

Correlation between Hop Count and Packet Transfer Time

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Abstract— This paper proves the correlation between hop count and packet transfer time. We first discuss how to calculate hop count and packet transfer time from TCP/IP packets. Hop count can be calculated from the TTL value of an IP packet. However, it is not a trivial calculation because many Operating Systems do not follow the Internet standard and have many different default TTL values. Packet round trip time can be measured by analyzing TCP connections. Using those methods, we calculate the correlation coefficients of hop count and packet roundtrip time. It is proved that these two values are strongly correlated. Then we introduce two new values, average hop count and marginal packet transfer time, which can be used to evaluate network characteristics.

I. INTRODUCTION

This paper gives proof that hop count is strongly correlated with packet transfer time. Hop count is the number of routers through which a packet passed to the destination host. Also this paper introduces two new values related to hop count, which can be used to evaluate network characteristics of measuring point: average hop count, and marginal packet transfer time.

It is not simple to compute the hop count of the host in the distance. We introduce a method to calculate hop count from the TTL (Time To Live) value of the IP (Internet Protocol) header in section II. We first survey the initial TTL values and find that there are six initial values. Because some of them are too close to distinguish, we decide to use four of them: 32, 64, 128, and 255. The hop count of the IP packet can be calculated by subtracting the TTL value from one of those initial values.

Section III. discusses how to compute roundtrip time between two hosts by observing TCP connections. Then, in section V., we prove that there is a strong correlation between hop count and packet roundtrip time. We also explain that hop count and data transfer speed are somehow related, and because of that hop count can be used to evaluate network performance of measuring point.

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Lastly, section VI., describes how to compute marginal packet transfer time, and explains that it can be used to represent two network characteristics: the congestion of the entire network, and the performance of neighboring networks.

II. COMPUTING HOP COUNT

A. TTL (Time To Live)

In order to compare hop count and packet transfer time, we have to establish a method to compute hop count and transfer time. In this paper, we use the TTL value in the IP header to compute hop count. Fig.1 illustrates the structure of the IP header. By the original definition [1], TTL indicates the time in which a packet can exist on the network. For example, a packet with TTL 64 is allowed to be on the network for 64 seconds. TTL is defined to prevent a packet from circling on the network forever in case a routing loop is created by accident or by improper configuration. Because it is difficult to calculate the exact time and because TTL does not have to be accurate as long as it eventually becomes zero, TTL is decremented by one when passing through one router. Therefore, it is possible to calculate hop count from the value of TTL.

Version 4 bits	Hlen 4 bits	Type of Service 8 bits	Packet length 16 bits	
Identification 16 bits		Flags 3 bits	Fragment Offset 13 bits	
TTL 8 bits	Protocol 8 bits	Checksum 16 bits		
Source IP Address 32 bits				
Destination IP Address 32 bits				

Fig. 1. IP Header

B. How to measure hop count

There are two methods to measure the hop count from a host. One is an active measurement, and the other is a passive measurement.

The first method is to use ICMP ECHO packets. In most cases this gives an accurate hop count. However, applying this method to thousands hosts is not realistic

because sending lots of ICMP packets is not recommended as a measuring method.

The second method is simply to subtract the TTL of a received IP packet from its initial value. This can be done without sending any sample packets, and therefore is ideal for measuring the hop counts of many hosts.

$$(\text{Hopcount}) = (\text{initial TTL}) - (\text{TTL})$$

We choose this method to calculate a hop count from the TTL. However, in order to use this method, the initial TTL values should be known in advance.

C. Problem with initial values of TTL

According to RFC 1700 [2], the recommended initial TTL value is 64. However, the rule is often ignored on the real Internet. Figs. 2 and 3 show the distributions of TTL values at two measuring points, Waseda University and APAN Tokyo XP (eXchange Point).

