CSE3214 Computer Network Protocols and Applications

Chapter 1

Examples and Homework Problems

Example 1 (review question 18)

(1) How long does it take a packet of length 1000 bytes to propagate over a link of distance 2500km, propagation speed 2.5×10^8 m/s, and transmission rate 2 Mbps?

(2) More generally, how long does it take a packet of length L to propagate over a link of distance d, propagation speed s, and transmission rate R bps?

(3) Dose this delay depend on packet length?

(4) Does this delay depend on transmission rate?

Solution to Example 1

(1) How long does it take a packet of length 1000 bytes to propagate over a link of distance 2500km, propagation speed 2.5×10^8 m/s, and transmission rate 2 Mbps?

Ans: (2500*10^3)/(2.5*10^8)=0.01s = 10ms

(2) More generally, how long does it take a packet of length *L* to propagate over a link of distance *d*, propagation speed *s*, and transmission rate *R* bps?

Ans: d/s

(3) Dose this delay depend on packet length? No

(4) Does this delay depend on transmission rate? No

Example 2 (R19)

Suppose Host A wants to send a large file to Host B. The path from Host A to Host B has three links, of rate $R1 = 500$kbps, $R2 = 2$Mbps, and $R3 = 1$Mbps.

a. Assuming no other traffic in the network, what is the throughput for the file transfer?
b. Suppose the file is 4 million bytes. Dividing the file size by the throughput, roughly how long will it take to transfer the file to Host B?
c. Repeat (a) and (b), but now with $R2$ reduce to 100kpbs.

Solution to Example 2

Suppose Host A wants to send a large file to Host B. The path from Host A to Host B has three links, of rate $R1$=500kbps, $R2$=2Mbps, and $R3$=1Mbps.

a. Assuming no other traffic in the network, what is the throughput for the file transfer?

Ans: 500kpbs.

a. Suppose the file is 4 million bytes. Dividing the file size by the throughput, roughly how long will it take to transfer the file to Host B?

Ans: (4*10^6)*8/(500*10^3)= 64 seconds

a. Repeat (a) and (b), but now with $R2$ reduce to 100kpbs.

Ans: 100kbps, 4*10^6*8/100*10^3= 320 seconds

Example 3 (R21)

Visit the Queuing and Loss applet at the companion web site. What is the maximum emission rate and the minimum transmission rate? With those rates, what is the traffic intensity?

Run the applet with these rates and determine how long it takes for packet loss to occur. Then repeat the experiment a second time and determine again how long it takes for packet loss to occur. Are the values different?

Packet Loss 1st Run

Queuing and Loss Applet

Queuing and Loss Applet

As we learned in Chapter 1, the most complicated and interesting component of end-to-end delay is queuing delay. In this applet, you specify the packet arrival rate and the link transmission speed. You'll then see packets arrive and queue for service. When the queue becomes full, you'll see the queue overflow--that is, packet loss.

A particularly interesting case is when the emission and transmission rates are the same, for example when both are 500 packets/sec. If you let the applet run for a very long time, you'll eventually see the queue fill up and overflow. Indeed when the two rates are the same (that is, ρ = 1), the queue grows without bound (with random inter-arrival times), as described in the text.

