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Project on

Delay Measurement and Analysis of Time-critical Multimedia Applications over the Internet

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Statement

I assure, that the following work is done by myself and with the help of listed literature.

July, 2004 Braunschweig Pepaala Bharath Kumar

Abstract

This project deals with the measurement and analysis of the end-to-end delay composition of real-time traffic across the global Internet and the factors that contribute to the total delay budget (e.g. end-host processing delay, single-hop delay at network nodes, etc.). The delay plays a vital role in the provisioning of Quality of Service (QoS) for time-critical interactive multimedia applications like real-time audio/video content distribution (e.g. video telephony), video conferencing and networked multi-player game.

The main goal is to measure the percentage of packets one-way-delay more than 100ms or the percentage of packets out of boundary delay is more than 200ms.To evaluate this we used two methods. First, to find the roundtriptime to game servers we used Visual route tool which uses the ICMP packets for traceroute. Second, we used tcptraceroute which uses TCP packet for traceroute. Using these methods we have collected data in different time slots and calculated the percentage of packet roundtriptime more than 200ms. And also in this report we have presented some results which are collected from different literatures in order to find the percentage of packet roundtrip time more than 200 ms. Finally we presented the results.

If the packets roundtriptime is more than 200 ms, we considered as obsolete packets. Because of these packets the end-user can get lot of problems for interactive multimedia applications. Based on these result we can control the packet lifetime for interactive multimedia applications like online music, video-conferencing and multi-player games etc.

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Introduction

The worldwide fever for interactive multimedia applications is soaring due to great proliferation of the Internet and abundance of broadband. Most of these applications (e.g. video telephony, networked multiplayer game) are highly delay-sensitive; therefore depend on the support from the networks. Packet loss and delay are more important factors and effects the quality of service of the interactive multimedia applications like online music, video-conferencing and multi-player games etc.Delay and Jitter introduces a lot of inconvenience when the user is watching movies or playing online games or listening music. In next sections we will give the detailed description of different delays existing in one-way-delay or round-trip-time and calculating end to end delay.

In this report, the focus is on the measurement and analysis on packets of one-way-delay or roundtriptime and packet loss in the internet. Using UDP or TCP transport of data, the loss of packets during transport affects the quality of service degradation when loss is excessive in real time applications such as internet telephony or video conferencing. Though Transmission Control Protocol is a reliable protocol and guarantees delivery of packets and in order, we use User Datagram Protocol. Because UDP is suitable for real time multimedia applications due to lower overhead and lower delay than TCP. Also, the retransmission nature of TCP is not well suited for real time data streams, such as MPEG video. While TCP provides a reliable end to end transmission, a single packet loss causes the TCP to decrease its transmission rate. So it is very difficult to maintain the high throughput over a lossy path. So we prefer the UDP transmission for interactive multimedia applications.

We have collected data in two ways. One is, using VisualRoute tool at different time intervals for different websites to calculate the average round trip time from Europe to different countries. The statistics related to roundtriptime is explained in detail in the further chapters. Second is, using tcptraceroute we collected data in different time slots and calculated the percentage of packets roundtriptime more than 200ms.

Regarding experimental approaches, using some tools which examine the variations of packet delay and loss for different paths, different time intervals of the day of the week, etc. Then we have to find out the percentage of packets loss in real time multimedia applications and percentage of packets roundtriptime more than 200ms or one way delay more than 100 ms.

The chapter2 describes the detailed description of delay and different types of delays. Chapter 3 describes the factors that effects on the packet loss and focus on the analysis from different literatures. Chapter 4 gives the details about tcptraceroute and its results.Chapter 5 gives the conclusion about the project work and finally discusses the further possible future extension work for this project .

2. Packet Delay

2.1 Motivation:

With the emergence of new applications on data networks, it is becoming increasingly important for customers to accurately predict the impact of new application rollouts. Not long ago, it was easy to allocate bandwidth to applications and let the applications adapt to the exploding nature of traffic flows through timeout and retransmission functions of the upper layer protocols. Now, however, new world applications, such as voice and video, are more susceptible to changes in the transmission characteristics of data networks. It is necessary to understand the traffic characteristics of the network before deployment of new world applications to ensure successful implementations.

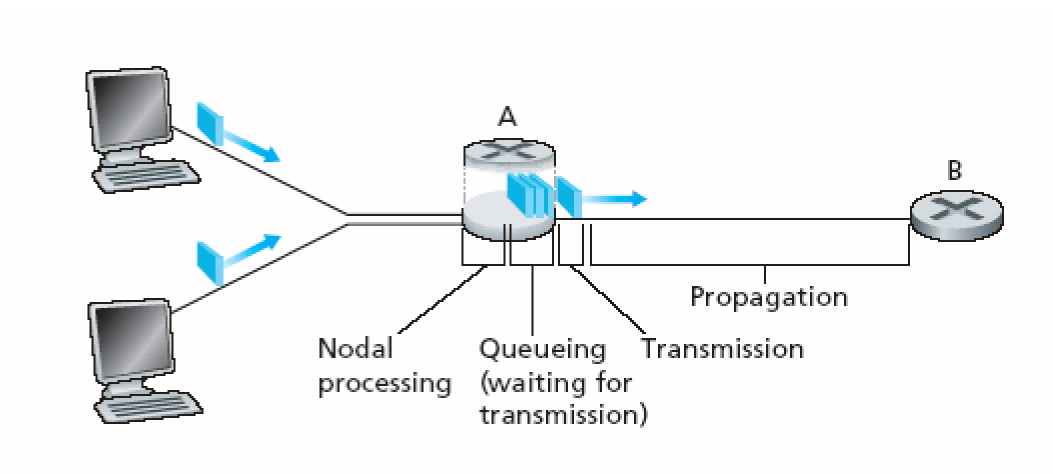
Interactive real time media applications like online music, movies, games and video conferencing etc. are susceptible to network behaviors, referred to as delay and loss which can degrade the quality of service of the applications. Delay is the time taken for traveling the packets from point-to-point in a network. Delay can be measured in either one-way or round-trip delay. One-way delay calculations require expensive sophisticated test gear and are beyond the budget and expertise of most enterprise customers. However, measuring round-trip delay is easier and requires less expensive equipment. To get a general measurement of one-way delay, measure round-trip delay and divide the result by two. While measuring the end to end delay we can take either one way delay or Roundtrip time.

The measurement of one-way delay instead of round-trip delay is motivated by the following factors:

- In today's internet, the path from a source to destination may be different the path from the destination back to the source (asymmetric path), that means different sequence of router are used in forward and reverse paths. Therefore round trip tine measurements actually measure the performance of two distinct paths together. Measuring the each path independently highlights the performance difference between the two paths which may traverse different internet service providers and even radically different types of networks.

- Even when two pats are symmetric, they may have different performance characteristics due to asymmetric queuing.
- Performance of an application may depend mostly on the performance in one direction. For example file transfer using TCP may depend on one direction than acknowledgement.
- In quality of service enabled networks, provisioning in one direction may be radically different than provisioning in the reverse direction, and thus the Qos guarantees differ.

In the data transmission each packet travels from its source to its destination. Recall that a packet starts in a host (source), passes through a series of routers, and ends its journey at another host (destination). The packet suffers from several different types of delay at each node along the path while it is traveling form one node (host or router) to subsequent node (host or router). The most important of these delays are the **nodal processing delay, queuing delay, transmission delay, and propagation delay**; together, these delay accumulate to give the total nodal delay.



The delay through router A (Ref [8])

2.2 Types of Delays:

As a part of end -to-end route between source and destination, packet is sent from upstream node through router A to router B. When the packet arrives at router A , router A examines the packet's header in order to place it in appropriate link. Then router A directs it to that particular link.

Processing Delay: The time required to examine the packet's header and determine where to direct the packet is part of the processing delay. The processing delay can also include other factors, such as the time needed to check for bit-level errors in the packet that occurred in transmitting the packet's bits from the upstream node to router A. Processing delays in high-speed routers are typically on the order of microseconds or less. After this nodal processing, the router directs the packet to the queue that precedes the link to router B.

Queuing Delay: At the queue, the packet experiences a queuing delay as it waits to be transmitted onto the link. The queuing delay of a specific packet will depend on the number of earlier-arriving packets that are queued and waiting for transmission across the link. The delay of a given packet can vary significantly from packet to packet. If the queue is empty and no other packet is currently being transmitted, then our packet's queuing delay is zero. On the other hand, if the traffic is heavy and many other packets are also waiting to be transmitted, the queuing delay will be long. Queuing delays can be on the order of microseconds to milliseconds in practice.

Transmission Delay: Assuming that packets are transmitted in first-come-first-served manner, as is common in packet-switched networks, our packet can be transmitted only after all the packets that have arrived before it have been transmitted. Denote the length of the packet by L bits, and denote the transmission rate of the link from router A to router B by R bits/sec. The rate R is determined by the transmission rate of the link to router B. For example, for a 10 Mbps Ethernet link, the rate is R = 10 Mbps; for a 100 Mbps Ethernet link, the rate is R = 100 Mbps. The transmission delay (also called the store-and-forward delay, is L/R. This is the amount of time required to push (that is, transmit) all of the packet's bits into the link. Transmission delays are typically on the order of microseconds to milliseconds in practice.

Propagation Delay: Once a bit is pushed onto the link, it needs to propagate to router B. The time required to propagate from the beginning of the link to router B is the propagation delay. The bit propagates at the propagation speed of the link. The propagation speed depends on the physical medium of the link (that is, fiber optics, twisted-pair copper wire, and so on) and is in the range of

2 • 108 meters/sec to 3 • 108 meters/sec

which is equal to, or a little less than, the speed of light. The propagation delay is the distance between two routers divided by the propagation speed. That is, the propagation delay is *d/s*, where *d* is the distance between router A and router B and *s* is the propagation speed of the link. Once the last bit of the packet propagates to node B, it and all the preceding bits of the packet are stored in router B. The whole process then continues with router B now performing the forwarding. In wide-area networks, propagation delays are on the order of milliseconds.

If we let *d*proc, *d*queue, *d*trans, and *d*prop denote the processing, queuing, transmission, and propagation delays, then the total nodal delay is given by

$d_{nodal} = d_{proc} + d_{queue} + d_{trans} + d_{prop}$

The contribution of these delay components can vary significantly.

End-System Delay: - It is the time required to put the bits on the network.

2.3 End-to-End Delay:

Our discussion up to this point has been focused on the nodal delay, that is, the delay

at a single router. Let us conclude our discussion by briefly considering the delay from source to destination. To get a handle on this concept, suppose there are *N*–1 routers between the source host and the destination host. Let us also suppose that the network is uncongested (so that queuing delays are negligible), the processing delay at each router and at the source host is *d*proc, the transmission rate out of each router and out of the source host is *R* bits/sec, and the propagation on each link is *d*prop. The nodal delays accumulate and give an end-to-end delay,

*d*end-end = *N* (*d*proc + *d*trans + *d*prop)

where, once again, *d*trans = *L/R*, where *L* is the packet size.

According to reference-1 end-to-end one way delay should be below 100 ms for the interactive real time multimedia applications.

According to reference-5, various studies concluded that delay:

	Excellent	Good	Acceptable	Poor	Very Poor	Bad
Loss	<0.1%	>=0.1% & & < 1%	> =1% & < 2.5%	>= 2.5% & < 5%	>= 5% & < 12%	> 12%
RTT		<62.5ms	>=62.5ms & < 125ms	>= 125ms & < 250ms	>=250ms & < 500ms	>500ms

- 100ms end to end one way delay should be good for human interactive applications

2.4 Delay and Routes in the Internet

To get a hands-on feel for the delay in a computer network, we can make use of the Traceroute diagnostic program. Traceroute is a simple program that can run in any Internet host. When the user specifies a destination host name, the program in the source host sends multiple, special packets toward that destination. As these packets work their way toward the destination, they pass through a series of routers. When a router receives one of these special packets, it sends a short message back to the source. This message contains the name and address of the router.

More specifically, suppose there are *N*–1 routers between the source and the destination. Then the source will send *N* special packets into the network, with eachpacket addressed to the ultimate destination. These *N* special packets are marked *1*through *N*, with the first packet marked *1* and the last packet marked *N*.

When the *n*th router receives the *n*th packet marked *n*, the router destroys the packet and sends a message back to the source. And when the destination host receives the *N*th packet, the destination destroys it as well, but again returns a message back to the source. The source records the time that elapses from when it sends a packet until when it receives the corresponding return message; it also records the name and address of the router (or the destination host) that returns the message. In this manner, the source can reconstruct the route taken by packets flowing from source to destination, and the source can determine the round-trip delays to all the intervening routers. Traceroute actually repeats the experiment just described three times, so the source actually sends *3•N* packets to the destination. RFC 1393 describes Traceroute in detail. Hers is an example of traceroute to www.games.yahoo.com

```
Tracing route to rc.yahoo.akadns.net [216.109.112.135]
over a maximum of 30 hops:
  1    <1 ms    <1 ms    <1 ms  corona.ibr.cs.tu-bs.de [134.169.34.1]
  2    <1 ms    <1 ms    <1 ms  infogate.rz.tu-bs.de [134.169.39.254]
  3    <1 ms    <1 ms    <1 ms  rzrouter.rz.tu-bs.de 134.169.246.14]
  4 <1 ms <1 ms <1 ms  ar-braunschweig3-ge4-0-222.g-win.dfn.de
[188.1.46.145]
  5     4 ms    3 ms    3 ms  cr-hannover1-po0-0.g-win.dfn.de [188.1.88.69]
  6     8 ms    8 ms    8 ms cr-frankfurt1-po9-3.g-win.dfn.de
[188.1.18.181]
  7    10 ms   10 ms   10 ms  so-6-0-0.ar2.FRA2.gblx.net [208.48.23.141]
  8    98 ms   98 ms   99 ms  so5-1-0-2488M.ar1.DCA3.gblx.net [67.17.68.37]
  9   128 ms 128 ms 128 ms  208.50.13.210
 10   128 ms 128 ms 128 ms  vlan220-msr2.dcn.yahoo.com [216.115.96.165]
 11   130 ms 131 ms 130 ms  UNKNOWN-216-109-120-207.yahoo.com
[216.109.120.207]
 12   130 ms 129 ms 130 ms  w2.rc.vip.dcn.yahoo.com [216.109.112.135]

Trace complete.
```

2.5 Analysis-1

To find the Roundtrip time from Europe to different countries in the world we used VisualRoute tool. Here we have taken 4 different websites from different countries and collected data using this tool at different time intervals and evaluated some statistics on it and plotted. The figures are shown.

