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Research on Application of CGAR Algorithm in Network Routing Protocols

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Abstract: In order to reduce the transmission delay of the data packet in the wireless network, so as to improve the performance of the network, this paper puts forward the CGAR algorithm of the coding gain aware routing protocol. The algorithm uses the network coding gain and expected transmission count of the wireless link, provides the time needed for the node to transmit a data packet from the new flow in the wireless link, and take it as the routing metrics. This metric takes into consideration the factors such as the transmission delay, the time of the node's contention channel and so on. The experimental results show that this algorithm can reduce the transmission delay of the data packet and improve the performance of the network.

Keywords: Gain; Routing protocol; Coding graph; Measurement

1. Introduction

Network coding is a kind of emerging information transmission technology which overall improves the performance of system, and it is showed that network coding increases the amount of information transmitted singly by the network data, so as to reduce the delivery times of the groups and improve the transmission performance of the network data, such as improving the throughput and bandwidth utilization of the network, and balance the network load and so on. The data transmission in this paper is described as the forwarding process of the data group in the network through multiple-hop transmission, while the forwarding process of the data group on a link is described as the network data transmission [1]. Network coding is initially applied in the wired networks. In recent years, many researchers also applied the network coding method to the wireless network, in order to improve the throughput, robustness, security, energy efficiency and so on of the wireless network. With the rapid development of the wireless network technology in recent years, more and more people begin to use the wireless network for data communication, and the requirements of the users on the diversity of the network services and the transmission quality of the wireless communication constantly increase at the same time [2]. Thus, how to improve the data transmission performance of the wireless network, namely to ensure the safety and reliability of the network data transmission, and at the same time improve the utilization rate of the bandwidth and the energy resource of the network at the most extent has become one of the important problems of the network communication research. For this, based on the radio feature of the data transmission of the wireless link and the local information processing strategies of the node, network coding has become an important method to improve the transmission performance of network. At present, the data transmission technology based on network coding has proven to be the effective method which can be approximate to the theoretical transmission limitation of the network capacity, and it is identified as the important means to solve the problem of network transmission by the international academia and the U.S. military [3-6].

The traditional routing protocols usually treat the shortest path, the minimum hop, the least expected transmission count, expected transmission time and so on as the routing metrics. The coding-aware routing metric (CRM) is directly related to this paper, it takes into account the coding situation with multiple streams and uses the coding graph to represent the mediocre coding situation of the node. This paper uses the network coding gain and expected transmission count of the wireless link, provides the time needed for the node to transmit a data packet from new stream in the wireless link, and take it as the routing metrics. This metric takes into consideration the factors such as the transmission delay, the time of the node's contention channel and so on. The routing algorithm presented in this paper can reduce the transmission delay of the data packet.

2. Network Coding

In this paper, the path $F = v_0 \to v_1 \to \ldots \to v_{i-1} \to v_i \to v_{i+1} \to \ldots$ for transmitting the data packet from the source node v_0 to the end node v_t is called a "information flow" (hereinafter referred to as "flow") from node v_0 to node v_t, in which $v_0, v_1, \ldots v_{i-1}, v_i, v_{i+1}, \ldots v_t$, represent the nodes in the network. Moreover, sets $w(i,F) = \{v_0, v_1, \ldots, v_{i-1}\}$ and

$D(i,F)=\{v_{i+1},v_{i+2},...,v_t\}$ are respectively called the upstream node set and the downstream node set of node i. In addition, $N(i)$ stands for the neighbor node set of node i.

In the network topology graph which is shown in figure 1, assume that the bandwidth of the network link is 2M bps, there exists the data flow if ($i=1,2,3$), 1f transfers from node 1 to node 10 at a speed of 1.5Mbps, 2f transfers from node 1 to node 9 at a speed of 0.5Mbps, and 3f transfers from node 2 to node 9 at a speed of 2M bps. Figure 2 is the model diagram where 2f and 3f realize the rate matching coding of data in coding node 6, in the figure 23f is the encoding data flow. The figure shows that because 1f and 3f have the matched data transmission rate, and the data rate, and the queue length are the statistically significant parameters, aiming at the single queue, the data rate describes the arrival rate of the queue data, and the forwarding rate of encoding describes the data service rate of the queue. In the network coding, the corresponding encoding forwarding rates of the data groups of the different data flows after encoding are the same, therefore, the corresponding queue length of each data flow depends on the arrival rate of data. And under the condition that different data flows have the matched data rates, the corresponding queue length of the encoding data are equal in the statistical sense, therefore, in the process of encoding of 1f and 3f in node 6, there exist the equal queue length, which creates conditions for the complete coding between the two kinds of data flows.