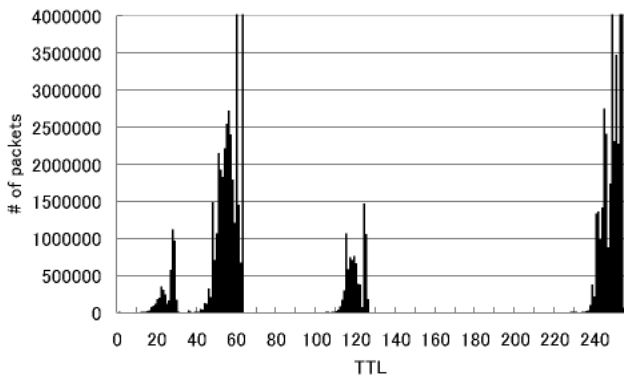


Fig. 2. TTL distribution (Waseda, Dec.98)

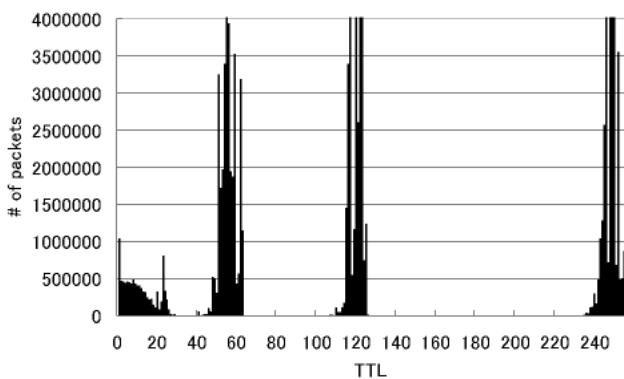


Fig. 3. TTL distribution (APAN Tokyo XP, Aug.99)

In both graphs, similar distribution patterns can be observed. There are roughly four areas in which most packets are gathered: below 30, around 60, around 120, and

around 250. This indicates that there are at least four initial TTL values, which are close to those four values.

Swiss Academic & Research Network (SWITCH) has researched initial TTL values of different OS (Operating Systems) [3]. Table I shows their results with some updates we added.

As a result, we are aware that there are six initial TTL values: 30, 32, 60, 64, 128, and 255.

TABLE I
Initial TTL Value of various OS

OS	Initial TTL of TCP	Initial TTL of UDP
AIX	60	30
DEC Pathworks V5	30	30
FreeBSD	64	64
BSD/OS	64	64
HP/UX 9.0x	30	30
HP/UX 10.01	64	64
Irix 5.3	64	64
Linux	64	64
MacOS/MacTCP 2.0.x	60	60
OS/2 TCP/IP 3.0	64	64
OSF/1 V3.2A	60	30
Solaris 2.x	255	255
SunOS 4.1.3/4.1.4	60	60
Ultrix V4.1/V4.2A	60	30
VMS/Multinet	64	64
VMS/TCPware	60	64
VMS/Wollongong 1.1.1.1	128	30
VMS/UCX (latest rel.)	128	128
MS WfW	32	32
MS Windows 95	32	32
MS Windows NT 3.x	32	32
MS Windows NT 4.0	128	128

From Figs. 2 and 3, it is known by intuition that the packets whose initial TTL is 255 and those whose initial TTL is 128 can be distinguished from others easily. However, it is more difficult to assess whether the initial value of the packets whose TTL are less than 60 is 60 or 64. The same problem occurs to the packets with TTL that are less than 30. In this paper, we decide to ignore the initial value 30 and 60 and to use only 32, 64, 128 and 255. The reasons are the following: today's popular OS, Microsoft Windows 95/98, Linux, and FreeBSD, are using 32 and 64 as initial values. Also it is common to use 2^n value in the computer world.

From the prospects described above, we create the following formula to convert TTL to hop count.

$$\text{Hop} = \begin{cases} 32 - \text{TTL} & (\text{TTL} \leq 32) \\ 64 - \text{TTL} & (\text{TTL} \leq 64) \\ 128 - \text{TTL} & (\text{TTL} \leq 128) \\ 255 - \text{TTL} & (\text{TTL} \leq 255) \end{cases}$$

III. COMPUTING PACKET TRANSFER TIME

It is also necessary to establish a method to measure packet transfer time between the measuring point and the target host. Since there is no time stamp field in IP

header, it is impossible to calculate packet transfer time by simply analyzing the packet header.