VisualRoute Personal Edition combines essential networking utilities, including traceroute, ping, WHOIS, and reverse DNS, to determine precisely where and how traffic is flowing on an Internet connection, providing a geographical map of the route and the performance of each segment. VisualRoute is recognized the leading Internet traceroute utility.

The IP address location can be found using tool *SmartWhoIs.* SmartWhois is a useful network information utility that allows you to find out all available information about an IP address, host name, or domain, including country, state or province, city, name of the network provider, administrator and technical support contact information.

Calculating Single Hop delay:

$$\Pr opagationDelay = \frac{Dis\tan cebetweencities}{propagationSpeed}.$$

To find distance between cities, there is *City Distance Tool* in http://www.geobytes.com/.

Light Speed $=3*10^8$ m/sec .We know the Round trip time each hop from the source from the VisualRoute Tool report. Here Time in milli seconds (ms)

For any packet n, the single hop delay through the router is the difference between its arrival and departure. It corresponds to the total time packet spends in a router, including IP address lookup time at the input port, transmission time, waiting time at the output port and transmission delay.

$$DifferenceDelay = currenthopRTT - previoushopRTT.$$

$$SingleHopdelay = \frac{DifferenceDelay}{2} - (d_{proc} + d_{queue} + d_{trans} + d_{prop})$$

The average round trip time can be calculated by taking game servers located in USA, Australia, UK, Norway ,China, Deutschland in three time intervals i.e. morning, afternoon and night. Finally calculated statistics on that data and plotted.

The following plots represents the average roundtrip time for the game servers. In the graphs we can observe the minimum and maximum average round trip time.

The game server is

www.4u-servers.co.uk located in UK

http://games.yahoo.de located in Germany.

http://games.yahoo.com located in USA.

http://cn.games.yahoo.com located in China.

http://au.games.yahoo.com located in Australia.

www.rtcw.no located in Norway.

In the x-axis we have taken Hop number and y-axis we have taken average delay at each hop. The Graphs are shown below.

Location of Server	average RTT around 9AM in ms	average RTT around 13PM in ms	average RTT around 21 PM in ms
Australia	339.7	342.5	338.36
Germany	23.1	26.9	27.7
Norway	50.2	44	44.1
China	813.6	637.3	393.1
UK	25	32.9	33.1
USA	165.1	167.8	167.54

Table 1:- Average roundtriptime to different server located in different continents using Visualroute tool

Report for www.4u-servers.co.uk [195.20.108.140]:

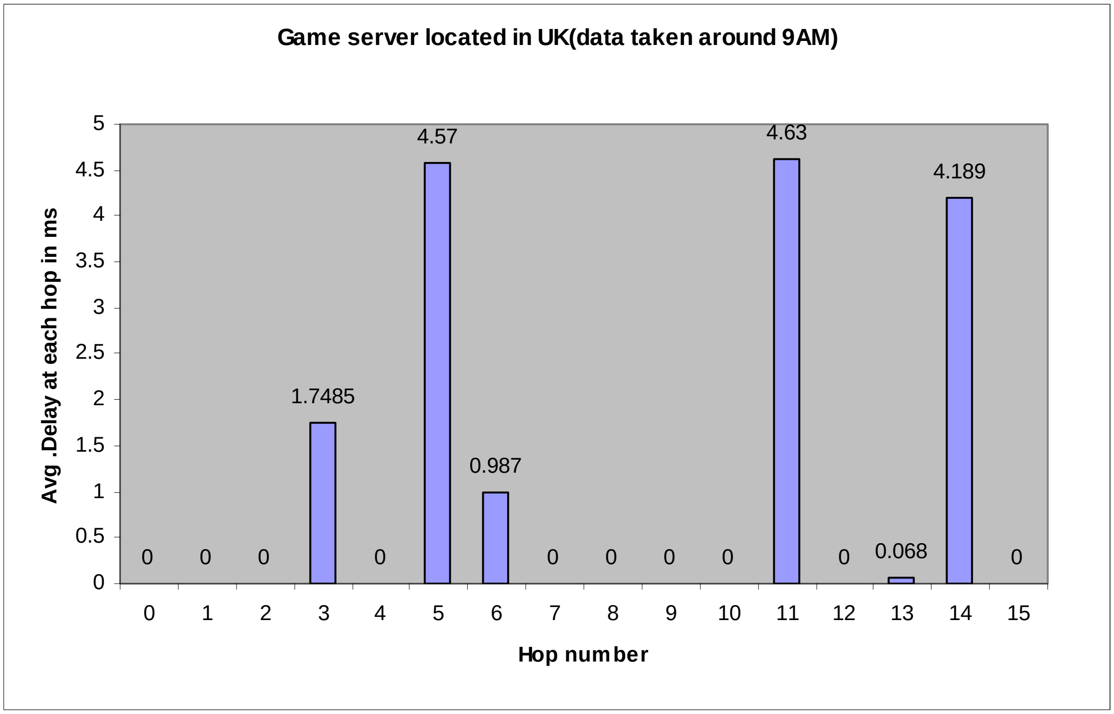


Figure.1

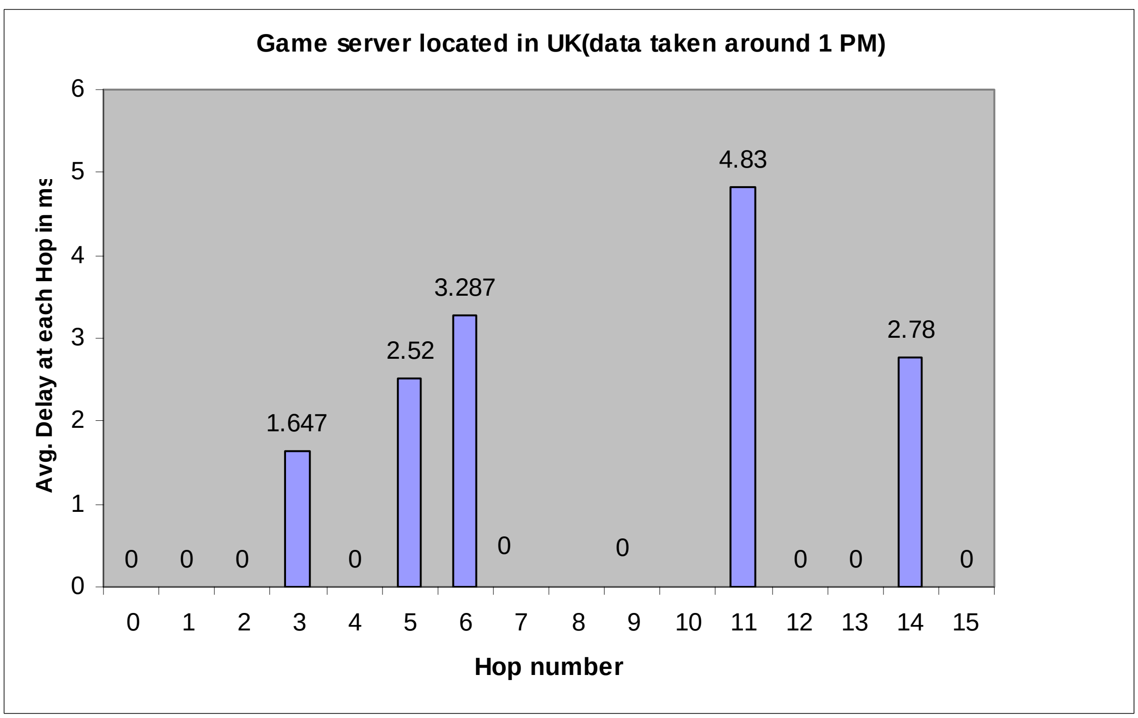


Figure .2

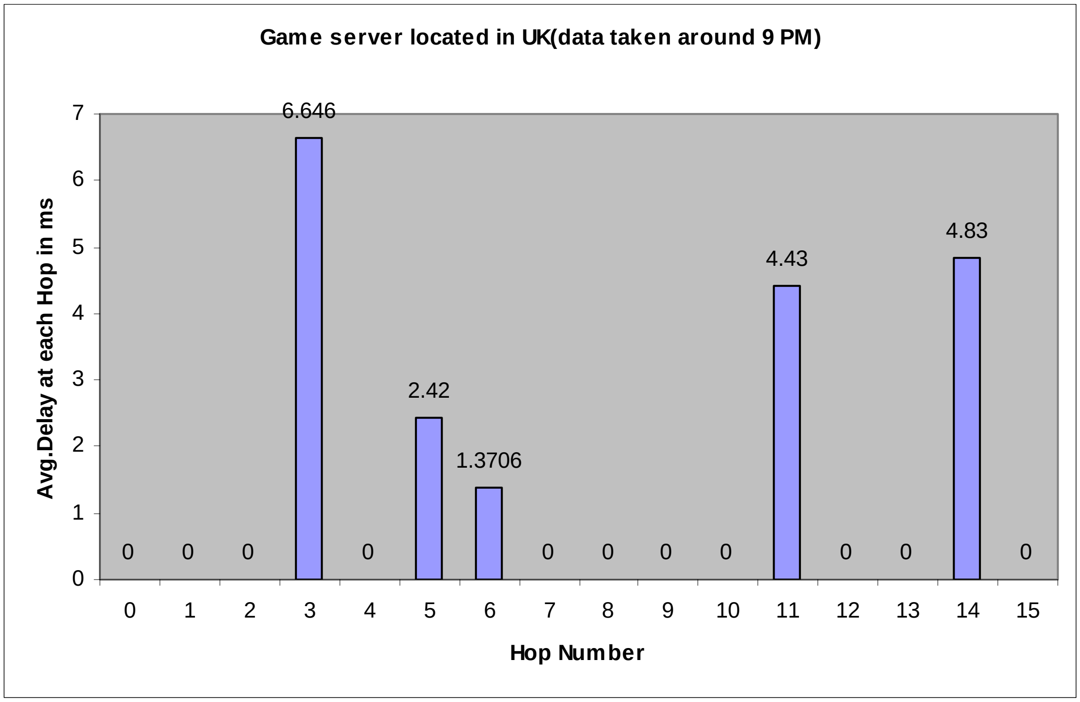


figure-3

Analysis: 'www.4u-servers.co.uk' was found in 15 hops. It is a HTTP server (running Apache/1.3.27 (UNIX) (Red-Hat/Linux) mod_ssl/2.8.12 OpenSSL/0.9.6b PHP/4.3.3).This server is located in London, UK

Average Roundtrip time for this server in morning = 25 ms

Average Roundtrip time for this server in night = 32.9 ms

Average Roundtrip time for this server in afternoon = 33.1 ms

Figure 1

Data can be taken for 10 times at night after 9O'Clock and calculated average round trip time at each hop for all test runs.

Figure.2

Report taken at morning after 9O'Clock.

Figure.3

Report taken in the afternoon.

