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SIMULATION STUDY OF A CELLULAR AIDED MOBILE AD-HOC NETWORK

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Abstract

This report studies the performance of a Cellular Aided Mobile Ad-hoc (CAMA) network by simulation in ns-2. The simulation results are extensions to the previous published results in the paper on CAMA ([1]). The link adaptation, the impact of position error, carrier sense threshold, duration time and mobility, the transmission of CBR, TCP, video and VoIP, and the cellular overhead are studied. The delivery ratio, throughput, network delay, data rate and hop count are presented and analyzed.

Gang Ding, Xiaoxin Wu, Bharat Bhargava, and Shan Lei

Nov. 6th, 2003

1 Simulator: ns-2 and extension

1.1 Mobile wireless network simulation in ns-2

The mobile wireless part of ns-2 is ported from CMU's Monarch group. MobileNode is the basic Node object with added functionalities like movement, ability to transmit and receive on a channel that allows it to be used to create mobile, wireless simulation environments. The network stack for a mobilenode consists of a link layer(LL), an ARP module connected to LL, an interface priority queue(IFq), a mac layer(MAC), a network interface(netIF), all connected to the channel. These network components are created and plumbed together in OTcl. Each component is briefly described here.

Link Layer The LL used by mobilenode is similar to the LL for wired network simulation in ns-2. The only difference is that the link layer for mobilenode has an ARP module connected to it which resolves all IP to hardware (Mac) address conversions. Normally for all outgoing packets, the packets are handed down to the LL by the Routing Agent. The LL hands down packets to the interface queue. For all incoming packets, the mac layer hands up packets to the LL.

ARP The Address Resolution Protocol module receives queries from Link layer. If ARP has the hardware address for destination, it writes it into the mac header of the packet. Otherwise it broadcasts an ARP query, and caches the packet temporarily. For each unknown destination hardware address, there is a buffer for a single packet. Incase additional packets to the same destination is sent to ARP, the earlier buffered packet is dropped. Once the hardware address of a packet's next hop is known, the packet is inserted into the interface queue.

Interface Queue The queue is implemented as a priority queue which gives priority to routing protocol packets, inserting them at the head of the queue. It supports running a filter over all packets in the queue and removes those with a specified destination address.

Mac Layer The IEEE 802.11 distributed coordination function (DCF) Mac protocol has been implemented. It uses a RTS/CTS/DATA/ACK pattern for all unicast packets and simply sends out DATA for all broadcast packets.

Network Interfaces The network interface layer serves as a hardware interface which is used by mobilenode to access the channel. The wireless shared media interface subjects to collisions and the radio propagation model receives packets transmitted by other node interfaces to the channel. The interface stamps each transmitted packet with the meta-data related to the transmitting interface like the transmission power, wavelength etc. This meta-data in packet header is used by the propagation model in receiving network interface to determine if the packet has minimum power to be received and/or captured and/or detected (carrier sense) by the receiving node. The model approximates the DSSS radio interface (Lucent WaveLan direct-sequence spread-spectrum).

Radio Propagation Model It uses Friss-space model at near distances and approximation to Two Ray Ground at far distances.

Antenna An omni-directional antenna having unity gain is used by mobilenodes.

Node movement generation The node-movement generator is available in ns-2, in which we can specify the number of mobilenodes, the maximum speed of movement, the average pause between movement, the simulation stop time and the topology boundary. The generator uses the random waypoint algorithm.

Traffic pattern generation Random traffic connections of TCP and CBR can be setup between mobilenodes using a traffic-scenario generator script, which can be used to define the type of traffic connection (CBR or TCP), the number of nodes and maximum number of connections to be setup between them, a random seed and incase of CBR connections, a rate whose inverse value is used to compute the interval time between the CBR packets. The start times for TCP/CBR connections are randomly generated

In addition to the above features, four ad-hoc routing protocols are currently supported at the network layer:

- Destination Sequence Distance Vector (DSDV),
- Dynamic Source Routing (DSR),
- Temporally ordered Routing Algorithm (TORA),
- Ad hoc On-demand Distance Vector (AODV).

1.2 Extensions to ns-2

In order to study the performance of CAMA in ns-2, we have completed following extensions to ns-2.

GPR routing protocol see [1]

Link adaptation in IEEE 802.11 In order to adapt the data sending rate in IEEE 802.11 wireless networks, sender sends RTS at the basic rate with the estimated duration time (NAV) based on local SNR. This duration time affects network performance. Upon receiving RTS, the receiver estimates the SNR and selects an appropriate transmission rate, then the receiver updates the duration time and sends back CTS with the selected data rate. The data rate is selected according to the receiving SNR:

- 48 Mbps for > 30 dB SNR
- 36 Mbps for > 26 dB SNR
- 24 Mbps for > 21 dB SNR
- 11 Mbps for > 18 dB SNR
- 5.5 Mbps for > 16 dB SNR
- 2 Mbps for > 14 dB SNR
- 1 Mbps for > 11 dB SNR
- Frame lost for < 11 dB SNR

Different duration time update methods will be compared in the following simulations.

Lognormal fading and Walfisch/Ikagami propagation model are implemented according to the method provided by Dr. Bonta.

Video traffic generation In order to generate video traffic for the simulation, we modified the video traffic generation script by Michael Savoric so that the video trace data can be fragmented in a given packet unit size.

2 Result

In addition to the simulation results already provided in [1], we introduce new simulations for four kinds of traffics

- CBR (Constant Bit Rate): *n* packets per second with a packet size of 512 bytes, where *n* can be changed for different simulations. The CBR data is transported over UDP.
- TCP: The continuous data packets of 512 bytes are transported over TCP so that the flow and congestion control algorithms are applied.
- Video: Trace data from movie *the Jurasic Park* with resolution of QCIF 176*144, frame rate of 25 frames/sec, frame sequence of IBBPBBPBBPBB, and color of YUV.
- VoIP: CBR with interval 20ms and 20 bytes per packet.

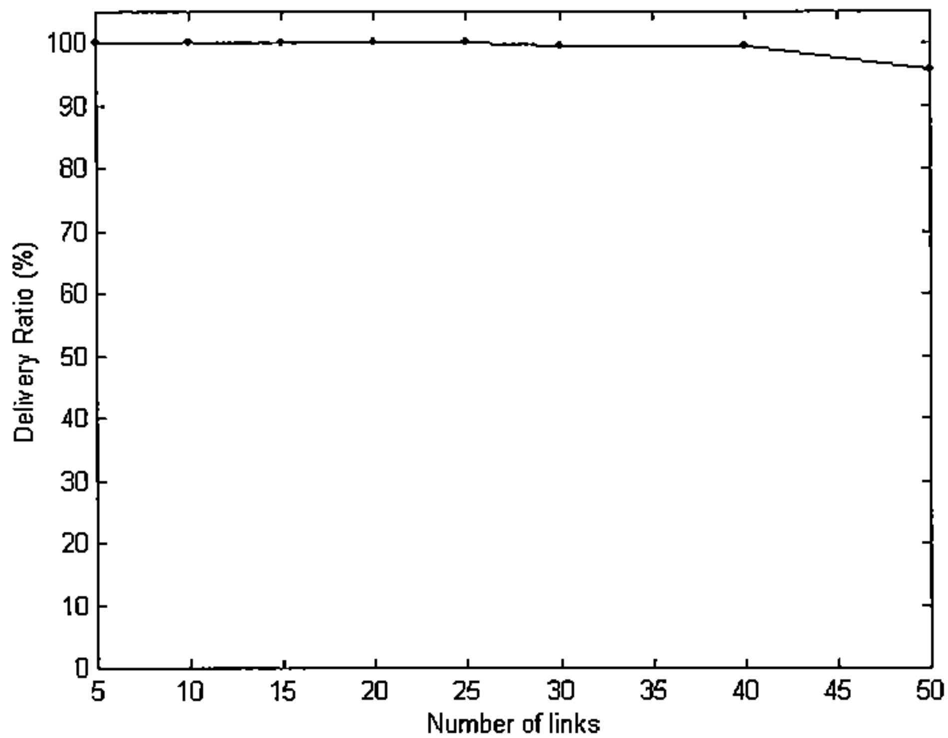
The start time of data transport is randomly generated and we define the seed of random generator at the beginning of simulation. We have tried different seed and found that it does not significantly affect the simulation results. So in the following simulation results, we only run program once for each data point. We have tried different simulation stop time up to 60 seconds, it turns out that 10 seconds is good enough for the scenario we are using and saves a lot of program running time. For different traffic and load, it takes about 1 to 3 minutes to obtain one data point, and there are hundreds of such points we need to collect.

2.1 CBR

For CBR and TCP, the mobile scenario is that 100 mobile nodes move in a 500m * 500m area with maximum moving speed = 3 m/s and pause time = 0.1 s. The number of CBR connections is 5, 10, 15, 20, 25, 30, 40 and 50.

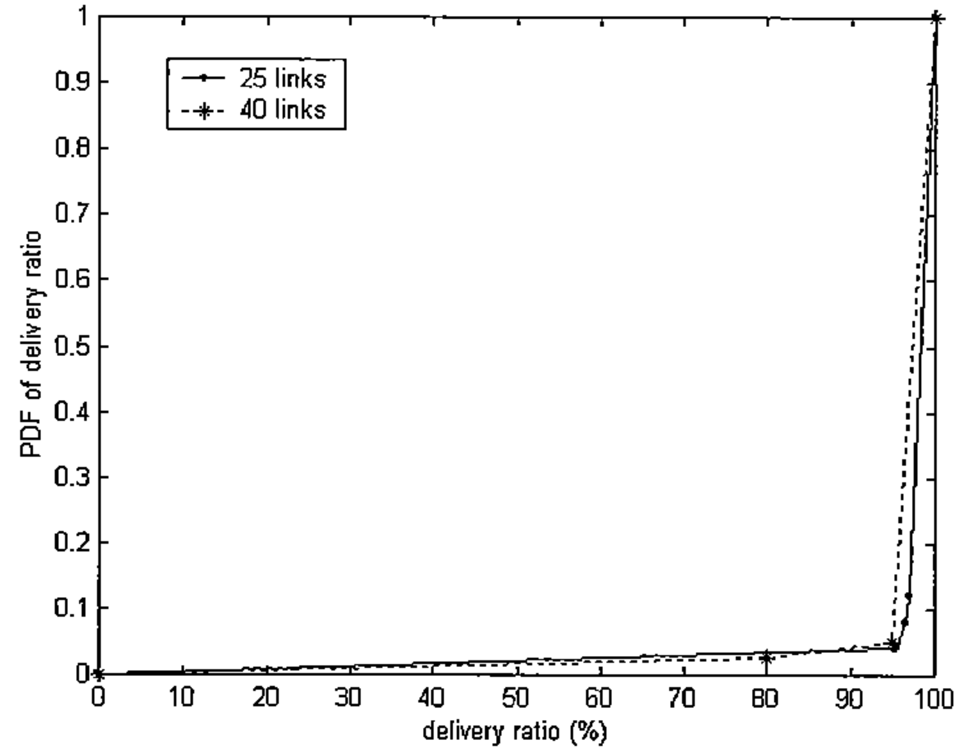
The simulation results are shown for

- data delivery ratio
- PDF of delivery ratio
- network transmission delay
- PDF of time delay
- variation of time delay
- data sending rate in the first 10 seconds
- PDF of data sending rate
- Throughput
- Hop count



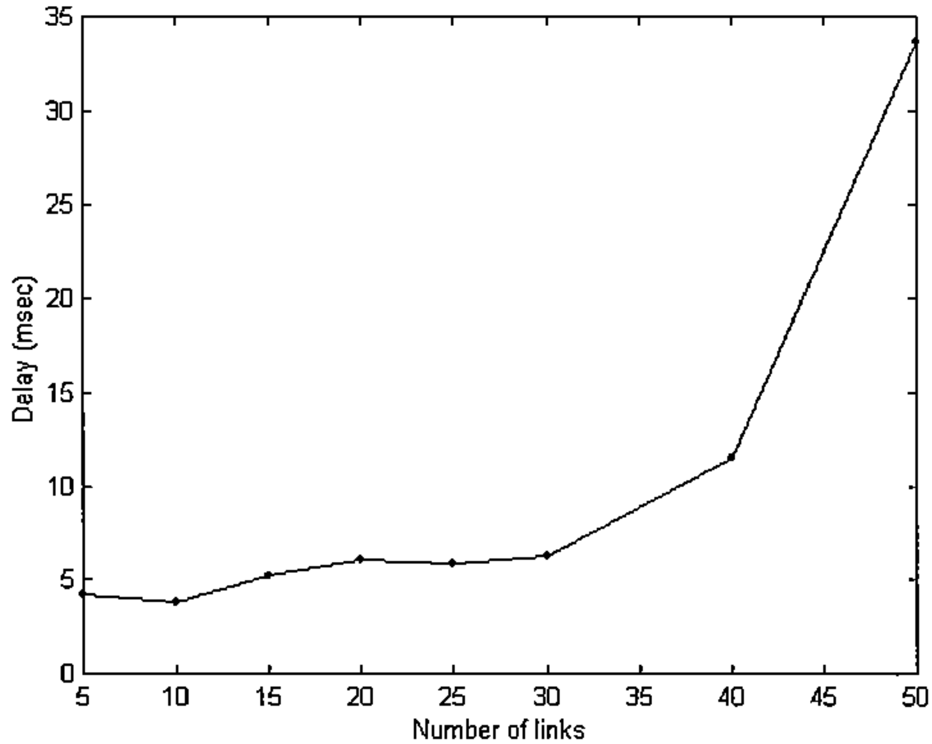
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- Almost all sent packets can be received by the receiver. The delivery ratio only drops a little bit for higher number of links.



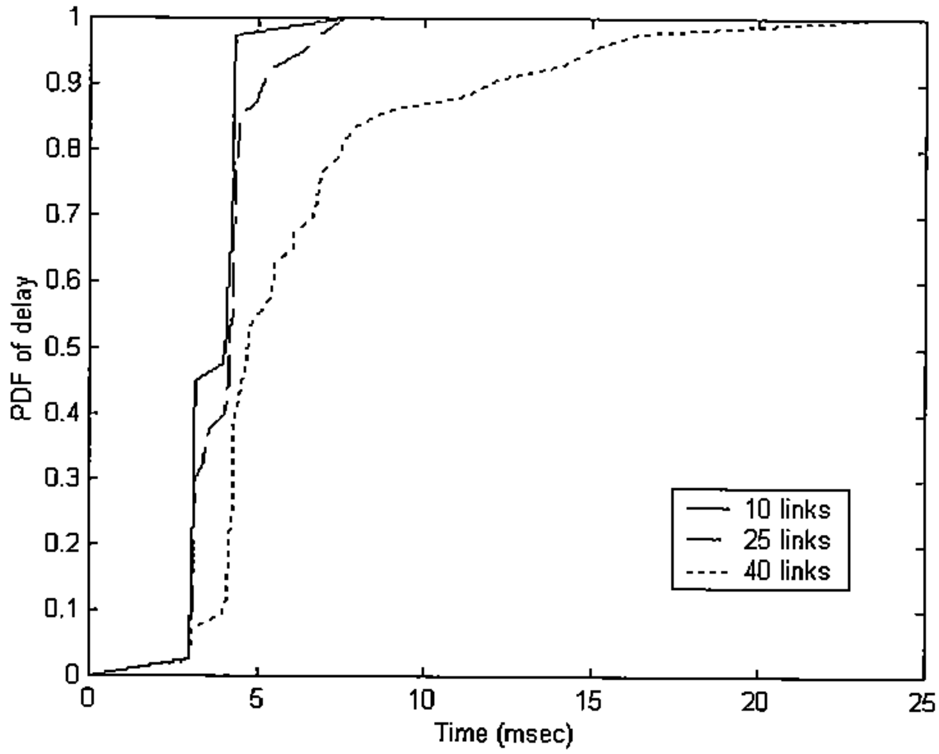
PDF of delivery ratio

- PDF of delivery ratio shows the probability distribution of all possible delivery ratios.
- PDF of delivery ratio is measured by counting the number of connections for every delivery ratio and dividing it by the total number connections.
- Two PDFs are displayed for loads of 25 links and 40 links. It is shown that more than 95% of connections can transport data at delivery ratio of 95% or higher.