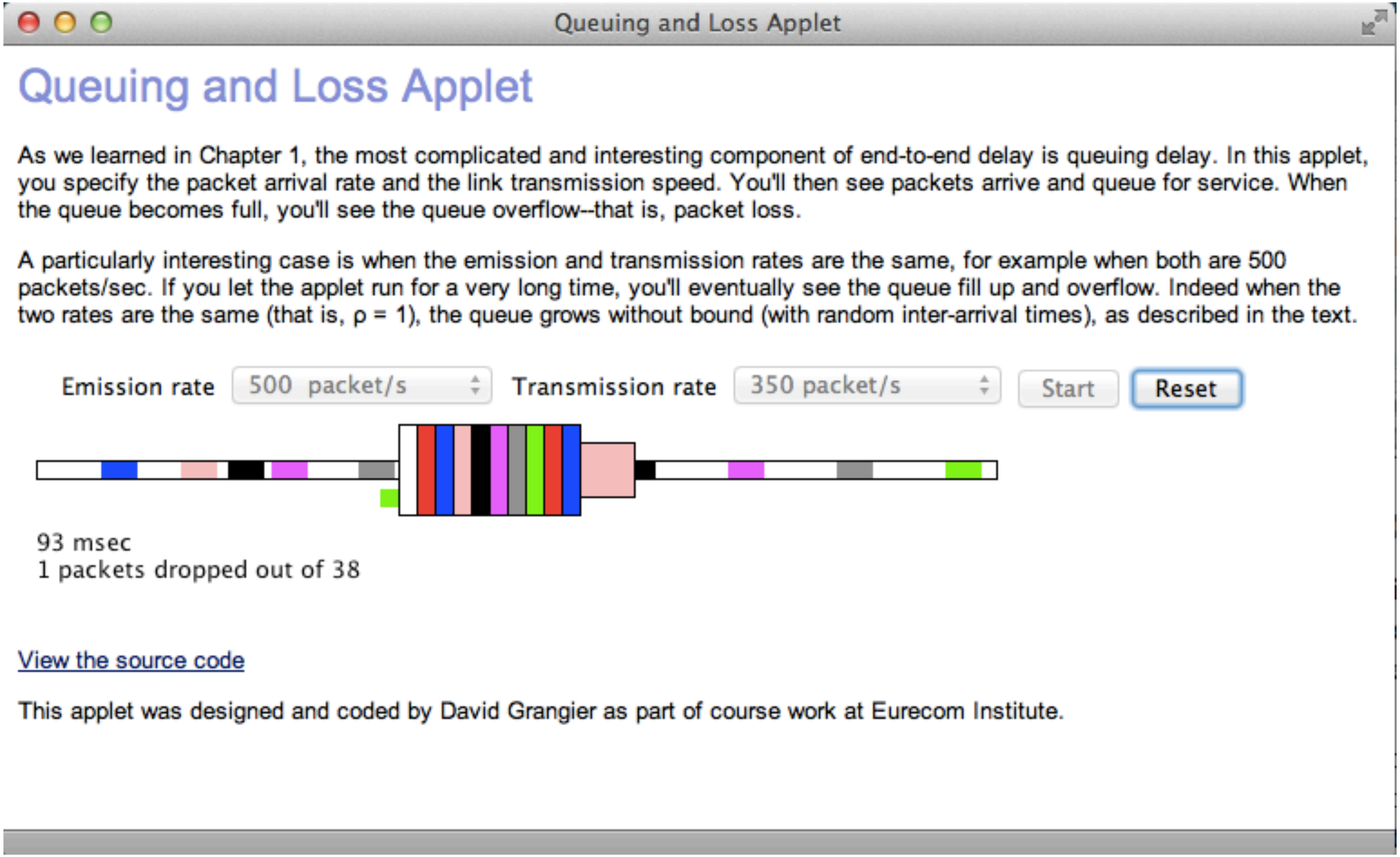
Emission rate 500 packet/s Transmission rate 350 packet/s Start Reset

64 msec
1 packets dropped out of 29

View the source code

This applet was designed and coded by David Grangier as part of course work at Eurecom Institute.

Packet Loss 2^{nd} Run



Solution to Example 3

Visit the Queuing and Loss applet at the companion web site. What is the maximum emission rate and the minimum transmission rate? With those rates, what is the traffic intensity?

Run the applet with these rates and determine how long it takes for packet loss to occur. Then repeat the experiment a second time and determine again how long it takes for packet loss to occur. Are the values different?

Ans: The maximum emission rate is 500 packets/sec and the maximum transmission rate is 350 packets/sec.

The corresponding traffic intensity is $500/350 = 1.43 > 1$.

Loss will eventually occur for each experiment; but the time when loss first occurs will be different from one experiment to the next due to the randomness in the emission process.