Report for http://games.yahoo.de [217.12.6.29]

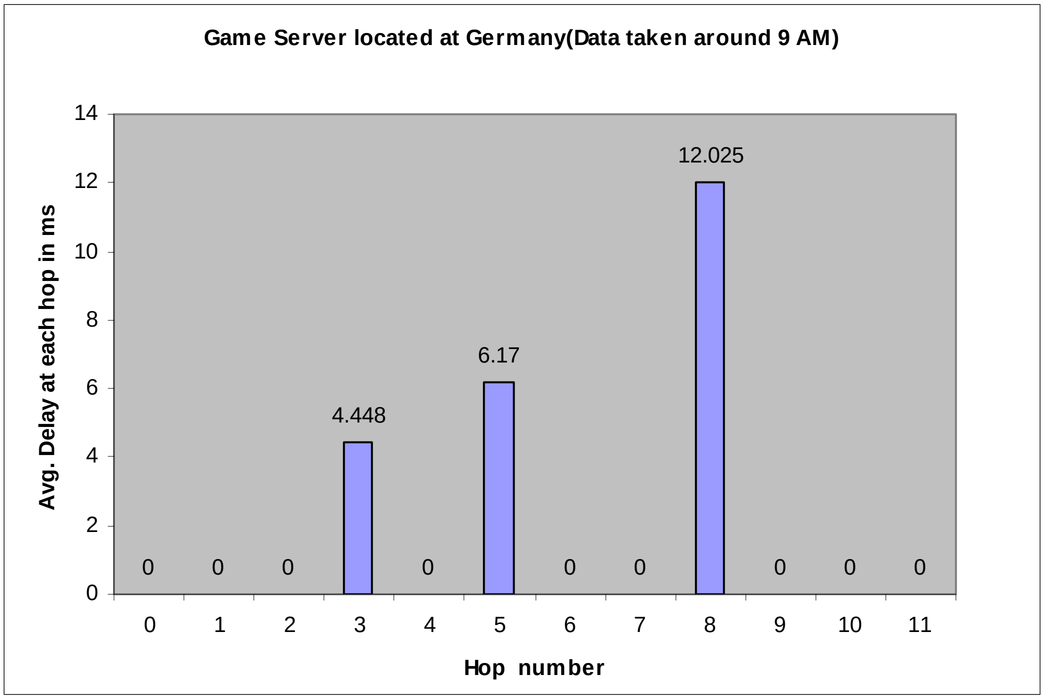


Figure 4

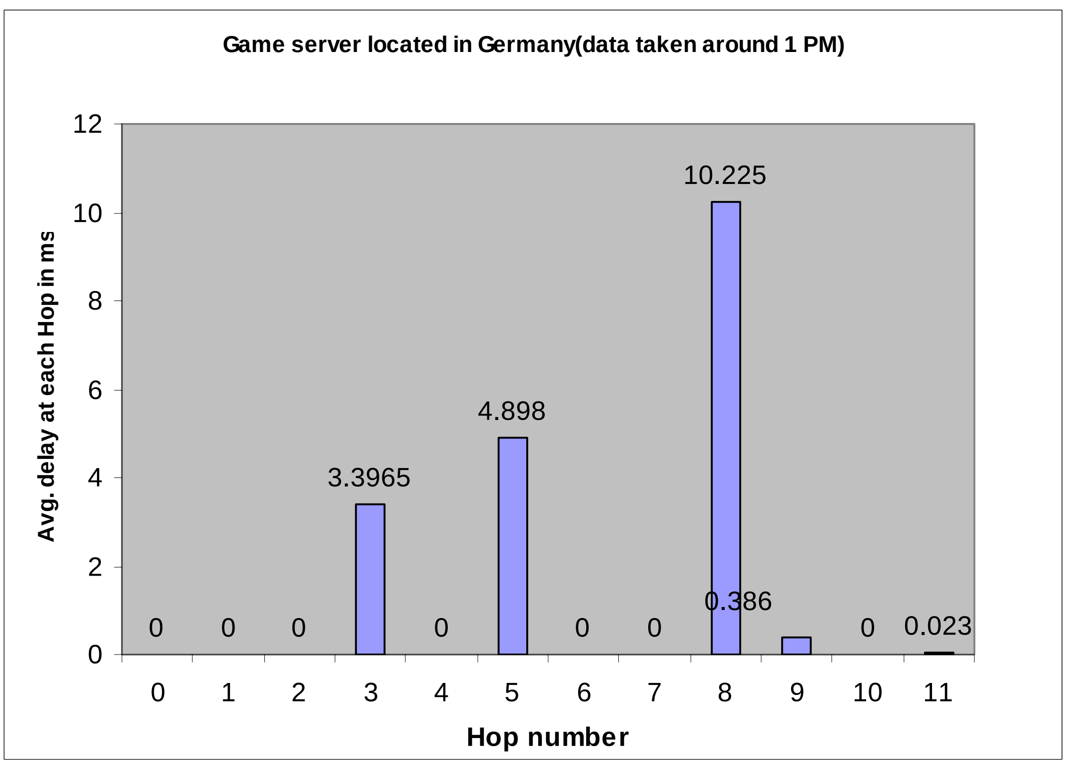


Figure 5

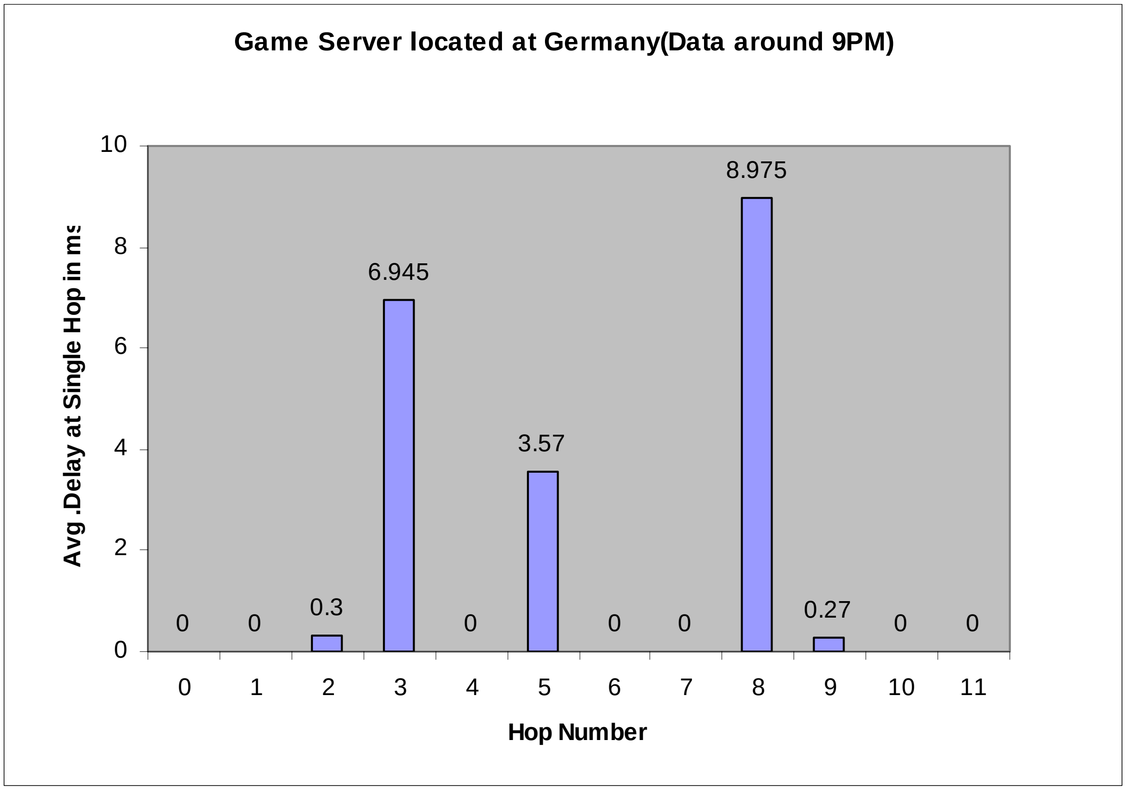


Figure-6

Analysis: 'http://games.yahoo.de' [rc1.vip.ukl.yahoo.com] was found in 11 hops (TTL=244). Connections to HTTP port 80 are working.

Average Round trip time for this server in morning = 23.1ms

Average Round trip time for this server in afternoon= 26.9 ms

Average Round trip time for this server in night =27.7 ms

Real-time report for games.yahoo.com [66.218.71.230]

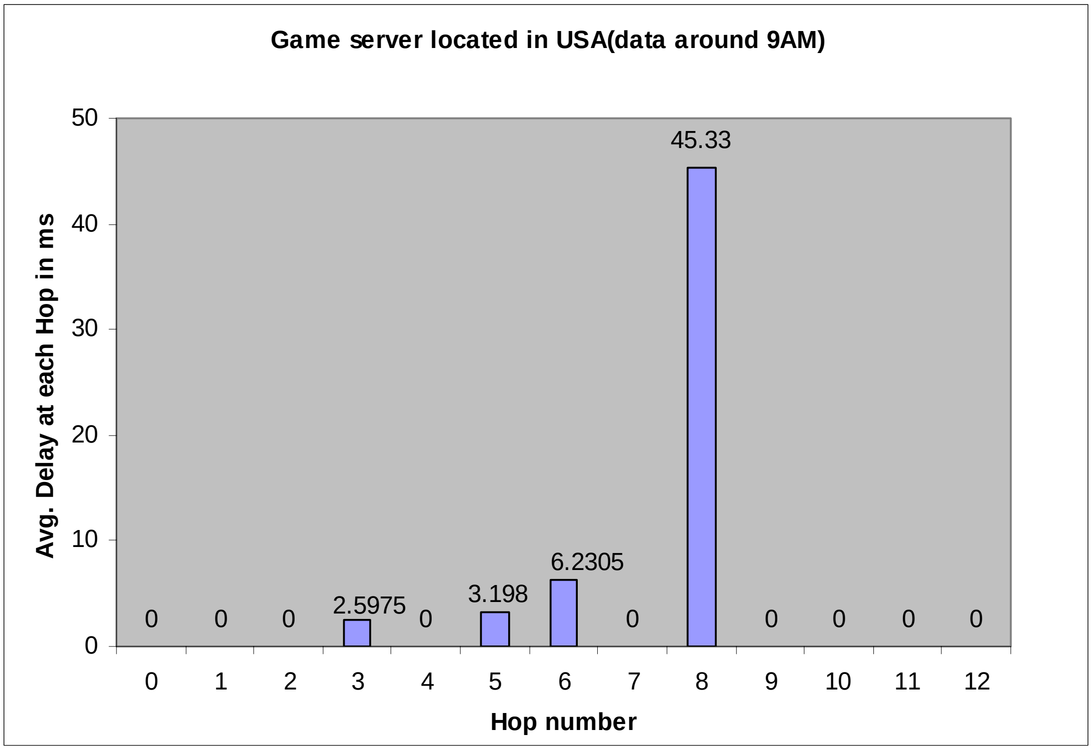


Figure 7

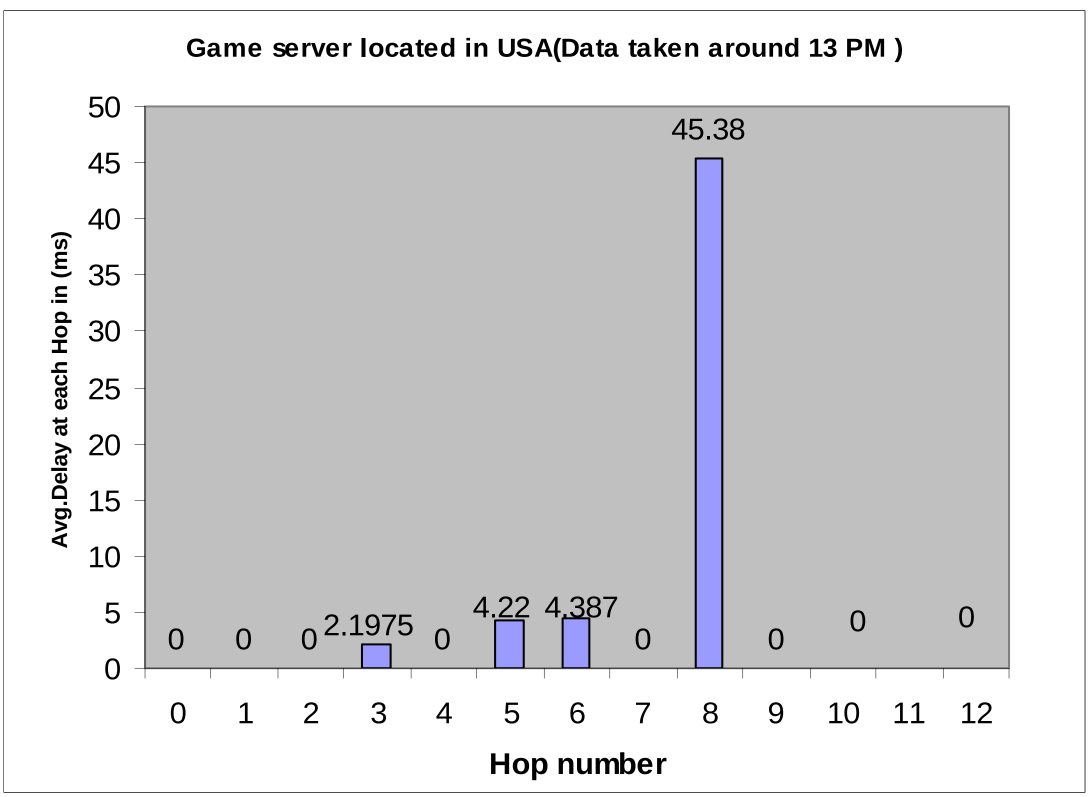


Figure-8

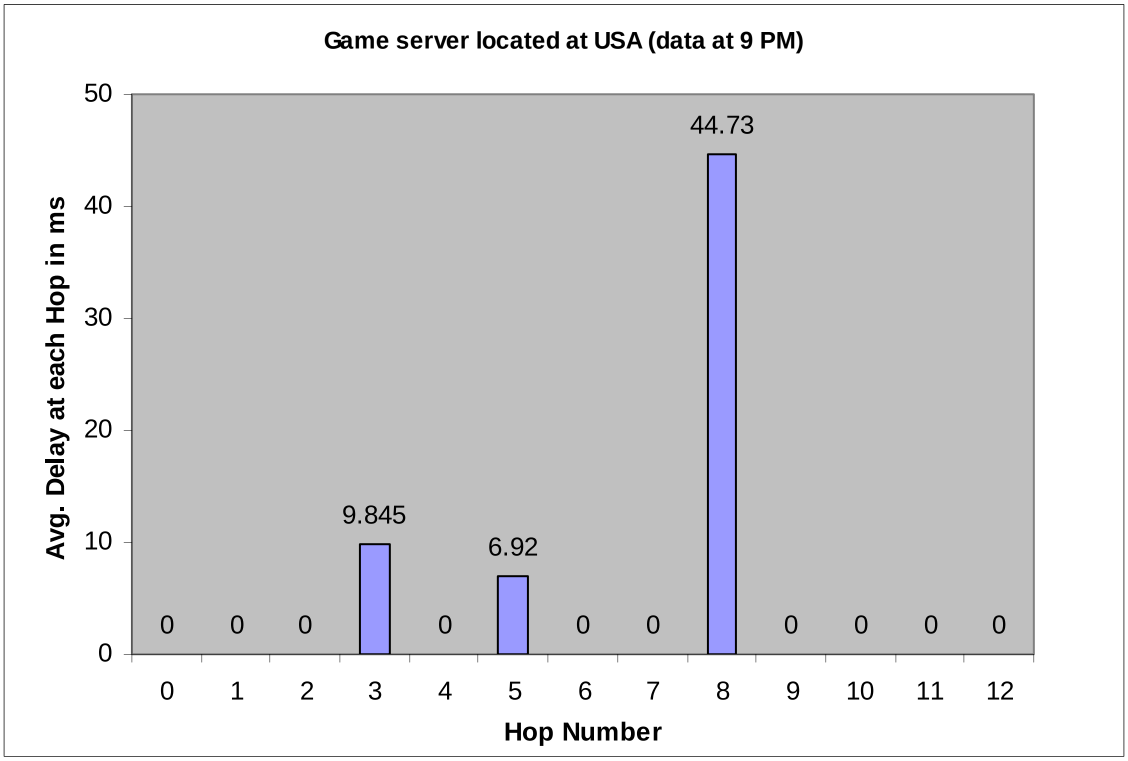


Figure 9

<u>Analysis</u>: 'http://games.yahoo.com' [games1.vip.scd.yahoo.com] was found in 12 hops (TTL=234). Connections to HTTP port 80 are working.

