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Qualitative Comparison Routing Algorithms of MANET Carrying Packet Telephony

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Abstract— Mobile Ad-hoc Networks are quite different from other wireless networks. The reason behind this difference is the mobile nature of participating nodes. Due to this network configuration keeps on changing and re-routing is frequently required. It has been difficult for researchers to realize Packet Telephony over Mobile Ad-hoc Networks A robust routing algorithm is prerequisite for successful implementation of Packet Telephony over Mobile Ad-hoc Networks. In this work authors are presenting a qualitative comparison of various popular routing algorithms in terms of various QoS parameters. During this work an environment is simulated using ns2 where Packet Telephony is implemented over a Mobile Ad-hoc Network.

Keywords— Mobile Ad-hoc Network, AODV, DSDV, DSR, Packet Telephony, Voice over Internet Protocol, Network Architecture, throughput, channel utilization, routing overhead.

I. INTRODUCTION

"A Mobile Ad-hoc Network is a collection of autonomous nodes or terminals that communicate with each other by forming multi-hop networks and maintaining connectivity in decentralized manner."

Each node, participating in a MANET, acts as a potential router and the system lacks a centralized routing system. Every other node within the range of a given node is termed as neighbor. Two neighbors communicate directly. For communication with some distant node, in the network, individual node(s) rely on the neighbors. A node needs to maintain and update route to every other node in the network so that it can forward the packets toward its (packet's) destination. The whole idea of MANETs is based on the assumption of cooperation between various participating nodes. The dynamic nature of nodes and open wireless channel make MANETs vulnerable to security threats.

Voice over Internet Protocol [1] represents a set of rules and techniques to transport telephonic conversation over Internet Protocol. Packet Telephony or Voice over Internet Protocol (VoIP) has been proposed as alternative to the traditional circuit switched telephony. This is intended to overcome the limitations of traditional telephony. In traditional circuit switched a permanent circuit is first setup between caller and callee. Afterwards the conversation can start. Due to the reservation of complete path traditional telephony is quiet costly. To reduce the cost of telephonic calls Packet Telephony was proposed and it has gained popularity for long distance calling.

The origin of VoIP can be dated back to 1995 when an Israeli company called Vocaltec launched first VoIP phone termed as 'Internet Phone'. The market share of VoIP as compared to PSTN has been increasing steeply since then. VoIP is one of the admired applications of Internet.

The process of VoIP or Packet Telephony is outline below:

Step 1: Caller dials the telephone number of Callee.

Step 2: Some handshaking process results in the Call setup and the session begins.

Step 3: Caller (or Callee) speaks into the mouthpiece. This result into generation of analog conversation data.

Step 4: The analog conversation are converted into digital form and compressed using some suitable codec.

Step 5: The compressed call packets are transported in terms of IP packets using some suitable routing algorithm.

Step 6: These conversation packets on arrival at destination are decoded and converted into analog voice.

Step 7: The Callee (or Caller) hears this voice through earpiece.

Step 8: After conversation is over Session is closed.

II. QUALITY OF SERVICE

Successful implementation of Packet Telephony over Mobile Ad-hoc Networks [2] is constrained by a number of service related parameters termed as QoS parameters, as defined by International Telecommunication Union (ITU). QoS parameters related to the Packet Telephony can be classified into two categories viz. Call Level Parameters and Packet Level Parameters. Various packet level parameters include Packet Drop Rate, End-to-End delay, Jitter, throughput, channel utilization, average hop per route, routing overhead, control packets sent/received and routing delay. Some QoS parameters are very critical and their value must remain in specified range for successful implementation of Packet Telephony. While for other parameters there may be no range limitation but the qualitative performance in terms of these parameters is important to realize packet telephony over the Mobile Ad-hoc Networks.

III.NETWORK ARCHITECTURE

Network architecture for implementation of Packet Telephony over Mobile Ad-hoc Networks is outlined in figure 1. At application layer corresponding voice data is produced which is encoded and compressed then forwarded towards the transport layer. One application layer solution for packet telephony over Mobile Ad-hoc Networks is G .729 [3] codec. Transport layer is responsible for end-to-end delivery of compressed voice from caller to callee. One common transport layer solution for packet telephony over Mobile Ad-hoc Networks is RTP/UDP [4]. Network layer is central to the success of the system. Major responsibilities with the network layer are Packetization and routing. The compressed voice coming from transport layer is packetized as IP packets for routing towards destination. A number of routing protocols [5] are available out of these most common ones are DSR, DSDV, TORA and AODV. In this work system architectures based on AODV, DSDV and DSR respectively are implemented to evaluate various routing algorithms. The role of data link layer is medium control and hiding the details of channel from the upper layers. Physical layer is responsible for moving the voice data. IEEE 802.11g [6] provides pretty popular specifications for both Physical and Data Link Layers. A number of devices working on these specifications are available in market. So, authors propose to use these specifications.