Packet Loss under TI=1

- When the emission and transmission rate are the same, TI=1, the queue grows without bound (with random inter-arrival times)

Queuing and Loss Applet

Queuing and Loss Applet

As we learned in Chapter 1, the most complicated and interesting component of end-to-end delay is queuing delay. In this applet, you specify the packet arrival rate and the link transmission speed. You'll then see packets arrive and queue for service. When the queue becomes full, you'll see the queue overflow--that is, packet loss.

A particularly interesting case is when the emission and transmission rates are the same, for example when both are 500 packets/sec. If you let the applet run for a very long time, you'll eventually see the queue fill up and overflow. Indeed when the two rates are the same (that is, ρ = 1), the queue grows without bound (with random inter-arrival times), as described in the text.

Emission rate 500 packet/s Transmission rate 500 packet/s Start Reset

5659 msec
29 packets dropped out of 2766

View the source code

This applet was designed and coded by David Grangier as part of course work at Eurecom Institute.

Example 4 (Problem 2)

The end-to-end delay of sending a packet consisting L bits from source to destination over a path consisting N links each of rate R is (NL/R). Generalize this formula for sending P such packets back-to-back over the N links.

Solution to Example 4

The end-to-end delay of sending a packet consisting L bits from source to destination over a path consisting N links each of rate R is (NL/R). Generalize this formula for sending P such packets back-to-back over the N links.

Ans:

At time N*(L/R) the first packet has reached the destination, the second packet is stored in the last router, the third packet is stored in the next-to-last router, etc. At time N*(L/R) + L/R, the second packet has reached the destination, the third packet is stored in the last router, etc. Continuing with this logic, we see that at time N*(L/R) + (P-1)*(L/R) = (N+P-1)*(L/R) all packets have reached the destination.

Homework Problems for Chapter 1

Solutions will posted on course website in week 4

Problem 1 (P3)

Consider an application that transmits data at a steady rate (for example, the sender generates an *N*-bit unit of data every *k* time units, where *k* is small and fixed). Also, when such an application starts, it will continue running for a relatively long period of time. Answer the following questions. Briefly justifying your answer:

a. Would a packet-switched network or a circuit-switched network be more appropriate for this application? Why?
b. Suppose that a packet-switched network is used and the only traffic in this network comes from such applications as described above. Furthermore, assume that the sum of the application data rates is less than the capacities of each and every link. Is some form of congestion control needed? Why?

Problem 2 (P6, c & g)

Consider two hosts, A and B, connected by a single link of rate *R* bps. Suppose that the two hosts are separated by *m* meters, and suppose the propagation speed along the link is *s* meters/sec. Host A is to send a packet of size *L* bits to Host B.

a. Ignoring processing and queuing delay, obtain an expression for the end-to-end delay.

b. Suppose $s=2.5*10^8$ meters/sec, $L=120$ bits, and $R=56$kbps. Find the distance *m* so that the propagation delay equals transmission delay.

Problem 3(P7)

We consider sending real-time voice from Host A to Host B over a packet-switched network (VoIP). Host A converts analog voice to a digital 64kpbs bit stream on the fly. Host A then groups the bits into 56-byte packets. There is one link between Hosts A and B, its transmission rate is 2Mbps and its propagation delay is 10msec. As soon as Host B receives an entire packet. It converts the packet's bit to an analog signal. How much time elapses from the time a bit is created (from the original analog signal at Host A) until the bit is decoded (as part of the analog signal at Host B)?

Problem 4 (P31)

In modern packet-switched networks, including the Internet, the source host segments long application-layer message (for example, an image or a music file) into smaller packets and sends the packets into the network. The receiver then reassembles the packets back into the original message. We refer to this process as message segmentation. Consider a message that is $8*10^6$ bits long that is to be sent from Host A to Host B with two packet-switches in between. Suppose each link is 2Mbps. Ignore propagation, queuing, and processing delays.

a. Consider sending the message from A to B without message segmentation. How long does it take to move the message from Host A to the first packet switch? Keeping in mind that each switch uses store-and-forward packet switching. What is the total time to move the message from A to B?

b. Now suppose that the message is segmented into 800 packets, with each packet being 10,000 bits long. How long does it take to move the first packet from A to the first switch? When the first packet is being sent from the first switch to the second switch, the second packet is being sent from A to the first switch. At what time will the second packet be fully received at the first switch?

c. How long does it take to move the file from A to B when message segmentation is used? Compare this result with your answer in part (a) and comment.

d. In addition to reducing delay, what are reasons to use message segmentation?

e. Discuss the draw-backs of message segmentation.