Average Round trip time for this server in the morning = 165.1 ms

Average Round trip time for this server in the night = 167.8 ms

Average Round trip time for this server in the morning = 167.54 ms

Report for au.games.yahoo.com [202.3.14.158]

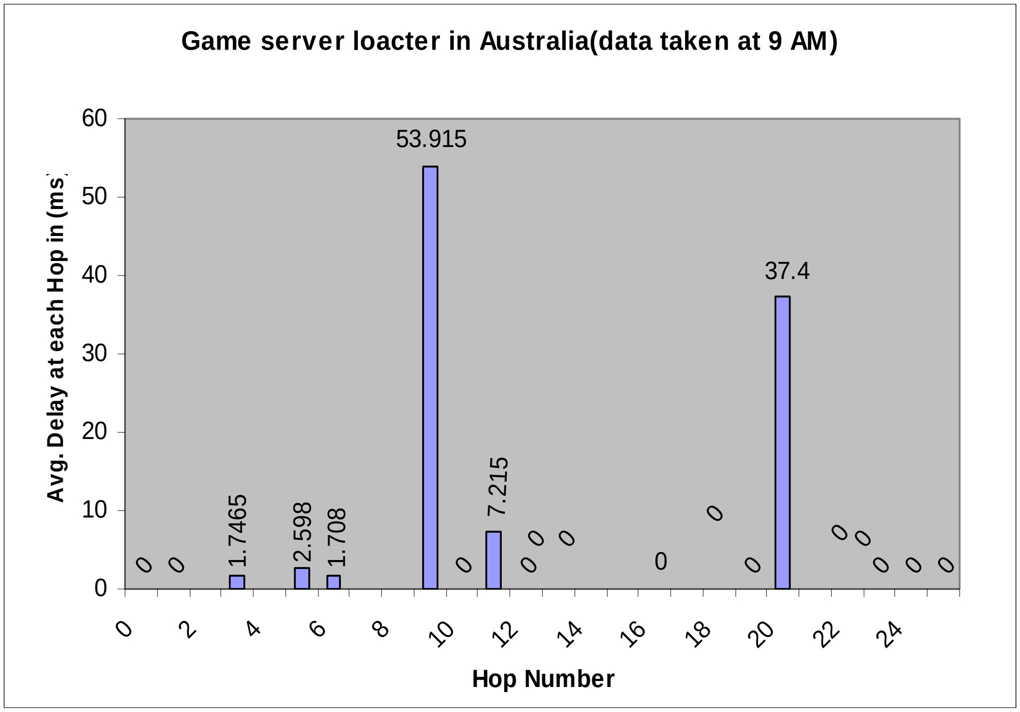


Figure-10

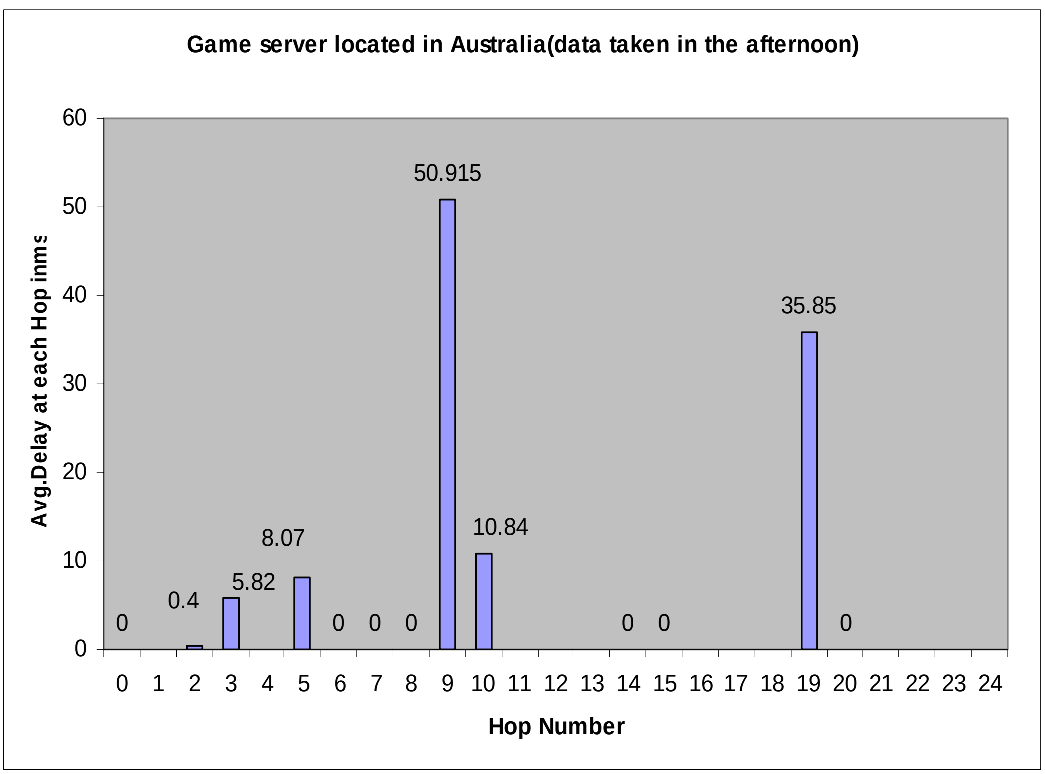


Figure-11

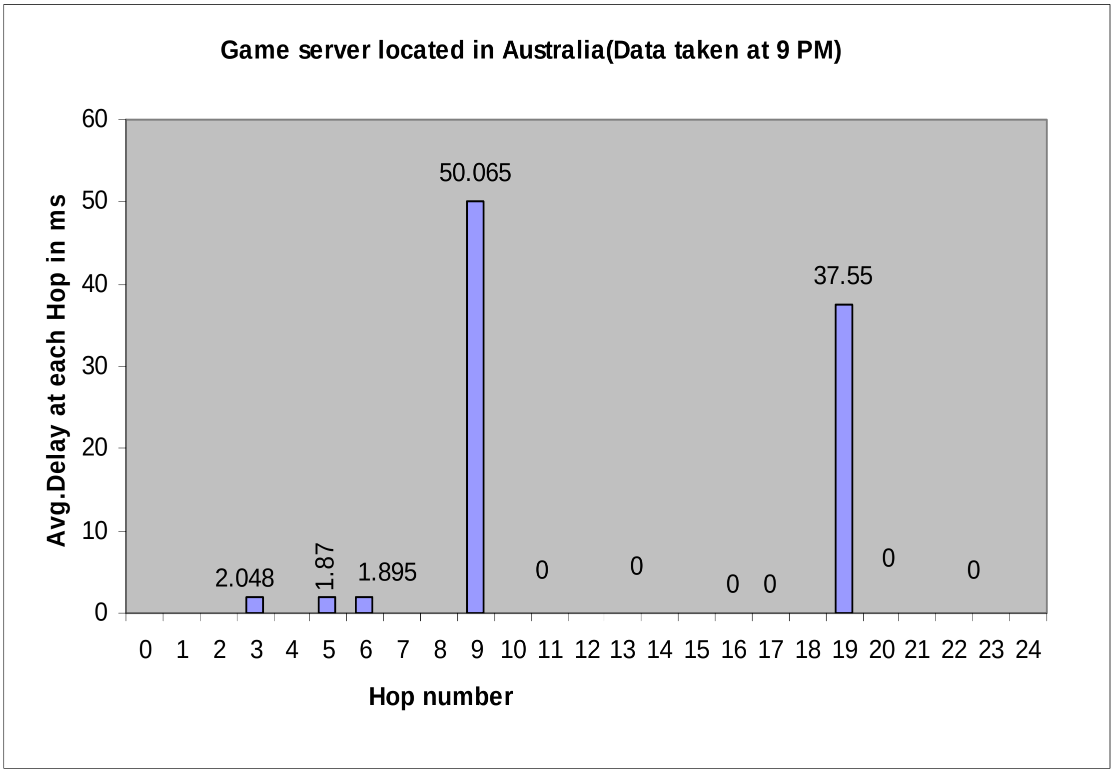


Figure-12

<u>Analysis</u>: 'au.games.yahoo.com' was found in 30 hops. Connections to HTTP port 80 are working.

Average Round Trip Time for this server in the morning =339.7 ms

Average Round Trip Time for this server in the night =338.3636364 ms

Average Round Trip Time for this server in the afternoon =342.5 ms

Overall average Round trip time in day = 340.18 ms

Report for www.rtcw.no [195.139.204.140]

