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Testing, Detection and Possible Solutions for the Bufferbloat Phenomenon on Networks.

Thesis Degree

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Definición de Tema de Memoria
para optar al Título de:
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1 Introduction

If a little salt makes food taste better, then a lot must make it taste great, right?. What happens if you apply the same statement to a network domain? It keeps been as good as it was? It improves the performance or makes it worse?.

Lets think of a network as a road system where everyone drives at the maximum speed. When the road gets full, there are only two choices: crash into other cars, or get off the road and wait until things get better. The former isn't as disastrous on a network as it would be in real life: losing packets in the middle of a communication session isn't a big deal. But making a packet wait for a short time is usually better than "dropping" it and having to wait for a re-transmission.[29]

At this point, the role of the router becomes important. It has to control the congestion effectively in networks. It is important to remember that the traffic in a network is inherently bursty, so the role of the buffers in the router is to smooth the flow of traffic. Without any buffering, to allocate the bandwidth evenly would be impossible. But there are some problems with current algorithms; they use tail-drop based queue management that has two big drawbacks: 1.- lockout 2.- full queue that impact with a high queue delay.

These problems are fixed with the creation of a group of FIFO based queue management mechanisms to support end-to-end congestion control in the internet. That procedure is called Active Queue Management (AQM). With AQM the loss of package and the average queue length is reduced; this impacts in a decreasing end-to-end delay by drooping packages before buffer comes full, using the exponential weighted average queue length as a congestion indicator. For the proper use of AQM, it has to be widely enabled and consistently configured the router.

Today's networks are suffering from unnecessary latency and poor system performance. The culprit is Bufferbloat, the existence of excessively large and frequently full buffers inside the network. Large buffers have been inserted all over the Internet without sufficient thought or testing. They damage or defeat the fundamental congestion-avoidance algorithms of the Internet's most common transport protocol. Long delays from bufferbloat are frequently attributed incorrectly to network congestion, and this misinterpretation of the problem leads to the wrong solutions being proposed.[6]

The existence of cheap memory and a misguided desire to avoid packet loss has led to larger and larger buffers being deployed in the hosts, routers, and switches that make up the Internet. It turns out that this is a recipe for bufferbloat. Evidence of bufferbloat has been accumulating over the past decade, but its existence has not yet become a widespread cause for concern.

2 The Transport Control Protocol

The Transport Control Protocol or TCP is main transport protocol of the internet, providing reliable host to host communication over unreliable transport media[15]. The advantages of its connectionless design, flexibility and robustness provides reliable, ordered delivery of a stream of bytes from a program on one computer to another program on another computer, but for that, the cost are: the needed of careful design so it provides a good service under heavy loads. Many modifications has been done since it was originally defined in 1981.

The root idea of any algorithm related to transport connections must be based into the "*packet conservation principle*". This principle claims:

Definition 2.1. A new packet isn't put into the network until an old packet leaves.

2.1 Fundamentals of TCP

TCP operates basically making two hosts exchange segments of data. The connection between these two hosts is identified uniquely at each end by the host's network address and a 16 bit port number. The communication between the hosts is initiated by a three way handshake between them. This is where the sequence number is synchronized between the participants. The sequence number is a 32-bit number and it is the basis for a reliable data transport, working as a mechanism to assure the exchange of data through an unreliable network medium counting the bytes transmitted making possible be checked by the receiver and be acknowledged. This is, starting from the initial sequence number, each data byte sent as part of the connection has a corresponding sequence number; and only after having being acknowledged by the receiver is the data considered to be transmitted successfully.

TCP made use of the idea of pipe size and the knowledge there was reasonable but not excessive buffering along the data path to send a window of packets at a time. To

control the amount of data that flows through the network path, the receiver sent the information of how much data can receive at once, so the network resource is utilized in a more efficiently way. This window of data means how many bytes the receiver can work with, given the available buffering and processing constraints. That is, window size represents how much data a device can handle from its peer at one time before it is passed to the application process. All the extra data in excess will be dropped. This window is also a constrain to the sender, because the sender is not allowed to transmits no more data than the window before waiting for the response of the host as acknowledgment of the sent data.

To manage the data sent and the acknowledgment received for those packets, TCP uses a cumulative scheme. After a packet is in flow from the server with its corresponding sequence number, sent data goes to a retransmission queue where it is held until the corresponding acknowledge from the other end has come in or, to be resent if not acknowledgment within a timeout. When the acknowledgment of a sequence number is received, the sender discard all data with sequence numbers bellow the sequence number in the acknowledgment that has arrive. For retransmission, TCP uses an adaptive scheme. The timeout is automatically set from the measured round trip time of the connection, taking into account the variance of the measured values[10]. This helps avoid retransmitting potentially lost segments too quickly or too slowly.