Solutions to Chapter 1 Problems

Q1. Consider an application that transmits data at a steady rate (for example, the sender generates an N-bit unit of data every k time units, where k is small and fixed). Also, when such an application starts, it will continue running for a relatively long period of time. Answer the following questions. Briefly justifying your answer:

a. Would a packet-switched network or a circuit-switched network be more appropriate for this application? Why?
b. Suppose that a packet-switched network is used and the only traffic in this network comes from such applications as described above. Furthermore, assume that the sum of the application data rates is less than the capacities of each and every link. Is some form of congestion control needed? Why?

(This question is taken from Kurose & Ross's book, Chapter 1 Problem 3)

Solution

a) A circuit-switched network would be well suited to the application, because the application involves long sessions with predictable smooth bandwidth requirements. Since the transmission rate is known and not bursty, bandwidth can be reserved for each application session without significant waste. In addition, the overhead costs of setting up and tearing down connections are amortized over the lengthy duration of a typical application session.

b) In the worst case, all the applications simultaneously transmit over one or more network links. However, since each link has sufficient bandwidth to handle the sum of all of the applications' data rates, no congestion (very little queuing) will occur. Given such generous link capacities, the network does not need congestion control mechanisms.

Q2. Consider two hosts, A and B, connected by a single link of rate R bps. Suppose that the two hosts are separated by m meters, and suppose the propagation speed along the link is s meters/sec. Host A is to send a packet of size L bits to Host B.

a. Ignoring processing and queuing delay, obtain an expression for the end-to-end delay.
b. Suppose s=2.5*10^8 meters/sec, L=120 bits, and R=56kbps. Find the distance m so that the propagation delay equals transmission delay.

(This question is taken from Kurose & Ross's book, Chapter 1 Problem 6, parts c and g)

Solution:

a) $d_{end-to-end} = (m/s + L/R)$ seconds.

d) Want

$$m = \frac{L}{R}s = \frac{120}{56 \times 10^3}\left(2.5 \times 10^8\right) = 536\,\text{km}.$$

Q3 We consider sending real-time voice from Host A to Host B over a packet-switched network (VoIP). Host A converts analog voice to a digital 64kpbs bit stream on the fly. Host A then groups the bits into 56-byte packets. There is one link between Hosts A and B, its transmission rate is 2Mbps and its propagation delay is 10msec. As soon as Host B receives an entire packet. It converts the packet's bit to an analog signal. How much time elapses from the time a bit is created (from the original analog signal at Host A) until the bit is decoded (as part of the analog signal at Host B)?

Solution:

Consider the first bit in a packet. Before this bit can be transmitted, all of the bits in the packet must be generated. This requires

$$\frac{56 \cdot 8}{64 \times 10^3} \text{sec=7msec.}$$

The time required to transmit the packet is

$$\frac{56 \cdot 8}{2 \times 10^6} \text{sec=} 224 \mu \text{ sec.}$$

Propagation delay = 10 msec.
The delay until decoding is

$$7\text{msec} + 224 \mu \text{ sec} + 10\text{msec} = 17.224\text{msec}$$

A similar analysis shows that all bits experience a delay of 17.224 msec.

Q4. In modern packet-switched networks, including the Internet, the source host segments long application-layer message (for example, an image or a music file) into smaller packets and sends the packets into the network. The receiver then reassembles the packets back into the original message. We refer to this process as message segmentation. Consider a message that is $8*10^6$ bits long that is to be sent from Host A to Host B with two packet-switches in between. Suppose each link is 2Mbps. Ignore propagation, queuing, and processing delays.

a. Consider sending the message from A to B without message segmentation. How long does it take to move the message from Host A to the first packet switch? Keeping in mind that each switch uses store-and-forward packet switching. What is the total time to move the message from A to B?
b. Now suppose that the message is segmented into 800 packets, with each packet being 10,000 bits long. How long does it take to move the first packet from A to the first switch? When the first packet is being sent from the first switch to the

second switch, the second packet is being sent from A to the first switch. At what time will the second packet be fully received at the first switch?

c. How long does it take to move the file from A to B when message segmentation is used? Compare this result with your answer in part (a) and comment.

d. In addition to reducing delay, what are reasons to use message segmentation? Discuss the draw-backs of message segmentation.