The most accurate method to measure transfer time is to use a measurement software which sends and receives sample packets. However, this method can be used only for specific hosts on which we can execute the measurement program, and therefore is not suitable to collect packet transfer times of a number of hosts. Another way is to send an ICMP ECHO packet and count the time till its reply returns from the host. This method seems to work well, but actually it doesn't, because ICMP packets are treated differently from normal IP packets, and therefore the roundtrip time of ICMP packets can be different from that of IP packets. Moreover, sending many ICMP packets is not good in terms of Internet manners.

In response to these problems, we adopt a passive method, which is to capture and analyze TCP (Transfer Control Protocol) handshaking packets. This method makes it possible to collect roundtrip times of a large number of hosts without sending any unnecessary packets.

When making a TCP connection, one host sends a TCP packet with a SYN (Synchronize) flag bit on, in order to request connection establishment. Immediately upon receiving the packet, another host sends back a TCP packet with ACK (Acknowledgement) and SYN flag bits on, to accept the request and to establish the connection in the reverse direction. The negotiation ends when the first host sends back an ACK packet (Fig. 4). Logically TCP tries to make both upstream and downstream connection separately to achieve full duplex transmission. This process is called "3-way handshake" [4].

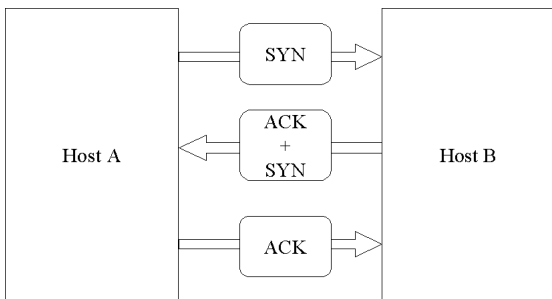


Fig. 4. 3-way handshake

Because the ACK packet is sent back immediately after the SYN packet is received, the ACK packet can be regarded as an "ECHO packet" of the SYN packet. Therefore, the difference between the time when the SYN packet is sent and the time when the ACK packet returns can be used as the approximation of the roundtrip time between two hosts.

IV. MEASUREMENT CONDITION

We capture and analyze IP packets at two measuring points: the gateway of Waseda University and the APAN (Asia Pacific Advanced Network) Tokyo XP (eXchange Point).

Waseda University is connected to the IMnet (Inter-Ministry Research Information network) with 100Mbps FDDI. We capture the traffic that flows on the 100Mbps Ethernet segment near the gateway with libpcap, LBNL's (Lawrence Berkeley National Laboratory) packet capture library [5]. The traffic from inside the university to outside is discarded because it does not match our interest. The measurements were conducted on Dec.15, 1998 and Nov.16, 1999, for 24 hours each.

APAN is a non-profit international consortium established on 3 June 1997, which is intended to be a high-performance network for research and development on advanced applications and services [6]. APAN Tokyo XP is directly connected to vBNS (NSF/MCI's very high performance Backbone Network Service) with OC-3. Using OC3mon [7], the measuring tool designed especially for OC-3, and our original analyzing tool, we analyze the SYN packets which are sent from APAN to vBNS, and the corresponding ACK packets from vBNS. The measurement is done on Aug.31, 1999, also for 24 hours.

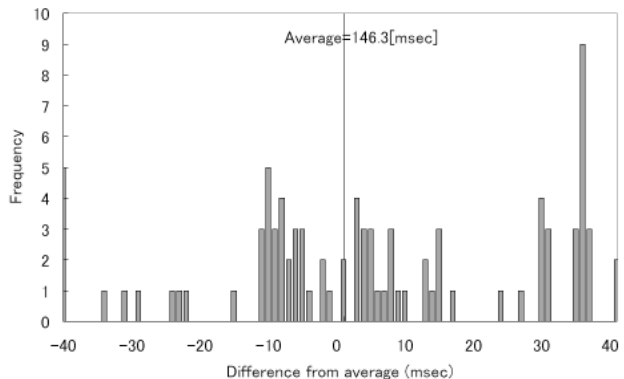


Fig. 5. Distribution of packet roundtrip time (www.yahoo.com)

The hop count for each host can be obtained from the TTL field and source address of each packet. Also the roundtrip time to the host is calculated by counting the time from when a TCP SYN packet is captured till the corresponding ACK packet comes back. Roundtrip time is not constant because it depends on the network congestion. (See Fig.5 about how the roundtrip times from Waseda to www.yahoo.com are scattered in 24 hours.) The best way to avoid the influence of congestion is to collect the roundtrip times in very short period of time. However, if the measurement time is very short, sufficient amount of data can not be collected. This time we avoid this problem by averaging the roundtrip times. If the mea-

suring time is long enough, this alleviates the difference of the congestions.