Because today's networks are dynamic and in different configurations, both topologically speaking as a bandwidth, TCP must handle these changes and still be able to maintain communication between the two hosts. The lost of communications can be an issue that TCP needs to take into account. In case of lost of one packet means subsequent packets cannot be acknowledged until the lost packet is retransmitted. This can lad to excessive retransmission and unnecessary load. TCP extension has been developed that allows the receiver to send selective acknowledgments of block of received

data with sequence numbers that are not cumulative with the data acknowledged in the traditional way[13].

Since networks are shared and conditions change along the path, the algorithms continually probe the network and adapt the number of packets in flight. It is not hard also (but it is often the case) to find along the way decreases in the bandwidth. This spots are the bottlenecks and they are important because the performance or capacity of the entire connection (connection as a state between the two hosts) is limited by the resources that this trace has. With this, controlling the optimal rate of data transmission is a hard work, and the receiver window as communicated by TCP is not a necessarily a very good indicator. To fix this issue, TCP received an addition to its specification: *congestion window*. The congestion window plays a crucial role in estimating the available bandwidth between the hosts. After this modification, the minimum between the receiver window and the congestion window is used as the transmit limit. All the TCP additions attempt to keep the network operating near the inflection point where throughput is maximized, delay is minimized, and little loss occurs.

2.2 TCP's Phases

To provide a good service, it's needed that TCP flows respond to orders given by the hosts machines that controls the connection during congestion. This characteristic of flows is called "*responsiveness*" and tells when flows must back-off during congestion. In the second half of the 80's, TCP began to experience a strange phenomenon that manifested itself through a dramatically diminish throughput. After a deep analysis, Van Jacobson showed that TCP needed a mechanism to limit transmission speed in the face of congestion[10]. This lead to the development of a better congestion control mechanism for TCP, which was added as a requirement for hosts connected to Internet[2]. Since then, the congestion controls methods implemented in TCP has been updated

several times, adding two new algorithms, the *slow start* and *congestion avoidance* were designed to keep in "equilibrium" the data that is in exchange. Later a third where added, the *fast retransmit and fast recovery.*

2.2.1 Slow start and Congestion avoidance

The idea behind bandwidth occupancy in TCP is to use as much as it is allow to use. With this in mind, slow start proves the network by increasing exponentially the rate of how many packets sends; the congestion window is increases by one packet and adds another for each packet acknowledged. This behavior continues until the slow start threshold (ssthrsh) is reached or congestion is signaled. If congestion is detected, the slow start threshold is reset to half of the amount of congestions window, at the time that congestion window is reset to a size of one segment, and slow start is restarted. In case that the slow start threshold is reached, the sender switches to congestion avoidance mode, which is employed to maintain the transmission rate, increasing by one segment each RTT[ii].

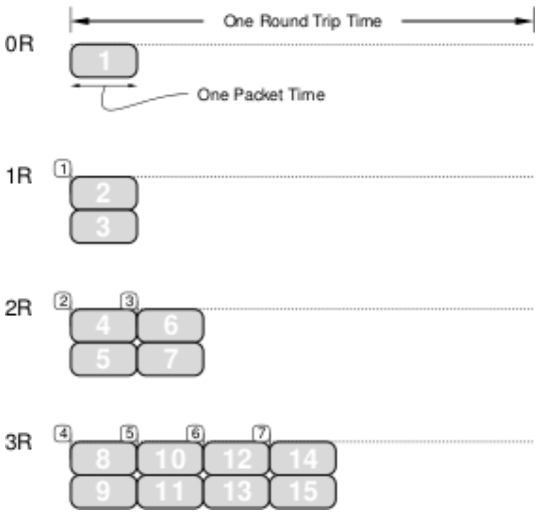

Figure 1: The Chronology of a Slow-Start.[10]

TCP must be smart enough to signal to the hosts when path is experiencing congestion, so it must have policies that decrease the network utilization or increase it if congestion signals are received or not. Also, based on the afore mentioned principle ("*packet conservation principle*"), congestion collapse would become the exception

[ii]Each ack increases the congestion window by $ss * ss/cwnd$. This results in a linear increase of the congestion window; that means in 1 RTT, congestion window will be increased by 1 [10] [16] [4]

rather than a rule. Thus congestion control involves finding place that violate conservation and fixing them. It is important detect early this misbehavior, since new packets are added exponentially also the congestion grows exponentially, so if is detected early, only small adjustments will cure it.