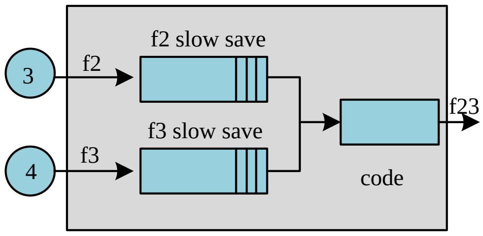

Figure 1. The Flows Enter into Node X

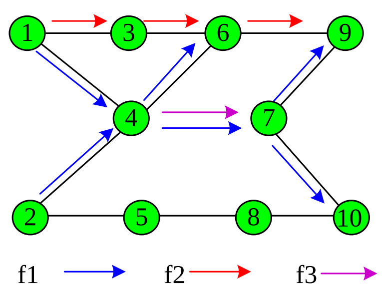

Figure 2. The Coding Graph G (V, E) of Node X

The coding graph of each node can be constructed by using the above "coding conditions". In the coding graph, if a certain sub graph of a graph meets the condition that there exists an edge between any two vertexes which can connect the two vertexes together, then the sub graph is a "complete sub graph". All the flows that constitute a complete sub graph can be encoded at the same time. As shown in figure 2, flow f_1, f_2 and f_3 constitute a complete sub graph of coding graph G, there is a connection between every two vertices.

Theorem 1 shows that if there is a complete sub graph in the coding graph of node y, then node y can obtain a data packet from each inflow involved in the complete sub graph to form a coding packet, and there is downstream node of node y in each flow. The downstream node can restore the data packet sent by the information source in its flow through the encoding of the coding packet, thus to make the information sink of the flow obtain the data packet sent by the information source.

Demonstration of theorem 1: when there are two inflows in node y, according to the "coding conditions", if the two flows can be encoded to generate a coding packet in node y, then there is downstream node of node y in each flow, which can use the overheard data packet to carry out the encoding of the coding packet to obtain the data packet sent by the information source in its flow. Thus, theorem 1 is established.

As shown in figure 3, when there are three inflows in node y, set that the three inflows f_1, f_2 and f_3 constitute a complete sub graph in the coding graph, namely they can be encoded with each other. Set A_1, B_1 and C_1 are respectively the information sources of flow f_1, f_2 and f_3, and the data packets sent are respectively PA, PB and PC, and the three data packets are encoded into a coding packet $PA\oplus PB\oplus PC$ by node y, in which $\oplus$ represents operation. Because flow f_1 and f_2 can be encoded in node y, it can be known from the "coding conditions" that there is a downstream node B2 of node y in f_2, and B2 can take advantage of the overheard data packet PA that comes from flow F1 to carry out the encoding and obtain that $PA\oplus(PA\oplus PB\oplus PC)=(PB\oplus PC)$. Then node B2 forwards $(PB\oplus PC)$ to the downstream node. In addition, because flow f_3 and f_2 can be encoded in node y, it can be known from the "coding conditions" that there is a downstream node B3 of node y in f_2, and it can make use of the overheard data packet PC of flow f_3 to encode the received data packet $(PB\oplus PC)$ and obtain that $PC\oplus(PB\oplus PC)=PB.(PB\oplus PC)$ is the data packet that sent from the information source of flow f_2 to the information sink. By the same token, the data packets sent from the information sources of flow f_1 and f_2 can be received by their respective information sinks.