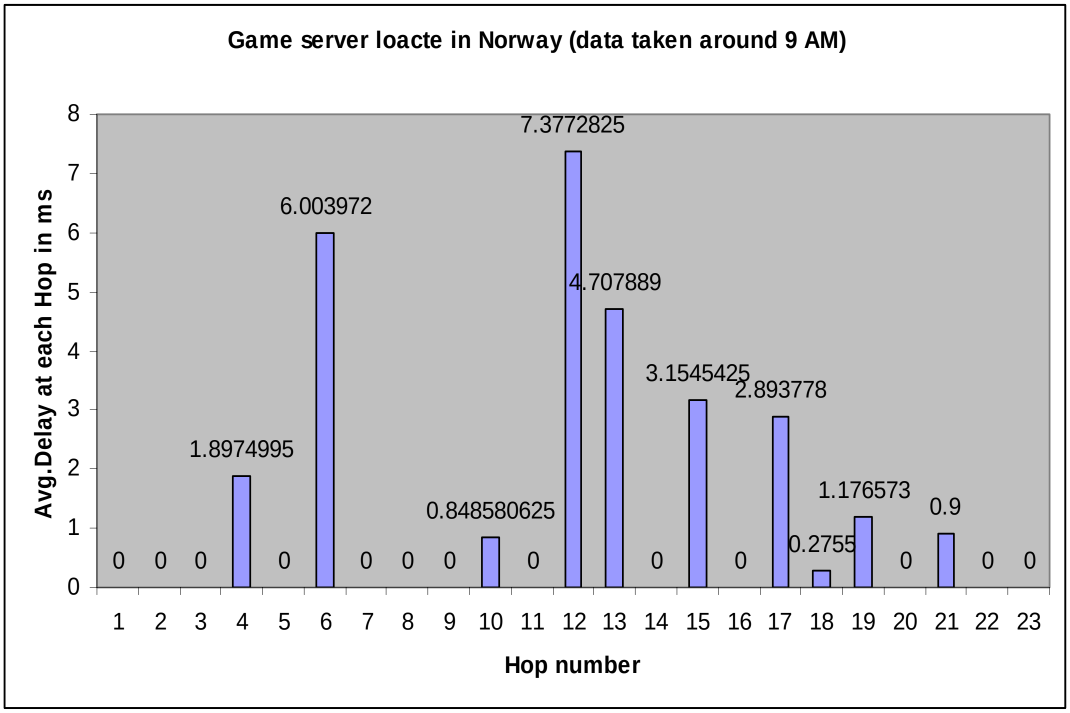


Figure-16

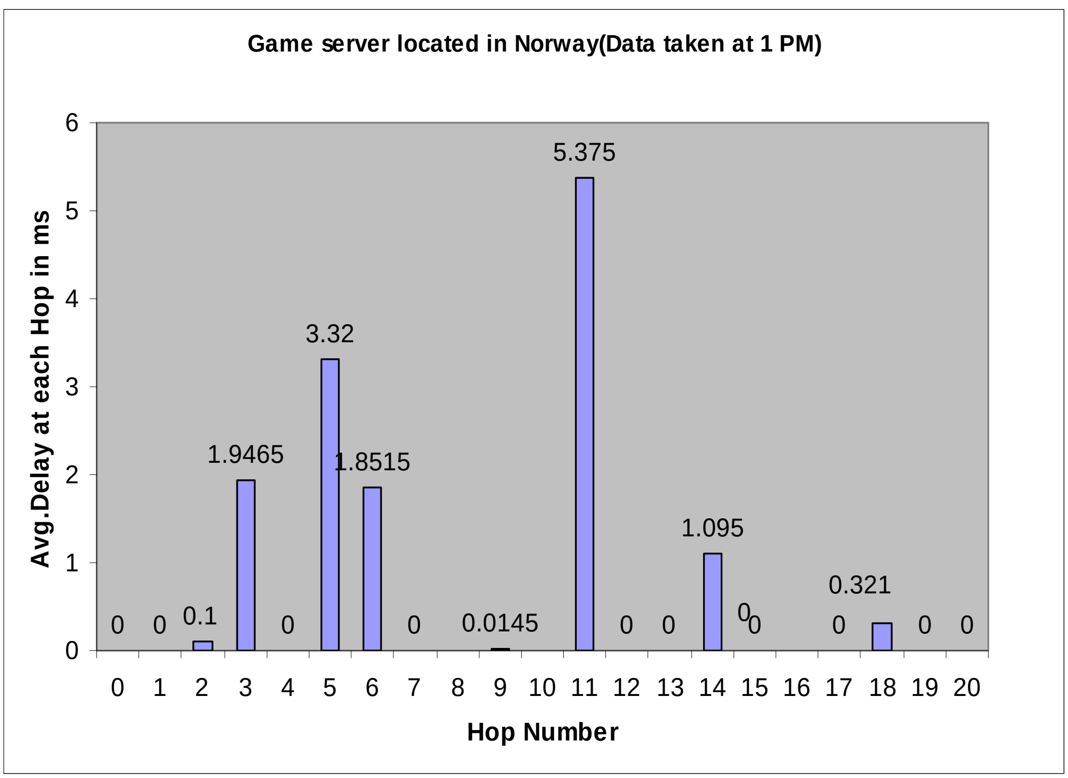


Figure-17

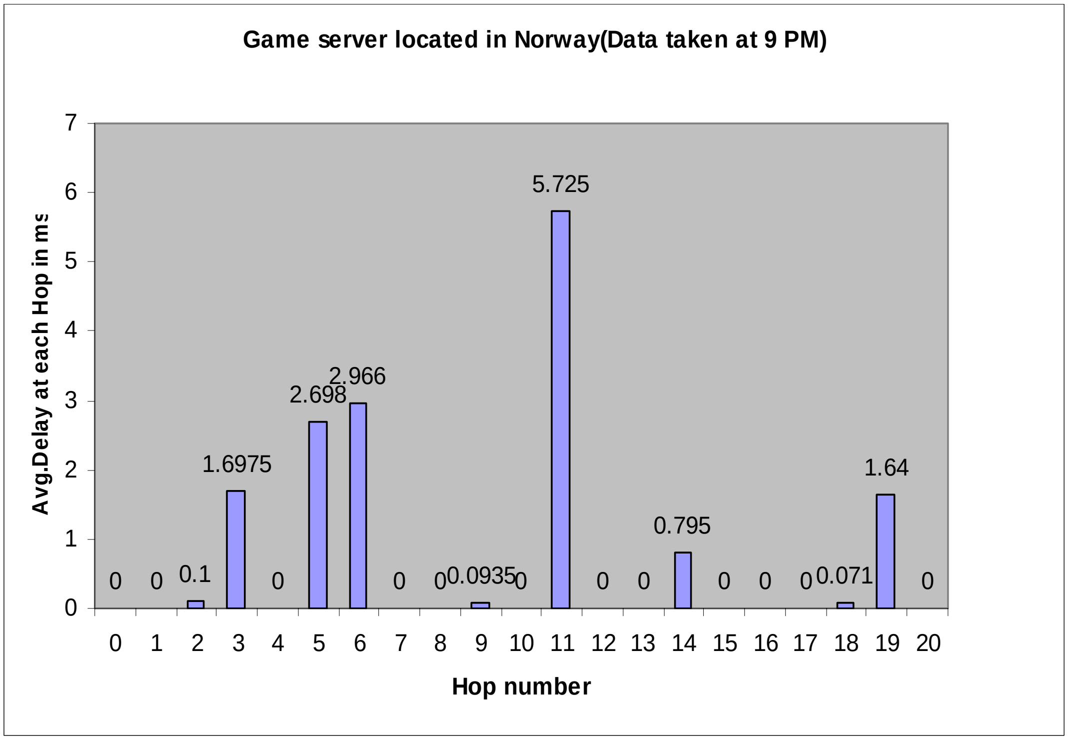


Figure-18

Analysis: 'www.rtcw.no' was found in 20 hops. It is a HTTP server (running Microsoft-IIS/5.0).

Average round trip time for this server in morning = 50.2 ms

Average round trip time for this server in afternoon = 44 ms

Average round trip time for this server in night = 44.1 ms

2.6 Analysis-2

Using PingER (Ping End to end Reporting) is the name given to the Internet End-to-end Performance Measurement (IEPM) project to monitor end-to-end performance of Internet links. The project now involves hundreds of sites in many countries all over the world .Reference:[3] The cell coloring indicates the quality of the performance:

Metric	White	Green	Yellow	Pink	Red
Loss	0-1.0%	1.0-2.5%	2.5-5.0%	5.0-12.0%	>12%
Response	0-62.5ms	62.5-125ms	125-250ms	250-500ms	>500ms
Zero Packet Loss	100-95%	95-85%	85-65%	65-45%	<45%

From Continent To Continent Average Round Trip Time Jun 2004 Submit

WORLD	South Asia	North America	Europe	Russia	East Asia	Oceania	South America
South Asia	42.42	556.75	658.84		418.06		603.51
North America	375.31	53.52	144.82	383.72	244.50	223.61	199.97
Europe	416.10	155.66	42.75	269.68	302.70	353.43	257.22
Africa		703.63	610.36				450.46
Russia		291.76	160.11	219.51	319.28	477.50	1403.00
Latin America		289.03	329.90	583.35	501.35	465.78	70.08
East Asia		248.41	325.32	425.92	107.17	414.54	398.36
Middle East		273.90	273.06	492.42	344.61	390.61	628.07
Balkans		173.63	48.60	277.74	313.08	356.69	
Baltics		171.63	61.79	249.16	324.68	362.39	
Oceania		287.17	340.98	558.50		70.57	
S.E. Asia		264.51					
Central Asia		870.19	813.67				
East Europe		309.54	167.05				
Caucasus		607.72	557.56				
South America		261.01					
.	South Asia	North America	Europe	Russia	East Asia	Oceania	South America

From the above table, the red colored box indicates the average roundtrip time is greater than 500ms.Within the continent the roundtrip time is less compared to outside the continent.

3. Packet Loss

3.1 Motivation :

We have assumed that the queue is capable of holding an infinite number of packets. In reality a queue preceding a link has finite capacity, although the queuing capacity greatly depends on the switch design and cost. Because the queue capacity is finite, packet delays do not really approach infinity as the traffic intensity approaches . Instead, a packet can arrive to find a full queue. With no place to store such a packet, a router will **drop** that packet; that is, the packet will be **lost.** From an end-system viewpoint, this will look like a packet having been transmitted into the network core but never emerging from the network at the destination. The fraction of lost packets increases as the traffic intensity increases. Therefore, performance at a node is often measured not only in terms of delay, but also in terms of the probability of packet loss. A lost packet may be retransmitted on an end-to-end basis, either by the application or by the transport layer protocol.

For getting the best quality characterization for any application we have focused mainly on these packet losses. Based on some observations, above 4-6 % packet loss for video conferencing becomes irritating and become unable to communicate.
According to ref-2, the quality levels for packet losses:
0 – 1% ➔ good
1 – 5% ➔ acceptable
5 – 12% ➔ poor
> 12% ➔ bad.
More recently these levels are refined
0 – 1% ➔ good
1 – 2.5% ➔ acceptable
2.5 – 5% ➔ poor
5 - 12% ➔ very poor
> 12% ➔ bad

Packet loss changes with some parameters.

- Loss rate increase with the nature of the traffic.
- Increasing the buffer, results in reduction of the loss rate.
- Large buffer gives the increase of the queuing delay, increases the roundtrip time.
- According to observations once loss occurs in multimedia applications, consecutive losses will occurs.

Packet loss versus time:-

Higher packet losses were observed during business hours and lower loss rates in the middle of night.

Packet loss versus Packet size:-

Generally, the transmitted IP packet may be reached at destination or may lose. Packets larger than the Ethernet maximum transfer unit (MTU) size of 1500 bytes may be divided into fragments. If any one of fragment lost during transmission, the entire packet is lost. It is therefore expected tat the packets larger than the MTU size will have a higher packet loss rate than packets smaller than the MTU size.

Packet loss versus Buffer capacity:-

More buffer capacity less packet loss but more delay and less buffer capacity more packet loss.

3.2 Analysis-1

The main mechanism used in PingER is Internet control message Protocol (ICMP**).** This allows you to send a packet of a user selected length to a remote node and have it echoed back. Nowadays it usually comes pre-installed on almost all platforms, so there is nothing to install on the clients. The server (i.e. the echo responder) runs at a high priority (e.g. in the kernel on UNIX) and so is more likely to provide a good measure of network performance than a user application. It is very modest in its network bandwidth requirements (~ 100 bits per second per monitoring-remote-host-pair for the way we use it)**.**

The following table gives the result of Packet loss from continent to continent taken in the month of June 2004. Using this PingER tool we can see the packet loss percentage from different country to country. According to ref [3] we can see the reports for packet loss from last 2 years.

From Continent To Continent Packet Loss Jun 2004 Submit

WORLD	South Asia	North America	Europe	Russia	East Asia	Oceania	South America
South Asia	1.38	3.14	2.56		0.00		5.00
North America	3.72	0.27	0.25	1.62	0.18	0.13	1.70
Europe	3.38	0.50	0.41	2.38	0.39	0.34	1.24
Africa		4.95	4.27				4.04
Russia		1.09	1.14	3.54	0.57	1.13	4.58
Latin America		1.56	1.00	3.45	1.11	0.91	2.08
East Asia		0.85	1.00	2.00	0.76	0.81	2.49
Middle East		1.69	1.74	3.29	1.61	1.69	0.00
Balkans		0.28	0.33	2.04	0.26	0.17	
Baltics		0.48	0.62	2.20	0.62	0.62	
Oceania		0.64	1.24	2.56		0.47	
S.E. Asia		4.00					
Central Asia		3.98	3.69				
East Europe		2.41	1.95				
Caucasus		2.01	2.31				
South America		3.10					
.	South Asia	North America	Europe	Russia	East Asia	Oceania	South America

Table : Percentage of packet loss

From the above table, we can say within the continent packet loss is less when compared with out side the continent.

3.3 Analysis-2

According ref -6, experiments conducted using 3 different hosts, DEC station 3100 running Ultrix3.1, DEC station 5000 running Ultrix 4.2, and IBM RT running BSD Unix 4.3, located at Maryland (UMD), Stanford, MIT respectively. All the experiments were conducted between the afternoon of May 29, 1992 and the morning of May 30, 1992. And the results are in the following:-

Packet size: packet size of 32 octets was used.

Interpacket delay: we sent packet from the source process every 39.06 milliseconds.

Number of packets: we send 100,000 packets in ever experiments.

Table1 indicates that there were 2.1% to 10.1% packets losses in different experiments.

Table 2 presents the number of packets with round-trip time of greater than 500ms and 1000ms.

No.	Source/Sink	Echohost	Time (EDT)	Losses	%loss	Duplicates
1	UMD	MIT	2:00 PM	6984	6.984	860
2	UMD	Stanford	3:30 PM	4825	4.825	921
3	Stanford	MIT	5:00 PM	10081	10.081	45
4	UMD	MIT	8:00 PM	3712	3.712	844
5	UMD	Stanford	9:30 PM	2857	2.857	913
6	Stanford	MIT	11:00 PM	4187	4.187	55
7	UMD	MIT	2:00 AM	3067	3.067	993
8	UMD	Stanford	3:30 AM	2164	2.164	1056
9	Stanford	MIT	5:00 AM	3144	3.144	39

Table 1: Losses, Duplicates and Reorders.

Percentage of average packet loss = 4.5578 %

	Roundtriptime in ms				Count	
No	Maximum	Minimum	Average	Standard deviation	>200ms	% >200 ms
1	19	2339	85	69	336	0.336
2	74	898	105	40	244	0.244
3	78	2070	114	56	380	0.38
4	19	1406	40	75	821	0.821
5	78	706	87	17	40	0.04
6	78	1464	98	80	1010	1.01
7	19	1371	42	89	1241	1.241
8	74	671	85	16	25	0.025
9	78	1550	105	118	2247	2.247

Table 2: Roundtrip times.

Percentage of path roudtriptime more than 500ms = 0.7048 %.

4. Analysis using Tcptraceroute

4.1 Motivation:

Tcptraceroute is a traceroute implementation using TCP packets.

The more traditional traceroute sends out either UDP or ICMP ECHO packets with a TTL of one, and increments the TTL until the destination has been reached. By printing the gateways that generate ICMP time exceeded messages along the way, it is able to determine the path packets are taking to reach the destination.