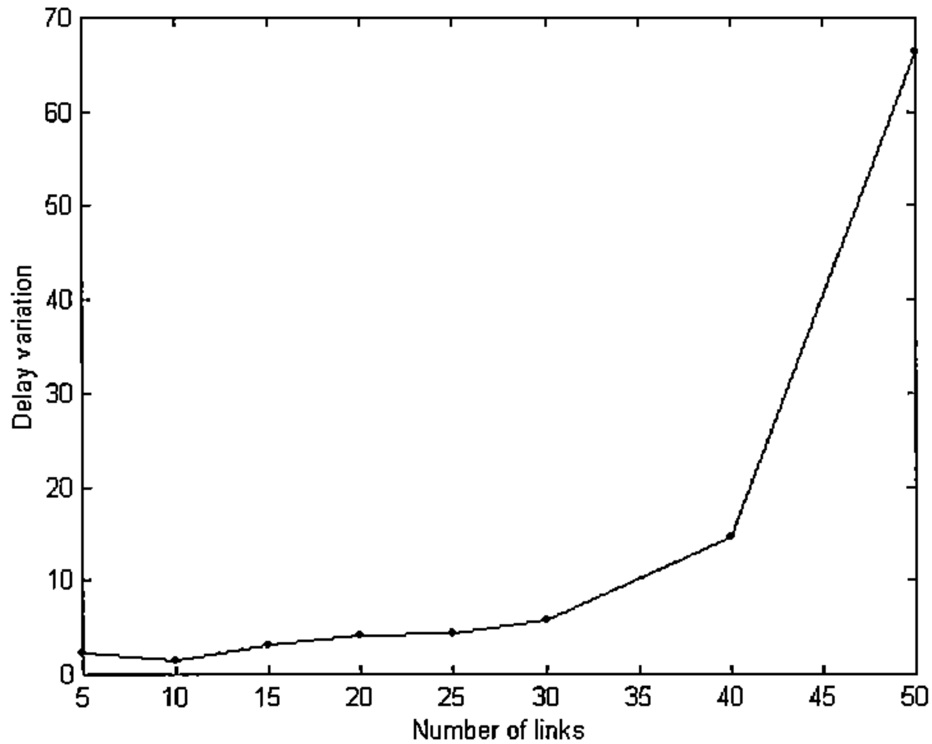
Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging them.
- The transmission delay increases when the number of connections increases. This is because different rates have been employed by the link adaptation algorithm, and for higher network load (more connections), smaller rates tend to be employed due to network congestion.



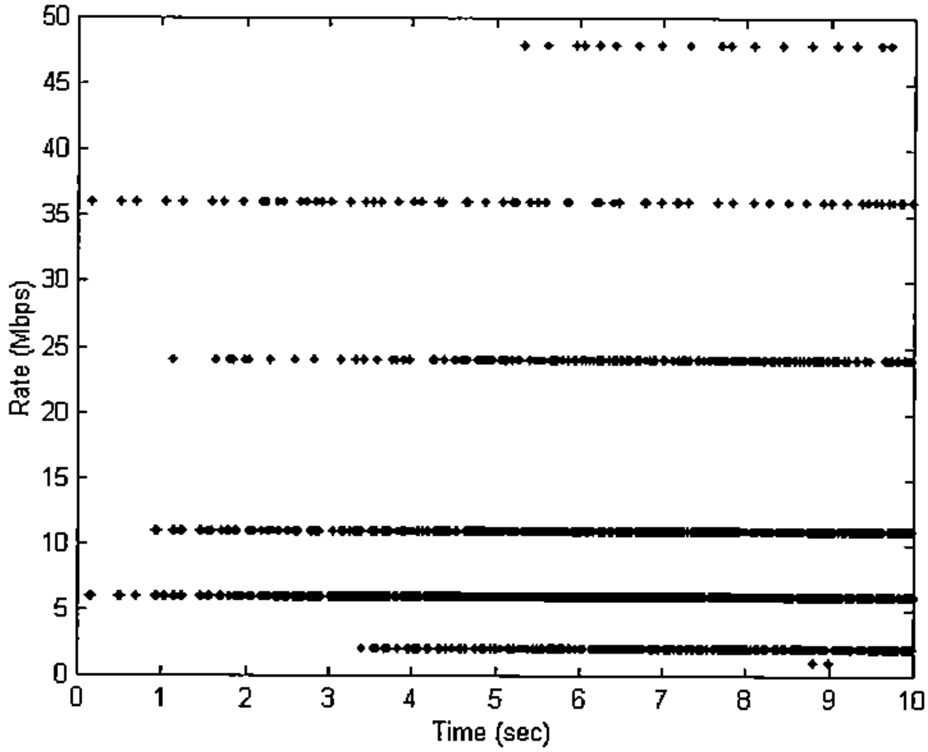
PDF of transmission delay

- PDF of transmission delay shows the probability distribution of all possible delays.
- PDF of transmission delay is measured by counting the number of packets for every transmission delay and dividing it by the total number packets.
- Three PDFs are displayed for loads of 10, 25 and 40 links. It is shown that all packets can be received within 7.5 msec. for 10 and 25 links, but it takes longer for some packets when there are 40 links. This matches the previous figure of transmission delay.



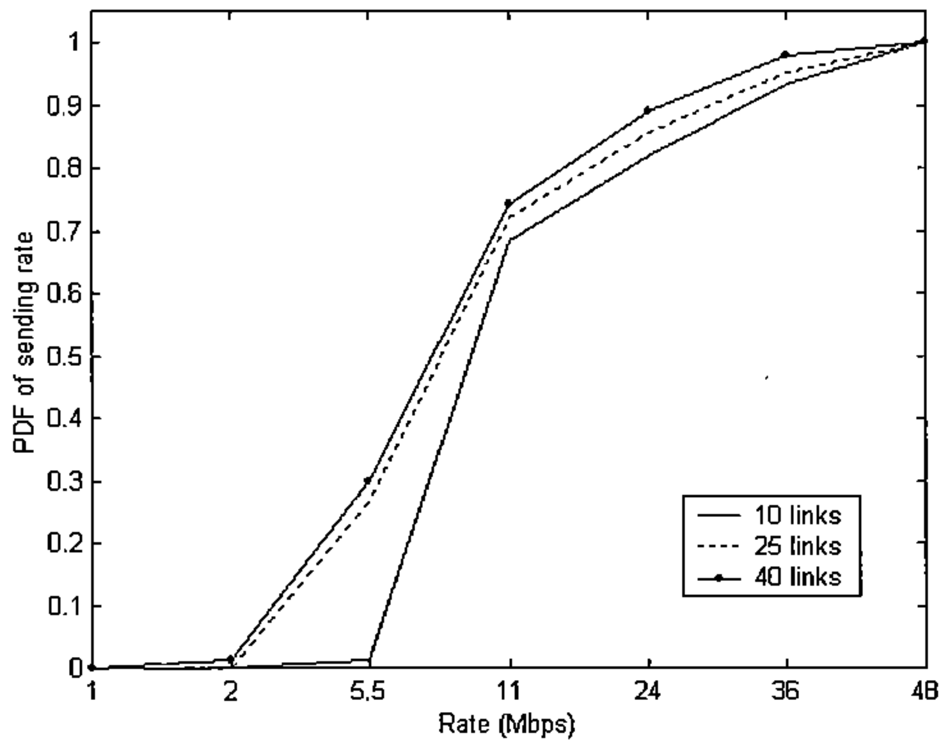
Variation of transmission delay for different loads

- Variation of transmission delay (or Jitter) is the standard deviation of transmission delay.
- Variation of transmission delay is measured by calculating the square root of the average of the square of the difference between a delay and the average transmission delay.
- It is shown that the transmission delay varies more significantly for higher loads. This is because that, at high load, different data sending rates are employed, depending on if there is network congestion.



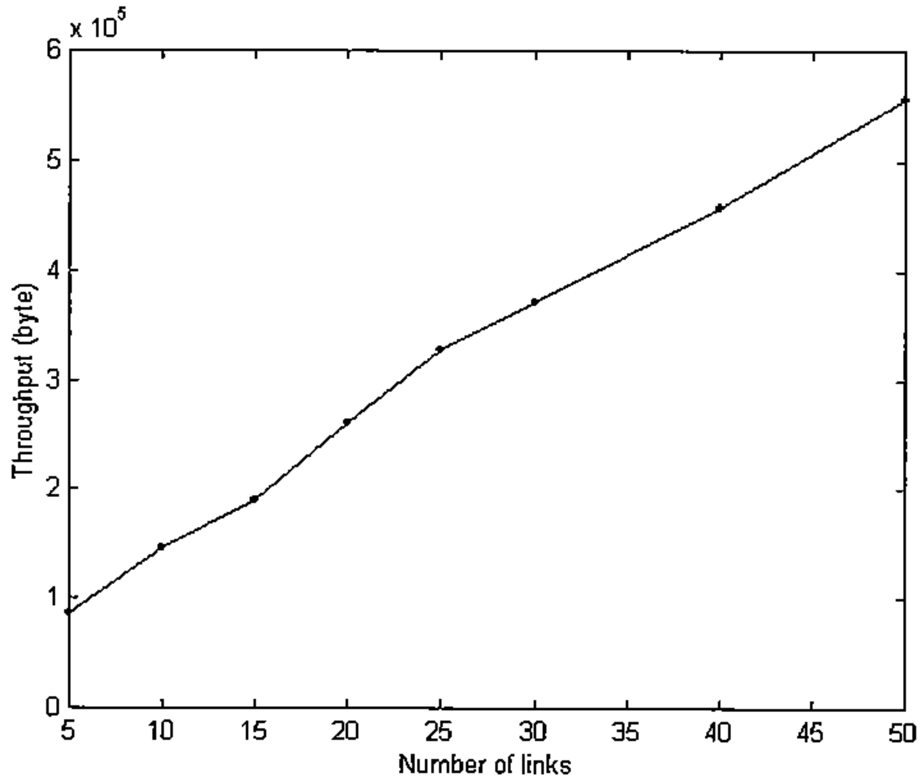
Data sending rate

- Data sending rate is the data transmission rate chosen by the sender at MAC layer of IEEE 802.11, according to the network condition.
- This figure shows the data sending rate chosen by all sent packets. Every single point represents that the corresponding rate is chosen by one packet sent at the corresponding time.
- The density of points for each rate represents how often this rate is chosen.



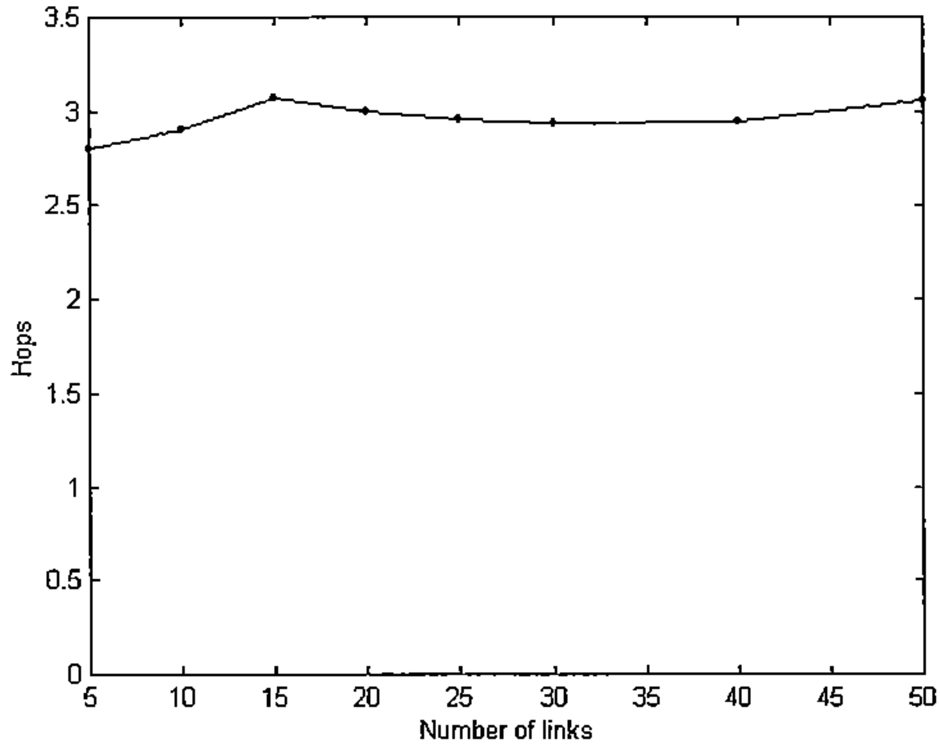
PDF of data sending rate

- PDF of data sending rate shows the probability distribution of all possible data sending rate.
- PDF of data sending rate is measured by counting the number of packets for every sending rate and dividing it by the total number of packets.
- Three PDFs are displayed for loads of 10, 25 and 40 links. It is shown that higher load tends to employ lower sending rates.



Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- Throughput only counts the successfully received data, so it is less or equal to the total number of sent data. Throughput increases for higher loads because there are more data being sent and received.

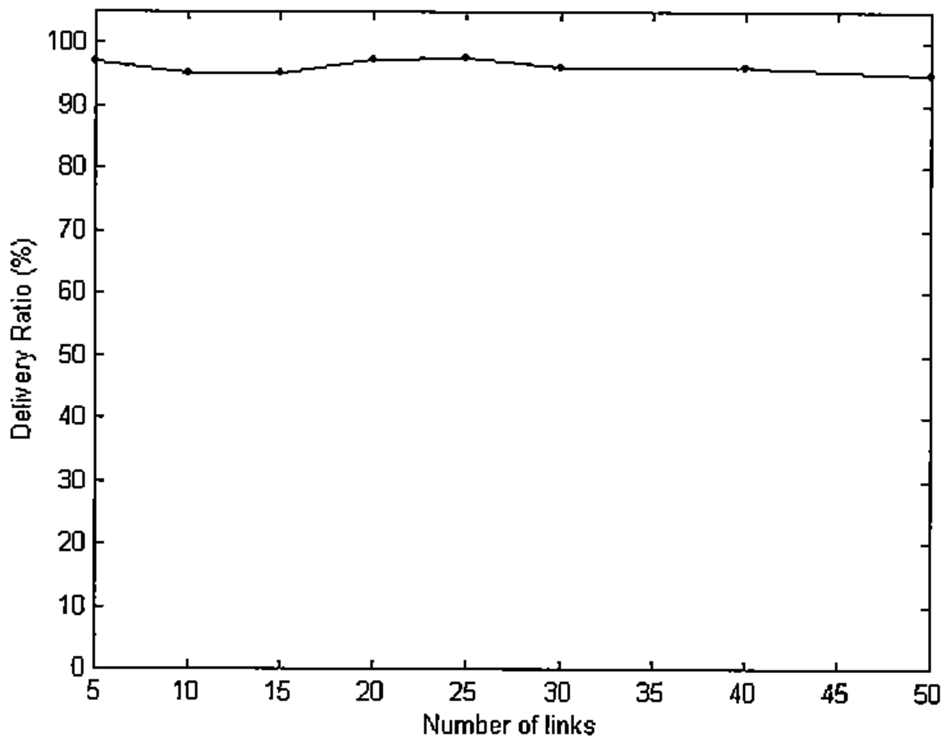


Hop count for different loads

- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them.
- It is shown that the hop count does not change much for different network loads.

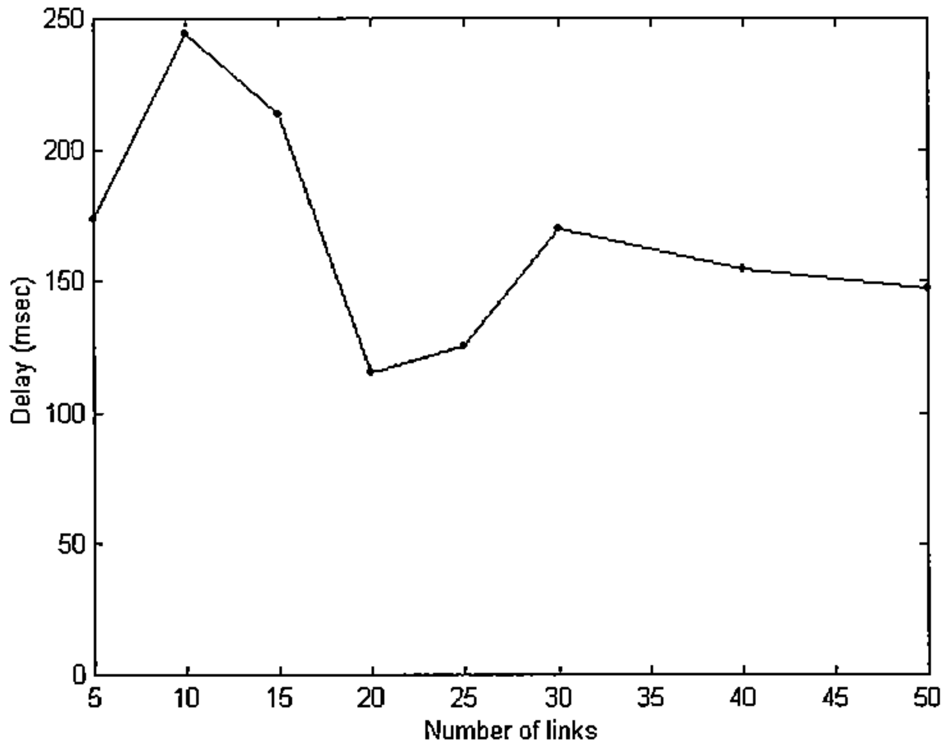
2.2 TCP

In the same scenario as CBR, TCP traffic can also maintain very good data delivery ratio for no more than 50 connections. But the time delay is much longer than that of CBR due to the congestion control in TCP.