Typically, when the sender is signaled that packets are been dropped, TCP/IP networks understand that as congestion, but typical effects include queuing delay, packet loss or the blocking of new connections. New implementations define a explicit congestion notification methods [17], that allows notification of congestion between the ends without any packet loss. TCP specification mandates initially setting the congestion window to between 2 and 4 segments of data depending on the segment size[1].

2.2.2 Congestion control algorithms

One of the biggest problems that TCP had in the past years for its development has been the achieve of optimal utilization preventing congestion associated with the differences of bandwidth existing in the medium through which communication is propagated. That is why lately there have been various implementations for proper handling of the original algorithm originally deployed in TCP for attaining prevent or lessen the effects of this problem. Among the algorithms that currently exist, the two predominant in modern operating systems will be mentioned: CUBIC and Compound TCP.

2.2.2.1 CUBIC TCP is the default implementation for congestion control in the Linux kernel since 2.6.19. *Scaling the window growth rate to match large BDP*[iii] *is rather straightforward, tackling the fairness issues of new protocols has remained as a major challenge. Although BDP implies the network capacity by packet count(or window size), packet count is not adequate to characterize TCP performance because growth rate of TCP depends on RTT*[8].

[iii] View definition in appendix A

Because the basis of their growth is a cubic function (hence its name), has a less aggressive behavior than their predecessors, while retaining excellent scalability, fairness and stability. As can be seen in figure 2, the cubic function is compound of three parts: a concave part where the window grows rapidly over small time values, a plateau set at the previous maximal congestion window size, and the end portion where the increasing bandwidth above the maximum which exhibits a convex shape.

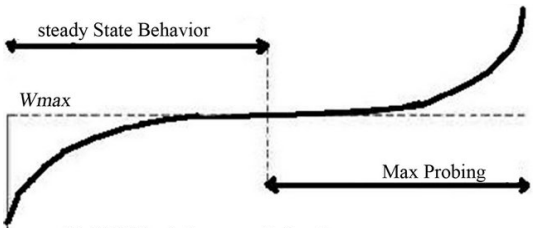

Figure 2: The window growth function of CUBIC

$$W_{cubic} = C(t - K)^3 + W_{max}$$

Figure 3: CUBIC's Congestion Window.[iv]

As can be better seen in the formula 3, the congestion window ain't dependent of its previous acknowledged packets; instead, the congestion window is computed at each step from a calibrated function. With this calibration, the plateau is at the previous maximal congestion window size. This feature allow the algorithm to respond faster to changes in available bandwidth. Since in previous algorithms throughput is defined by the packet loss rate as well as RTT, the throughput in CUBIC is defined by only the packet loss rate.

Another feature that CUBIC has with respect to its predecessor in the Linux kernel, it also changes the behavior of the Slow Start algorithm, replacing it with one called

[iv] C is a scaling factor, textitt is the elapsed time from the last window reduction. W_{max} is the window size before the last window reduction. $K = sqrt[3]W_{max}beta/C$. And $beta$ is a constant multiplication decrease factor applied for window reduction at the time of loss event

Hybrid Slow Start (or HyStart)[7]. The idea behind this algorithm is to prevent the extra drop of packets caused by the overshoot of bandwidth done by the regular slow start. This extra amount of data cause unnecessary on both ends hosts and network, and taking long time to recover from, that can be worst if the packet drop is made at a late stage of the slow start[v].

2.2.2.2 Compound TCP

under dev [24] and [25]

2.2.3 Fast retransmit and fast recovery

Timers are an important part of this self-regulated protocol. TCP maintains four timers[vi]. One of them, the retransmission timer, causes the sender to retransmit the segment over which at the expiry of this timer(which is a function of the estimated RTT), has not yet received their respective acknowledge, assuming the segment is lost in the network. When this action is unleashed, the server will interpret it as a sign of congestion, but data still flowing.

As already seen in **2.2.2.1**, slow start can do a lot of damage to our networks, but fast retransmit helps to recover from lost without hurting the network that much. It is triggered after a specified number of acknowledgments, usually is at the third, with the same acknowledge number are received by the sender. This gives makes the sender reasonably confident that the next higher sequence number was dropped and needs to be retransmitted; all this without the need for the action triggered by the timeout.