The problem is that with the widespread use of firewalls on the modern Internet, many of the packets that traceroute sends out end up being filtered, making it impossible to completely trace the path to the destination. However, in many cases, these firewalls will permit inbound TCP packets to specific ports that hosts sitting behind the firewall are listening for connections on. By sending out TCP SYN packets instead of UDP or ICMP ECHO packets, tcptraceroute is able to bypass the most common firewall filters.

4.2 tracetcp vs. tracert

Many sites block ICMP pings, which makes the use of the traditional route tracing tools difficult, as can be seen by the following trace, which uses the standard Windows tracert utility:

```
Tracing route to www.ebay.co.uk [66.135.208.41]
over a maximum of 30 hops:
 1    <1 ms    <1 ms    <1 ms  corona.ibr.cs.tu-bs.de [134.169.34.1]
 2    <1 ms    <1 ms    <1 ms  infogate.rz.tu-bs.de [134.169.39.254]
 3    <1 ms    <1 ms    <1 ms  rzrouter.rz.tu-bs.de [134.169.246.14]
 4    <1 ms    <1 ms    <1 ms  ar-braunschweig3-ge4-0-222.g-win.dfn.de [188.1.46.145]
 5     3 ms     3 ms     3 ms  cr-hannover1-po3-0.g-win.dfn.de [188.1.88.65]
 6     3 ms     3 ms     4 ms  ir-hannover2-ge4-0.g-win.dfn.de [188.1.88.62]
 7     2 ms     2 ms     2 ms  188.1.62.2
 8   170 ms   171 ms   171 ms  paix-gw12.SFO.US.net.DTAG.DE [62.154.5.245]

          9     *        *        *     Request timed out.
 10     *        *        *     Request timed out.
 11     *        *        *     Request timed out.
 12     *        *        *     Request timed out.
 13     *        *        *     Request timed out.
 14  ……..continue until maximum number of hops reached.
```

Looking at the above trace it is impossible to determine if an intermediate router (at hop 9) has failed or the probes are being blocked.

By using **tracetcp** we are able to probe all the way to the host machine. Note, however some routers do not report time-to-live exceeded messages as can be by hop #11. From this trace we can see that our packets are reaching the host machine and that it is accepting connections on the specified port (in this case 80).

```
Tracing route to 66.135.192.41 [www.ebay.co.uk] on port 80
Over a maximum of 30 hops.
1  0 ms     0 ms     0 ms     134.169.34.1       [corona.ibr.cs.tu-bs.de]
2  1 ms     0 ms     2 ms     134.169.39.254     [infogate.rz.tu-bs.de]
3  1 ms     1 ms     0 ms     134.169.246.14     [rzrouter.rz.tu-bs.de]
4  2 ms     0 ms     1 ms     188.1.46.145       [ar-braunschweig3-ge4-0-222.g-
win.dfn.de]
5  4 ms     5 ms     4 ms     188.1.88.65        [cr-hannover1-po3-0.g-win.dfn.de]
6  5 ms     4 ms     4 ms     188.1.88.62        [ir-hannover2-ge4-0.g-win.dfn.de]
7  3 ms     2 ms     3 ms     188.1.62.2
8  171 ms   171 ms   171 ms   62.154.5.245       [paix-gw12.SFO.US.net.DTAG.DE]
9  178 ms   178 ms   177 ms   62.159.124.58
10          178 ms   177 ms   176 ms   66.135.207.54
11          *        *        *        Request timed out.
12          *        *        *        Request timed out.
13          Destination Reached in 180 ms.
Connection established to 66.135.192.41
Trace Complete.
```

4.3 Analysis-1

The following results are obtained by running the tcptraceroute continuously for an hour at different times in a day.

Tcptraceroute is open source and modified as per requirements.

Data taken in Night, Afternoon and Morning respectively. Here the Average is equal to the sum of the numbers divided by the number of data in the set.

Location of Server	average RTT around 9AM in ms	average RTT around 13PM in ms	average RTT around 21 PM in ms
Australia	328.8534483	328.9145299	332.8608696
China	349.125	346.9056604	362.2363636
Germany	20.65068493	20.72108844	23.23972603
Norway	42.59016393	42.15447154	46.00833333
UK	25.60843373	31.91666667	29.38888889
USA	176.3313253	176.2349398	180.9207317

Table 2. Average roundtriptime to different server located in different continents

The Result of the Test Program:-

```
OS time:                        13:43:55
OS date:                        09/13/04

Tracing route to 66.218.71.230 [games1.vip.scd.yahoo.com] on port
80
Over a maximum of 30 hops.
1 0 ms  0 ms  0 ms  134.169.34.1      [corona.ibr.cs.tu-bs.de]
2 1 ms  0 ms  0 ms  134.169.39.254    [infogate.rz.tu-bs.de]
3 1 ms  91 ms 0 ms  134.169.246.14    [rzrouter.rz.tu-bs.de]
4 1 ms  0 ms  0 ms  188.1.46.145      [ar-braunschweig3-ge4-0-
222.g-win.dfn.de]
5 5 ms  4 ms  4 ms  188.1.88.65 [cr-hannover1-po3-0.g-win.dfn.de]
6 8 ms  7 ms  8 ms  188.1.18.181      [cr-frankfurt1-po9-3.g-
win.dfn.de]
7 26 ms 9 ms  8 ms  188.1.80.46 [ir-frankfurt2-po4-0.g-win.dfn.de]
8 127 ms      136 ms      127 ms      213.228.221.18
9 172 ms      173 ms      173 ms      216.115.96.34
 [vl140.pat1.pao.yahoo.com]
10      173 ms      173 ms      171 ms      66.218.82.193
 [vl34.bas1-m.scd.yahoo.com]
11      183 ms      175 ms      174 ms      66.218.82.230
 [UNKNOWN-66-218-82-230.yahoo.com]
12      Destination Reached in 173 ms.
Connection established to 66.218.71.230
Trace Complete.

OS time:                        13:44:21
OS date:                        09/13/04

Tracing route to 66.218.71.230 [games1.vip.scd.yahoo.com] on port
80
Over a maximum of 30 hops.
1 1 ms  0 ms  0 ms  134.169.34.1      [corona.ibr.cs.tu-bs.de]
```

```
2 1 ms  0 ms  0 ms  134.169.39.254    [infogate.rz.tu-bs.de]
3 1 ms  10 ms 1 ms  134.169.246.14    [rzrouter.rz.tu-bs.de]
4 1 ms  1 ms  0 ms  188.1.46.145      [ar-braunschweig3-ge4-0-
222.g-win.dfn.de]
5 5 ms  63 ms 4 ms  188.1.88.65 [cr-hannover1-po3-0.g-win.dfn.de]
6 8 ms  7 ms  7 ms  188.1.18.181      [cr-frankfurt1-po9-3.g-
win.dfn.de]
7 10 ms 8 ms  8 ms  188.1.80.46 [ir-frankfurt2-po4-0.g-win.dfn.de]
8 128 ms      127 ms      127 ms      213.228.221.18
9 173 ms      173 ms      172 ms      216.115.96.34
 [vl140.pat1.pao.yahoo.com]
10      173 ms      173 ms      171 ms      66.218.82.201
 [vl36.bas2-m.scd.yahoo.com]
11      173 ms      174 ms      173 ms      66.218.82.234
 [vl44.bas1-m.scd.yahoo.com]
12      Destination Reached in 172 ms.
Connection established to 66.218.71.230
Trace Complete.
OS time:                        13:44:42
OS date:                        09/13/04

Tracing route to 66.218.71.230 [games1.vip.scd.yahoo.com] on port
80
Over a maximum of 30 hops.
1 1 ms  0 ms  0 ms  134.169.34.1      [corona.ibr.cs.tu-bs.de]
2 1 ms  0 ms  0 ms  134.169.39.254    [infogate.rz.tu-bs.de]
3 1 ms  1 ms  1 ms  134.169.246.14    [rzrouter.rz.tu-bs.de]
4 1 ms  1 ms  0 ms  188.1.46.145      [ar-braunschweig3-ge4-0-
222.g-win.dfn.de]
5 4 ms  4 ms  4 ms  188.1.88.65 [cr-hannover1-po3-0.g-win.dfn.de]
6 8 ms  108 ms      7 ms  188.1.18.181      [cr-frankfurt1-po9-3.g-
win.dfn.de]
7 9 ms  8 ms  14 ms 188.1.80.46 [ir-frankfurt2-po4-0.g-win.dfn.de]
8 128 ms      128 ms      127 ms      213.228.221.18
9 172 ms      172 ms      171 ms      216.115.96.34
 [vl140.pat1.pao.yahoo.com]
10      172 ms      171 ms      172 ms      66.218.82.193
 [vl34.bas1-m.scd.yahoo.com]
11      174 ms      196 ms      175 ms      66.218.82.234
 [vl44.bas1-m.scd.yahoo.com]
12      Destination Reached in 173 ms. Connection established to
66.218.71.230 Trace Complete.

OS time:                        13:45:04
OS date:                        09/13/04

Tracing route to 66.218.71.230 [games1.vip.scd.yahoo.com] on port
80
Over a maximum of 30 hops.
1 0 ms  0 ms  0 ms  134.169.34.1      [corona.ibr.cs.tu-bs.de]
2 1 ms  1 ms  0 ms  134.169.39.254    [infogate.rz.tu-bs.de]
3 4 ms  0 ms  1 ms  134.169.246.14    [rzrouter.rz.tu-bs.de]
4 2 ms  1 ms  0 ms  188.1.46.145      [ar-braunschweig3-ge4-0-
222.g-win.dfn.de]
5 6 ms  5 ms  4 ms  188.1.88.65 [cr-hannover1-po3-0.g-win.dfn.de]
```

```
6 8 ms  7 ms  7 ms  188.1.18.181      [cr-frankfurt1-po9-3.g-
win.dfn.de]
7 9 ms  9 ms  8 ms  188.1.80.46 [ir-frankfurt2-po4-0.g-win.dfn.de]
8 304 ms      238 ms      127 ms      213.228.221.18
9 173 ms      173 ms      172 ms      216.115.96.34
 [vl140.pat1.pao.yahoo.com]
10      172 ms      174 ms      177 ms      66.218.82.201
 [vl36.bas2-m.scd.yahoo.com]
11      173 ms      173 ms      174 ms      66.218.82.238
 [UNKNOWN-66-218-82-238.yahoo.com]
12      Destination Reached in 171 ms.Connection established to
66.218.71.230
Trace Complete.
```

```
OS time:                        13:45:25
OS date:                        09/13/04

Tracing route to 66.218.71.230 [games1.vip.scd.yahoo.com] on port
80
Over a maximum of 30 hops.
1 0 ms  0 ms  0 ms  134.169.34.1      [corona.ibr.cs.tu-bs.de]
2 1 ms  0 ms  0 ms  134.169.39.254    [infogate.rz.tu-bs.de]
3 86 ms 1 ms  1 ms  134.169.246.14    [rzrouter.rz.tu-bs.de]
4 1 ms  1 ms  1 ms  188.1.46.145      [ar-braunschweig3-ge4-0-
222.g-win.dfn.de]
5 4 ms  45 ms 4 ms  188.1.88.65 [cr-hannover1-po3-0.g-win.dfn.de]
6 7 ms  15 ms 8 ms  188.1.18.181      [cr-frankfurt1-po9-3.g-
win.dfn.de]
7 9 ms  9 ms  29 ms 188.1.80.46 [ir-frankfurt2-po4-0.g-win.dfn.de]
8 128 ms      128 ms      127 ms      213.228.221.18
9 172 ms      173 ms      172 ms      216.115.96.34
 [vl140.pat1.pao.yahoo.com]
10      172 ms      179 ms      173 ms      66.218.64.150
 [vl17.bas2.scd.yahoo.com]
11      173 ms      173 ms      179 ms      66.218.82.226
 [vl42.bas1-m.scd.yahoo.com]
12      Destination Reached in 172 ms.Connection established to
66.218.71.230
Trace Complete.
```

.... Continued till 2 hours.

4.4 Data Collection and Analysis:-

By creating the test program for the Tcptraceroure, data can be taken and graphs are drawn. Recent measurements indicate that the number of hosts in the internet is fast approaching. Clearly, it is impossible to study the delay characteristics for all connections, i.e. for all source-destination pairs. In this report we examine specific connections in details. All the results are taken by running the test program three different times in a day. They are morning around 9 AM, afternoon i.e. after lunch around 1 PM and night i.e. after dinner around 8 PM. Generally all the game servers have heavy traffic in the night time i.e. after dinner because many users play games that time. During the heavy traffic there may be lot of delays and packet losses. So we preferred this time for getting the better results.

For collecting the data we have taken three game servers i.e. *http://games.yahoo.com* and *http://ca.games.yahoo.com* located in USA and *www.kidsdomain.com* located in Canada. These results are taken by sending the tcp packets to the game servers which are located in USA and Canada from the Technical University of Braunschweig, Germany.

We assumed each tcptraceroute as one packet. Using the program we stored all the roundtriptimes to each sever in an array. Now the array contains the set of data of roundtriptimes. By importing these data into the excel sheets we presented the descriptive statistics analysis on that data set. These results are obtained by running the program continuously for as mentioned earlier.

Analysis on the collected data can be calculated in the following way:

1) Importing data to Excel sheet.
2) Then go to Excel menu bar ->Tools ->Add-Ins -> Analysis Tool Park ->OK.
   We can get Data Analysis option under the Tools menu bar.
3) Then Excel Menu Bar -> Tools -> Data Analysis.