V. CORRELATION BETWEEN HOP COUNT AND TRANSFER TIME

A. Computing correlation coefficients

At both measuring points, a strong correlation of hop count and roundtrip time can be observed.

Figs. 6 and 7 show the average roundtrip time for each hop count at Waseda University. Fig. 8 is of APAN Tokyo XP. In every graph, it is clear that roundtrip time increases as the hop count grows. It can be said only by observing those graphs that hop count and transfer time are correlated. To make this more obvious, we compute correlation coefficients of each graph.

Correlation coefficient is a number mainly used for statistics. Its range is from -1.0 to 1.0. Positive correlation coefficient means that two values are positively correlated, that is, when one value increase the other also increases. If two values are inversely proportional, correlation coefficient becomes negative. Its absolute value shows how strong the correlation is. Correlation coefficient of two sequences X_i and Y_i can be calculated by following formula.

$$r = \frac{\sum_{i=1}^n (X_i - \bar{X})(Y_i - \bar{Y})}{\sqrt{(\sum_{i=1}^n (X_i - \bar{X})^2)(\sum_{i=1}^n (Y_i - \bar{Y})^2)}}$$

We calculate the correlation coefficients of each graph. We only take the hosts whose hop count is in the effective range, i.e. when the percentage of the host number is more than 0.5 %, into account. The results are 0.9549, 0.9541 and 0.9127 respectively (Table II). These correlation coefficients are high enough to say that there is a strong correlation between hop count and packet transfer time.

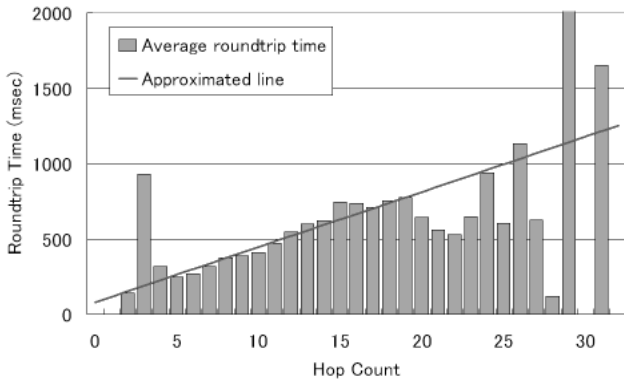


Fig. 6. Packet Roundtrip Time (Waseda, Dec.98)

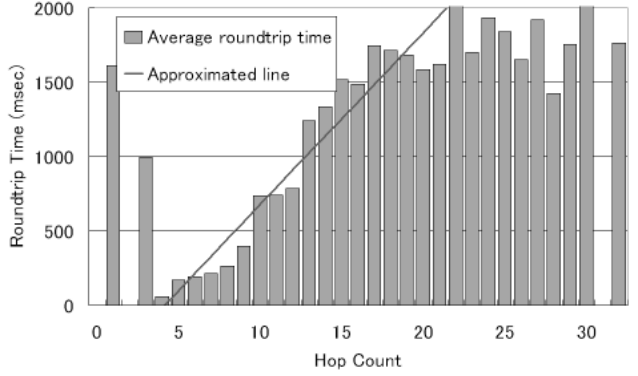


Fig. 7. Packet Roundtrip Time (Waseda, Nov.99)