[v] since slow start adds one extra packet for each acknowledged

[vi] The timers that TCP has for each connection are: (1)Persist Timer: Ensures that window size information is transmitted even if no data is transmitted. (2)Keepalive Timer: Detects crashes on the other end of the connection. (3)2MSL Timer: Measures the time that a connection has been in the TIME_WAIT state. (4)Retransmission Timer: The timer is started during a transmission. A timeout causes a retransmission

Fast recovery manages the continuous transmitting of new data on subsequent duplicate acknowledgments, until the retransmitted packet is received. and after a non-duplicated acknowledge is received, the congestion window is reset back to the value it had before the fast retransmit was initiated, and the fast recovery mode is ended, going back to congestion avoidance mode.

2.3 Hidden Flaws and Bufferbloat

3 Active Queue Management

aqm section

3.1 BLUE

blue subsection

3.2 RED

red subsection

3.3 CoDel

codel subsection

4 Interaction with Buffers

5 Experimental Work

The goal of the experiments outlined in this section is to examine different residential and public networks with the objective to proof the existence of the phenomenon under an uncontrolled scenario. This will be done by running a set of tests with different tools, first to define and characterize the network, then to measure and compare how the network behave with and without load. More specifically, the factor to be tested is the latency under load and analyzed to identify if the latency that occurs is due to an excess of buffers or due to some other problem.

This section aims to explain the setup that will be used to perform the tests, the selected tools for each of these tests and what is to be achieved and expected from each one of these tests.

5.1 The Test Setup

The tests will be ran under a pseudo controlled environment, using one physical machine and a second virtual machine hosted in first machine. These systems runs under a regular OS without any modification beside the ones that the own OS as by default. Also, for some tests two other devices will be added, one acting as an Iperf Server and a regular Android Tablet that will be used to add some extra load to the network when the Ethernet cable is used as medium.

All of these tests will be carried out in a real-world scenario, where no packet prioritization is done by the server against our flows, the routes can vary between each iteration of the same test, and many different flows will collide with other flows from different sizes and types. Nor there is more information about how the flows are treated by the queue manager algorithms or about how they are configured.

5.1.1 Hardware Characterization

Physical Machine :

The physical machine runs as host OS, Windows 7 SP1, that works with an Intel(R) Core(TM) i7-2670QM CPU 2.20GHz with 8GB of available RAM. This machine will always be connected through its wireless adapter, a *Broadcom Corp. BCM4313 802.11b/g/n Wireless LAN Controller (rev 01)*. The Ethernet controller is a *Realtek Semiconductor Co., Ltd. RTL8111/8168 PCI Express Gigabit Ethernet controller (rev 06)* adapter, and will be bridged to the virtual machine for some tests.

Virtual Machine :

The virtual machine is hosted using VMWare Player 6.0.1, with 4 processors assigned for use plus 4GB of RAM, and with the Ethernet adapter connected only for certain tests. The OS selected is a Debian based OS called Kali Linux, and using the kernel release identified as Debian 3.12.6-2kali.

The wireless adapter is a AIR-802 USB adapter with Zydas chipset and a TP-link 8dbi antenna. This adapter is directly connected to the virtual machine and hooked to the physical machine without the USB extension, this way any extra signal loss is avoided.

Iperf Server :

This machine will be used as a server for the Iperf test. This is a VPS hosted by Digital Ocean [vii] with 512MB Ram, 20GB SSD Disk, and located in New York data center. This machine runs Ubuntu 12.04.3 x64 under KVM software using as a processor an Intel Hex-Core 3 GHz.

[vii] `https://digitalocean.com`

The idea of using an external device to connect the virtual machine to the network and not using a bridged configuration provided by software, is mainly because with an USB device the machine will take care of all the management and administration of the device, avoiding any possibility that the host machine modify or manages any flow. Also, by using a second machine to overload the uplink, causes to avoid overflow the queue on the machine that is performing the tests. With this, it is expected to minimize the possibility that our testing machine is causing extra latency, either by the saturation of the wireless channel, or by the queue in the traffic control subsystem into the kernel.

For the tests, the Bufferbloat community[28] has created a set of best practices[21] to follow so the results are consistent and repeatable, but in this case, computers and routers will not be modified as it will attempt to analyze what an every-day-user experience. Those practices will be taken in consideration if it is the QOS present in routers and deactivate it.