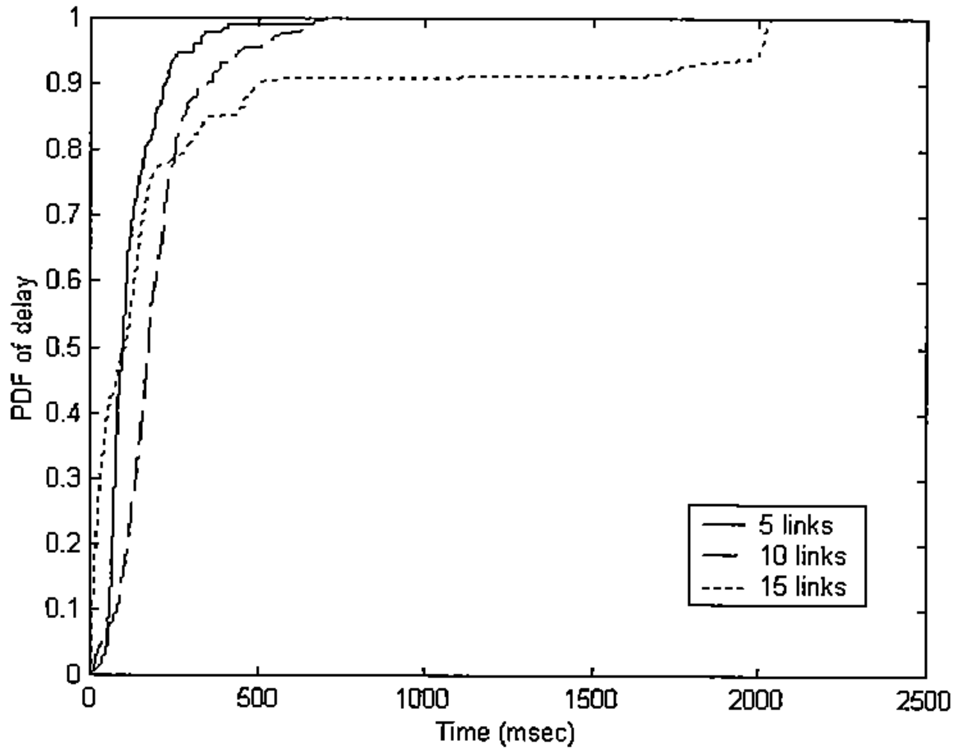
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- At least 95% of sent packets can be received by the receiver. The delivery ratio only drops a little bit for higher number of links.



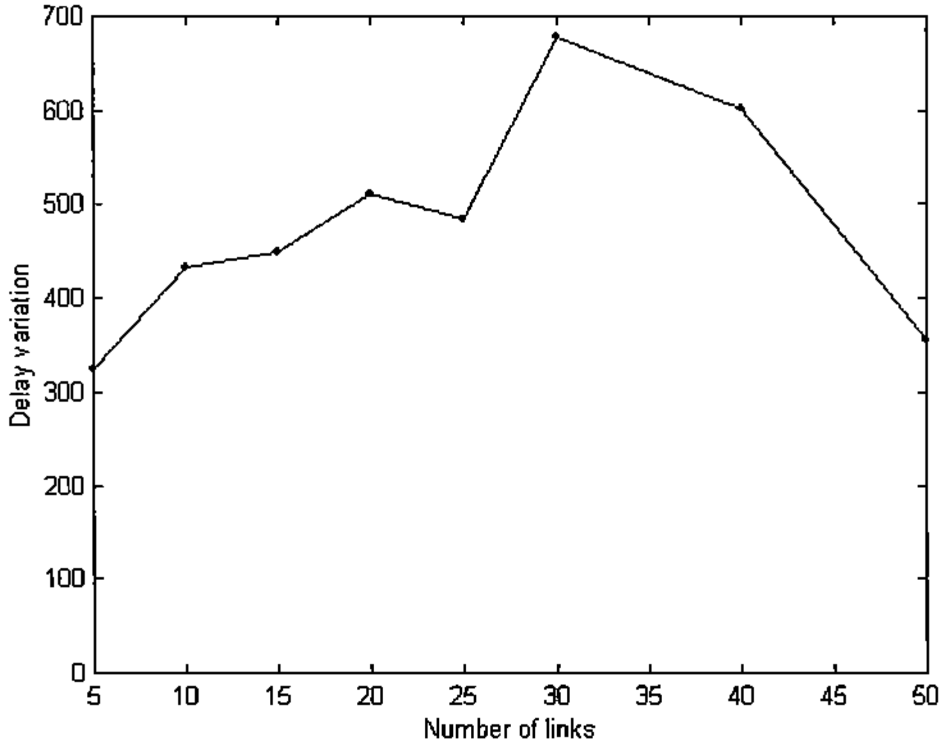
Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging them.
- In contrast to CBR, the transmission delay for TCP traffic is longer even for low network loads. This can be explained as that when high data rate is chosen at low loads, it may introduce congestion in the intermediate routers and the congestion control mechanism of TCP is triggered, which will delay the data transmission.



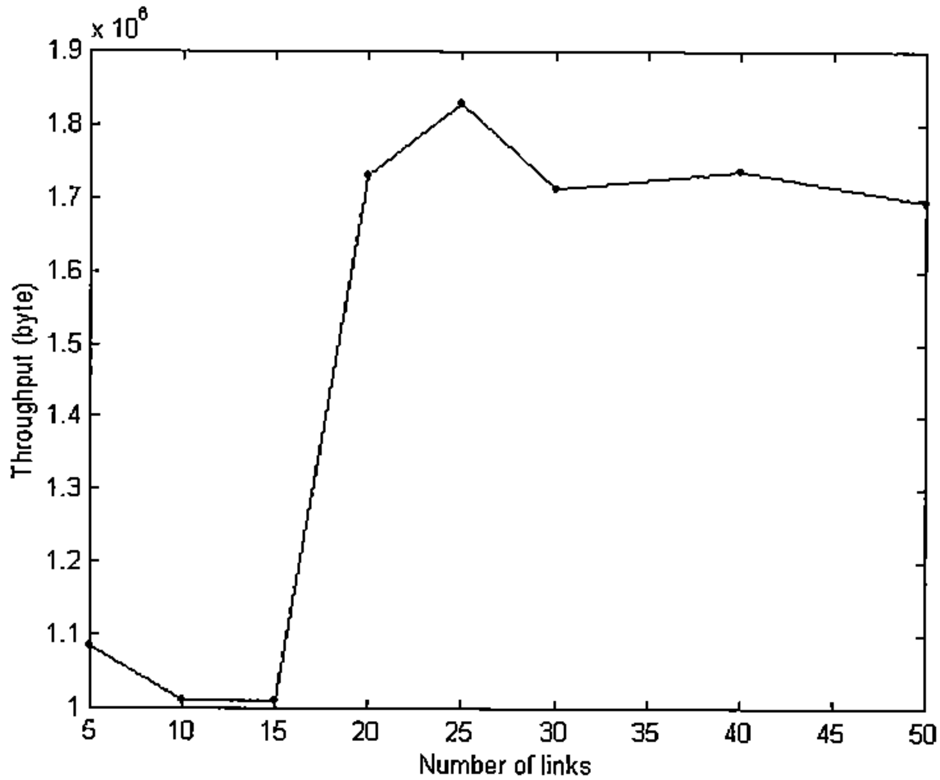
PDF of transmission delay

- PDF of transmission delay shows the probability distribution of all possible delays.
- PDF of transmission delay is measured by counting the number of packets for every transmission delay and dividing it by the total number packets.
- It is shown that all packets can be received within 0.7 sec. for 5 and 10 links, but it takes longer for some packets when there are 15 links.



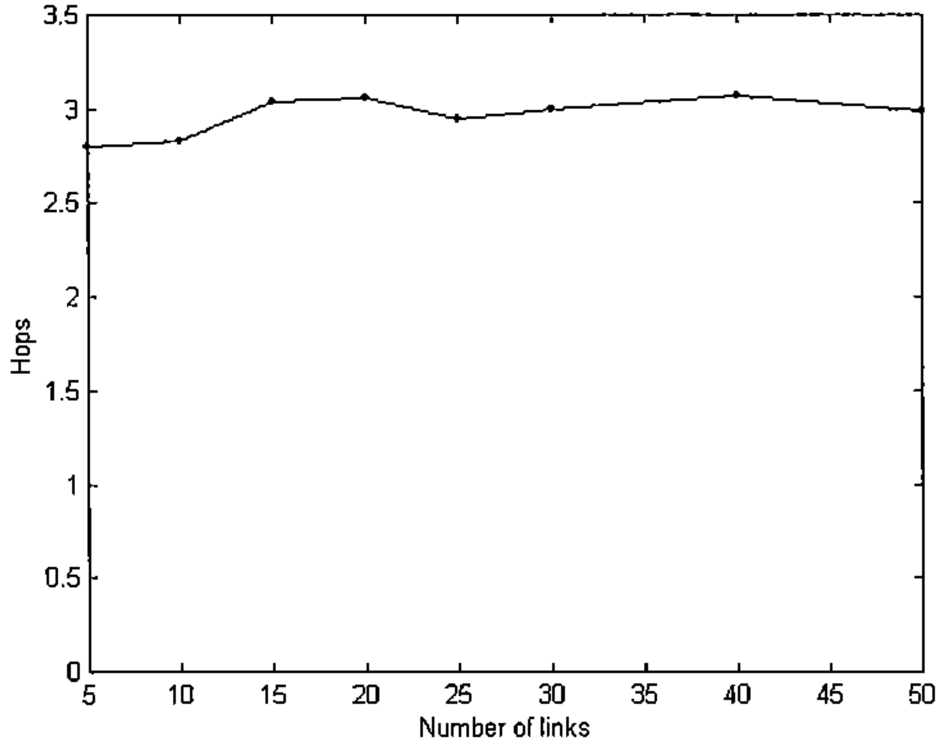
Variation of transmission delay for different loads

- Variation of transmission delay (or Jitter) is the standard deviation of transmission delay.
- Variation of transmission delay is measured by calculating the square root of the average of the square of the difference between a delay and the average transmission delay.
- In contrast to CBR, the variation of transmission delay for TCP traffic is high even for low network loads. This is due to the same reason as the transmission delay.



Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- Throughput increases for higher loads because there are more data being sent and received. But when the delivery ratio for higher loads drops, the throughput may also drops.



Hop count for different loads

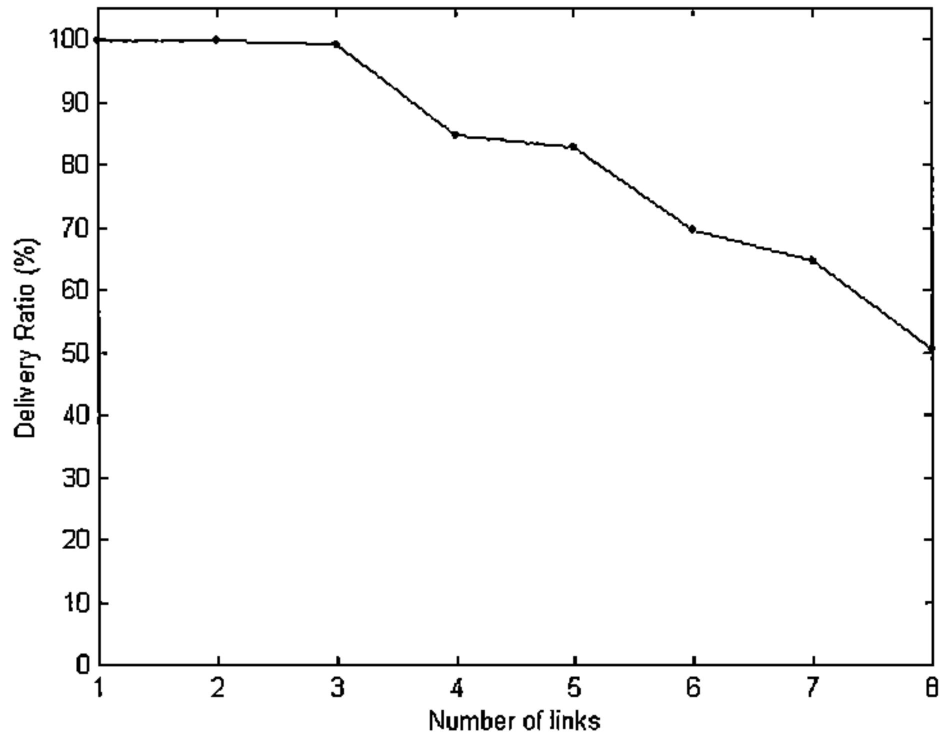
- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them.
- It is shown that the hop count does not change much for different network loads.

2.3 Video

For video and VoIP, the mobile scenario is that 50 mobile nodes move in a 250m * 250m area with maximum moving speed = 3 m/s and pause time = 0.1 s. The number of connections is 1 to 8.

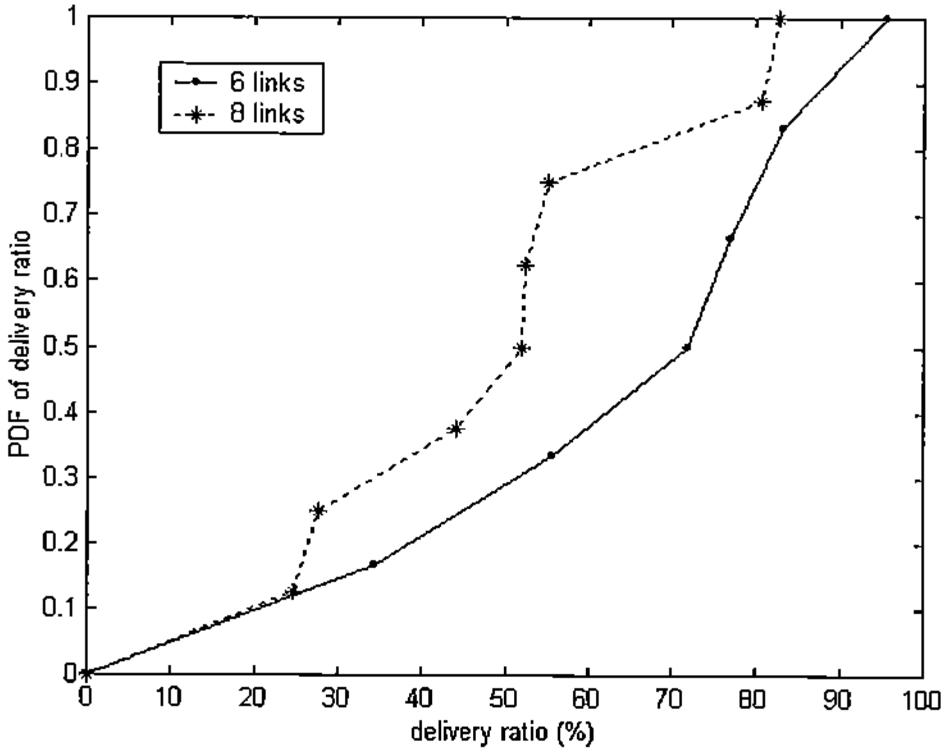
In addition to the previous observations for CBR and TCP, heavy video traffic introduces significant packet loss and transmission delay. This is mainly due to the high data rate of video. Another possible reason is that the current implementation of IEEE 802.11 in ns-2 is based on the IEEE standard in late 1990's, so it may not be as accurate as we expect to be used for the simulation of newly approved standards like IEEE 801.11g or 801.11a. We did update some parameters, such as the slot time, but we are not sure if the current update is complete. Hence the given simulation results only demonstrate the system performance qualitatively so that a particular value for some parameters does not imply the exact same thing in the real system.

One more notation about network transmission delay is that it is not equivalent to the time interval of packet, instead, the network transmission delay only represents the time difference between the sending time and receiving time of the same packet. For example, although the video packets are sent at interval of 0.04 second and the average network transmission delay for 5 connections is about 0.35 second, but this does not mean that the QoS of video applications is necessarily violated. Actually, the 0.35-second delay is still acceptable for most video applications. For video streaming applications, some kind of buffer mechanisms are usually employed to deal with transmission delay so that a certain period time of video is stored before it is actually played. For interactive video applications, I cannot tell the exact tolerable limit for time delay because it is a kind of subjective criteria depending on different users.



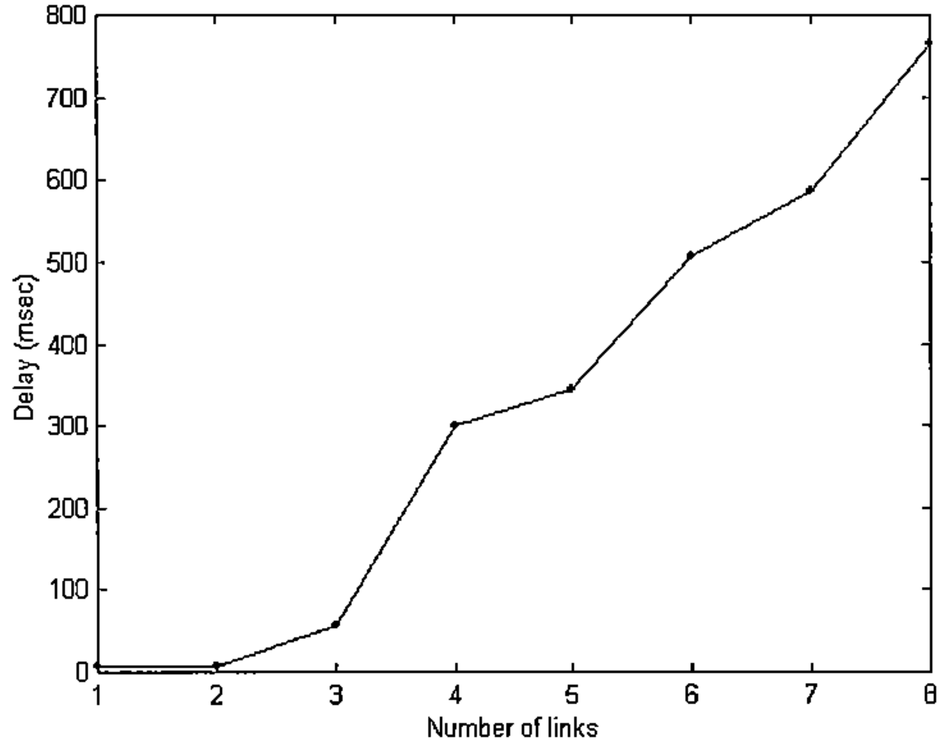
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- The delivery ratio drops significantly when network load increases as explained at the beginning of this subsection.