IV. QUALITATIVE ANALYSIS OF ROUTING ALGORITHMS

Performance of proposed network architecture is evaluated by varying routing algorithm. Evaluation is done in terms of different Quality-of-Service parameters that affect the quality of performance. These parameters include

A. Throughput
B. Channel utilization
C. Average hop per route
D. Routing overhead
E. Fraction of Control packets sent & received in traffic
F. Routing delay

During this work various routing algorithms are compared, in terms of throughput, channel utilization and number of hops, in an environment with Mobile Ad-hoc Network carrying Packet Telephony.

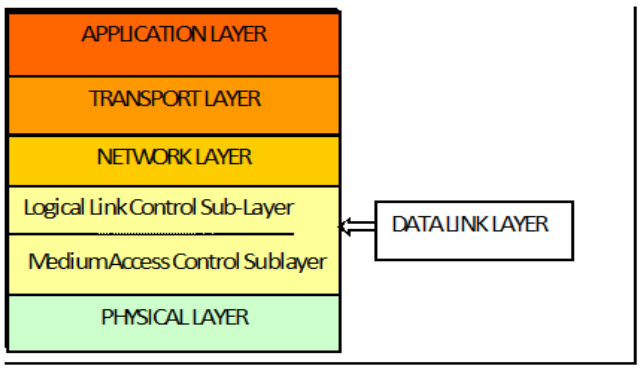

Figure 1: Proposed Network Architecture

V. DSDV ROUTING ALGORITHM

DSDV [7] is a table driven algorithm designed by C.Perkins and P. Bhagwat. In this Bellman-Ford algorithm is employed. It is a proactive routing algorithm where each node maintains its copy of routing table. Each record in the table contains details of some node in the network with number of hops and next neighbor address to reach that.

Each record has a sequence number to identify the recentness of the route. Routing tables are regularly updated through periodic transfer of routing information in the network. These results in large traffic in the network, to decrease the amount of traffic two type of packets are used viz. full dump (containing whole routing information) and incremental (Carrying only changes in last information).

The routing packets are broadcast and contain the address of destination. The recentness of a route, in the table, is identified by its sequence number. If two routing updates have same sequence number then one with lesser number of hops is preferred. New routes discovered by a node are further advertised by various nodes with routing information. The performance of DSDV is highly dependent upon the time duration between route updates. Routes that are broken due to the dynamics of network are reported by the MAC layer and communicated to the network layer where the hop count is replaced with infinity for representing broken link.

Due to frequent route discovery in won't allow nodes to go in sleep mode for saving energy thereby it is harsh on battery. It consumes a lot of bandwidth due to frequent routing activities. It is not suitable for larger networks.

VI. AODV ROUTING ALGORITHM

AODV[8] is one of the popular on-demand routing algorithms designed through a collaborative effort of Nokia, University of California, University of Cincinnati for Mobile Ad-hoc Networks and was proposed by C.Perkins, E.B.Royer and S.Das .

A. Route Discovery: This is a purely demand based algorithm and nodes remain silent until some route is requested. The only thing a node does, during silence period, is to keep track of its neighbors through special "HELLO" packets.

When some source node has message for a destination node, to which path is unknown, first of all it broadcasts a route request packet, RREQ to its neighbors which is forwarded by them to their neighbors. This process continues until either the destination is reached or an node is located that has a recently used route to the destination.

Each route request is identified by its 'broadcast_id'. This unique ID helps in avoiding routing loops as well as ensures that multiple copies of same request do not exist in the network. In the process of forwarding the request packet, intermediate nodes record the address of neighbor from which the first copy of broadcast packet is received. This information helps to find the reverse path.

B. Route Reply: After the destination (or node with appropriate route) is found, it responds by unicasting a route reply packet back to the neighbor from which it first received the request packet. This traces back to the source and the route is established.

A route expires if not used within the specified lifetime. Source may receive more than one route reply. It makes use of destination sequence number in reply to decide upon the route to be used. It does not employ source routing, intermediate nodes simply sends route reply to the previous node from which they received the route request.

