mx then**
8. **return i**

```
break
End if
End for
Else
for i ←0, i ≤n, i ←i + 1 do
hop_delay[i] ← no.of.hop[i]
end for
mx=maxvalue (hop_delay [0], hop_delay [1]…
hop_delay [no of path])
for i ←0, i ≤n, i ←i + 1 do
If hop_delay [i] eq Mx then
return i
break
End if
End for
```

4. RESULTS AND DISCUSSION

To evaluate the performance of our protocol, we have developed our own simulator in C++. We generated topologies of the network where 100 nodes are distributed uniformly randomly in a region of area 1000m × 1000m. Each node is assumed to have a transmission range of 250m. The simulation time is 100 simulated seconds. The mobility model is Random Way Point (RWP). In the RWP model, a node chooses a random speed uniformly distributed between a min speed and a max speed. It starts moving towards a randomly selected destination. When the node arrives at the destination, it takes a pause; and then again starts moving following the similar steps. The time for which the node takes a pause is known as the pause time. The medium access control (MAC) layer is assumed to be IEEE 802.11. The bandwidth of each link is assumed to be 2 Mbps. The traffic is constant bit rate (CBR). The packet size is 512 bytes. While evaluating the performance of the protocol,



. Fig.7 Average end-to-end delay as a function of the number of packets sent by the source.

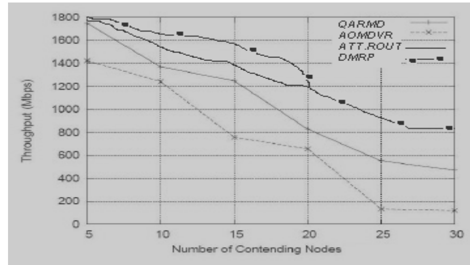


Fig.8 Throughput as a function of the number of contending nodes

We focused on the following parameters: (i) Average end-to-end delay: It denotes average latencies experienced by packets from the source to the destination. (ii) Throughput: It denotes the average bandwidth occupied by packets that are sent to the destination. (iii) QOS success ratio: It represents how many packets arrive at the destination before the expiry of their respective deadlines divided by the total number of packets sent by the source. In other words, it represents the ratio of the number of packets whose QOS requirements (in terms of end-to-end delays) are satisfy to that of the total number of packets sent by the source. We compare the performance of the proposed protocol (DMRP), with (QARMD), Attentive routing protocol and AOMDV. The reason behind choosing DMRP is also a multipath routing protocol. In addition, we modified previous protocols in such a manner so that Data volume, end-to-end delay and capacity can be used as a QOS parameter.

Fig shows average end-to-end delays and capacity as a function of the number of packets sent by the source for a varying number of paths identified by the protocols. We observe that as the number of packets transmitted is increased, the average end-to- end delay decreases. The reason is that with the increase in the deadlines, thereby, increasing the QOS success ratio. Number of packets, the queueing delay is decreased. However, DMRP incurs fewer delays as compared to AOMDV and QARMD. The reason is that in case of DMRP , the path is selected in such a manner so that the contention delays and re-sequencing delays are mitigated.

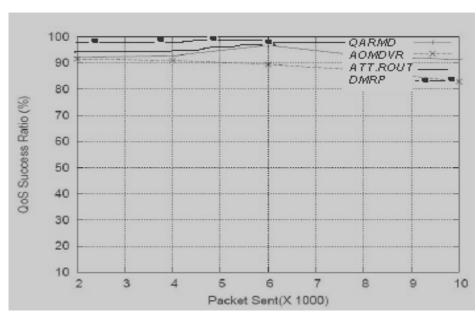


Fig.9 QOS success ratio as a function of the number of packets sent by the source.

The fig shows that the throughput as a function of the number of contending nodes that are within carrier sense transmission range, Rcs, of the sender. We observed that when the number of contending nodes increases, the throughput of the network is decreased. The reason is that as the number of contending nodes increases, the possibilities of collisions and hidden terminal problems are significantly increased. As a result, the throughput of the network is decreased. As it is mentioned DMRP tries to identify node-disjoint paths with few contending nodes for mitigating the collisions and the hidden terminal problems, thus, the DMRP has significantly better throughput than AOMDV and QARMD. Figure shows the QOS success ratio as a function of the total number of packets sent by the source. We observe that the QOS success ratio for DMRP is larger than that of QARMD and AOMDV. The reason is that in case of DMRP packets follow paths with low contention delays. This reduces end-to-end delays experienced by packets and simultaneously enables more number of packets to reach before their respective throughput of a single hop network to a multi-hop network then we tried to maximize it. In the simulation, the performance of the proposed protocol, DMRP is compared with AOMDV and QARMD. We observed that DMRP provides an improvement over AOMDV and QARMD in terms of end-to-end delay, QOS success ratio, and throughput.

5. CONCLUSION

In this paper, the proposed DMRP protocol will provide path selection mechanism based on the data volume, which means it effectively utilizes the node computation. The source tries to identify the data volume that has to be sent. If it's a small volume of data, our protocol reduces the work load and increases the performance by calculating only the end to end delay. But for large volume of data, the DMRP protocol identifies multiple node-disjoint paths that satisfy QOS requirements in terms of end-to-end delays and Capacity estimation at the same time of achieving high throughput while sending packets. Since the contention delay at each node plays an important role and contributes to the major part of end-to-end delay, we focus on contention delays along a path and tried to minimize it.

The source selects the paths that are expected to incur relatively low contention delays and high capacity. Hence, the data packets are sent according to the estimated capacity and the delay of each path. We extend the analysis of the throughput of a single hop network to a multi-hop network and then tried to maximize it. In this simulation, the performance of the proposed protocol and DMRP routing is compared with AOMDV, Attentive routing protocol and QARMD. We observed that DMRP routing provides better improvement over AOMDV, Attentive routing protocol and QARMD in terms of end-to-end delay, Capacity, QOS success ratio, and throughput.

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