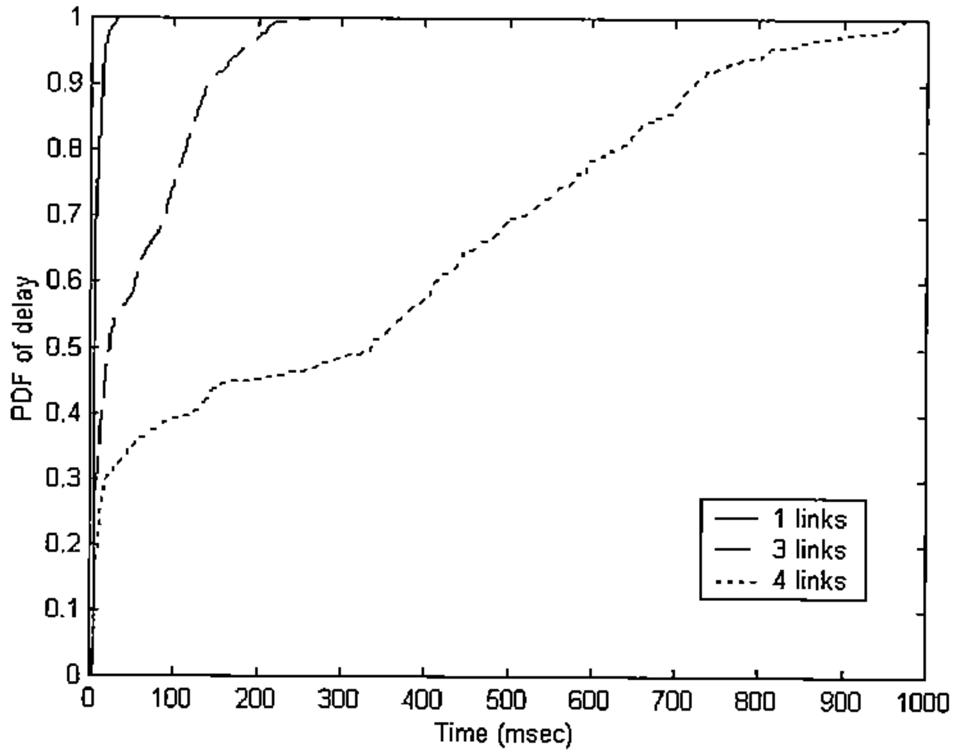
PDF of delivery ratio

- PDF of delivery ratio shows the probability distribution of all possible delivery ratios.
- PDF of delivery ratio is measured by counting the number of connections for every delivery ratio and dividing it by the total number connections.
- Two PDFs are displayed for loads of 6 links and 8 links. It is shown that higher load tends to reduce the delivery ratio.



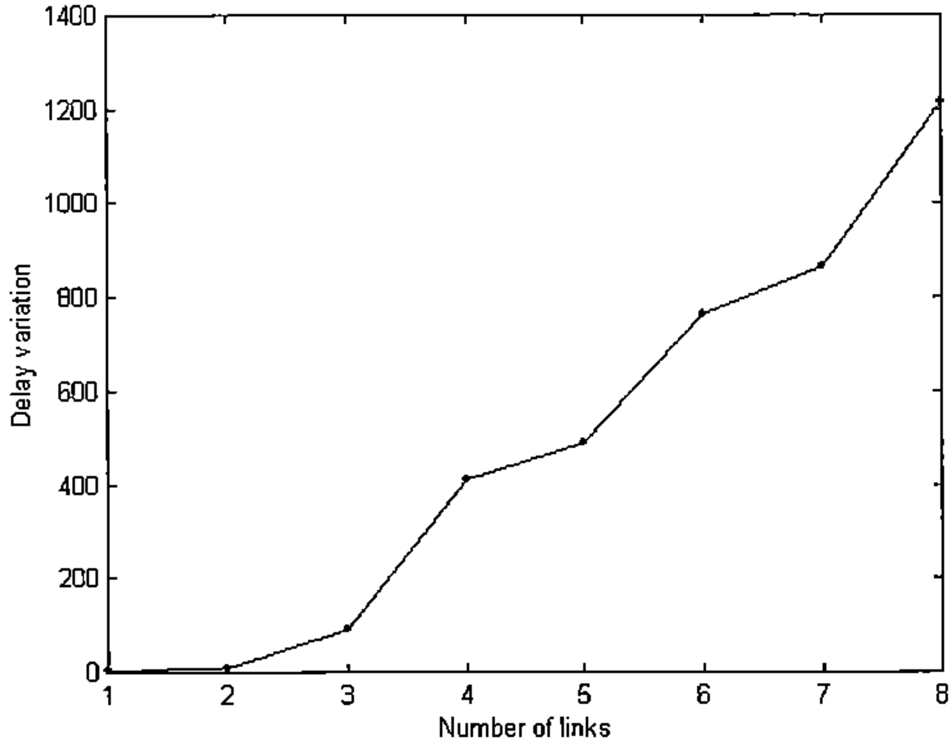
Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging them.
- The transmission delay increases when the number of connections increases. This is because different rates have been employed by the link adaptation algorithm, and for higher network load (more connections), smaller rates tend to be employed due to network congestion.



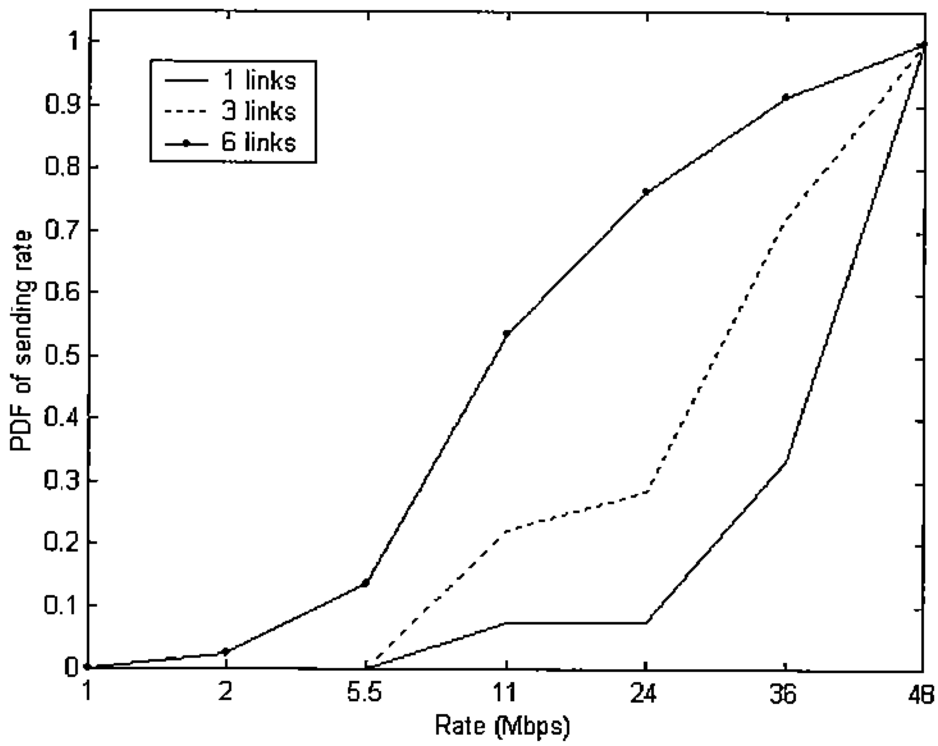
PDF of transmission delay

- PDF of transmission delay shows the probability distribution of all possible delays.
- PDF of transmission delay is measured by counting the number of packets for every transmission delay and dividing it by the total number packets.
- It is shown that higher load tends to take longer time to transmit data, which matches the previous figure of transmission delay.



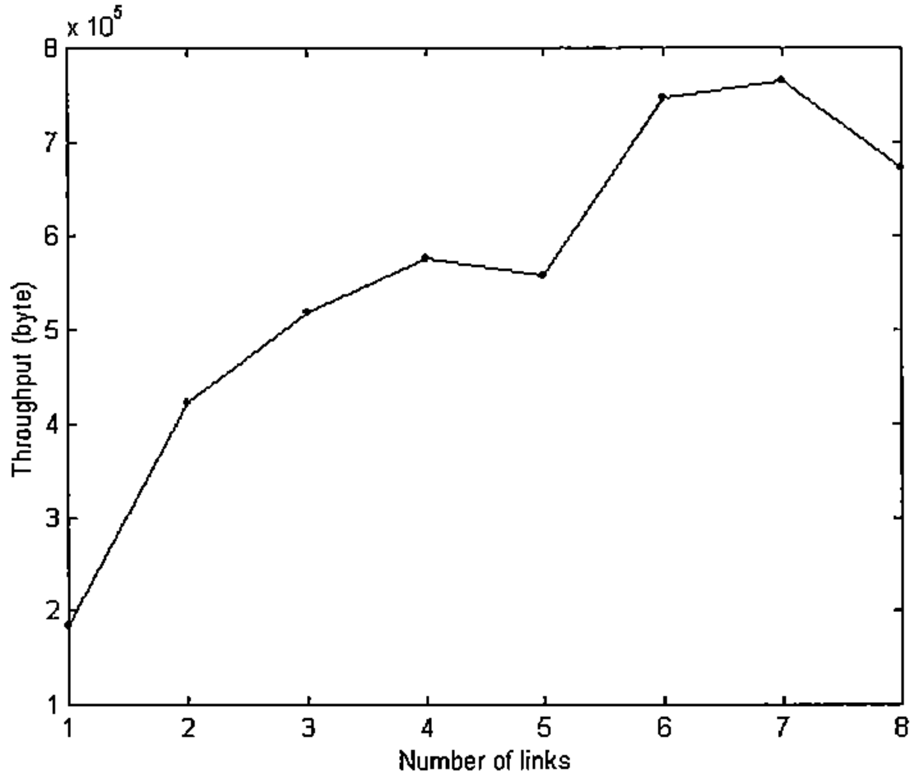
Variation of transmission delay for different loads

- Variation of transmission delay (or Jitter) is the standard deviation of transmission delay.
- Variation of transmission delay is measured by calculating the square root of the average of the square of the difference between a delay and the average transmission delay.
- It is shown that the transmission delay varies more significantly for higher loads. This is because that, at high load, different data sending rates are employed, depending on if there is network congestion.



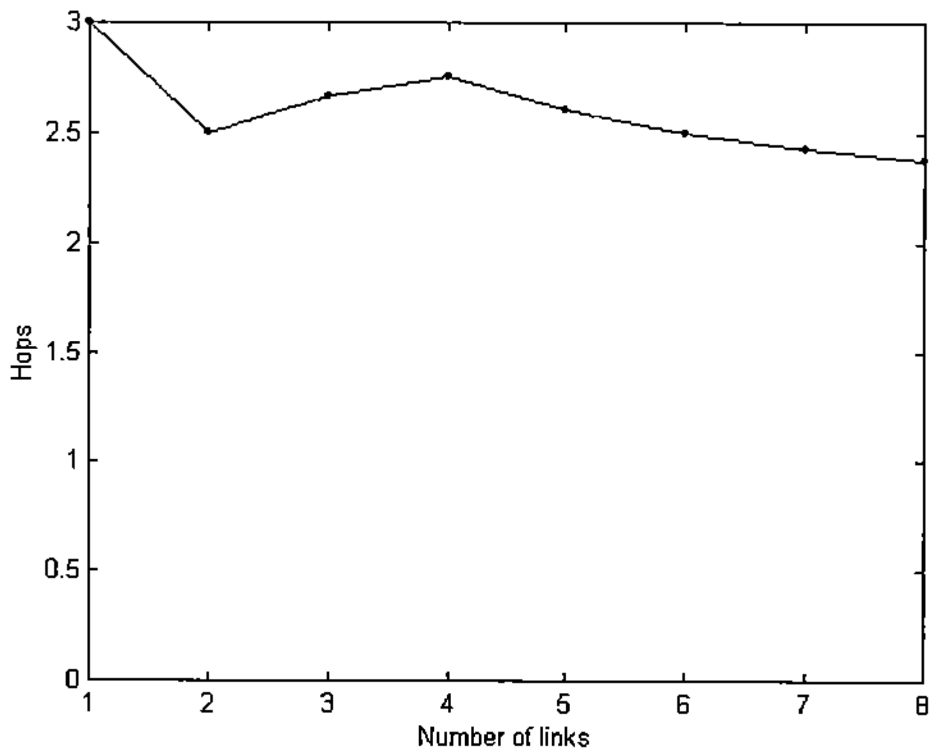
PDF of data sending rate

- PDF of data sending rate shows the probability distribution of all possible data sending rate.
- PDF of data sending rate is measured by counting the number of packets for every sending rate and dividing it by the total number of packets.
- Three PDFs are displayed for loads of 1, 3 and 6 links It is shown that higher load tends to employ lower sending rates.



Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- In most cases, throughput increases for higher loads because there are more data being sent and received. But since the packet size for video is varying in a large range, there might be the case when the throughput decreases although the number of packets increases, such as 5-link case in the figure.

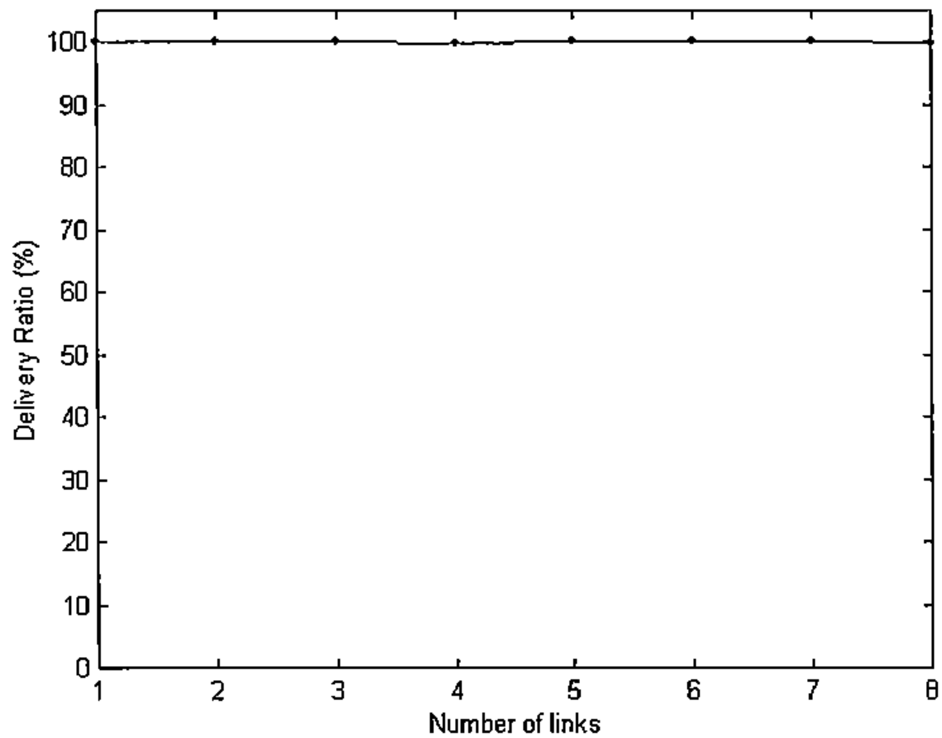


Hop count for different loads

- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them
- It is shown that the hop count does not change much for different network loads.

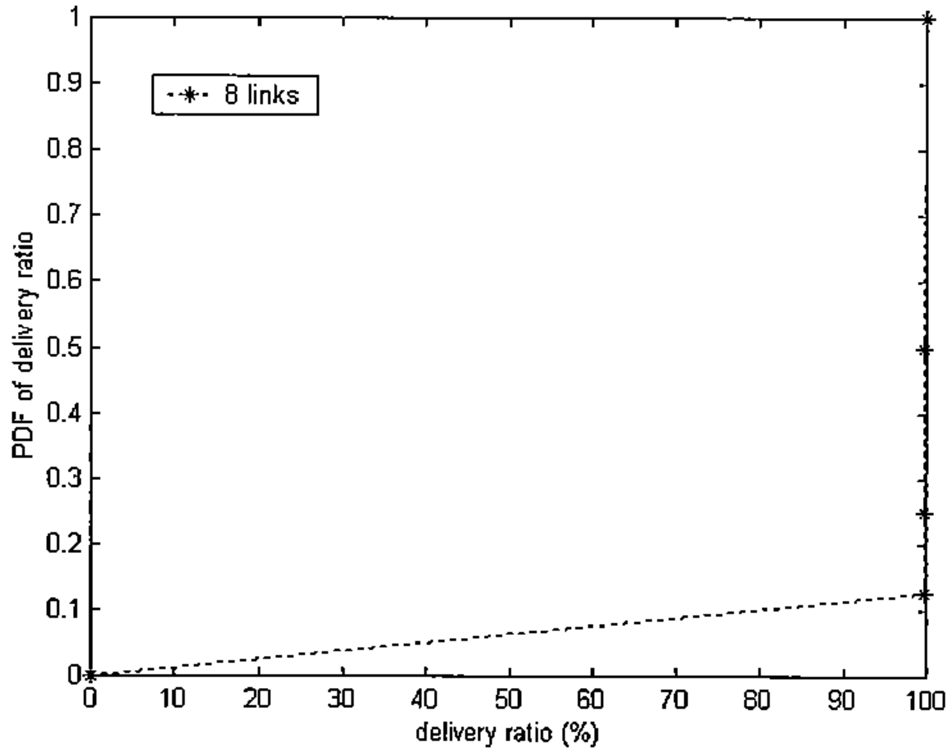
2.4 VoIP

In the same scenario as video, VoIP performs excellent.