On route error, the information is sent back to the neighbor from whom the route request arrived at this node. The process continues till the source node.

VII. DSR ROUTING ALGORITHM

DSR [9] is another on-demand routing algorithm, but its working it quite different from AODV. It makes use of 'source routing' rather than depending upon routing tables of intermediate nodes. This means each node constructs routing information independent of other and routing packets contain whole route rather than the address of immediate node.

In this no "HELLO" packets are employed to discover the neighbors thus it would generate lesser control traffic than AODV. In this the mobile nodes maintain route caches. This cache contains the known routes. This is updated when some new route is learned. The overall algorithm consists of two phases:

A. ***Route Discovery:*** If some node (say caller) wants to communicate with some node (say Callee) in the network, first of all it checks its route cache. If it does not contain recent enough route then the route discovery is initiated. To start with a route request packet is broadcast. Every other node that receives this packet will check its route-cache. If it knows of a route to the destination then route reply is generated.

Otherwise it will add own address to the route record of the packet and forward it. A route reply is generated when the route request reaches either the destination itself or the present node knows some usable route to the destination. Now the problem is to find the reverse route. If it is in the route cache of the node, it may use that. Otherwise if symmetric links are supported then node may reverse the path just found. If symmetric paths are not supported then yet another route discovery, with discovered route piggybacked on route reply packet, initiates.

B. **Route Maintenance:** Each node sends periodic packets to see that all nodes in the cache are intact, if any problem with some route is found, it tries to find alternative.

Full network architecture, discussed above, with DSR as routing algorithm was implemented in ns2 (Network Siumator-2) [10-12] and following observations were made.

VIII. THROUGHPUT

Throughput is an important Quality of Service parameter, it represents amount of data successfully transmitted from source to destination per unit of time. For this work throughput will be specified as

"The voice packet successfully transmitted per unit time from caller to receiver."

Performance of the proposed architecture with respect to important network characteristics, viz. network area, number of nodes, number of simultaneous telephonic calls, maximum node speed are plotted in figure 4. In every plot all other network characteristic parameters except for one are kept constant.