5.2 Tools Definition

The tools used for the benchmark were selected by the capability to determinate the presence of the Bufferbloat phenomenon in the network of study by the analysis of its indicator, the latency or the round trip time[viii]. So, with this as a main consideration, the tools selected for the benchmark will be chosen by the complexity and accuracy to measure and determine the RTT into an IP/TCP connection, and how hard is to consistently replicate the results under similar contexts. Other tools were selected to help determine, characterize, and proof the capability of the network to cause Bufferbloat.

[viii] To SUBTEL, organization responsible for control and supervision in the performance of telecommunications in Chile, RTT and Latency correspond to the same thing [18]

The tools selected to be used in this tests are the following:

- Speedtest test by Ookla
- Netalyzr by ICSI
- Iperf Tool with Tcptrace/Xplot.org
- Page Benchmarker extension for Google Chrome
- Smokeping Latency Tool

5.2.1 Speedtest

Developed by Ookla[ix], this tool is used for most of the ISPs and many users in Chile to test their broadband's connections globally. It can be used not only in their website *www.speedtest.net* but also in Android, iOS or Windows Phone. A command line interface developed in Python can be used too for testing Internet bandwidth.

The server that will be selected to perform every test, either has or not the fastest ping, is the one hosted by the Pontificia Universidad Católica de Valparaíso (this host is the default selected most of times by the site also).

5.2.2 Netalyzer

For end-users, little is revealed about how ISPs manage their networks. That is why since 2009 ICSI has developed Netalyzr, a "two click " tool developed to test networks that runs in web browser as a Java applet. Once downloaded, the applet contacts the back-end server using a range of protocols and mechanisms employed as part of the testing, and then conducts a series of tests. Once completed, it uploads its findings to the back-end where they are distilled into a detailed report breaking down the findings into correctly operating aspects, those that show potential signs of trouble, and those

[ix] `https://www.ookla.com/about`

that are downright broken. Figure **??** is an example of the output.

The primary goal in developing Netalyzr's tests was to provide a new kind of diagnostic tool, *one that particularly illuminates under what sort of restrictions a user's Internet connection operates, like both forms of filtering (blocking) and proxying imposed by the user's ISP, and performance issues that arise from the nature of the user's Internet access setup*[12]. Among the performance considerations, Netalyzr's measures are packet loss, latency, bandwidth. Also, it compute in-path forwarding device buffer size by comparing small-packet latencies under idle and loaded states in networks (the perfect time to occur Bufferbloat). Other tests like general TCP and UDP service reachability.

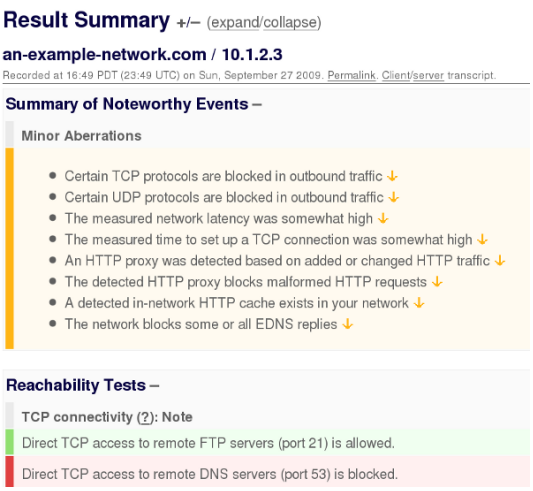

Figure 4: Result Summary example from Netalyzr. Source: `www.dslreports.com`

5.2.3 Iperf

Iperf is a well known and commonly used network testing tool. It can create a TCP and UDP data streams and measure the bandwidth and the quality of a network link. It can perform multiple tests like Latency, Jitter or Datagram Loss.

Iperf basically tries to send as much information down a connection as quickly as possible reporting on the throughput achieved. This tool is especially useful in determining the volume of data that links between two machines can supply. This two machines define the network, one acting like a server and the second as the client. For

this scenario, the server will be the VPS that only will receive the Iperf connections (also will be running ssh but without further interaction). The VM Linux machine will work as an Iperf Client.

As mentioned in Iperf users mailing list " *When one runs TCP tests, there are 2 things that block Iperf from having clear view of real throughput: buffering on sender's side (TCP/IP stack) and TCP behavior itself (acking). What Iperf can measure is the pace with which it sends data to TCP/IP stack; TCP/IP stack will only accept data from application when buffers are not full. If the buffer is huge, Iperf will see high throughput initially, then it will drop. If there's congestion or retrasmission going on, Iperf will see it as lower throughput*"[11], but the data generated by Iperf won't be further analyzed because the idea behind using this tool is a TCP's packet generator. This means that the packets generated by Iperf will captured and analyzed with tcpdump, tcptrace and xplot.org.