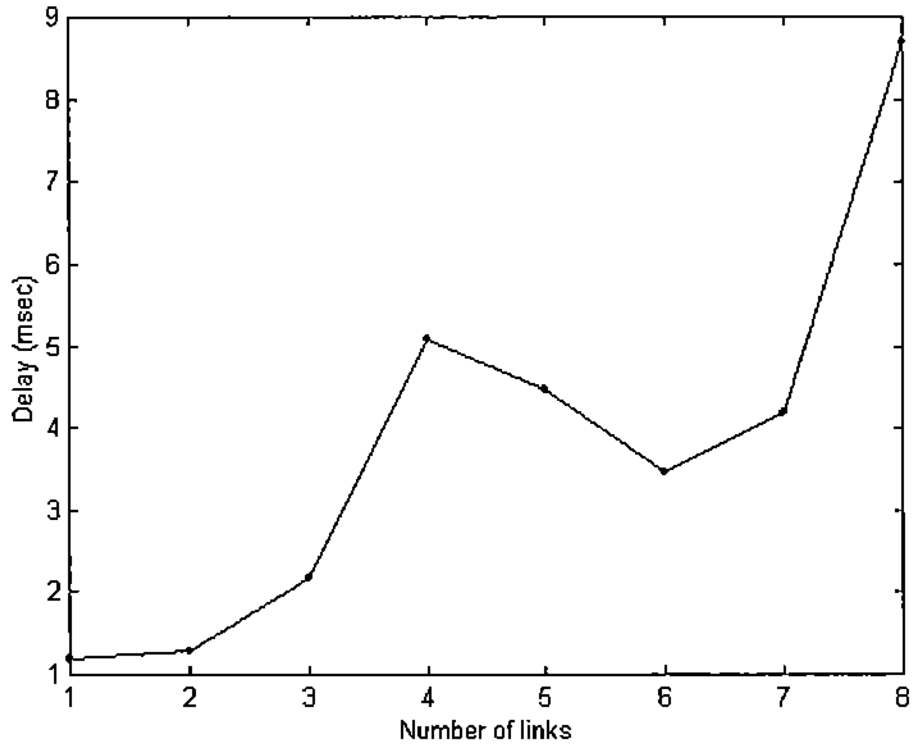
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- Almost all sent packets can be received by the receiver. The delivery ratio only drops a little bit for higher number of links.



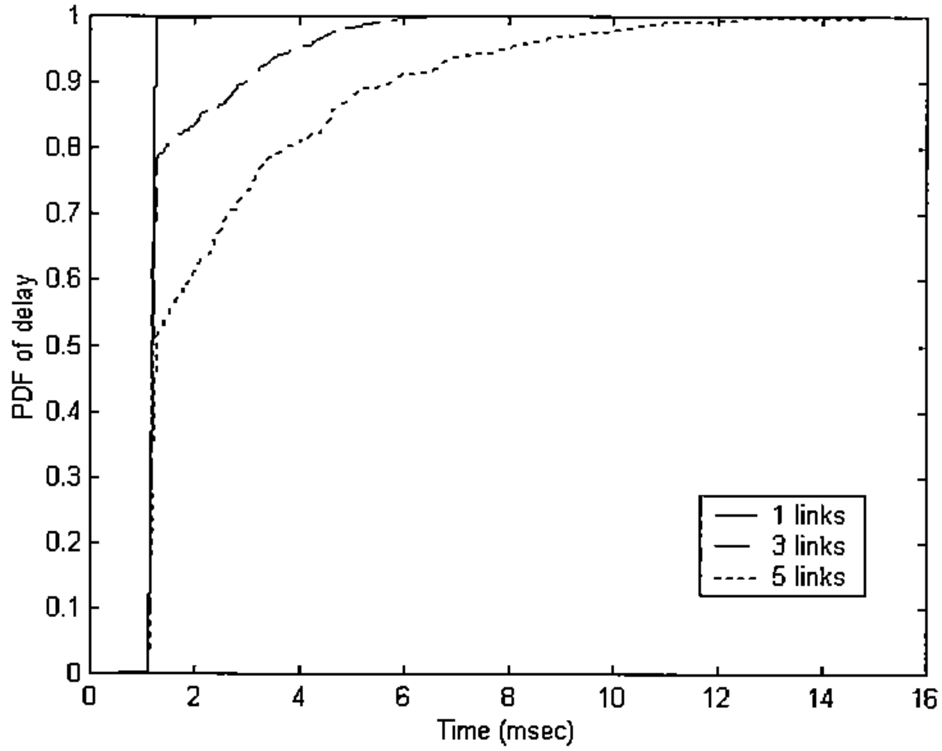
PDF of delivery ratio

- PDF of delivery ratio shows the probability distribution of all possible delivery ratios.
- PDF of delivery ratio is measured by counting the number of connections for every delivery ratio and dividing it by the total number connections.
- It is shown that almost 90% of connections can successfully transport all data.



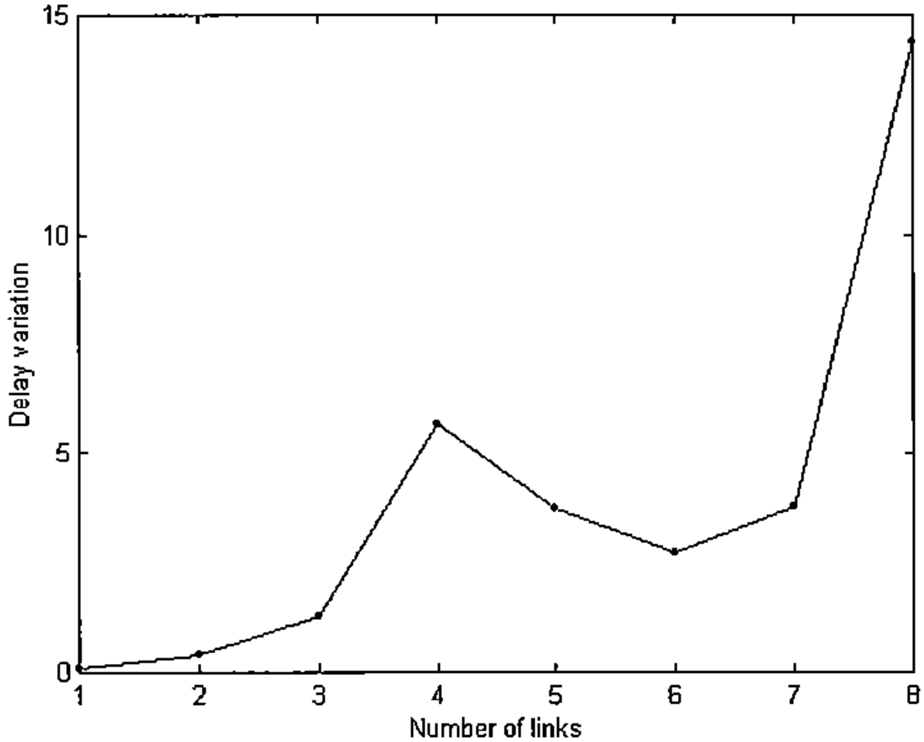
Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging them.
- In most cases, the transmission delay increases when the number of connections increases. An interesting phenomenon in the figure is that the time delays for 4 and 5 links are higher than those for 6 and 7 links. This is probably due to the mobility of mobile nodes, which might introduce lower signal-to-noise (SNR) ratio even if the number of mobile nodes decreases so that lower sending rate is selected.



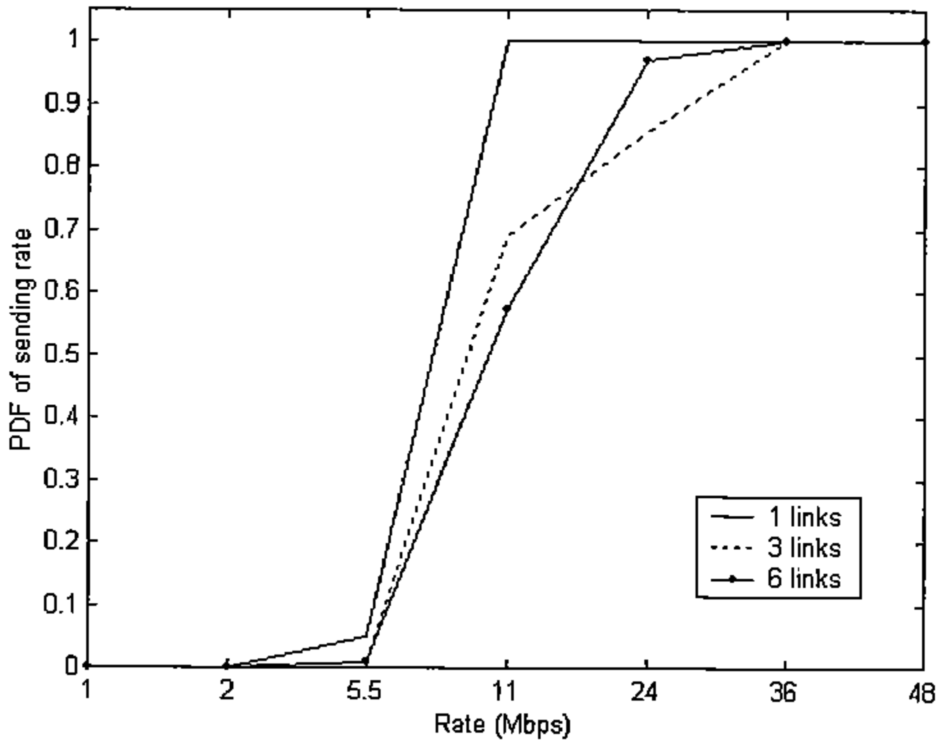
PDF of transmission delay

- PDF of transmission delay shows the probability distribution of all possible delays.
- PDF of transmission delay is measured by counting the number of packets for every transmission delay and dividing it by the total number packets.
- Three PDFs are displayed for loads of 1, 3 and 5 links. It is shown that it usually takes longer to transmit data when the network load increases.



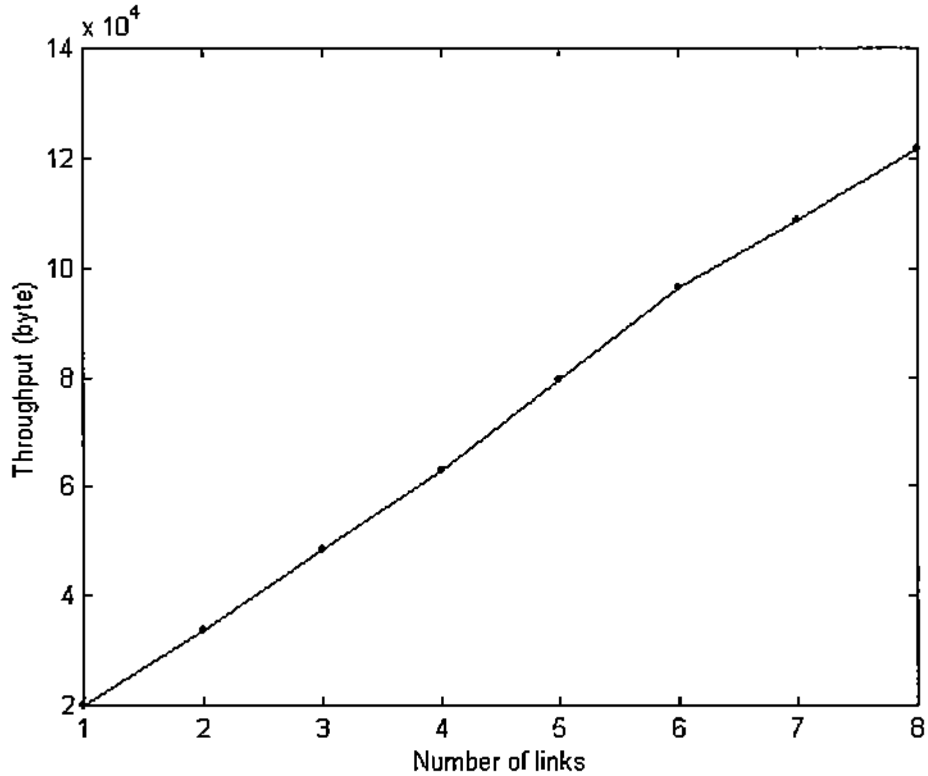
Variation of transmission delay for different loads

- Variation of transmission delay (or Jitter) is the standard deviation of transmission delay.
- Variation of transmission delay is measured by calculating the square root of the average of the square of the difference between a delay and the average transmission delay.
- It is shown that the transmission delay usually varies more significantly for higher loads. The unsmooth delay variation at 4 and 5 links can be explained similarly as under the figure of transmission delay.



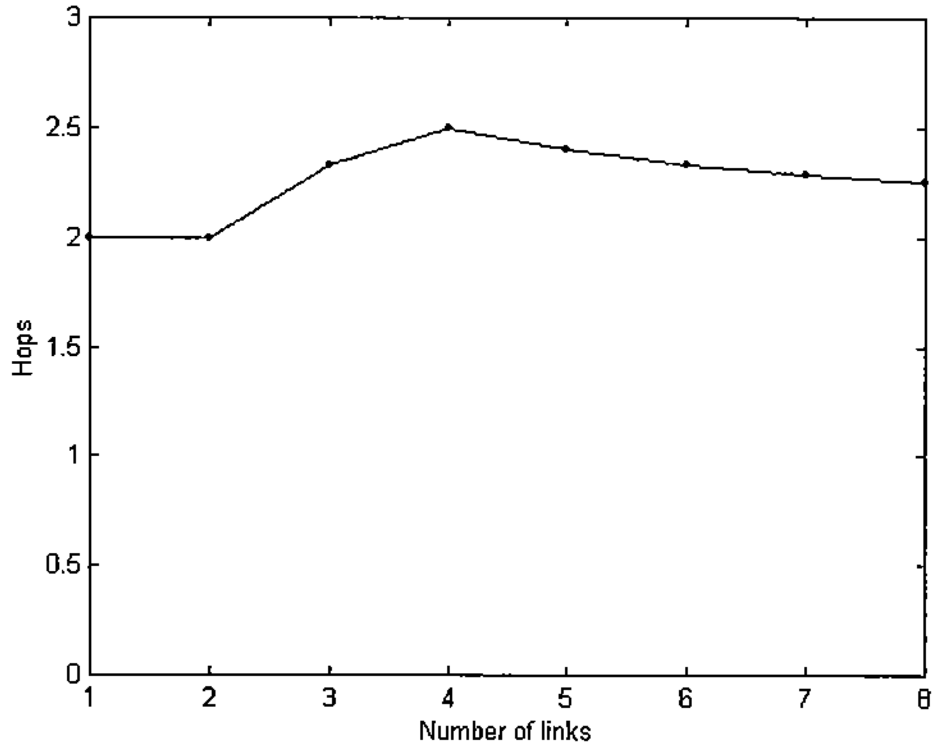
PDF of data sending rate

- PDF of data sending rate shows the probability distribution of all possible data sending rate.
- PDF of data sending rate is measured by counting the number of packets for every sending rate and dividing it by the total number of packets.
- Three PDFs are displayed for loads of 1, 3 and 6 links.



Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- Throughput increases for higher loads because there are more data being sent and received.

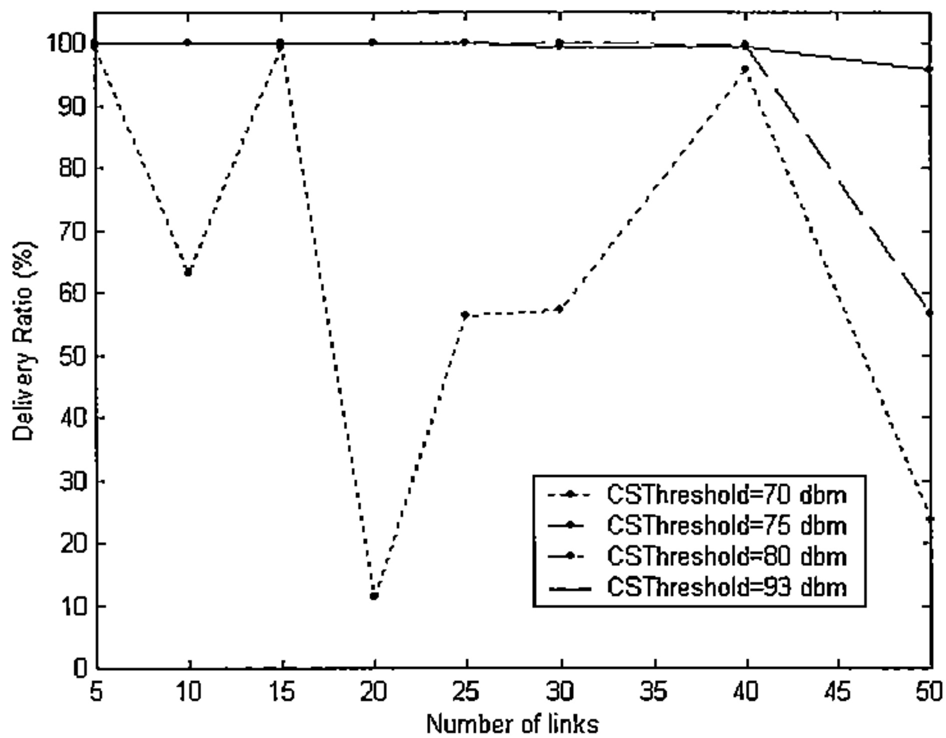


Hop count for different loads

- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them.
- It is shown that the hop count does not change much for different network loads.