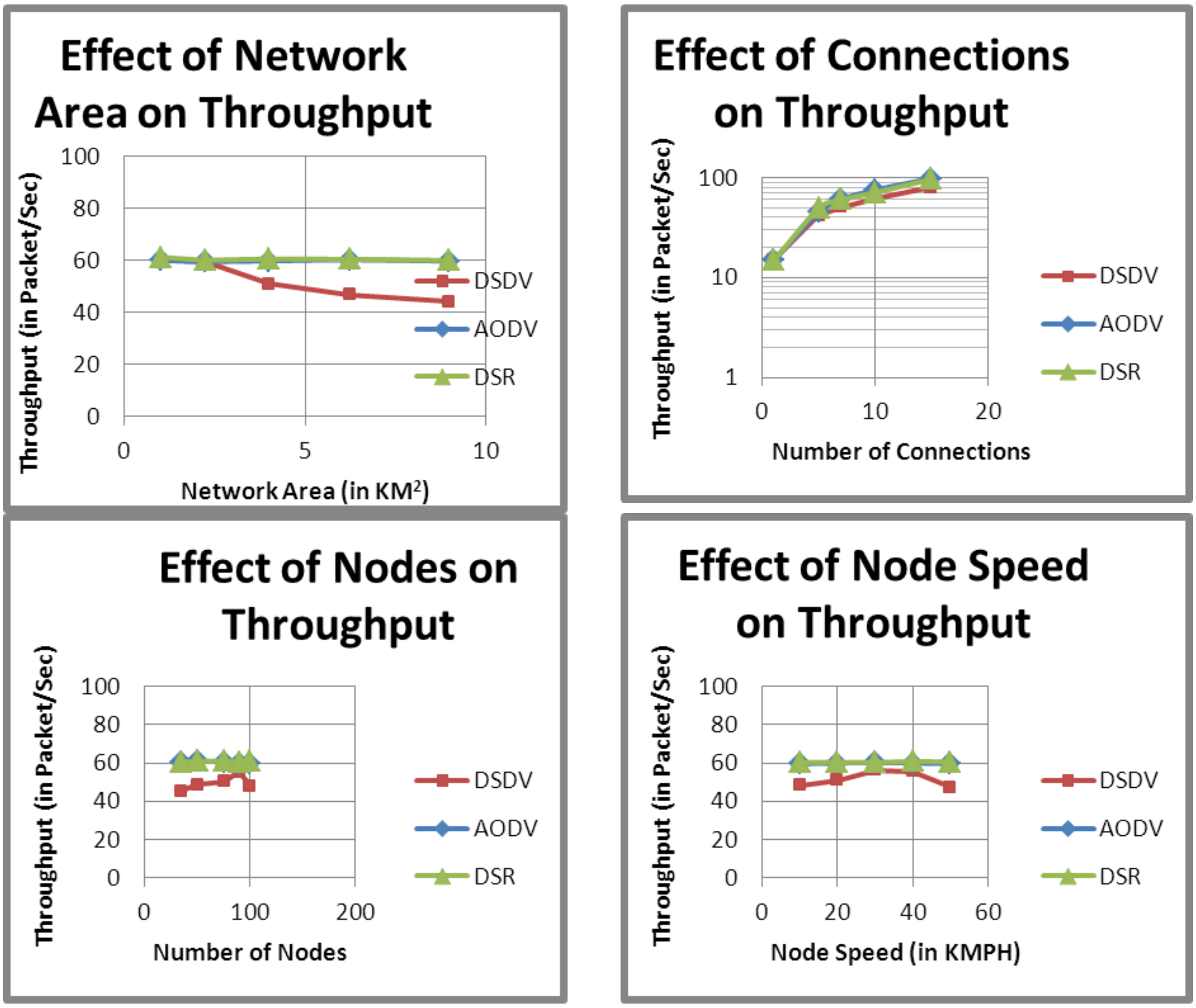

Figure 4: Study of Throughput with Various Network Characteristics

Observation(s)

A. It was observed that throughput decreases with area as expected due to the increase in network size, available router per area decreases and hence packet delivery rate and hence throughput decreases. It is important to note that throughput included delivered voice packets only.
B. It was observed that the throughput increases exponentially with number of connections as expected because more connections means more packet to transfer and hence throughput increases steeply.
C. With increase in number of nodes throughput increases, although this increase is not that prominent in most cases. This can be attributed to the fact that with increase of number of nodes user density increases and hence number of routers per unit area increases. This leads to more successful packet delivery and thus increases throughput.
D. No specific inference could be drawn by the plot of node speed versus throughput. But overall on-demand algorithms show better performance in terms of throughput.

IX. CHANNEL UTILIZATION

Channel utilization is another important QoS parameter to study the performance of a network offering real time service like packet telephony. In this study Channel utilization is evaluated for comparison of various routing algorithms and attention will be limited to the portion of channel used for voice packets. The channel portion used by the voice packers would be plotted against various network parameters and performance of each algorithm would be studied.

Performance of the proposed architecture with respect to important network characteristics, viz. network area, number of nodes, number of simultaneous telephonic calls, maximum node speed are plotted in figure 5. In every plot all other network characteristic parameters except for one are kept constant.