5.2.4 Page Benchmarker

Page Benchmarker is a Google Chrome extension that intent to test page load time performance within Chrome. Measures time-to-first-paint, overall page load time KB read/written, and several other metrics, and with its capability to clear the cache and existing connections between each page load, makes this tool one of the main sources of income to analysis.

5.2.5 Smokeping

SmokePing is a latency logging and graphing tool that consists of a running daemon which organizes the latency measurements and a CGI which presents the graphs. SmokePing give us the ability to measure latency and packet loss in the current network,

and with RRDtool, is capable to maintain a long term data store and to draw different graphs with the giving up-to-the-minute information on the state of each network connection.

Smokeping can be configured to perform a wide range of latency measurement probes each one directed to an independent target or over a set of targets selected for each proof.

5.3 Test Description

To test the existence of the Bufferbloat phenomenon, five tests are conducted as described below. Each test will be repeated under the following contexts:

1. Twice on the same day in one network to determine if does exists a considerable variance in latency for different times in a day.
2. Select different public and private networks with different *"speeds"*.
3. Use the Ethernet cable in order to compare the results with those previously obtained using Wireless.

Because it is intended to prove the existence of the phenomenon under circumstances experienced by everyday users, no kernel parameters will be amended nor changed, and only the ability to perform QOS on routers that have this feature is disabled. Furthermore, the overall question these tests seek to answer is the following:

Theorem. *"The networks that we use every day, have the necessary characteristics to generate the Bufferbloat phenomenon whether under low loads and if does exists, the how serious are the effects ?"*

5.3.1 Speed test

The idea under this test, is to set a baseline by comparing the speed offered by the ISP and the one at the moment of testing. To find the speed provided, the tool used is the Speedtest in the web site of Ookla.

The benefits of using this web sites is that not only it will reveal not only the available the national uplink and downlink, also it will set the baseline for the ping. The expected pings, independently of the connection bandwidth, should be around $\sim 12ms$, based on the data presented in [31] and [27] by the two most used ISP in Chile[x].

As data, it is estimated that the ratio of the average speed as compared to the rated speed of the service offered for domestic bonds has a variation of 85%, while for international links decreases drastically to 35%[19]. Due this factor, none of the tests will be performed on servers located outside our country, as it is believed that the data will not very representative regarding actual service.

5.3.2 Signs of trouble

After characterize our network based on the speed, it is needed to try to define the state of the service based on the quality in which our network operates, in example: what kind of traffic is passing through our network, and give also get a little more detail on the average rates of delay and buffers with which the traffic can find along the path. To collect all this information will be used Netalyzr.

While laws present in Chile assure consumers networks free of traffic shaping barriers and filters, ISPs apply traffic management measures[30][26] to deliver an optimum experience of the service's use and thus, achieve an efficient use of the network, thus

[x]Telefónica has a 39.2% market share while VTR owns 38.8% respectively. [20]

further to protect the safety of users while maintaining network stability.

This is why it is expected that some ports or services are partially or completely blocked, but the most important is that this test will first light on the existence of the phenomenon under study.

5.3.3 Collapse test

Already having a clear idea of the stat of the studied network, the next will be try to check empirically that the results obtained by the previous test above described are valid. For this, Iperf will be used to generate as much traffic as possible for 5 minutes between servers (both national and international). But the main goal is not to measure the throughput of the network; instead to capture the packets that Iperf produces and study them. So before run Iperf, tcpdump will capture this information. Subsequently, the tcp's trace is taken and extracted with tcptrace tool and generate the RTT graph.

This exercise will be conducted three times as part of the test, running the first time as the sole source of network load. For the second and third time, after 50 seconds after Iperf was started, from the Windows machine will be performed an upload to a Dropbox account with a file big enough to the upload take more than the time defined to this test. This upload intends to saturate the upstream link, leading to an overload of existing buffer (indistinct whether the server is national or not, there are several levels where the route is common). The upload will be stopped 50 seconds before the time limit.

The time gap established before loading and the end of the test is to let Iperf Iperf frames reach a steady state flow and thus to generate a basis on which to analyze the time when the traffic is maximum.

The expected RTT are around $\sim 12ms$ and $\sim 100ms$ without load, for national and international servers respectively. About the expected value under load, it is hard to estimated, further that their expected behavior is to be stable round a certain value. In case of no stable behavior, the possible causes will be analyzed and tracked.