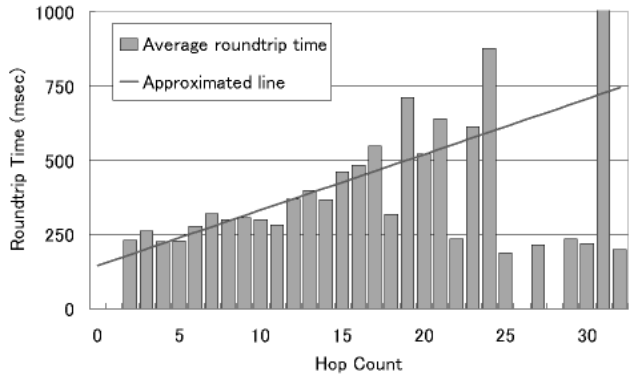


Fig. 8. Packet Roundtrip Time (APAN, Aug.99)

TABLE II
Correlation coefficients of hop count and packet roundtrip time

Measuring point	Effective hop area	Correlation coefficient
Waseda Univ. (15/12/98)	5 - 20	0.9559
Waseda Univ. (16/11/99)	3 - 21	0.9450
APAN Tokyo (31/08/99)	4 - 16	0.9127

B. Meaning of hop count

In the last section, a strong correlation between hop count and transfer time is proved. This means that hop count is one of the big factors which determines network communication speed.

In connection-oriented protocol, such as TCP, ACK packets are sent back to the sender in order to make sure that the receiver got the data packets correctly. Usually the sender does not send all data packet at once, but sends several packets and then waits for ACK, because that makes the overhead of retransmission smaller. Therefore, no matter how wide the connection is, if packet transfer takes long, the ACK waiting period becomes big and the throughput gets limited.

Hop count and packet transfer time are highly correlated according to our result. Therefore, hop count can determine data transfer speed. From this theory, we suggest that hop count can be used to evaluate network performance.

C. Evaluation of network with average hop count

Using the method described in section II., it is not difficult to calculate the average hop count of all the hosts with which the measuring point is communicating. The average hop count at a certain point indicates how far the point is from the hosts to which most traffic is concentrated. The small average means that those hosts can be reached in little time. Thus, it can be one of the values which indicate the network performance of a measuring point.

Figs. 9, 10 and 11 show the distributions of hosts for each hop count, and table III shows the average hop counts of two measuring points. Because the average hop count of APAN Tokyo XP is smaller than that of Waseda, it can be declared that APAN Tokyo XP exists at the better location than Waseda in terms of communication speed.

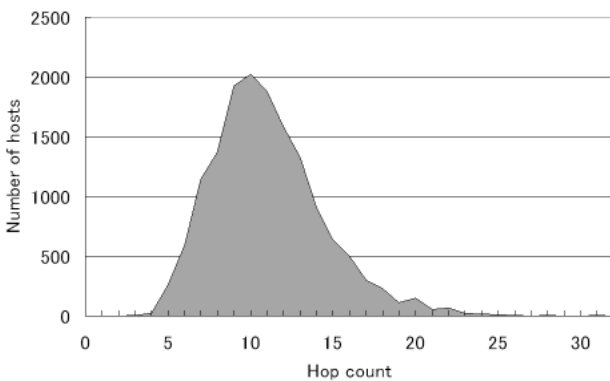


Fig. 9. Host number distribution (Waseda, Dec.98)

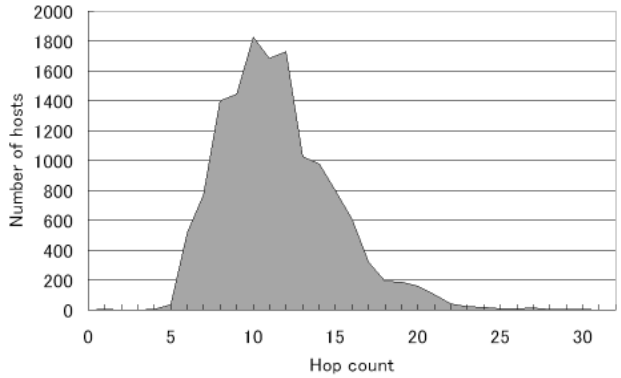


Fig. 10. Host number distribution (Waseda, Nov.99)