3. Gain

In the process of network coding, the improving of coding opportunities shows that the amount of data transmission reduces, and the reducing of the amount of data transmission reduces the cost of network data transmission, and obtains the improving of coding gains. Under the condition that the network topology is stable, for the network coding between the convections, the coding level of data in the coding node directly affects the size of the network coding opportunities, and it depends on the data rate matching state of the data flow in the coding node in a large extent. Based on this, the paper discusses the establishing of the relationship between the data rate and the amount of data transmission, and then establishes the cost efficiency function of the network based on the data rate, so as to describe the change of the network coding gain. Based on this, the paper adopts the linear programming model of the correlation data rate to solve the data rate matching problem in network coding. The specific description is as follows: set that the rate of data flow f_i is x_i, and the transfer rate of f_i on path y is x_i^y, and it meets the following formula: $\sum_{y \in y_i} x_i^y = x_i \left(x_i^y \in x_i, x_i \in x, x = \left[x_1, x_2, \ldots x_F \right] \right)$, y_i is the route set passed by f_i, and F is the set of the network data flow, the success transfer rate of data on link $\left(n_i, n_j \right)$ is $a_{ij}, \left(a_{ij} \right)$, define the probability that node n_j, successfully receives the data groups of node n_i, and it is the size of cost of transmission on $\left(n_i, n_j \right)$ in the paper. For path $y\left(n_i, n_i, n_j \right)$ and path $y\left(n_i, n_i, n_j \right)$ n_k is the coding node, after introducing the network encoding mechanism, the transmission costs produced by the data flows transferred with x_i^y, x_j^y on path p and q are as follows:

$$u_{n_i}\left(x_i, y_i \right) = \max\left\{ \alpha_{ij}, a_{ki} \right\} y_i$$
$$= x_{n_i} x_i^y + x_{n_i} x_j^y - \left[\left(\alpha_{ij} + a_{ki} \right) - \max\left(\alpha_{ij}, a_{ki} \right) y_i \right] \quad (1)$$

In formula (1), there is $u_k = \min\left\{ x_i^y, x_j^y, y_i \right\}, y_i$ the capacity of path p, and y_i is the vector of the network link state, namely the capacity of link, then the transmission cost of the network system is described as follows:

$$u\left(x, y \right) = \sum_{f_i \in F_x} \sum_{x \in x_i} \left(\theta_i^x y_j^r - R \right) \quad (2)$$

In formula (2), there is $u_k = \min\left\{ x_i^y, x_j^y, y_i \right\}, y_i, \theta_i^y$ *and* CN_p f_i is respectively the transmission cost of i_j vector and the set of encoding nodes of f_i on path p. At the same time, the required amount if transfer rate of the non-coding data and coding data of if on the route of c_n is less than the currently available capacity x_y of the path, namely $\sum_{x \in x_i} x_i^y \leq y_c$, and it can be known from (2) that to make the transmission cost of the network data obtains the minimum value, the matching rate of network data coding should be obtained the maximum value under the condition of meeting the capacity limit of the link, namely be expressed as the following linear programming model:

$$\max \sum_{f_i \in F_x} \sum_{x \in x_i} T \quad (3)$$

4. The CGAR Algorithm

4.1. The route discovery

The routing protocol of CGAR is similar to that of AODV. The path-finding process is initiated by the source node, who broadcasts the routing request packet RREQ (Route Request) to the neighbor node, the intermediate node forwards RREQ to the neighbor node, and the information sink (the destination node) returns RREP(Route Reply) to the information source after receiving RREQ, after the information source receives RREP, it chooses a path. When there are multiple paths between the information source and the information sink to be chosen from, the information source will the path

Each node is deployed with the “path-finding table” and the “flow chart”, which respectively record the information used in the path-finding process and the information of the existing flow in the node. The former contains the domains such as the ReqID, RepID, SRC, DST, UpSet, UpNbSet, DnSet, DnNb - Set, NextNode To SRC (the next hop node from the node to the information source), NextNode to DST (the next hop node from the node to the information sink), Valid-Time, which shows the survival time of the corresponding record in the table, and the unit is ms. The later contains the domains such as the FID (identifier of the flow), SRC, DST, UpSet, UpNbSet, DnSet, DnNbSet, Q (the flow passed within Δt time, and the unit is Mb), NextHopTo SRC (the next hop node to the information source), NextHop to DST (the next hop node to the information sink). For example, for node 3 in Figure 4, its flow chart is shown in table 1, in which the value of Q is refreshed every Δ t time.