5.3.4 Load benchmark test

With the effects of excess buffers already determined, this tests seeks to show how this phenomenon affects users who surfs through a web browser. This is why the extension of Google Chrome Page Benchmarker comes in and it will be used to do 5 iterations of 10 loads to a website and analyze the behavior of these.

As in the previous experiment, the first will be without any load on the network. Then, again a file will be uploaded to Dropbox from the second machine and after 30 seconds from the start of the load, and with the load active, will proceed to a second iteration. The third will begin the after a minute, canceling the file upload and after waiting 30 seconds after cancel the upload, the benchmark will be measured again. The fourth and fifth will be a minute and a half since the previous iteration have finished. The site chosen is `http://www.usm.cl`.

As like Jim Gettys showed in his video demonstration[5], the benchmark increases between 10 to 15 times with load against the original times. For this test, the expected proportions are lower, this mainly because the kernel machine that will run the tests has implemented already some improvements to mitigate this problem.

5.3.5 Smoke the path

To verify that the effect of Bufferbloat, if it is the case, does not just happen on a single server or with a single TCP flow, the Smokeping tool will be configured to perform two types of tests: Fping and echopinghttp. These tests will be directed to different types of servers (ie: physical and VPS machines) which have different connection settings and/or type of services (some servers has dedicated link), and located in both Chile and the United States.

The tool will be left couple of minutes to track and save the state of the connection under no load(except the required for testing). After that, a upload of a big file will be performed from a second machine until it finish or the behavior has become stable and then re-enter to a phase without load for couple of more minutes and repeat the upload.

With this test, in addition to verifying the validity of the previously captured data is expected to determine the presence of other factors that can contribute to the degradation of the quality of service on the network, whether factors such as quality of service provided by the host, channel/path problems, problems related to the way handling packets by the router or the machine, and/or any other that may arise.

6 Results

Here goes a description of what will be shown in this section

6.1 Speed test

Here goes the information related to test speed getting the differences and results gathered by this tests.

Will be a table with the current speed and pings. Also will be difference table with the speed that must has.

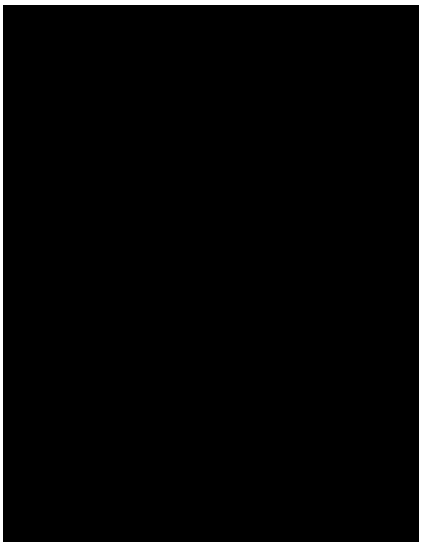

Figure 5: Speed and Pings based on Appendix B, Table 4

6.2 Netalyzr test

Here will be a resume of what netalyrz give us

6.3 Iperf test

Graphs and what happend with the rtts

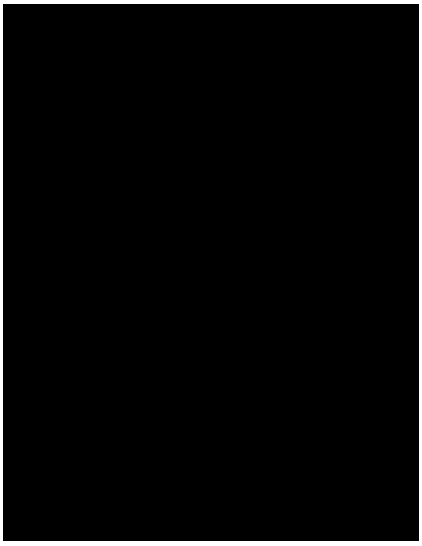

Figure 6: Variation ratio of offered vs measured speed based on Appendix B, Table 5

6.4 Benchmark test

Page benchmark results with tables with differences and mean results

Table 1: Total load mean times.

Lugar	1	2	3	4	5
4low	3415,1	20932,3	4490,5	4170,1	4087,1
casa	1654,8	3053,6	1718	1877,4	1666,8
mbahamon	2248,4	23868,5	2248	2182,6	2223,8
polmos	1623,4	3701,4	2420,6	1947	1659,3
polmos2	1912,4	4861,8	1738,5	2136,1	1827,5
nalucem	6421,4	26048,6	6280,8	6335,8	6306,4
casa2	1682,9	2354,4	1682,3	1667,5	1646,6
nalucem2	8199	32151,5	6479,1	6537,1	6349,4

6.5 Smokeping Test

Graph results

Table 2: Variation ratio over own iteration.