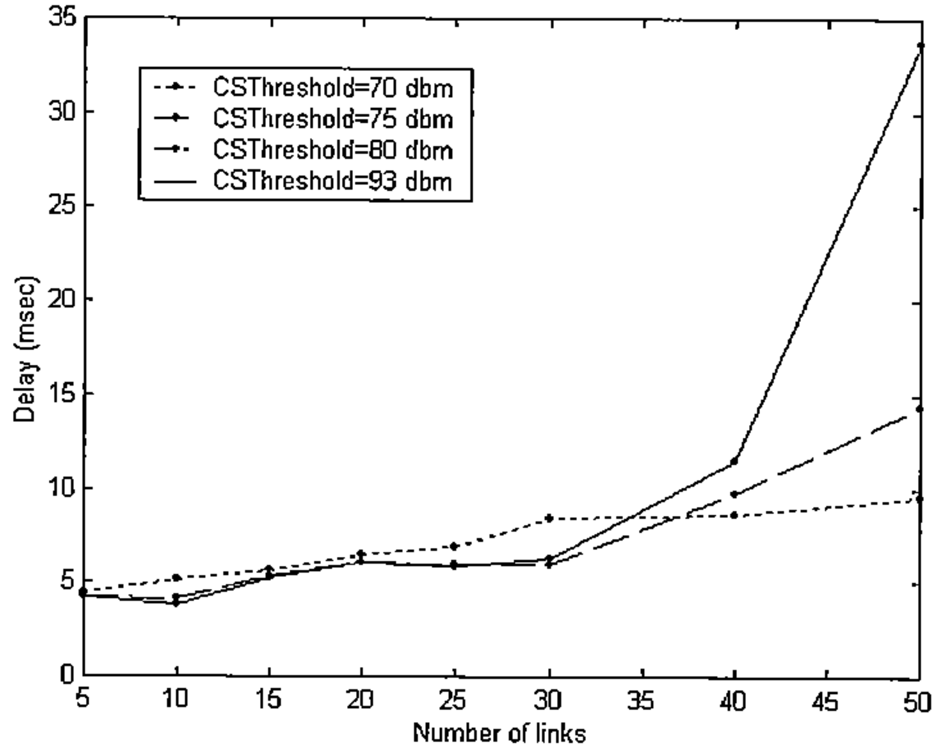
2.5 Carrier sense threshold

We compare the system performance of different carrier sense threshold (CSThresh) at 70, 75, 80, and 93 dbm.



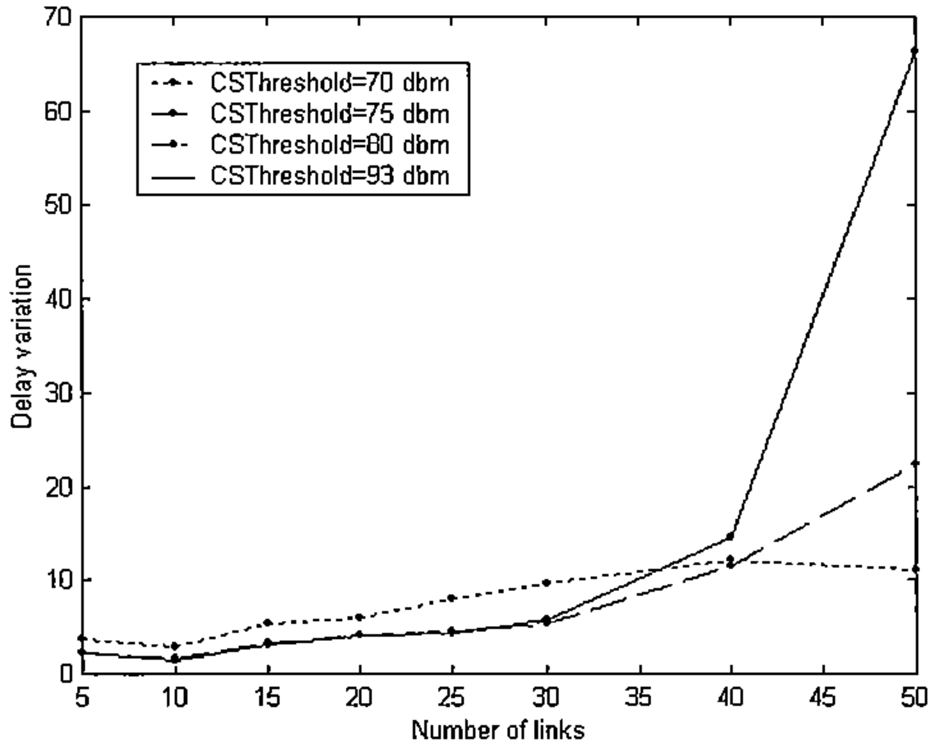
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- It is obvious that higher CSThresh maintains higher delivery ratio. For low CSThreshold such as 70 dbm, a high percentage of packets can be lost. But we have not found a good reason for the unsmooth curve in this case.



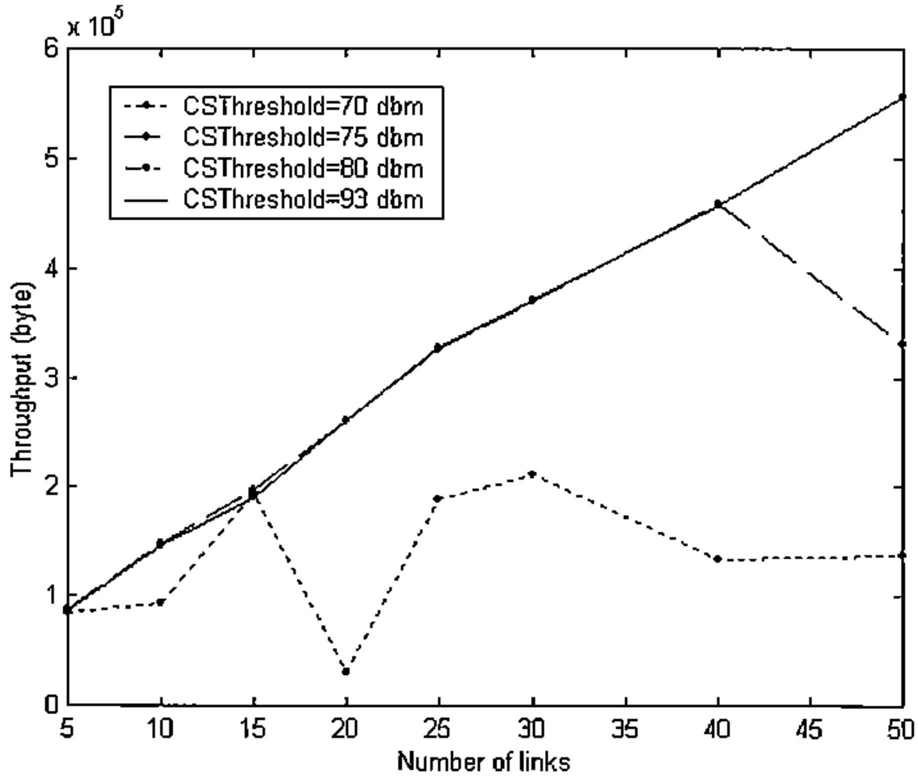
Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging them.
- The transmission delay increases when the number of connections increases. This is because different rates have been employed by the link adaptation algorithm, and for higher network load (more connections), smaller rates tend to be employed due to network congestion.
- Since the delivery ratio for lower CSThreshold is low at high network loads, there is actually very small number of packets being employed for calculation of transmission delay. So we could get a smaller average delay for lower CSThreshold. If we set the transmission delay of all the lost packets to be a very large value, then the delay is actually very high.



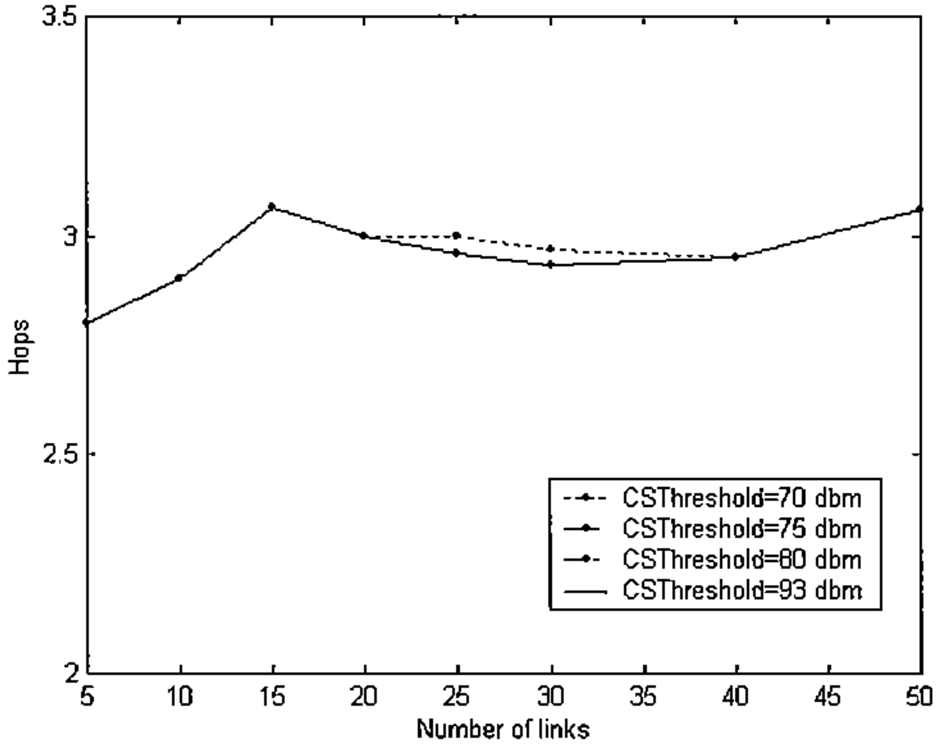
Variation of transmission delay for different loads

- Variation of transmission delay (or Jitter) is the standard deviation of transmission delay.
- Variation of transmission delay is measured by calculating the square root of the average of the square of the difference between a delay and the average transmission delay.
- It is shown that the transmission delay varies more significantly for higher loads. This is because that, at high load, different data sending rates are employed, depending on if there is network congestion.



Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- Throughput at low CSThreshold such as 70 dbm is unsmooth because the corresponding delivery ratio is low and there are actually very small number of packets being successfully received

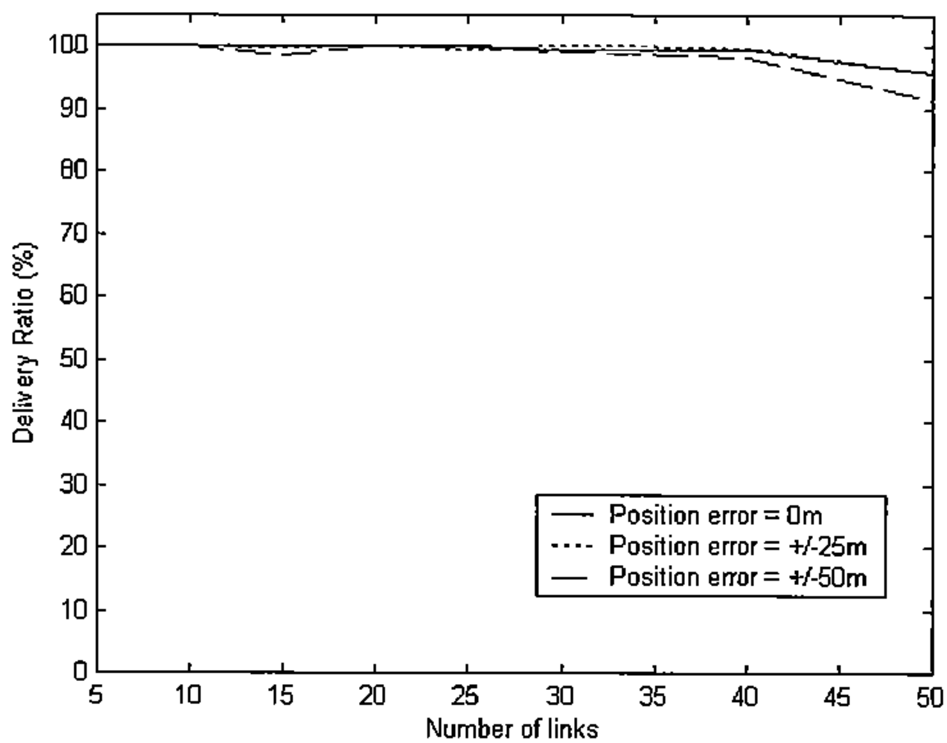


Hop count for different loads

- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them.
- It is shown that the hop count does not change much for different network loads and CSThresholds.

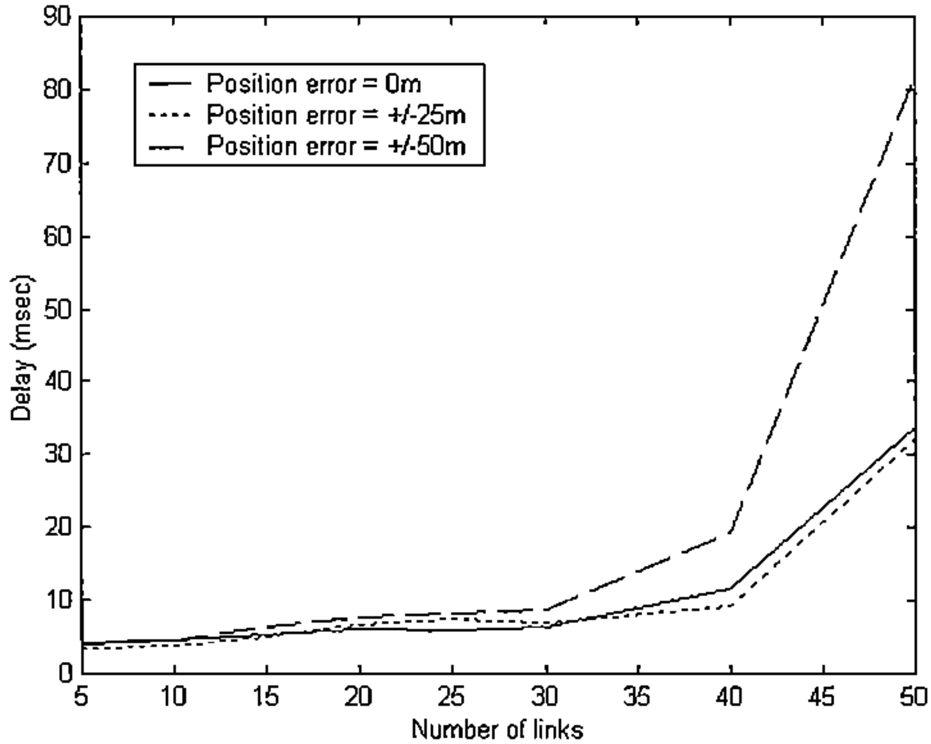
2.6 Position inaccuracy

In order to testify the robustness of GPR to the position error introduced by GPS, we compare the system performance using different position errors as 0m, ±25m and ±50m. The moving speed of mobile nodes is 3 m/s. It is shown that the system works pretty well for position error of ±25m, but ±50m position error does deteriorate the performance significantly, especially for the time delay.