Table 1. The flow chart

NextHop-To SRC	UpSet	DnNbSet	UpNbSet	ReqID	DnSet
4	2	8	2	9	5
8	5	5	6	3	6

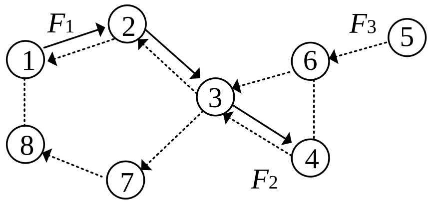

Figure 3. The Flows through Node 3

The route discovery process of CGAR protocol is as follows:

The source node broadcasts RREQ to the neighbor node, in which ReqID is the unique identifier of RREQ packet, NID and SRC both are the numbers of the source node, DST is the number of the destination node, and UpSet and UpNbSe are empty. At the same time, add a record (namely add a line) in the path-finding table of the destination node, and fill the above contents in the corresponding field.

After the intermediate node receives the RREQ packet, handle the RREQ package according to the following logic:

Step 1 if in the received RREQ, the triple (SRC, DST, ReqID) is the same with the certain record in the path-finding table of the node, then abandon RREQ and go to Step 7.

Step 2 if the bigram (SRC, DST) in RREQ is the same with the corresponding content recorded in the path-finding table, go to step 3, otherwise, go to the Step4.

Step 3 if the domain of UpSet in the path-finding table is the subset of the UpSet in RREQ, then abandon this RREQ packet and go to Step7; otherwise, delete the corresponding record in the path-finding table, and go to Step4.

Step 4 add a new record in the path-finding table, copy the domains ReqID, SRC, DST, UpSet and UpNbSet into the path-finding table, and copy the RREQ.NID into the NextNode To SRC,FID domain in the path-finding table, the content will be automatically generated, and increased by 1 every time.

Step 5 modify the RREQ packet. Modify the NID domain into the number of the node, add the number of the node to UpSet, and add the next hop node of the node to the UpNbSet domain.

Step 6 broadcast the RREQ packet to the neighbor node.

Step 7 END.

5. The Experimental Simulation and Analysis

5.1. The experimental settings

Use the design language of the Matlab programming to compile the simulation program to carry out the experiment, and analyze the experimental results. In figure 1, the numbers next to the link show the distances between the nodes. The same as the literature [9], the loss index y of the path is 2, and there are $Eelec = 52nJ/bit$ and $Eamp = 12pj/bit/m2$. When solving the combinatorial optimization problem (17), the CGAR algorithm described in section 3 is used, and the bytes occupied by the header of the data packet and the ACK packet both are 30 bytes, namely, $v = m = 30$ bytes. Realize the throughput and delay of the three routing algorithms, CGAR, DCAR, and COPE. The simulated experiment scenario is shown in Figure 4.

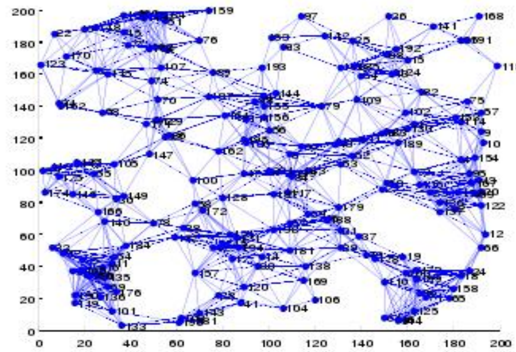

Figure 4. The Simulation Diagram

5.2. The analysis of the results

Respectively in view of the condition that there are 1,2,…10 flows ((randomly select a pair of nodes to form the information source and the information sink of a flow) in the network and the condition of the change of the flow number (hereinafter referred to as "flow"), the simulation results of the throughput and the delay of the three algorithms CGAR, DCAR and COPE are shown in Figure 5 and Figure 6.