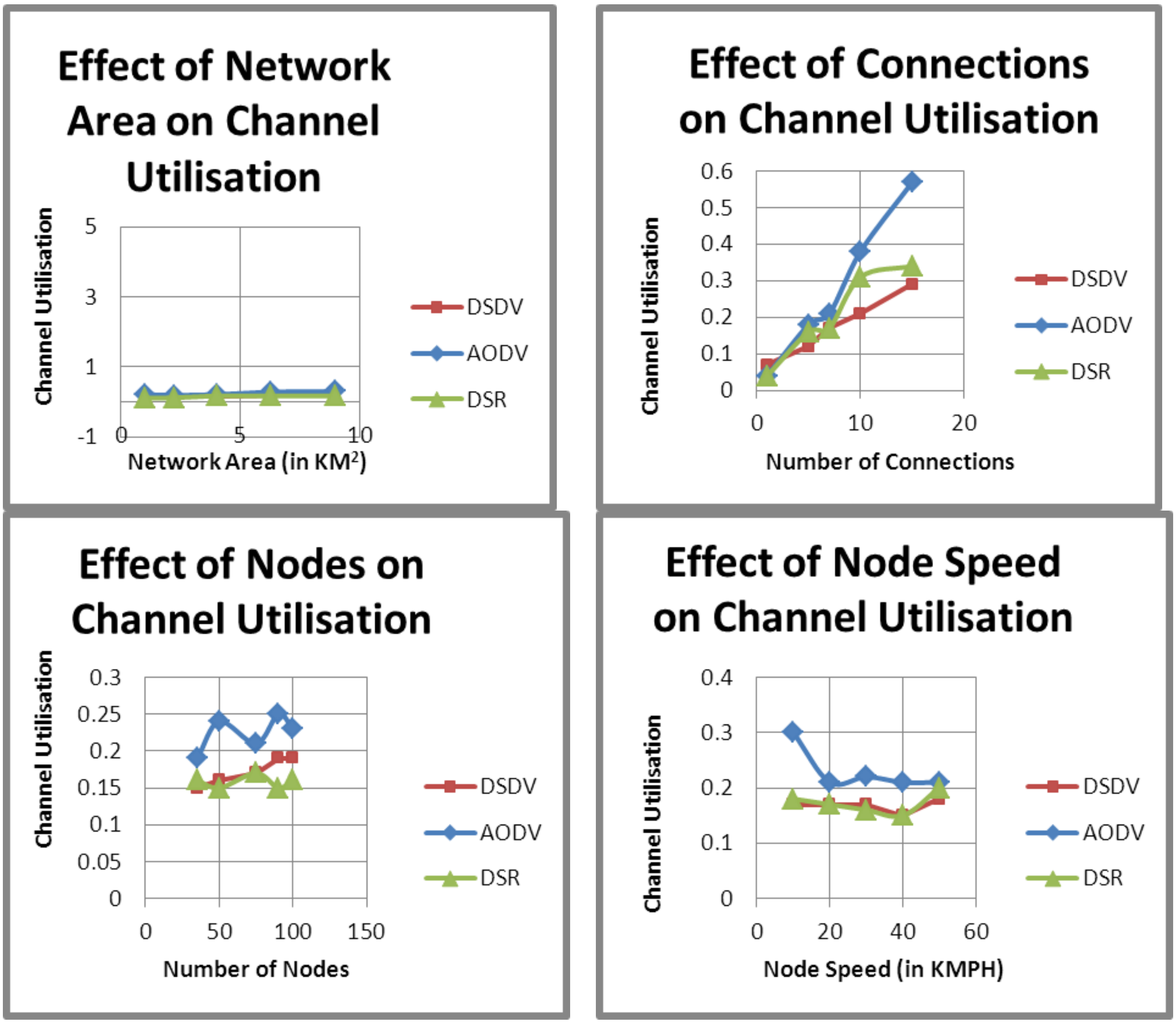

Figure 5: STUDY OF PACKET DELIVERY RATIO WITH VARIOUS NETWORK CHARACTERISTICS

Observations:

A. It was observed that channel utilization for the purpose of packet telephony is not more than 1%, this means that with available wireless technologies it is possible to realize the packet telephony in mobile ad-hoc networks. The problem in implementation, if any, would come from routing algorithm. Due to the mobile nature of networks route keeps on changing and for successful delivery of packets proper routing algorithms are required.
B. The channel utilization increases with number of connections as expected because more connections means more packet to transfer and hence utilization increases steeply.
C. With increase in number of nodes channel utilization does not vary that much.
D. No specific inference could be drawn by the plot of node speed versus channel utilization.

X. HOP(S) PER ROUTE

Another QoS parameter to evaluate the performance of a routing algorithm is number of hop(s) per route. Number of hop(s) in a route has direct impact on overall delay as more hops means more queues the data packets have to stand in and hence more delay. The number of hop(s) was plotted for various routing algorithms in different scenarios are earlier; the details of these are given next:

"The hop represents number of nodes voice/data packets go through from source to destination (including destination)."

Performance of the proposed architecture with respect to important network characteristics, viz. network area, number of nodes, number of simultaneous telephonic calls, maximum node speed are plotted in figure 6. In every plot all other network characteristic parameters except for one are kept constant.

Observations

A. The outcomes are plotted in figure 6. It was observed that Hop(s) increases with area. Average hop per route ranges between 1 to 5 for all protocols.
B. It was observed that average hop per route ranges between 1 to 4 for all protocols.
C. It was observed that average hop per route ranges between 1 to 4 for all protocols.
D. The observations are plotted in figure 6. It was observed that average hop per route ranges between 1 to 4 for all protocols.