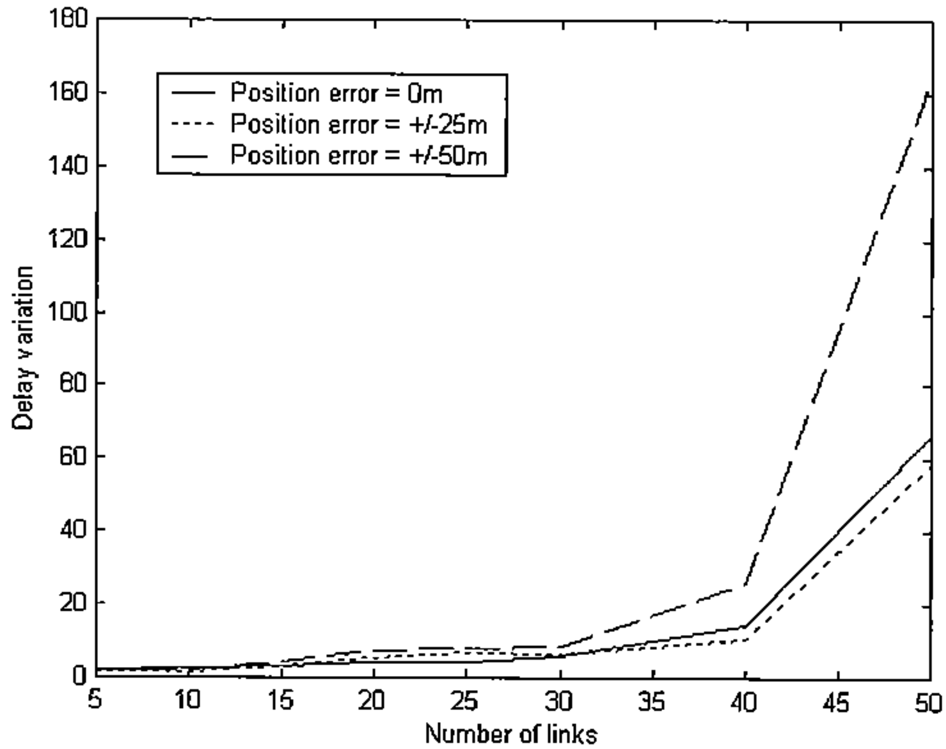
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- Almost all sent packets can be received by the receiver. The delivery ratio only drops a little bit for higher number of links. Bigger position error introduces a little bit more packet loss.



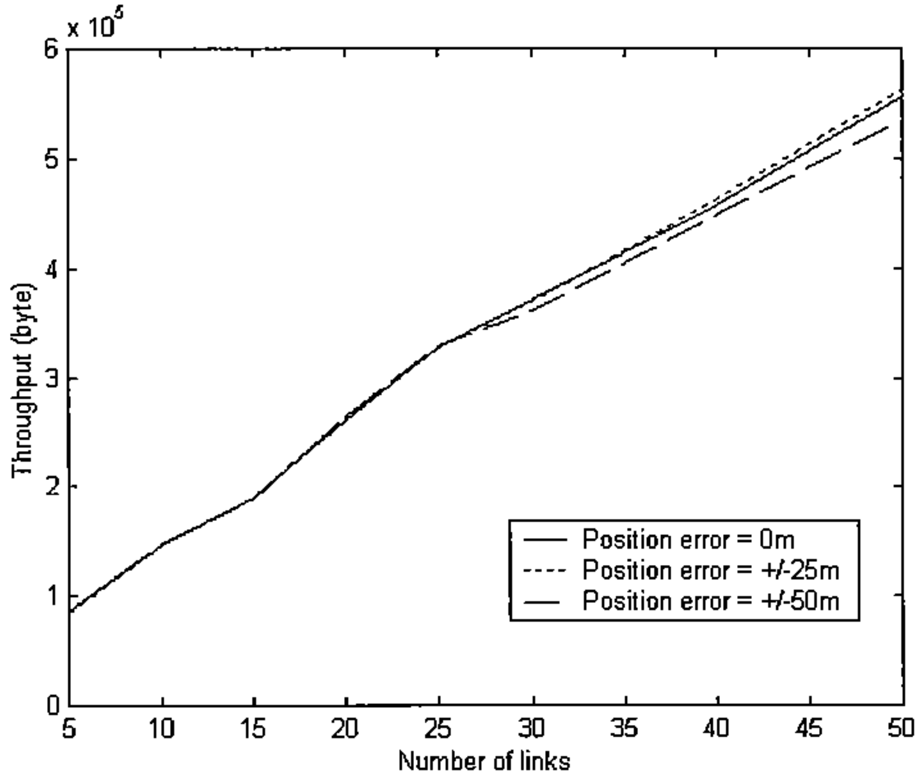
Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging.
- As expected, the transmission delay significantly increases when the inaccuracy of position information increases.



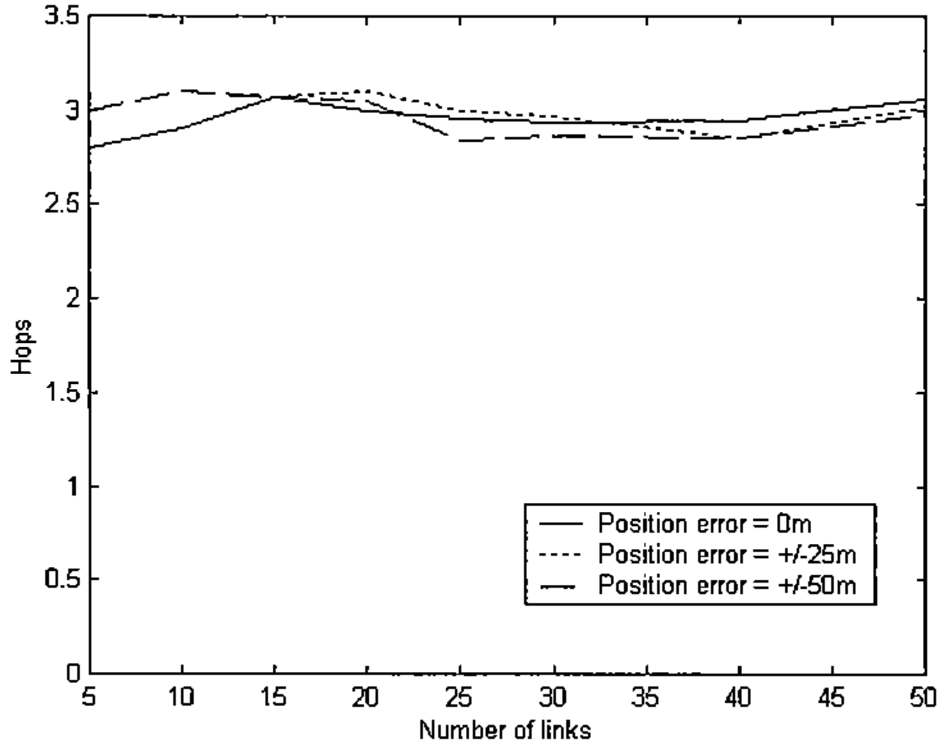
Variation of transmission delay for different loads

- Variation of transmission delay (or Jitter) is the standard deviation of transmission delay.
- Variation of transmission delay is measured by calculating the square root of the average of the square of the difference between a delay and the average transmission delay.
- It is shown that the transmission delay varies more significantly when the position error increases.



Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- Throughput changes very little for different position errors.

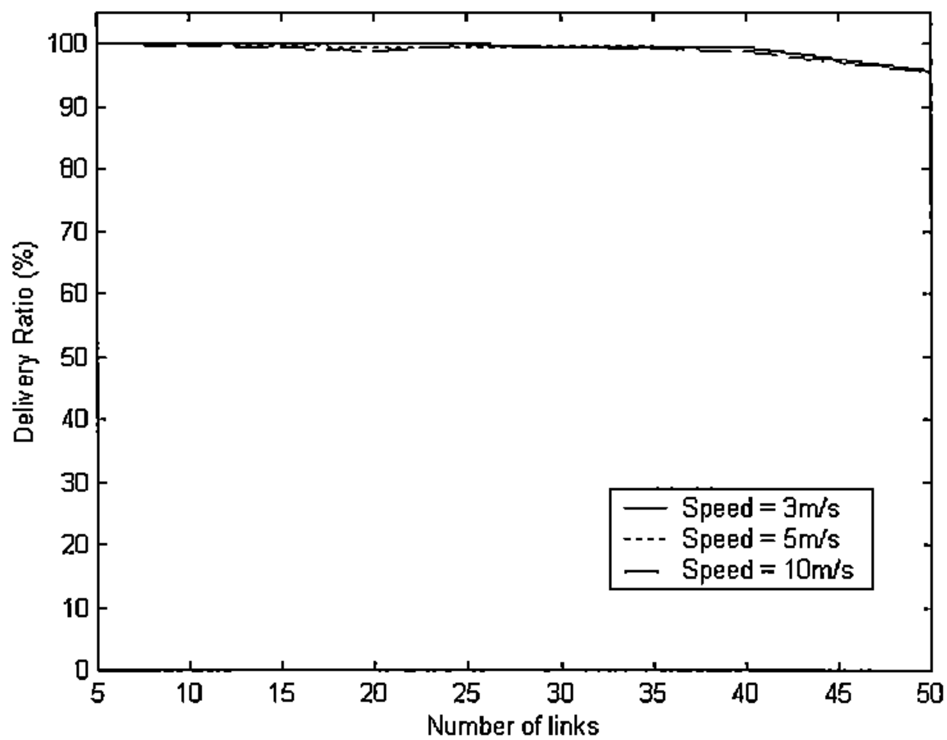


Hop count for different loads

- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them.
- It is shown that the hop count does not change much for different network loads and position errors.

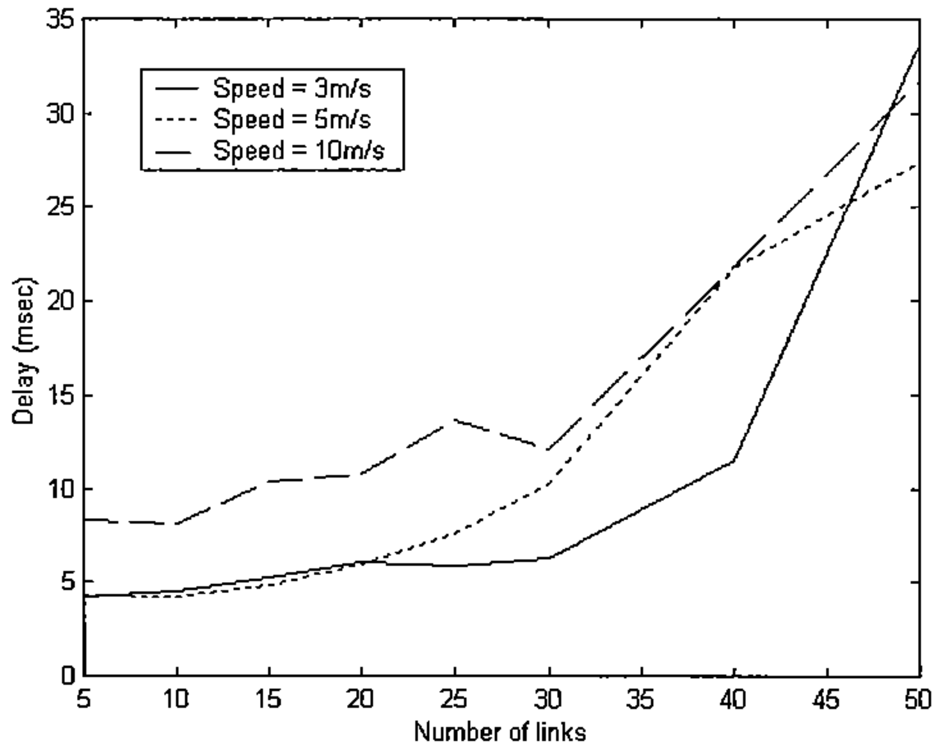
2.7 Mobility

The system performance is tested when mobile nodes are moving at different maximum speeds of 3m/s, 5m/s, and 10m/s. The rate of position updates for these simulations is not fixed because it is updated whenever it is necessary in the routing protocol. It is shown from the figures that the system works well for all these speeds.



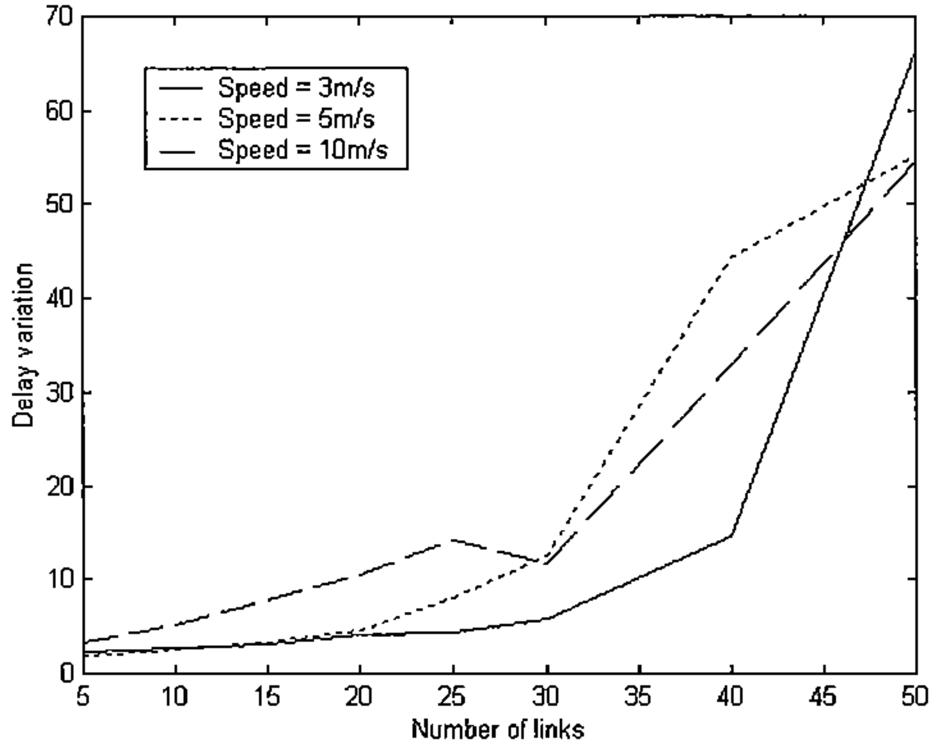
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- Almost all sent packets can be received by the receiver. The delivery ratio only drops a little bit for higher number of links.



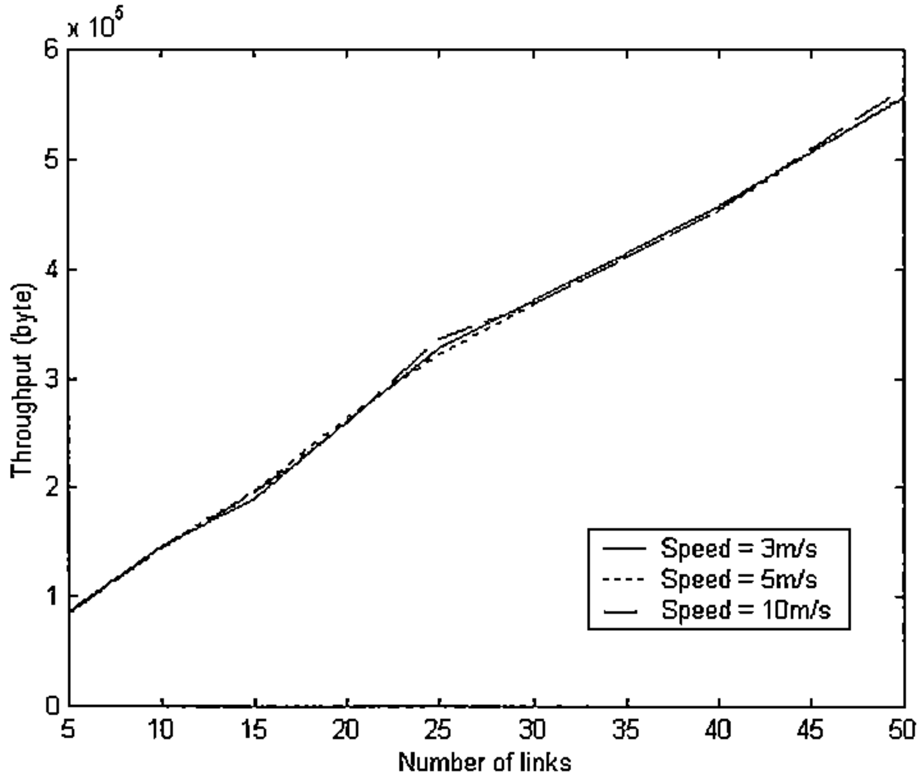
Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging them.
- The transmission delay increases when the number of connections increases. The higher moving speed introduces a little longer delay.



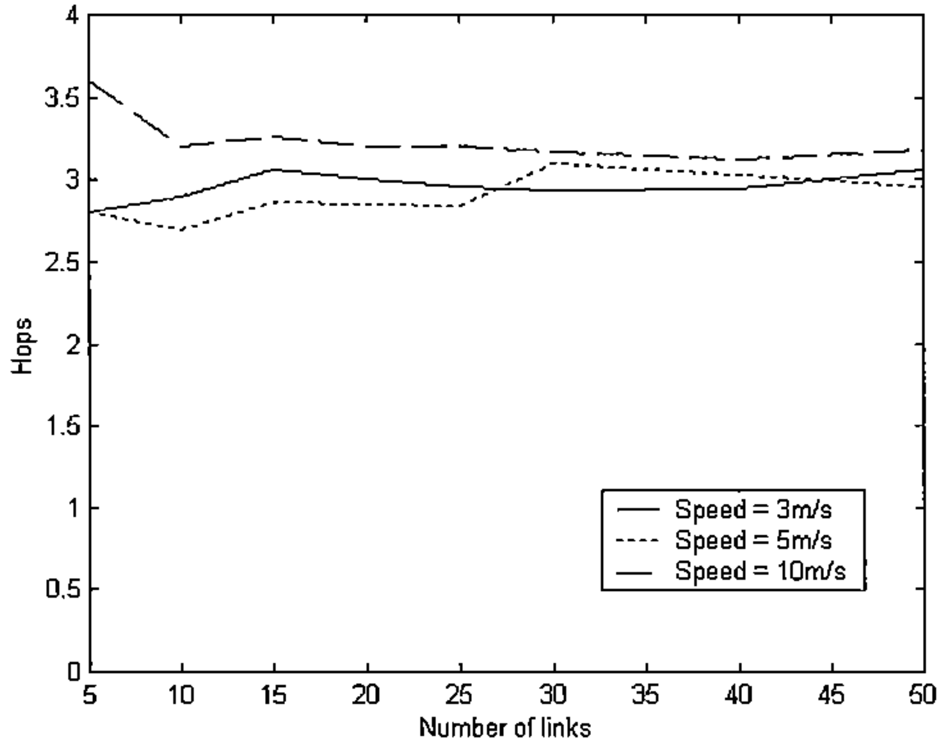
Variation of transmission delay for different loads

- Variation of transmission delay (or Jitter) is the standard deviation of transmission delay.
- Variation of transmission delay is measured by calculating the square root of the average of the square of the difference between a delay and the average transmission delay.
- It is shown that the transmission delay varies more significantly for higher loads. The higher moving speed also introduces a little more delay variation.



Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- Different moving speed does not introduce significant change of throughput.



Hop count for different loads

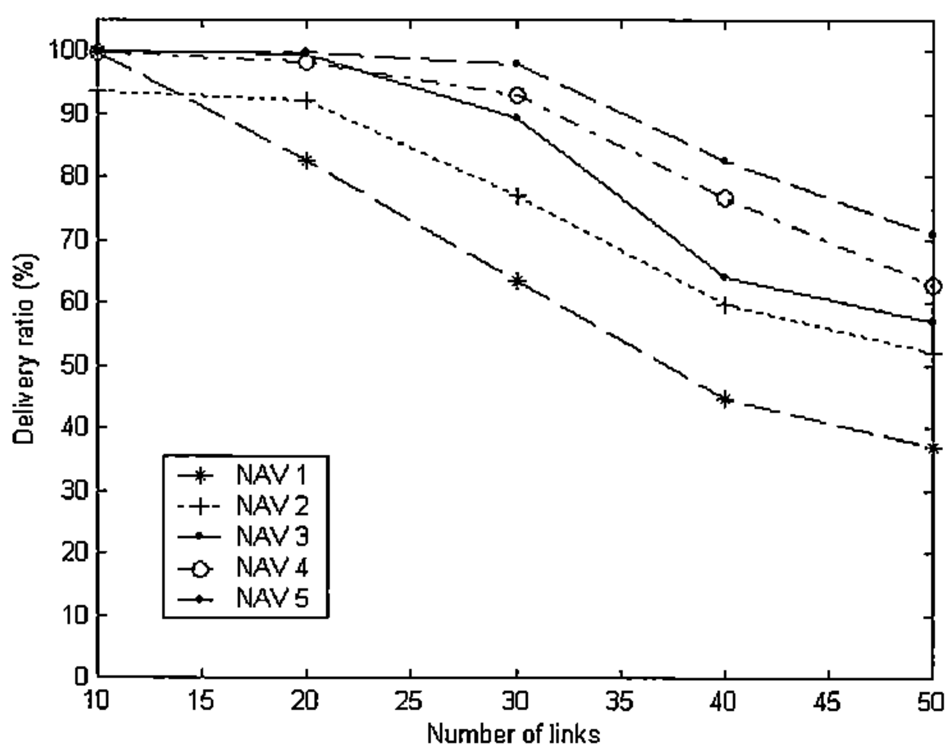
- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them.
- The hop count does not change much for different network loads and moving speeds.

2.8 NAV update methods

Due to the adaptive rate control, the NAV duration time in IEEE 802.11 should be estimated in RTS and probably updated later in CTS. Five different methods are compared

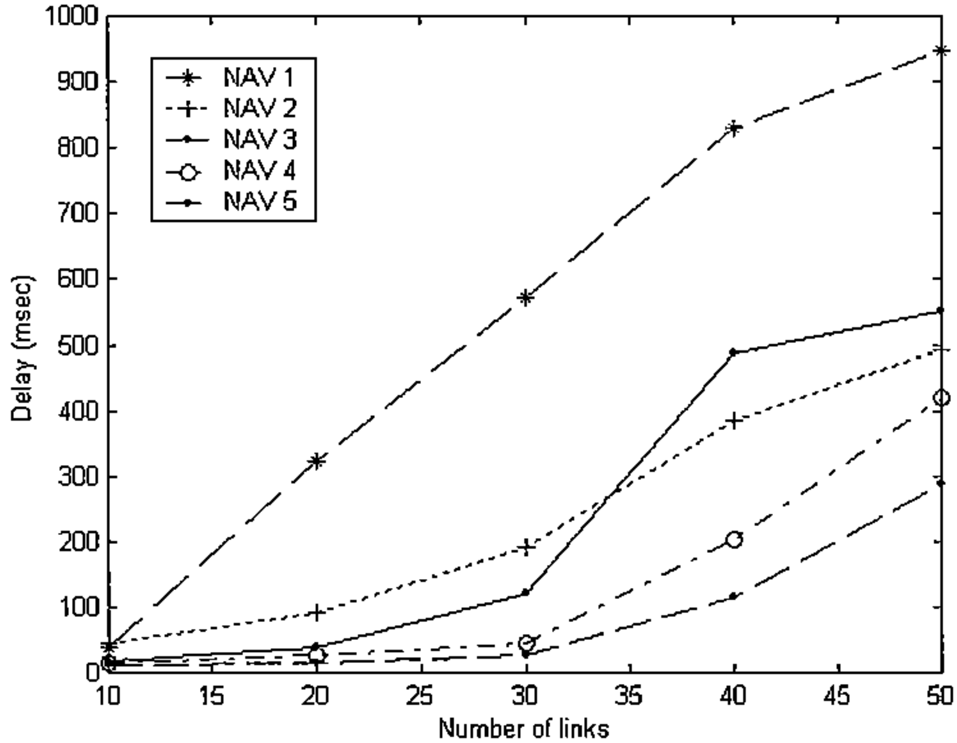
- NAV 1: Use 54 Mbps to estimate NAV in RTS, no update in CTS
- NAV 2: Use 1 Mbps to estimate NAV in RTS, no update in CTS
- NAV 3: Use 54 Mbps to estimate NAV in RTS, update NAV in CTS based on the new rate
- NAV 4: Use the last sending rate to estimate NAV in RTS, no update in CTS
- NAV 5: Use appropriate rate to get correct NAV in RTS so that it is not necessary to update it in CTS, the best but not realistic

It is proved from the following simulation results that the performance for these methods is ordered as NAV 1 < NAV 2 < NAV 3 < NAV 4 < NAV 5.



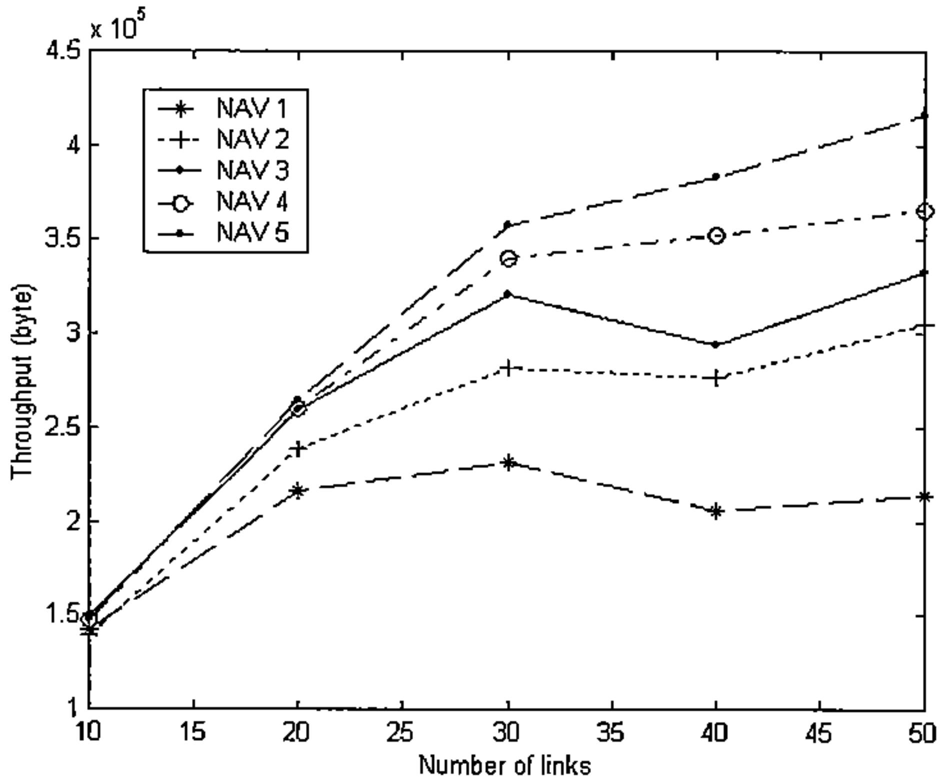
Delivery ratio for different loads

- Delivery ratio is the ratio between the number of successfully received packets and the number of sent packets.
- Delivery ratio is measured by counting all received packets and sent packets and then calculating the ratio.
- The delivery ratio increases in the order from NAV 1 to NAV 5.



Transmission delay for different loads

- Transmission delay is the difference between the data receiving time and sending time.
- Transmission delay is measured by calculating the difference between receiving time and sending time for every packet and then averaging them.
- The transmission delay decreases in the order from NAV 1 to NAV 5.

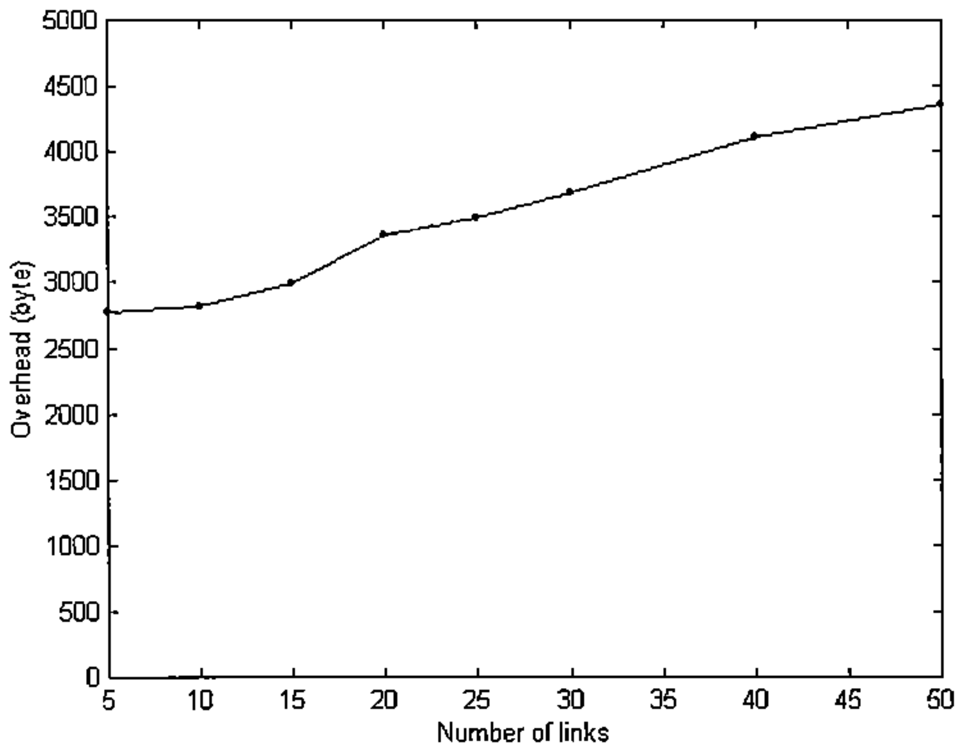


Throughput for different loads

- Throughput is the total number of data bytes successfully received.
- Throughput is measured by counting the total number of data bytes received by receivers in all connections.
- Throughput increases in the order from NAV 1 to NAV 5.

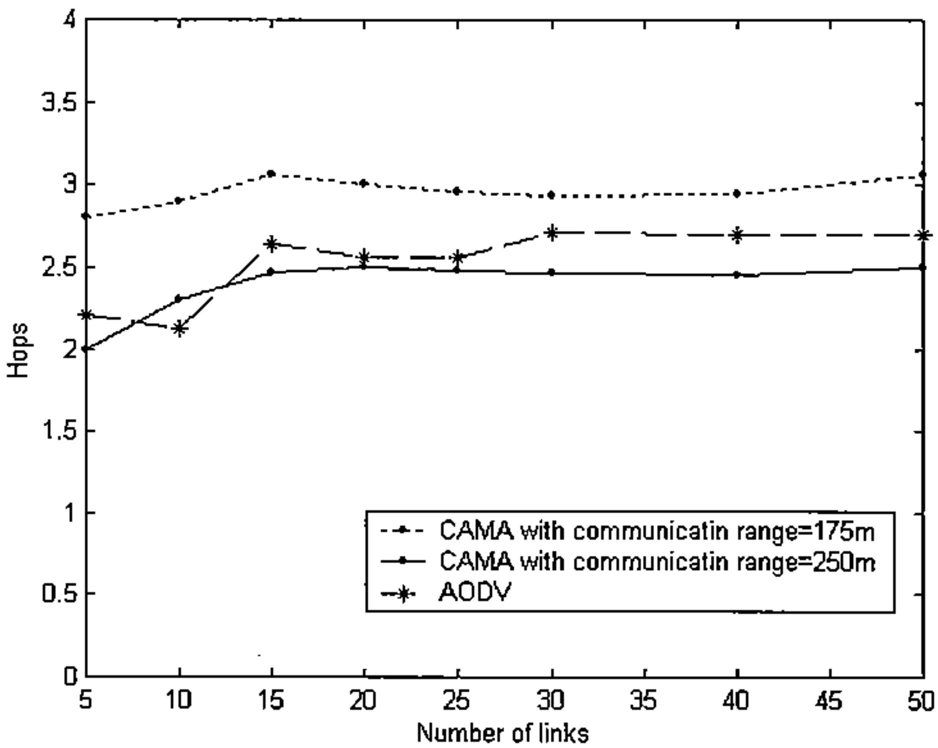
2.9 Others

The overhead of CAMA is illustrated in the figure below. The overhead does include the data transmitted for position information exchange between CAMA agent and mobile nodes. The number of hops is compared between CAMA and AODV. For CAMA, two different communication ranges of 175m and 250m are used. The communication range means the maximum propagation distance that can be reached by a mobile node.



Overhead of CAMA

- Overhead is the number of control data bytes.
- Overhead is measured by counting the size of all control packets, including those among mobile nodes and between mobile nodes and CAMA agents.
- Overhead increases a little when network load increases.



Hop count comparison

- Hop count is the average number of hops traversed by a packet from sender to receiver.
- Hop count is measured by counting the number hops for every packet and then averaging them.
- It is shown that the CAMA using communication range of 250m mostly needs the least number of hops. The CAMA using communication range of 175m needs a little more hops than AODV, but provides much better system performance than AODV because 175m is approximately the optimal communication range in this case, as stated in a paper we just submitted ([2]). It is also shown that the hop count does not change much for different network loads.

Conclusion

In this report, extensive simulation studies on CAMA are presented. Following conclusions are made based on the simulation results.

- CAMA enables centralized control of ad hoc networks by cellular networks, which increases the routing efficiency, reliability and security (The security issues are discussed in [3]). But also due to the central control mechanism, CAMA is not a pure ad hoc network so that the routing and security problems in CAMA are not so challenging as in traditional ad hoc networks. CAMA is also far different from the previous geographic-based ad hoc networks where the position information is distributed without assistance of central controllers.
- In practice, since the position information of all mobile nodes is supposed to be available to CAMA agents, the proposed routing protocol, GPR, can be replaced by some simpler table-driven protocols.
- It is proven by simulations that CAMA performs much better than the traditional ad hoc networks, but it is still not that satisfactory for high load networks. For example, the delivery ratio is only about 50% for eight simultaneous video transmissions. This might be due to the fact that we are using IEEE 802.11 as the lower layer wireless access network for CAMA. But the current implementation of IEEE 802.11 in ns-2 is not that up-to-date and complete because ns-2 is focusing on the network layer ad hoc routing protocols. So a more accurate simulator of current IEEE 802.11 standards is needed for future study.
- CAMA does not provide details of signalings between mobile nodes and CAMA agents. But there will be significant work to specify and implement them in the real system in the environment of a particular cellular network.
- Other issues that are in need of further study are: power control of mobile nodes; MAC layer QoS of IEEE 802.11; Scalability of centralized routing protocol in CAMA agents; and so on.

Reference

[1] B. Bhargava, X. Wu, Y. Lu and W. Wang, "Integrating Heterogeneous Wireless Technologies: A Cellular Aided Mobile Ad hoc Network (CAMA)," accepted for publication in ACM Special Issues of the Journal on Special Topics in Mobile Networking and Applicaitons (MONET).

[2] X. Wu, G. Ding, B. Bhargava and S. Lei, "Improving Throughput by Link Distance Control in a Multi-rate Ad Hoc Network," submitted, 2003.

[3] X. Wu and B. Bhargava, "CAMA Security Report," 2003.