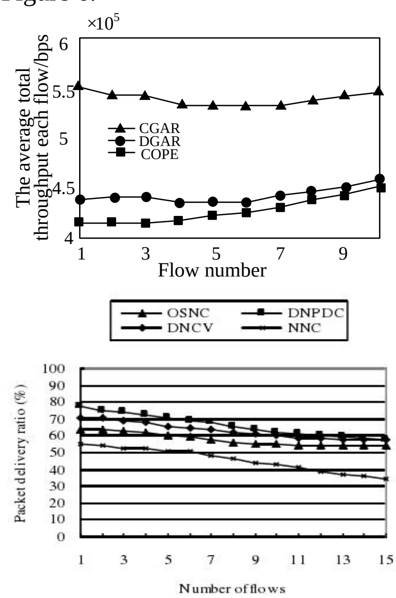

Figure 5. The Average Thoughput of Each Flow

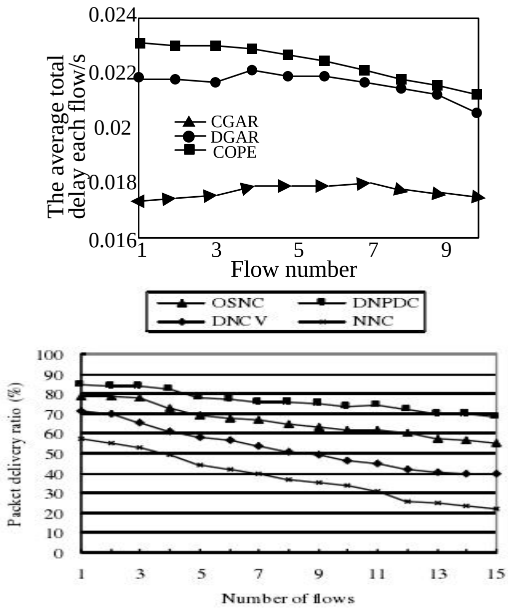

Figure 6. The average delay of each flow

It can be seen from the two figures that, (1) for the given number of flow, CGAR is better than DCAR and COPE in terms of throughput and delay; (2) when the number of flow is relatively small in the network, the delay of CGAR is on the rise with the increasing of the number of the flow. But when the number of the flow increases to a certain degree, the delay gradually shows a trend of declining, this is mainly because with the increasing of the number of the flow, the node can encode the data packets from multiple flows to generate a coding packet, so as to improve the efficiency and save time.

Let the success rate of the probe packet changes between 0.90~1.03, namely the packet loss rate changes between 0 ~ 0, 18, the simulation results of the changes of the delay can be obtained, and it is shown in figure 7. This shows that: (1) CGAR is still better than DCAR and COPE in the aspect of delay; (2) the delay of CGAR declines with the increasing of the success rate of the data packet, this is because the success rate is large, the number of the retransmission of the data packet declines and the delay reduces accordingly.

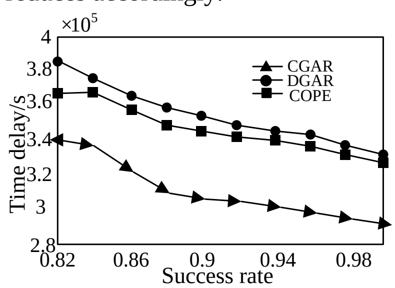

Figure 7. The relationship between success rate and delay

Under the condition that the number of the flow changes, the simulation results of the path failure rate of the CGAR, DCAR and COPE routing are shown in figure 8. The so-called path failure refers to that there is node in the path which cannot send the data packet due to the lack of channel capacity. In addition, the path failure rate is defined as the proportion of the number of the failed paths in the total number of the paths determined by the routing algorithm. Figure 8 shows that: (1) for the given number of the flow, the path failure rate of CGAR is lower than that of DCAR and COPE; (2) with the increasing of the number of the flow, the path failure rate of the three types of routing protocols increases, which is in line with the intuitive. Because the load of the each path's transmission of the data packet increases with the increasing of the number of the flow, as so to make the lack of bandwidth, which leads to the failure of the path.

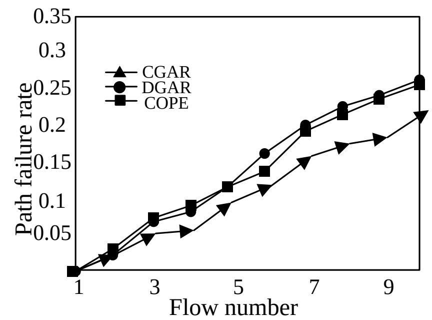

Figure 8. The path failure

6. Conclusion

Routing protocol is a hot research topic in the current wireless sensor network, this paper studies the application of the proposed CGAR algorithm in the wireless network routing protocol. The algorithm can effectively reduce the energy consumption, relieve the radio interference between the nodes, save the transmission delay of the node used in retransmitting the cache of the needed configuration of the data packet and the data packet, and reduce the loss rate of the path, so as to improve the reliability of data transmission in the process of coding. To further explore the effects of the distribution of coding nodes and the configuration of the network data flow model on the data transmission delay during coding, and the time synchronization problem of the nodes during coding, it remains to be further implementation in the future research.

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