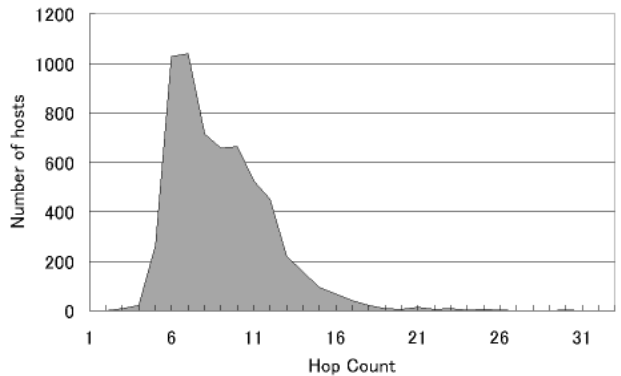


Fig. 11. Host number distribution (APAN, Aug.99)

TABLE III
Average hop count

Measuring point	Average hop count
Waseda Univ. (15/12/98)	11.5127
Waseda Univ. (16/11/99)	11.0505
APAN Tokyo (31/08/99)	7.9975

VI. MARGINAL PACKET TRANSFER TIME

A. Computing marginal transfer time

In this paper we assume that packet transfer time increases linearly. It is not difficult to calculate the approximated line of transfer time.

Solid lines in Figs. 6 – 8 are the approximated lines of roundtrip time, computed by least squares estimates. The gradients of the approximated line at two measuring points are shown in Table IV. These gradients are equal to the average increase of packet transfer time per hop. We call these values “marginal transfer time.”

Interestingly, the marginal transfer times are different even at the same measuring point. The next section argues the meanings of marginal transfer time, and how it is related to network characteristics.

TABLE IV
Slope of approximated line

Measuring point	Slope [msec]
Waseda Univ. (15/12/98)	36.6647
Waseda Univ. (16/11/99)	115.6569
APAN Tokyo (31/08/99)	18.7637

B. Evaluation of network performance with marginal transfer time

The marginal transfer time means how much packet transfer time increases when hop count increments by one. Two factors can be considered as affecting this value: the congestion of the entire network at the measuring time, and the network topology of the measuring points.

If the entire network is more congested, the marginal transfer time becomes bigger. It should be noted here that marginal transfer time is not affected by the congestion of the measuring point. It changes the interception of the approximated line of packet transfer time, but not the gradient.

The network topology of the measuring point is another factor determining marginal transfer time. If the measuring point is connected to the networks composed with wide bandwidth links and high-performance routers, the degree of change of transfer time per one hop must be small. That is, marginal transfer time is also affected by the performance of the networks connected to the measuring point.

Because marginal transfer time is determined by two factors, it is difficult to evaluate just one of those factors by using marginal transfer time. A good way to distinguish the mixed factors is to compare the results of the situations in which one of the factors is the same.

The difference of marginal transfer time at the same measuring point includes little influence of network topology, and therefore indicates the congestion of the entire network. For example, the measurement results at

Waseda show that the Internet was more congested on November, 1999 than on December, 1998.

The difference of marginal transfer time at different measuring points in the same time is hardly influenced by the congestion of the entire network, and thus indicates the difference between the performance of neighboring networks. That means, APAN Tokyo XP is located in better position than Waseda in terms of network performance.

VII. CONCLUSION

Through this paper, we have focused on hop count. First we have established the way to calculate hop count from TTL. Then we have proved the correlation of hop count and packet transfer time, and discussed the availability of hop count as reference value of network performance, because throughput and packet transfer time are closely related. In order to realize network evaluation by hop count, we have introduced two values: average hop count and marginal transfer time. Average hop count indicates the average access time to the major hosts. Marginal transfer time represents two characteristics, the congestion of entire network and the performance of neighboring networks.

Until recently bandwidth was believed to be the biggest factor to determine the quality of the network. Most people would prefer the ISPs with wider connection to the backbone networks. Certainly bandwidth has great effect to data transfer speed, which is normally equal to the network's quality. However, that is not the only factor. Some start caring about the topology of the Internet when choosing ISP, because they notice that the closer to the backbone the faster, and therefore the better.

It is hard to tell how close a point of the network is to the backbone from the network topology. Moreover, it is almost impossible to obtain the entire topology of the Internet. However, the topology-specific characteristics of a measuring point can be clarified by the average hop count and marginal transfer time. These characteristics which can be seen by simply observing bandwidth or utilization of the circuit. Thus these values can be good measures to evaluate network performance from a new point of view.

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