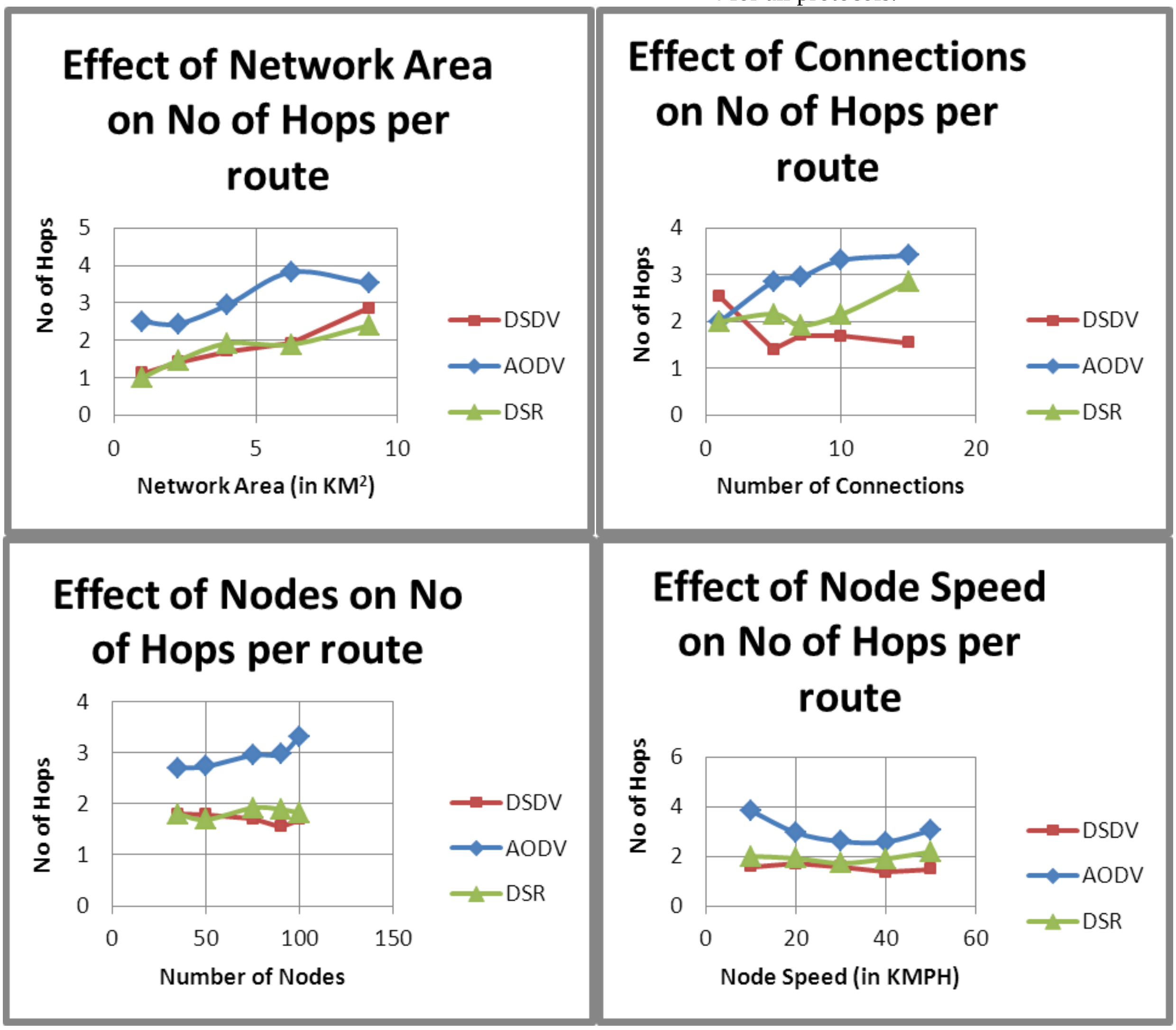

Figure 6: STUDY OF PACKET DELIVERY RATIO WITH VARIOUS NETWORK CHARACTERISTICS

XI. CONCLUSIONS

A. In terms of throughput on-demand algorithms in DSR and AODV outperforms proactive routing algorithm, DSDV. Throughput would have a direct effect on Packet Delivery Ratio and End-to-End delay values.
B. In terms of channel utilization DSR utilizes minimum channel. This would mean lesser chances of congestion and hence lesser chance of delay or packet loss.
C. One interesting observation comes for number of hops (from source to destination). It was observed that in case of DSDV average number of hops is least. One important point worth mentioning here is that this value is calculated for successfully delivered packets only

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