This will take all the data as input then it will give the all the statistics about that data.

The formulas for the statistics:-

Mean: The mean is also called the Average. The mean is equal to the sum of the numbers divided by the number of data in the set

Mode: The number(s) that appears most frequently in a set.

Median: The middle number of a set of data when it is ordered least to greatest.

Range: The difference between the highest and lowest number in the set.

Standard Deviation: The **standard deviation** tells you how tightly all the various examples are clustered around the mean in a set of data.The standard deviation is the square root of the average squared deviation from the mean.

Suppose we are given a dataset $x_1,...,x_N$ of values (which are real numbers). The mean of this population is defined as

$$Mean\ \bar{x} = \frac{1}{N}\sum_{i=1}^{N} x_i$$

$$S\tan dardDeviation(\sigma) = \sqrt{\frac{1}{N}\sum_{i=1}^{N}(x_i - \bar{x})^2}$$

Simple Variance: Variance is the square of standard deviation.

To calculate the roundtriptime to each server, I took Mode value on the data set as the roudtriptime value because mode is defined as the number that appear most frequently in the set.

Packet number can be taken in x-axis and roundtriptime to each packet in y-axis.

Graphs for the Packet roundtriptime for http://games.yahoo.com :

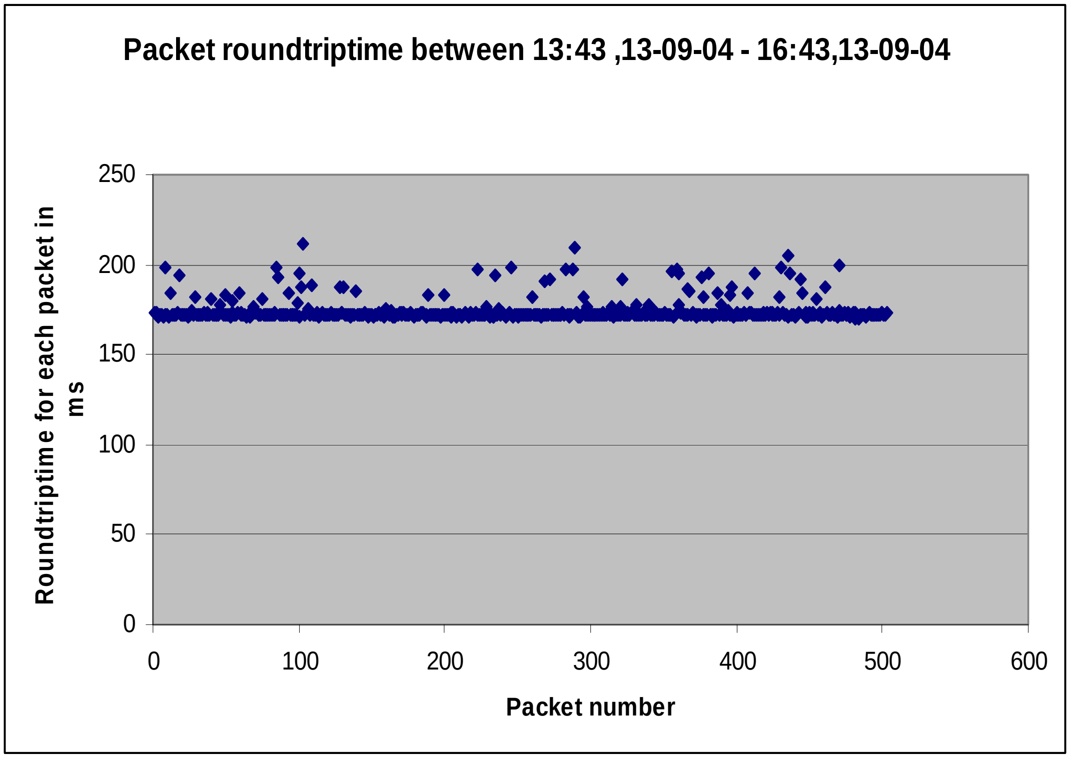


figure: packet roundtriptime between 13:43, 13-09-04 – 16:43,13-09-04 for the http://games.yahoo.com server

Analysis :

Mean	174.169
Standard Error	0.274444
Median	172
Mode	172
Standard Deviation	6.155138
Sample Variance	37.88573
Range	42
Minimum	170
Maximum	212

From the above statistics on data, the Roundtriptime to http://games.yahoo.com from Technical university of Braunschweig, Germany is 172 ms. But as per our assumption if the packet roundtriptime is more than 200 ms, we considered as obsolete packet and we are calculating that percentage.

Number of Packet sent = 503

Number of packet roundtriptime more than 200 ms = 4

Percentage of packets roundtriptime more than 200 ms = 0.8 %

Graphs for the Packet roundtriptime for http://games.yahoo.com

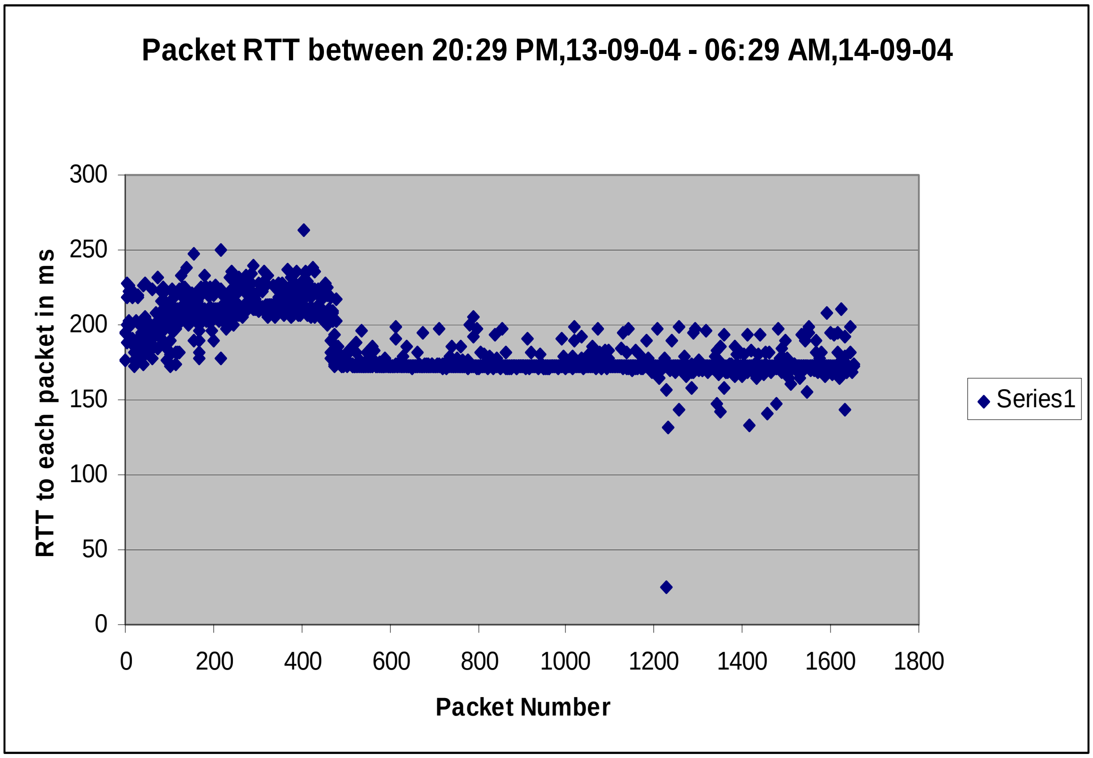


figure : Roundtriptime between 20:29PM,13-09-04 – 6:29 AM ,14-09-04

Analysis:-

Mean	183.9073289
Median	173
Mode	172
Standard Deviation	19.45251906
Sample Variance	378.4004978
Range	238
Minimum	25
Maximum	263

From the above statistics on the collected data, the Roundtriptime to http://games.yahoo.com from Technical university of Braunschweig, Germany is 172 ms.

Number of Packet sent = 1668

Number of packet roundtriptime more than 200 ms = 399

Percentage of packets roundtriptime more than 200 ms = 24.46 %

Report for the Packet roundtriptime for http://games.yahoo.com

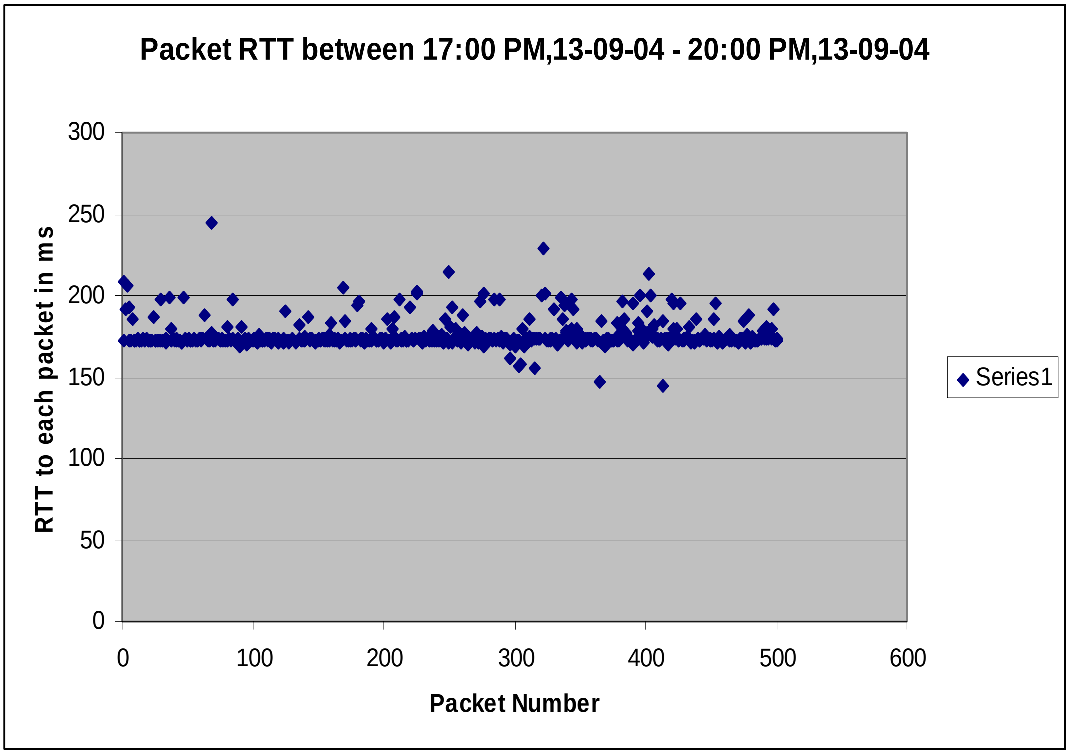


figure: packet roundtriptime between 17:00 PM, 13-09-04 – 20:00 PM,13-09-04 for the http://games.yahoo.com server

Analysis:

Mean	175.4630739
Median	173
Mode	172
Standard Deviation	8.790058801
Sample Variance	77.26513373
Range	100
Minimum	145
Maximum	245
Sum	87907
Count	501

Number of Packet sent = 501

Number of packet roundtriptime more than 200 ms = 14

Percentage of packets roundtriptime more than 200 ms = 2.98 %

Report for the Packet roundtriptime for http://ca.games.yahoo.com

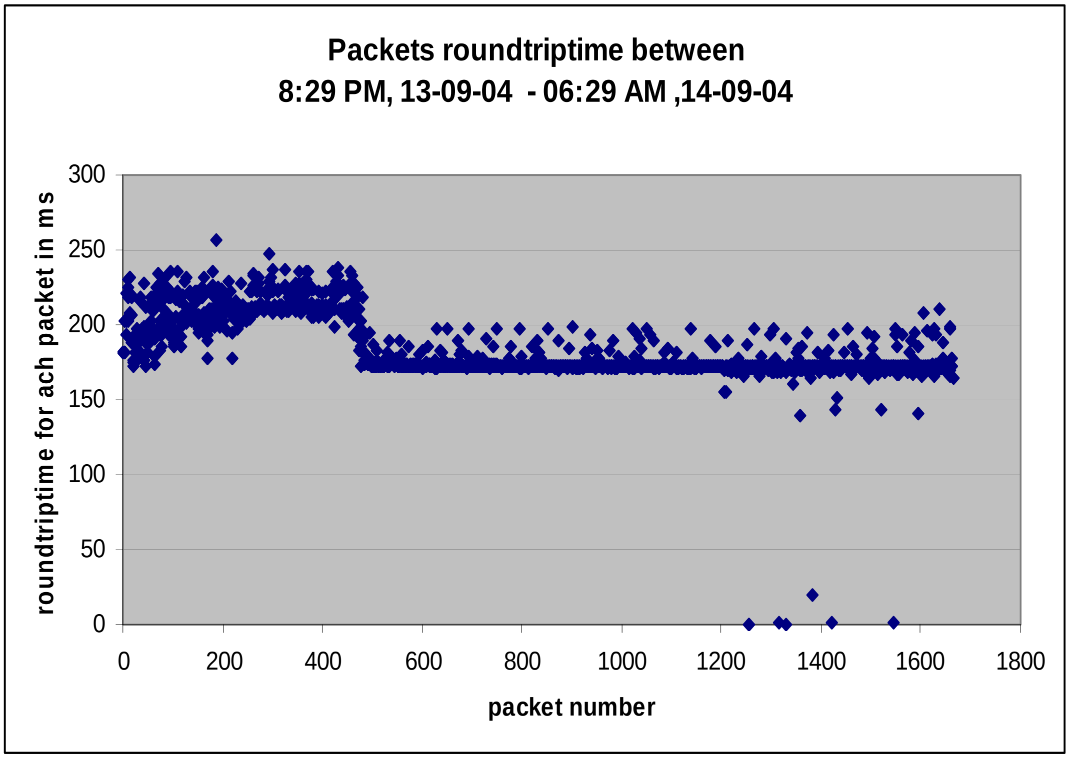


figure: packet roundtriptime between 8:29 PM, 13-09-04 – 6:29,14-09-04 for the http://ca.games.yahoo.com server

Analysis :

Mean	184.1053907
Median	173
Mode	172
Standard Deviation	19.27972183
Sample Variance	371.707674
Range	237
Minimum	20
Maximum	257

From the above statistics on the collected data, the Roundtriptime to http://ca.games.yahoo.com from Technical university of Braunschweig, Germany is 172 ms.