Solution:

a) Time to send message from source host to first packet switch = $\frac{8\times10^{6}}{2\times10^{6}}\sec = 4\sec$

With store-and-forward switching, the total time to move message from source host to destination host = $4\sec\times 3\ hops = 12\sec$

b) Time to send 1st packet from source host to first packet switch = . $\frac{1\times10^{4}}{2\times10^{6}}\sec = 5\ m\sec$. Time at which 2nd packet is received at the first switch = time at which 1st packet is received at the second switch = $2\times 5m\sec = 10\ m\sec$

c) Time at which 1st packet is received at the destination host = $5\ m\sec\times 3\ hops = 15\ m\sec$. After this, every 5msec one packet will be received; thus time at which last (800th) packet is received = $15\ m\sec + 799 * 5m\sec = 4.01\sec$. It can be seen that delay in using message segmentation is significantly less (almost 1/3rd).

d)

i. Without message segmentation, if bit errors are not tolerated, if there is a single bit error, the whole message has to be retransmitted (rather than a single packet).

ii. Without message segmentation, huge packets (containing HD videos, for example) are sent into the network. Routers have to accommodate these huge packets. Smaller packets have to queue behind enormous packets and suffer unfair delays.

e)

i. Packets have to be put in sequence at the destination.

ii. Message segmentation results in many smaller packets. Since header size is usually the same for all packets regardless of their size, with message segmentation the total amount of header bytes is more.

Computer Networks - CS132/EECS148 - Spring 2013
Instructor: Karim El Defrawy
Assignment 1
Deadline : April 18th – 9:30pm (hard and soft copies required)
Each problem has 10 points.

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Problem 1 - Consider a packet switching architecture.
a. What are the main components of delay when we use packet switching?
Processing delay, queuing delay, transmission delay and propagation delay

b. What is the difference between transmission delay and propagation delay?
Transmission delay is the time needed to put the entire packet on the link and is dependent on the length of the packet, while the propagation delay is needed time for one bit to reach to the other end of the link.

c. How is propagation delay affected if the length of the packet is increased?
This delay is just a function of link length and its physical characteristics, so it is NOT affected.

Problem 2 - Consider an application that transmits data at a steady rate (for example the sender generates an N-bit unit of data every K time units, where K is small and fixed). Also, when such an application starts, it will continue running for a relatively long period of time. Answer the following questions, briefly justifying your answers.
a. Would a packet-switched network or a circuit-switched network be more appropriate for this application? Why?
A circuit-switched network would be well suited to the application, because the application involves long sessions with predictable smooth bandwidth requirements. Since the transmission rate is known and not bursty, bandwidth can be reserved for each application session without significant waste. In addition, the overhead costs of setting up and tearing down

connections are amortized over the lengthy duration of a typical application session.

b. Suppose a packet-switched network is used and the only traffic in this network comes from this application as described above. Furthermore, assume that the sum of the application data rates is less than the capacities of each and every link. Is some form of congestion control needed? Why?

In the worst case, all the applications simultaneously transmit over one or more network links. However, since each link has sufficient bandwidth to handle the sum of all of the applications' data rates, no congestion (very little queuing) will occur. Given such generous link capacities, the network does not need congestion control mechanisms.

Problem 3 (Similar to Problem 9 in Ch1 in Kurose/Ross) - Consider the difference between circuit switching and packet switching. Assume the link's rate is 2 Mbps and users are generating data at a rate of 100 Kbps when busy. Users are busy only %1 of time.

Problem 9 of chapter 1. Numbers are changed.

a. What is the maximum number of users that a circuit switching architecture can support simultaneously?

2 Mbps / 100 Kbps = 20 users

b. Write down the formula to calculate the probability of having more than 5 active users, assuming that we have 20 users in total.