Lugar	1	2	3	4	5
4low	55,55	607,623	96,647	132,874	3202,323
casa	8,353	199,264	16,087	256,497	18,852
mbahamon	63,959	83,863	53,778	39,558	219,521
polmos	6,365	163,106	3802,472	8343,236	4050,064
polmos2	59,754	1891,192	8106,53	317,45	8126,291
nalucem	161,954	154,734	9,858	6,96	5,233
casa2	13,998	181,652	27,18	3,102	17,715
nalucem2	880,662	215,955	154,129	184,86	10,703

Table 3: Variation ratio over all iterations.

Lugar	min	max	%
4low	3415,1	20932,3	612,933
casa	1654,8	3053,6	184,529
mbahamon	2182,6	23868,5	1093,581
polmos	1623,4	3701,4	228,002
polmos2	1738,5	4861,8	279,654
nalucem	6280,8	26048,6	414,733
casa2	1646,6	2354,4	142,985
nalucem2	6349,4	32151,5	506,37

6.6 Summarising Results

all test

7 Conclusions

conclusions

http://lalith.in/2012/02/15/fun-with-tcp-cubic/ Examples of iperf and wireless

8 Further Work

further

analysis in specific of how the three different flows (Elephant, mice and ant)[7] [3] reacts to the bufferbloat

How the linux kernel now reacts to bufferbloat. [9]

The effects under laying the use of CTCP[24][25] in Windows and the recommendations of the Bufferbloat community[22]

ver notas en [14]

cerowrt project [23]

A Definitions

Bandwidth Delay Product (BDP) :

An amount of data measured in bits. It is equivalent to the maximum amount of data on the network circuit at any given time. Commonly measured by the RTT*bandwidth.

Bottleneck bandwidth :

The smallest bandwidth along the path. Packets cannot arrive at the destination any faster than the time it takes to send a packet at the bottleneck rate.

Bufferbloat :

Refers to excess buffering inside a network, resulting in high latency and reduced throughput.

Congestion Window (cwin) :

One of the factors that determines the number of bytes that can be outstanding at any time. It prevents the overload of the link with too much traffic. The size is calculated by estimating how much congestion there is between the two places.

Long Fat Network (LFN) :

A network with a large bandwidth- delay product. Often $>> 10^5$ bits (12500 bits).

Maximum segment size (mss) :

Largest amount of data (in octets) that a communication device can receive in a single TCP segment (single IP datagram). Established by pass on the syn packet.

Slow start threshold (ssthrsh) :

It determine whether the TCP should do Slow Start or Congestion Avoidance. It is initialized to a large value, and after a congestion is signaled, cwin is divided in half and ssthrsh is set to cwin.

System Throughput :

Corresponds to the fastest rate at which the count of packets transmitted to the destination by the network is equal to the number of packets sent into the network.

Throughput :

Measure of the efficiency of a network expressed as the data(bits or packets) transfer rate of useful and non-redundant information. It depends on factors such as bandwidth, line congestion, error correction, etc.

B Speed Test results

Table 4: Speeds and Pings measured.

Location	Medium	Downlink (Mbps)	Uplink (Mbps)	Ping (ms)
4low	Wireless	0,6	8,03	29
mbahamon	Wireless	0,6	10,21	24
mhbrisas	Wireless	1,19	17,96	19
mgallard	Wireless	x	x	x
garriag	Wireless	x	x	x
casa	Wireless	4,95	15,48	18
casa2	Wireless	5,09	13,46	20
polmos	Wireless	2,32	19,61	26
polmos2	Wireless	2,34	18,72	22
nalucem	Wireless	0,57	3,97	35
nalucem2	Wireless	0,56	4,06	36
mcatala	Wireless	x	x	x
mcatala2	Wireless	x	x	x

Table 5: Variation ratio of offered vs measured speed.

Location	Uplink ratio	Downlink ratio
4low	120,00	80,30
mbahamon	120,00	102,10
mhbrisas	119,00	89,80
mgallard	0,00	0,00
garriag	0,00	0,00
casa	99,00	103,20
casa2	101,80	89,73
polmos	116,00	49,03
polmos2	117,00	46,80
nalucem	114,00	99,25
nalucem2	112,00	101,50

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