Number of Packet sent = 1657

Number of packet roundtriptime more than 200 ms = 393

Percentage of packets roundtriptime more than 200 ms = 24.03 %

Report for the Packet roundtriptime for http://ca.games.yahoo.com

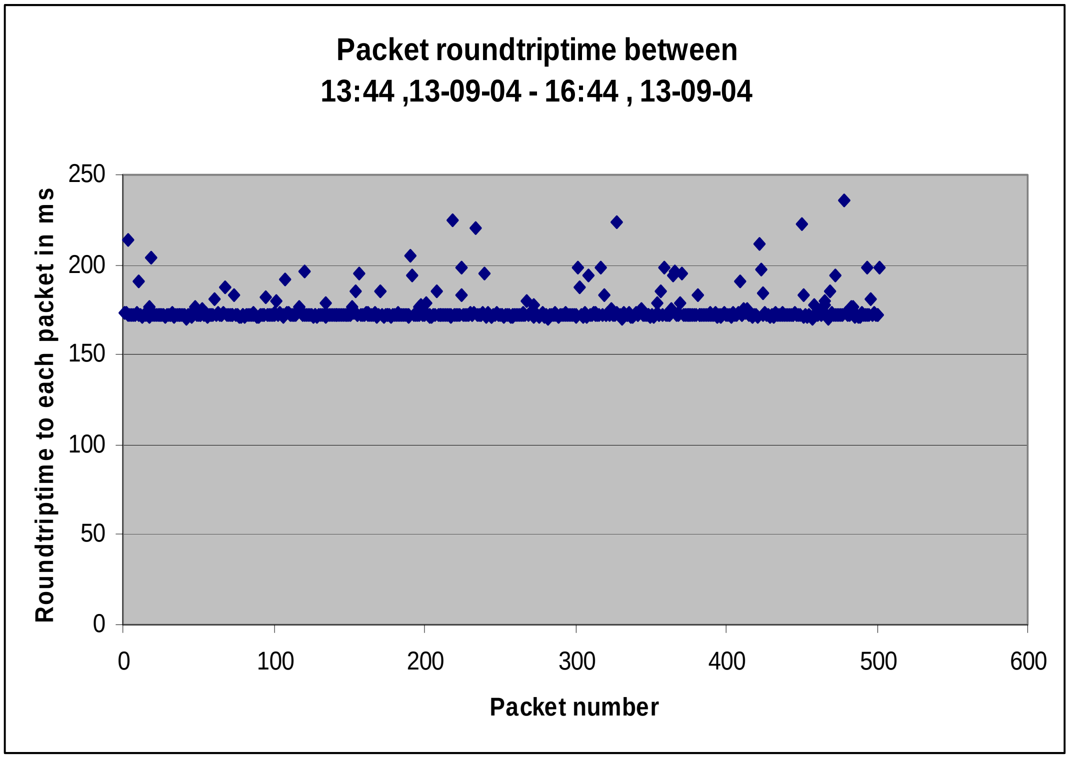


figure: packet roundtriptime between 13:44, 13-09-04 – 16:22,13-09-04 for the http://ca.games.yahoo.com server

Analysis :

Mean	174.3466135
Median	172
Mode	172
Standard Deviation	7.880303302
Sample Variance	62.09918013
Range	66
Minimum	170
Maximum	236

Number of Packet sent = 502

Number of packet roundtriptime more than 200 ms = 15

Percentage of packets roundtriptime more than 200 ms = 2.98%

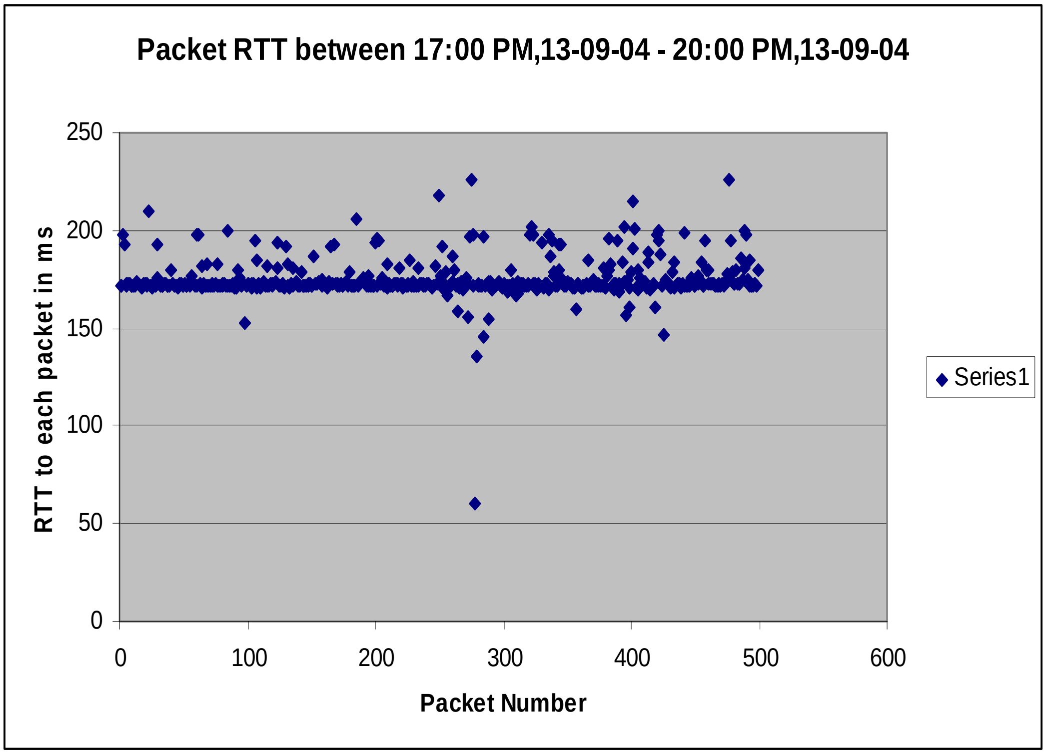


figure: packet roundtriptime between 17:00 PM, 13-09-04 – 20:00 PM,13-09-04 for the http://ca.games.yahoo.com server

Analysis :

Mean	175.1583166
Median	173
Mode	172
Standard Deviation	10.34121603
Sample Variance	106.940749
Range	166
Minimum	60
Maximum	226

Number of Packet sent = 499

Number of packet roundtriptime more than 200 ms = 12

Percentage of packets roundtriptime more than 200 ms = 2.40%

Report for the Packet roundtriptime for http://kidsdomain..com

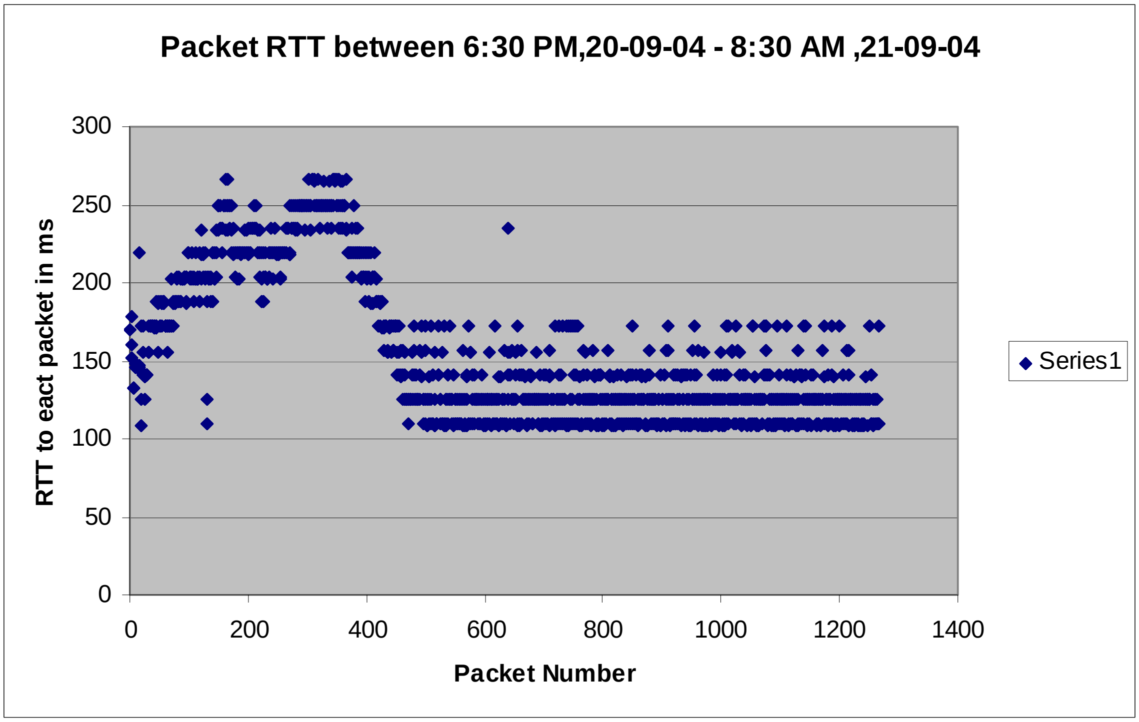


figure: packet roundtriptime between 6:30 PM,20-09-04 to 8:30AM,21-09-04 for the www.kidsdomain.com server

Analysis :

Mean	156.2383583
Standard Error	1.331597817
Median	140
Mode	125
Standard Deviation	47.39814902
Sample Variance	2246.584531
Range	157
Minimum	109
Maximum	266
Sum	197954
Count	1267

Number of Packet sent = 1267

Number of packet roundtriptime more than 200 ms = 321

Percentage of packets roundtriptime more than 200 ms = 25.33 %

4.5 Results :

Game Server Name	Time	No. packets sent	No. of packets RTT >=200 ms	Percentage of packets RTT >=200
games.yahoo.com (Located in USA)	13.09.04 20:29-14.09.04 06.29	1668	408	24.26%
games.yahoo.com	13.09.04 13:43 - 13.09.04 16:43	500	4	0.80%
games.yahoo.com	13.09.04 17:06 - 13.09.04 20.06	503	15	2.98%
games.yahoo.com	20-09-04 18:30-21-09-04 8:00 AM	2338	917	39.22%
games.yahoo.com	21-09-04 18:00-22-09-04 8:00 AM	2301	918	39.89%
games.yahoo.com	14.12.04 9:00 – 12:30	703	0	0
ca.games.yahoo.com (Located in USA)	13.09.04 20:29 - 14.09.04 06.29	1664	400	24.03%
ca.games.yahoo.com	13.09.04 13:43 - 13.09.04 16:43	509	9	1.07%
ca.games.yahoo.com	13.09.04 17:06 - 13.09.04 20.06	499	12	2.40%
ca.games.yahoo.com	20-09-04 18:30-21-09-04 8:00 AM	1267	321	25.22%
ca.games.yahoo.com	21-09-04 18:00-22-09-04 8:00 AM	2297	923	40.18%
www.kidsdomai.com (located in Canada)	20-09-04 18:30-21-09-04 8:00 AM	1267	321	25.33%
www.kidsdomai.com	21-09-04 18:00-22-09-04 8:00 AM	1265	340	26.87%
www.kidsdomain.co m	14.12.04 8:51 AM-14.12.04 12:30 AM	357	0	0

table 3: overall results from tcptraceroute

If we observe the above results , in the night times more than 20 % of packets are taking more than 200 ms roundtriptime to the game servers located in USA and Canada.

5. Conclusion

In this report we have presented the results of the average roundtriptime for the game servers in order to check whether the overall roundtriptime is more than 200 ms or not. The packet roundtriptime more than 200ms or one way delay more than 100ms are varied based on location of servers, traffic, and packet size etc factors. For example. Within Europe this two factor is less percentage compared with the servers located in other countries when traffic is less. Data can be collected by running the test tcptraceroute between Germany and USA. By searching different literature we have presented some analysis on packet loss in the internet also.

According to my results the percentage of packets roundtriptime more than 200 ms is more than 20 % in the night times that means after. The game severs have heavy traffic in that times. But in morning and afternoon this percentage is very less. The packets which are having delayed more then 200 ms are called obsolete packets. Because of these obsolete packets the user gets lot of incontinence. We have to reduce these obsolete packets by explicitly control the delay. To do this we have to control the packet lifetime explicitly in the IP packet.

Delay and packet loss are very important factors to provide best quality of service to interactive multimedia applications like video-conferencing, online music, movie and games etc. Finally the results for overall percentage of packet roundtriptime more than 200 ms for different game servers are calculated and graphs are drawn.

6. References

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