(Taught in discussion class)

1 - (C(20,0)0.99^20 + C(20,1)0.01*0.99^19 + ... + C(20,5)0.01^5*0.99^15)

c. Explain the effects of having more users in a packet switching architecture with the above characteristics.

The probability of having more than 20 users simultaneously increases, so the buffers in routers become fuller and gradually packets will be dropped.

Problem 4 (Similar to Problem 25 in Ch1 in Kurose/Ross) -Suppose two hosts, A and B, are separated by 10,000 kilometers and are connected by a direct link of R = 2 Mbps. Suppose the propagation speed over the link is 2.5 * 10^8 meters/sec.

Problem 25 - Chapter 1 (numbers are changed)

a. Calculate the bandwidth-delay product , R * d_prop.

2 Mpbs * (10000*1000 / (2.5*10^8)) = 0.8 Mb

b. Consider sending a file of 1000,000 bits from host A to host B. Suppose the file is sent continuously as one large message. What is the maximum number of bits that will be in the link at any given time?

Maximum number of bits : 0.8 M bits

c. Provide an interpretation of the bandwidth-delay product.

It is the maximum number of bits that the link can contain in the same time.

Problem 5 - Suppose a user wants to load a simple static HTML page into his/her browser. Also assume RTT is 5 seconds and the total needed time for the page to be transferred is 4 seconds. Draw a diagram showing different steps of this process and calculate the time it takes from the initial request to the server until the user receives the file completely.

Draw the diagram (or similar one) provided in page 102 in the book. The total time is: 2*RTT + total file transfer time = 2*5 + 4 = 14 seconds.

Problem 6 - Consider visiting Google's main search page. Write down the HTTP request/response format and fill out different parts of the packet (method, URL, version , status code ....) according to this case.

Note: You can use Firebug plugin for Firefox or Inspect Element feature of Chrome to see what you have in your sending/receiving packets.

Write down the HTTP request/response format provided in pages 105 and 107 and fill them out with information from request to get www.google.com (method is GET, version depends on browser, URL is www.google.com , no headers , status should be 200 OK ...)

[Optional, Extra Credit] Problem 7 - Perform a traceroute between source and destination on the same continent at three different hours of the day.

a. Find the average and standard deviation of the round-trip delays at each of the three hours.
b. Find the number of routers in the path at each of the three hours. Did the path change during any of the hours?
c. Try to identify the number of ISP networks that the traceroute packets pass through from source to destination. Routers with similar names and/or similar IP addresses should be considered as part of the same ISP. In your experiment do the largest delays occur at the peering interfaces between adjacent ISPs?
d. Repeat the above for a source and destination on different continents. Compare the intra-continent and inter-continent results.
On linux you can use the command
traceroute www.targethost.com
and in the Windows command prompt you can use
tracert www.targethost.com

In either case, you will get three delay measurements. For those three measurements you can calculate the mean and standard deviation. Repeat the experiment at different times of the day and comment on any changes.

Here is an example solution:

Traceroutes between San Diego Super Computer Center and www.poly.edu

a) The average (mean) of the round-trip delays at each of the three hours is 71.18 ms, 71.38 ms and 71.55 ms, respectively. The standard deviations are 0.075 ms, 0.21 ms, 0.05 ms, respectively.

b) In this example, the traceroutes have 12 routers in the path at each of the three hours. No, the paths didn't change during any of the hours.

c) Traceroute packets passed through four ISP networks from source to destination. Yes, in this experiment the largest delays occurred at peering interfaces between adjacent ISPs.

Traceroutes from www.stella-net.net (France) to www.poly.edu (USA).

d) The average round-trip delays at each of the three hours are 87.09 ms, 86.35 ms and 86.48 ms, respectively. The standard deviations are 0.53 ms, 0.18 ms, 0.23 ms, respectively. In this example, there are 11 routers in the path at each of the three hours. No, the paths didn't change during any of the hours. Traceroute packets passed three ISP networks from source to destination. Yes, in this experiment the largest delays occurred at peering interfaces between adjacent ISPs.